Polycom™

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Polycom® RealPresence® Collaboration Server, Virtual Edition Administrator’s Guide

Trademark Information

Patent Information

This software has not achieved UC APL certification.

This document provides the latest information for security-conscious users running Version 8.1 software. The information in this document is not intended to imply that DoD or DISA certifies Polycom RealPresence Collaboration Server systems.

End User License Agreement

Use of this software constitutes acceptance of the terms and conditions of the Polycom® RealPresence® Collaboration Server 800s, Virtual Edition system end-user license agreements (EULA).

The EULA for your version is available on the Polycom Support page for the Polycom® RealPresence® Collaboration Server 800s, Virtual Edition system.

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RealPresence Collaboration Server Virtual Edition Overview

About the RealPresence Collaboration Server Virtual Edition Administrator's Guide

The *Polycom® RealPresence Collaboration ServerVirtual Edition Administrator's Guide* provides instructions for configuring, deploying, and administering Polycom Multipoint Control Units (MCUs) for video conferencing. This guide will help you understand the Polycom video conferencing components, and provides descriptions of all available conferencing features. This guide will help you perform the following tasks:

- Optional. Customize the Collaboration Server conferencing entities such as conference Profiles, IVR Services, Meeting Rooms, Entry Queues, etc., to your organization's needs. In the CloudAxis solution environment, these entities should be defined in the Polycom RealPresence Distributed Media Application (DMA) system.
- Define Collaboration Server Users.
- Advanced conference Management
- Define Video Protocols and Resolution Configuration for CP Conferencing
- Optional. Configure Templates and the Address Book. In the CloudAxis solution environment, these entities should be defined in the RealPresence DMA system.
- Record Conferences
- Configure the Collaboration Server to support special call flows and conferencing requirements, such as Cascading Conferences.
- Configure the Collaboration Server for special applications and needs by setting various system flags.
- Manage and troubleshoot the Collaboration Server's performance.

The *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide* provides description of basic conferencing operations. It will help you perform the following tasks:

- Perform basic configuration procedures.
- Start a new conference and connect participants/endpoints to it.
- Monitor ongoing conferences
- Perform basic operations and monitoring tasks

Who Should Read This Guide?

System administrators and network engineers should read this guide to learn how to properly set up Polycom Collaboration Server systems. This guide describes administration-level tasks.

For detailed description of first time installation and configuration, description of the *Collaboration Server Web Client*, and basic operation of your Collaboration Server system, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*.

Prerequisites

This guide assumes the user has the following knowledge:

- Familiarity with Windows® XP or Windows 7 operating systems and interface.
- Familiarity with Microsoft® Internet Explorer® Version 7, 8 or 9.
- Basic knowledge of video conferencing concepts and terminology.

How This Guide is Organized

The following typographic conventions are used in this guide to distinguish types of in-text information.

Table 1-1 *Typographic Conventions*

Convention	Description
Bold	Highlights interface items such as menus, soft keys, flag names, and directories. Also used to represent menu selections and text entry to the phone.
Italics	Used to emphasize text, to show example values or inputs, file names and to show titles of reference documents available from the Polycom Support Web site and other reference sites.
Underlined Blue	Used for URL links to external Web pages or documents. If you click on text in this style, you will be linked to an external document or Web page.
Blue Text	Used for cross referenced page numbers in the same or other chapters or documents. If you click on blue text, you will be taken to the referenced section. Also used for cross references. If you click the italic cross reference text, you will be taken to the referenced section.
<variable name>	Indicates a variable for which you must enter information specific to your installation, endpoint, or network. For example, when you see <IP address>, enter the IP address of the described device.
>	Indicates that you need to select an item from a menu. For example, **Administration > System Information** indicates that you need to select **System Information** from the *Administration* menu.

About the Polycom® RealPresence Collaboration ServerVirtual Edition System

The *RealPresence Collaboration Server Virtual Edition* system is a high performance, scalable, IP-network (H.323 and SIP) MCU that provides feature-rich and easy-to-use multipoint voice and video conferencing.

The MCU can be used as a standalone device to run voice and video conferences or it can be used as part of a solution provided by Polycom. This solution may include the following components:

- *Polycom® RSS™ 4000* - provides one-touch recording and secure playback on video conferencing systems, tablets and smartphones, or from your Web browser.
- *Polycom® Distributed Media Application*™ (DMA™) system - provides call control and MCU virtualization with carrier-grade redundancy, resiliency and scalability.
- *Polycom Real Presence Resource Manager* - centrally manages, monitors and delivers Cloud based Video as a Service (VaaS) and enterprise video collaboration.
- *Polycom® RealPresence® Access Director™ (RPAD)* - removes communication barriers and enables internal and external teams to collaborate more easily and effectively over video.

The following diagram describes the multipoint video conferencing configuration with the Collaboration Server as a standalone system.

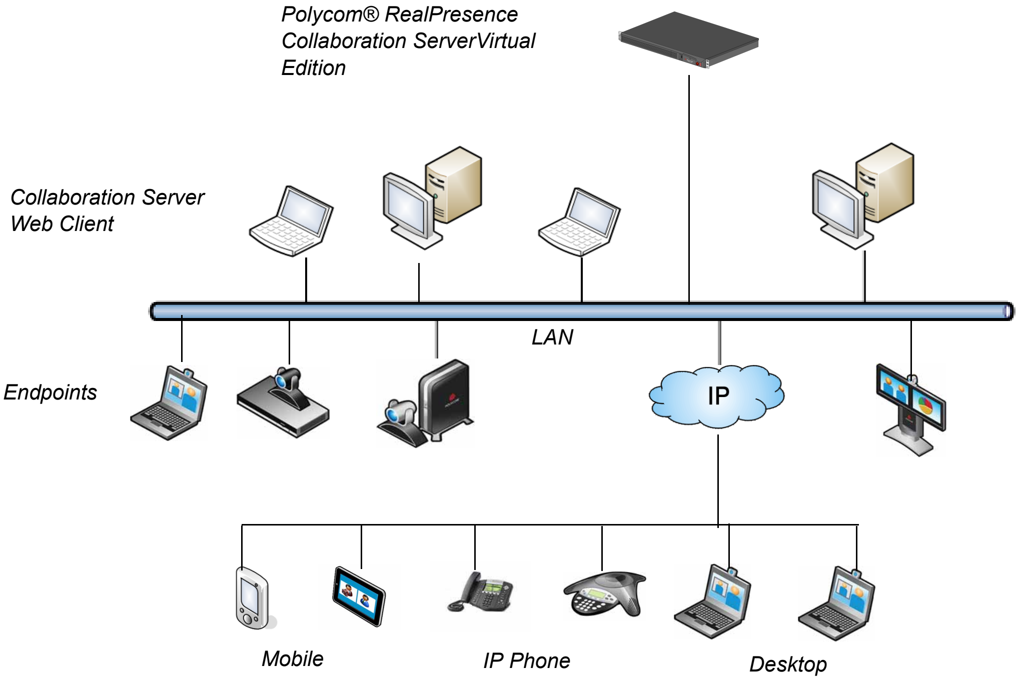

Polycom® RealPresence Collaboration ServerVirtual Edition
Collaboration Server Web Client
LAN
Endpoints
IP
Mobile
IP Phone
Desktop

Figure 1-1 *Multipoint Video Conferencing using a Polycom Collaboration Server*

The RealPresence Collaboration Server Virtual Edition unit can be controlled via the LAN by the *Collaboration Server Web Client* application using Internet Explorer® installed on the user's workstation, or the RMX Manager application. The RMX Manager can control several Collaboration Server units. For more information about the RMX Manager see "*RMX Manager Application"* on page **18-1**.

IP Networks

In the Polycom® RealPresence Collaboration ServerVirtual Edition, system management and IP conferencing are performed via a single LAN port.

Workstation Requirements

The *Collaboration Server Web Client* and *RMX Manager* applications can be installed in an environment that meets the following requirements:

- **Minimum Hardware** – Intel® Pentium® III, 1 GHz or higher, 1024 MB RAM, 500 MB free disk space.
- **Workstation Operating System** – Microsoft® Windows® XP, Windows® 7, and Windows® 8.
- **Network Card** – 10/100/1000 Mbps.
- **Web Browser** - Microsoft® Internet Explorer® Version 7, 8, 9, and 10.
- Collaboration Server Web client and RMX Manager are optimized for display at a resolution of 1280 x 800 pixels and a magnification of 100%.

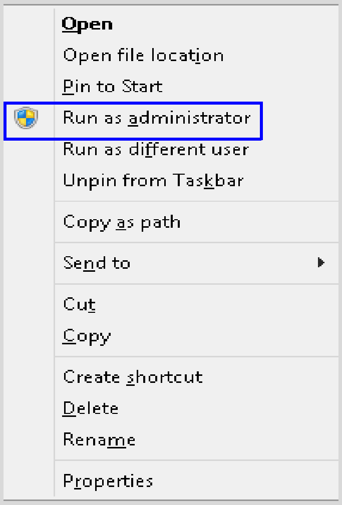
Internet Explorer must be enabled to allow running Signed ActiveX.
If ActiveX installation is blocked please see "*ActiveX Bypass"* on page **19-55**.

Collaboration Server Web Client does not support larger Windows text or font sizes. It is recommended to set the text size to 100% (default) or Normal in the Display settings in Windows Control Panel on all workstations. Otherwise, some dialog boxes might not appear properly aligned. To change the text size, select **Control Panel>Display**. For Windows XP, click the **Appearance** tab, select **Normal** for the Font size and click **OK**. For Windows 7, click the **Smaller - 100%** option and click **OK**.

When installing the *Collaboration Server Web Client*, Windows Explorer **>Internet Options> Security Settings** must be set to *Medium* or less.

It is not recommended to run *Collaboration Server Web Client* and *Polycom CMAD* applications simultaneously on the same workstation.

If you have problems getting the Collaboration Server Web Client to work with Windows 8, it is recommended to run Internet Explorer as an administrator by holding the shift key and right-clicking on the IE icon, and then select Run as Administrator.



For Windows 7™ Security Settings, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Microsoft Windows 7™ Security Settings"* on page **1-12**.

For Internet Explorer 8 configuration, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Microsoft Windows 7™ Security Settings"* on page **1-12**.

2

Conference Profiles

In the *RealPresence CloudAxis Solution*, the Conference Profiles are defined in the RealPresence DMA system component and should not be defined directly in the RealPresence Collaboration Server Virtual Edition component.

Profiles stored on the Collaboration Server enable you to define all types of conferences. Profiles include conference parameters such as Conferencing Mode, Conference Session Type, Conference Line Rate, video and Video Layout, Encryption, Lost Packet Recovery (LPR), etc.

The maximum number of *Conference Profiles* that can be defined is 80.

Conference Profiles are assigned to Conferences, Meeting Rooms, and Entry Queues. The same *Profile* can be assigned to different conferencing entities. When modifying the *Profile* parameters, the changes will be applied to all the conferencing entities to which the profile is assigned.

Conference Profile options differ according to the selected *Conferencing Mode* and *Conference Type*. Profiles can be defined for AVC (Advanced Video Codec) CP conferencing Mode or SVC (Scalable Video Codec) conferencing Mode. AVC Conferencing Mode, offers two Video Session types: Continuous Presence (CP) conferences and Video Switching (VSW) conferences, and a special functional conference - Operator Conferences.

Conference Profiles can be saved to *Conference Templates* along with all participant parameters, including their *Personal Layout* and *Video Forcing* settings. It enables administrators and operators to create, save, schedule and activate identical conferences quickly and easily. For more information see Chapter 10, *"Using Conference Templates"* .

Conferencing Modes

The MCU system offers the following Conferencing Modes:

- Transcoding - AVC Conferencing
- Media Relay - SVC Conferencing
- Mixed Mode - Mixed AVC (CP) and SVC Conferencing

CP Transcoding - AVC-based Conferencing

A transcoded CP (Continuous Presence) conference is also described as an AVC (Advanced Video Coding) conference. It supports the standard video protocols. In this mode, video is received from all the endpoints using different line rates, different protocols (SIP, H.323) and video parameters:

- Video protocols: H.263, H.264 Base and High profile and RTV
- Video Resolutions: from QCIF, CIF and up to 720p
- Frame rates up to 30 fps

All endpoints that do not support the H.264 SVC protocol such as H.263, H.264, or RTV, are considered AVC endpoints.

The MCU processes the received video, transcodes it and sends the resulting video streams to the endpoints. The video processing that is required differs according to the video session set for the conference, with all the processing performed by the MCU. For more details, see "*AVC Conferencing - Video Session Types"* on page **2-4**.

Media Relay - SVC Conferencing

Media Relay SVC Conferencing is based on the SVC (Scalable Video Codec) video protocol and SAC audio protocol. It offers high resolution video conferencing with low end-to-end latency, improved Error Resiliency and higher system capacities.

The Polycom multipoint media server, serves as an integrated media relay engine that provides media streams for displaying conferences at low latency video experience in video conferences. For more details see, "*SVC-based Conferencing"* on page **2-9**.

Mixed CP and SVC Conferencing

This type of conference enables participants with SVC-enabled endpoints and AVC endpoints to participate in the same conference.

Each endpoint connects according to its capabilities. The MCU processes the AVC video streams and converts them into SVC video Streams and relays them to the SVC participants that constructs the video layout on the endpoint.

In the same way, the MCU processes the video streams received from the SVC participants, converts them into AVC video and then transcodes all the video streams to compose the video layout that is sent to the AVC endpoints.

Conferencing Capabilities in the Various Conferencing Modes

The following table summarizes the conferencing capabilities and options available in the different Conferencing Modes.

Table 2-1 *Conferencing Capabilities in the Different Conferencing Modes*

Feature	CP Only	Mixed CP & SVC	SVC Only
Operator Conferences	✓	✗	✗
Entry Queues	✓*	✓*	✓*
Permanent Conference	✓	✓	✓
Cascading	✓**	✓**	✗
IVR	✓	✓	✓ Reduced IVR set for SVC endpoints
Dial Out	✓	✗	✗

Table 2-1 Conferencing Capabilities in the Different Conferencing Modes (Continued)

Feature	CP Only	Mixed CP & SVC	SVC Only
Auto Redial	✓	✓	✗
LPR	✓	✓***	✓***
Content	✓ All Content Settings, All Content Protocols	✓ Graphics Only, H.264 Cascade & SVC Optimized	✓ Graphics Only, H.264 Cascade & SVC Optimized
Presentation Mode	✓	✗	✗
Lecture Mode	✓	✗	✗
Same Layout	✓	✓	✗
Layout Selection	✓	✓ AVC endpoints only	Layout set to Auto Layout and defined on the endpoint
Skins	✓	✓ AVC endpoints only	✗
Encryption	✓	✓	✓
Recording	✓	✓ AVC recording only	✗
Site Names	✓	✓ AVC endpoints only	Managed by the endpoint (not via MCU)

* Entry Queue & Destination Conference must have the same profile (i.e. SVC only to SVC only, Mixed CP and SVC to Mixed CP and SVC)

** Only Basic Cascading is available

*** For AVC, the LPR error resiliency is used, however for SVC endpoints, new error resiliency methods are used.

AVC Conferencing - Video Session Types

All endpoints have AVC capabilities and can connect to AVC conferences running on the MCU. AVC-based Endpoints can connect using different signaling protocols and different video protocols.

Continuous Presence (CP) Conferencing

The dynamic Continuous Presence (CP) capability of the Collaboration Server system enables viewing flexibility by offering multiple viewing options and window layouts for video conferencing.

Endpoints can connect to the conference using any signaling protocol (H.323, SIP and RTV), line rate (up to a maximum line rate defined for the conference), Video Protocol (H.261, H.263, H.264 Base and High Profile) and at any resolution and frame rate (provided they meet the minimum requirements set for the conference).

In Continuous Presence conferences, the MCU receives the video stream from each endpoint at the video rate, video resolution and frame rate that it is capable of sending, and it superimposes all the received streams into one video stream that includes the input from the other endpoints arranges in the selected video layout.

Participants do not see themselves in the video layout. By Default, the speaker is shown in the top left layout cell in symmetric layouts, in the larger cell in asymmetric layouts, or in full screen. The speaker sees the previous speakers (their number depends on the number of cells on the speaker’s layout.

The Continuous Presence video session offers layouts to accommodate different numbers of participants and conference settings including support of the VUI annex to the H.264 protocol for endpoints that transmit wide video instead of 4CIF resolution. Each participant can select his/her layout for viewing during the conference, as can be seen in **Figure 2-1**.

For conferences with more participants than display squares, the Collaboration Server dynamic video mix capability allows the viewed sites to be modified throughout the conference. The displayed layout can be changed during an ongoing conference, allowing a participant to view different screen layouts of the other conference participants. These layout options allow conferences to have greater flexibility when displaying a large number of participants and maximizes the screen’s effectiveness.

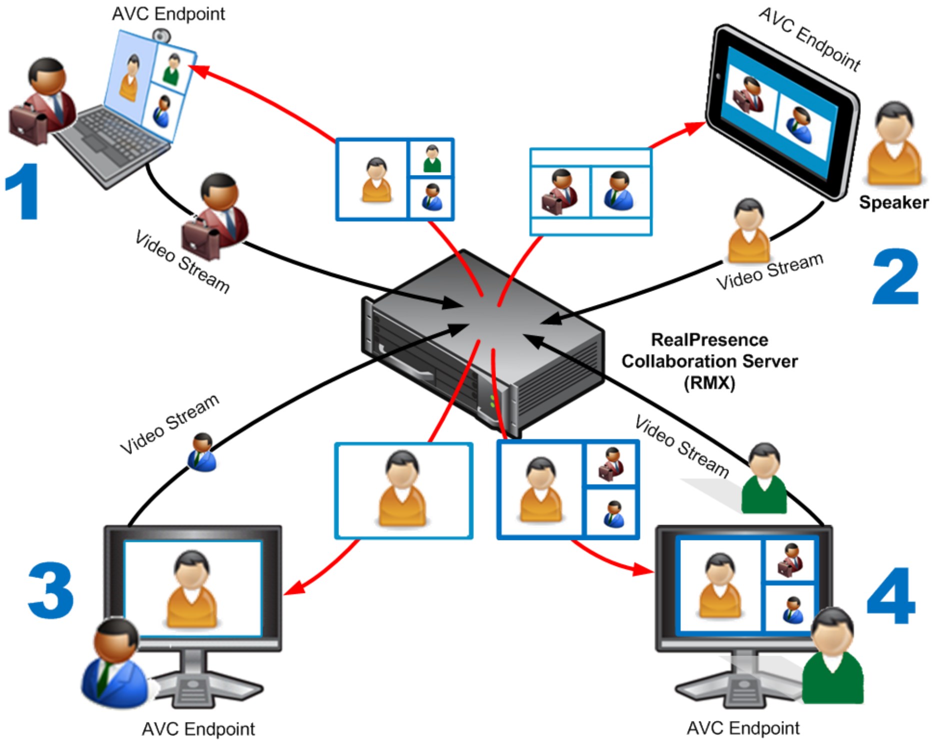


Figure 2-1 AVC Continuous Presence (CP) video streams and built layouts

Video quality in Continuous Presence conferences is affected by the conference line rate (that determines the maximum line rate to be used by the connecting endpoints), and the video capabilities of the endpoints such as the video protocol, video resolution and frame rate. Content sharing is available in all CP conferences.

This requires extensive processing of the video sent to each participant in the conference. The higher the video rate and resolution, the more processing power is required.

By default every conference, Entry Queue and Meeting Room has the ability to declare the maximum CP resolution as defined for the system. This includes conferences launched by the *Collaboration Server Web Client* and conferences started via the API.

CP conferencing is defined in the Conference profile by setting the following main features:

- Setting the *Conferencing Mode* to **AVC only**
- Conference Line Rate
- Video Layout

Video Protocol Support in CP Conferences

The video protocol selected by the system determines the video compression standard used by the endpoints. In Continuous Presence conferences, the system selects the best video protocol for each of the endpoint according to he endpoint's capabilities.

The following Video protocols are supported in CP conferences:

- **H.261 -** the legacy video compression algorithm mandatory to all endpoints. It is used by endpoints that do not support other protocols.
- **H.263 -** a video compression algorithm that provides a better video quality than H.261. This standard is not supported by all endpoints.

- **H.264** Base Profile **-** a video compression standard that offers improved video quality, especially at line rates lower than 384 Kbps.

  H.264 High Profile allows higher quality video to be transmitted at lower line rates.
- **RTV -** a video protocol that provides high quality video conferencing capability to *Microsoft OCS (Office Communicator Server)* endpoints at resolutions up to *HD720p30*. (SIP only).

AVC Conferencing Parameters

When defining a new video Profile, you select the parameters that determine the video display on the participant's endpoint and the quality of the video. When defining a new conference Profile, the system uses default values for Continuous Presence (CP) standard conferencing. Continuous Presence conferencing enables several participants to be viewed simultaneously and each connected endpoint uses its highest video, audio and data capabilities up to the maximum line rate set for the conference.

Basic Conferencing Parameters

The main parameters that define the quality of a video conference are:

- **Line (Bit) Rate -** The transfer rate of video and audio streams. The higher the line (bit) rate, the better the video quality.
- **Audio Algorithm -** The audio compression algorithm determines the quality of the conference audio.
- **Video protocol, video format, frame rate, annexes, and interlaced video mode** - These parameters define the quality of the video images. The Collaboration Server will send video at the best possible resolution supported by endpoints regardless of the resolution received from the endpoints.
  - When *Sharpness* is selected as the *Video Quality* setting in the *Conference Profile*, the Collaboration Server will send 4CIF (H.263) at 15fps instead of CIF (H.264) at 30fps.
  - *H.264 High Profile* protocol provides better compression of video images in line rates lower than 384 Kbps and it will be automatically selected for the endpoint if it supports H.264 *High Profile*. If the endpoint does not support H.264 High Profile, the Collaboration Server will try H.264 Base Profile which provides good compression of video images in line rates lower than 384 Kbps (better than H.263 and not as good as H.264 *High Profile).*
  - When working with Collaboration Servers at low bit rates (128, 256, or 384Kbps), HDX endpoints will transmit SD15 resolution instead of 2CIF resolution.

  When using a full screen (1x1) conference layout, the Collaboration Server transmits the same resolution it receives from the endpoint.
- Supported resolutions:
  - **H.261 CIF/QCIF –** Is supported in Continuous Presence (CP) conferences at resolutions of 288 x 352 pixels (CIF) and 144 x 176 pixels (QCIF). Both resolutions are supported at frame rates of up to 30 frames per second.
  - **H.263 4CIF** - A high video resolution available to H.263 endpoints that do not support H.264. It is only supported for conferences in which the video quality is set to sharpness and for lines rates of 384kbps to 1920kbps.
  - **Standard Definition (SD)** - A high quality video protocol which uses the H.264 and H.264 High Profile video algorithms. It enables compliant endpoints to connect to Continuous Presence conferences at resolutions of 720 x 576 pixels for

PAL systems and 720 x 480 pixels for NTSC systems. For more information, see "*Video Resolutions in AVC-based CP Conferencing"* on page **4-1**.

- — **High Definition (HD)** – HD is an ultra-high quality video resolution that uses the H.264 and H.264 High Profile video algorithms. Depending on the Collaboration Server's Card Configuration mode compliant endpoints are able to connect to conferences at the following resolutions:
  - **720p** (1280 x 720 pixels)

"*Video Resolutions in AVC-based CP Conferencing"* on page **4-1**.

- **Lost Packet Recovery (LPR) -** LPR creates additional packets that contain recovery information used to reconstruct packets that are lost during transmission.

Supplemental Conferencing Features

In addition to *basic parameters* that determine the quality of the video, additional features can be enabled, adding capabilities to the conference, or enabling special conferencing modes:

- **Content Sharing (H.239) –** Allows compliant endpoints to transmit and receive two simultaneous streams of conference data to enable Content sharing. H.239 is also supported in cascading conferences. Both H.263 and H.264 Content sharing protocols are supported. If all endpoints connected to the conference have H.264 capability, Content is shared using H.264, otherwise Content is shared using H.263.
  For more information, see **"***Content Sharing"* **on page 3-1.**
- **Encryption –** Used to enhance media security at conference and participant levels. For more information, see "*Media Encryption"* on page **3-26**.
- **Conference Recording -** The Collaboration Server enables audio and video recording of conferences using Polycom RSS recording system.
- **Lecture Mode –** The lecturer is seen by all participants in full screen while the lecturer views all conference participants in the selected video layout.
  For more information, see **"***Lecture Mode (AVC CP Only)"* **on page 3-38.**
- **Presentation Mode (CP Conferences only) –** When the current speaker's speech exceeds a predefined time (30 seconds), the conference layout automatically changes to full screen, displaying the current speaker as the conference lecturer on all the participants' endpoints. During this time the speaker's endpoint displays the previous conference layout. When another participant starts talking, the Presentation Mode is cancelled and the conference returns to its predefined video layout. Presentation mode is available with *Auto Layout* and *Same Layout*.
  - — If the speaker in a video conference is an Audio Only participant, the Presentation Mode is disabled for that participant.
  - — Video forcing works in the same way as in Lecture Mode when Presentation Mode is activated, that is, forcing is only enabled at the conference level, and it only applies to the video layout viewed by the lecturer.

Default Profile Settings in CP Conferencing Mode

The Collaboration Server is shipped with a default *Conference Profile* for CP conferences which allows users to immediately start standard ongoing CP conferences. These are also the default settings when creating a new Profile. The default settings are as follows:

Table 2-2 Default CP Only Conference Profile Settings

Setting	Value
Profile Name	Factory_Video_Profile
Line Rate	384Kbps
Operator Conference	Disabled
Encryption	Disabled
Packet Loss Compensation (LPR and DBA)	Enabled
Auto Terminate	• After last participant quits - Enabled • When last participant remains - Disabled
Auto Redialing	Disabled
Exclusive Content Mode	Disabled
Enable FECC	Enabled
Video Quality	Sharpness
Maximum Resolution	Auto
Content Settings	HiResGraphics (High Res Graphics)
Content Protocol	H.264 HD
Presentation Mode	Disabled
Same Layout	Disabled
Lecturer View Switching	Disabled
Auto Scan Interval	Disabled (10)
Auto Layout	Enabled
Mute participants except the lecturer	Disabled
Skin	Polycom
IVR Name	Conference IVR Service
Recording	Disabled
Site Names display	Disabled
Network Services - SIP Registration	Disabled
Network Services - Accept Calls	Enabled

SVC-based Conferencing

The SVC-Based conferencing mode provides video without transcoding by the MCU, hence requiring less video resources while providing better error resiliency and lower latency.

Using the SVC video protocol, SVC conferences provide video bit streams at different resolutions, frame rates and line rates to SVC-enabled endpoints with various display capabilities and layout configurations.

In the SVC-based conference, each SVC-enabled endpoint transmits multiple bit streams, called simulcasting, to the Polycom® RealPresence® Collaboration Server. Simulcasting enables each endpoint to transmit at different resolutions and frame rates such as 720p at 30fps, 15fps, and 7.5fps, 360p at 15fps and 7.5fps, and 180p at 7.5fps.

The Polycom SVC-enabled endpoints (such as Polycom® RealPresence® Desktop and Polycom® RealPresence® Mobile) compose the layout according to their layout settings and video capabilities. This enables the MCU to send or relay the selected video streams to each endpoint without processing the video streams and sending the composite video layout to the endpoints.

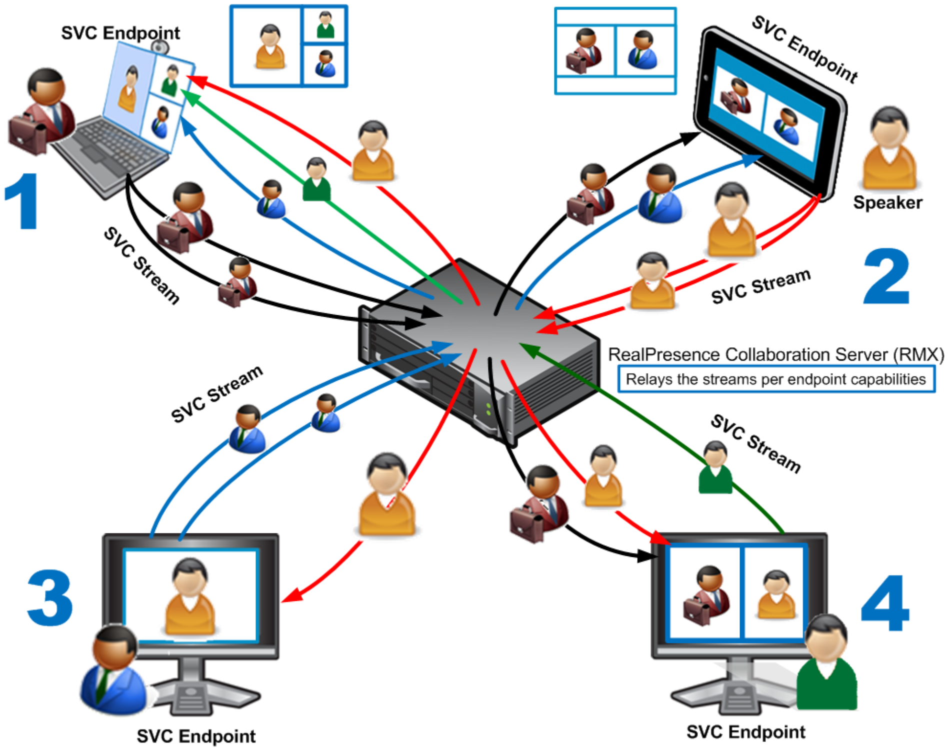


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Figure 2-2 SVC video streams and Layouts

The video streams displayed in the conference layout on each endpoint is obtained from the different streams received from each of the endpoints displayed in the layout. Depending on the size of the video cell in the configured layout, the endpoint requests the video stream in the required resolution from the RealPresence Collaboration Server. The higher the display quality and size, the higher the requested resolution will be sent to the endpoint. The endpoint creates the displayed layout from the different video streams it receives.

For instance, an SVC endpoint might want to receive three video streams at different frame rates and resolutions, and create a conference layout with the received video streams. Each SVC-enabled endpoint sends encoded SVC bit streams to the MCU to relay to the other SVC-enabled endpoints in the conference.

The endpoints encode the video in multiple resolutions and decodes the multiple video input streams.

For example:

RealPresence mobile client (**2**) will transmit two resolutions; one that is suited for RealPresence Desktop client (**3**) and a second that is suited for two other endpoints: RealPresence Desktop client (**4**) and (**1**).

RealPresence Desktop client (**1**) transmits two resolutions; one that is suited for RealPresence Mobile client (**2**) and a second that is suited for RealPresence Desktop client (**4**).

The MCU determines which of the incoming resolutions to send to each endpoint. It does not perform any SVC encoding and decoding, or any transcoding of the video streams. The RealPresence Collaboration Server functions as the multipoint media relay to the endpoints. For voice activated selection of the video streams, the RealPresence Collaboration Server determines which of the incoming bit streams to send to each endpoint.

Advantages of SVC Conferencing

SVC increases the scalability of video networks and enables mass desktop video deployments. Some of the advantages of SVC conferencing are:

- Offers high-resolution video conferencing with low end-to-end latency, improved error resiliency and higher system capacities.
- Allows the SVC-enabled video endpoints to manage display layouts, supporting multiple line rates, resolutions and frame rates.
- The RealPresence Collaboration Server functions as a media relay server providing low cost production benefits. The RealPresence Collaboration Server reduces bandwidth usage by only selecting the necessary video stream to be sent to the endpoints.

Guidelines

- SVC conferences are supported only with the following:
  - — SVC Licensing
  - — SIP over UDP signaling
  - — SIP over TLS Signaling
  - — Polycom SVC-enabled endpoints (Polycom® RealPresence® Desktop, Polycom® RealPresence® Mobile)
- SVC Only conferences can run on the same MCU as AVC Only conferences.
- End-to-end latency on a local network (same site), is around 200mSec to ensure AV sync (also known as Lip-sync).
- Dial-out is not available in SVC Only conference.
- Dial-in is available as follows:
  - — AVC endpoints (participants) can only connect to an AVC conference or Mixed CO and SVC conference. When dialing into SVC Only conferences they will be disconnected and the calls fail.

- — SVC endpoints support both AVC and SVC video protocols.
  When dialing into SVC Only conferences, they connect as SVC endpoints.
  When dialing into AVC Only conferences, they connect as AVC endpoints. They cannot connect to an AVC conference using the SVC capabilities.
- SVC endpoints can connect to conferences via Entry Queues, however:
  - — The Entry Queue and Conference Modes must match - both SVC Only or both Mixed.
  - — Both the Entry Queue and the Conference must have the same line rate.
- SVC endpoints cannot be moved between conferences.
- Content is supported in H.264 (AVC).
  - — Only the *H.264 Cascade and SVC Optimized* option is supported.
  - — LPR and DBA are not supported for SVC content sharing.
- In SVC Only conferences and Mixed CP and SVC conferences, Auto Layout is the default and the layout display for SVC endpoints is controlled from the endpoint application.
- Site names display on SVC endpoints is controlled from the SVC endpoints.
- When DMA is part of the solution, the DMA is used as the SIP proxy and the SVC endpoint subscribes to DMA for call control. If a DMA is not part of the solution, the SVC endpoint dial directly to the Collaboration Server using IP addresses is the SIP dialing strings.
- When Hot backup is enabled, all the conferences are created on the Slave MCU.
- When Hot Backup is activated and the Slave MCU becomes the Master MCU:
  - — All AVC endpoints will be reconnected to the AVC conferences. SVC endpoints connected to AVC conferences using their AVC capabilities will be reconnected to their AVC conferences.
  - — SVC endpoints cannot be reconnected to their SVC Only conferences as dial-out is not supported for SVC endpoints. These endpoints will have to manually reconnect to their SVC conferences.
- Cascading between SVC Only conferences or between AVC and SVC Only conferences is not supported.
- The following functionality and features are not supported during SVC Only conferences:
  - — FECC
  - — Skins. The video cells are displayed on the endpoint's default background.
  - — Conference Gathering phase
  - — Password protected conferences as DTMF input for passwords cannot be processed
  - — Manual selection of video layout
  - — Recording of SVC Only conferences
  - — Text messaging using Message Overlay

MCU Supported Resolutions for SVC Conferencing

The MCU automatically selects the resolution and frame rate according to the conference line rate. Table 2-3 details the maximum resolution and frame rates supported by the MCU for each conference line rate. The actual video rate, resolution and frame rates displayed on each endpoints is determined by the endpoint's capabilities.:

Table 2-3 SVC Conferencing - Maximum Supported Resolutions per Simulcast Stream

Conference Line Rate (kbps)	Profile	Maximum Resolution	Max. Frame Rate (fps)	Audio Rate (kbps)
1472 - 2048	High Profile	720p	30fps	48
1024 - 1472	High Profile	720p	15fps	48
768 - 1024	High Profile	720p	15fps	48
512 - 768	High Profile	360p	30fps	48
256 - 512	Base Profile	180p	15fps	48
192 - 256	Base Profile	180p	30fps	48
128 - 192	Base Profile	180p	15fps	48

Default Profile Settings in SVC Only Conferencing Mode

The Collaboration Server is shipped with a default *Conference Profile* for SVC Only conferences which allows users to immediately start standard ongoing SVC Only conferences. These are also the default settings when creating a new Profile. The default settings are as follows:

Table 2-4 Default SVC Only Conference Profile Settings

Setting	Value
Profile Name	Factory_SVC_Video_Profile
Line Rate	1920Kbps
Operator Conference	Not supported
Encryption	Disabled
Packet Loss Compensation (LPR and DBA)	Not supported
Auto Terminate	• After last participant quits - Enabled • When last participant remains - Disabled
Auto Redialing	Not supported
Exclusive Content Mode	Disabled

Table 2-4 Default SVC Only Conference Profile Settings (Continued)

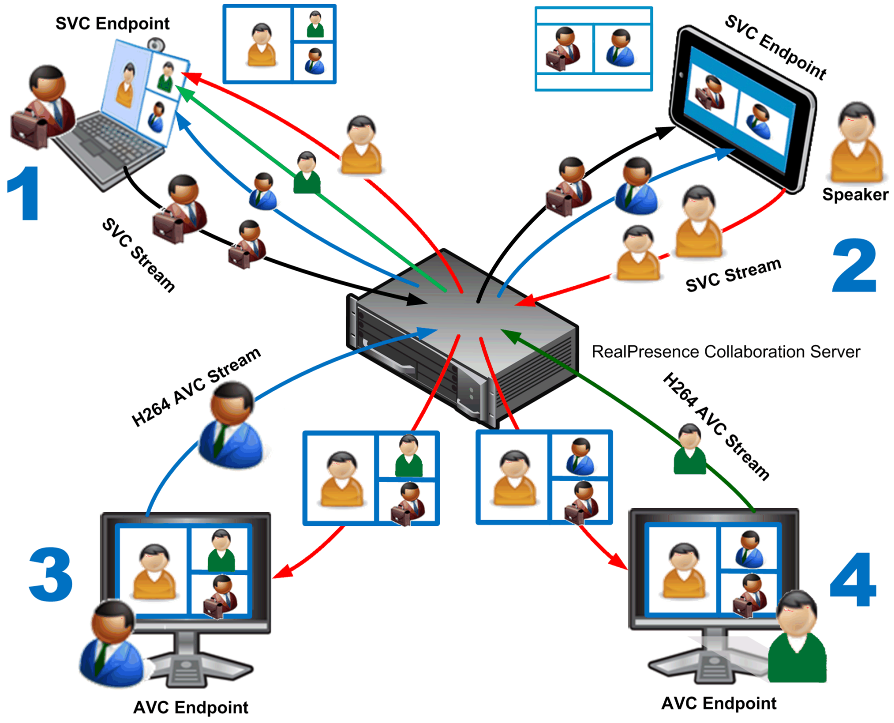
Setting	Value
Enable FECC	Disabled
Video Quality	Sharpness
Maximum Resolution	Auto
Content Settings	Graphics
Content Protocol	H.264 Cascading and SVC Optimized
Presentation Mode	Not applicable
Same Layout	Not applicable
Lecturer View Switching	Not applicable
Auto Scan Interval	Not applicable
Auto Layout	Enabled (Only available option)
Mute participants except the lecturer	Not applicable
IVR Name	Conference IVR Service
Network Services - SIP Registration	Disabled
Network Services - Accept Calls	Enabled

Mixed CP and SVC Conferencing

In a mixed CP (AVC) and SVC conference, AVC-based endpoints and SVC-enabled endpoints can be supported in the same conference.

In a mixed CP (AVC) and SVC conference, SVC endpoints transmit multiple resolutions and temporal layers to the RealPresence Collaboration Server like the SVC-based conferences, while AVC endpoints, for example, send only one H.264 AVC video stream to the Collaboration Server. Other endpoints (also referred to as AVC endpoints as opposed to SVC endpoints) can send different video protocols, such as H.263. The Collaboration Server relays SVC-decoded video bit streams to the SVC-enabled endpoints in the conference according to their display capabilities. This enables the video conference layouts to be automatically assembled by the endpoint. AVC endpoints connected to the conference send a single H.264 AVC video bit stream to the Collaboration Server, which is then transcoded to SVC video streams. SVC-enabled endpoints receive the AVC converted video bit streams through the Collaboration Server from the AVC endpoints as a single SVC video bit stream. Alternatively, AVC endpoints receive a single video bit stream with the defined video conference layout from the Collaboration Server. In this mixed mode conferencing, both SVC and AVC endpoints in the conference receive the same CP layout.

The following diagram illustrates an example of a mixed CP and SVC conferencing mode:



In this example, an SVC endpoint (**1**) receives three video streams at different frame rates and resolutions, and creates the conference layout with the received video streams. The video bit stream that the SVC endpoint receives from the AVC endpoint (**3**) is transcoded in the Collaboration Server and then encoded into an SVC bit stream in the required resolution.

Alternatively, an AVC endpoint (**4**) sends a single resolution video stream to the Collaboration Server. The Collaboration Server first converts the SVC bit stream into AVC, then transcodes the video received from the other endpoints to the required resolution. The Collaboration Server composes the video layout for the AVC endpoint and sends a single resolution video stream with the video layout to the participant. In the displayed example, the Collaboration Server creates different video layouts for each AVC endpoint.

Default Profile Settings in a Mixed CP and SVC Conferencing Mode

The Collaboration Server is shipped with a default *Conference Profile* (CP and SVC) for mixed CP and SVC conferences which enables users to immediately start a standard ongoing mixed CP and SVC conference. These are also the default settings when creating a new Profile. (During mixed SVC & CP conferences, PSTN (Audio Only) calls are supported.) Dial-out is not available in Mixed CP and SVC conferences.

The default settings are as follows:

Table 2-5 Default Mixed CP and SVC Conference Profile Settings

Setting	Value
Profile Name	Factory_Mix_SVC_CP_Video_Profile

Table 2-5 *Default Mixed CP and SVC Conference Profile Settings (Continued)*

Setting	Value
Line Rate	1920Kbps
Operator Conference	Disabled
Encryption	Enabled
Packet Loss Compensation (LPR and DBA)	Enabled for AVC participants only
Auto Terminate	• After last participant quits - Enabled • When last participant remains - Disabled
Auto Redialing	Disabled
Font for text over video	Enabled for AVC participants only
Exclusive Content Mode	Disabled
Enable FECC	Enabled
Video Quality	Sharpness
Maximum Resolution	Auto
Content Settings	Graphics
Content Protocol	H.264 Cascade and SVC Optimized
Presentation Mode	Disabled
Same Layout	Enabled
Lecturer View Switching	Disabled
Auto Scan Interval	Disabled
Auto Layout	Enabled
Mute participants except the lecturer	Disabled
Skin	Classic (for AVC participants)
IVR Name	Conference IVR Service
Recording	Enabled
Site Names display	Enabled for AVC participants only
Network Services - SIP Registration	Disabled
Network Services - Accept Calls	Enabled
Network quality indication	Enabled for AVC participants only

This *Profile* is automatically assigned to the following conferencing entities:

Name	ID
Meeting Rooms	
Maple_Room	*1001*
Oak_Room	*1002*
Juniper_Room	*1003*
Fig_Room	*1004*
Entry Queue	
Default EQ	*1000*

Resource Capacities for Mixed CP and SVC Conferences

In a mixed CP and SVC conference, video resources are allocated according to the MCU type and the translation pools (AVC to SVC and SVC to AVC) used to convert video streams. Translation pools are dynamically allocated, when the conference becomes a mixed CP and SVC conference; resources are not released when the conference stops being a mixed CP and SVC conference. The translation pools send one SVC to AVC stream with a resolution of 360p, two AVC to SVC streams with a resolution of 360p and 180p for AVC HD endpoints, and one video stream with a resolution of 180p for AVC SD endpoints. When a video stream with a resolution of 360p is not available, a video stream with a resolution of 180p is sent instead.

Translations between different endpoints can be done without using the highest resolution, thus saving translation resources. CP video layouts in mixed CP and SVC conferences support the standard resolutions as in normal CP conferences.

Taking these factors into consideration and the type of MCU deployed in the environment, the resource capacities for a mixed CP and SVC conference can vary.

The following table describes an example of the resource capacity allocations for the RealPresence Collaboration Server:

Table 2-7 *Estimated Resource Capacity Allocations*

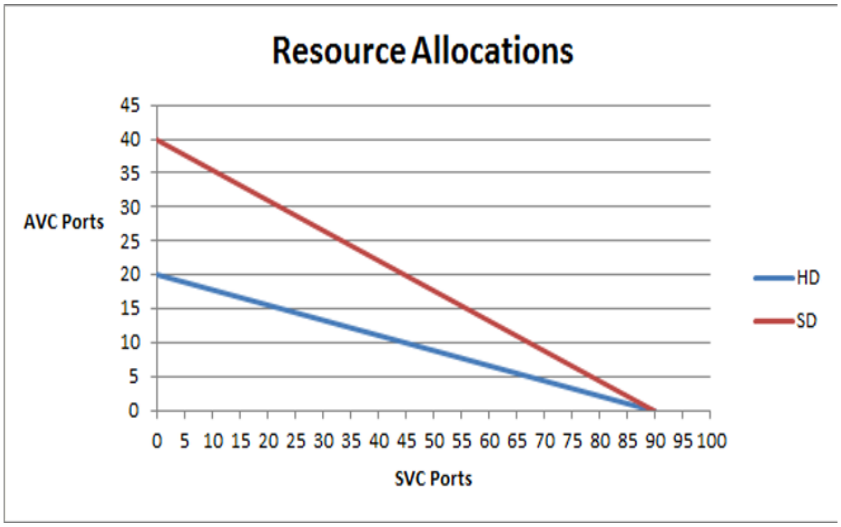
Resource Type	Number of Available Ports
Mixed CP and SVC (HD) (Example)	20 AVC 90 SVC
HD720p30	40
SD (@ 30 fps)	40
SVC Only	60
CIF (@ 30 fps)	60

The first four resource types in the resource capacity allocations table are endpoints in a CP only conference or a mixed CP and SVC conference before the actual resource allocations occur.

In a mixed CP and SVC conference, video resources are used according to the amount of both AVC and SVC participants in the conference and according to the actual type of the conference - mixed CP and SVC conferences or CP only conferences. The ratio of resources in a mixed conference is one AVC HD (720p30) video resource to three SVC video resources, meaning for each AVC HD video resource, three SVC video resources can be allocated.

In this resource capacity allocations example, the mixed CP and SVC conference can allocate a combination of AVC and SVC ports depending on the endpoints that are defined in the actual conference. For example, a conference can be defined as a mixed CP and SVC conference but will only allocate resources as a mixed conference when both AVC and SVC endpoints join the conference. When there are only one resource type of endpoints participating in the conference, such as AVC or SVC, the resource allocations are assigned according to the type of endpoint. For instance, a mixed CP and SVC conference with HD endpoints assigned, can have 60 or 120 ports allocated depending on the server configuration. When an SVC endpoint joins the conference, the conference becomes an actual mixed conference and the resource allocations are divided between the AVC and SVC endpoints. The Resource Report will reflect this by showing an increase in the resource usage.

The following diagram illustrates the amount of AVC to SVC port resources that are used in an actual mixed CP and SVC conference:



Viewing Profiles

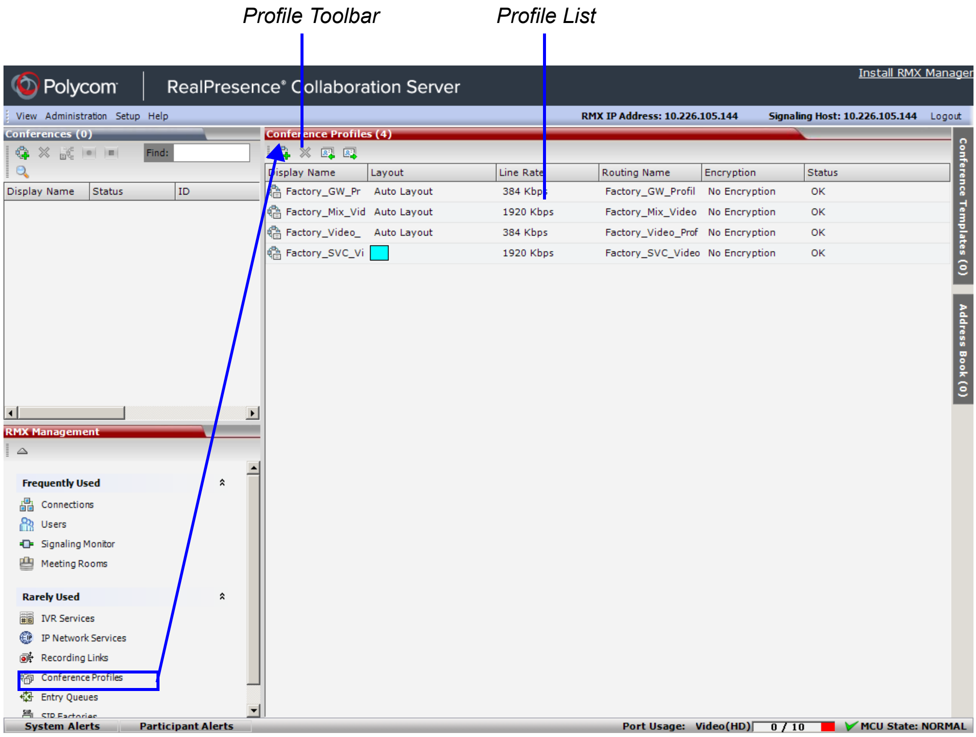
Conference Profiles are listed in the *Conference Profiles* list pane.

To list Conference Profiles:

1 In the *RealPresence Collaboration Server Management* pane, expand the *Rarely Used* list.

2 Click the **Conference Profiles** button.

The *Conference Profiles* are displayed in the *List* pane.



Display Name	Layout	Line Rate	Routing Name	Encryption	Status
Factory_GW_Pr	Auto Layout	384 Kbps	Factory_GW_Profil	No Encryption	OK
Factory_Mix_Vid	Auto Layout	1920 Kbps	Factory_Mix_Video	No Encryption	OK
Factory_Video_	Auto Layout	384 Kbps	Factory_Video_Prof	No Encryption	OK
Factory_SVC_Vi	🟦	1920 Kbps	Factory_SVC_Video	No Encryption	OK

The number of the currently defined Conference Profiles appears in the title of the list pane.

The following *Conference Profile* properties are displayed in the *List* pane:

Table 2-8 Conference Profiles Pane Columns

Field	Description
Name	The name of the *Conference Profile*.
Layout	Displays either “*Auto Layout*” or an icon of the layout selected for the profile. For information about video layouts, see Table 2-15 "*Video Layout Options”* on page **2-31**.
Line Rate	The maximum bit rate in kbps at which endpoints can connect to the conference.
Routing Name	Displays the Routing Name defined by the user or automatically generated by the system.
Encryption	Displays if media encryption is enabled for the Profile. For more information see "*Media Encryption”* on page **3-26**.

Profiles Toolbar

The Profile toolbar provides quick access to the Profile functions:

Table 2-9 Profile Tool bar buttons

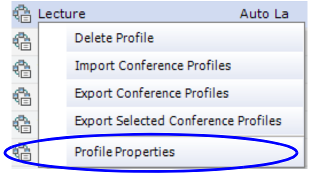
Button	Button Name	Description
	New Profile	To create a new Profile.
✖	*Delete Profile*	To delete a Profile, click the Profile name and then click this button.
	Import Profile	To import Conference Profiles from another MCU in your environment.
	Export Profile	To export Conference Profiles to a single XML file that can be used to import the Conference Profiles on multiple MCUs.

Modifying an Existing Profile

You can modify any of the Profile's parameters but you cannot rename the *Profile*.

To modify the Profile Properties:

1 In the *Conference Profiles* list, double -click the *Profile* icon or right-click the *Profile* icon, and then click **Profile Properties**.


Lecture
Auto La
Delete Profile
Import Conference Profiles
Export Conference Profiles
Export Selected Conference Profiles
Profile Properties

The Profile *Properties - General* dialog box opens.

Deleting a Conference Profile

To delete a Conference Profile:

1 In the *Conference Profiles* list, select the *Conference Profile* you want to delete.

2 Click the **Delete Profile** (✖) button.
or
Right-click the *Conference Profile* to be deleted and select **Delete Profile** from the drop-down menu.

A confirmation dialog box is displayed.

3 Click **OK** in the confirmation dialog box.

4 The *Conference Profile* is deleted.

A *Conference Profile* cannot be deleted if it is being used by Meeting Rooms, Entry Queues, and SIP Factories. A Profile that is assigned to only one ongoing conference and no other conferencing entity can be deleted.

Defining New Profiles

In the *RealPresence CloudAxis Solution*, the Conference Profiles are defined in the RealPresence DMA system component and should not be defined directly in the RealPresence Collaboration Server Virtual Edition component.

Profiles are the basis for the definition of all ongoing conferences, *Meeting Rooms*, *Entry Queues*, and *Conference Templates* and they contain only conference properties.

Conference Profile options differ according to the selected *Conferencing Mode* and *Conference Type*. Profiles can be defined for AVC (Advanced Video Codec) conferencing Mode or SVC (Scalable Video Codec) conferencing Mode.

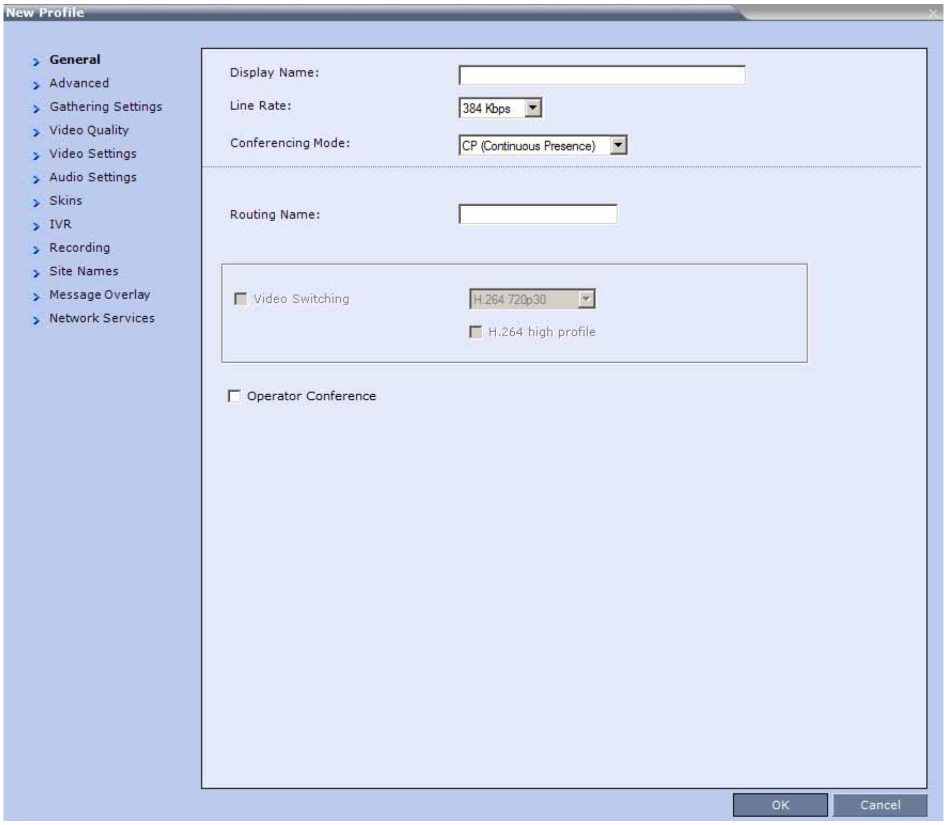
AVC Conferencing Mode offers the following Video session types: Continuous Presence (CP) conferences, Mixed CP and SVC conferences, and a special functional conference - Operator Conferences in CP mode.

Defining AVC CP Conferencing Profiles

To define a new Profile:

1 In the *RealPresence Collaboration Server Management* pane, click **Conference Profiles**.

2 In the *Conference Profiles* pane, click the **New Profile** button.
The *New Profile – General* dialog box opens.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

Routing Name:

□ Video Switching H.264 720p30

□ H.264 high profile

□ Operator Conference

3 Define the *Profile* name and, if required, the *Profile - General* parameters:

Table 2-10 *New AVC CP Profile - General Parameters*

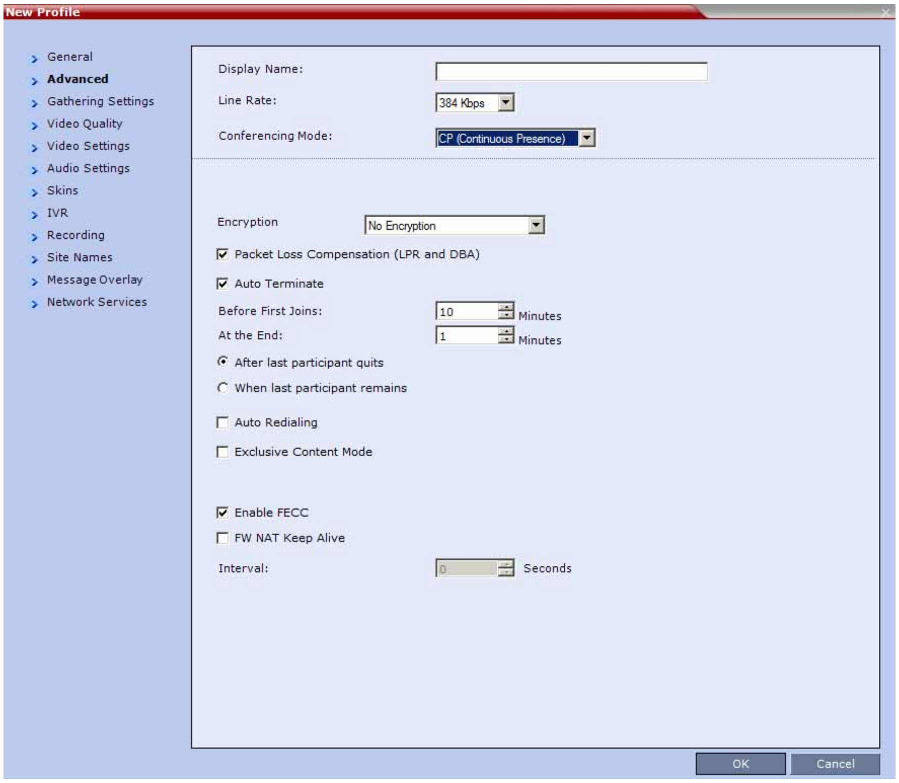
Field/Option	**Description**
Display Name	Enter a unique Profile name, as follows: • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. It is recommended to use a name that indicates the Profile type, such as Operator conference or Video Switching conference. **Note:** This is the only parameter that must be defined when creating a new profile. **Note:** This field is displayed in all tabs.

Table 2-10 New AVC CP Profile - General Parameters (Continued)

Field/Option	Description
Line Rate	Select the conference bit rate. The line rate represents the combined video, audio and Content rate. The default setting is 384 Kbps. **Notes:** • This field is displayed in all tabs.
Conferencing Mode	For CP conferencing, make sure that **CP (Continuous Presence)** is selected to define a CP conference Profile (it is the default option). **Note:** This field is displayed in all tabs.
Routing Name	Enter the *Profile* name using ASCII characters set. The Routing Name can be defined by the user or automatically generated by the system if no Routing Name is entered as follows: • If an all ASCII text is entered in Display Name, it is used also as the Routing Name. • If any combination of Unicode and ASCII text (or full Unicode text) is entered in Display Name, the ID (such as Conference ID) is used as the Routing Name.
Operator Conference	Select this option to define the profile of an Operator conference. For more information, see Chapter 9, "*Operator Assistance & Participant Move"* on page **9-1**.

4 Click the **Advanced** tab.

The *New Profile – Advanced* dialog box opens.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

Encryption: No Encryption

☑ Packet Loss Compensation (LPR and DBA)

☑ Auto Terminate

Before First Joins: 10 Minutes

At the End: 1 Minutes

● After last participant quits

○ When last participant remains

□ Auto Redialing

□ Exclusive Content Mode

☑ Enable FECC

□ FW NAT Keep Alive

Interval: 0 Seconds

5 Define the following parameters:

Table 2-11 New AVC CP Profile - Advanced Parameters

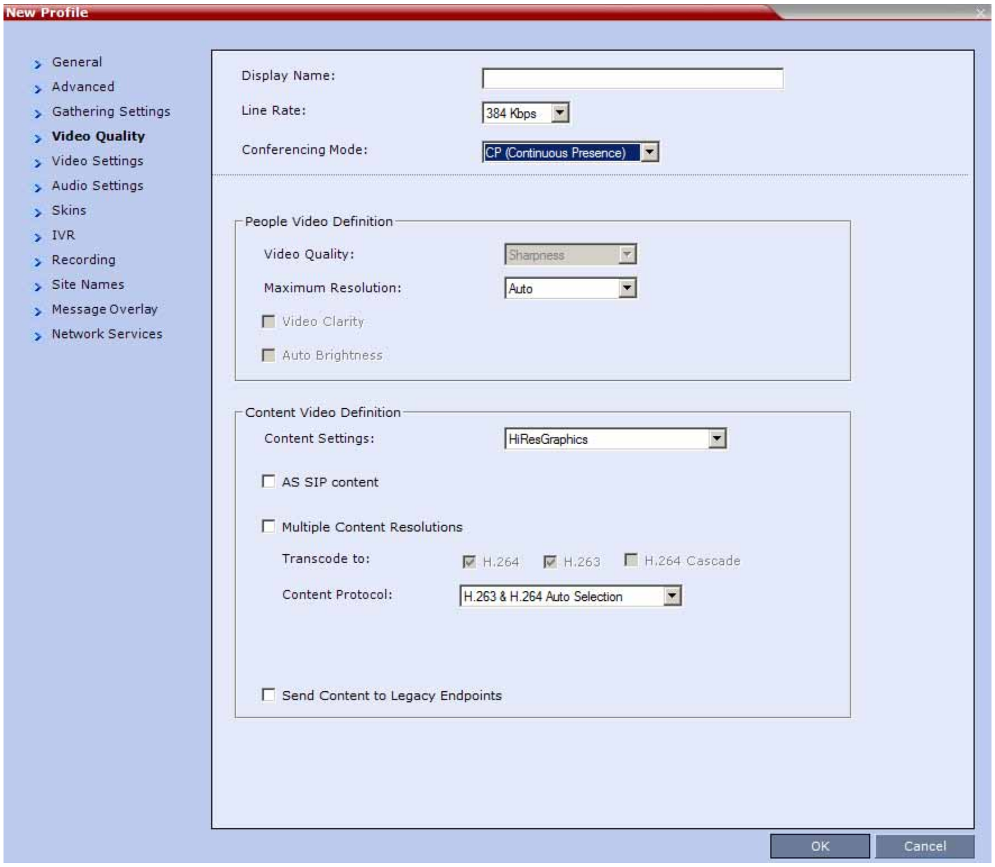
Field/Option	Description
Encryption	Select the Encryption option for the conference: • **Encrypt All** - Encryption is enabled for the conference and all conference participants must be encrypted. • **No Encryption** - Encryption is disabled for the conference. • **Encrypt when Possible** - enables the negotiation between the MCU and the endpoints and let the MCU connect the participants according to their capabilities, where encryption is the preferred setting. For connection guidelines see "*Mixing Encrypted and Non-encrypted Endpoints in one Conference"* on page **3-27**. For more information, see "*Media Encryption"* on page **3-26**.
LPR	When selected (default for CP conferences), *Lost Packet Recovery* creates additional packets that contain recovery information used to reconstruct packets that are lost during transmission. For more information, see "*Packet Loss Compensation (LPR and DBA) AVC CP Conferences"* on page **3-34**.

Table 2-11 New AVC CP Profile - Advanced Parameters (Continued)

Field/Option	Description
Auto Terminate	When selected (default), the conference automatically ends when the termination conditions are met: ***Before First Joins*** — No participant has connected to a conference during the *n* minutes after it started. Default idle time is 10 minutes. ***At the End - After Last Quits*** — All the participants have disconnected from the conference and the conference is idle (empty) for the predefined time period. Default idle time is 1 minute. ***At the End - When Last Participant Remains*** — Only one participant is still connected to the conference for the predefined time period (excluding the recording link which is not considered a participant when this option is selected). **Note:** The selection of this option is automatically cleared and disabled when the *Operator Conference* option is selected. The Operator conference cannot automatically end unless it is terminated by the Collaboration Server User.
Auto Redialing	The *Auto Redialing* option instructs the Collaboration Server to automatically redial *H.323* and *SIP* participants that have been abnormally disconnected from the conference. • *Auto Redialing* is disabled by default. • *Auto Redialing* can be enabled or disabled during an ongoing conference using the *Conference Properties – Advanced* dialog box. • The Collaboration Server will not redial an endpoint that has been disconnected from the conference by the participant. • The Collaboration Server will not redial an endpoint that has been disconnected or deleted from the conference by an operator or administrator.
Exclusive Content Mode	Select the *Exclusive Content Mode* check box to limit the Content broadcasting to one participant, preventing other participants from interrupting the Content broadcasting while it is active. For more details, see "*Exclusive Content Mode”* on page **3-15**.
Enable FECC	This option is enabled by default, allowing participants in the conference to control the zoom and PAN of other endpoints in the conference via the FECC channel. Clear this check box to disable this option for all conference participants.
FW NAT Keep Alive	The MCU can be configured to send a *FW NAT Keep Alive* message at specific Intervals for the *RTP*, *UDP* and *BFCP* channels. For more information see "*FW (Firewall) NAT Keep Alive”* on page **16-79**.
Interval	If needed, modify the *NAT Keep Alive Interval* field within the range of 1 - 86400 seconds.For more information see "*FW (Firewall) NAT Keep Alive”* on page **16-79**.

6 Click the **Video Quality** tab.

The *New Profile – Video Quality* dialog box opens.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

People Video Definition

Video Quality: Sharpness

Maximum Resolution: Auto

□ Video Clarity

□ Auto Brightness

Content Video Definition

Content Settings: HiResGraphics

□ AS SIP content

□ Multiple Content Resolutions

Transcode to: ☑ H.264 ☑ H.263 □ H.264 Cascade

Content Protocol: H.263 & H.264 Auto Selection

□ Send Content to Legacy Endpoints

7 Define the following parameters:

Table 2-12 New AVC CP Profile - Video Quality Parameters

Field/Option	Description
People Video Definition	
Video Quality	Sharpness is the only supported content format that supports higher video resolutions. **Note:** When Sharpness is selected as the *Video Quality* setting in the conference Profile, the Collaboration Server will send 4CIF (H.263) at 15fps instead of CIF (H.264) at 30fps. For more information, see "*Content Settings”* on page **3-7**"*Video Resolutions in AVC-based CP Conferencing”* on page **4-1**.

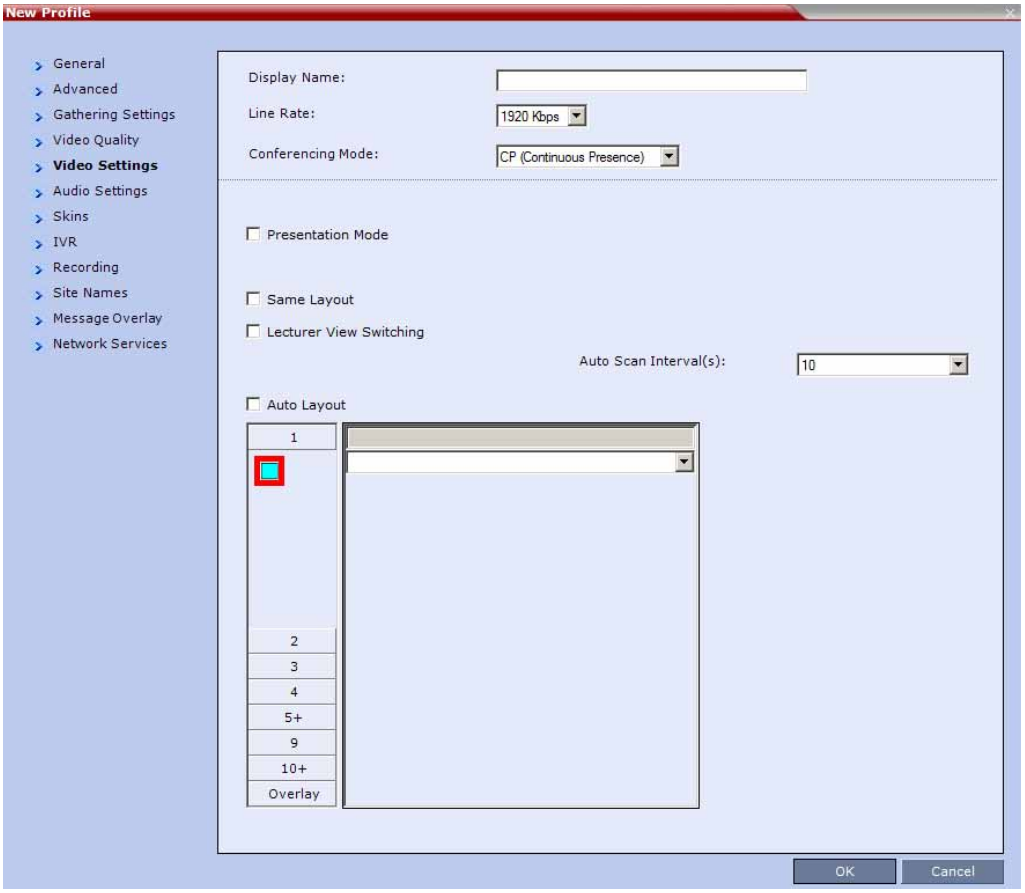
***Table 2-12** New AVC CP Profile - Video Quality Parameters (Continued)*

Field/Option	Description
Maximum Resolution	This setting overrides the *Maximum Resolution* setting of the *Resolution Configuration* dialog box. The administrator can select one of the following *Maximum Resolution* options: • *Auto* (default) - The *Maximum Resolution* remains as selected in the *Resolution Configuration* dialog box. • *CIF* • *SD* • *HD720* *Maximum Resolution* settings can be monitored in the *Profile Properties - Video Quality* and *Participant Properties - Advanced* dialog boxes. **Notes:** • The *Resolution* field in the *New Participant - Advanced* dialog box allows *Maximum Resolution* to be **further limited** per participant endpoint. • The *Maximum Resolution* settings for conferences and participants cannot be changed during an ongoing conference.
Content Video Definition	
Content Settings	Select the transmission mode for the Content channel: • ***Graphics*** — basic mode, intended for normal graphics • ***Hi-res Graphics*** (AVC CP Only) — a higher bit rate intended for high resolution graphic display • ***Live Video*** (AVC CP Only) — Content channel displays live video • ***Customized Content Rate*** (AVC CP Only) — manual definition of the Conference Content Rate**,** mainly for cascading conferences. Selection of a higher bit rate for the *Content* results in a lower bit rate for the people channel. For a detailed description of each of these options, see "*Content Sharing Parameters in Content Highest Common (Content Video Switching) Mode"* on page **3-6**.
AS SIP Content	This Content Sharing option is not supported with RealPresence Collaboration Server Virtual Edition..
Multiple Resolutions	This Content Sharing option is not supported with RealPresence Collaboration Server Virtual Edition.

Table 2-12 *New AVC CP Profile - Video Quality Parameters (Continued)*

Field/Option	Description
Content Protocol	Select the Content Protocol to be used for content sharing in Highest Common Content Sharing Mode. • **H.263**(AVC CP only) • *Content* is shared using the *H.263* protocol. • Use this option when most of the endpoints support *H.263* and some endpoints support *H.264*. • **H.263 & H.264 Auto Selection** (AVC CP only) • *Content* is shared using *H.263* if a mix of H.263-supporting and *H.264*-supporting endpoints are connected. • *Content* is shared using *H.264* if all connected endpoints have *H.264* capability. • **H.264 Cascade and SVC Optimized** • All *Content* is shared using the *H.264* content protocol and is optimized for use in *Cascaded Conferences*. • **H.264 HD** (AVC CP only, default) • Ensures high quality *Content* when most endpoints support *H.264* and *HD Resolutions*. When *Multiple Resolutions* is selected, this feature is hidden. For more information, see "*Content Protocols”* on page **3-8** and "*Defining Content Sharing Parameters for a Conference”* on page **3-3**.
Content Resolution	Select the Content Resolution and frame rate according to the selected Content Sharing Mode (Highest common Content or Multiple Resolution Contents) and the video protocol. For more information, see "*Defining Content Sharing Parameters for a Conference”* on page **3-3**.
Send Content to Legacy Endpoints (CP only)	This Content Sharing option is not supported with RealPresence Collaboration Server Virtual Edition.

8 Click the **Video Settings** tab.
The *New Profile - Video Settings* dialog box opens.



Display Name:

Line Rate: 1920 Kbps

Conferencing Mode: CP (Continuous Presence)

☐ Presentation Mode

☐ Same Layout

☐ Lecturer View Switching

Auto Scan Interval(s): 10

☐ Auto Layout

9 Define the video display mode and layout using the following parameters:

Table 2-13 New AVC CP Profile - Video Settings Parameters

Field/Option	Description
Presentation Mode (CP only)	Select this option to activate the Presentation Mode. In this mode, when the current speaker speaks for a predefined time (30 seconds), the conference changes to Lecture Mode. When another participant starts talking, the Presentation Mode is cancelled and the conference returns to the previous video layout.
Same Layout (CP only)	Select this option to force the selected layout on all participants in a conference. Displays the same video stream to all participants and personal selection of the video layout is disabled. In addition, if participants are forced to a video layout window, they can see themselves.

Table 2-13 New AVC CP Profile - Video Settings Parameters (Continued)

Field/Option	Description
Lecture View Switching	Select this option to enable automatic switching of participants on the Lecturer's screen when Lecture Mode is enabled for the conference. The automatic switching is enabled when the number of participants exceeds the number of video windows displayed on the Lecturer's screen. **Note:** Lecture Mode is enabled in the *Conference Properties – Participants* tab. For more information, see "*Lecture Mode (AVC CP Only)"* on page **3-38**.
Auto Scan Interval(s) **(CP only)**	Select the time interval, 5 - 300 seconds, that *Auto Scan* uses to cycle the display of participants that are not in the conference layout in the selected cell. *Auto Scan* is often used in conjunction with *Customized Polling* which allows the cyclic display to be set to a predefined order for a predefined time period.
Auto Layout **(CP only)**	When selected (default), the system automatically selects the conference layout based on the number of participants currently connected to the conference. When a new video participant connects or disconnects, the conference layout automatically changes to reflect the new number of video participants. For more information, see Table 2-14 "*Auto Layout – Default Layouts in CP Conferences"* on page **2-29**. Clear this selection to manually select a layout for the conference. The default Auto Layout settings can be customized by modifying default Auto Layout system flags in the System Configuration file. For more information see, "*Auto Layout Configuration"* on page **20-34**. **Note:** In some cases, the default layout automatically selected for the conference contains more cells than the number of connected participants, resulting in an empty cell. For example, if the number of connected participants is 4, the default layout is 2x2, but as only 3 participants are displayed in the layout (the participants do not see themselves), one cell is empty.

Table 2-14 Auto Layout – Default Layouts in CP Conferences

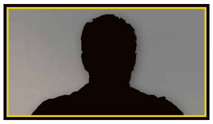
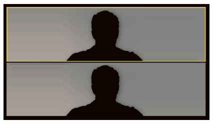
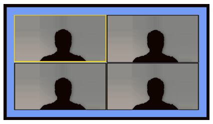
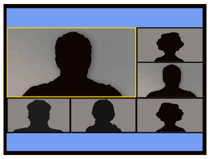
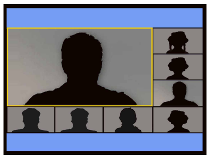
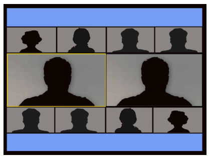
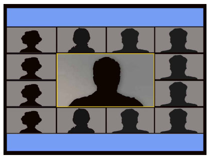
Number of Video Participants	Auto Layout Default Settings
0–2	
3	

Table 2-14 Auto Layout – Default Layouts in CP Conferences (Continued)

Number of Video Participants	Auto Layout Default Settings
4–5	
6–7	
8-10	
11	
12+	

In layout 2+8, the two central windows display the last two speakers in the conference: the current speaker and the "previous" speaker. To minimize the changes in the layout, when a new speaker is identified the "previous" speaker is replaced by the new speaker while the current speaker remains in his/her window.

The Collaboration Server supports the VUI addition to the H.264 protocol for endpoints that transmit wide video (16:9) in standard 4SIF resolution.

When there is a change of speaker in a Continuous Presence conference, the transition is set by default to fade in the current speaker while fading out the previous speaker.

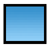
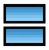
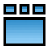
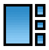
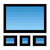
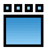
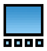
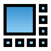
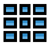
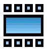
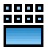
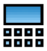
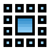
To make this transition visually pleasant, fading in the current speaker while fading out the previous speaker is done over a period of 500 milliseconds.

The *Fade In / Fade Out* feature can be disabled by adding a new flag to the *System Configuration*. The *Value* of the new flag must be: FADE_IN_FADE_OUT=NO.

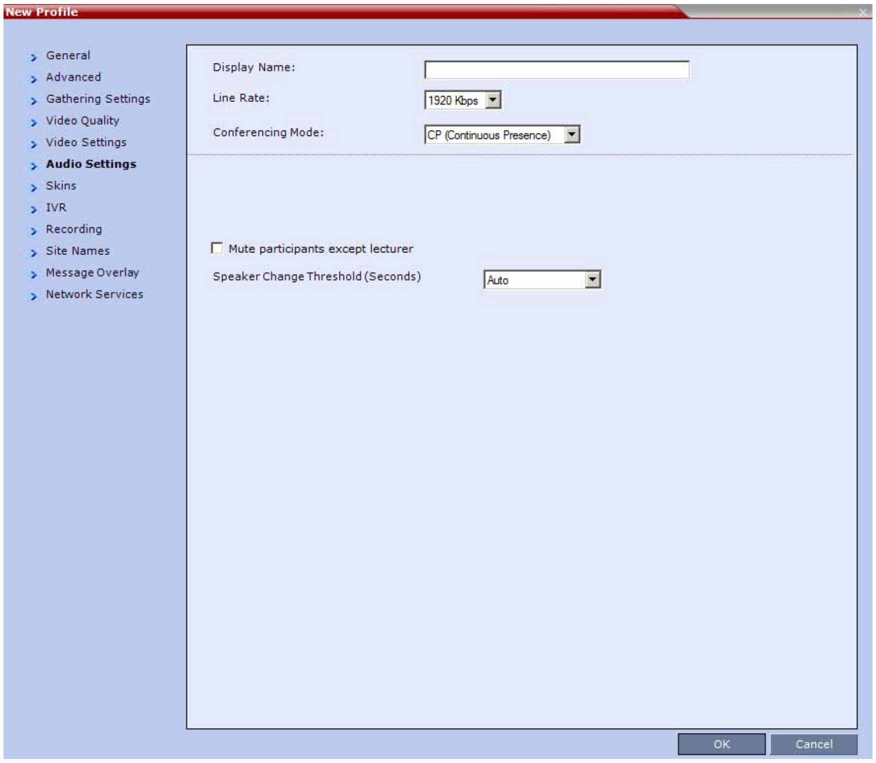
For more information about *System Flags*, see the *Polycom® RealPresence Collaboration Server Virtual Edition Administrator's Guide*, "*Modifying System Flags"* on page **20-1**.

10 To select the *Video Layout* for the conference, click the required number of windows from the layouts bar and then select the windows array. The selected layout is displayed in the *Video Layout* pane.

Table 2-15 *Video Layout Options*

Number of Video Windows	Available Video Layouts				
1					
2					
3					
4					
5+					
9					
10+					

11 Click the **Audio Settings** tab.
The *New Profile - Audio Settings* dialog box opens.



Display Name:

Line Rate: 1920 Kbps

Conferencing Mode: CP (Continuous Presence)

□ Mute participants except lecturer

Speaker Change Threshold (Seconds) Auto

12 Define the following parameters:

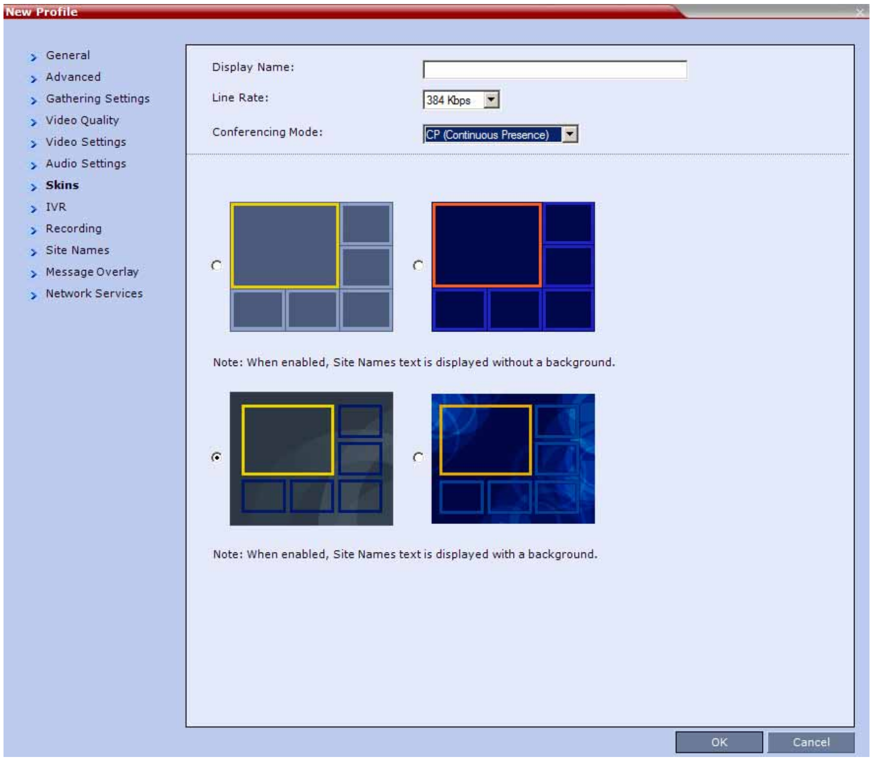
Table 2-16 New AVC CP Profile - Audio Settings Parameters

Field/Option	Description
Mute participant except lecturer	When the *Mute Participants Except Lecturer* option is enabled, the audio of all participants in the conference except for the lecturer can be automatically muted upon connection to the conference. This prevents other conference participants from accidentally interrupting the lecture, or from a noisy participant affecting the audio quality of the entire conference. Muted participants cannot unmute themselves unless they are unmuted from the Collaboration Server Web Client/ RMX Manager. You can enable or disable this option during the ongoing conference. **Notes:** • When enabled, the mute indicator on the participant endpoints are not visible because the mute participants was initiated by the MCU. Therefore, it is recommended to inform the participants that their audio is muted by using the Closed Caption or Message Overlay functions. In the Collaboration Server Web Client/RMX Manager the mute by MCU indicator is listed for each muted participant in the *Audio* column in the *Participants* pane. • The *Mute Participants Except Lecturer* option can be disabled during an ongoing conference, thereby unmuting all the participants in the conference. • If the endpoint of the designated lecturer is muted when the lecturer connects to the conference, the lecturer remains muted until the endpoint has been unmuted. • When you replace a lecturer, the MCU automatically mutes the previous lecturer and unmutes the new lecturer.
Mute participant except lecturer (continued)	• When you disconnect a lecturer from the conference or the lecturer leaves the conference, all participants remain muted but are able to view participants in regular video layout until the you disable the *Mute Participants Except Lecturer* option. • A participant can override the *Mute Participants Except Lecturer* option by activating the *Mute All Except Me option* using the appropriate DTMF code, provided the participant has authorization for this operation in the *IVR Services*. The lecturer audio is muted and the participant audio is unmuted. You can reactivate the *Mute Participants Except Lecturer* option after a participant has previously activated the Mute All Except Me option. The participant is muted and the lecturer, if designated, is unmuted. • In cascaded conferences, all participants (including the link participants) are muted. Only the lecturer is not muted.

Table 2-16 *New AVC CP Profile - Audio Settings Parameters (Continued)*

Field/Option	Description
Speaker Change Threshold	The Speaker Change Threshold is the amount of time a participant must speak continuously before becoming the speaker. When defining or editing a conference profile, you can define the Speaker Change Threshold. Select the desired threshold: • Auto (Default, 3 seconds) • 1.5 seconds • 3 seconds • 5 seconds

13 Click the **Skins** tab to modify the background and frames.
The *New Profile - Skins* dialog box opens.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

○ ○

Note: When enabled, Site Names text is displayed without a background.

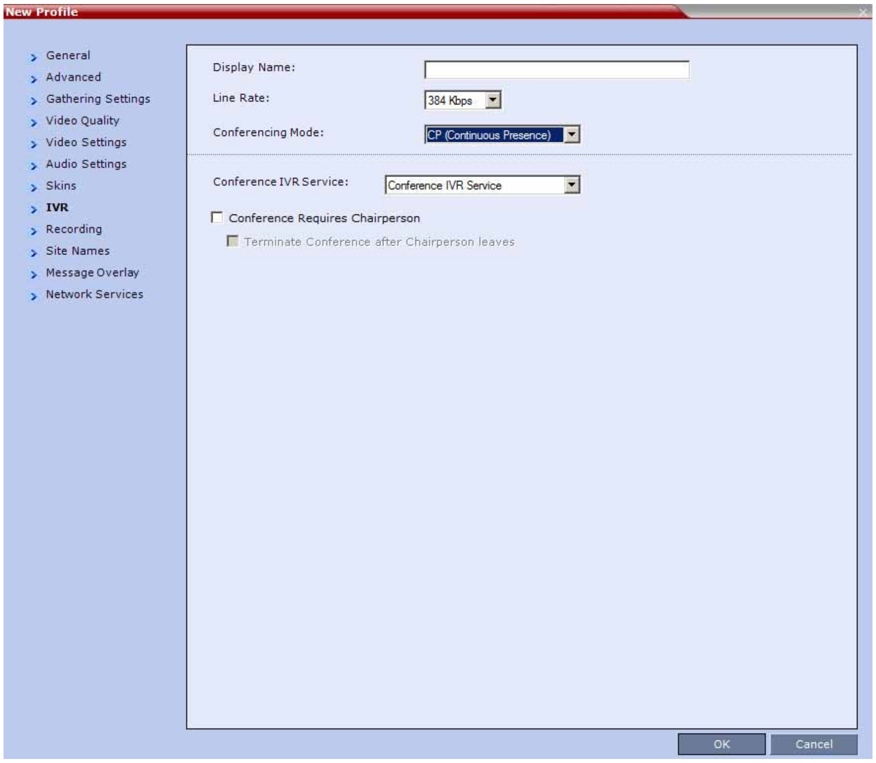
● ○

Note: When enabled, Site Names text is displayed with a background.

14 Select one of the *Skin* options.

15 Click **IVR** tab.

The *New Profile - IVR* dialog box opens.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

Conference IVR Service: Conference IVR Service

□ Conference Requires Chairperson

□ Terminate Conference after Chairperson leaves

16 If required, set the following parameters:

Table 2-17 New AVC CP Profile - IVR Parameters

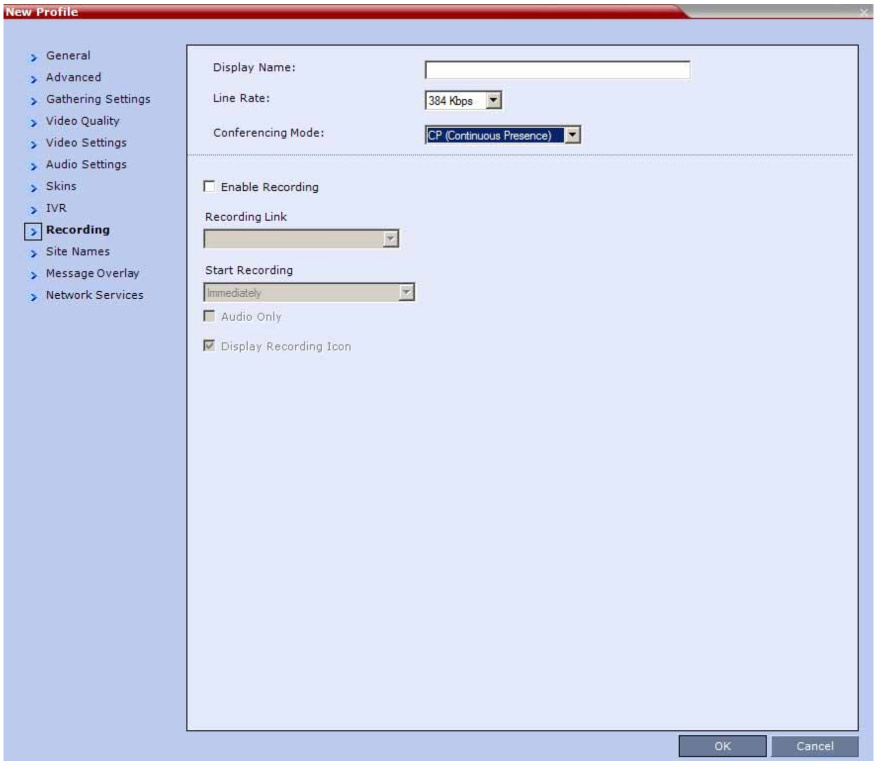
Field/Option	Description
Conference IVR Service	The default conference IVR Service is selected. You can select another conference IVR Service if required.
Conference Requires Chairperson	Select this option to allow the conference to start only when the chairperson connects to the conference and to automatically terminate the conference when the chairperson exits. Participants who connect to the conference before the chairperson are placed on *Hold* and hear background music (and see the *Welcome* video slide). Once the conference is activated, the participants are automatically connected to the conference. When the check box is cleared, the conference starts when the first participant connects to it and ends at the predefined time or according to the *Auto Terminate* rules when enabled.

Table 2-17 *New AVC CP Profile - IVR Parameters (Continued)*

Field/Option	Description
Terminate conference after chairperson leaves	Select this check box to automatically terminate the conference after the chairperson leaves. When the chairperson leaves, the *“Chairperson Has Left”* IVR message is played to all participants, at which point the conference terminates. This way an operator does not need to monitor a conference to know when to terminate it manually. If there is a single chairperson in the conference who is changed to a regular participant the conference will be terminated as if the chairperson left. If there is more than one chairperson, then changing one chairperson to a regular participant will not terminate the conference. It is therefore recommended that before changing a single chairperson to regular participant, another participant first be changed to chairperson. **Note:** The Chairperson can be either an AVC-enabled or SVC-enabled endpoint.

17 **Optional**. Click the **Recording** tab to enable conference recording with *Polycom RSS 2000/4000*.

The *New Profile - Recording* tab opens.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

□ Enable Recording

Recording Link

Start Recording: Immediately

□ Audio Only

☑ Display Recording Icon

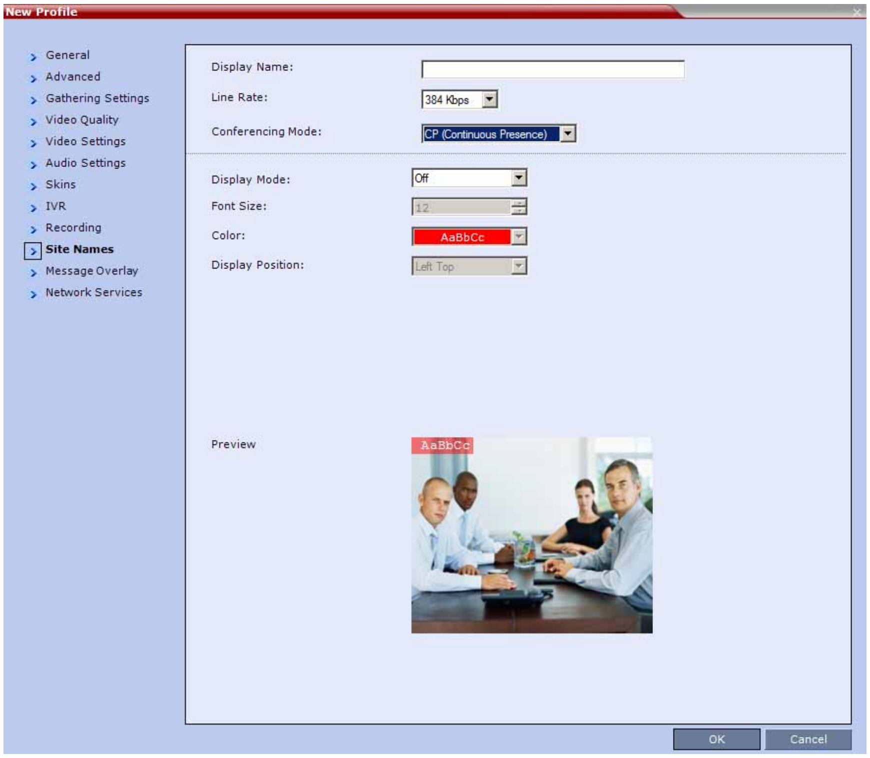
18 Define the following parameters:

Table 2-18 New AVC CP Profile - Recording Parameters

Parameter	Description
Enable Recording	Select this check box to enable the *Recording* settings. If no *Recording Links* are found an error message is displayed.
Recording Link	*Select the Recording Link to be used for conference recording.* *Recording Links* defined on the Collaboration Server can be given a descriptive name and can be associated with a *Virtual Recording Room (VRR)* saved on the *Polycom® RSS™ 4000 Version 6.0 Recording and Streaming Server* (*RSS*). For more information see "*Recording Conferences”* on page **13-1**.
Start Recording	Select one of the following: • **Immediately** – conference recording is automatically started upon connection of the first participant. • **Upon Request** – the operator or chairperson must initiate the recording (manual).
Display Recording Icon	This option is automatically selected to display a *Recording Indication* to all conference participants informing them that the conference is being recorded.

The Recording participant does not support H.264 High Profile. If recording a conference that is set to H.264 High Profile, the Recording participant connects as Audio Only and records the conference Audio.

19 Click the **Site Names** tab to display the *Site Names* dialog box.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

Display Mode: Off

Font Size: 12

Color: AaBbCc

Display Position: Left Top

Using the *Site Name* dialog box, you can control the display of the site names by defining the font, size, color, background color and transparency and position within the Video Window. For a detailed description of the site names options see "*Site Names Definition"* on page 2-55.

Define the following parameters:

***Table 2-19** New AVC CP Profile - Site Names Parameters*

Field	Description
Display Mode	Select the display mode for the site names: • **Auto** - Display the *Site Names* for 10 seconds whenever the *Video Layout* changes. • **On** - Display the *Site Names* for the duration of the conference. • **Off** (default) - Do not display the *Site Names*. **Note:** The *Display Mode* field is grayed and disabled if *Video Switching* mode is selected in the *Profile - General* tab. If *Display Mode* is **Off**, all other fields in this tab are grayed and disabled. Selecting **Off** enables *Video Switching* for selection in the *Profile - General* tab (if the conference is not already ongoing).

***Table 2-19** New AVC CP Profile - Site Names Parameters (Continued)*

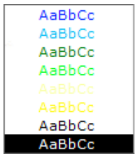
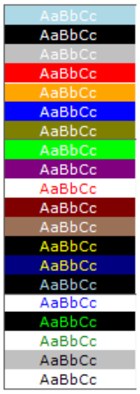
Field	Description
Font Size	Click the arrows to adjust the font size (in points) for the *Site Names* display. **Range:** 9 - 32 points **Default:** 12 **Note:** Choose a *Font Size* that is suitable for viewing at the conference's video resolution. For example, if the resolution is *CIF*, a larger *Font Size* should be selected for easier viewing.
Background Color	Select the color of the *Site Names* display text. The color and background for *Site Names* display text is dependent on whether a *Plain Skin* or a *Picture Skin* was selected for the conference in the *Profile - Skins* tab. The choices are: **Plain Skin**  **Default:** White Text No Background (For contrast, no background is shown as black when the text is white.) **Picture Skin**  AaBbCc **Default:** White Text Red Background **Note:** Choose a *Background Color* combination that is suitable for viewing at the conference's video resolution. At low resolutions, it is recommended to select brighter colors as dark colors may not provide for optimal viewing.

Table 2-19 New AVC CP Profile - Site Names Parameters (Continued)

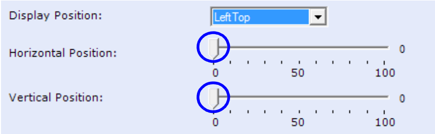
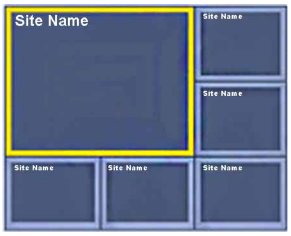
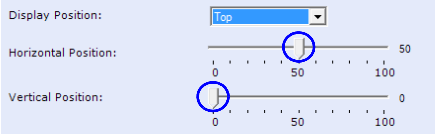
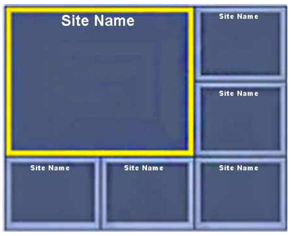
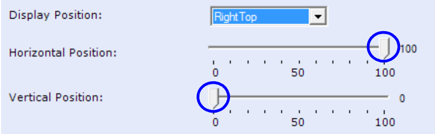
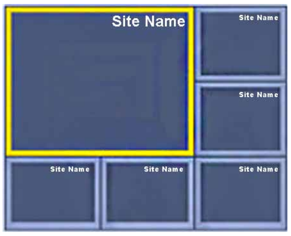
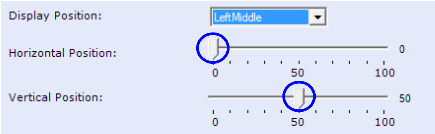
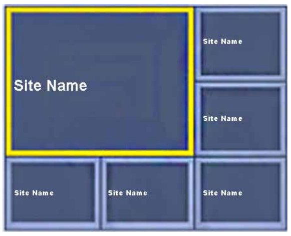
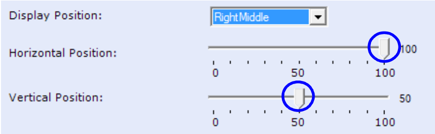
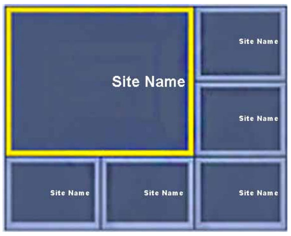
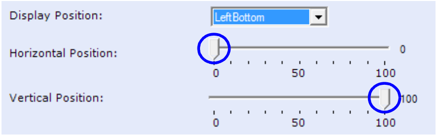
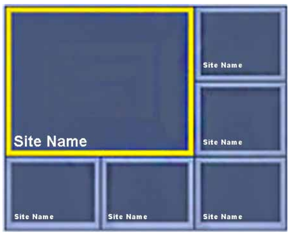
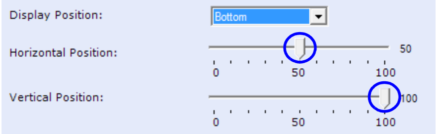
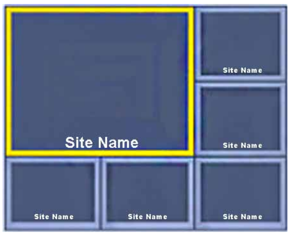
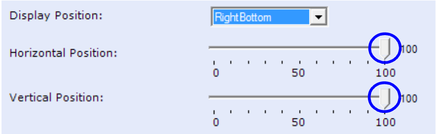
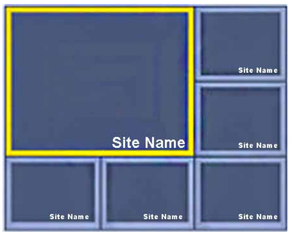
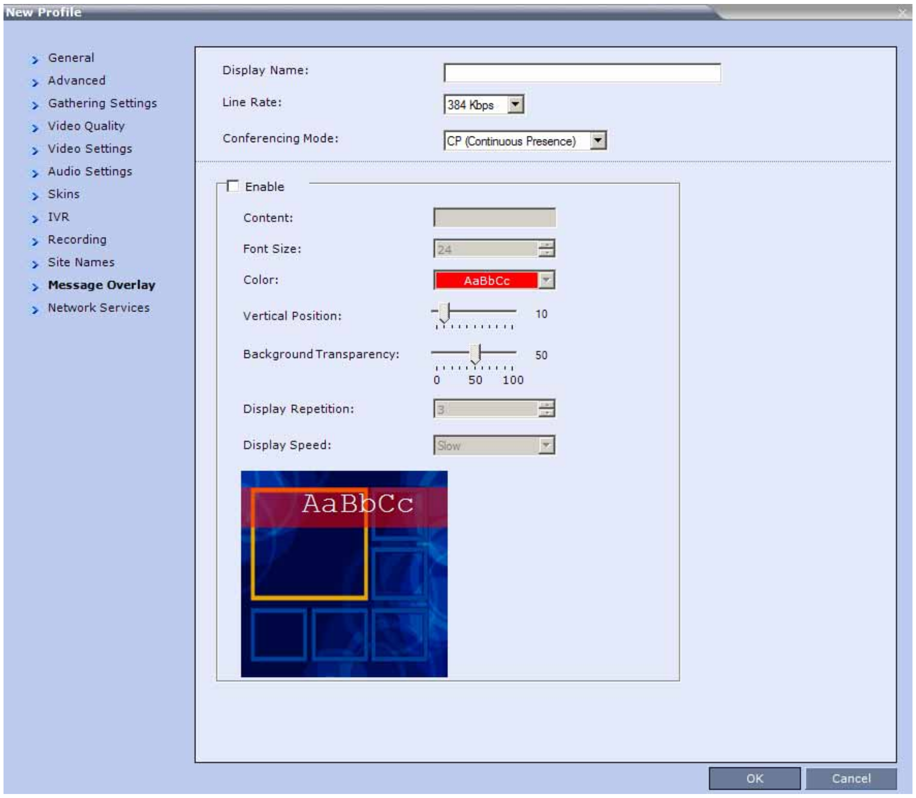
Field	Description	
Display Position	Select the pre-set position for the display of the *Site Names.*	
	Selection	***Site Names Position***
	LeftTop (Default)  Display Position: LeftTop Horizontal Position: 0 (0 50 100) Vertical Position: 0 (0 50 100)	 Site Name Site Name Site Name Site Name Site Name Site Name
	Top  Display Position: Top Horizontal Position: 50 (0 50 100) Vertical Position: 0 (0 50 100)	 Site Name Site Name Site Name Site Name Site Name Site Name
	RightTop  Display Position: RightTop Horizontal Position: 100 (0 50 100) Vertical Position: 0 (0 50 100)	 Site Name Site Name Site Name Site Name Site Name Site Name
	LeftMiddle  Display Position: LeftMiddle Horizontal Position: 0 (0 50 100) Vertical Position: 50 (0 50 100)	 Site Name Site Name Site Name Site Name Site Name Site Name
	RightMiddle  Display Position: RightMiddle Horizontal Position: 100 (0 50 100) Vertical Position: 50 (0 50 100)	 Site Name Site Name Site Name Site Name Site Name Site Name

Table 2-19 *New AVC CP Profile - Site Names Parameters (Continued)*

Field	Description	
Display Position (cont.)	LeftBottom  Display Position: LeftBottom Horizontal Position: 0 Vertical Position: 100 	 Site Name Site Name Site Name Site Name Site Name Site Name
	Bottom  Display Position: Bottom Horizontal Position: 50 Vertical Position: 100 	 Site Name Site Name Site Name Site Name Site Name Site Name
	RightBottom  Display Position: RightBottom Horizontal Position: 100 Vertical Position: 100 	 Site Name Site Name Site Name Site Name Site Name Site Name
	Custom	The current *Site Names* display position becomes the initial position for *Site Names* position adjustments using the *Horizontal* and *Vertical Position* sliders.
Horizontal Position	Move the slider to the **left** to move the horizontal position of the *Site Names* to the **left** within the *Video Windows.* Move the slider to the **right** to adjust the horizontal position of the *Site Names* to the **right** within the *Video Windows.*	**Note:** Use of these sliders will set the *Display Position* selection to **Custom**.
Vertical Position	Move the slider to the **left** to move the vertical position of the *Site Names* **upward** within the *Video Windows.* Move the slider to the **right** to move the vertical position of the *Site Names* **downward** within the *Video Windows.*	

20 **For CP Conferences only:** Click the **Message Overlay** tab to display the *Message Overlay* dialog box.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

☐ Enable

Content:

Font Size: 24

Color: AaBbCc

Vertical Position: 10

Background Transparency: 50

Display Repetition: 3

Display Speed: Slow

Message Overlay enables you to send text messages to all participants during ongoing Continuous Presence conferences.

The text message is seen as part of the in the participant's video layout on the endpoint screen or desktop display.

For more details, see "*Message Overlay for Text Messaging"* on page 2-58.

Define the following fields:

Table 2-20 *New AVC CP Profile - Message Overlay Parameters*

Field	Description
Enable	Select this check box to enable *Message Overlay*. Clear this check box to disable *Message Overlay*. **Default:** Cleared. **Note:** • The *Message Overlay* field is shaded and disabled when *Video Switching* mode is selected in the *New Profile - General* tab. All other fields in this tab are also disabled. • Clearing the *Enable* check box enables *Video Switching* for selection in the *New Profile - General* tab. • If *Message Overlay* is selected, the *Video Switching* check box in the *New Profile - General* tab is disabled and cannot be selected.

***Table 2-20** New AVC CP Profile - Message Overlay Parameters (Continued)*

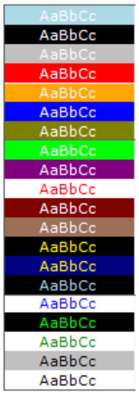
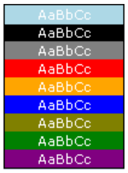
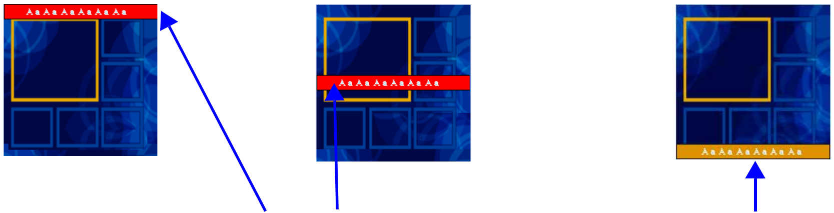
Field	Description
Content	Enter the message text. The message text can be up to 50 Chinese characters.
Font Size	Click the arrows to adjust the font size (points) for the *Message Overlay* display. **Range:** 9 - 32 **Default:** 24 **Note:** In some languages, for example Russian, when a large font size is selected, both rolling and static messages may be truncated if the message length exceeds the resolution width.
Color	From the drop-down menu select the color and background of the *Message Overlay* display text. The choices are:  **MPMx Mode** Color Options  **Default:** White Text on Red Background.
Vertical Position	Move the slider to the **right** to move the vertical position of the *Message Overlay* **downward** within the *Video Layout.* Move the slider to the **left** to move the vertical position of the *Message Overlay* **upward** within the *Video Layout.* **Default:** Top Left (10)
Background Transparency	Move the slider to the **left** to **decrease** the transparency of the background of the *Message Overlay* text. 0 = No transparency (solid background color). Move the slider to the **right** to **increase** the transparency of the background of the *Message Overlay* text. 100 = Full transparency (no background color). **Default:** 50
Display Repetition	Click the arrows to increase or decrease the number of times that the text message display is to be repeated. **Default:** 3

Table 2-20 New AVC CP Profile - Message Overlay Parameters (Continued)

Field	Description
Display Speed	Select whether the text message display is static or moving across the screen, the speed in which the text message moves: • Static • Slow • Fast **Default:** Slow

As the fields are modified the *Preview* changes to show the effect of the changes.
For example:

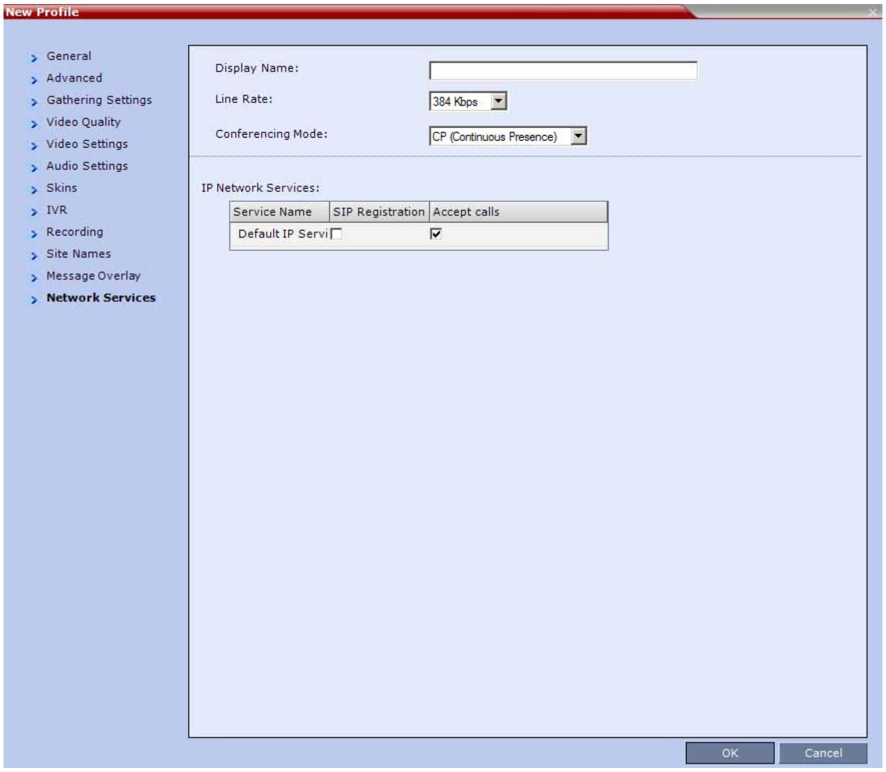


Small Text, White on red, Top, Middle

Small Text, White on yellow, Bottom

21 Click the **Network Services** tab.

The *New Profile - Network Services* tab opens.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

IP Network Services:

Service Name	SIP Registration	Accept calls
Default IP Servi	☐	☑

Registration of conferencing entities such as ongoing conferences, Meeting Rooms, Entry Queues and SIP Factories with SIP servers is done per conferencing entity. This allows better control on the number of entities that register with each SIP server.

Selective registration is enabled by assigning a conference Profile in which registration is configured to the required conferencing entities. Assigning a conference Profile in which registration is not configure to conferencing entities will prevent them from registering. By default, Registration is disabled in the Conference Profile, and must be enabled in Profiles assigned to conferencing entities that require registration.

22 Define the following parameters:

Table 2-21 New AVC CP Profile - Network Services Parameters

Parameter	Description
IP Network Services:	
Service Name	This column lists all the defined *Network Services,* one or several depending on the system configuration.
SIP Registration	To register the conferencing entity to which this profile is assigned with the SIP Server of the selected *Network Service*, click the check box of that *Network Service* in this column. When SIP registration is not enabled in the conference profile, the Collaboration Server's registering to SIP Servers will each register with an URL derived from its own signaling address.
Accept Calls	To prevent dial in participants from connecting to a conferencing entity when connecting via a *Network Service*, clear the check box of the *Network Service* from which calls cannot connect to the conference.

23 Click **OK** to complete the *Profile* definition.
A new *Profile* is created and added to the *Conference Profiles* list.

Defining SVC Conferencing Profiles

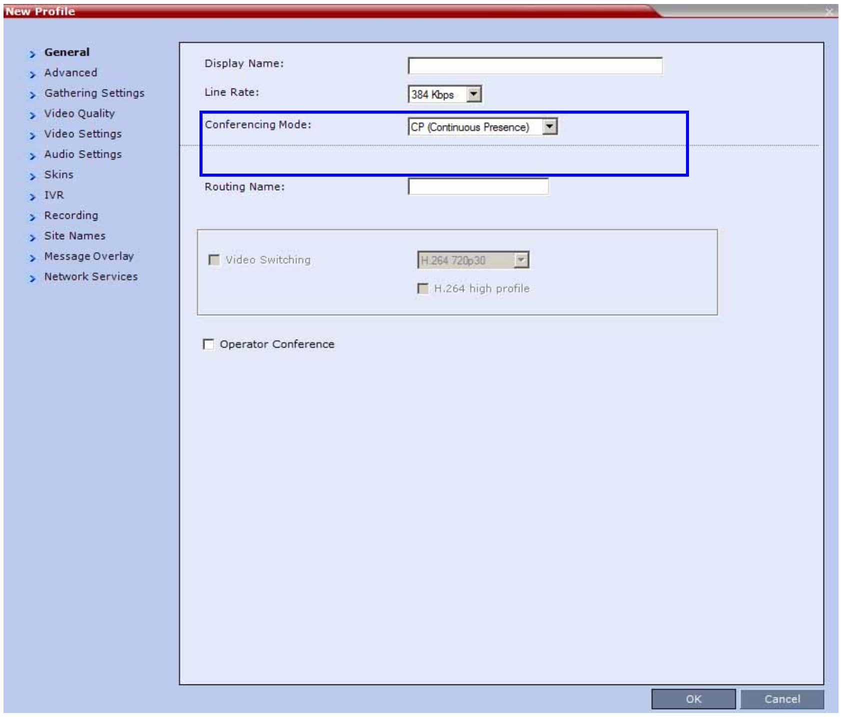
> In the *RealPresence CloudAxis Solution*, the Conference Profiles are defined in the RealPresence DMA system component and should not be defined directly in the RealPresence Collaboration Server Virtual Edition component.

To define SVC Only Profile:

1 In the *Collaboration Server Management* pane, click **Conference Profiles**.

2 In the *Conference Profiles* pane, click the **New Profile** button.

The *New Profile – General* dialog box opens.



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

Routing Name:

□ Video Switching H.264 720p30

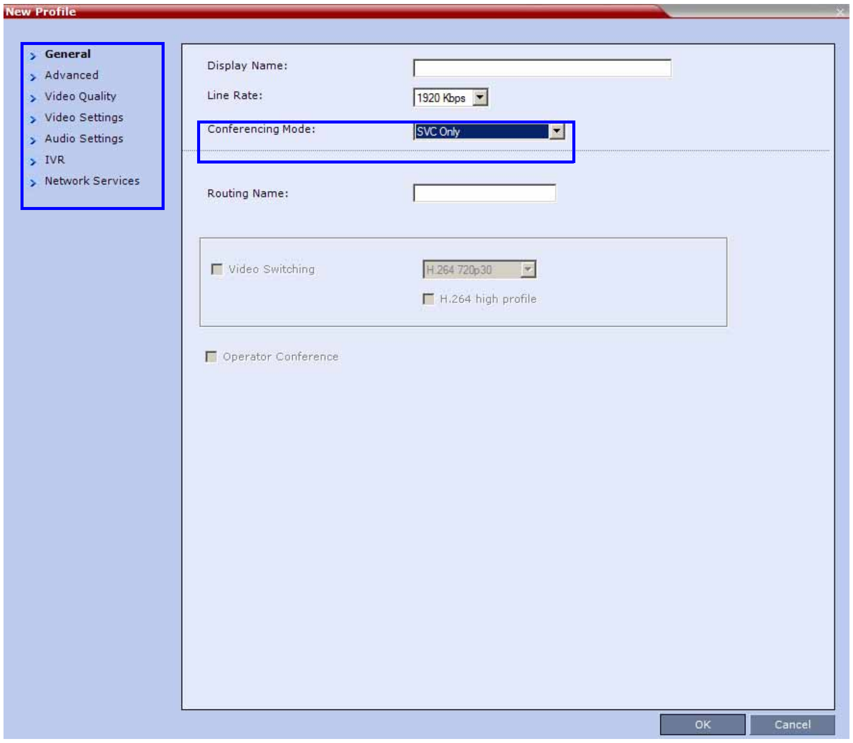
□ H.264 high profile

□ Operator Conference

By default, the Profile is set to *CP Conferencing Mode*.

3 In the *Conferencing Mode* list, select **SVC Only** to define the SVC Profile.

The profile tabs and options change accordingly and only supported options are available for selection. Unsupported options are disabled (grayed out).



Display Name:

Line Rate: 1920 Kbps

Conferencing Mode: SVC Only

Routing Name:

□ Video Switching H.264 720p30

□ H.264 high profile

□ Operator Conference

4 Define the *Profile* name and, if required, the *Profile - General* parameters:

Table 2-22 *New SVC Profile - General Parameters*

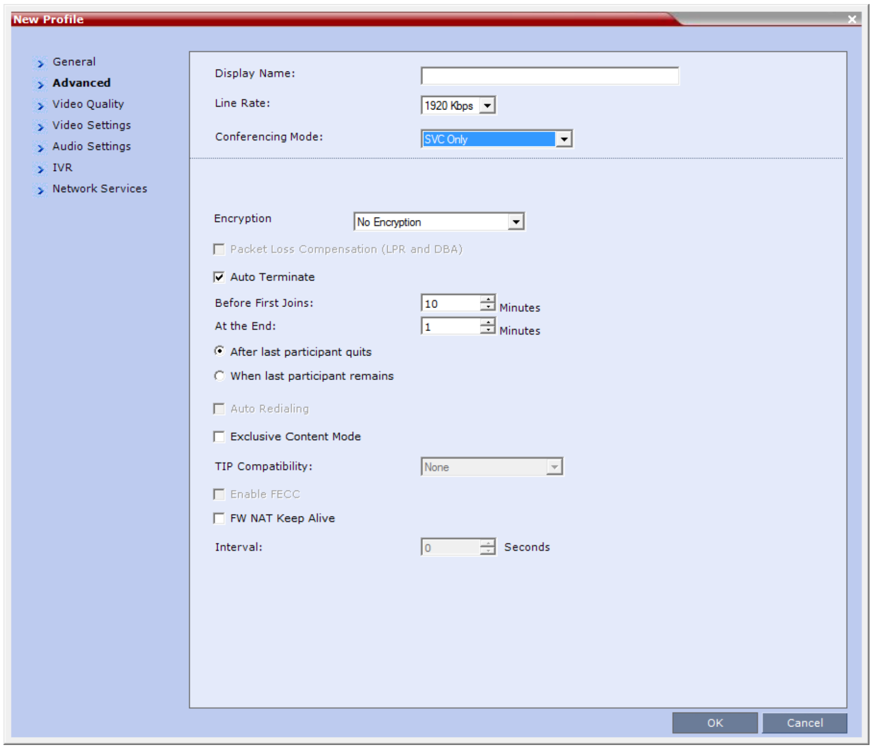
Field/Option	Description
Display Name	Enter a unique Profile name, as follows: • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. It is recommended to use a name that indicates the Profile type, such as Operator conference or Video Switching conference. **Note:** This is the only parameter that must be defined when creating a new profile. **Note:** This field is displayed in all tabs.
Line Rate	Select the conference bit rate. The line rate represents the combined video, audio and Content rate. The default setting for SVC Only conference is 1920kbps. **Notes:** • This field is displayed in all tabs.

Table 2-22 New SVC Profile - General Parameters (Continued)

Field/Option	Description
Routing Name	Enter the *Profile* name using ASCII characters set. The Routing Name can be defined by the user or automatically generated by the system if no Routing Name is entered as follows: • If an all ASCII text is entered in Display Name, it is used also as the Routing Name. • If any combination of Unicode and ASCII text (or full Unicode text) is entered in Display Name, the ID (such as Conference ID) is used as the Routing Name.

5 Click the **Advanced** tab.

The *New Profile – Advanced* dialog box opens.



Display Name:

Line Rate: 1920 Kbps

Conferencing Mode: SVC Only

Encryption: No Encryption

□ Packet Loss Compensation (LPR and DBA)

☑ Auto Terminate

Before First Joins: 10 Minutes

At the End: 1 Minutes

● After last participant quits

○ When last participant remains

□ Auto Redialing

□ Exclusive Content Mode

TIP Compatibility: None

□ Enable FECC

□ FW NAT Keep Alive

Interval: 0 Seconds

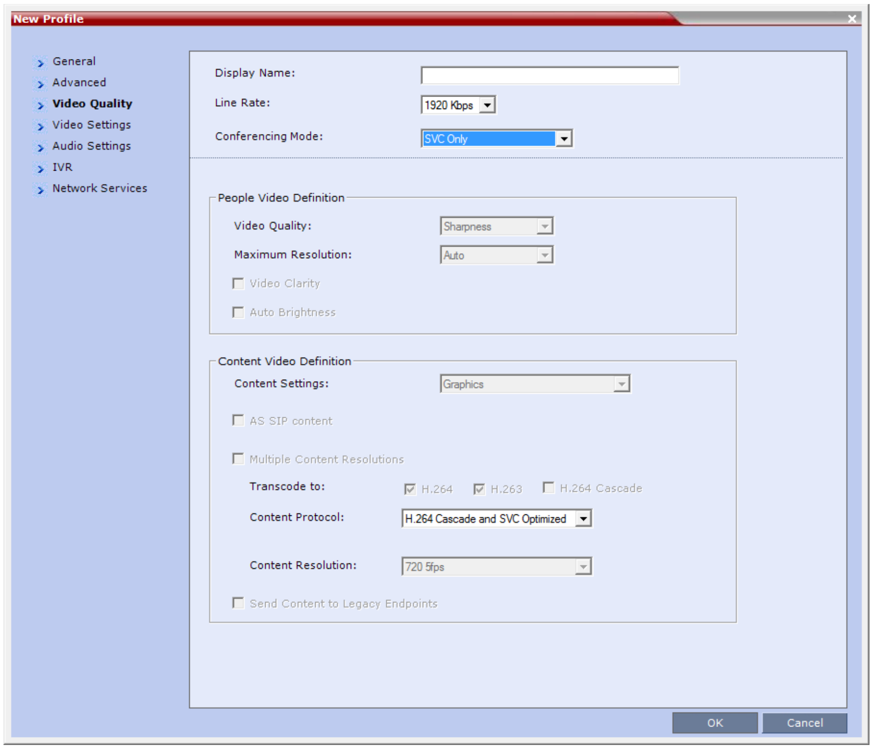
6 Define the following supported parameters:

Table 2-23 New SVC Profile - Advanced Parameters

Field/Option	Description
Encryption	Select the Encryption option for the conference: • **Encrypt All** - Encryption is enabled for the conference and all conference participants must be encrypted. • **No Encryption** - Encryption is disabled for the conference. • **Encrypt when Possible** - enables the negotiation between the MCU and the endpoints and let the MCU connect the participants according to their capabilities, where encryption is the preferred setting. For connection guidelines see "*Mixing Encrypted and Non-encrypted Endpoints in one Conference”* on page **3-27**. For more information, see "*Media Encryption”* on page **3-26**.
Auto Terminate	When selected (default), the conference automatically ends when the termination conditions are met: ***Before First Joins*** — No participant has connected to a conference during the *n* minutes after it started. Default idle time is 10 minutes. ***At the End - After Last participant Quits*** — All the participants have disconnected from the conference and the conference is idle (empty) for the predefined time period. Default idle time is 1 minute. ***At the End - When Last Participant Remains*** — Only one participant is still connected to the conference for the predefined time period (excluding the recording link which is not considered a participant when this option is selected). It is not recommended to select this option for SVC Conferences. Default idle time is 1 minute.
Exclusive Content Mode	When selected, *Content* broadcasting is limited to one participant preventing other participants from interrupting the Content broadcasting while it is active. For more details, see
FW NAT Keep Alive	When selected, an *FW NAT Keep Alive* message is sent at an interval defined in the field below the check box.
Interval	The time in seconds between *FW NAT Keep Alive* messages.

7 Click the **Video Quality** tab.

The *New Profile – Video Quality* dialog box opens.



Display Name:

Line Rate: 1920 Kbps

Conferencing Mode: SVC Only

People Video Definition

Video Quality: Sharpness

Maximum Resolution: Auto

□ Video Clarity

□ Auto Brightness

Content Video Definition

Content Settings: Graphics

□ AS SIP content

□ Multiple Content Resolutions

Transcode to: ☑ H.264 ☑ H.263 □ H.264 Cascade

Content Protocol: H.264 Cascade and SVC Optimized

Content Resolution: 720 5fps

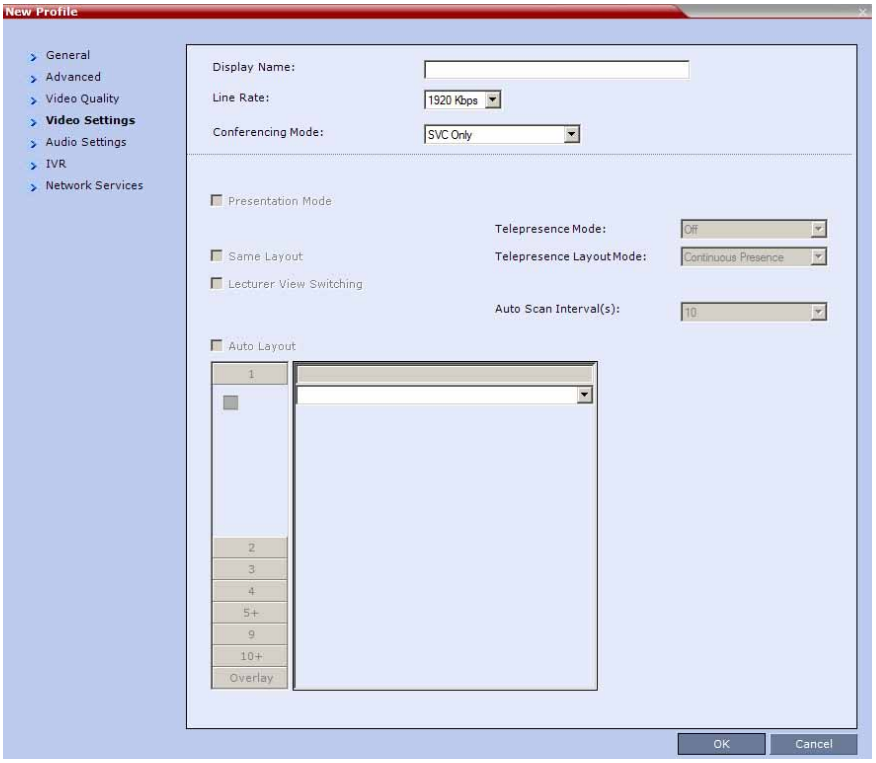
□ Send Content to Legacy Endpoints

8 In SVC Profiles, the video and Content sharing parameters cannot be modified and they are set to the following parameters:

Table 2-24 New SVC Profile - Video Quality Parameters

Field/Option	Description
Content Video Definition	
Content Settings	Only **Graphics** is available in SVC Conferencing Mode for transmission of Content. It offers the basic mode, intended for normal graphics. For more information, see "*Content Sharing"* on page 3-1.
Content Protocol	**H.264 Cascade and SVC Optimized** is the only available Content Protocol for content sharing during SVC-based conferences. In this mode, all *Content* is shared using the *H.264* content protocol and all endpoints must use the set video resolution and frame rate (720p 5fps). Endpoints that do not support these settings cannot share content.

9 Click the **Video Settings** tab.



Display Name:

Line Rate: 1920 Kbps

Conferencing Mode: SVC Only

□ Presentation Mode

Telepresence Mode: Off

□ Same Layout

Telepresence Layout Mode: Continuous Presence

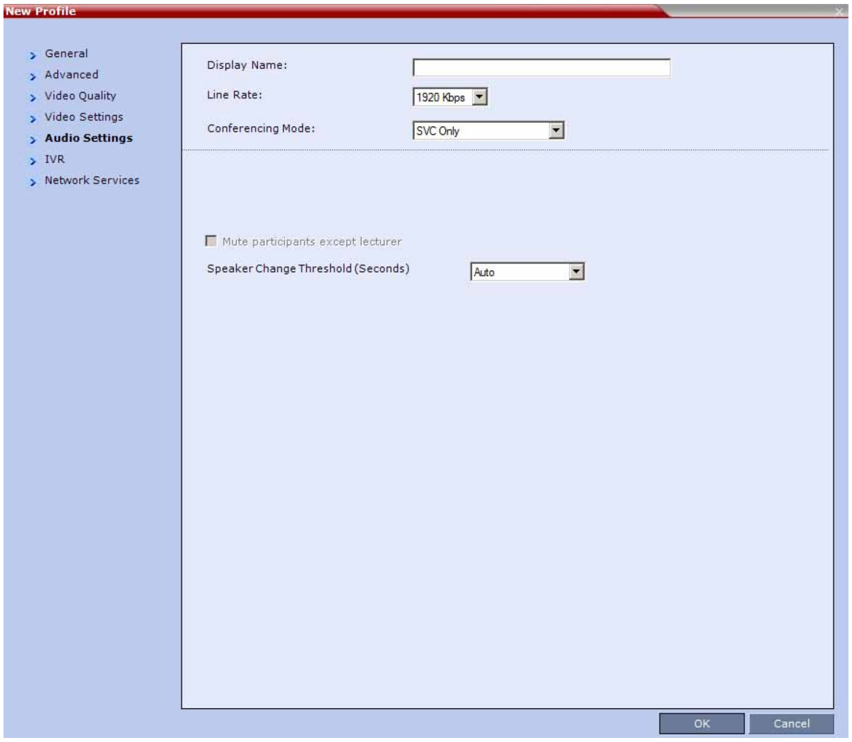
□ Lecturer View Switching

Auto Scan Interval(s): 10

□ Auto Layout

By default, the layout is set to **Auto** Layout and it cannot be changed. All other video settings options are unavailable in SVC Only conferences.

10 Click the **Audio Settings** tab.



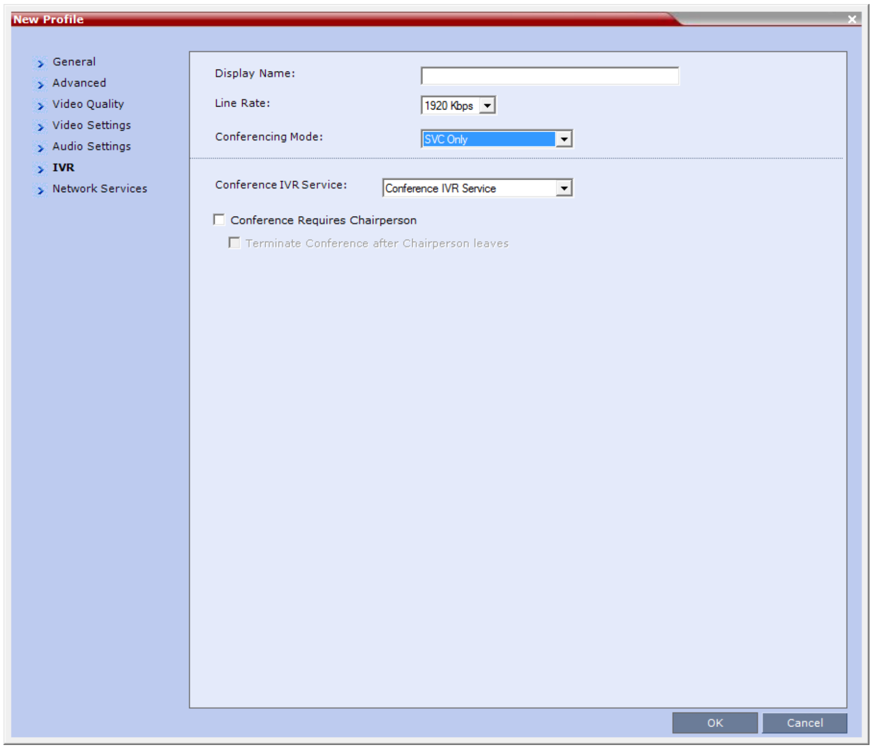
New Profile

Display Name:
Line Rate: 1920 Kbps
Conferencing Mode: SVC Only

□ Mute participants except lecturer
Speaker Change Threshold (Seconds) Auto

All the Audio options are disabled in SVC Only Profiles.

11 Click the **IVR** tab.



New Profile

Display Name:
Line Rate: 1920 Kbps
Conferencing Mode: SVC Only

Conference IVR Service: Conference IVR Service
□ Conference Requires Chairperson
□ Terminate Conference after Chairperson leaves

12 If required, set the following parameters:

Table 2-25 New AVC CP Profile - IVR Parameters

Field/Option	Description
Conference IVR Service	The default conference IVR Service is selected. You can select another conference IVR Service if required.
Conference Requires Chairperson	Select this option to allow the conference to start only when the chairperson connects to the conference and to automatically terminate the conference when the chairperson exits. Participants who connect to the conference before the chairperson are placed on *Hold* and hear background music (and see the *Welcome* video slide). Once the conference is activated, the participants are automatically connected to the conference. When the check box is cleared, the conference starts when the first participant connects to it and ends at the predefined time or according to the *Auto Terminate* rules when enabled.
Terminate conference after chairperson leaves	Select this check box to automatically terminate the conference after the chairperson leaves. When the chairperson leaves, the *"Chairperson Has Left"* IVR message is played to all participants, at which point the conference terminates. This way an operator does not need to monitor a conference to know when to terminate it manually. If there is a single chairperson in the conference who is changed to a regular participant the conference will be terminated as if the chairperson left. If there is more than one chairperson, then changing one chairperson to a regular participant will not terminate the conference. It is therefore recommended that before changing a single chairperson to regular participant, another participant first be changed to chairperson. **Note:** The Chairperson can be either an AVC-enabled or SVC-enabled endpoint.

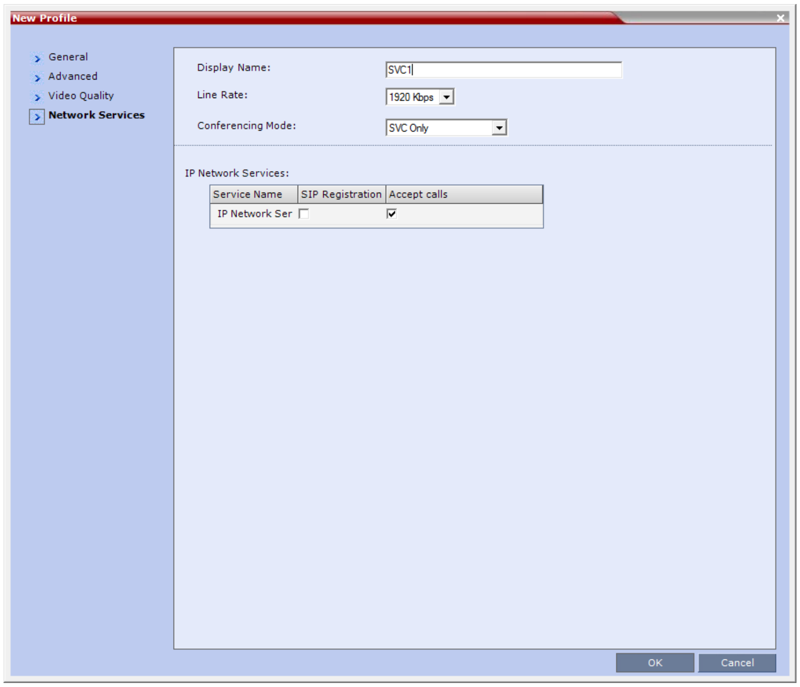
The following IVR features are not supported during SVC conferences:

— Invite Participants

— Entry and Exit tones

13 Click the **Network Services** tab.

The *New Profile - Network Services* tab opens.



Display Name: SVC1

Line Rate: 1920 Kbps

Conferencing Mode: SVC Only

IP Network Services:

Service Name	SIP Registration	Accept calls
IP Network Ser	☐	☑

Registration of conferencing entities such as ongoing conferences, Meeting Rooms, and SIP Factories with SIP servers is done per conferencing entity. This allows better control of the number of entities that register with each SIP server. Selective registration is enabled by assigning a conference Profile in which registration is configured for the required conferencing entities. Assigning a conference Profile in which registration is not configure for conferencing entities will prevent them from registering. By default, Registration is disabled in the Conference Profile, and must be enabled in Profiles assigned to conferencing entities that require registration.

14 Define the following parameters:

Table 2-26 New SVC Profile - Network Services Parameters

Parameter	Description
IP Network Services:	
Service Name	This column lists all the defined *Network Services*, one or several depending on the system configuration.
SIP Registration	To register the conferencing entity to which this profile is assigned with the SIP Server of the selected *Network Service*, click the check box of that *Network Service* in this column. When SIP registration is not enabled in the conference profile, the Collaboration Server's registering to SIP Servers will each register with an URL derived from its own signaling address.
Accept Calls	To prevent dial in participants from connecting to a conferencing entity when connecting via a *Network Service*, clear the check box of the *Network Service* from which calls cannot connect to the conference.

15 Click **OK** to complete the *Profile* definition.
A new *Profile* is created and added to the *Conference Profiles* list.

Defining Mixed CP and SVC Conferencing Profiles

> In the *RealPresence CloudAxis Solution*, the Conference Profiles are defined in the RealPresence DMA system component and should not be defined directly in the RealPresence Collaboration Server Virtual Edition component.

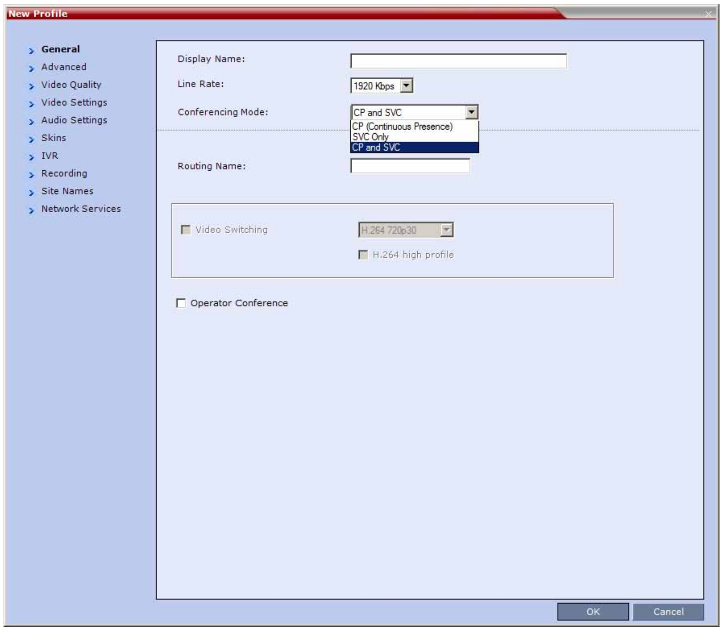
The mixed CP and SVC Profile is based on the CP Profile with a few of the CP options disabled for compatibility between AVC and SVC and to enable the media conversion between these two modes.

To configure a mixed AVC and SVC conference:

1 In the *RealPresence Collaboration Server Management* pane, click **Conference Profiles**.

2 In the *Conference Profiles* pane, click the **New Profile** button.

The *New Profile - General* dialog box is displayed.



Display Name:
Line Rate: 1920 Kbps
Conferencing Mode: CP and SVC
Routing Name:
□ Video Switching H.264 720p30
□ H.264 high profile
□ Operator Conference

3 In the *Conferencing Mode* list, select **CP and SVC** to define a mixed AVC and SVC conference.

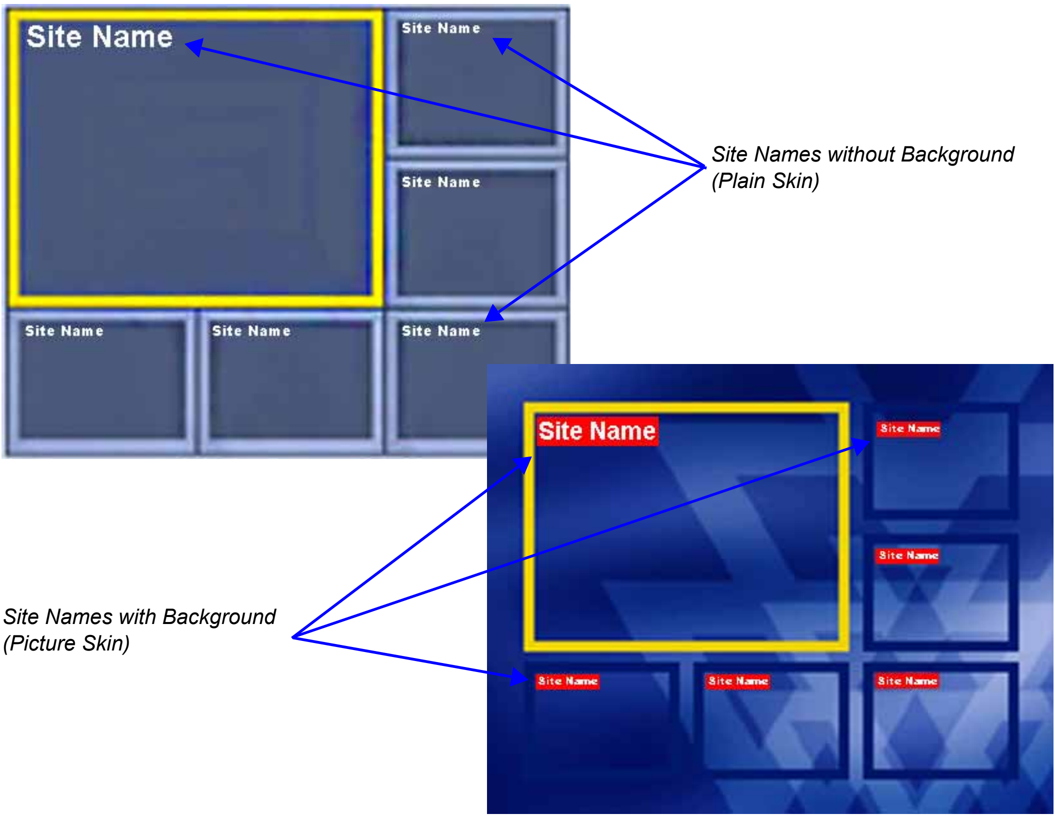
For a detailed description of the CP and SVC Profile options refer to "*Defining AVC CP Conferencing Profiles"* on page **2-21**.

CP Conferencing Additional Information

This section includes detailed explanation of various CP Profile settings.

Site Names Definition

Using the *Site Name* dialog box, you can control the display of the site names by defining the font, size, color, background color and transparency and position within the *Video Window*.



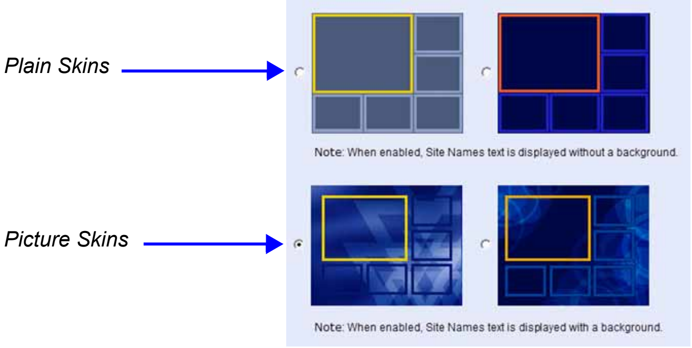
Site Names without Background (Plain Skin)

Site Names with Background (Picture Skin)

Guidelines

- *Site Names* display is **Off** by default in a new profile.
- *Site Names* can be enabled to function in one of two modes:
  — **Auto** – Site names are displayed for 10 seconds whenever the conference layout changes.
  — **On** – Site names are displayed for the duration of the conference.
- During the display of the site names, the video frame rate is slightly reduced
- *Site Names* display characteristics (position, size, color) can by modified during an ongoing conference using the *Conference Properties - Site Names* dialog box. Changes are immediately visible to all participants.
- *Site Names* display text and background color is dependent on the *Skin* selected for the conference:
  — **Plain Skins** - *Site Names* text is displayed without a background.

— **Picture Skins** - *Site Names* text is displayed with a background.



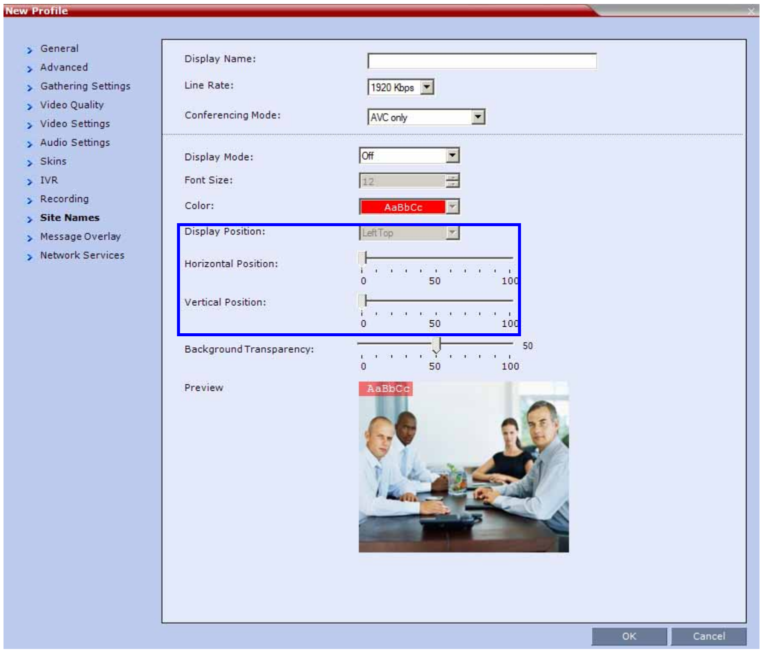
Note: When enabled, Site Names text is displayed without a background.

Note: When enabled, Site Names text is displayed with a background.

Site Names Display Position

The *Site Names* display position is controlled using three fields in the *Site Names* tab:

- *Display Position* drop-down menu
- *Horizontal Position* slider
- *Vertical Position* slider



Display Name:

Line Rate: 1920 Kbps

Conferencing Mode: AVC only

Display Mode: Off

Font Size: 12

Color: AaBbCc

Display Position: Left Top

Horizontal Position:

Vertical Position:

Background Transparency: 50

Using these three fields, the position at which the *Site Names* are displayed in the *Video Windows* can be set by:

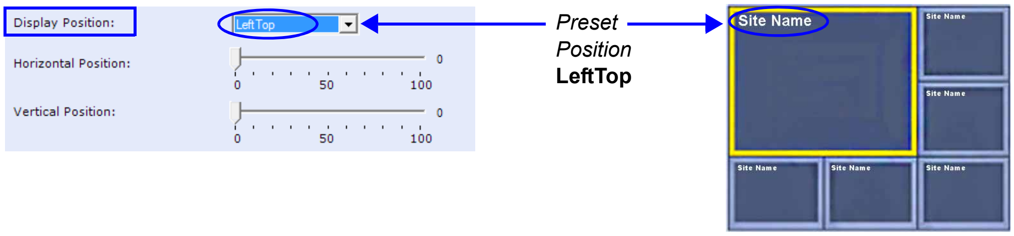
- Selecting a preset position from the drop-down menu in the *Display Position* field.
- Moving the *Horizontal* and *Vertical Position* sliders.
- Selecting **Custom** and moving the *Horizontal* and *Vertical Position* sliders.

Selecting a preset position from the drop-down menu in the Display Position field

>> In the *Display Position* drop-down menu select a preset position for *Site Names* display. Preset positions include:

LeftTop	*Top*	*RightTop*
LeftMiddle		*RightMiddle*
LeftBottom	*Bottom*	*RightBottom*
Custom		

When *Custom* is selected, the current position becomes the initial position for *Site Names* position adjustments using the *Horizontal* and *Vertical Position* sliders.



Display Position: LeftTop

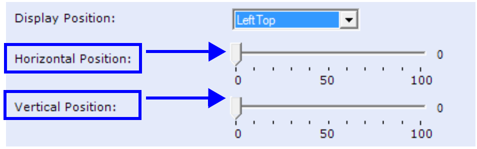
Horizontal Position: 0

Vertical Position: 0

The *Horizontal* and *Vertical Position* sliders are automatically adjusted to match the *Display Position* drop-down menu preset selection.

Moving the Horizontal and Vertical Position sliders

>> Drag the *Horizontal* and *Vertical Position* sliders to adjust the position of the *Site Names* display.



Display Position: LeftTop

Horizontal Position: 0

Vertical Position: 0

The *Site Names* display moves from its current position according to the slider movement.
Dragging the sliders causes the *Display Position* drop-down menu field to be set **Custom**.

Selecting Custom and moving the Horizontal and Vertical Position sliders

1. In the *Display Position* drop-down menu select **Custom**.

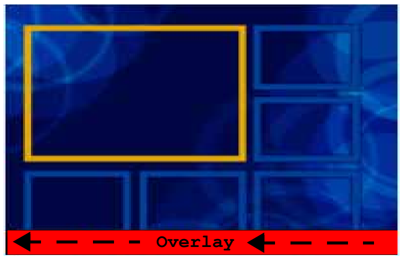
   The current *Site Names* position becomes the initial position for *Site Names* position. Dragging the *Horizontal* and *Vertical Position* sliders moves the *Site Names* from this position.

2. Drag the *Horizontal* and *Vertical Position* sliders to adjust the position of the *Site Names* display.

Message Overlay for Text Messaging

Message Overlay allows the operator or administrator to send text messages to a single, several or all participants during an ongoing conference.

The text message is seen as part of the in the participant's video layout on the endpoint screen or desktop display.



Guidelines

- *Message Overlay* messaging is supported in:
  - Continuous Presence (CP) conferences
  - in *Same Layout* mode
  - in encrypted conferences
- *Messages Overlay* can be enabled or disabled during the ongoing conference.
- Text messages *Content* can be changed on the fly during the ongoing conference.
- *Message Overlay* text messages are supported in *Unicode* or *ASCII* characters.
- The number of characters for each language can vary due to the type of font used, for example, the available number of characters for Chinese is 18, while for English and Russian it is 48.
- In an *Event Mode* conference with *Lecture Mode* enabled, only the *Lecturer* can see the *Message Overlay* text, while other participants on the lecturer's level cannot. Participants on other levels can see the *Message Overlay* text.
  - In some languages, for example Russian, when large font size is selected, both rolling and static messages may be truncated if the message length exceeds the resolution width.
- Changes to the *Message Overlay* display characteristics (position, size, color and speed) are immediately visible to all participants.
- Changes to the *Message Overlay Content* are immediately visible to all participants. When there is a current *Message Overlay*:
  - The current message is stopped immediately.
  - The *Display Repetition* count is reset to 1.
  - The new message *Content* is displayed <*Display Repetition*> times or until it is stopped and replaced by another *Content* change.
- If a *Repeating Message* is modified before it has completed all its repetitions, it is changed immediately without completing all of its repetitions. The modified *Repeating Message* is displayed starting with repetition one.
- *Message Overlay* messaging is not supported in *Lecture* mode.
- If during the ongoing conference the **Show Number of Participants** DTMF option (default DTMF ***88**) is used, when the displayed number of participants is removed, the message overlay text is also removed.

- Participants that have their video suspended do not receive *Message Overlays* messages.
- *Message Overlay* text messages cannot be sent via the *Content* channel.
- *Message Overlay* messages are not displayed when the *PCM* menu is active.
- *Message Overlay* text settings are not saved in the *Conference Template* when saving an ongoing conference as a *Conference Template*.
- Sending text messages using *Message Overlay* can be enabled or disabled during the ongoing conference. For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Sending Text Messages to All Participants Using Message Overlay (AVC-based Conferences)"* on page **3-63**.
- Text messages can be sent to individual or several participants during the ongoing conferences. For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Sending Messages to Selected Participants Using Message Overlay"* on page **3-70**.

For a detailed description of the Message Overlay parameters, see "*New AVC CP Profile - Message Overlay Parameters"* on page **2-41**.

Exporting and Importing Conference Profiles

Conference Profiles can be exported from one MCU and imported to multiple MCUs in your environment, enabling you to copy the *Conference Profiles* definitions to other systems. This can save configuration time and ensures that identical settings are used for conferences running on different MCUs. This is especially important in environments using cascading conferences that are running on different MCUs.

Guidelines

- Administrators can export and import *Conference Profiles*. Operators are only allowed to export *Conference Profiles*.
- You can select a single, multiple, or all *Conference Profiles* to be exported.
- *Conference Templates* and their related *Conference Profiles* can be exported and imported simultaneously using the *Conference Templates* export and import function. For more information, see the **Exporting and Importing Conference Templates** section.

Exporting Conference Profiles

Conference Profiles are exported to a single XML file that can be used to import the *Conference Profiles* on multiple MCUs.

Using the Export Conference Profile feature, you can:

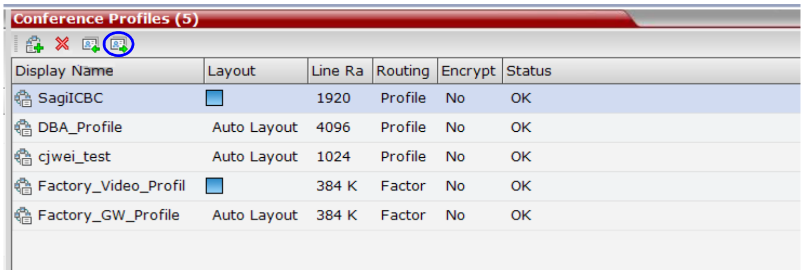
- Export all *Conference Profiles* from an MCU
- Export selected *Conference Profiles*

Exporting All Conference Profiles from an MCU

To export all Conference Profiles from an MCU:

1. In the *RealPresence Collaboration Server Management* pane, expand the *Rarely Used* list.
2. Click the **Conference Profiles** button.

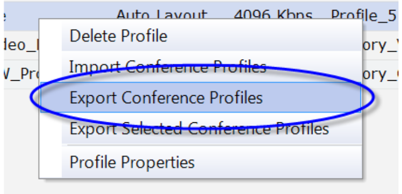
The *Conference Profiles* are displayed in the *List* pane.



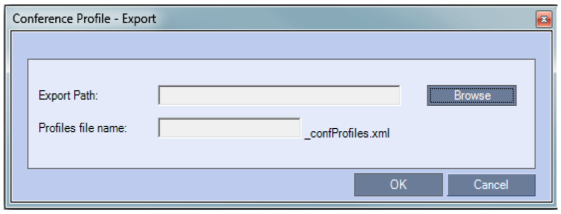
Conference Profiles (5)

Display Name	Layout	Line Ra	Routing	Encrypt	Status
SagiICBC	■	1920	Profile	No	OK
DBA_Profile	Auto Layout	4096	Profile	No	OK
cjwei_test	Auto Layout	1024	Profile	No	OK
Factory_Video_Profil	■	384 K	Factor	No	OK
Factory_GW_Profile	Auto Layout	384 K	Factor	No	OK

3 Click the **Export Conference Profiles** button or right-click the *Conference Profiles* pane, and then click **Export Conference Profiles**.


Auto Layout
4096 Kbps
Profile_5
Delete Profile
Import Conference Profiles
Export Conference Profiles
Export Selected Conference Profiles
Profile Properties

The *Conference Profile - Export* dialog box is displayed.



Conference Profile - Export

Export Path:

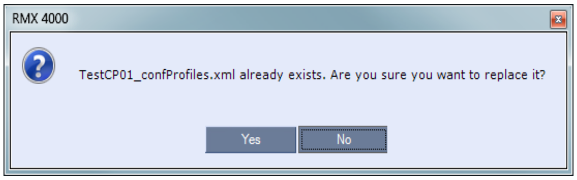
Profiles file name: _confProfiles.xml

4 In the *Export Path* field, click **Browse** to navigate to the location of the desired path where you want to save the exported file.

5 In the *Profiles file name* field, type the file name prefix. The file name suffix (_confProfiles.xml) is predefined by the system. For example, if you type *Profiles01*, the exported file name is defined as *Profiles01_confProfiles.xml*.

6 Click **OK** to export the *Conference Profiles* to a file.

If the export file with the same file name already exists, a prompt is displayed.



RMX 4000

TestCP01_confProfiles.xml already exists. Are you sure you want to replace it?

7 Click **Yes** to replace the exported file or click **No** to cancel the export operation and return to the *Conference Profiles* list. You can modify the export file name and restart the export operation.

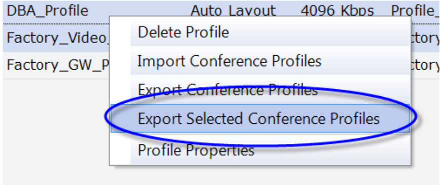
Exporting Selected Conference Profiles

You can select a single Conference Profile or multiple Conference Profiles and export them to a file to be imported to other MCUs in your environment.

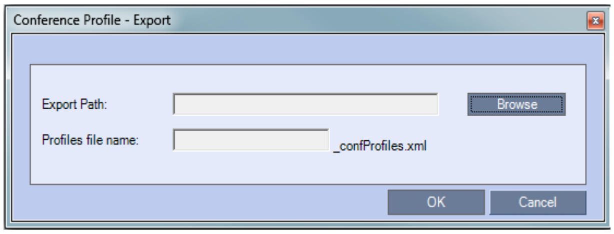
To export selected Conference Profiles:

1 In the *Conference Profiles* pane, select the profiles you want to export.

2 Right-click the selected *Conference Profiles*, and then click **Export Selected Conference Profiles**.



The *Conference Profile - Export* dialog box is displayed.

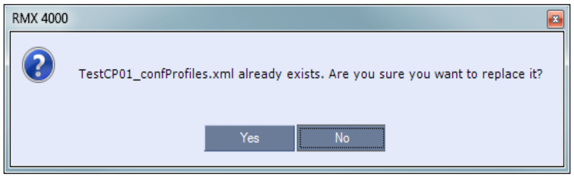


3 In the *Export Path* field, click **Browse** to navigate to the location of the desired path where you want to save the exported file.

4 In the *Profiles file name* field, type the file name prefix. The file name suffix (_confProfiles.xml) is predefined by the system. For example, if you type *Profiles01*, the exported file name is defined as *Profiles01_confProfiles.xml*.

5 Click **OK** to export the *Conference Profiles* to a file.

If the export file with the same file name already exists, a prompt is displayed.



TestCP01_confProfiles.xml already exists. Are you sure you want to replace it?

6 Click **Yes** to replace the exported file or click **No** to cancel the export operation and return to the *Conference Profiles* list. You can modify the export file name and restart the export operation.

Importing Conference Profiles

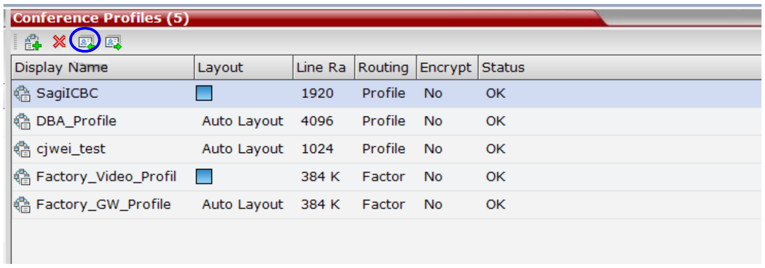
You can import Conference Profiles from another MCU in your environment.

To import Conference Profiles:

1 In the *RealPresence Collaboration Server Management* pane, expand the *Rarely Used* list.

2 Click the **Conference Profiles** button.

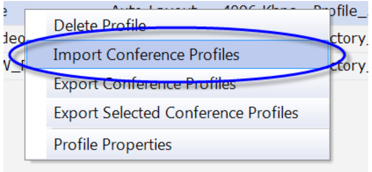
The *Conference Profiles* are displayed in the *List* pane.



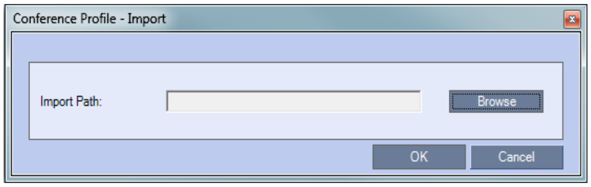
Conference Profiles (5)

Display Name	Layout	Line Ra	Routing	Encrypt	Status
SagiICBC	🟦	1920	Profile	No	OK
DBA_Profile	Auto Layout	4096	Profile	No	OK
cjwei_test	Auto Layout	1024	Profile	No	OK
Factory_Video_Profil	🟦	384 K	Factor	No	OK
Factory_GW_Profile	Auto Layout	384 K	Factor	No	OK

3 Click the **Import Conference Profiles** button or right-click the Conference Profiles pane, and then click **Import Conference Profiles**.


Delete Profile
Import Conference Profiles
Export Conference Profiles
Export Selected Conference Profiles
Profile Properties

The *Conference Profile - Import* dialog box is displayed.



Conference Profile - Import

Import Path:

4 In the *Import Path* field, click **Browse** to navigate to the path and file name of the exported *Conference Profiles* you want to import.

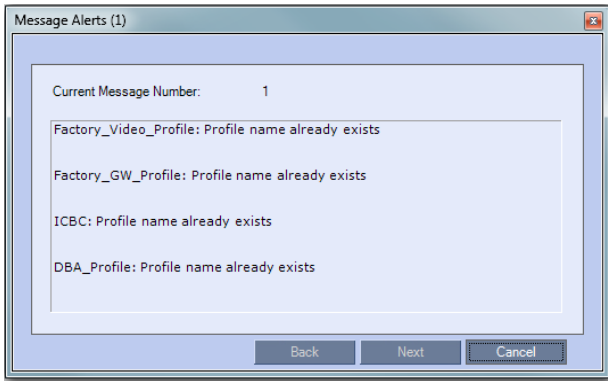
5 Click **OK** to import the *Conference Profiles*.

Conference Profiles are not imported when:

— A *Conference Profile* already exists

— An IVR Service does not exist for the related *Conference Profile*

When *Conference Profiles* are not imported into the *Conference Profiles* list, a Message Alert window is displayed with the profiles that were not imported.



Message Alerts (1)

Current Message Number: 1

Factory_Video_Profile: Profile name already exists

Factory_GW_Profile: Profile name already exists

ICBC: Profile name already exists

DBA_Profile: Profile name already exists

Conference Profiles that are not problematic are imported.

6 Click **Cancel** to exit the *Message Alerts* window.

The imported *Conference Profiles* appear in the *Conference Profiles* list.

3

Additional Conferencing Information

In the *RealPresence CloudAxis Solution*, the conferencing parameters are defined in the RealPresence DMA system component and should not be defined directly in the RealPresence Collaboration Server Virtual Edition component.

Various conferencing modes and video features require additional settings, such as system flag settings, conference parameters and other settings. In depth explanations of these additional settings are described in the following sections:

- "*Content Sharing"* on page **3-1**
- "*Video Preview (AVC Only Participants)"* on page **3-20**
- "*Media Encryption"* on page **3-26**
- "*Packet Loss Compensation (LPR and DBA) AVC CP Conferences"* on page **3-34**
- "*Lecture Mode (AVC CP Only)"* on page **3-38**
- "*Lecture Mode (AVC CP Only)"* on page **3-38**
- "*Audio Algorithm Support"* on page **3-45**
- "*Permanent Conference"* on page **3-49**

Content Sharing

Content such as graphics, presentations, documents or live video can be shared with conference participants using the H.239 (H.323) or BFCP (SIP) protocol, which is the standard protocols for content sharing.

H.239 Protocol

The *H.239* protocol allows compliant endpoints to transmit and receive two simultaneous video streams:

- **People video stream** – video is displayed in Continuous Presence conferences or in Media Relay (SVC) conferencing Mode
- **Content video stream** – Video Switching mode for content sharing

By default, all conferences, *Entry Queue*s, and *Meeting Rooms* launched on the *Collaboration Server* have *H.239* capabilities.

To view *Content*, endpoints must use the same Bit Rate, Protocol, and Resolution. Endpoints may not send *Content* while connecting to an *Entry Queue*.

Endpoints without *H.239* capability can connect to the video conference without *Content*.

Cascade links declare *H.239* capabilities and they are supported in *Star* cascading topologies. For more details, see "*Basic Cascading"* on page **5-3**.

SIP BFCP Content Capabilities

SIP Clients supporting *BFCP* over *UDP*, when connected to conferences on the Collaboration Server, can share *Content* with endpoints supporting the following *Content* sharing protocols:

- *BFCP/TCP*
- *BFCP/UDP*
- *H.323/ H.239*
- *H.323 /Polycom People+Content*

Guidelines

For *SIP Clients* that support both *BFCP/TCP* and *BFCP/UDP*:

- The preferred protocol is *BFCP/UDP*.
- When used in *Cascading* conferences, the *Cascade Link* must be *H.323*.
- *BFCP/UDP* is supported in both *IPv4* and *IPv6* addressing modes.
- *BFCP* utilizes an unsecured channel (port 60002/TCP) even when *SIP TLS* is enabled. If security is of higher priority than *SIP* content sharing, *SIP People+Content* can be disabled. To do this manually add the **ENABLE_SIP_PEOPLE_ PLUS_CONTENT** *System Flag* to the *System Configuration* and set its value to **NO**.
- *SIP People+Content* and *BFCP* capabilities are by default declared to all endpoints. If, however, the endpoint identity is hidden by a proxy server, these capabilities will not be declared by the Collaboration Server. Capabilities declaration is controlled by the **ENABLE_SIP_PPC_FOR_ALL_USER_AGENT** *System Flag*.

  The default value of the **ENABLE_SIP_PPC_FOR_ALL_USER_AGENT** *System Flag* is **YES** resulting in *BFCP* capability being declared with all vendors' endpoints unless it is set to **NO**. When set to **NO**, the Collaboration Server will declare *SIP People+Content* and *BFCP* capabilities to *Polycom* and *Avaya* endpoints.
- The **ENABLE_FLOW_CONTROL_REINVITE** *System Flag* should be set to **NO** when *SIP BFCP* is enabled.
- If these *System Flags* don't exist in the system, they must be manually added. For more information see "*Modifying System Flags"* on page **20-1**.
- BFCP capabilities are not supported in Microsoft ICE environment.

Dial-out Connections:

- For dial-out connections to *SIP Clients*, *BFCP/UDP* protocol can be given priority by adding the adding the **SIP_BFCP_DIAL_OUT_MODE** *System Flag* to *system.cfg* and setting its value to UDP.

The *Collaboration Server’ s Content* sharing determined by the *System Flag’s* settings and *SIP Client* capabilities are summarized in Table 3-1.

Table 3-1 *System Flag - SIP_BFCP_DIAL_OUT_MODE*

Flag Value	**SIP Client: BFCP Support**		
	UDP	**TCP**	**UDP and TCP**
AUTO (Default)	BFCP/**UDP** selected as *Content* sharing protocol.	BFCP/**TCP** selected as *Content* sharing protocol.	BFCP/**UDP** selected as *Content* sharing protocol.
UDP		Cannot share *Content*.	
TCP	Cannot share *Content*.	BFCP/**TCP** selected as *Content* sharing protocol.	

For more information see "*Manually Adding and Deleting System Flags"* on page **20-12**.

Dial-in Connections:

- The Collaboration Server will share content with *Dial-in SIP Clients* according to their preferred *BFCP* protocol.
- *SIP Clients* connected as *Audio Only* cannot share *Content*.

Defining Content Sharing Parameters for a Conference

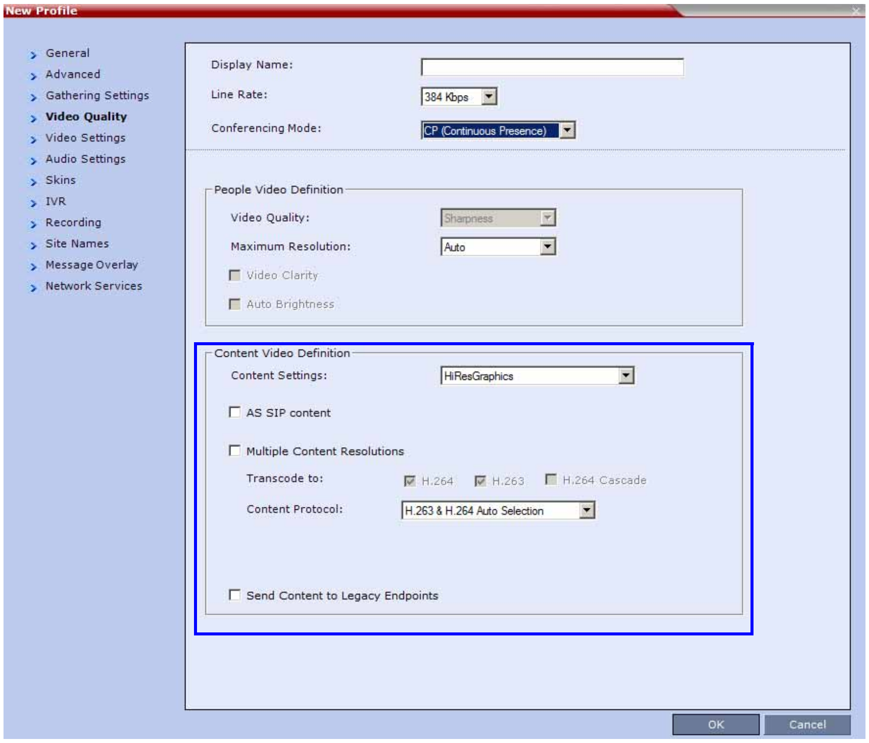
Content can be shared using two main methods:

- Content Highest common parameters (also known as Content Video Switching)
- Multiple Content Resolutions (not supported with RealPresence Collaboration Server Virtual Edition)

In ***Content Video Switching*** mode, the content is negotiated to highest common capabilities supported by the endpoints connected to the conference. If the conference includes participants that support lower content capabilities (such as H.263) and higher content capabilities (H.264), content will be sent at the lower capabilities supported by all endpoints, resulting in lower content quality seen by all endpoints.

In this mode, the content is set according to the capabilities of all the participants currently connected to the conference. If the all the connected participant support the H.264 protocol, the Content will be started with H.264 capabilities. If then an endpoint supporting only H.263 protocol connects, the content is stopped in order to switch to H.263 and has to be resent. If the H.263 participant leaves the conference and only H.264 capable endpoints remain connected, content is stopped in order to switch back to H.264 and has to be resent.

Content parameters are defined in the *Conference Profiles - Video Quality* dialog box. The parameters change according to the *Conferencing Mode*.

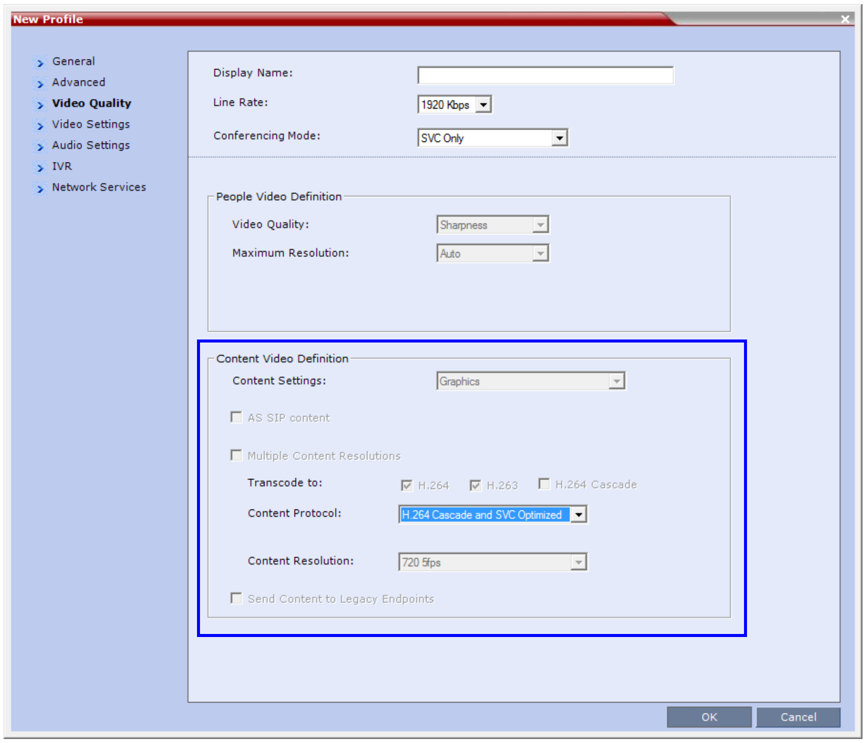


Display Name:
Line Rate: 384 Kbps
Conferencing Mode: CP (Continuous Presence)

People Video Definition
Video Quality: Sharpness
Maximum Resolution: Auto
□ Video Clarity
□ Auto Brightness

Content Video Definition
Content Settings: HiResGraphics
□ AS SIP content
□ Multiple Content Resolutions
Transcode to: ☑ H.264 ☑ H.263 □ H.264 Cascade
Content Protocol: H.263 & H.264 Auto Selection
□ Send Content to Legacy Endpoints

AVC CP Conferencing Mode



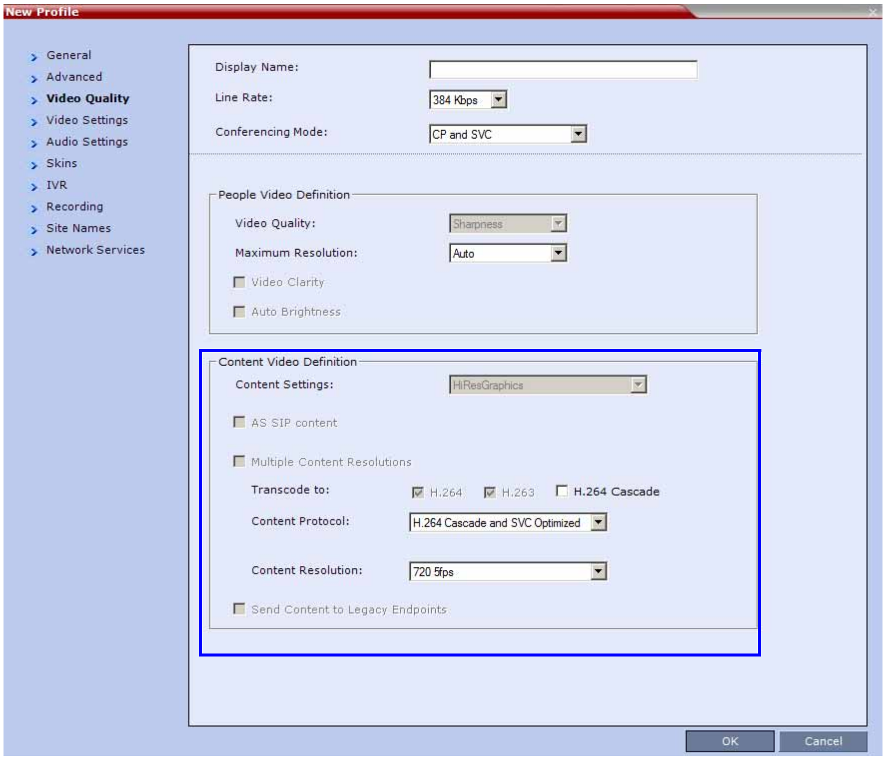
Display Name:
Line Rate: 1920 Kbps
Conferencing Mode: SVC Only

People Video Definition
Video Quality: Sharpness
Maximum Resolution: Auto

Content Video Definition
Content Settings: Graphics
□ AS SIP content
□ Multiple Content Resolutions
Transcode to: ☑ H.264 ☑ H.263 □ H.264 Cascade
Content Protocol: H.264 Cascade and SVC Optimized
Content Resolution: 720 5fps
□ Send Content to Legacy Endpoints

SVC Conferencing Mode

Mixed CP and SVC Conferencing Mode



Display Name:

Line Rate: 384 Kbps

Conferencing Mode: CP and SVC

People Video Definition

Video Quality: Sharpness

Maximum Resolution: Auto

□ Video Clarity

□ Auto Brightness

Content Video Definition

Content Settings: HiResGraphics

□ AS SIP content

□ Multiple Content Resolutions

Transcode to: ☑ H.264 ☑ H.263 □ H.264 Cascade

Content Protocol: H.264 Cascade and SVC Optimized

Content Resolution: 720 5fps

□ Send Content to Legacy Endpoints

1 In the *Content Video Definition* section, select the *Content Settings* and *Protocol* as follows:

Table 3-2 *H.239 Content Options*

Field	Description
Content Settings	Select the transmission mode for the Content channel: • ***Graphics*** — basic mode, intended for normal graphics • ***Hi-res Graphics*** (AVC CP Only) — a higher bit rate intended for high resolution graphic display • ***Live Video*** (AVC CP Only) — Content channel displays live video • ***Customized Content Rate*** (AVC CP Only) **-** manual definition of the Conference Content Rate**,** mainly for cascading conferences. Selection of a higher bit rate for the *Content* results in a lower bit rate for the people channel. For a detailed description of each of these options, see "*Content Sharing Parameters in Content Highest Common (Content Video Switching) Mode"* on page **3-6**.
AS-SIP Content	This Content Sharing option is not supported with RealPresence Collaboration Server Virtual Edition.
Multiple Resolutions	This Content Sharing option is not supported with RealPresence Collaboration Server Virtual Edition.

Table 3-2 H.239 Content Options (Continued)

Field	Description
Content Protocol	• **H.263** (AVC CP Only) • *Content* is shared using the *H.263* protocol. • Use this option when most of the endpoints support *H.263* and some endpoints support *H.264*. • **H.263 & H.264 Auto Selection** (AVC CP Only) • *Content* is shared using *H.263* if a mix of H.263-supporting and *H.264*-supporting endpoints are connected. • *Content* is shared using *H.264* if all connected endpoints have *H.264* capability. • **H.264 HD** (AVC CP Only, default) • Ensures high quality *Content* when most endpoints support *H.264* and *HD Resolutions*. • **H.264 Cascade and SVC Optimized** • All *Content* is shared using the *H.264* content protocol and is optimized for use in *SVC only* and *Cascaded Conferences*. For a detailed description of each of these settings, see "*Content Protocols"* on page 3-8.
Content Resolution	Select a *Content Resolution* from the drop-down menu. The *Content Resolutions* that are available for selection are dependent on the content sharing mode (Highest Common Content or Multiple Content Resolutions), *Line Rate* and *Content Settings* that have been selected for the conference. For a full list of *Content Resolutions* see "*Defining Content Sharing Parameters for a Conference"* on page 3-3. **Note:** This field is displayed only when ***H.264 Cascade** is selected for Multiple Content Resolution or when **H.264 Cascade and SVC Optimized*** option is selected as the *Content Protocol* (in the Highest Common Content Mode) and is enabled for selection in CP conferences (AVC CP Only). This option is disabled in SVC conferences.
Send Content to Legacy Endpoints	This Content Sharing option is not supported with RealPresence Collaboration Server Virtual Edition.

2 Click **OK.**

Content Sharing Parameters in Content Highest Common (Content Video Switching) Mode

This section describes the possible content sharing parameters when content Highest Common mode is used:

- Content Settings
- Content Protocol
- Content Resolution

Content Settings

The Content channel can transmit one of the following modes:

- **Graphics** – for standard graphics. This is the default mode in AVC conferences and the only supported mode for SVC conferences.
- **Hi-res Graphics** (AVC CP Only) – requiring a higher bit rate, for high quality display or highly detailed graphics.
- **Live Video** (AVC CP Only) – highest bit rate, for video clips or live video display.
- **Customized Content Rate** (AVC CP Only) - that allows manual definition of the *Conference Content Rate*.

AVC CP Content Setting

For *Graphics*, *Hi-res Graphics* and *Live Video*, the highest common Content bit rate is calculated for the conference each time an endpoint connects. Therefore, if an endpoint connects to an ongoing conference at a lower bit rate than the current bit rate, the Content bit rate for the current conference is re-calculated and decreased.

Bit rate allocation by the MCU is dynamic during the conference and when the Content channel closes, the video bit rate of the *People conference* is restored to its maximum.

During a conference the MCU will not permit an endpoint to increase its bit rate, it can however change its Content resolution. The Collaboration Server can decrease the allocated Content bit rate during a conference.

The following table summarizes the bit rate allocated to the Content channel from the video channel in each of the *Content Settings* according to the conference line rate:

Table 3-3 Decision Matrix - Bit Rate Allocation to Content Channel per Conference Line Rate

Content Settings	Content Bit Rate Allocation per Conference Line Rate (kbps)								
	64 96	128	256	384	512	768 832	1024 1152	1472 1728	1920
Graphics		64	64	128	128	256	256	256	256
Hi Resolution Graphics		64	128	192	256	384	384	512	768
Live Video		64	128	256	384	512	768	768	1152
Customized Content Rate	The Content Bit Rate is selected from a menu in the *Content Video Definition* pane. See "*Selecting a Customized Content Rate in AVC CP Conferences"* on page 3-13.								

Table 3-4 summarizes the *Maximum Resolution* of *Content* and *Frames per Second (fps)* for *Bit Rate Allocations* to the *Content Channel*.

Table 3-4 *Content - Maximum Resolution, Frames/Second per Bit Rate Allocation*

Bit Rate Allocated to Content Channel (Kbps)	Content	
	Maximum Resolution	**Frames/Second**
*From **64** and less than **512***	H.264 HD720p	5
*From **512** and up to **1536***	H.264 HD720p	30

SVC Only and Mixed CP and SVC Content Setting

The Content channel is transmitted in **Graphics** mode only.

Content Protocols

Two *Content Protocols* can be used for sharing content:

- H.263 (CP and mixed CP and SVC)
- H.264 (all conferencing modes)

H.264 offers higher quality content, but is not supported by legacy endpoints. Depending on the endpoints capabilities, you can determine the content sharing experience by selecting the appropriate protocol and system behavior from the *Content Protocol* list:

- *H.263 & H.264 Auto Selection* (*AVC CP Only)*
- *H.263 (AVC CP Only)*
- *H.264 HD (AVC CP Only)*
- *H.264 Cascade and SVC Optimized* (all conferences)

H.263 & H.264 Auto Selection (AVC CP Conferences)

The **H.263 & H.264 Auto Selection** option should be selected when *Content* is to be shared using a mix of *H.263*-supporting and *H.264*-supporting endpoints.

Bit rate allocation to the *Content* channel by the Collaboration Server is dynamic according to the conference line rate and *Content Setting* selected for the conference.

If an endpoint that supports only *H.263* for Content Sharing connects to a conference with Content Protocol set to *H.263 & H.264 Auto Selection*:

- *Content* is shared using *H.263* even if *H.264*-supporting endpoints are connected.
- *Content* is shared using *H.264* if all connected endpoints have *H.264* capability.
- If the first endpoint to connect to the conference only supports *H.263,* the *H.263* protocol is used for *Content* for all conference participants.
- If *Content* is already being shared using the *H.264* protocol when a *H.263* endpoint connects, *Content* sharing is stopped and must be manually restarted using *H.263* (i.e. the endpoint using H.263 Content Protocol must connect first), for all participants to receive content. If the *H.263* endpoint disconnects, *Content* sharing must be manually stopped and restarted and will automatically upgrade to the *H.264* protocol.
- Endpoints that do not have at least *H.263* capability can connect to the conference but cannot share *Content*.
- This option is not available in *SVC Conferencing Mode* and *CP and SVC Conferencing Mode*.

H.263 (AVC CP Conferences)

Select this option when most of the endpoints support *H.263* and some endpoints support *H.264*. In such a case, all endpoints will share content using the H.263 protocol, and this protocol will not change throughout the conference (fixed mode).

Bit rate allocation to the *Content* channel by the Collaboration Server is dynamic according to the conference line rate and *Content Settings* selected for the conference. For more information see "*Content Sharing Parameters in Content Highest Common (Content Video Switching) Mode"* on page **3-6**.

This option is not available in *SVC Conferencing Mode* and *CP and SVC Conferencing Mode*.

H.264 HD (AVC CP Conferences default)

The **H.264 HD** option should be selected only if most endpoints in the conference support *H.264* to ensure high quality *Content*.

When this protocol option is selected, endpoints must connect at *Content* bit rates above a minimum as specified by specific *System Flags* to ensure high quality *Content* for all participants. For more information about *System Flags* see "*Setting the Minimum Content Rate for Each Content Quality Setting for H.264 HD"* on page **3-9**.

Bit rate allocation to the *Content* channel by the Collaboration Server is dynamic according to the conference line rate and *Content Setting* selected for the conference. For more information see "*Content Sharing Parameters in Content Highest Common (Content Video Switching) Mode"* on page **3-6**.

Endpoints that do not support *H.264*, or those that do not meet the minimum line rate threshold for the *Content Setting* will not receive content.

Guidelines

- Only endpoints that support *H.264* capability at a resolutions of *HD720p5* or higher will be able to receive and send *Content*.
- This option is not available in *SVC Conferencing Mode* and *CP and SVC Conferencing Mode*.
- Maximum supported content resolution is HD 720p.
- The minimum *Content Rate* required for allowing a participant to share *Content* is the lower valued parameter when comparing the *System Flag* setting *(*Table 3-5*)* and the *content bit rate allocation* derived from the conference line rate (Table 3-6*)*.

  When the flag settings enable an endpoint to share Content at a content rate that is lower than the conference content rate (Table 3-6), the content rate of the entire conference is reduced to the content rate supported by that endpoint.

Setting the Minimum Content Rate for Each Content Quality Setting for H.264 HD

The following *System Flags* determine the minimum content rate required for endpoints to share *H.264* high quality content via the *Content* channel.

A *System Flag* determines the minimum line rate for each *Content Setting*:

- *Graphics*
- *Hi Resolution Graphics*
- *Live Video*

In order to change the *System Flag* values, the flags must be manually added to the *System Configuration*. For more information see "*Modifying System Flags"* on page **20-1**.

Table 3-5 *H.264 HD System Flags*

Content Settings	Flag Name	Range	Default
Graphics	*H264_HD_GRAPHICS_MIN_CONTENT_RATE*	0-1536	128
Hi Resolution Graphics	*H264_HD_HIGHRES_MIN_CONTENT_RATE*	0-1536	256
Live Video	*H264_HD_LIVEVIDEO_MIN_CONTENT_RATE*	0-1536	384

Example

Table 3-6 summarizes an example of two participants trying to connect to a conference running at a *Line Rate* of 1024Kbps. The *Content Setting* for the conference is **Hi Resolution Graphics** and the **H264_HD_HIGHRES_MIN_CONTENT_RATE** *System Flag* setting are used to determine if *Content* will be shared with the participant.

Table 3-6 *Participant Content Sharing Based on Connection Line Rate and System Flag Setting*

	Participant		Conference		Flag Value	Result
	Line Rate	Bit Rate Allocation to Content Channel (Table 3-5)	Line Rate	Bit Rate Allocation to Content Channel (Table 3-5)		
Participant 1	384	192	1024	384	**128**	Participant and entire conference share content at 192Kbps
					512	Participant will not receive content
Participant 2	1024	384			**128**	Participant and entire conference share content at 384Kbps
					512	Participant and entire conference share content at 384Kbps

H.264 Cascade and SVC Optimized

The **H.264 Cascade and SVC Optimized** option maintains content quality and minimizes the amount of content refreshes that occur in large cascading conferences when participants connect or disconnect from the conference.

This option is the only available option automatically selected in *SVC Only conferencing Mode* and mixed *CP and SVC Conferencing Mode*.

The *H.264 Cascade and SVC Optimized* option uses fixed resolution and frame rate for *SVC Only* and mixed *CP and SVC* conferences. In AVC CP conferences, each content *Line Rate* and *Content Setting* has its own resolution and frame rate as summarized in Table 3-4.

In AVC CP conferences, endpoints that do not support the required content parameters (*Content* line rate, *H.264* protocol and *Content Resolution)*, will not receive content.

Endpoints that do not support the required content parameters (*Content* line rate and *Content Resolution)* cannot share content.

Guidelines

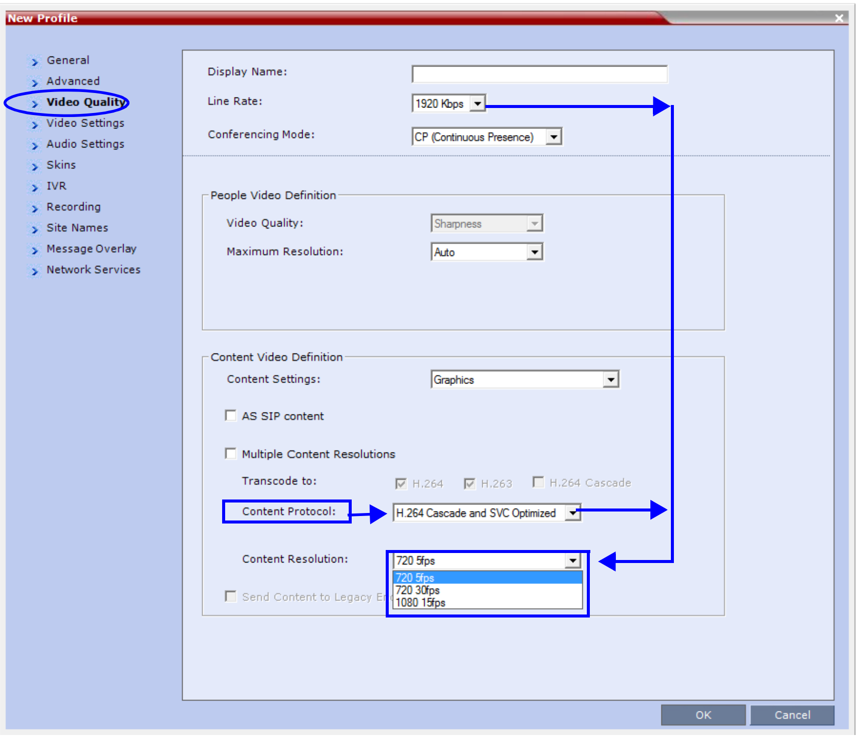
- In Cascading conferences, the cascade link must be *H.323.*
- This is the only available Content sharing mode in *SVC Conferencing Mode* and *CP and SVC Conferencing Mode*.
- *H.264 High Profile* is not supported.

Enabling H.264 Cascade and SVC Optimized Content Sharing in AVC CP Conferences

When *H.264 Cascade and SVC Optimized* is selected as the *Content Protocol*, an additional field, *Content Resolution* is displayed in the *Content Video Definition* pane.

In *SVC Conferencing Mode* and *CP and SVC Conferencing Mode,* the *Content Resolution* option is disabled.

The *Content Resolution* is a fixed resolution and frame rate for *Content* sharing in a Cascaded Conference. The *Content Resolution* that are available for selection are dependent on the *Line Rate* and *Content Settings* that have been selected for the conference.



Display Name:

Line Rate: 1920 Kbps

Conferencing Mode: CP (Continuous Presence)

People Video Definition

Video Quality: Sharpness

Maximum Resolution: Auto

Content Video Definition

Content Settings: Graphics

☐ AS SIP content

☐ Multiple Content Resolutions

Transcode to: ☑ H.264 ☑ H.263 ☐ H.264 Cascade

Content Protocol: H.264 Cascade and SVC Optimized

Content Resolution: 720 5fps

720 5fps
720 30fps
1080 15fps

☐ Send Content to Legacy E

CP Conferencing Mode

The following table summarizes the interaction of these parameters.

The selection of the appropriate *Content Resolution* option, when several options are available, should be based on the line rate and capabilities that can be used by most or all endpoints connecting to the conference.

Examples:

- If the conference *Line Rate* is **1024** kbps.

  and

- If the *Content Settings* selection is **Graphics**.

  — **Content Resolutions** of **HD720/5** and **HD1080/15** are selectable with **256** kbps and **768** kbps allocated as the *Conference Content Rate* respectively.

  The higher *Content Resolution,* **HD1080/15** should be selected only if most of the endpoints connecting to the conference can support a *Content Rate* of 768Kbps, which requires the participant to connect to the conference at a *Line Rate* of 1024kbps.

When the lower *Content Resolution* **HD720/5** is selected, the conference *Content Rate* is set to 256 kbps. This will enable the endpoints that connect to the conference at a *Line Rate* of at least 768 kbps to receive content in the Content channel. Endpoints that connect to the conference at a line rate lower than 768Kbps, will not receive content.

- If the *Content Settings* selection is **Hi Resolution Graphics**.
  - — Only **HD720/5** can be selected as the *Content Resolution* with **384** kbps allocated as the conference *Content Rate*.

    Only endpoints that connect at a *Line Rate* of 768 kbps that is required to support a *Content Rate* of 384 kbps will receive content in the Content channel. Endpoints that connect to the conference at a line rate lower than 768 kbps, will not receive content.
- If the *Content Settings* selection is **Live Video**.

  HD720/5, **HD720/30** or **HD1080/15** can be selected as the *Content Resolution* with **768** kbps allocated the as the *Conference Content Rate.* The higher *Content Resolution* should be selected according to the resolution capabilities of the majority of the endpoints connecting to the conference. Endpoints that cannot support the selected *Content Resolution* are considered *Legacy Endpoints* and will not receive content.

Selecting a Customized Content Rate in AVC CP Conferences

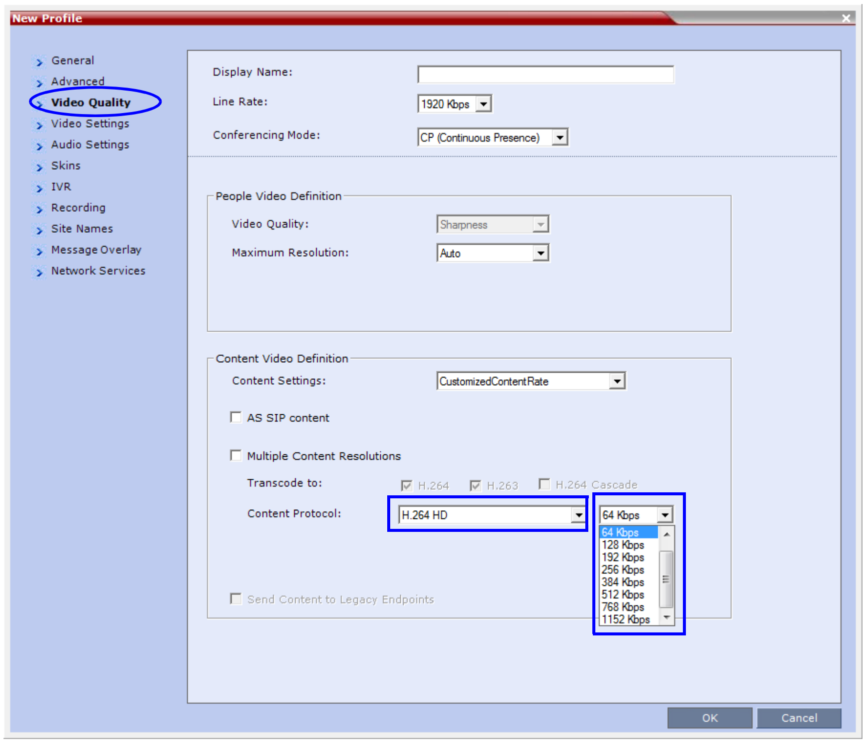
Customized Content Rate functionality can be implemented when a *Conference Content Rate*, that is automatically calculated by the Collaboration Server, may not be suitable in a *Cascaded Environment*, where conference line rates may vary widely between the cascaded conferences. For example, one conference may have a line rate of 2 Mbps, and the other a line rate of is 512kbps.

Guidelines:

- Cascaded conferences may have different *Conference Line Rates*.
- The *Customized Content Rate* must be the same for all cascaded conferences.

To Select the Customized Content Rate:

Customized Content Rate is enabled in the *Profile - Video Quality* dialog box.



Display Name:

Line Rate: 1920 Kbps

Conferencing Mode: CP (Continuous Presence)

People Video Definition

Video Quality: Sharpness

Maximum Resolution: Auto

Content Video Definition

Content Settings: CustomizedContentRate

□ AS SIP content

□ Multiple Content Resolutions

Transcode to: ☑ H.264 ☑ H.263 □ H.264 Cascade

Content Protocol: H.264 HD 64 Kbps

□ Send Content to Legacy Endpoints

1 In the *Content Settings* list, select **Customized Content Rate**.
When selected, a drop-down menu of the available *Conference Content Rates* is displayed. These *Content Line Rates* are based on and will vary according to the selected *Conference Line Rate*.

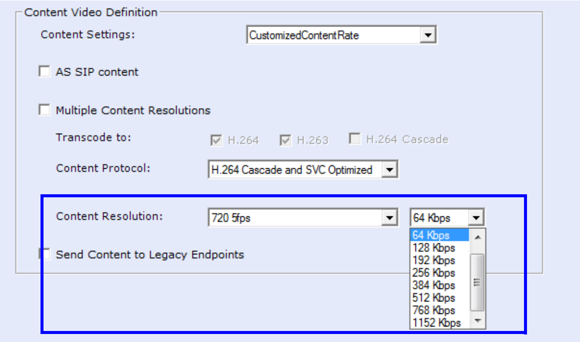
The largest selectable *Content Line Rate* is 66% of the *Conference Line Rate*.

If the *Conference Line Rate* is 64kbps or 96kbps, the only available *Conference Content Rate* is **0***,* indicating that *Content* is not supported at these rates.

2 Select the required content rate.

When selecting a *Conference Line Rate* (after selecting *Customized Content Rate)* that is too low for the selected *Customized Content Rate*, the following error message is displayed:
"The selected content line rate should be modified. To update content line rate press Cancel. To return to automatic mode (Graphics) press OK."
You can then modify either the *Content Line Rate* or the *Conference Line Rate* or select another *Content Setting* option.

3 If **H.264 Cascade and SVC Optimized** is the selected *Content Protocol*, a **Content Resolution** must be selected.



Content Video Definition

Content Settings: CustomizedContentRate

□ AS SIP content

□ Multiple Content Resolutions

Transcode to: ☑ H.264 ☑ H.263 □ H.264 Cascade

Content Protocol: H.264 Cascade and SVC Optimized

Content Resolution: 720 5fps 64 Kbps

□ Send Content to Legacy Endpoints

Table 3-7 lists the *Cascade Resolutions* available for the various *Conference Content Rates*.

Table 3-7 *H.264 Cascade and SVC Optimized - Cascade Resolutions*

H.264 Cascade Optimized		
Conference Content Rate (Kbps)	**Available Resolutions**[*]	
64	HD720p5 Content Not Supported	
128	HD720p5	
192	HD720p5	
256	HD720p5	
384	HD720p5	
512	HD720p5	HD720p30
768	HD720p5	HD720p30
1152	HD720p5	HD720p30
1536	HD720p5	HD720p30

[*] The default resolution for all *Content Rates* is *HD720p5*.

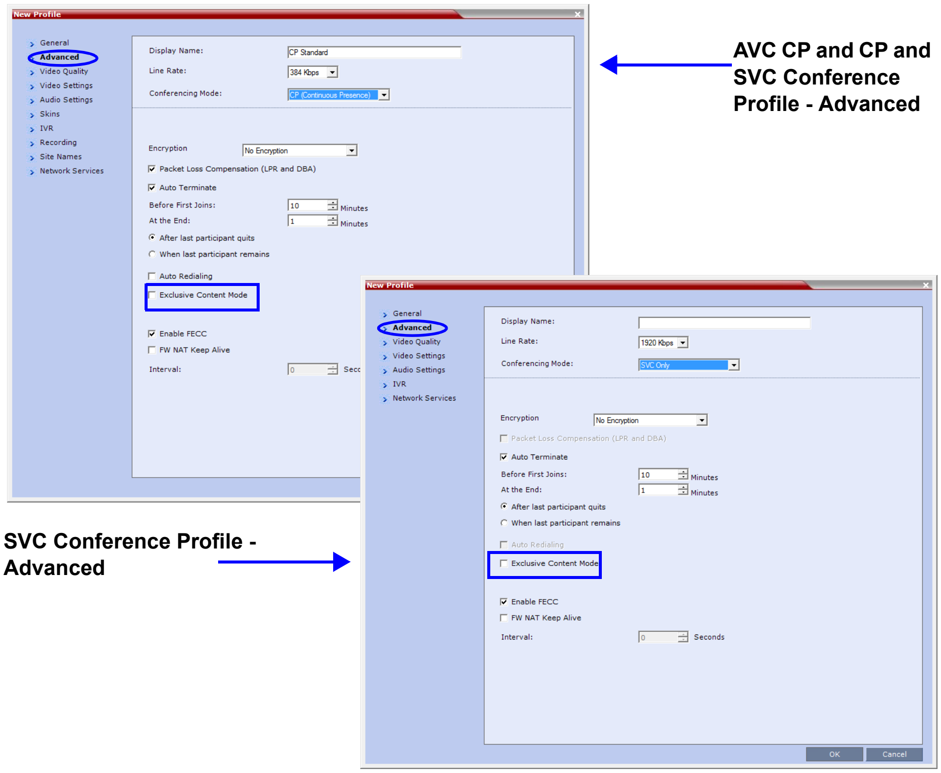
Exclusive Content Mode

Exclusive Content Mode allows you to limit *Content* broadcasting to one participant, preventing other participants from interrupting the *Content* broadcasting while it is active.

Guidelines

- *Exclusive Content Mode* is available in all Conferencing Modes.

- The *Exclusive Content Mode* is enabled or disabled by a check box in the in the *Advanced* tabs of the *Conference Profile*. The check box is cleared (feature is disabled) by default.



AVC CP and CP and SVC Conference Profile - Advanced

New Profile

Display Name: CP Standard
Line Rate: 384 Kbps
Conferencing Mode: CP (Continuous Presence)

Encryption: No Encryption
☑ Packet Loss Compensation (LPR and DBA)
☑ Auto Terminate
Before First Joins: 10 Minutes
At the End: 1 Minutes
● After last participant quits
○ When last participant remains
□ Auto Redialing
□ Exclusive Content Mode
☑ Enable FECC
□ FW NAT Keep Alive
Interval: 0 Seconds

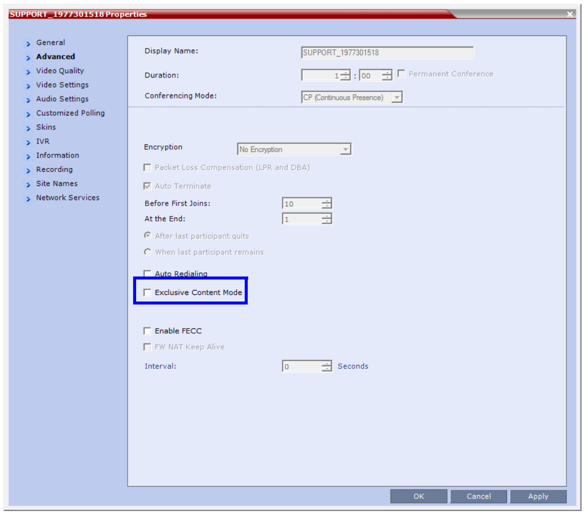
SVC Conference Profile - Advanced

New Profile

Display Name:
Line Rate: 1920 Kbps
Conferencing Mode: SVC Only

Encryption: No Encryption
□ Packet Loss Compensation (LPR and DBA)
☑ Auto Terminate
Before First Joins: 10 Minutes
At the End: 1 Minutes
● After last participant quits
○ When last participant remains
□ Auto Redialing
□ Exclusive Content Mode
☑ Enable FECC
□ FW NAT Keep Alive
Interval: 0 Seconds

- *Exclusive Content Mode* can be enabled or disabled during an ongoing conference using the *Conference Properties - Advanced* dialog box.



SUPPORT_1977301518 Properties

Display Name: SUPPORT_1977301518
Duration: 1 : 00 □ Permanent Conference
Conferencing Mode: CP (Continuous Presence)

Encryption: No Encryption
□ Packet Loss Compensation (LPR and DBA)
☑ Auto Terminate
Before First Joins: 10
At the End: 1
● After last participant quits
○ When last participant remains
□ Auto Redialing
□ Exclusive Content Mode
□ Enable FECC
□ FW NAT Keep Alive
Interval: 0 Seconds

- In *Exclusive Content Mode,* if the RESTRICT_CONTENT_BROADCAST _ TO_LECTURER *System Flag* is set to:
  - — **NO** - the first participant to send content becomes the *Content Token* holder and has to release the *Content Token* before any other participant can acquire the token and begin transmitting *Content*.

  - **YES** - only the designated *Lecturer* can be the *Content Token* holder.
- In *Exclusive Content Mode*, if an endpoint attempts to send *Content* a few seconds after another endpoint sent *Content*, the *Content* stream it is receiving is momentarily interrupted by a slide which is displayed for a few seconds before the normal *Content* stream is resumed.

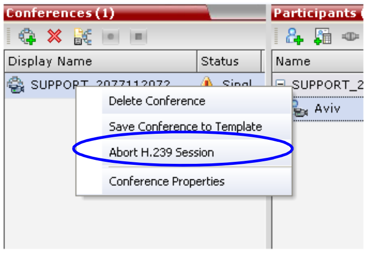
Stopping a Content Session

In some cases, when one participant ends the Content session from his/her endpoint, the Content token is not released and other participants cannot send Content.

The Collaboration Server User can withdraw the Content token from the current holder and to return it to the MCU for assignment to other endpoints.

To end the current Content session:

>> In the *Conferences* list pane, right-click the conference icon and then click **Abort H.239 Session**.



Content Broadcast Control

Content Broadcast Control prevents the accidental interruption or termination of *H.239 Content* that is being shared in a conference.

Content Broadcast Control achieves this by giving *Content Token* ownership to a specific endpoint via the *Collaboration Server Web Client*. Other endpoints are not able to send content until *Content Token* ownership has been transferred to another endpoint via the *Collaboration Server Web Client*.

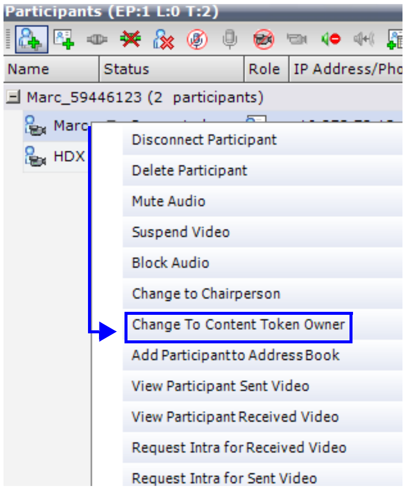
Guidelines

- *Content Broadcast Control* is supported in *CP* conferences.
- *Content Broadcast Control* is supported in H.323 environments.
- Only the selected *Content Token* owner may send content and *Content Token* requests from other endpoints are rejected.
- *Content Token* ownership is valid until:
  - It is canceled by the Collaboration Server User via the *Collaboration Server Web Client*.
  - The owner releases it.
  - The endpoint of the *Content Token* owner disconnects from the conference.
- The Collaboration Server User can cancel *Content Token* ownership.
- In cascaded conferences, a participant functioning as the cascade link cannot be given token ownership.

Giving and Cancelling Token Ownership (AVC Participants)

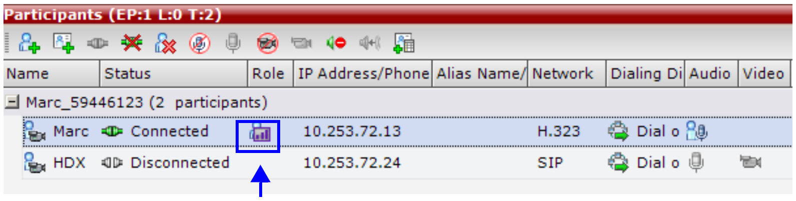
To give token ownership:

1 In the Participants list, right click the AVC-enabled endpoint that is to receive Content Token ownership.


Participants (EP:1 L:0 T:2)
Name
Status
Role
IP Address/Pho
Marc_59446123 (2 participants)
Marc
HDX
Disconnect Participant
Delete Participant
Mute Audio
Suspend Video
Block Audio
Change to Chairperson
Change To Content Token Owner
Add Participantto Address Book
View Participant Sent Video
View Participant Received Video
Request Intra for Received Video
Request Intra for Sent Video

2 Select **Change To Content Token Owner** in the drop-down menu.

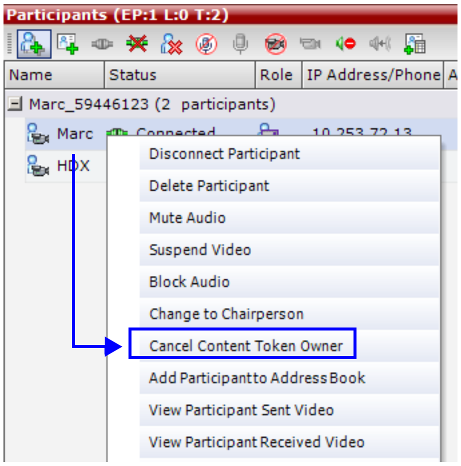
The endpoint receives ownership of the *Content Token* and an indication icon is displayed in the Role column of the participant's entry in the Participants list.



Name	Status	Role	IP Address/Phone	Alias Name/	Network	Dialing Di	Audio	Video
Marc_59446123 (2 participants)								
Marc	Connected		10.253.72.13		H.323	Dial o		
HDX	Disconnected		10.253.72.24		SIP	Dial o		

To cancel token ownership:

1 In the *Participants* list, right click the endpoint that currently has *Content Token* ownership.


Participants (EP:1 L:0 T:2)
Name
Status
Role
IP Address/Phone
Marc_59446123 (2 participants)
Marc
Connected
10.253.72.13
HDX
Disconnect Participant
Delete Participant
Mute Audio
Suspend Video
Block Audio
Change to Chairperson
Cancel Content Token Owner
Add Participantto Address Book
View Participant Sent Video
View Participant Received Video

2 Select **Cancel Content Token Owner** in the drop-down menu**.**
Content Token ownership is cancelled for the endpoint.

Managing Noisy Content Connections

The system can identify participants who send frequent requests to refresh their Content display usually as a result of a problematic network connection. The frequent refresh requests cause frequent refresh of the Content display and degrade the viewing quality.

When the system identifies the noisy participants, the system will automatically suspend the requests to refresh the sent Content to avoid affecting the quality of the Content viewed by other conference participants. This process is controlled by System flags.

Content Display Flags

- **MAX_INTRA_REQUESTS_PER_INTERVAL_CONTENT**

  Enter the maximum number of refresh (intra) requests for the Content channel sent by the participant's endpoint in a 10 seconds interval that will be dealt by the Collaboration Server system. When this number is exceeded, the Content sent by this participant will be identified as noisy and his/her requests to refresh the Content display will be suspended.
  Default setting: 3
- **MAX_INTRA_SUPPRESSION_DURATION_IN_SECONDS_CONTENT**

  Enter the duration in seconds to ignore the participant's requests to refresh the Content display.
  Default setting: 10
- **CONTENT_SPEAKER_INTRA_SUPPRESSION_IN_SECONDS**

  This flag controls the requests to refresh (intra) the Content sent from the Collaboration Server system to the Content sender as a result of refresh requests initiated by other conference participants.

  Enter the interval in seconds between the Intra requests sent from the Collaboration Server to the endpoint sending the Content to refresh the Content display. Refresh requests that will be received from endpoints within the defined interval will be postponed to the next interval.
  Default setting: 5

Forcing Other Content Capabilities

- The **H239_FORCE_CAPABILITIES** *System Flag* in *system.cfg* gives additional control over Content sharing:
  - When the flag is set to **NO (default)**, the Collaboration Server only verifies that the endpoint supports the content protocols: *H.263* or *H.264*.
  - When set to **YES**, the Collaboration Server checks frame rate, bit rate, resolution, annexes and all other parameters of the Content mode as declared by an endpoint during the capabilities negotiation phase. If the endpoint does not support the Content capabilities of the MCU, the participant will not be able to send or receive content over a dedicated content channel.

Content Sharing via the Polycom CCS Plug-in for Microsoft Lync Clients

From version 8.1, Polycom CCS (Content Collaboration Solution) Plug-in for Lync clients allows Lync clients to receive and send *Content* on a separate channel, without having to use the video channel. *Content* is transmitted using SIP BFCP.

For more information, see Appendix 3, "*Exclusive Content Mode*"on page **3-15**.

Video Preview (AVC Only Participants)

Collaboration Server users can preview the video sent from the participant to the conference (MCU) and the video sent from the conference to the participant. It enables the Collaboration Server users to monitor the quality of the video sent and received by the participant and identify possible quality degradation.

The video preview is displayed in a separate window independent to the *Collaboration Server Web Client*. All Web Client functionality is enabled and conference and participant monitoring as well as all other user actions can be performed while the video preview window is open and active. Live video is shown in the preview window as long as the window is open. The preview window closes automatically when the conference ends or when participant disconnects from the conference. It can also be closed manually by the Collaboration Server user.

Video Preview Guidelines

- Video preview is available for *AVC* participants. It is not available for *SVC* participants.
- Video preview window size and resolution are adjusted to the resolution of the PC that displays the preview.
- Video Preview of the video sent from the conference to the participant is shown according to the line rate and video parameters of the level threshold to which the participant is connected.
- All users can view a video preview.
- Only one preview window can be displayed for each *Collaboration Server Web Client* connection (workstation).
- Only one preview window can be displayed for a single conference and up to four preview windows can be displayed for each system on different workstations (one per workstation and one per conference).
- Live video that is shown in the preview window does not include the Content when it is sent by the participant.
- Video Preview is supported in cascaded conferences.
- If the video preview window is opened when the IVR slide is displayed to the participant, it will also be displayed in the video preview window.
- Video Preview is supported with *H.264 High Profile.*
- Video Preview is not supported for endpoints using the *RTV* protocol.
- Video Preview is disabled in encrypted conferences.
- Video preview cannot be displayed when the participant's video is suspended.
- Participant's video preview and the CMAD window cannot be open and running simultaneously on the same PC as both require the same DirectDraw resource.

Workstation Requirements

To be able to display the video preview window, the following minimum requirements must be met:

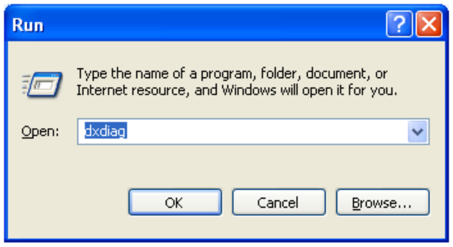
- Windows XP, Windows Vista and Windows 7
- Internet Explorer 7 and later
- DirectX is installed
- DirectDraw Acceleration must be enabled and no other application is using the video resource
- Hardware acceleration must be enabled

Testing your Workstation

To ensure that your workstation can display the video preview window:

1 In Windows, click **Start > Run**.
The *Run* dialog box opens.

2 In the *Open* field, type **dxdiag** and press the **Enter** key or click **OK**.



Type the name of a program, folder, document, or Internet resource, and Windows will open it for you.

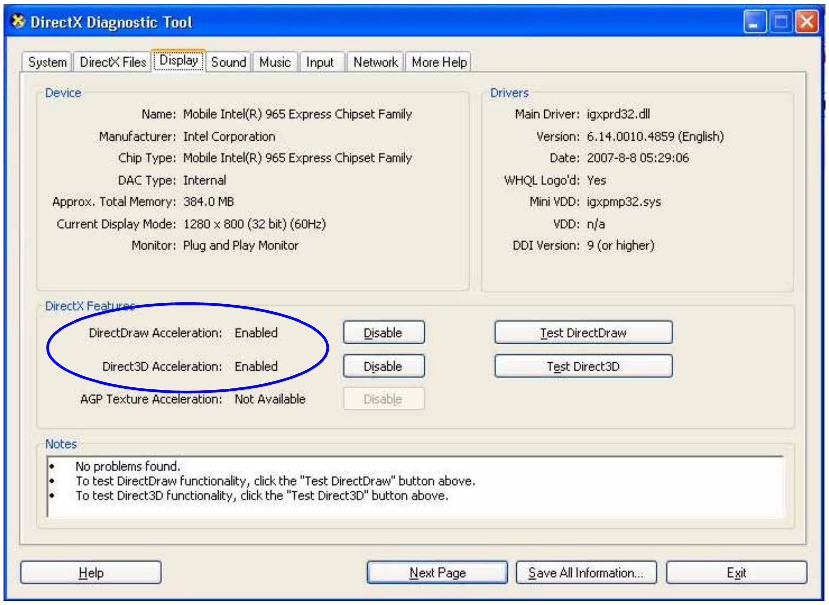
Open: dxdiag

A confirmation message is displayed.

3 Click **Yes** to run the diagnostics.
The *DirectX Diagnostic Tool* dialog box opens.

4 Click the **Display** tab.

To be able to display the video preview window, the **DirectDraw Acceleration** and **Direct3D Acceleration** options must be **Enabled**.



Device

Name: Mobile Intel(R) 965 Express Chipset Family
Manufacturer: Intel Corporation
Chip Type: Mobile Intel(R) 965 Express Chipset Family
DAC Type: Internal
Approx. Total Memory: 384.0 MB
Current Display Mode: 1280 x 800 (32 bit) (60Hz)
Monitor: Plug and Play Monitor

Drivers

Main Driver: igxprd32.dll
Version: 6.14.0010.4859 (English)
Date: 2007-8-8 05:29:06
WHQL Logo'd: Yes
Mini VDD: igxpmp32.sys
VDD: n/a
DDI Version: 9 (or higher)

DirectX Features

DirectDraw Acceleration: Enabled
Direct3D Acceleration: Enabled
AGP Texture Acceleration: Not Available

Notes

- No problems found.
- To test DirectDraw functionality, click the "Test DirectDraw" button above.
- To test Direct3D functionality, click the "Test Direct3D" button above.

If the video card installed in the PC does not support DirectDraw Acceleration, a black window may be viewed in the Video Preview window.

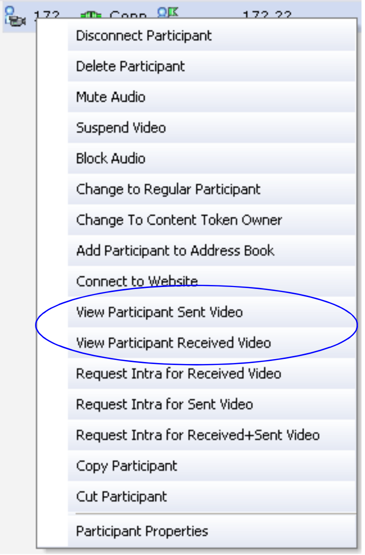
5 Click the **Exit** button.

Previewing the Participant Video

To preview the participant video:

1 List the conference participants in the *Participants* pane.

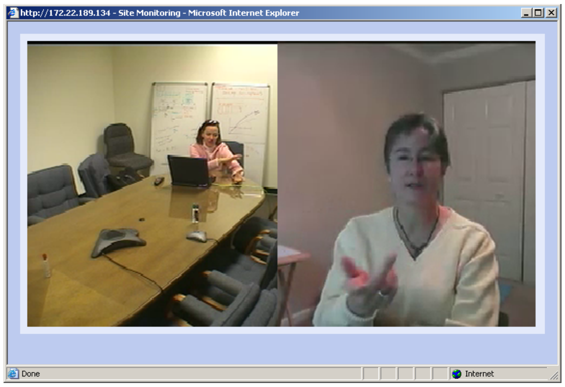
2 Right-click the participant whose video you want to preview and then click one of the following options:


Disconnect Participant
Delete Participant
Mute Audio
Suspend Video
Block Audio
Change to Regular Participant
Change To Content Token Owner
Add Participant to Address Book
Connect to Website
View Participant Sent Video
View Participant Received Video
Request Intra for Received Video
Request Intra for Sent Video
Request Intra for Received+Sent Video
Copy Participant
Cut Participant
Participant Properties

— **View Participant Sent Video** - to display the video sent from the participant to the conference.

— **View Participant Received Video** - to display the video sent from the conference to the participant.

The *Video Preview* window opens.


http://172.22.189.134 - Site Monitoring - Microsoft Internet Explorer
Done
Internet

If the video card installed in the PC does not support DirectDraw Acceleration, a black window may be viewed.

Auto Scan and Customized Polling in Video Layout (CP Only)

Auto Scan enables a user to define a single cell in the conference layout to cycle the display of participants that are not in the conference layout.

Customized Polling allows the cyclic display to be set to a predefined order for a predefined time period. The cyclic display only occurs when the number of participants is larger than the number of cells in the layout.

Guidelines

- *Auto Scan* and *Customized Polling* are supported in AVC CP conferences only.
- Participants that are in the conference layout will not appear in the *Auto Scan* enabled cell.
- If *Customized Polling* is not used to define the order of the *Auto Scan* it will proceed according to order in which the participants connected to the conference.
- If the user changes the conference layout, the *Auto Scan* settings are not exported to the new layout. If the user changes the conference layout back to the layout in which *Auto Scan* was enabled, *Auto Scan* with the previous settings will be resumed.

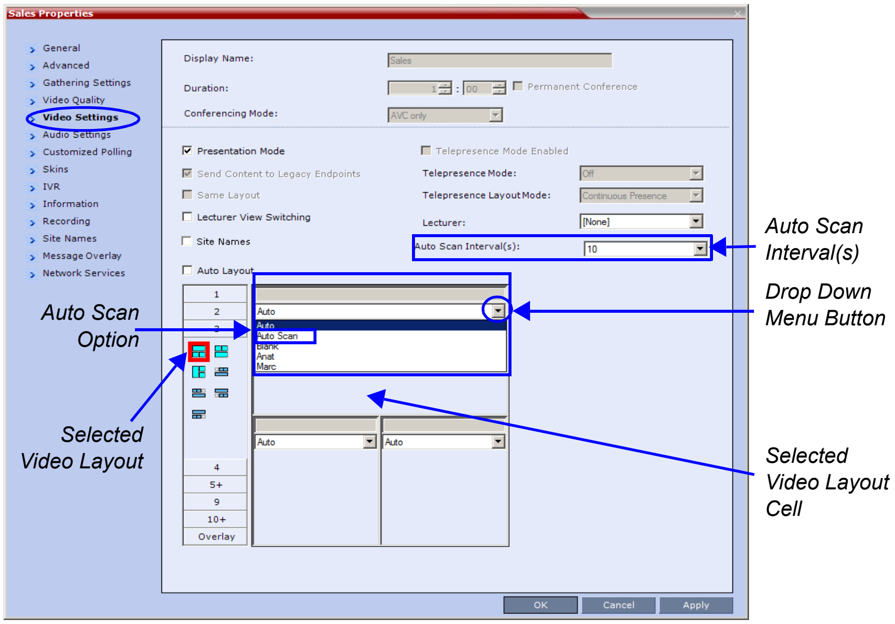
Enabling Auto Scan and Customized Polling (CP Only)

Auto Scan

To enable Auto Scan:

1 In the *Collaboration Server Web Client Main Screen - Conference* list pane, double-click the conference or right-click the conference and then click **Conference Properties**.

2 In the *Conference Properties - General* dialog box, click **Video Settings**.
The *Video Settings* tab is displayed.



Sales Properties

Display Name: Sales

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: AVC only

☑ Presentation Mode

☑ Send Content to Legacy Endpoints

□ Same Layout

□ Lecturer View Switching

□ Site Names

□ Auto Layout

□ Telepresence Mode Enabled

Telepresence Mode: Off

Telepresence Layout Mode: Continuous Presence

Lecturer: [None]

Auto Scan Interval(s): 10

3 If **Auto Layout** check box is selected, clear it.

4 In the video layout cell to be designated for *Auto Scan*, click the drop-down menu button and select **Auto Scan**.

5 Select from the *Auto Scan Interval(s)* drop-down list the scanning interval in seconds.

6 Click the **Apply** button to confirm and keep the *Conference Properties* dialog box open.

-or-

Click **OK** to confirm and close the *Conference Properties* dialog box.

Customized Polling

The order in which the Auto Scanned participants are displayed in the *Auto Scan* enabled cell of the video layout can be customized.

1 Open the *Customized Polling* tab:

a If the *Video Settings* tab is open click the **Customized Polling** tab.

or

b In the *Conference* list pane, double-click the conference or right-click the conference and then click **Conference Properties**.

c In the *Conference Properties - General* dialog box, click **Customized Polling**. The *Customized Polling* tab is displayed.

All conference participants are listed in the left pane (*All Participants*) while the participants that are to be displayed in the Auto Scan enabled cell of the video layout are listed in the right pane (*Scanning Order*).

The dialog box buttons are summarized in the following table:

Button	Description
Add	Select a participant and click this button to *Add* a the participant to the list of participants to be Auto Scanned. The participants name is removed from the *All Participants* pane.
Delete	Select a participant and click this button to *Delete* the participant from the list of participants to be *Auto Scanned*. The participants name is moved back to the *All Participants* pane.
Add All	Add all participants to the list of participants to be *Auto Scanned*. All participants' names are removed from the *All Participants* pane.
Delete All	Delete all participant from the list of participants to be *Auto Scanned*. All participants' names are moved back to the *All Participants* pane.
Up	Select a participant and click this button to move the participant *Up* in the *Scanning Order*.
Down	Select a participant and click this button to move the participant *Down* in the *Scanning Order*.

2 **Optional.** Add a participant to the list of participants to be *Auto Scanned*:

— Click on the participant's name in the *All Participants* list and then click the **Add** button to move the participant to the *Scanning Order* pane.

3 **Optional.** Delete a participant from the list of participants to be *Auto Scanned*:

— Click on a participant's name in the *Scanning Order* list and then click the **Delete** button to move the participant back to the *All Participants* pane.

4 **Optional.** Add all participants to the list of participants to be *Auto Scanned* by clicking the **Add All** button.

5 **Optional.** Delete all participant from the list of participants to be *Auto Scanned* by clicking the **Delete All** button.

6 **Optional.** Move the participant up in the *Scanning Order* by clicking the **Up** button.

7 **Optional.** Move the participant down in the *Scanning Order* by clicking the **Down** button.

8 Click the **Apply** button to confirm and keep the *Conference Properties* dialog box open, or click the **OK** the button to confirm and return to the *Collaboration Server Web Client Main Screen*.

Media Encryption

Encryption is available at the conference and participant levels, based on AES 128 (Advanced Encryption Standard) and is fully H.233/H.234 compliant and the Encryption Key exchange DH 1024-bit (Diffie-Hellman) standards.

Media Encryption Guidelines

- Encryption is not available in all countries and it is enabled in the MCU license. Contact Polycom Support to enable it.
- Media encryption is supported in CP, SVC Only and mixed CP and SVC Conferencing Modes.
- Endpoints must support both AES 128 encryption and DH 1024 key exchange standards which are compliant with H.235 (H.323) to encrypt and to join an encrypted conference.
- The encryption mode of the endpoints is not automatically recognized, therefore the encryption mode must be set for the conference or the participants (when defined).
- Conference level encryption must be set in the Profile, and cannot be changed once the conference is running.
- If an endpoint connected to an encrypted conference stops encrypting its media, it is disconnected from the conference.
- In Cascaded conferences, the link between the cascaded conferences must be encrypted in order to encrypt the conferences.
- The recording link can be encrypted when recording from an encrypted conference to the RSS that is set to encryption. For more information, see "*Recording Link Encryption"* on page **13-7**.
- Encryption of SIP Media is supported using *SRTP* (*Secured Real-time Transport Protocol*) and the *AES* key exchange method.
- Encryption of SIP Media requires the encryption of SIP signaling - TLS Transport Layer must be used.
- Encryption of SIP Media is supported in conferences as follows:
  - — All media channels are encrypted: video, audio and FECC.
  - — Collaboration Server SRTP implementation complies with Microsoft SRTP implementation.
  - — LPR is not supported with SRTP.
  - — The **ENABLE_SIRENLPR_SIP_ENCRYPTION** *System Flag* enables the *SirenLPR* audio algorithm when using encryption with the *SIP* protocol. The default value of this flag is **NO** meaning *SirenLPR* is disabled by default for *SIP* participants in an encrypted conference. To enable *SirenLPR* the *System Flag* must be added to *system.cfg* and its value set to **YES**.
  - — The **SEND_SRTP_MKI** *System Flag* enables or disables the inclusion of the *MKI* field in *SRTP* packets sent by the Collaboration Server. The default value of the flag is **YES**.

  Add the flag to *system.cfg* and set its value set to **NO** to disable the inclusion of the *MKI* field in *SRTP* packets sent by the Collaboration Server when using endpoints that cannot decrypt *SRTP*-based audio and video streams if the *MKI* (*Master Key Identifier*) field is included in *SRTP* packets sent by the Collaboration Server. When all conferences on the RMX will not have MS-Lync clients participating and will have 3rd party endpoints participating. This setting is recommended for Maximum Security Environments.

Add the flag to *system.cfg* and set its value set to **YES** when *Microsoft Office Communicator* and *Lync Clients.* When any conferences on the RMX will have both MS-Lync clients and Polycom endpoints participating. Some 3rd party endpoints may be unsuccessful in participating in conferences with this setting.

Polycom endpoints function normally regardless of the setting of this flag.

For more information, see "*Modifying System Flags"* on page **20-1**.

Mixing Encrypted and Non-encrypted Endpoints in one Conference

Mixing encrypted and non-encrypted endpoints in one conference is possible, based on the Encryption option "Encrypt When Possible" in the *Conference Profile - Advance* dialog box.

The option *"Encrypt When Possible"* enables the negotiation between the MCU and the endpoints and let the MCU connect the participants according to their capabilities, where encryption is the preferred setting. Defined participants that cannot connect encrypted are connected non-encrypted, with the exception of dial-out SIP participants.

- When the conference encryption is set to "*Encrypt when possible*", SIP dial out participants whose encryption is set to AUTO can only connect with encryption, otherwise they are disconnected from the conference.

When the conference encryption is set to "*Encrypt when possible*", SIP dial out participants whose encryption is set to AUTO can only connect with encryption, otherwise they are disconnected from the conference.

The same system behavior can be applied to undefined participants, depending on the setting of the System Flag FORCE_ENCRYPTION_FOR_UNDEFINED_PARTICIPANT_IN_WHEN_AVAILABLE_MODE:

- When set to **NO** and the conference encryption in the Profile is set to *"Encrypt When Possible"*, both *Encrypted* and *Non-encrypted undefined participants* can connect to the same conferences, where encryption is the preferred setting.
- When set to **YES** (default), *Undefined participants* must connect encrypted, otherwise they are disconnected.

For *defined participants*, connection to the conference is decided according to the encryption settings in the conference *Profile,* the *Defined Participant's* encryption settings.

For *undefined participants*, connection to the conference is decided according to the encryption settings in the conference *Profile,* the System Flag setting and the connecting endpoint's *Media Encryption* capabilities.

Direct Connection to the Conference

Table 3-8, summarizes the connection status of participants, based on the encryption settings in the conference *Profile,* the *Defined Participant's* encryption settings or the System Flag setting for undefined participants and the connecting endpoint's *Media Encryption* capabilities.

Table 3-8 *Connection of Defined and Undefined H.323, SIP Participants to the Conference Based on the Encryption Settings*

Conference Encryption Setting	**Defined Participant**		**Undefined Participant**	
	Encryption Setting	**Connection status**	**Connection Status *Flag = No**	**Connection Status *Flag = YES**
No Encryption	**Auto**	Connected, non-encrypted	Connected non-encrypted (Encryption is not declared by the Collaboration Server, therefore the endpoint does not use encryption)	Connected non-encrypted (Encryption is not declared by the Collaboration Server, therefore the endpoint does not use encryption)
	No	Connected, non-encrypted		
	Yes	Connected only if encrypted. Non-encrypted endpoints are disconnected as encryption is forced for the participant.		
Encrypt All	**Auto**	Connected, encrypted. Non-encrypted endpoints are disconnected	Connect only if encrypted. Non-encrypted endpoints are disconnected	Connect only if encrypted. Non-encrypted endpoints are disconnected
	No	Disconnected (cannot be added to the conference)		
	Yes	Connected, encrypted		
Encrypt When Possible	**Auto**	*All defined participants* ***except*** *dial-out SIP participants:* Connect encrypted - Endpoints with encryption capabilities. Connect non-encrypted - endpoints without encryption capabilities. *Defined dial-out SIP participant:* Connect only if encrypted. Non-encrypted endpoints are disconnected.	Connect encrypted - Endpoints with encryption capabilities. Connect non-encrypted - endpoints without encryption capabilities	Connect only if encrypted. Non-encrypted endpoints are disconnected.
	No	Connected, non-encrypted		
	Yes	Connected, encrypted		

* System Flag = FORCE_ENCRYPTION_FOR_UNDEFINED_PARTICIPANT_IN_WHEN_AVAILABLE_MODE

Connection to the Entry Queue

An undefined participant connecting to an *Entry Queue* inherits the encryption characteristics of the *Entry Queue* as defined in the *Entry Queue's* profile.

Participants can be moved from the Entry Queue and the destination conference if both conferencing entities have the same Profile settings, i.e. from SVC Only Entry Queue to SVC Only conference and from mixed CP and SVC Entry Queue to a mixed CP and SVC conference, etc.

Table 3-9 summarizes the connection possibilities for a participant that is to be moved from an *Entry Queue* to a destination conference for each of the conference *Profile* and Entry Queue encryption options.

Table 3-9 Connection of Undefined Participants to the Entry Queue Based on the Encryption Settings

Entry Queue Encryption Setting	**Undefined Participant Connection to the Entry Queue**	
	***Flag = No**	***Flag = YES**
No Encryption	Connected, non-encrypted (Encryption is not declared by the Collaboration Server, therefore endpoint does not use encryption)	Connected, non-encrypted (Encryption is not declared by the Collaboration Server, therefore endpoint does not use encryption)
Encrypt All	Connected only if encrypted. Non-encrypted endpoints are disconnected	Connected only if encrypted. Non-encrypted endpoints are disconnected
Encrypt When Possible	Connected encrypted - Endpoints with encryption capabilities. Connected non-encrypted - endpoints without encryption capabilities	Connected only if encrypted. Non-encrypted endpoints are disconnected.

* System Flag = FORCE_ENCRYPTION_FOR_UNDEFINED_PARTICIPANT_IN_WHEN_AVAILABLE_MODE

Moving from the Entry Queue to Conferences or Between Conferences

Participants can be moved from the Entry Queue and the destination conference if both conferencing entities have the same Profile settings, i.e. from SVC Only Entry Queue to SVC Only conference and from mixed CP and SVC Entry Queue to a mixed CP and SVC conference, etc.

When moving participants from the Entry Queue to the destination conference, or when the Collaboration Server user moves AVC participants from one conference to another (SVC participants cannot be moved between conferences), the connection rules are similar and they are summarized in Table 3-10:

Table 3-10 Moving Participants from the Entry Queue to the Destination conference or between conferences Based on the Encryption Settings

Destination Conference Encryption Setting	Current Participant Encryption Status			
	Encrypted		Non-Encrypted	
	*Flag = NO	*Flag = YES	*Flag = NO	*Flag = YES
No Encryption	Move succeeds, connected encrypted		Move succeeds, connected non-encrypted	
Encrypt All	Move succeeds, connected encrypted.		Move fails, disconnected.	
Encrypt When Possible	Move succeeds, connected encrypted	Move succeeds, connected encrypted	Move succeeds, connected non-encrypted	Connected only if endpoint was a defined participant in the source conference. Otherwise, move fails.

* System Flag = FORCE_ENCRYPTION_FOR_UNDEFINED_PARTICIPANT_IN_WHEN_AVAILABLE_MODE

Recording Link Encryption

Recording Links are treated as regular participants, however the ALLOW_NON_ENCRYPT_RECORDING_LINK_IN_ENCRYPT_CONF *System Flag* must be set to YES if a non-encrypted *Recording Link* is to be allowed to connect to an encrypted conference.

Table 3-11 summarizes the connection possibilities for a *Recording Link* that is to be connected to a conference for each of the conference *profile* and *Entry Queue* encryption options.

Table 3-11 Connections by Recording Link and Conference Encryption Settings

Conference Profile Setting	Recording Link Connection Status according to flag: ALLOW_NON_ENCRYPT_RECORDING_ LINK_IN_ENCRYPT_CONF	
	YES	NO
Encrypt All	Connected encrypted if possible, otherwise connected non-encrypted.	Connected only if encrypted, otherwise disconnected
No Encryption	Connected non-encrypted	Connected non-encrypted

Table 3-11 Connections by Recording Link and Conference Encryption Settings (Continued)

Conference Profile Setting	Recording Link Connection Status according to flag: ALLOW_NON_ENCRYPT_RECORDING_ LINK_IN_ENCRYPT_CONF	
	YES	**NO**
Encrypt when possible	Connected encrypted if possible, otherwise connected non-encrypted.	Connected encrypted if possible, otherwise connected non-encrypted.

Encryption Flag Settings

To modify the Encryption flags:

1 Click **Setup>System Configuration**.
The *System Flags* dialog box opens.

2 Set the **FORCE_ENCRYPTION_FOR_UNDEFINED_PARTICIPANT_IN_WHEN_AVAILABLE_MODE** flag to **YES** or **NO**.

3 If recording will be used in encrypted conferences, set the **ALLOW_NON_ENCRYPT_RECORDING_LINK_IN_ENCRYPT_CONF** flag to **YES** or **NO**.

4 Click **OK**.

For more information, see "*Modifying System Flags"* on page **20-1**.

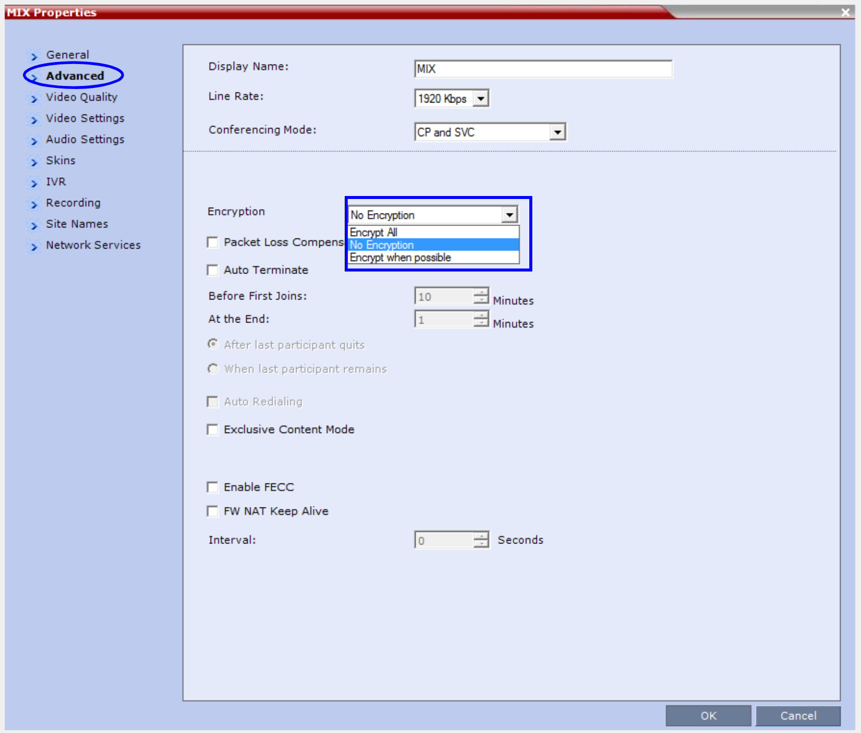
5 Reset the MCU for flag changes to take effect.

Enabling Encryption in the Profile

Encryption for the conference is in the Profile and cannot be changed once the conference is running.

To enable encryption at the conference level:

>> In the *Conference Profile Properties – Advanced* dialog box, select one of the following Encryption options:



Display Name: MIX

Line Rate: 1920 Kbps

Conferencing Mode: CP and SVC

Encryption: No Encryption
- Encrypt All
- No Encryption
- Encrypt when possible

□ Packet Loss Compens

□ Auto Terminate

Before First Joins: 10 Minutes

At the End: 1 Minutes

● After last participant quits

○ When last participant remains

□ Auto Redialing

□ Exclusive Content Mode

□ Enable FECC

□ FW NAT Keep Alive

Interval: 0 Seconds

— **Encrypt All** - Encryption is enabled for the conference and all conference participants must be encrypted.

— **No Encryption** - Encryption is disabled for the conference.

— **Encrypt when possible** - enables the negotiation between the MCU and the endpoints and let the MCU connect the participants according to their capabilities, where encryption is the preferred setting. For connection guidelines see "*Mixing Encrypted and Non-encrypted Endpoints in one Conference"* on page 3-27.

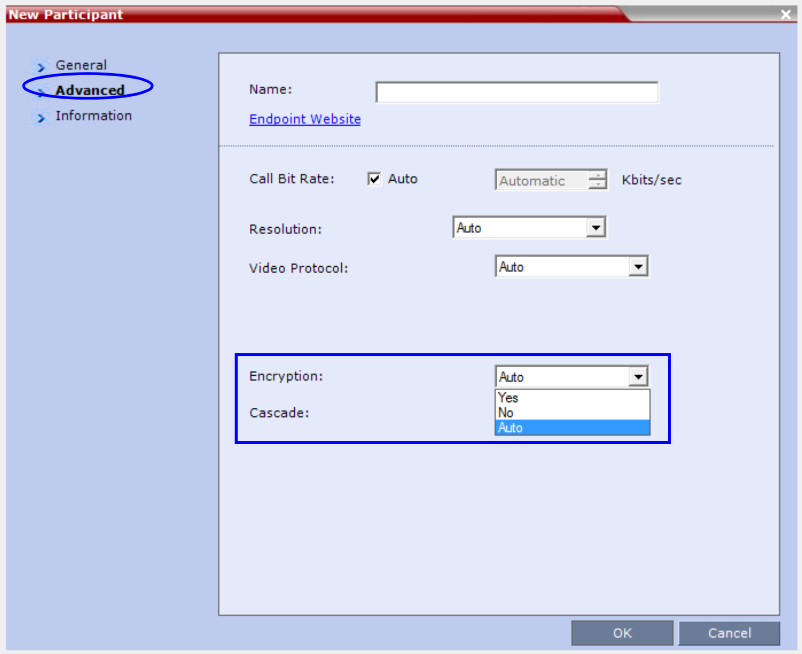
For more information about recording encrypted conferences, see "*Recording Link Encryption"* on page 13-7.

Enabling Encryption at the Participant Level

You can select the encryption mode for each of the defined participants. Encryption options are affected by the settings of the flag in the system configuration. Undefined participants are connected with the Participant *Encryption* option set to **Auto**, inheriting the conference/ Entry Queue encryption setting.

To enable encryption at the participant level:

>> In the *Participant Properties – Advanced* dialog box, in the *Encryption* list, select one of the following options: **Auto**, **On**, or **Off**.



New Participant

Name:

Endpoint Website

Call Bit Rate: ☑ Auto Automatic Kbits/sec

Resolution: Auto

Video Protocol: Auto

Encryption: Auto

Cascade:

— **Auto** - The participant inherits the conference/Entry Queue encryption setting. The participant connects as encrypted only if the conference is defined as encrypted.

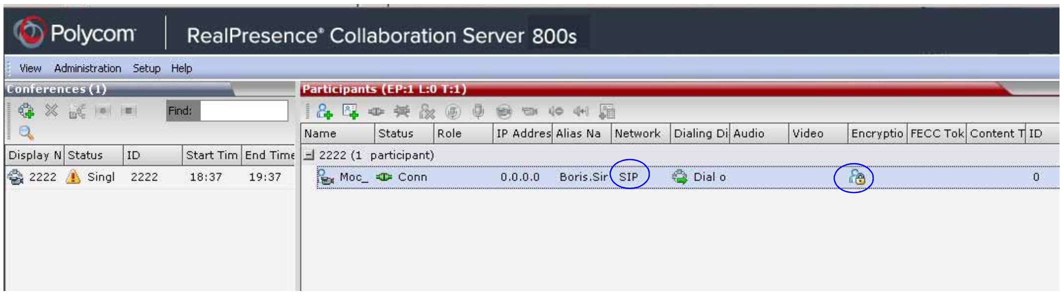
— **Yes** - The participant joins the conference/Entry Queue as *encrypted.*

— **No** - The participant joins the conference/Entry Queue as *non-encrypted*.

Monitoring the Encryption Status

The conference encryption status is indicated in the *Conference Properties - General* dialog box.

The participant encryption status is indicated by a check mark in the *Encryption* column in the *Participants* list pane.



Polycom
RealPresence® Collaboration Server 800s
View Administration Setup Help
Find:

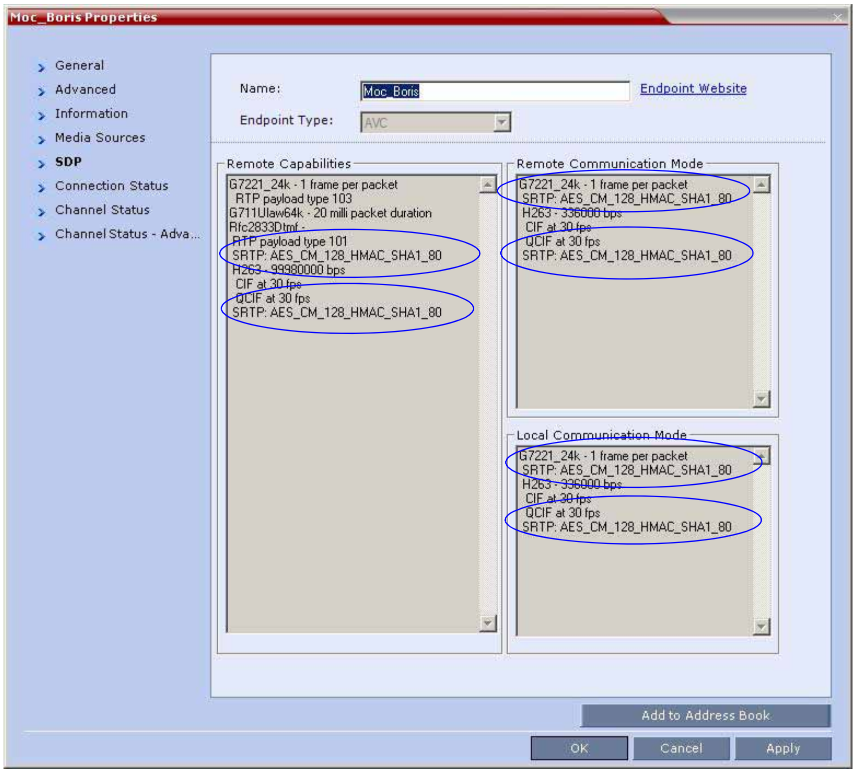
Conferences (1)

Display N	Status	ID	Start Tim	End Time
2222	Singl	2222	18:37	19:37

Participants (EP:1 L:0 T:1)

Name	Status	Role	IP Addres	Alias Na	Network	Dialing Di	Audio	Video	Encryptio	FECC Tok	Content T	ID
2222 (1 participant)												
Moc_	Conn		0.0.0.0	Boris.Sir	SIP	Dial o			🔒			0

The participant encryption status is also indicated in the *Participant Properties – SDP* tab, where SRTP indication is listed for each encrypted channel (for example, audio and video).



Name: Moc_Boris

Endpoint Type: AVC

Remote Capabilities

```
G7221_24k - 1 frame per packet
 RTP payload type 103
G711Ulaw64k - 20 milli packet duration
Rfc2833Dtmf -
 RTP payload type 101
 SRTP: AES_CM_128_HMAC_SHA1_80
 H263 - 99980000 bps
  CIF at 30 fps
  QCIF at 30 fps
 SRTP: AES_CM_128_HMAC_SHA1_80
```

Remote Communication Mode

```
G7221_24k - 1 frame per packet
 SRTP: AES_CM_128_HMAC_SHA1_80
 H263 - 336000 bps
  CIF at 30 fps
  QCIF at 30 fps
 SRTP: AES_CM_128_HMAC_SHA1_80
```

Local Communication Mode

```
G7221_24k - 1 frame per packet
 SRTP: AES_CM_128_HMAC_SHA1_80
 H263 - 336000 bps
  CIF at 30 fps
  QCIF at 30 fps
 SRTP: AES_CM_128_HMAC_SHA1_80
```

An encrypted participant who is unable to join a conference is disconnected from the conference. The disconnection cause is displayed in the *Participant Properties – Connection Status* tab, *Security Failure* indication, and the *Cause* box identifies the encryption related situation.

For more information about monitoring, see "*Conference and Participant Monitoring"* on page **12-1**.

Packet Loss Compensation (LPR and DBA) AVC CP Conferences

Lost Packet Recovery (*LPR*) and *Dynamic Bandwidth Allocation* (*DBA*) help minimize media quality degradation that can result from packet loss in the network. *Packet loss Compensation* is available in *AVC CP Conferencing Mode* only and is not supported in *SVC Conferencing Mode* or *CP and SVC Conferencing Mode.*

Packet Loss

Packet Loss refers to the failure of data packets, transmitted over an IP network, to arrive at their destination. *Packet Loss* is described as a percentage of the total packets transmitted.

Causes of Packet Loss

Network congestion within a LAN or WAN, faulty or incorrectly configured network equipment or faulty cabling are among the many causes of Packet Loss.

Effects of Packet Loss on Conferences

Packet Loss affects the quality of:

- **Video –** frozen images, decreased frame rate, flickering, tiling, distortion, smearing, loss of lip sync
- **Audio –** drop-outs, chirping, audio distortion
- **Content –** frozen images, blurring, distortion, slow screen refresh rate

Lost Packet Recovery

The *Lost Packet Recovery* (*LPR*) algorithm uses *Forward Error Correction* (*FEC*) to create additional packets that contain recovery information. These additional packets are used to reconstruct packets that are lost, for whatever reason, during transmission. *Dynamic Bandwidth Allocation* (*DBA*) is used to allocate the bandwidth needed to transmit the additional packets.

Lost Packet Recovery Guidelines

- If packet loss is detected in the packet transmissions of either the video or Content streams:
  - *LPR* is applied to both the video and Content streams.
  - *DBA* allocates bandwidth from the video stream for the insertion of additional packets containing recovery information.
- *LPR* is supported in H.323 and *SIP* networking environments only.
- In *LPR*-enabled *Continuous Presence* conferences:
  - Both *LPR*-enabled and non-*LPR*-enabled endpoints are supported.
  - The *LPR* process is not applied to packet transmissions from non-*LPR*-enabled H.323 and SIP endpoints.
  - Non-LPR-enabled endpoints can be moved to *LPR*-enabled conferences.
  - LPR-enabled endpoints **cannot** be moved to non-*LPR*-enabled conferences.
- When connecting via an *Entry Queue:*
  - A participant using an *LPR*-enabled endpoint can be moved to a non-*LPR*-enabled conference. The participant is connected with LPR enabled.

Enabling Lost Packet Recovery

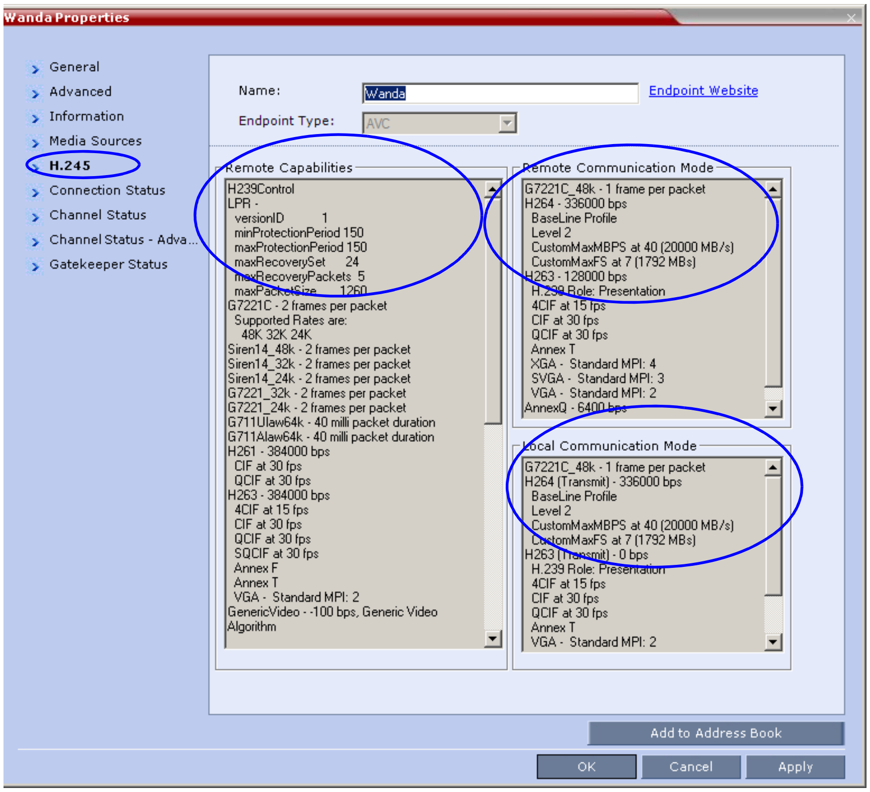
LPR is enabled or disabled in the *Conference Profile* dialog box.

- **CP Conferences –** *LPR* is enabled by default in the *New Profile – Advanced* dialog box.

  For more information, see "*Defining New Profiles"* on page **2-20**.

Monitoring Lost Packet Recovery

In the *Participant Properties – H.245* tab, *LPR* activity is displayed in all three panes.



Name: Wanda

Endpoint Type: AVC

Remote Capabilities

```
H239Control
LPR -
  versionID          1
  minProtectionPeriod 150
  maxProtectionPeriod 150
  maxRecoverySet      24
  maxRecoveryPackets  5
  maxPacketSize       1260
G7221C - 2 frames per packet
  Supported Rates are:
    48K 32K 24K
Siren14_48k - 2 frames per packet
Siren14_32k - 2 frames per packet
Siren14_24k - 2 frames per packet
G7221_32k - 2 frames per packet
G7221_24k - 2 frames per packet
G711Ulaw64k - 40 milli packet duration
G711Alaw64k - 40 milli packet duration
H261 - 384000 bps
  CIF at 30 fps
  QCIF at 30 fps
H263 - 384000 bps
  4CIF at 15 fps
  CIF at 30 fps
  QCIF at 30 fps
  SQCIF at 30 fps
  Annex F
  Annex T
  VGA -  Standard MPI: 2
GenericVideo - -100 bps, Generic Video
Algorithm
```

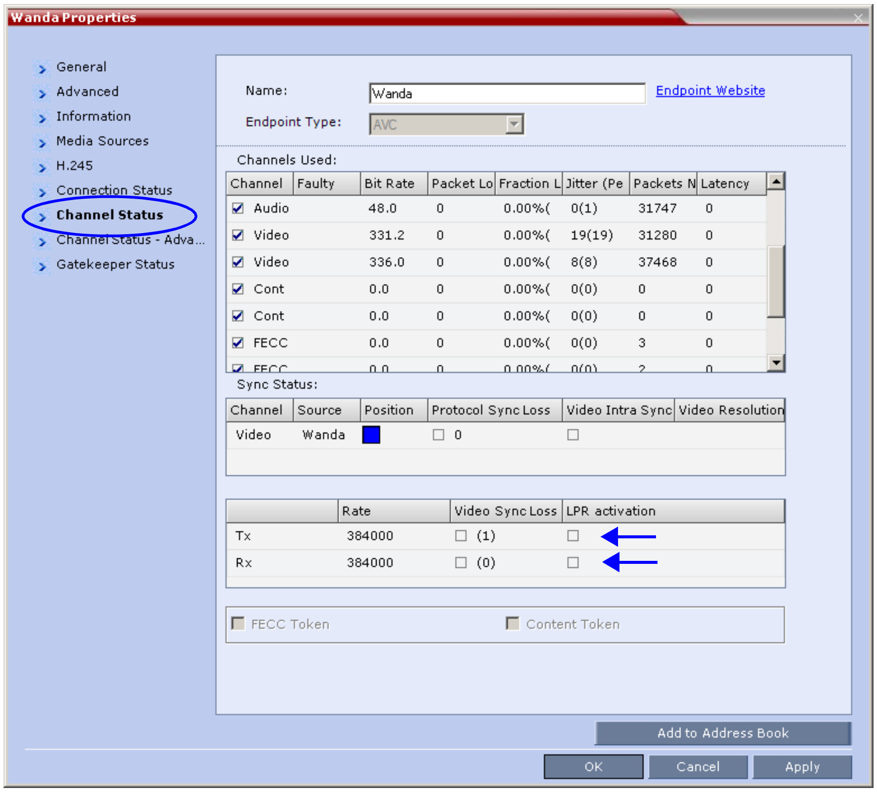
Remote Communication Mode

```
G7221C_48k - 1 frame per packet
H264 - 336000 bps
  BaseLine Profile
  Level 2
  CustomMaxMBPS at 40 (20000 MB/s)
  CustomMaxFS at 7 (1792 MBs)
H263 - 128000 bps
  H.239 Role: Presentation
  4CIF at 15 fps
  CIF at 30 fps
  QCIF at 30 fps
  Annex T
  XGA -  Standard MPI: 4
  SVGA -  Standard MPI: 3
  VGA -  Standard MPI: 2
AnnexQ - 6400 bps
```

Local Communication Mode

```
G7221C_48k - 1 frame per packet
H264 (Transmit) - 336000 bps
  BaseLine Profile
  Level 2
  CustomMaxMBPS at 40 (20000 MB/s)
  CustomMaxFS at 7 (1792 MBs)
H263 (Transmit) - 0 bps
  H.239 Role: Presentation
  4CIF at 15 fps
  CIF at 30 fps
  QCIF at 30 fps
  Annex T
  VGA -  Standard MPI: 2
```

In the *Participant Properties – Channel Status* tab, check box indicators show *LPR* activation in the local and remote (transmit and receive) channels.



Name: Wanda

Endpoint Type: AVC

Channels Used:

Channel	Faulty	Bit Rate	Packet Lo	Fraction L	Jitter (Pe	Packets N	Latency
☑ Audio		48.0	0	0.00%(	0(1)	31747	0
☑ Video		331.2	0	0.00%(	19(19)	31280	0
☑ Video		336.0	0	0.00%(	8(8)	37468	0
☑ Cont		0.0	0	0.00%(	0(0)	0	0
☑ Cont		0.0	0	0.00%(	0(0)	0	0
☑ FECC		0.0	0	0.00%(	0(0)	3	0
☑ FECC		0.0	0	0.00%(	0(0)	2	0

Sync Status:

Channel	Source	Position	Protocol Sync Loss	Video Intra Sync	Video Resolution
Video	Wanda	🟦	□ 0	□	

	Rate	Video Sync Loss	LPR activation
Tx	384000	□ (1)	□
Rx	384000	□ (0)	□

□ FECC Token □ Content Token

Lecture Mode (AVC CP Only)

Lecture Mode enables all participants to view the lecturer in full screen while the conference lecturer sees all the other conference participants in the selected layout while he/she is speaking. When the number of sites/endpoints exceeds the number of video windows in the layout, switching between participants occurs every 15 seconds. Conference participants cannot change their Personal Layouts while Lecture Mode is enabled.

Automatic switching is suspended when one of the participants begins talking, and it is resumed automatically when the lecturer resumes talking.

Lecture Mode is available only in *AVC CP Conferencing Mode.*

Enabling Lecture Mode

Lecture Mode is enabled at the conference level by selecting the lecturer. Conference participants cannot change their Personal Layouts while Lecture Mode is enabled.

Automatic switching between participants viewed on the lecturer's screen is enabled in the conference Profile.

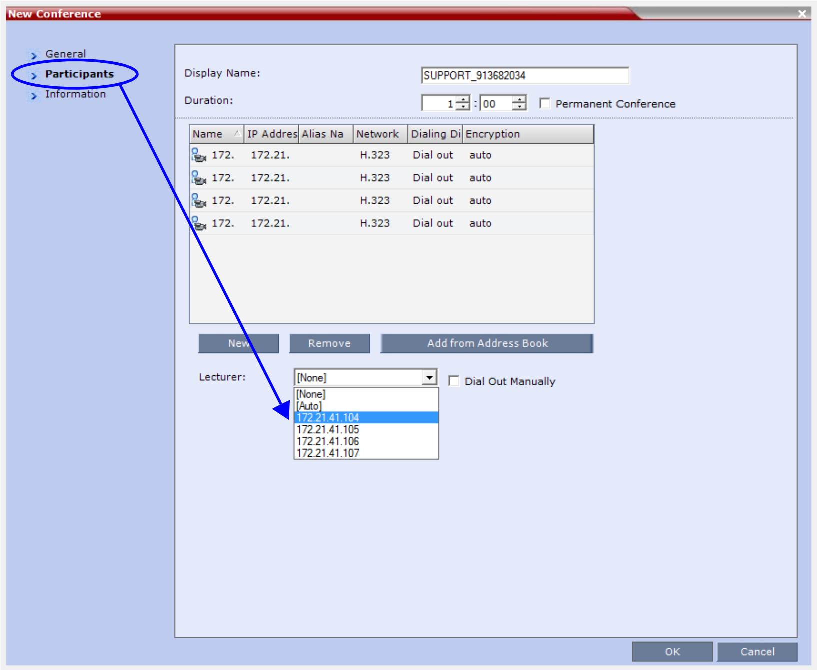
Selecting the Conference Lecturer

Selecting a lecturer for the ongoing conference, enables the Lecture Mode. You can select the lecturer:

- during the definition of the ongoing conference
- after the conference has started and the participants have connected to the conference.

To select the lecturer and enable the Lecture Mode while starting the conference:

>> In the *Conference Properties - Participant* dialog box, enable the Lecture Mode in one of the following methods:



Display Name: SUPPORT_913682034

Duration: 1 : 00 □ Permanent Conference

Name	IP Addres	Alias Na	Network	Dialing Di	Encryption
172.	172.21.		H.323	Dial out	auto
172.	172.21.		H.323	Dial out	auto
172.	172.21.		H.323	Dial out	auto
172.	172.21.		H.323	Dial out	auto

Lecturer: [None] □ Dial Out Manually

[None]
[Auto]
172.21.41.104
172.21.41.105
172.21.41.106
172.21.41.107

Selecting a defined participant:

a Add participants to the conference either from the Address book or by defining new participants.

b In the **Lecturer** field, select the lecturer from the list of the defined participants.

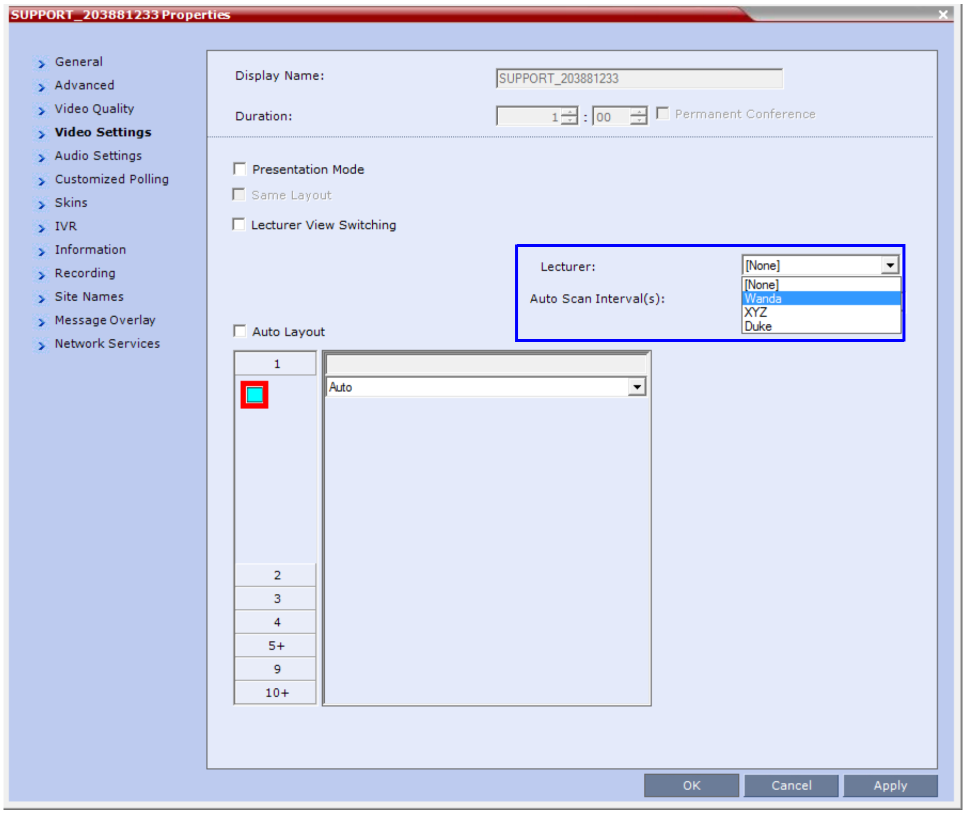
Automatic selection of the lecturer:

— In the **Lecturer** field, select **[Auto]**.
In this mode, the conference speaker becomes the lecturer.

To select the lecturer and enable the Lecture Mode during the ongoing conference:

1 Make sure that the participant you want to designate as the lecturer has connected to the conference.

2 In the *Conference Properties - Video Settings* dialog box, in the **Lecturer** field, select the lecturer from the list of the connected participants.



Display Name: SUPPORT_203881233

Duration: 1 : 00 □ Permanent Conference

□ Presentation Mode

□ Same Layout

□ Lecturer View Switching

Lecturer: [None]

Auto Scan Interval(s):

□ Auto Layout

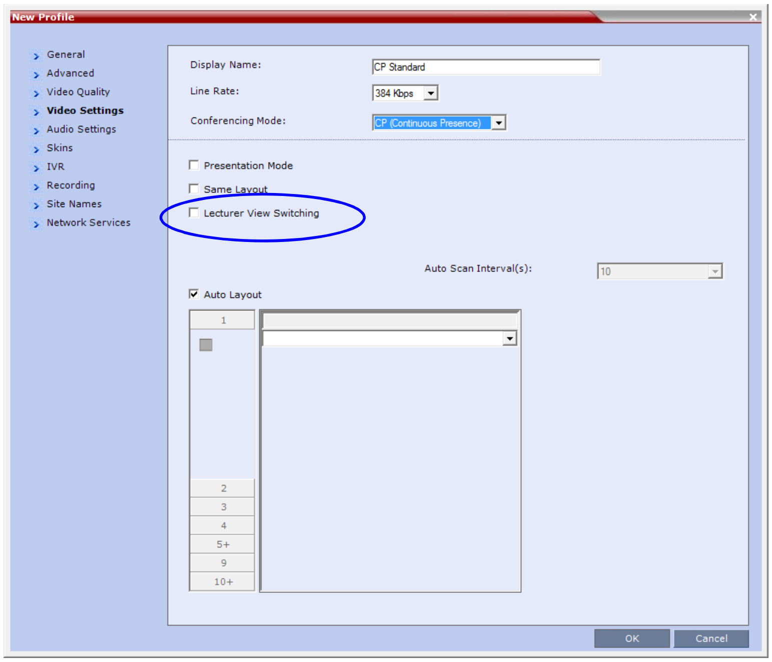
Auto

Defined dial out participants and dial in participants are considered to be two separate participants even if they have the same IP address/number. Therefore, if a defined dial-out participant is added to the conference and the same participant then dials in (before the system dialed out to that participant) the system creates a second participant in the Participants list and tries to call the dial-out participant. If the dial-out participant was designated as the conference lecturer, the system will not be able to replace that participant with the dial-in participant that is connected to the conference.

Enabling the Automatic Switching

Automatic switching between participants viewed on the lecturer's screen is enabled in the conference Profile, or during the ongoing conference, in the Conference Properties.

>> In the *Profile Properties - Video Settings* dialog box, select the **Lecturer View Switching** check box.



Display Name: CP Standard

Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

☐ Presentation Mode

☐ Same Layout

☐ Lecturer View Switching

Auto Scan Interval(s): 10

☑ Auto Layout

This option is activated when the conference includes more sites than windows in the selected layout. If this option is disabled, the participants will be displayed in the selected video layout without switching.

For more information about Profile definition, see "*Defining AVC CP Conferencing Profiles"* on page **2-21**.

>> Once the conference is running, in the *Conference Properties - Video Settings* dialog box, select the **Lecturer View Switching** check box.

Lecture Mode Monitoring

A conference in which the Lecture Mode is enabled is started as any other conference. The conference runs as an audio activated Continuous Presence conference until the lecturer connects to the conference. The selected video layout is the one that is activated when the conference starts. Once the lecturer is connected, the conference switches to the Lecture Mode.

When *Lecturer View Switching* is activated, it enables automatic switching between the conference participants in the lecturer's video window. The switching in this mode is not determined by voice activation and is initiated when the number of participants exceeds the number of windows in the selected video layout. In this case, when the switching is performed, the system refreshes the display and replaces the last active speaker with the current speaker.

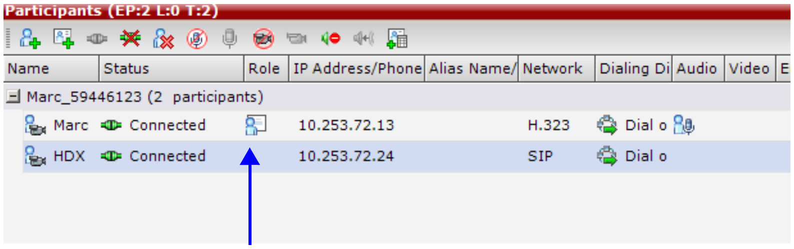
When one of the participants is talking, the automatic switching is suspended, showing the current speaker, and it is resumed when the lecturer resumes talking.

If the lecturer is disconnected during an Ongoing Conference, the conference resumes standard conferencing.

Forcing is enabled at the Conference level only. It applies only to the video layout viewed by the lecturer as all the other conference participants see only the lecturer in full screen.

If an asymmetrical video layout is selected for the lecturer (i.e. 3+1, 4+1, 8+1), each video window contains a different participant (i.e. one cannot be forced to a large frame and to a small frame simultaneously).

When *Lecture Mode* is enabled for the conference, the lecturer is indicated by an icon ( ) in the *Role* column of the *Participants* list.



Participants (EP:2 L:0 T:2)

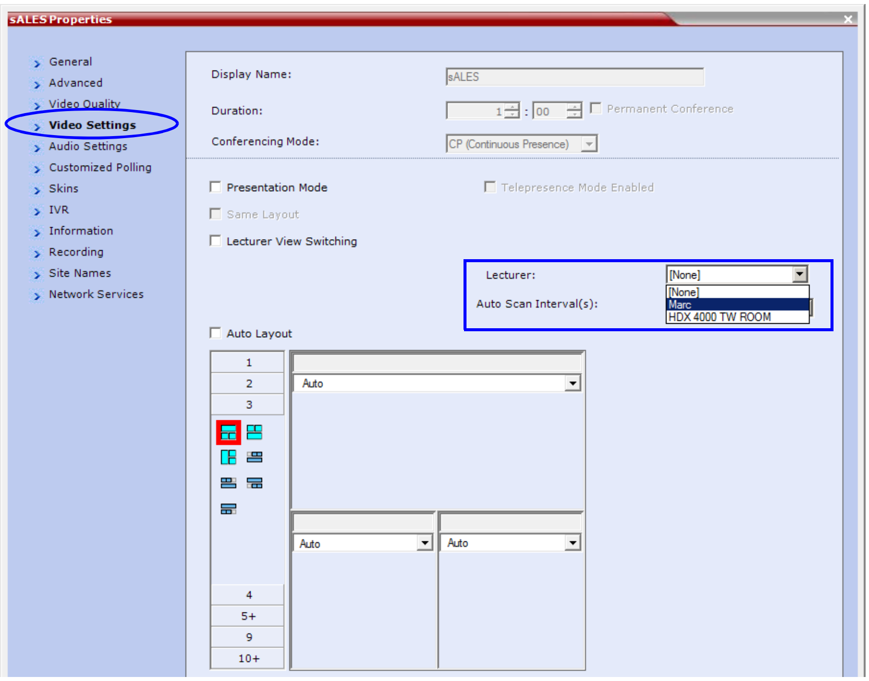
Name	Status	Role	IP Address/Phone	Alias Name/	Network	Dialing Di	Audio	Video	E
Marc_59446123 (2 participants)									
Marc	Connected		10.253.72.13		H.323	Dial o			
HDX	Connected		10.253.72.24		SIP	Dial o			

Participant designated as the Lecturer

To control the Lecture Mode during an Ongoing Conference:

During the Ongoing Conference, in the *Conference Properties - Video Settings* dialog box you can:

- Enable or disable the Lecture Mode and designate the conference lecturer in the *Lecturer* list; select **None** to disable the Lecture Mode or select a participant to become the lecturer to enable it.
- Designate a new lecturer.



sALES Properties

Display Name: sALES

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: CP (Continuous Presence)

□ Presentation Mode

□ Telepresence Mode Enabled

□ Same Layout

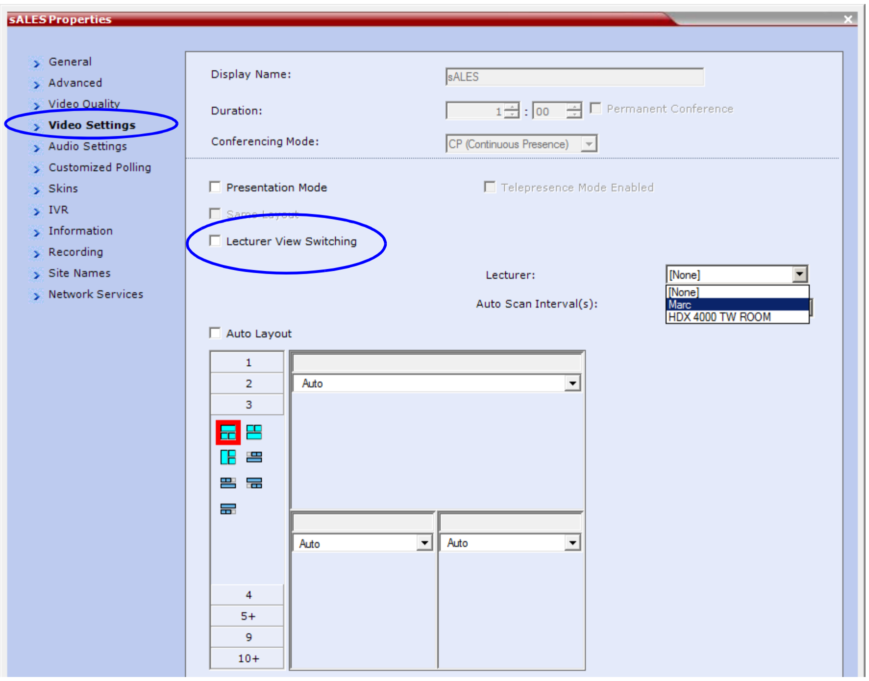
□ Lecturer View Switching

Lecturer: [None]

Auto Scan Interval(s):

□ Auto Layout

- Enable or disable the *Lecturer View Switching* between participants displayed on the lecturer monitor by selecting or clearing the **Lecturer View Switching** check box.



Display Name: sALES

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: CP (Continuous Presence)

□ Presentation Mode

□ Telepresence Mode Enabled

□ Same Layout

□ Lecturer View Switching

Lecturer: [None]

[None]
Marc
HDX 4000 TW ROOM

Auto Scan Interval(s):

□ Auto Layout

Auto

Auto

Auto

- Change the video layout for the lecturer by selecting another video layout.

Restricting Content Broadcast to Lecturer

Content broadcasting can be restricted to the conference lecturer only, when one of the conference participants is set as the lecturer (and not automatically selected by the system). Restricting the Content Broadcast prevents the accidental interruption or termination of H.239 Content that is being shared in a conference.

Content Broadcast restriction is enabled by setting the **RESTRICT_CONTENT_BROADCAST_TO_LECTURER** *system flag* to **ON**. When set to OFF (default) it enables all users to send Content.

When enabled, the following rules apply:

- Content can only be sent by the designated lecturer. When any other participant tries to send Content, the request is rejected.
- If the Collaboration Server user changes the designated lecturer (in the *Conference Properties - Video Settings* dialog box), the Content of the current lecturer is stopped immediately and cannot be renewed.
- The Collaboration Server User can abort the H.239 Session of the lecturer.
- Content Broadcasting is not implemented in conferences that do not include a designated lecturer and the lecturer is automatically selected by the system (for example, in *Presentation Mode*).

Muting Participants Except the Lecturer (AVC CP Only)

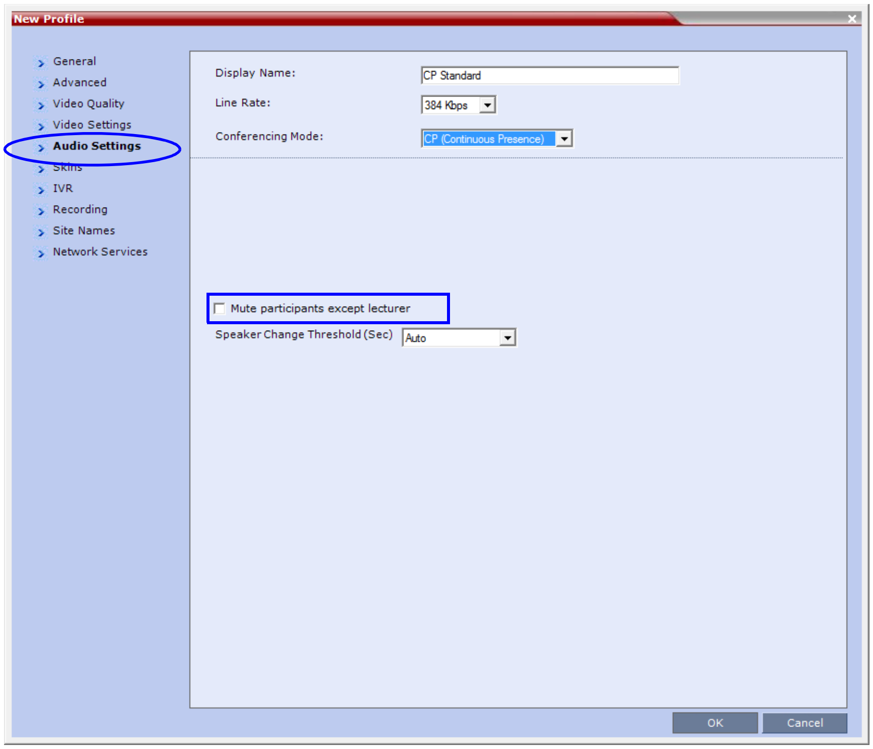
When the *Mute Participants Except Lecturer* option in the *Conference Profile* is enabled, the audio of all participants in the conference except for the lecturer can be automatically muted upon connection to the conference. This prevents other conference participants from accidentally interrupting the lecture, or from a noisy participant affecting the audio quality of the entire conference. Muted participants cannot unmute themselves unless they are unmuted from the Collaboration Server Web Client/RMX Manager.

Guidelines

- Both administrators and operators (users) are allowed to set the *Mute Participants Except Lecturer* option.
- When the *Mute Participants Except Lecturer* option is enabled, the mute indicator on the participant endpoints are not visible because the mute participants was initiated by the MCU. Therefore, it is recommended to inform the participants that their audio is muted by using the *Closed Caption* or *Message Overlay* functions.
- When the *Mute Participants Except Lecturer* option is enabled in the *Conference Profile* settings, all conferences to which this profile is assigned will start with this option enabled. All participants, except for the designated lecturer, are muted.
- The *Mute Participants Except Lecturer* option can be enabled or disabled at any time after the start of the conference. When enabled, it allows all the conference participants to converse before the lecturer joins the conference or before they are muted. When disabled, it unmutes all the participants in the conference.
- If the endpoint of the designated lecturer is muted when the lecturer connects to the conference, the lecturer remains muted until the endpoint has been unmuted.
- When you replace a lecturer, the MCU automatically mutes the previous lecturer and unmutes the new lecturer.
- When you disconnect a lecturer from the conference or the lecturer leaves the conference, all participants remain muted but are able to view participants in regular video layout until the you disable the *Mute Participants Except Lecturer* option.
- A participant can override the *Mute Participants Except Lecturer* option by activating the *Mute All Except Me* option using the appropriate DTMF code, provided the participant has authorization for this operation in the IVR Services properties. The lecturer audio is muted and the participant audio is unmuted. You can reactivate the *Mute Participants Except Lecturer* option after a participant has previously activated the *Mute All Except Me* option. The participant is muted and the lecturer, if designated, is unmuted.
- In cascaded conferences, all participants (including the link participants) except the lecturer are muted. Only the lecturer is not muted.

Enabling the Mute Participants Except Lecturer Option

The *Mute Participants Except Lecturer* option is enabled or disabled (default) in the *Conference Profile* or in an ongoing conference in the *Profile Properties - Audio Settings* tab.



Display Name: CP Standard

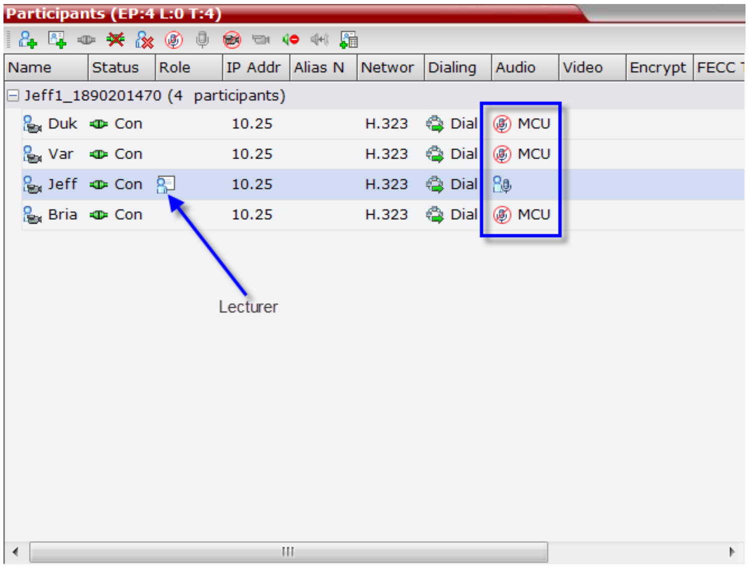
Line Rate: 384 Kbps

Conferencing Mode: CP (Continuous Presence)

□ Mute participants except lecturer

Speaker Change Threshold (Sec) Auto

When the *Mute Participants Except Lecturer* option is enabled and a conference has started, the **Mute by MCU** icon is displayed in the *Audio* column in the *Participants* pane of each participant that is muted.



Jeff1_1890201470 (4 participants)

Name	Status	Role	IP Addr	Alias N	Networ	Dialing	Audio	Video	Encrypt	FECC
Duk	Con		10.25		H.323	Dial	MCU			
Var	Con		10.25		H.323	Dial	MCU			
Jeff	Con		10.25		H.323	Dial				
Bria	Con		10.25		H.323	Dial	MCU			

Audio Algorithm Support

The Collaboration Server supports the following audio algorithms in **AVC conferences:** G.711, G. 719, G.722, G.722.1, G.722.1C, G.729A, G.723.1, Polycom Siren 7 (in mono), Siren14, Siren 22 (in mono or stereo) and SirenLPR.

Polycom's proprietary *Siren 22* and industry standard *G.719* audio algorithms are supported for participants connecting with *Polycom* endpoints.

The *Siren 22* audio algorithm provides CD-quality audio for better clarity and less listener fatigue with audio and visual communication applications. *Siren 22* requires less computing power and has much lower latency than alternative wideband audio technologies.

The SirenLPR audio algorithm provides CD-quality audio for better clarity and less listener fatigue with audio and visual communication applications.

In **SVC conferences**, the system supports SAC (Scalable Audio Coding) audio algorithm.

Guidelines

- *Siren 22* and *G.719* are supported in both mono and stereo.
- Stereo is supported in *H.323* calls only.
- *Siren 22* is supported by Polycom HDX endpoints, version 2.0 and later.
- *G.728* is supported in *H.323* and *SIP* environments.
- *SirenLPR* is enabled by default and can be disabled by setting the system flag, **ENABLE_SIRENLPR**, to **NO**.
- *SirenLPR* is supported:
  - In IP (H.323, SIP) calls only.
  - In CP and VSW conferences.
  - With *Polycom CMAD* and *HDX* "*Canyon 3.0.1*" endpoints.
  - For mono audio at audio line rates of 32Kbps, 48Kbps and 64Kbps.
  - For stereo audio at audio line rates of 64Kbps, 96Kbps and 128Kbps.

SIP Encryption

The **ENABLE_SIRENLPR_SIP_ENCRYPTION** *System Flag* enables the *SirenLPR* audio algorithm when using encryption with the *SIP* protocol.

The default value of this flag is **NO** meaning *SirenLPR* is disabled by default for *SIP* participants in an encrypted conference. To enable *SirenLPR* the *System Flag* must be added to *system.cfg* and its value set to **YES**.

Mono

The *Siren 22*, *G.719* and *SirenLPR* mono audio algorithms are supported at the following bit rates:

Table 3-12 *Siren22, G.719 and SirenLPR Mono vs Bitrate*

Audio Algorithm	Minimum Bitrate (kbps)
Siren22 64k	
Siren22 48K	
Siren22_32k	

Table 3-12 Siren22, G.719 and SirenLPR Mono vs Bitrate (Continued)

Audio Algorithm	Minimum Bitrate (kbps)
G.719_64k	384
G.719_48k	
G.719_32k	
G.728 16K	
G.719_64k	384
SirenLPR_48k	256
Siren22_48K	
G.719_48k	
G.7221C_48k	
Siren14_48k	
SirenLPR_32k	128
Siren22_32k	
G.719_32k	
G.7221C_32k	
Siren14_32k	
SirenLPR	64
SirenLPR	48
SirenLPR	32

Stereo

The *Siren 22Stereo, G.719Stereo* and *SirenLPR* audio algorithms are supported at the following bit rates.

Table 3-13 Siren22Stereo, G.719Stereo and SirenLPR vs Bitrate

Audio Algorithm	Minimum Bitrate (kbps)
Siren22Stereo_128k	1024
SirenLPRStereo_128k	
G.719Stereo_128k	
Siren22Stereo_96k	512
SirenLPRStereo_96k	
G.719Stero_96k	
Siren14Stero_96k	

Table 3-13 Siren22Stereo, G.719Stereo and SirenLPR vs Bitrate (Continued)

Audio Algorithm	Minimum Bitrate (kbps)
SirenLPRStereo_64k	384
G.719Stereo_64k	
Siren22Stereo_64k	
Siren14Stereo_64k	

Monitoring Participant Audio Properties

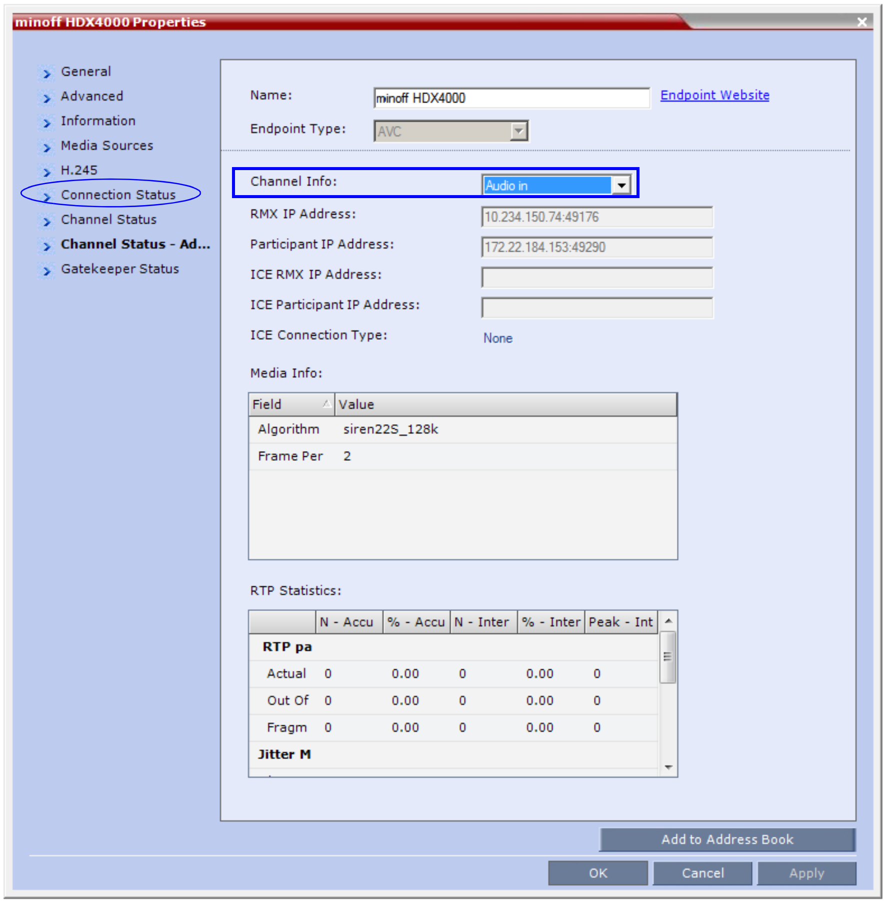
The audio algorithm used by the participant's endpoint can be verified in the Participant Properties - Channel Status dialog box.

To view the participant's properties during a conference:

1 In the *Participants* list, right click the desired participant and select **Participant Properties**.

2 Click the **Channel Status - Advanced** tab.
The *Participant Properties - Channel Status - Advanced* dialog box is displayed.

3 In the *Channel Info* field, select **Audio In** or **Audio Out** to display the audio parameters.



Name: minoff HDX4000

Endpoint Type: AVC

Channel Info: Audio in

RMX IP Address: 10.234.150.74:49176

Participant IP Address: 172.22.184.153:49290

ICE RMX IP Address:

ICE Participant IP Address:

ICE Connection Type: None

Media Info:

Field	Value
Algorithm	siren22S_128k
Frame Per	2

RTP Statistics:

	N - Accu	% - Accu	N - Inter	% - Inter	Peak - Int
RTP pa					
Actual	0	0.00	0	0.00	0
Out Of	0	0.00	0	0.00	0
Fragm	0	0.00	0	0.00	0
Jitter M					

4 Click the **OK** button.

Permanent Conference

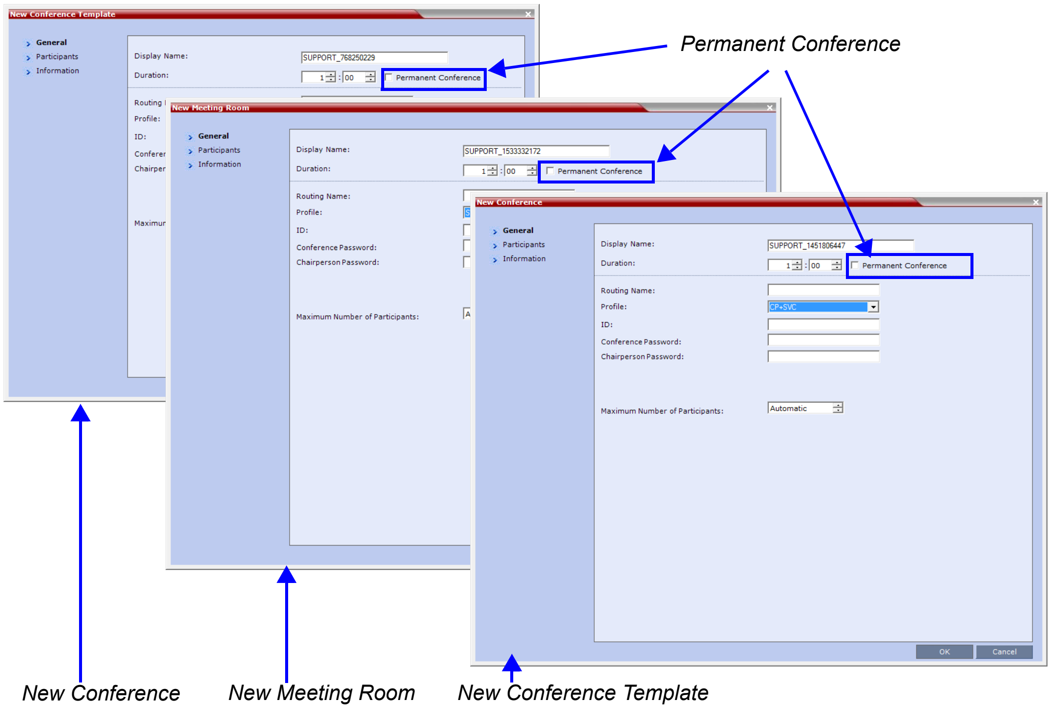
A *Permanent Conference* is any ongoing conference with no pre-determined *End Time* continuing until it is terminated by an administrator, operator or chairperson.

Guidelines

- *Auto Terminate* is disabled in *Permanent Conferences*.
- If participants disconnect from the *Permanent Conference*, resources are released.
- *Entry Queues*, *Conference Reservations* and *SIP Factories* cannot be defined as *Permanent Conferences*.
- Additional participants can connect to the conference, or be added by the operator, if sufficient resources are available.
- The maximum size of the *Call Detail Record* (*CDR*) for a *Permanent Conference* is 1MB.

Enabling a Permanent Conference

The *Permanent Conference* option is selected in the *New Conference, New Meeting Room* or *New Conference Templates* dialog boxes.



Display Name: SUPPORT_768250229
Duration: 1 : 00 □ Permanent Conference

Display Name: SUPPORT_1533332172
Duration: 1 : 00 □ Permanent Conference
Routing Name:
Profile:
ID:
Conference Password:
Chairperson Password:
Maximum Number of Participants:

Display Name: SUPPORT_1451806447
Duration: 1 : 00 □ Permanent Conference
Routing Name:
Profile: CP+SVC
ID:
Conference Password:
Chairperson Password:
Maximum Number of Participants: Automatic

4

Video Protocols and Resolution Configuration for CP Conferencing

Video Resolutions in AVC-based CP Conferencing

> The following video resolution information applies to AVC Conferencing Mode.
> For a description of resolutions for SVC Conferencing Mode see **"SVC-based Conferencing"**.

The Polycom® RealPresence® Collaboration Server always attempts to connect to endpoints at the highest line rate defined for the conference. If the connection cannot be established using the conference line rate, the Collaboration Server attempts to connect at the next highest line rate at its highest supported resolution.

Depending on the line rate, the Collaboration Server sends video at the best possible resolution supported by the endpoint regardless of the resolution received from the endpoint.

The video resolution is also defined by the *Video Quality* settings in the *Profile.*

The combination of **frame rate** and **resolution** affects the number of video resources required on the MCU to support the call.

The following resolutions are supported:

CIF	352 x 288 pixels.
SD	720 x 576 pixels.
HD 720p	1280 x 720 pixels.

Video Display with CIF, SD and HD Video Connections

Although any combination of CIF, SD and HD connections is supported in all CP conferences, the following rules apply:

- In a 1X1 *Video Layout*:
  - **SD:** If the speaker transmits CIF, the MCU will send CIF to all participants, including the SD participants. In any other layout the MCU will transmit to each participant at the participant's sending resolution.
  - **HD:** The MCU transmits speaker resolution (including input from HD participants) at up to SD resolution. If 1x1 is the requested layout for the entire duration of the conference, set the conference to HD *Video Switching* mode.
- In asymmetrical *Video Layouts*:
  - **SD:** A participant in the large frame that sends CIF is displayed in CIF.
  - **HD:** Where participants' *video windows* are different sizes, the Collaboration Server transmits HD and receives SD or lower resolutions.

- In panoramic *Video Layouts*:
  - **SD**: Participants that send CIF also receive CIF.
  - **HD**: the Collaboration Server transmits HD and receives SD or lower resolutions, the Collaboration Server scales images from SD to HD resolution.

H.264 High Profile Support in CP Conferences

The *H.264 High Profile* is a new addition to the *H.264* video protocol suite. It uses the most efficient video data compression algorithms to even further reduce bandwidth requirements for video data streams.

Video quality is maintained at bit rates that are up to 50% lower than previously required. For example, a 512Kbps call will have the video quality of a 1Mbps HD call while a 1Mbps HD call has higher video quality at the same (1Mbps) bit rate.

H.264 High-Profile should be used when all or most endpoints support it.

Guidelines

- *H.264 High Profile* is supported in *H.323* and *SIP* networking environments.
- *H.264 High Profile* is supported in *Continuous Presence* conferences at all bit rates, video resolutions and layouts.
- *H.264 High Profile* is the first protocol declared by the Collaboration Server, to ensure that endpoints that support the protocol will connect using it.

  Setting minimum bit rate thresholds that are lower than the default may affect the video quality of endpoints that do not support the *H.264 High Profile.*
- For monitoring purposes, the Collaboration Server and endpoint *H.264 High Profile* capability is listed in the *Participant Properties - H.245* and *SDP* tabs for *H.323* participants and *SIP* participants respectively.
  For more information see "*IP Participant Properties"* on page **12-20**.
- *H.264 High Profile* is not supported:
  - For *Content Sharing*
  - As an *RSS Recording* link
  - With *Video Preview*

CP Conferencing with H.263 4CIF

The video resolution of 4CIF in H.263 endpoints is only supported for line rates of 384 Kbps to 1920 Kbps as shown in Table 4-1.

Table 4-1 Video Quality vs. Line Rate

Endpoint Line Rate Kbps	Video Quality	
	Resolution	Frame Rate
128	CIF	30
256	CIF	30

Table 4-1 Video Quality vs. Line Rate (Continued)

Endpoint Line Rate Kbps	Video Quality	
	Resolution	Frame Rate
384 - 1920+	4CIF	15

The Collaboration Server Web Client supports monitoring of H.263 4CIF information. The H.245 or SDP tab includes the additional information.

The creation of a new H.263 4CIF slide is supported in the IVR Service in addition to the current H.263 IVR slide. If users utilize the default Polycom slides that are delivered with the RealPresence Collaboration Server, the slide's resolution will be as defined in the profile, i.e. SD, HD, CIF, etc.

For more information see "*High Resolution Slides*" on page **16-14**.

H.263 4CIF Guidelines

- H.263 4CIF is supported with *H.323* and *SIP* connection endpoints.
- H.263 4CIF is supported in CP mode only.
- Click & View is supported in H.263 4CIF.
- AES encryption is supported with H.263 4CIF.
- H.263 4CIF is supported in recording by the RSS2000 and other recording devices.
- All video layouts are supported in H.263 4CIF, except 1x1 layout. In a 1x1 layout, the resolution will be CIF.
- For information about Resource Usage see Table 19-7 on page **19-8**.
- H.239 is supported in H.263 4CIF and is based on the same bandwidth decision matrix as for HD.

The CP Resolution Decision Matrix

All the CP resolution options and settings are based on a decision matrix which matches video resolutions to connection line rates, with the aim of providing the best balance between resource usage and video quality at any given line rate.

The following factors affect the decision matrices:

- The video protocol used: *H.264 base Profile* or H*.264 High Profile*. The *H.264 High Profil*e maintains the Video quality at bit rates that are up to 50% lower than previously required. For example, a 512 kbps call will have the video quality of a 1Mbps HD call while a 1Mbps HD call has higher video quality at the same (1Mbps) bit rate.

By default, the system shipped with three pre-defined settings of the decision matrix for H.264 *Base Profile* and three pre-defined settings of the decision matrix for *H.264 High Profile*:

- **Resource-Quality Balanced (default)**

  A balance between video quality and resource usage.
- **Resource Optimized**

  System resource usage is optimized by allowing high resolution connections only at high line rates and may result in lower video resolutions (in comparison to other resolution configurations) for some line rates. This option allows to save MCU resources and increase the number of participant connections.

- **Video Quality Optimized**

  Video is optimized through higher resolution connections at lower line rates increasing the resource usage at lower line rates. This may decrease the number of participant connections.

Video Resource Usage

Video resource usage is dependent on the participant's line rate, resolution and *Video Quality* settings.

H.264 Base Profile and High Profile Comparison

The following illustrations show a comparison between the resolutions used at various line rates for H.264 Baseline and the H.264 High Profile Video Quality setting.

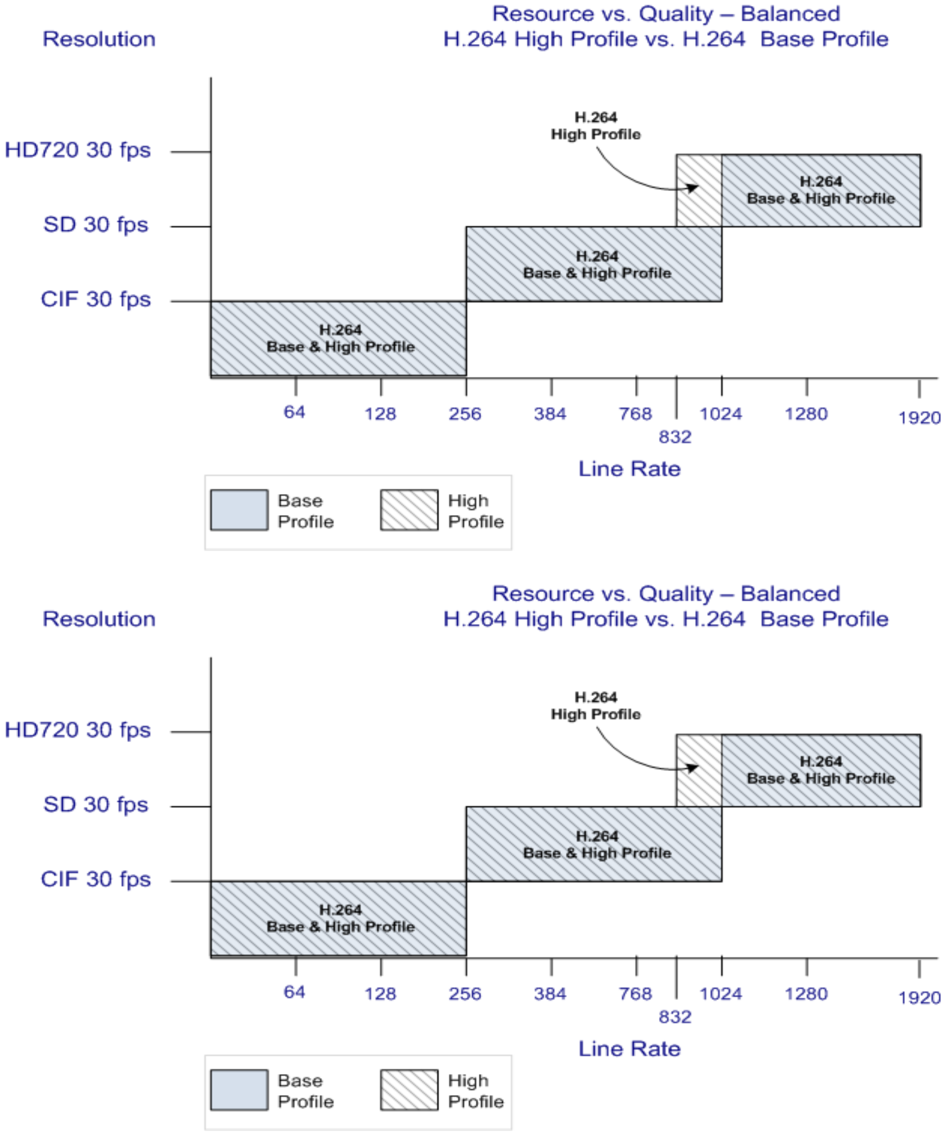
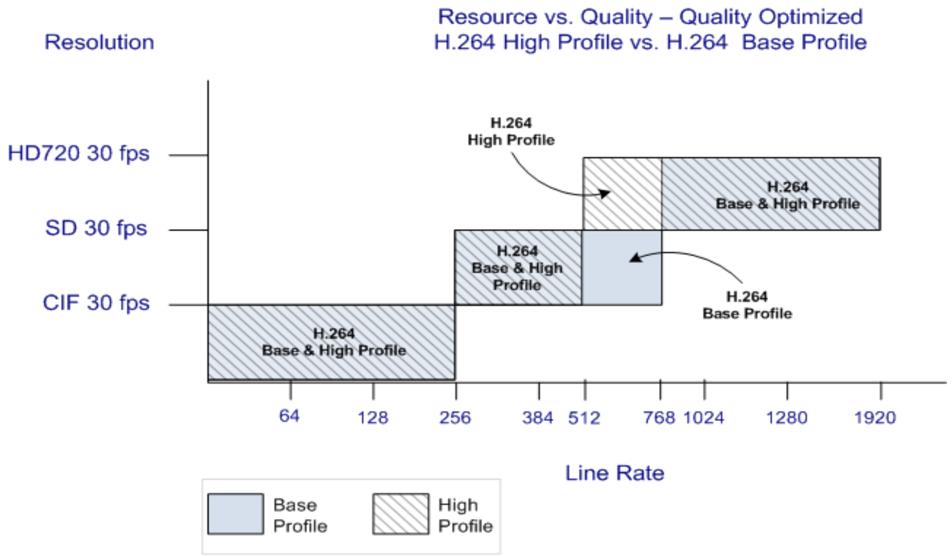


Figure 4-1 *Resolution usage for H.264 High Profile and H.264 Base Profile at various line rates when Resolution Configuration is set to Resource-Quality Balanced*



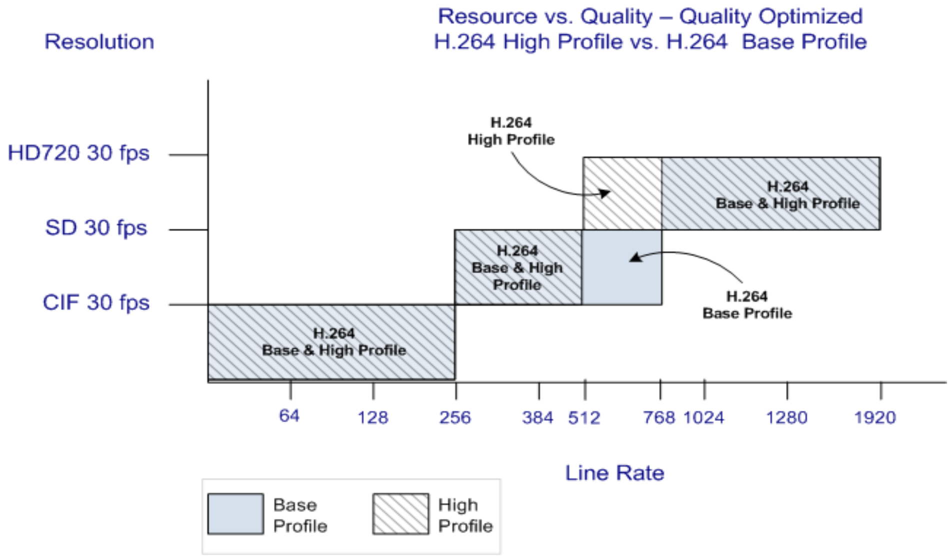


Figure 4-2 *Resolution usage for H.264 High Profile and H.264 Base Profile at various line rates when Resolution Configuration is set to Video Quality Optimized*

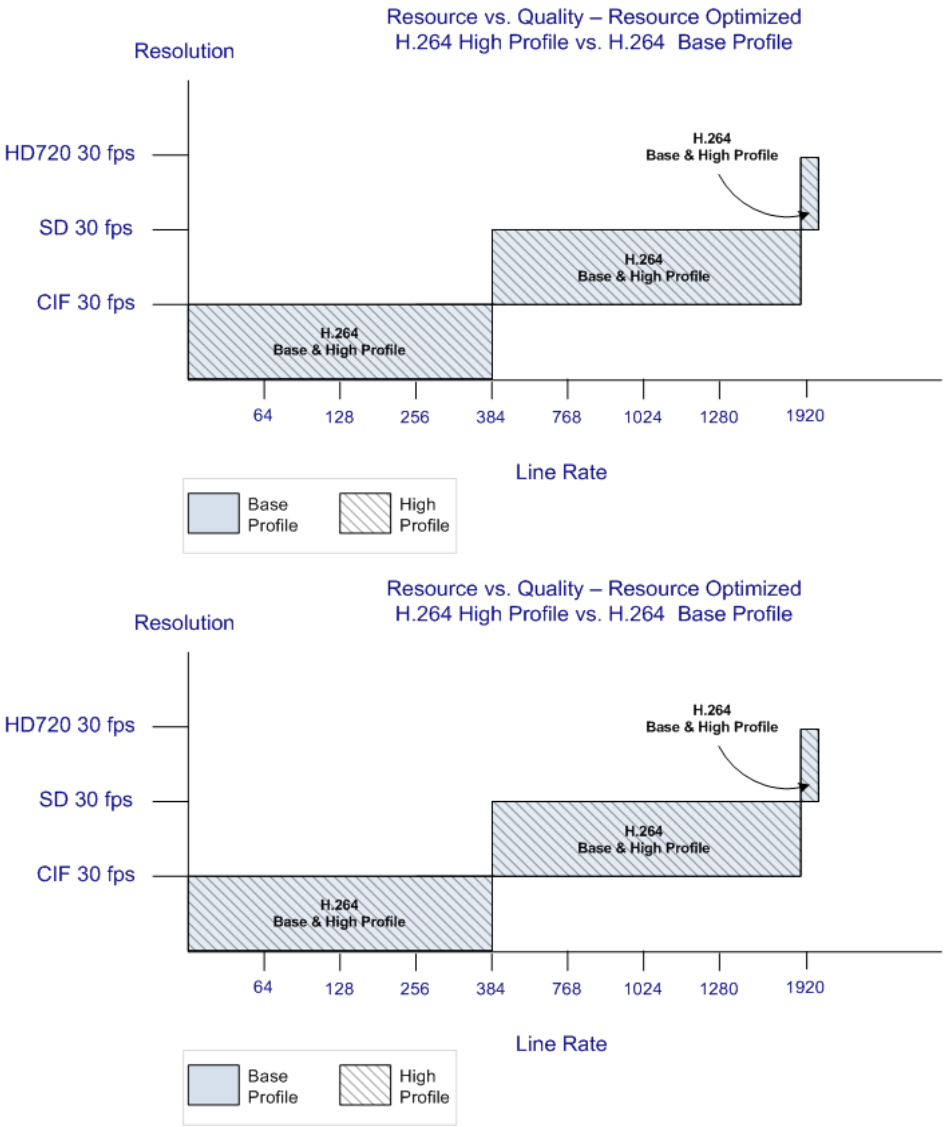
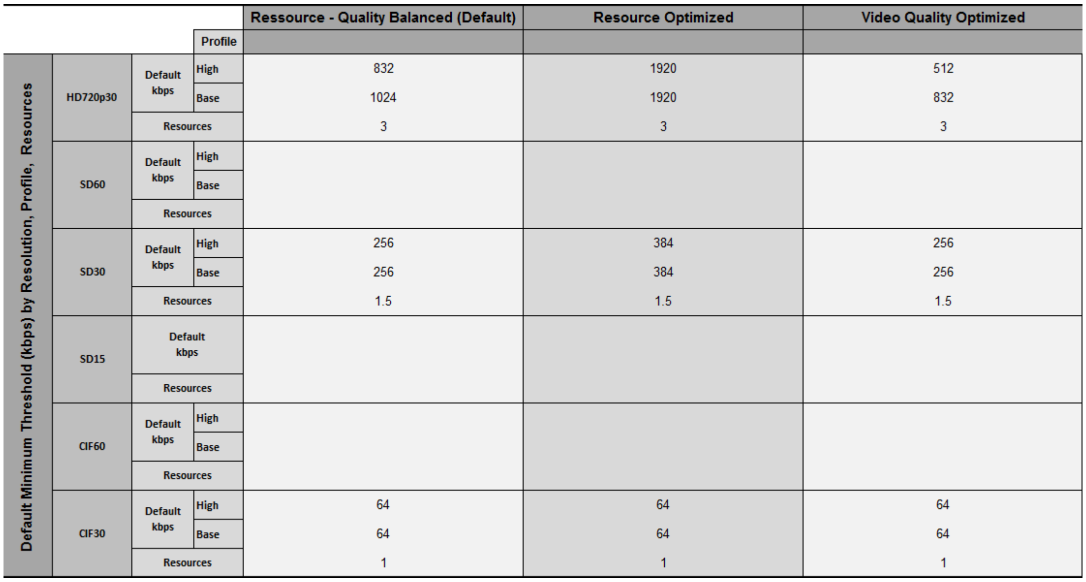


Figure 4-3 Resolution usage for H.264 High Profile and H.264 Base Profile at various line rates when Resolution Configuration is set to Resource Optimized

Default Minimum Threshold Line Rates and Resource Usage Summary

The following Table summarizes the *Default Minimum Threshold Line Rates* and *Video Resource* usage for each of the pre-defined optimization settings for each *Resolution*, *H.264 Profile*, *Video Quality* setting.



				Ressource - Quality Balanced (Default)	Resource Optimized	Video Quality Optimized
			Profile			
Default Minimum Threshold (kbps) by Resolution, Profile, Resources	HD720p30	Default kbps	High	832	1920	512
			Base	1024	1920	832
		Resources		3	3	3
	SD60	Default kbps	High			
			Base			
		Resources				
	SD30	Default kbps	High	256	384	256
			Base	256	384	256
		Resources		1.5	1.5	1.5
	SD15	Default kbps				
		Resources				
	CIF60	Default kbps	High			
			Base			
		Resources				
	CIF30	Default kbps	High	64	64	64
			Base	64	64	64
		Resources		1	1	1

Table 4-2 *Default Minimum Threshold Line Rates and Video Resource Usage*

Resolution			**Optimization Mode**		
		Profile	**Balanced**	**Resource**	**Video Quality**
HD720p30	Default kbps	High	832	1920	512
		Base	1024	1920	832
	Resources		2	2	2
SD 30	Default kbps	High	256	384	256
		Base	256	384	256
	Resources		1	1	1
CIF 30	Default kbps	High	64	64	64
		Base	64	64	64
	Resources		1	1	1

The table above lists resource consumption for *H.264*:

- CIF resolution consumes 1 resources.
- 4CIF resolution consumes 1 resources.
- HD720p resolution consumes 2 resources.

Resolution Configuration for CP Conferences

The *Resolution Configuration* dialog box enables Collaboration Server administrators to override the default video resolution decision matrix, effectively creating their own decision matrix. The minimum threshold line rates at which endpoints are connected at the various video resolutions can be optimized by adjusting the resolution sliders.

System resource usage is also affected by the *Resolution Configuration* settings. For more information see "*Video Resource Usage"* on page **4-4** and "*Default Minimum Threshold Line Rates and Resource Usage Summary"* on page **4-6**.

Guidelines

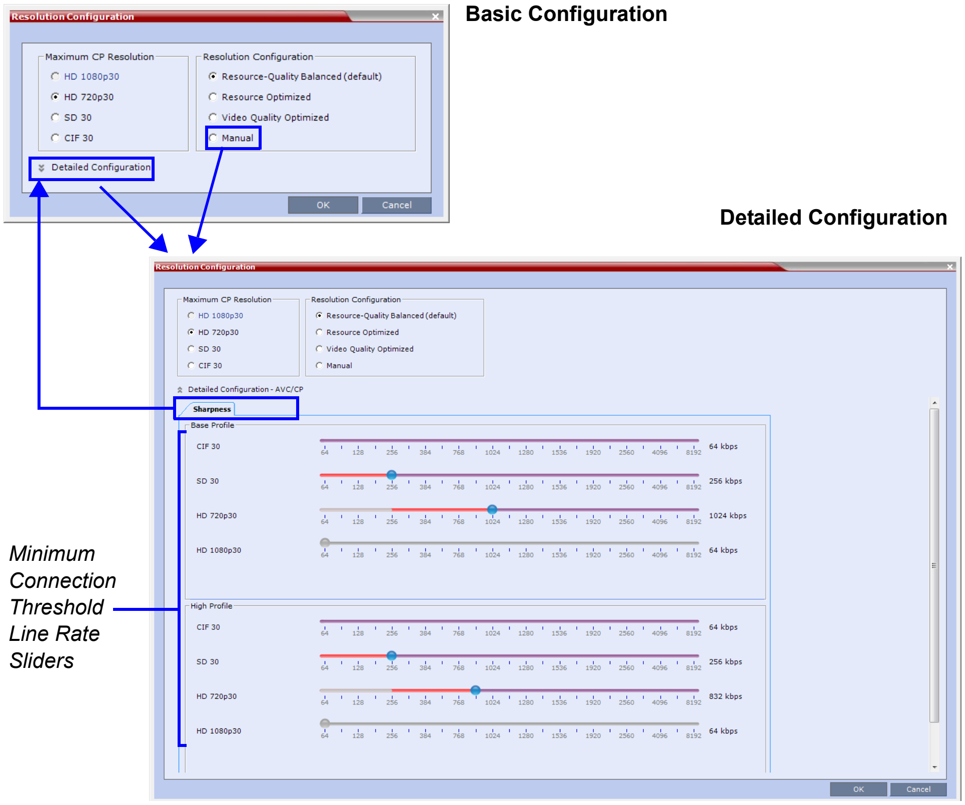
- *Resolution Slider* settings affect all *Continuous Presence* (*CP*) conferences running on the Collaboration Server. *Video Switched* conferences are not affected.
- A system restart is not needed after changing the *Resolution Slider* settings.
- *Resolution Slider* settings cannot be changed if there are ongoing conferences running on the Collaboration Server.

Modifying the Resolution Configuration

The *Resolution Configuration* dialog box is accessed by clicking **Setup > Resolution Configuration** in the *Collaboration Server Setup* menu.

Clicking the **Detailed Configuration** button toggles the display of the *Detailed Configuration* pane, which displays sliders for modifying minimum connection threshold line rates for endpoints that support *H.264 Base Profile* or *High Profile.*

The *Detailed Configuration* pane can also be opened by clicking the **Manual** radio button in the *Resolution Configuration* pane.



Basic Configuration

Resolution Configuration

Maximum CP Resolution
○ HD 1080p30
● HD 720p30
○ SD 30
○ CIF 30

Resolution Configuration
● Resource-Quality Balanced (default)
○ Resource Optimized
○ Video Quality Optimized
○ Manual

Detailed Configuration

Detailed Configuration

Resolution Configuration

Maximum CP Resolution
○ HD 1080p30
● HD 720p30
○ SD 30
○ CIF 30

Resolution Configuration
● Resource-Quality Balanced (default)
○ Resource Optimized
○ Video Quality Optimized
○ Manual

Detailed Configuration - AVC/CP

Sharpness

Base Profile
CIF 30 64 kbps
SD 30 256 kbps
HD 720p30 1024 kbps
HD 1080p30 64 kbps

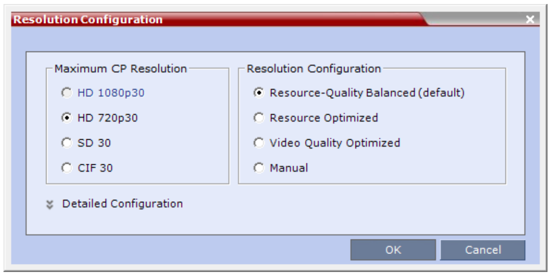
High Profile
CIF 30 64 kbps
SD 30 256 kbps
HD 720p30 832 kbps
HD 1080p30 64 kbps

Minimum Connection Threshold Line Rate Sliders

Resolution Configuration - Basic

The *Resolution Configuration -Basic* dialog box contains the following panes:

- *Max CP Resolution Pane*
- *Resolution Configuration Pane*



Resolution Configuration

Maximum CP Resolution
○ HD 1080p30
● HD 720p30
○ SD 30
○ CIF 30

Resolution Configuration
● Resource-Quality Balanced (default)
○ Resource Optimized
○ Video Quality Optimized
○ Manual

Detailed Configuration

Maximum CP Resolution Pane

The Collaboration Server can be set to one of the following *Maximum CP Resolutions*:

- HD 720p30
- SD 30
- CIF 30

Limiting Maximum Resolution

Before a selection is made in this pane, the *Maximum CP Resolution* of the system is determined by the MAX_CP_RESOLUTION *System Flag*.

Resolution Configuration - Detailed

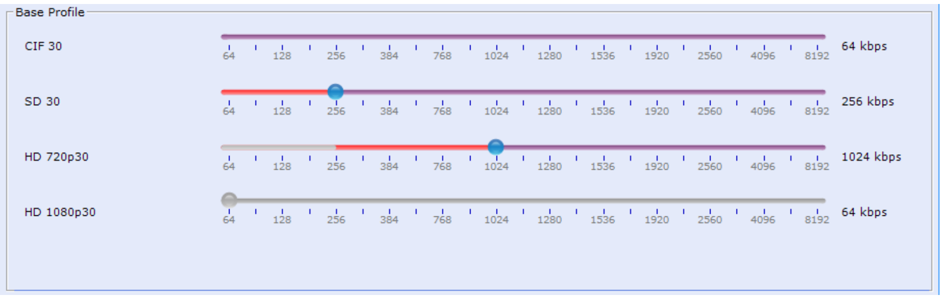
H.264 High Profile allows higher quality video to be transmitted at lower bit rates.

However, setting minimum bit rate thresholds that are lower than the default may affect the video quality of endpoints that do not support the *H.264 High Profile*. The Collaboration Server uses two decision matrices (*Base Profile*, *High Profile)* to enable endpoints to connect according to their capabilities.

Resolution Configuration Sliders

The *Detailed Configuration* dialog box allows the administrator to configure minimum connection threshold bit rates for endpoints that support *H.264 High Profile* and those that do not support *H.264 High Profile* by using the following slider panes:

- Base Profile - Endpoints that do not support H.264 High Profile connect at these minimum threshold bit rates.



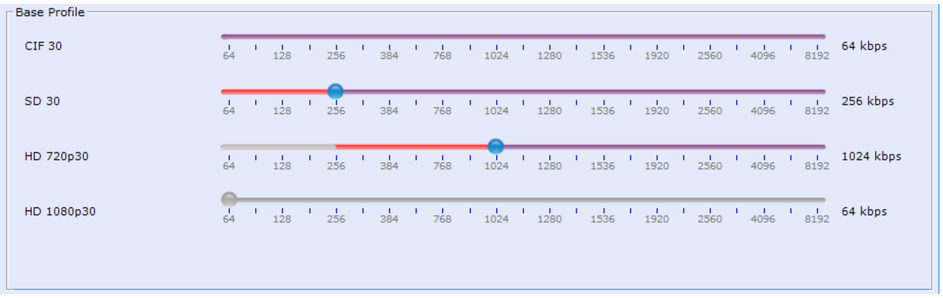
Base Profile

CIF 30 64 kbps

SD 30 256 kbps

HD 720p30 1024 kbps

HD 1080p30 64 kbps



Base Profile

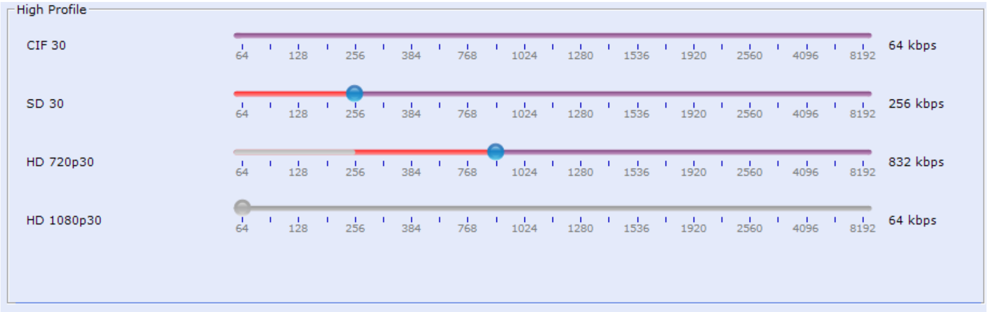
CIF 30 64 kbps

SD 30 256 kbps

HD 720p30 1024 kbps

HD 1080p30 64 kbps

- *High Profile* - Endpoints that support *H.264 High Profile* connect at these minimum threshold bit rates.



High Profile

CIF 30 — 64 kbps

SD 30 — 256 kbps

HD 720p30 — 832 kbps

HD 1080p30 — 64 kbps

High Profile

CIF 30 — 64 kbps

SD 30 — 256 kbps

HD 720p30 — 832 kbps

HD 1080p30 — 64 kbps

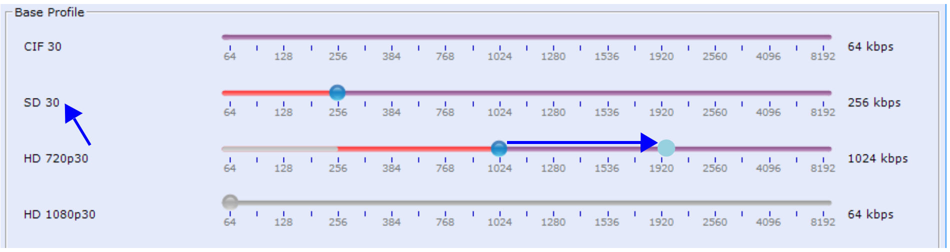
Although the default minimum threshold bit rates provide acceptable video quality, the use of higher bit rates usually results in better video quality.

These *Video Quality* settings are selected per conference and are defined in the conference *Profile* and they determine the resolution matrix that will be applied globally to all conferences. The resolution matrix is determined by the resolution configuration and can be viewed in the *Resolution Configuration* sliders.

System Resource usage is affected by the *Resolution Configuration* settings.

Example

As shown in following diagram:



Base Profile

CIF 30 — 64 kbps

SD 30 — 256 kbps

HD 720p30 — 1024 kbps

HD 1080p30 — 64 kbps

- Moving the *HD720p30* resolution slider from 1024kbps to 1920kbps increases the minimum connection threshold line rate for that resolution. Endpoints connecting at line rates between 1024kbps and 1920kbps that would have connected at *HD 720p30* resolution will instead connect at *SD 30* resolution. Each of the affected endpoints will connect at lower resolution but will use 1 system resource instead of 2 system resources.

Flag Settings

Setting the Maximum CP Resolution for Conferencing

The **MAX_CP_RESOLUTION** flag value is applied to the system during *First-time Power-up* and after a system upgrade. The default value is *HD720p30*.

All subsequent changes to the *Maximum CP Resolution* of the system are made by selections in the *Max Resolution* pane of the ***Resolution Configuration*** dialog box.

The Collaboration Server can be set to one of the following resolutions:

- HD720p30
- SD 30
- CIF 30

Minimum Frame Rate Threshold for SD Resolution

The **MINIMUM_FRAME_RATE_THRESHOLD_FOR_SD** *System Flag* can be added and set to prevent low quality, low frame rate video from being sent to endpoints by ensuring that an *SD* channel is not opened at frame rates below the specified value. For more information see "*Modifying System Flags"* on page **20-1**.

Additional Video Resolutions

The following higher video quality resolutions are available:

- CIF 352 x 288 pixels at 50 fps.
- WCIF 512 x 288 pixels at 50 fps.
- WSD 848 x 480 pixels at 50 fps.
- W4CIF 1024 x 576 pixels at 30 fps.
- HD 720p 1280 x 720 pixels at 30fps

Additional Intermediate Video Resolutions

Two higher quality, intermediate video resolutions replace the transmission of CIF (352 x 288 pixels) or SIF (352 x 240 pixels) resolutions to endpoints that have capabilities between:

- **CIF** (352 x 288 pixels) and **4CIF** (704 x 576 pixels) – the resolution transmitted to these endpoints is **432 x 336** pixels.
- **SIF** (352 x 240 pixels) and **4SIF** (704 x 480 pixels) – the resolution transmitted to these endpoints is **480 x 352** pixels.
- The frame rates (depending on the endpoint's capability) for both intermediate resolutions are 25 or 30 fps.

Microsoft RTV Video Protocol Support in CP Conferences

Microsoft RTV (*Real Time Video*) protocol provides high quality video conferencing capability to *Microsoft OC* (*Office Communicator*) Client endpoints at resolutions up to *HD720p30*. Interoperability between *Polycom HDX* and *OCS* endpoints is improved.

Guidelines

- The *RTV* protocol is supported:
  - — In *SIP* networking environments only
  - — In CP mode only
- *OCS (Wave 13)* and *Lync Server (Wave 14)* clients are supported.
- *RTV* is supported in *Basic Cascade* mode.
- *RTV* is the default protocol for *OCS* endpoints and *Lync Server* clients connecting to a conference.
- *RTV* participants are supported in recorded conferences.
- *RTV* participant encryption is supported using the *SRTP* protocol.
- *Video Preview* is not supported for *RTV* endpoints.
- *Custom Slides* in *IVR Services* are not supported for *RTV* endpoints.

- *HD720p30* resolution is supported at bit rates greater than 600 kbps. The following table summarizes the resolutions supported at the various bit rates.

Table 4-3 *RTV - Resolution by Bit Rate*

Resolution	Bitrate
QCIF	**Bitrate** <180kbps
CIF30	180kbps < **Bitrate** < 250kbps
VGA (SD30)	250kbps < **Bitrate** < 600kbps *
HD720p30	600kbps < **Bitrate** *

* Dependant on the PC’s capability

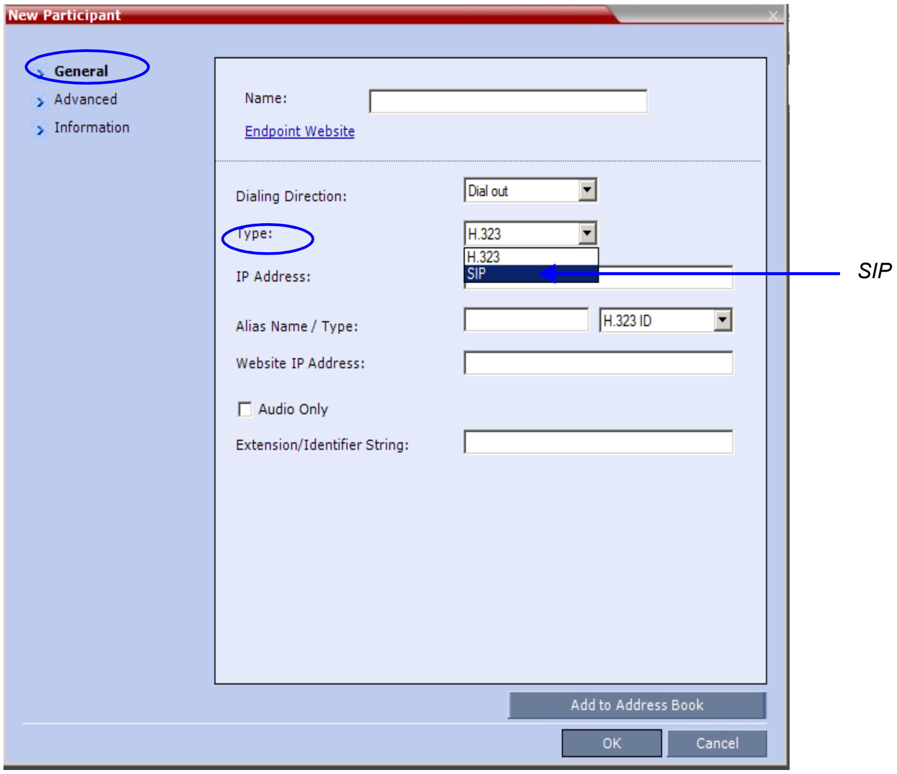
- *System Resource* usage is the same as for the *H.264* protocol. Table 4-4 summarizes *System Resource* usage for each of the supported resolutions.

Table 4-4 *RTV - Resources by Resolution*

Resolution	HD Video Resources Used
QCIF / CIF30	0.333
VGA (SD30) / W4CIF	0.5
HD720p30	1

Participant Settings

When defining a new participant or modifying an existing participant, select **SIP** as the participant's networking environment *Type* in the *New Participant* or *Participant Properties - General* tab.



New Participant

Name:

Dialing Direction: Dial out

Type: H.323

IP Address:

Alias Name / Type: H.323 ID

Website IP Address:

□ Audio Only

Extension/Identifier String:

The participants *Video Protocol* in the *New Participant* or *Participant Properties - Advanced* tab should be left at (or set to) its default value: **Auto**.

The **Auto** setting allows the video protocol to be negotiated according to the endpoint's capabilities:

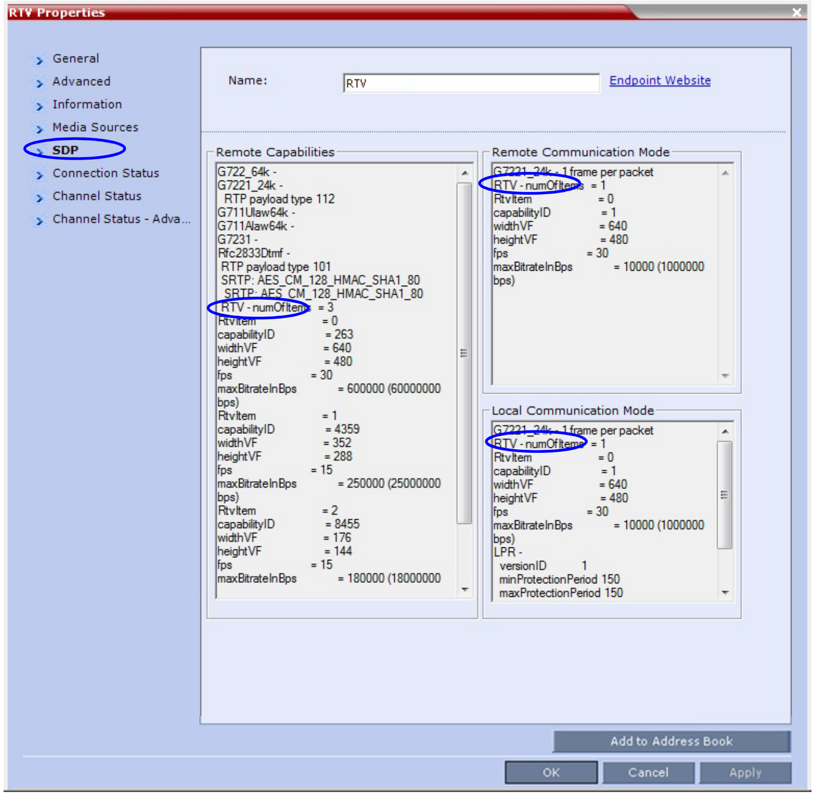
- *OCS* endpoints and *Lync Server* clients connect to the conference using the *RTV* protocol.
- Other endpoints negotiate the video protocol in the following sequence: *H.264*, followed by *RTV*, followed by *H.263* and finally *H.261*.

Protocol Forcing

Selecting *H.264*, *RTV*, *H.263* or *H.261* as the *Video Protocol* results in endpoints that do not support the selected *Video Protocol* connecting as *Secondary* (audio only).

Monitoring RTV

RTV information appears in all three panes of the *Participant Properties* - SDP tab.



Name: RTV

Remote Capabilities

```
G722_64k -
G7221_24k -
 RTP payload type 112
G711Ulaw64k -
G711Alaw64k -
G7231 -
Rfc2833Dtmf -
 RTP payload type 101
 SRTP: AES_CM_128_HMAC_SHA1_80
  SRTP: AES_CM_128_HMAC_SHA1_80
 RTV - numOfItems  = 3
RtvItem              = 0
capabilityID          = 263
widthVF              = 640
heightVF              = 480
fps               = 30
maxBitrateInBps          = 600000 (60000000
bps)
RtvItem              = 1
capabilityID          = 4359
widthVF              = 352
heightVF              = 288
fps               = 15
maxBitrateInBps          = 250000 (25000000
bps)
RtvItem              = 2
capabilityID          = 8455
widthVF              = 176
heightVF              = 144
fps               = 15
maxBitrateInBps          = 180000 (18000000
```

Remote Communication Mode

```
G7221_24k - 1 frame per packet
RTV - numOfItems  = 1
RtvItem              = 0
capabilityID          = 1
widthVF              = 640
heightVF              = 480
fps               = 30
maxBitrateInBps          = 10000 (1000000
bps)
```

Local Communication Mode

```
G7221_24k - 1 frame per packet
RTV - numOfItems  = 1
RtvItem              = 0
capabilityID          = 1
widthVF              = 640
heightVF              = 480
fps               = 30
maxBitrateInBps          = 10000 (1000000
bps)
LPR -
  versionID        1
  minProtectionPeriod 150
  maxProtectionPeriod 150
```

Controlling Resource Allocations for Lync Clients Using RTV Video Protocol

The number of resources used by the system to connect a Lync client with RTV is determined according to the conference line rate and the Maximum video resolution set in the *Conference Profile*.

The system flag **MAX_RTV_RESOLUTION** enables you to override the Collaboration Server resolution selection and limit it to a lower resolution. Resource usage can then be minimized the 1 or 1.5 video resources per call instead of 3 resources, depending on the selected resolution.
Possible flag values are: **AUTO** (default), **QCIF**, **CIF**, **VGA** or **HD720**.

For example, if the flag is set to VGA, conference line rate is 1024Kbps, and the Profile Maximum Resolution is set to Auto, the system will limit the Lync RTV client to a resolution of VGA instead of HD720p and will consume only 1.5 video resources instead of 3 resources.

When set to **AUTO** (default), the system uses the default resolution matrix based on the conference line rate.

To change the default flag setting, add the MAX_RTV_RESOLUTION flag to the *System Configuration* flags and set its value. For information, see .

The following table summarizes the Collaboration Server resources allocated to a Lync Client based on the MAX_RTV_RESOLUTION flag setting, the connection line rate and the video resolution.

Table 4-5 Selected video resolution based on flag setting and conference line rate and core processor

Maximum Resolution Value	Line Rate	Selected Video Resolution Per Core Processor		
		Quad	**Dual**	**Single**
AUTO	> 600 kbps	HD720p 30fps	VGA 30fps	VGA 15fps
	250 kbps - 600 kbps	VGA 30fps	VGA 30fps	VGA 15fps
	180 kbps - 249 kbps	CIF	CIF	CIF
	64 kbps - 179 kbps	QCIF	QCIF	QCIF
HD720p	> 600 kbps	HD720p 30fps	HD720p 13fps	VGA 15fps
	250 kbps - 600 kbps	VGA 30fps	VGA 30fps	VGA 15fps
	180 kbps - 249 kbps	CIF	CIF	CIF
	64 kbps - 179 kbps	QCIF	QCIF	QCIF
VGA	> 600 kbps	VGA 30fps	VGA 30fps	VGA 15fps
	250 kbps - 600 kbps	VGA 30fps	VGA 30fps	VGA 15fps
	180 kbps - 249 kbps	CIF	CIF	CIF
	64 kbps - 179 kbps	QCIF	QCIF	QCIF
CIF	> 600 kbps	CIF	CIF	CIF
	250 kbps - 600 kbps	CIF	CIF	CIF
	180 kbps - 249 kbps	CIF	CIF	CIF
	64 kbps - 179 kbps	QCIF	QCIF	QCIF
QCIF	> 600 kbps	QCIF	QCIF	QCIF
	250 kbps - 600 kbps	QCIF	QCIF	QCIF
	180 kbps - 249 kbps	QCIF	QCIF	QCIF
	64 kbps - 179 kbps	QCIF	QCIF	QCIF

When the MAX_ALLOWED_RTV_HD_FRAME_RATE flag equals 0 (default value), Table 1-1 for the MAX_RTV_RESOLUTION flag applies. When the MAX_ALLOWED_RTV_HD_FRAME_RATE flag does not equal 0, see "*Threshold HD Flag Settings using the RTV Video Protocol”* on page **3-30** for more information.

The following table describes the number of allocated video resources for each video resolution when using the RTV protocol.

Table 4-6 Allocated video resolutions per video resolution

Selected Video Resolution	Number of Allocated Video Resources
HD720p	3
VGA	1.5
CIF	1
QCIF	1

Threshold HD Flag Settings using the RTV Video Protocol

The system flag **MAX_ALLOWED_RTV_HD_FRAME_RATE** defines the threshold Frame Rate (fps) in which RTV Video Protocol initiates HD resolutions.

Flag values are as follows:

- Default: **0** (fps) - Implements any Frame Rate based on Lync RTV Client capabilities

If the MAX_RTV_RESOLUTION flag is set to AUTO dual core systems always view VGA. For more information on Lync RTV Client capabilities, see , "*Controlling Resource Allocations for Lync Clients Using RTV Video Protocol*" on page 3-23 for more information.

- Range: **0-30** (fps)

For example, when the flag is set to 15 and the Lync RTV Client declares HD 720P at 10fps, because the endpoint's frame rate (fps) of 10 is less than flag setting of 15, then the endpoint's video will open VGA and not HD.

In another example, when the flag is set to a frame rate of 10 and the Lync RTV Client declares HD 720P at 13fps, because the endpoint's frame rate (fps) of 13 is greater than flag setting of 10, then the endpoint's video will open HD and not VGA.

- Single core PC's cannot view HD and always connect in VGA.
- Dual Core Processor - The threshold for flag settings on Dual Core systems is 13 (fps) and less for viewing HD. When system flag is set to 14 (fps) or higher, the RTV Video Protocol shall connect in VGA.
- Quad Core PC systems always view HD, even when flag settings are set anywhere from to 0-30.
- The number of resources used by the system to connect a Lync client with RTV is determined according to the conference line rate and the maximum video resolution set in the Conference Profile. For more information, see "*Microsoft RTV Video Protocol Support in CP Conferences*" on page 3-20.

5

Cascading Conferences

> Cascading information applies to AVC Conferencing Mode (CP and mixed CP and SVC) only. Cascading is not supported with SVC Conferencing Mode.

Cascading enables administrators to connect one conference directly to one or several conferences, depending on the topology, creating one large conference. The conferences can run on the same MCU or different MCUs.

There are many reasons for cascading conferences, the most common are:

- Connecting two conferences on different MCUs at different sites.
- Utilizing the connection abilities of different MCUs, for example, different communication protocols.

The following cascading topologies are available for cascading:

- **Basic Cascading** - only two conferences are connected (usually running on two different Collaboration Servers). The cascaded MCUs reside on the same network.
- **Star Cascading** - one or several conferences are connected to one master conference. Conferences are usually running on separate MCUs. The cascaded MCUs reside on the same network.

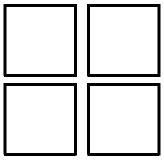
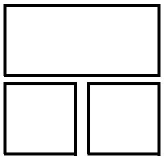
System configuration and feature availability change according to the selected cascading topology.

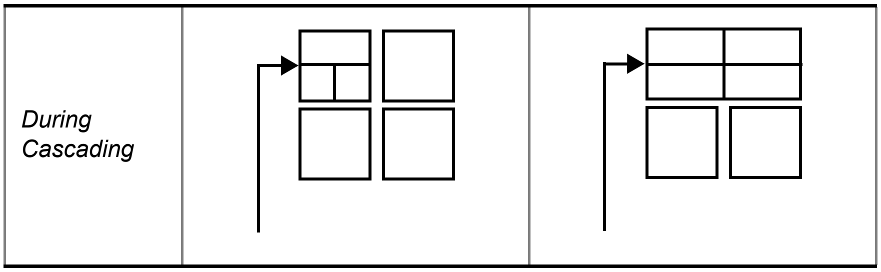
Video Layout in Cascading conferences (CP and mixed CP and SVC)

Cascade links are treated as endpoints in CP conferences and are allocated resources according to "*Resolution Configuration for CP Conferences"* on page 4-8. Cascaded links in 1x1 video layout are in SD resolution.

When cascading two conferences, the video layout displayed in the cascaded conference is determined by the selected layout in each of the two conferences. Each of the two conferences will inherit the video layout of the other conference in one of their windows.

In order to avoid cluttering in the cascaded window, it is advised to select appropriate video layouts in each conference before cascading them.

	Conference A	**Conference B**
Without Cascade		



Video layout of Conf. B in window of Conf. A

Video layout of Conf. A in window of Conf. B

Figure 5-1 Video Layouts in Cascaded Conferences

Guidelines

To ensure that conferences can be cascaded and video can be viewed in all conferences the following guidelines are recommended:

- The same version installed on all MCUs participating the cascading topology
- The same license installed on all MCUs participating the cascading topology
- Same Conference Parameters are defined in the Profile of the conferences participating in the cascading topology
  - Conference line rates should be identical
  - Content rate should be identical
  - Same encryption settings
- DTMF codes should be defined with the same numeric codes in the IVR services assigned to the cascading conferences
- DTMF forwarding is suppressed
- The video layout of the link is set to 1x1 by the appropriate system flag.
- When the Mute Participants Except Lecturer option is enabled in the Conference Profile, all participants (including the link participants) except the lecturer are muted. Only the lecturer is not muted.

Flags controlling Cascade Layouts

- Setting the **FORCE_1X1_LAYOUT_ON_CASCADED_LINK _CONNECTION** *System Flag* to **YES** (default) automatically forces the cascading link to Full Screen (1x1) in CP conferences, hence displaying the speaker of one conference to a full window in the video layout of the other conference.

  Set this flag to **NO** when cascading between an Collaboration Server and an MGC that is functioning as a Gateway, if the participant layouts on the MGC are not to be forced to 1X1.

- Setting the **AVOID_VIDEO_LOOP_BACK_IN_CASCADE** *System Flag* to **YES** (default) prevents the speaker's image from being sent back through the participant link from the cascaded conference. This can occur in cascaded conferences with conference layouts other than 1x1. It results in the speaker's own video image being displayed in the speaker's video layout.

  This option is supported with *Basic Cascading*. If a *Master MCU* has two slave MCUs, participants connected to the slave MCUs will not receive video from each other.

For more details on defining system flags, see "*Modifying System Flags"* on page **20-1**.

Basic Cascading

In this topology, a link is created between two conferences, usually running on two different MCUs. The MCUs are usually installed at different locations (states/countries) to save long distance charges by connecting each participant to their local MCU, while only the link between the two conferences is billed as long distance call.

- This is the only topology that enables IP cascading links:
  - — When linking two conferences using an IP connection, the destination MCU can be indicated by:
    - IP address
    - H.323 Alias
  - — If IP cascading link is used to connect the two conferences, both MCUs must be located in the same network.
- One MCU can be used as a gateway.
- The configuration can include two Collaboration Servers.

Basic Cascading using IP Cascaded Link

In this topology, both MCUs can be registered with the same gatekeeper or the IP addresses of both MCUs can be used for the cascading link. Content can be sent across the Cascading Link.

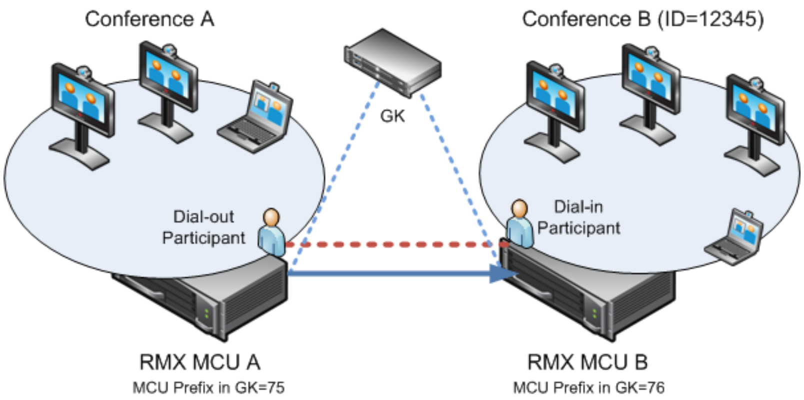


Figure 5-2 *Basic Cascading Topology - IP Cascading Link*

For example, MCU B is registered with the gatekeeper using 76 as the MCU prefix.

The connection between the two conferences is created when a dial out IP participant is defined (added) to conference A whose dial out number is the dial-in number of the conference or Entry Queue running on MCU B.

Dialing Directly to a Conference

Dial out IP participant in conference A dials out to the conference running on MCU B entering the number in the format:
[MCU B Prefix/IP address][conference B ID].

For example, if MCU B prefix is 76 and the conference ID is 12345, the dial number is **7612345**.

Automatic Identification of the Cascading Link

The system automatically identifies that the dial in participant is an MCU and creates a Cascading Link and displays the link icon for the participant (🔗). The master-slave relationship is randomly defined by the MCUs during the negotiation process of the connection phase.

6

Meeting Rooms

A Meeting Room is a conference saved on the MCU in passive mode, without using any of the system resources. A Meeting Room is automatically activated when the first participant dials into it. Meeting Rooms can be activated as many times as required. Once activated, a Meeting Room functions as any ongoing conference.

The conferencing Mode of the Meeting Room is determined by the Profile assigned to it.

In SVC Conferencing Mode, dial-in is available as follows:

- AVC-capable endpoints (participants) can only connect to an AVC CP Meeting Room. When dialing into SVC Only Meeting Room the calls fail.
- SVC-capable endpoints support both AVC and SVC video protocols. When dialing into SVC Only conferences, they connect as SVC endpoints. When dialing into AVC CP Only conferences, they connect as AVC endpoints.
- Both AVC and SVC endpoints can connect to a mixed CP and SVC conference.

In AVC CP Conferences, dial-out participants can be connected to the conference automatically, or manually. In the automatic mode the system calls all the participants one after the other. In the manual mode, the Collaboration Server user or meeting organizer instructs the conferencing system to call the participant. Dial-out participants must be defined (mainly their name) and added to the conference. This mode can only be selected at the conference/Meeting Room definition stage and cannot be changed once the conference is ongoing.

A Meeting Room can be designated as a Permanent Conference. For more information see "*Audio Algorithm Support*" on page **3-45**.

The maximum of number of Meeting Rooms that can be defined is 1000.

The system is shipped with four default Meeting Rooms as shown in Table 6-1.

Table 6-1 *Default Meeting Rooms List*

Meeting Room Name	ID	Default Line Rate
Maple_Room	1001	1920 Kbps
Oak_Room	1002	1920 Kbps
Juniper_Room	1003	1920 Kbps
Fig_Room	1004	1920 Kbps

Meeting Rooms List

Meeting Rooms are listed in the *Meeting Room* list pane.

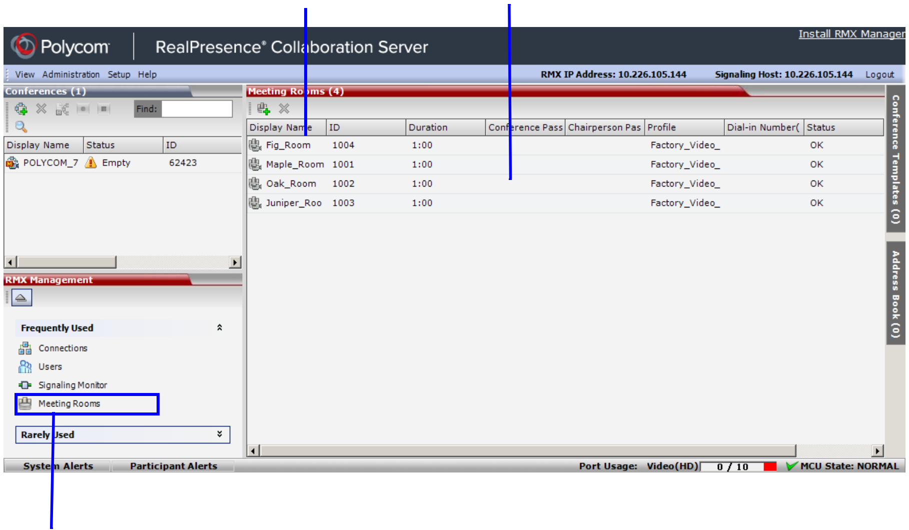
To list Meeting Rooms:

>> In the *RealPresence Collaboration Server Management* pane, in the *Frequently Used* list, click the **Meeting Rooms** button .

The *Meeting Rooms List* is displayed.

Meeting Room Toolbar

Meeting Room List



Display Name	Status	ID
POLYCOM_7	Empty	62423

Display Name	ID	Duration	Conference Pass	Chairperson Pas	Profile	Dial-in Number(	Status
Fig_Room	1004	1:00			Factory_Video_		OK
Maple_Room	1001	1:00			Factory_Video_		OK
Oak_Room	1002	1:00			Factory_Video_		OK
Juniper_Roo	1003	1:00			Factory_Video_		OK

Port Usage: Video(HD) 0 / 10 MCU State: NORMAL

Access to Meeting Rooms

An active Meeting Room becomes an ongoing conference and is monitored in the same way as any other conference.

The *Meeting Room List* columns include:

Table 6-2 *Meeting Rooms List Columns*

Field	Description	
Display Name	Displays the name and the icon of the Meeting Room in the *Collaboration Server Web Client*.	
	(green)	An active video Meeting Room that was activated when the first participant connected to it.
	(gray)	A passive video Meeting Room that is waiting to be activated.
Routing Name	The ASCII name that registers conferences, Meeting Rooms, Entry Queues and SIP Factories in the various gatekeepers and SIP Servers. In addition, the Routing Name is also: • The name that endpoints use to connect to conferences. • The name used by all conferencing devices to connect to conferences that must be registered with the gatekeeper and SIP Servers.	

Table 6-2 *Meeting Rooms List Columns (Continued)*

Field	Description	
ID	Displays the Meeting Room ID. This number must be communicated to H.323 conference participants to enable them to dial in.	
Duration	Displays the duration of the Meeting Room in hours using the format HH:MM (default 01:00).	
Conference Password	The password to be used by participants to access the Meeting Room. If blank, no password is assigned to the conference. This password is valid only in conferences that are configured to prompt for a conference password in the IVR Service.	The Collaboration Server can be configured to automatically generate conference and chairperson passwords when these fields are left blank. For more information, see the "*Automatic Password Generation Flags"* on page 20-37.
Chairperson Password	Displays the password to be used by the users to identify themselves as *Chairpersons.* They are granted additional privileges. If left blank, no chairperson password is assigned to the conference. This password is valid only in conferences that are configured to prompt for a chairperson password.	
Profile	Displays the name of the Profile assigned to the Meeting Room. For more information, see "*Defining New Profiles"* on page 2-9.	
SIP Registration	The status of registration with the SIP server: • **Not configured** - Registration with the SIP Server was not enabled in the Conference Profile assigned to this conferencing Entity. In Multiple Networks configuration, If one service is not configured while others are configured and registered, the status reflects the registration with the configured Network Services. The registration status with each SIP Server can be viewed in the *Properties - Network Services* dialog box of each conferencing entity. When SIP registration is not enabled in the conference profile, the Collaboration Server's registering to SIP Servers will each register with an URL derived from its own signaling address. • **Failed** - Registration with the SIP Server failed. This may be due to incorrect definition of the SIP server in the IP Network Service, or the SIP server may be down, or any other reason the affects the connection between the Collaboration Server or the SIP Server to the network. • **Registered** - the conferencing entity is registered with the SIP Server. • **Partially Registered** - This status is available only in Multiple Networks configuration, when the conferencing entity failed to register to all the required Network Services if more than one Network Service was selected.	

Meeting Room Toolbar & Right-click Menu

The Meeting Room toolbar and right-click menus provide the following functionality:

Table 6-3 Meeting Room Toolbar and Right-click Menus

Toolbar button	Right-click menu	Description
✚	*New Meeting Room*	Select this button to create a new Meeting Room.
✖	*Delete Meeting Room*	Select any Meeting Room and then click this button to delete the Meeting Room.

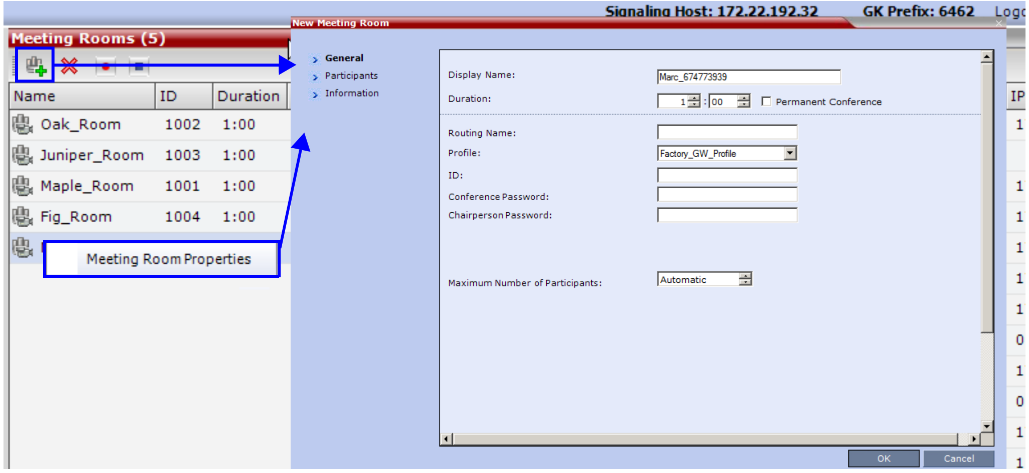
Dial out to AVC participants assigned to a Meeting Room will only start when the dial in participant who has activated it has completed the connection process and the Meeting Room has become an ongoing conference.

Creating a New Meeting Room

To create a new meeting room:

>> In the *Meeting Rooms* pane, click the **New Meeting Room** ✚ button *or* right-click an empty area in the pane and then click **New Meeting Room**.

The *New Meeting Room* dialog box is displayed.



Signaling Host: 172.22.192.32   GK Prefix: 6462

Meeting Rooms (5)

Name	ID	Duration
Oak_Room	1002	1:00
Juniper_Room	1003	1:00
Maple_Room	1001	1:00
Fig_Room	1004	1:00

New Meeting Room

Display Name: Marc_674773939

Duration: 1 : 00 □ Permanent Conference

Routing Name:

Profile: Factory_GW_Profile

ID:

Conference Password:

Chairperson Password:

Maximum Number of Participants: Automatic

The definition procedure is the same as for the new conference.

If SIP Factories are being used do not assign a Meeting Room the ID 7001. This ID is reserved for the default SIP Factory.

For more information, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Starting an AVC CP Conference from the Conferences Pane"* on page 3-13.

Microsoft Lync users can connect a Collaboration Server Meeting Room to a conference running on the Microsoft A/V MCU. This allows Collaboration Server Lync users to connect with a conference in progress on the A/V MCU and be an active participant in the conference.

For more information, see "*Connecting a Collaboration Server Meeting Room to a Microsoft AV-MCU Conference"* on page **G-31**.

7

Entry Queues, Ad Hoc Conferences and SIP Factories

Entry Queues

An Entry Queue (EQ) is a special routing lobby to access conferences. Participants connect to a single-dial lobby and are routed to their destination conference according to the Conference ID they enter. The Entry Queue remains in a passive state when there are no callers in the queue (in between connections) and is automatically activated once a caller dials its dial-in number.

Participants can be moved from the Entry Queue and the destination conference if both conferencing entities are set to the same conferencing parameters: Conferencing Mode, Line rate and video parameters. For example, participants can be moved from SVC Only Entry Queue to SVC Only conference, or from a mixed CP and SVC Entry Queue to a mix CP and SVC conference, from CP only Entry Queue to CP only conference.

The maximum of number of Entry Queues that can be defined is 40.

The parameters (bit rate and video properties) with which the participants connect to the Entry Queue and later to their destination conference are defined in the Conference Profile that is assigned to the Entry Queue. For example, if the Profile Bit Rate is set to 384kbps, all endpoints connect to the Entry Queue and later to their destination conference using this bit rate even if they are capable of connecting at higher bit rates.

An *Entry Queue IVR Service* must be assigned to the Entry Queue to enable the voice prompts guiding the participants through the connection process. The Entry Queue IVR Service also includes a video slide that is displayed to the participants while staying in the Entry Queue (during their connection process).

Different Entry Queues can be created to accommodate different conferencing modes, conferencing parameters (by assigning different Profiles) and prompts in different languages (by assigning different *Entry Queue IVR Services*).

For more information, see "*IVR Services List"* on page **16-1**.

The Entry Queue can also be used for Ad Hoc conferencing. If the Ad Hoc option is enabled for the Entry Queue, when the participant enters the target conference ID the system checks whether a conference with that ID is already running on the MCU. If not, the system automatically creates a new ongoing conference with that ID. For more information about Ad Hoc conferencing, see "*Ad Hoc Conferencing"* on page **7-11**.

An Entry Queue can be designated as Transit Entry Queue to which calls with dial strings containing incomplete or incorrect conference routing information are transferred. For more information, see "*Transit Entry Queue"* on page **7-6**.

Default Entry Queue properties

The system is shipped with a default Entry Queue whose properties are:

Table 7-1 Default Entry Queue Properties

Parameter	Value
Display Name	DefaultEQ The user can change the name if required.
Routing Name	DefaultEQ The default *Routing Name* cannot be changed.
ID	1000
Profile name	Factory_Mixd_CP_SVC_Video_Profile. Profile Bit Rate is set to 1920Kbps.
Entry Queue Service	Entry Queue IVR Service. This is default Entry Queue IVR Service shipped with the system and includes default voice messages and prompts in English.
Ad Hoc	Enabled

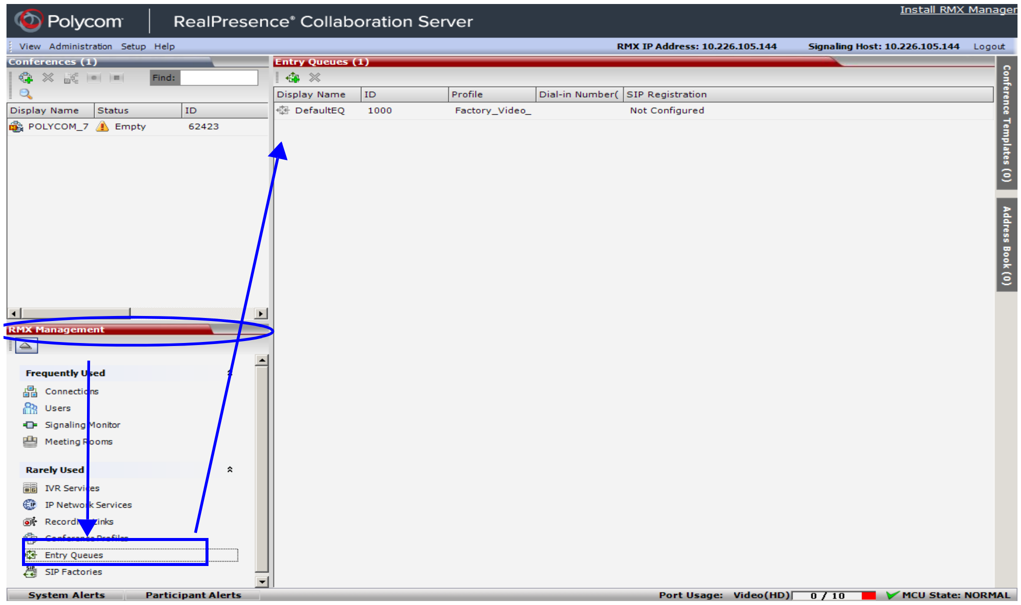
Defining a New Entry Queue

In the *RealPresence CloudAxis Solution*, virtual Entry Queues and ad-hoc conferences are defined in the RealPresence DMA system component and should not be defined directly in the RealPresence Collaboration Server Virtual Edition component.

You can modify the properties of the default Entry Queue and define additional Entry Queues to suit different conferencing requirements.

To define a new Entry Queue:

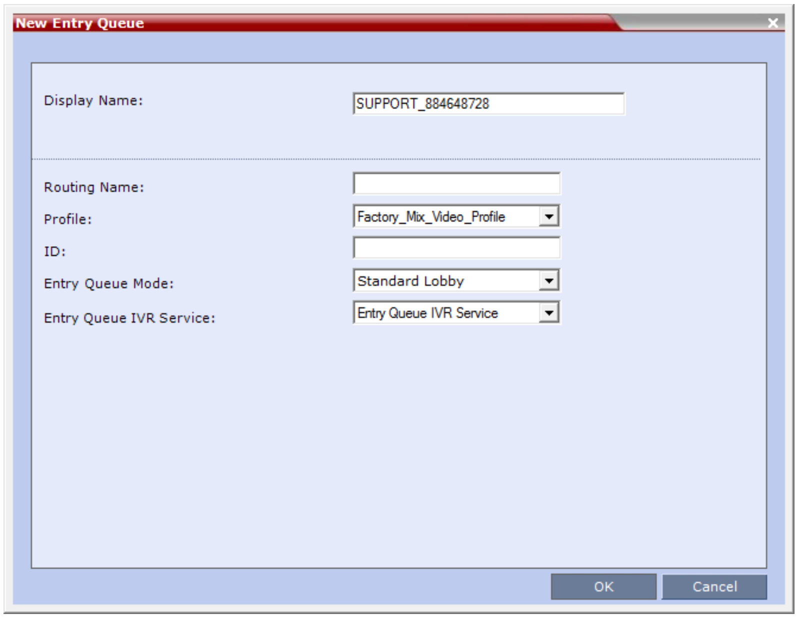
1 In the *RealPresence Collaboration Server Management - Rarely Used* pane, click **Entry Queues**.



Display Name	Status	ID
POLYCOM_7	Empty	62423

Display Name	ID	Profile	Dial-in Number(	SIP Registration
DefaultEQ	1000	Factory_Video_		Not Configured

2 In the *Entry Queues* list pane, click the **New Entry Queue** ✚ button. The *New Entry Queue* dialog box opens.



3 Define the following parameters:

Table 7-2: Entry Queue Definitions Parameters

Option	Description
Display Name	The Display Name is the conferencing entity name in native language character sets to be displayed in the Collaboration Server Web Client. In conferences, Meeting Rooms, Entry Queues and SIP factories the system automatically generates an ASCII name for the *Display Name* field that can be modified using Unicode encoding. • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. The maximum length of text fields also varies according to the mixture of character sets (Unicode and ASCII). Maximum field length in ASCII is 80 characters. If the same name is already used by another conference, Meeting Room or Entry Queue, the Collaboration Server displays an error message requesting you to enter a different name.

Table 7-2: Entry Queue Definitions Parameters (Continued)

Option	Description
Routing Name	Enter a name using ASCII text only. If no *Routing Name* is entered, the system automatically assigns a new name as follows: • If an all ASCII text is entered in *Display Name*, it is used also as the *Routing Name.* • If any combination of Unicode and ASCII text (or full Unicode text) is entered in *Display Name*, the *ID* (such as Conference ID) is used as the *Routing Name.*
Profile	Select the Profile to be used by the Entry Queue. The default Profile is selected by default. This Profile determines the Bit Rate and the video properties with which participants connect to the Entry Queue and destination conference. In Ad Hoc conferencing, it is used to define the new conference properties.
ID	Enter a unique number identifying this conferencing entity for dial in. Default string length is 4 digits. If you do not manually assign the ID, the MCU assigns one after the completion of the definition. The ID String Length is defined by the flag NUMERIC_CONF_ID_LEN in the System Configuration.
Entry Queue Mode	Select the mode for the Entry Queue:
	Standard Lobby (default)-When selected, the Entry Queue is used as a routing lobby to access conferences. Participants connect to a single-dial lobby and are routed to their destination conference according to the Conference ID they enter.
	Ad Hoc - Select this option to enable the Ad Hoc option for this Entry Queue. In this mode, when the participant enters the target conference ID the system checks whether a conference with that ID is already running on the MCU. If not, the system automatically creates a new ongoing conference with that ID.
	IVR Service Provider Only - When selected, designates this Entry Queue as a special Entry Queue that provides IVR Services to SIP calls on behalf of the RealPresence DMA system. The IVR service provider only Entry Queue does not route the SIP calls to a target conference. Instead the RealPresence DMA system handles the call. For more details, see "” on page 7-7.
	External IVR Control - Not Supported with RealPresence Collaboration Server Virtual Edition.
Entry Queue IVR Service	The default Entry Queue IVR Service is selected. If required, select an alternate Entry Queue IVR Service, which includes the required voice prompts, to guide participants during their connection to the Entry Queue.

4 Click **OK**.

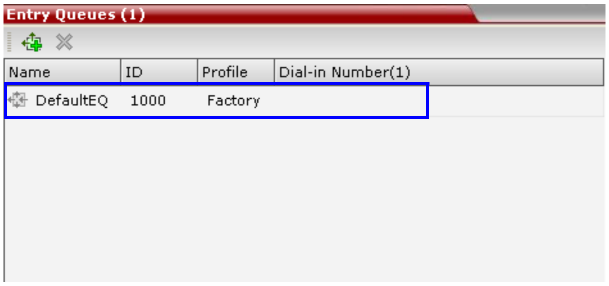
The new *Entry Queue* is added to the *Entry Queues* list.

Listing Entry Queues

To view the list of Entry Queues:

>> In the *RealPresence Collaboration Server Management - Rarely Used* pane, click **Entry Queues**.

The *Entry Queues* are listed in the *Entry Queues* pane.



Name	ID	Profile	Dial-in Number(1)
DefaultEQ	1000	Factory	

You can double-click an Entry Queue to view its properties.

Modifying the EQ Properties

To modify the EQ:

>> In the *Entry Queues* pane, either double-click or right-click and select **Entry Queue Properties** of the selected *Entry Queue* in the list.

The Entry Queue Properties dialog box is displayed. All the fields may be modified except **Routing Name**.

Transit Entry Queue

A *Transit Entry Queue* is an Entry Queue to which calls with dial strings containing incomplete or incorrect conference routing information are transferred.

IP Calls are routed to the *Transit Entry Queue* when:

- A gatekeeper is not used, or where calls are made directly to the Collaboration Server's *Signaling IP Address*, with incorrect or without a Conference ID.
- When a gatekeeper is used and only the prefix of the Collaboration Server is dialed, with incorrect or without a Conference ID.
- When the dialed prefix is followed by an incorrect conference ID.

When no *Transit Entry Queue* is defined, all calls containing incomplete or incorrect conference routing information are rejected by the Collaboration Server.

In the *Transit Entry Queue*, the *Entry Queue IVR Service* prompts the participant for a destination conference ID. Once the correct information is entered, the participant is transferred to the destination conference.

Setting a Transit Entry Queue

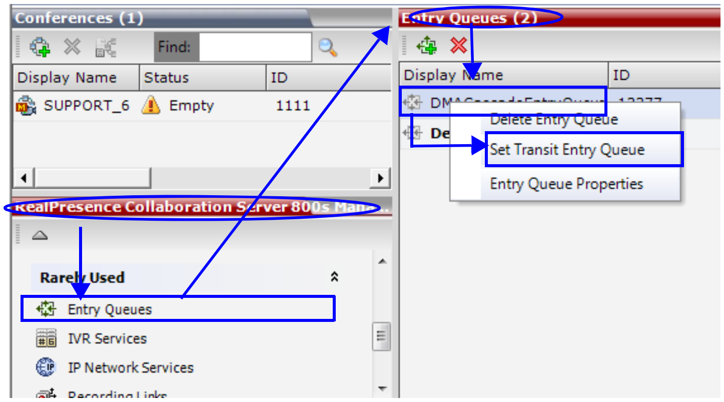
The Collaboration Server factory default settings define the *Default Entry Queue* also as the *Transit Entry Queue*. You can designate another Entry Queue as the *Transit Entry Queue*.

Only one *Transit Entry Queue* may be defined per Collaboration Server and selecting another Entry Queue as the *Transit Entry Queue* automatically cancels the previous selection.

To designate an Entry Queue as Transit Entry Queue:

1 In the *Collaboration Server Management - Rarely Used* pane, click **Entry Queues**.

2 In the *Entry Queues* list, right-click the Entry Queue entry and then click **Set Transit Entry Queue**.

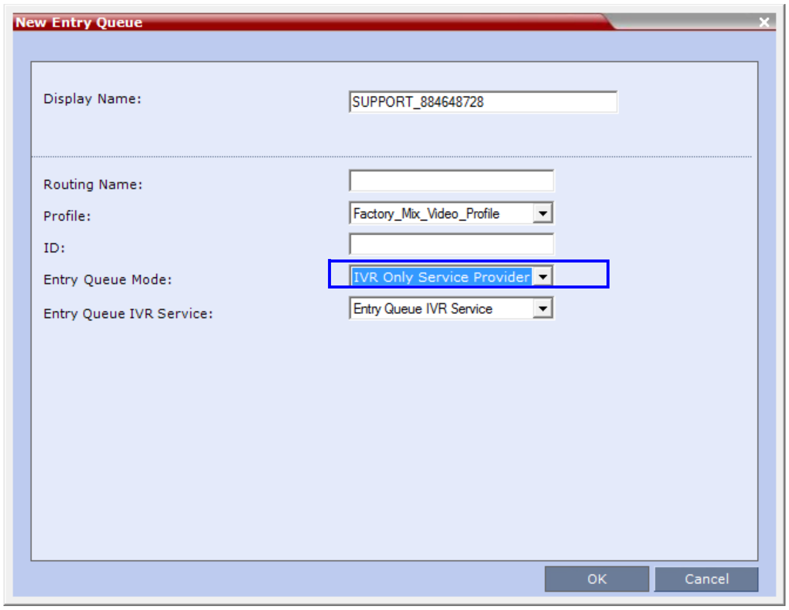


The Entry Queue selected as *Transit Entry Queue* is displayed in bold.

To cancel the Transit Entry Queue setting:

1 In the *Collaboration Server Management - Rarely Used* pane click **Entry Queues**.

2 In the *Entry Queues* list, right-click the *Transit Entry Queue* entry and then click **Cancel Transit Entry Queue**.



New Entry Queue

Display Name: SUPPORT_884648728

Routing Name:

Profile: Factory_Mix_Video_Profile

ID:

Entry Queue Mode: IVR Only Service Provider

Entry Queue IVR Service: Entry Queue IVR Service

SIP Factories

A SIP Factory is a conferencing entity that enables SIP endpoints to create Ad Hoc conferences. The system is shipped with a default SIP Factory, named DefaultFactory.

> The default SIP Factory uses the conferencing ID 7001. If a SIP Factory is being used do not assign this ID to any conferencing entity, including conferences, and meeting rooms.

When a SIP endpoint calls the SIP Factory URI, a new conference is automatically created based on the Profile parameters, and the endpoint joins the conference.

The SIP Factory URI must be registered with the SIP server to enable routing of calls to the SIP Factory. To ensure that the SIP factory is registered, the option to register *Factories* must be selected in the Default IP Network Service.

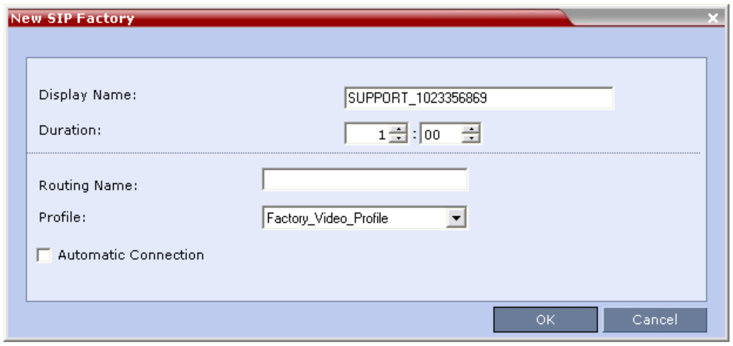
The maximum of number of SIP Factories that can be defined is 40.

Creating SIP Factories

To create a new SIP Factory:

1 In the *RealPresence Collaboration Server Management - Rarely Used* pane, click **SIP Factories**.

2 In the *SIP Factories* list pane, click the **New SIP Factory** button.
The *New Factory* dialog box opens.



Display Name: SUPPORT_1023356869

Duration: 1 : 00

Routing Name:

Profile: Factory_Video_Profile

□ Automatic Connection

3 Define the following parameters:

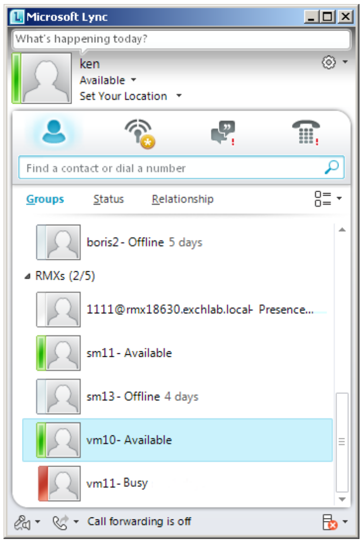
Table 7-3: New Factory Properties

Option	Description
Display Name	Enter the SIP Factory name that will be displayed. The Display Name is the conferencing entity name in native language character sets to be displayed in the Collaboration Server Web Client. In conferences, Meeting Rooms, Entry Queues and SIP factories the system automatically generates an ASCII name for the *Display Name* field that can be modified using Unicode encoding. • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. The maximum length of text fields also varies according to the mixture of character sets (Unicode and ASCII). Maximum field length in ASCII is 80 characters. If the same name is already used by another conference, Meeting Room or Entry Queue, the Collaboration Server displays an error message requesting you to enter a different name.
Routing Name	The *Routing Name* is defined by the user, however if no *Routing Name* is entered, the system will automatically assign a new name when the Profile is saved as follows: • If an all ASCII text is entered in *Display Name*, it is used also as the *Routing Name.* • If any combination of Unicode and ASCII text (or full Unicode text) is entered in *Display Name*, the *ID* (such as Conference ID) is used as the *Routing Name.*
Profile	The default Profile is selected by default. If required, select the conference Profile from the list of Profiles defined in the MCU. A new conference is created using the parameters defined in the Profile.
Automatic Connection	Select this check box to immediately accept the conference creator endpoint to the conference. If the check box is cleared, the endpoint is redirected to the conference and then connected.

4 Click **OK**.
The new SIP Factory is added to the list.

SIP Registration & Presence for Entry Queues and SIP Factories

Entry Queues and *SIP Factories* can be registered with *SIP* servers. This enables *Office Communication Server* or *Lync* server client users to see the availability status (*Available, Offline,* or *Busy*) of these conferencing entities and to connect to them directly from the *Buddy List*.

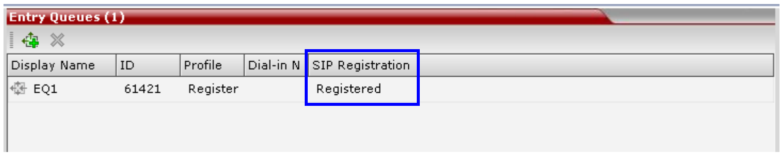


Guidelines

- The *Entry Queue* or *SIP Factory* must be added to the *Active Directory* as a *User*.
- *SIP Registration* must be enabled in the *Profile* assigned to the *Entry Queue* or *SIP Factory*. For more information see *Step* of "*Defining New Profiles”* on page **2-20**.

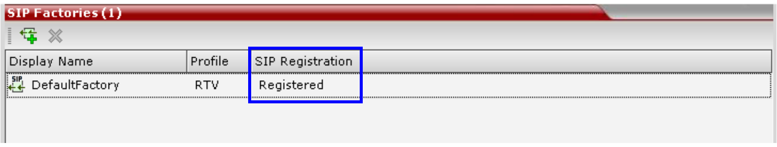
Monitoring Registration Status

The *SIP* registration status can be viewed in the *Entry Queue* or *SIP Factory* list panes.



Entry Queues (1)

Display Name	ID	Profile	Dial-in N	SIP Registration
EQ1	61421	Register		Registered



SIP Factories (1)

Display Name	Profile	SIP Registration
DefaultFactory	RTV	Registered

The following statuses are displayed:

- **Not configured** - *Registration* with the *SIP* Server was not enabled in the *Conference Profile* assigned to the *Entry Queue* or *SIP Factory*.

When SIP registration is not enabled in the conference profile, the Collaboration Server's registering to SIP Servers will each register with an URL derived from its own signaling address.

- **Failed** - *Registration* with the *SIP Server* failed.
This may be due to incorrect definition of the *SIP* server in the *IP Network Service*, or the *SIP Server* may be down, or any other reason the affects the connection between the Collaboration Server or the *SIP Server* to the network.
- **Registered** - the conferencing entity is registered with the *SIP Server*.
- **Partially Registered** - This status is available only in *Multiple Networks* configuration, when the conferencing entity failed to register to all the required *Network Services* if more than one *Network Service* was selected for *Registration*.

Ad Hoc Conferencing

The Entry Queue can also be used for Ad Hoc conferencing. If the Ad Hoc option is enabled for the Entry Queue, when the participant enters the target conference ID the system checks whether a conference with that ID is already running on the MCU. If not, the system automatically creates a new ongoing conference with that ID. The conference parameters are based on the Profile linked to the Entry Queue. As opposed to Meeting Rooms, that are predefined conferences saved on the MCU, Ad Hoc conferences are not stored on the MCU. Once an Ad Hoc conference is started it becomes an ongoing conference, and it is monitored and controlled as any standard ongoing conference.

An external database application can be used for authentication with Ad Hoc conferences. The authentication can be done at the Entry Queue level and at the conference level. At the Entry Queue level, the MCU queries the external database server whether the participant has the right to create a new conference. At the conference level the MCU verifies whether the participant can join the conference and if the participant is the conference chairperson. The external database can populate certain conference parameters.

For more information about Ad Hoc conferencing, see *Appendix D, "Ad Hoc Conferencing and External Database Authentication"* on page **D-1**.

8

Address Book

The Address Book stores information about the people and businesses you communicate with. The Address Book stores, among many other fields, IP addresses, phone numbers and network communication protocols used by the participant's endpoint. By utilizing the Address Book you can quickly and efficiently assign or designate participants to conferences. Groups defined in the Address Book help facilitate the creation of conferences. Participants can be added to the Address Book individually or in Groups.

The maximum of number of Address Book entries that can be defined on the Collaboration Server is 4000.

When using the Polycom CMA/RealPresence Resource Manager Global Address Book, all entries are listed.

The Address Book can be organized into a multi-level hierarchical structure. It can be used to mirror the organizational layout of the enterprises and it is especially suitable for large-scale enterprises with a considerable number of conference participants and organizational departments and divisions. Groups in the Address Book can contain sub-groups or sub-trees, and individual address book participant entities.

The Address Book provides flexibility in arranging conference participants into groups in multiple levels and the capabilities to add groups or participants, move or copy participants to multiple groups within the address book, and use the address book to add groups and participants to a conference or *Conference Template*.

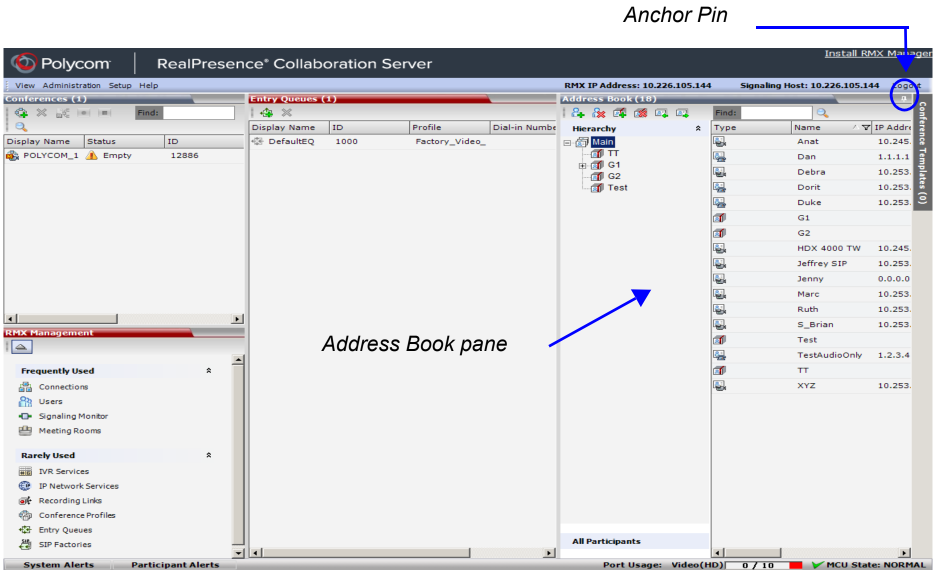
Importing and exporting of Address Books enables organizations to seamlessly distribute up-to-date Address Books to multiple Collaboration Server units. It is not possible to distribute Address Books to external databases running on applications such as *Polycom's RealPresence Resource Manager (XMA)* or *Polycom CMA*. External databases can run in conjunction with Collaboration Server units, but must be managed from the external application. For example, new participants cannot be added to the external database from the Collaboration Server Web Client. To enable the Collaboration Server to run with an external database such as Polycom *RealPresence Resource Manager (XMA)* or *CMA*, the appropriate system configuration flags must be set.

For more information, see "*Modifying System Flags"* on page **20-1**.

Integration with the Global Address Book of the Polycom RealPresence Resource Manager (XMA) or CMA is supported. For more information, see "*Integrating the Global Address Book (GAB) of Polycom RealPresence Resource Manager (XMA) or Polycom CMA™ with the Collaboration Server"* on page **8-23**. Integration with the *SE200* GAB (Global Address Book) is not supported.

Viewing the Address Book

You can view the participants currently defined in the Address Book. The first time the *Collaboration Server Web Client* is accessed, the *Address Book* pane is displayed.


Anchor Pin

Address Book pane

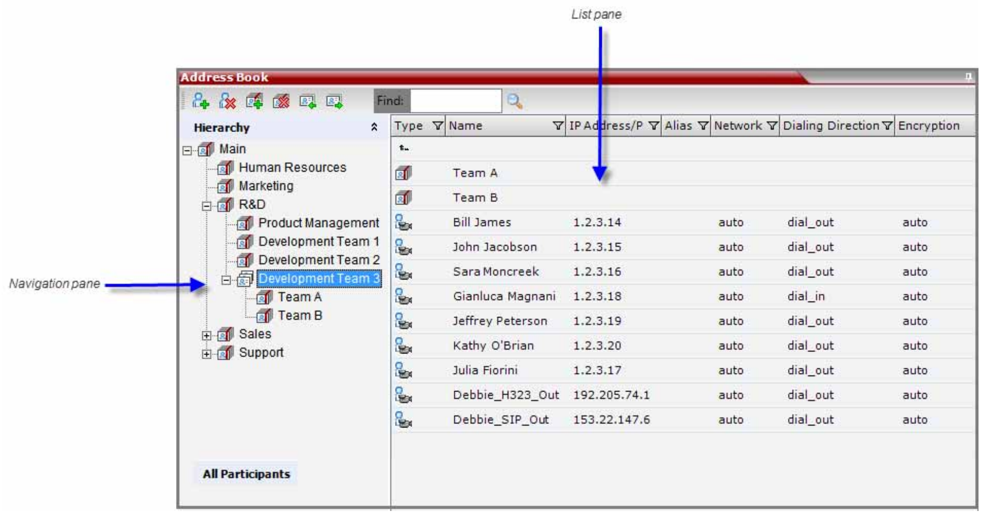
Display Name	Status	ID
POLYCOM_1	Empty	12886

Display Name	ID	Profile	Dial-in Numbe
DefaultEQ	1000	Factory_Video_	

Type	Name	IP Addre
	Anat	10.245.
	Dan	1.1.1.1
	Debra	10.253.
	Dorit	10.253.
	Duke	10.253.
	G1	
	G2	
	HDX 4000 TW	10.245.
	Jeffrey SIP	10.253.
	Jenny	0.0.0.0
	Marc	10.253.
	Ruth	10.253.
	S_Brian	10.253.
	Test	
	TestAudioOnly	1.2.3.4
	TT	
	XYZ	10.253.

The *Address Book* contains two panes:

- *Navigation pane* - contains the hierarchical tree and *All Participants* list
- *List pane* - displays the list of all the members of the selected group and sub-groups.



Type	Name	IP Address/P	Alias	Network	Dialing Direction	Encryption
	Team A					
	Team B					
	Bill James	1.2.3.14		auto	dial_out	auto
	John Jacobson	1.2.3.15		auto	dial_out	auto
	Sara Moncreek	1.2.3.16		auto	dial_out	auto
	Gianluca Magnani	1.2.3.18		auto	dial_in	auto
	Jeffrey Peterson	1.2.3.19		auto	dial_out	auto
	Kathy O'Brian	1.2.3.20		auto	dial_out	auto
	Julia Fiorini	1.2.3.17		auto	dial_out	auto
	Debbie_H323_Out	192.205.74.1		auto	dial_out	auto
	Debbie_SIP_Out	153.22.147.6		auto	dial_out	auto

The *Navigation pane* of the *Address Book* contains the following types of lists:

- **Hierarchical** — displays a multi-level hierarchical tree of groups and participants. Double-clicking a group on the navigation pane displays the group participants and sub-groups in the *List* pane.
- **All Participants** — double-clicking this selection displays the single unique entity of all the participants in a single level. When adding a participant to a group, the system adds a link to the participant's unique entity that is stored in the All Participants list. The same participant may be added to many groups at different levels, and all these

participant links are associated with the same definition of the participant in the *All Participants* list. If the participant properties are changed in one group, they will be changed in all the groups accordingly.

Displaying and Hiding the Group Members in the Navigation Pane

The currently selected group, whose group members are displayed in the Address Book List pane is identified by a special icon .

To expand the group to view the group members:

>> Double-click the group name or click the **Expand** ⊞ button.

The address book entities and sub-groups of the group is displayed in the right group list pane. You can drill down the sub-group to view address book entities in the sub-group.

To move up to the next level and view the members in the upper level:

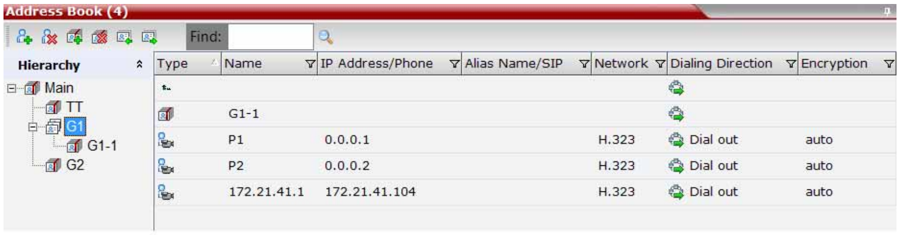
>> Double-click the **navigation arrow** ↑ button in the group members pane.

To collapse a group:

>> Double-click the group name or click the **Collapse** ⊟ button.

Participants List Pane Information

The *Participants List* pane displays the following information for each participant:



Address Book (4)

Find:

Hierarchy
- Main
  - TT
  - G1
    - G1-1
  - G2

Type	Name	IP Address/Phone	Alias Name/SIP	Network	Dialing Direction	Encryption
↑						
	G1-1					
	P1	0.0.0.1		H.323	Dial out	auto
	P2	0.0.0.2		H.323	Dial out	auto
	172.21.41.1	172.21.41.104		H.323	Dial out	auto

Table 8-1 Docked Address Book List Columns

Field/Option	Description
Type	Indicates whether the participant is a video ( ) or voice ( ).
Name	Displays the name of the participant.
IP Address/Phone	Enter the IP address of the participant's endpoint. • For H.323 participant define either the endpoint IP address or alias. • For SIP participant define either the endpoint IP address or the SIP address.
Network	The network communication protocol used by the endpoint to connect to the conference: *H.323* or *SIP* .

Table 8-1 *Docked Address Book List Columns (Continued)*

Field/Option	Description
Dialing Direction	*Dial-in* – The participant dials in to the conference. *Dial-out* – The Collaboration Server dials out to the participant.
Encryption	Displays whether the endpoint uses encryption for its media. The default setting is *Auto*, indicating that the endpoint must connect according to the conference encryption setting.

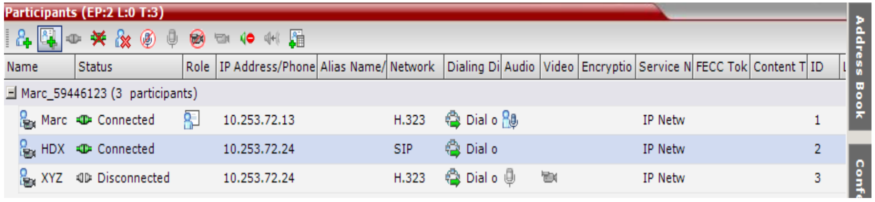
For information on adding and modifying participants in the Address Book, see "*Managing the Address Book"* on page 8-7.

Displaying and Hiding the Address Book

The Address Book can be hidden it by clicking the anchor pin (⊕) button in the pane header. The *Address Book* pane closes and a tab is displayed at the right edge of the screen.

>> Click the tab to re-open the *Address Book*.

Click tab to open Address Book



Participants (EP:2 L:0 T:3)

Name	Status	Role	IP Address/Phone	Alias Name/	Network	Dialing Di	Audio	Video	Encryptio	Service N	FECC Tok	Content T	ID	l
Marc_59446123 (3 participants)														
Marc	Connected		10.253.72.13		H.323	Dial o				IP Netw			1	
HDX	Connected		10.253.72.24		SIP	Dial o				IP Netw			2	
XYZ	Disconnected		10.253.72.24		H.323	Dial o				IP Netw			3	

Adding Participants from the Address Book to Conferences

You can add individual participants or a group of participants from the Address Book to a conference.

Adding Individual Participants from the Address Book to Conferences

You can add a participant or multiple participants to a new conference, ongoing conferences, or to *Conference Templates* by using the drag-and-drop operation.

> Multiple selection of group levels is not available.

To add a participant to a new conference or an ongoing conference:

1 In the *Address Book Navigation* pane, select the group from which to add participants.

2 In the *Address Book List* pane, select the participant or participants you want to add to the conference.

3 Click and hold the left mouse button and drag the selection to the Participants pane of the conference.

The participants are added to the conference.

Adding a Group from the Address Book to Conferences

You can add a group of participants to a new conference, ongoing conferences, or to *Conference Templates* by using the drag-and-drop operation.

To add a group to a new conference or an ongoing conference:

1 In the *Address Book Navigation* pane, select the group you want to add to the conference.

2 Click and hold the left mouse button and drag the selection to the *Participants* pane of the conference.

The participants in the group level and all sub-levels are added to the conference.

Participant Groups

A group is a predefined collection of participants. A group provides an easy way to manage clusters of participants that are in the same organizational structure and to connect a combination of endpoints to a conference. For example, if you frequently conduct conferences with the marketing department, you can create a group called "Marketing Team" that contains the endpoints of all members of the marketing team.

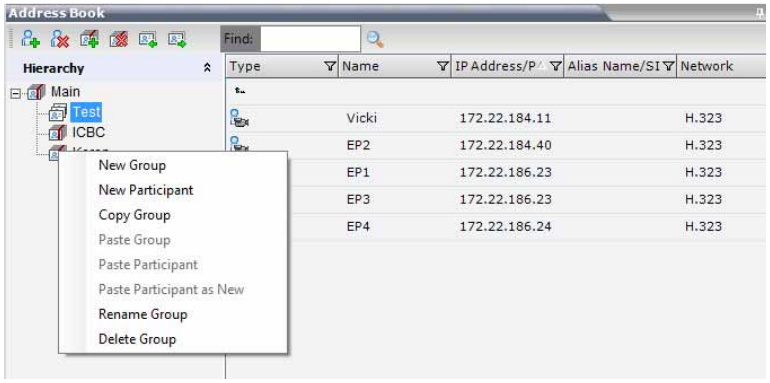
Groups can contain participants and sub-groups. You can define up to ten levels in the "Main" group.

Managing Groups in the Address Book

To manage the groups in the Address Book:

1 In the *Address Book Navigation* pane, right-click the group you want to manage.

The *Groups* menu is displayed.



Type	Name	IP Address/P	Alias Name/SI	Network
	Vicki	172.22.184.11		H.323
	EP2	172.22.184.40		H.323
	EP1	172.22.186.23		H.323
	EP3	172.22.186.23		H.323
	EP4	172.22.186.24		H.323

2 Select one of the following actions:

Table 8-2 Groups Drop-down Menu Actions

Action	Description
New Group	Creates a new group within the current group.
New Participant	Adds a new participant within the current group.
Copy Group	Copies the current group to be pasted as an additional group.
Paste Group	Places the copied group into the current group. The group name of the copied group is defined with "*Copy*" at the end of the group name. This action is only available after a **Copy Group** action has been implemented.
Paste Participant	Places the copied participant into the current selected group. This action is available after a **Copy** or **Cut** action was activated when selecting a single participant or multiple participants.
Paste Participant as New	Pastes as a new participant into the selected group. This paste action adds "*Copy*" at the end of the participant name. This action is only available after a **Copy** action was activated for a single participant.
Rename Group	Renames the group name.

Table 8-2 Groups Drop-down Menu Actions (Continued)

Action	Description
Delete Group	Deletes the group and all of its members. This action displays a message requesting confirmation to delete the group and all members connected with the group.

Additionally, you can drag a group from one location in the Address Book to another location, moving the group and all its members, including sub-groups, to its new location using the drag-and-drop operation. Moving a group to a new location can be done in the navigation pane or the list pane.

To drag a group from a location in the address book to another location:

1. Select the group you want to move.
2. Click and hold the left mouse button and drag the selection to the new location. The new location can be either the "Main" root level or another group level.

   The group and all its members (participants and groups) are moved to the new address book location.

Managing the Address Book

Guidelines

- The multi-level *Address Book* can only be used in a local configuration on the Collaboration Server. The hierarchical structure cannot be implemented with the *Global Address Book* (GAB).
- Up to ten levels can be defined in the hierarchical structure of the Address Book.
- The default name of the root level is "Main". The "Main" root level cannot be deleted but the root level name can be modified.
- Address Book names support multilingual characters.
- Participants in the *Address Book* can be copied to multiple groups. However, only one participant exists in the *Address Book*. Groups that contain the same participants refer to the same definition of the participant entity.

Adding a Participant to the Address Book

Adding participants to the Address Book can be performed by the following methods:

- Directly in the Address Book.
- Moving or saving a participant from an ongoing conference to the Address Book.

When adding dial-out participants to the ongoing conference, the system automatically dials out to the participants using the Network Service defined for the connection in the participant properties.

Adding a New participant to the Address Book Directly

You can add a new participant to the “Main” group or to a group in the *Address Book*. Additionally, you can add a participant from a new conference, ongoing conference, or *Conference Template*.

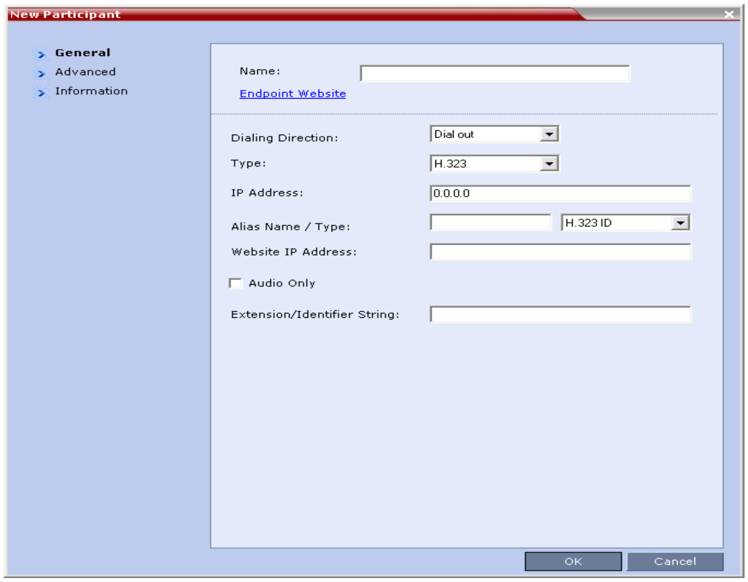
To add a new participant to the Address Book:

1 In the *Address Book - Navigation* pane, select the group to where you want to add the new participant.

2 Click the **New Participant** button ( ) or right-click the group to where you want to add the participant and select the **New Participant** option.

— Alternatively, click anywhere in the *List* pane and select the **New Participant** option.

The *New Participant - General* dialog box opens.



Name:

Dialing Direction: Dial out

Type: H.323

IP Address: 0.0.0.0

Alias Name / Type: H.323 ID

Website IP Address:

□ Audio Only

Extension/Identifier String:

3 Define the following fields:

Table 8-3 *New Participant – General Properties*

Field	Description
Name	Enter the name of the participant or the endpoint as it will be displayed in the Collaboration Server Web Client. The *Name* field can be modified using Unicode encoding. • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. Maximum field length in ASCII is 80 characters. The maximum length of text fields varies according to the mixture of character sets used (Unicode and ASCII). This field may not be left blank. Duplicate participant names, comma, and semi-colon characters may not be used in this field. This name can also become the endpoint name that is displayed in the video layout. For more details about endpoint (site) names, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, *"Audio and Visual Indications (AVC CP Conferencing)"* on page **3-36**". **Note:** This field is displayed in all tabs.
Endpoint Website	Click the Endpoint Website hyperlink to connect to the internal website of the participant's endpoint. It enables you to perform administrative, configuration and troubleshooting activities on the endpoint. The connection is available only if the IP address of the endpoint's internal site is defined in the *Website IP Address* field.
Dialing Direction	Select the dialing direction: • ***Dial-in*** – The participant dials in to the conference. This field applies to IP participants only. • ***Dial-out*** – The MCU dials out to the participant.
Type	The network communication protocol used by the endpoint to connect to the conference: *H.323* or *SIP*. The fields in the dialog box change according to the selected network type.
IP Address	Enter the IP address of the participant's endpoint. • For H.323 participant define either the endpoint IP address or alias. • For SIP participant define either the endpoint IP address or the SIP address. For Collaboration Servers registered to a gatekeeper, the Collaboration Server can be configured to dial and receive calls to and from H.323 endpoints using the IP address in the event that the Gatekeeper is not functioning.

Table 8-3 *New Participant – General Properties (Continued)*

Field	Description
Alias Name/Type **(H.323 Only)**	If you are using the endpoint's alias and not the IP address, first select the type of alias and then enter the endpoint's alias: • H.323 ID (alphanumeric ID) • E.164 (digits 0-9, * and #) • Email ID (email address format, e.g. abc@example.com) • Participant Number (digits 0-9, * and #) **Notes:** • Although all types are supported, the type of alias is dependent on the gatekeeper's capabilities. The most commonly supported alias types are H.323 ID and E.164. • This field is used to enter the Entry Queue ID, target Conference ID and Conference Password when defining a cascaded link. • Use of the E.164 Number is dependent on the setting of the REMOVE_IP_IF_NUMBER_EXISTS System Flag. For more information see "*Substituting E.164 Number in Dial String"* on page **8-13**.
SIP Address/Type **(SIP Only)**	Select the format in which the SIP address is written: • ***SIP URI*** - Uses the format of an E-mail address, typically containing a user name and a host name: *sip:[user]@[host]*. For example, sip:dan@polycom.com. **Note:** If the *SIP Address* field contains an *IPv6* address, it must be surrounded by square brackets, for example, [::1]. • ***TEL URI*** - Used when the endpoint does not specify the domain that should interpret a telephone number that has been input by the user. Rather, each domain through which the request passes would be given that opportunity. For example, a user in an airport might log in and send requests through an outbound proxy in the airport. If the users enters "411" (this is the phone number for local directory assistance in the United States), this number needs to be interpreted and processed by the outbound proxy in the airport, and not by the user's home domain. In this case, tel: 411 is the correct choice.
Endpoint Website IP Address	Enter the IP address of the endpoint's internal site to enable connection to it for management and configuration purposes. This field is automatically completed the first time that the endpoint connects to the Collaboration Server. If the field is blank it can be manually completed by the system administrator. The field can be modified while the endpoint is connected
Audio Only	Select this check box to define the participant as a voice participant, with no video capabilities.

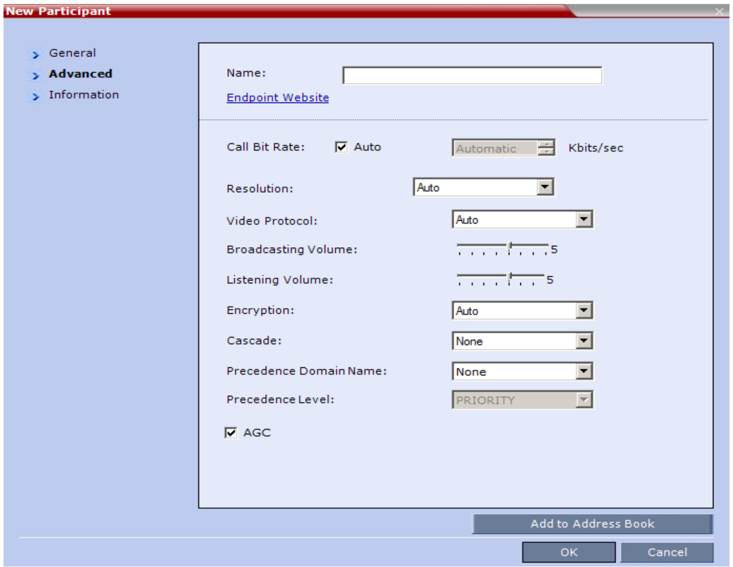
Table 8-3 *New Participant – General Properties (Continued)*

Field	Description
Extension/Identifier String	Dial-out participants that connect to an external device such as Cascaded Links or Recording Links may be required to enter a conference password or an identifying string to connect. Enter the required string as follows: **[p]…[p][string]** For example: pp4566# **p** - optional - indicates a pause of one second before sending the **DTMF string**. Enter several concatenated [p]s to increase the delay before sending the string. The required delay depends on the configuration of the external device or conference IVR system. **String** - enter the required string using the digits 0-9 and the characters * and #. The maximum number of characters that can be entered is identical to the H.323 alias length. If the information required to access the device/conference is composed of several strings, for example, the conference ID and the conference password, this information can be entered as one string, where pauses [p] are added between the strings for the required delays, as follows: **[p]…[p][string][p]…[p] [string]...** For example: p23pp*34p4566#
Extension/Identifier String (continued)	The Collaboration Server automatically sends this information upon connection to the destination device/conference. The information is sent by the Collaboration Server as DTMF code to the destination device/conference, simulating the standard IVR procedure.

4 Usually, additional definitions are not required and you can use the system defaults for the remaining parameters. In such a case, click **OK**.

To modify the default settings for advanced parameters, click the **Advanced** tab.

5 Define the following *Advanced* parameters:



Name:

Call Bit Rate: ☑ Auto Automatic Kbits/sec

Resolution: Auto

Video Protocol: Auto

Broadcasting Volume: 5

Listening Volume: 5

Encryption: Auto

Cascade: None

Precedence Domain Name: None

Precedence Level: PRIORITY

☑ AGC

Table 8-4 New Participant – Advanced Properties

Field	Description
Video Bit Rate / Auto	The *Auto* check box is automatically selected to use the Line Rate defined for the conference. **Note:** This check box cannot be cleared when defining a new participant during an ongoing conference. To specify the video rate for the endpoint, clear this check box and then select the required video rate.
Video Protocol	Select the video compression standard that will be forced by the MCU on the endpoint when connecting to the conference: *H.261*, *H.263*, *H.264* or *RTV*. Select **Auto** to let the MCU select the video protocol according to the endpoint's capabilities.
Resolution	The *Auto* check box is automatically selected to use the Resolution defined for the conference. To specify the Resolution for the participant, select the required resolution from the drop-down menu.
Encryption	Select whether the endpoint uses encryption for its connection to the conference. **Auto** (default setting) indicates that the endpoint will connect according to the conference encryption setting.

Table 8-4 New Participant – Advanced Properties (Continued)

Field	Description
Cascaded	If this participant is used as a link between conferences select: • **Slave**, if the participant is defined in a conference running on a Slave MCU. • **Master**, if the participant is defined in a conference running on the Master MCU. It enables the connection of one conference directly to another conference using an H.323 connection only. The conferences can run on the same MCU or different MCU's. For more information, see "*Basic Cascading using IP Cascaded Link"* on page 5-3.
Precedence Domain Name (**Dial-out SIP Only)**	Not supported with RealPresence Collaboration Server Virtual Edition.
Precedence Level (**Dial-out SIP Only)**	Not supported with RealPresence Collaboration Server Virtual Edition.
AGC	The Audio Gain Control (AGC) protocol that reduces noises is enabled by default for the participants. Clear this check box to disable the AGC feature.

6 To add general information about the participant, such as e-mail, company name, and so on, click the **Information** tab and type the necessary details in the **Info 1-4** fields. Text in the *info* fields can be added in Unicode format (length: 31 characters).

7 Click **OK**.

The new participant is added to the selected group in the address book.

Substituting E.164 Number in Dial String

Between the time a conference is scheduled and when it becomes active, the IP of an endpoint may change, especially in an environment that uses DHCP. The MCU can be set to ignore the IP address of a participant when the conference starts. Instead, the alternative E.164 number will be used.

A new flag, REMOVE_IP_IF_NUMBER_EXISTS, has been added to control this option. This flag must be manually added to change its value. The values of this flag are:

- **YES** (default) - The IP address of an endpoint will be ignored.
- **NO** - The IP address of an endpoint will be used.

Guidelines

- When this feature is enabled, the IP address field of participants in scheduled conferences and conference templates will be empty.
- In order for the MCU to ignore the IP of H.323 participants, the following requirements must be met:
  — A gatekeeper must be defined.
  — The alias of the participant must be defined.
  — The alias type must be defined (not set to *None*).
- If an H.323 gatekeeper is defined but is not connected, the MCU will fail to connect to H.323 dial-out participants.

- In order for the MCU to ignore the IP of SIP participants, the following requirements must be met:
  — A SIP proxy must be defined.
  — The SIP address must be defined.
- If a SIP proxy is defined but is not connected, the MCU will fail to connect to SIP dial-out participants.

Adding a Participant from an Ongoing Conference to the Address Book

You can add a participant to the Address Book directly from an ongoing conference.

> When adding a participant to the address book from a new conference, *Participants* list of an ongoing conference or *Conference Template*, the participant is always added to the "Main" group.

To add a participant from the conference to the Address Book:

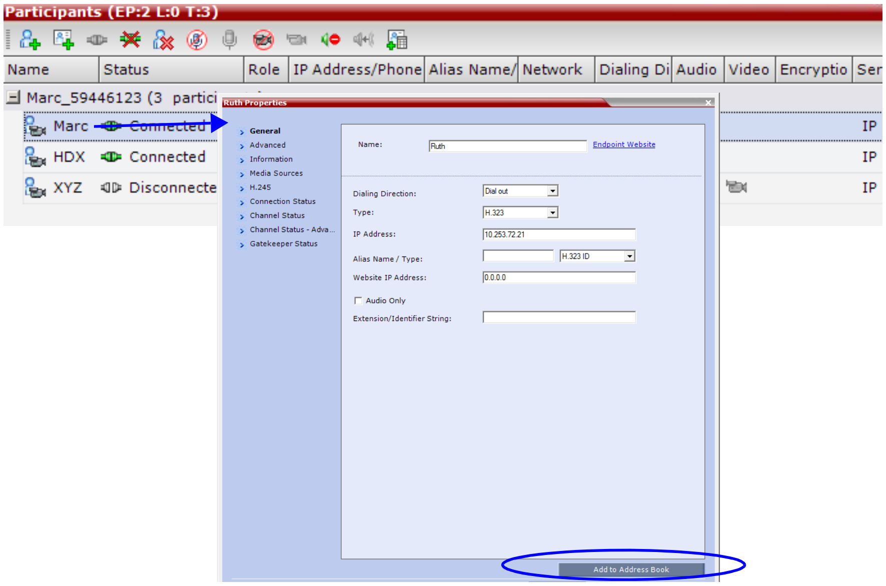
1 During an ongoing conference, select the participant in the *Participant* pane and either click the **Add Participant to Address Book** button ( ) or right-click and select **Add Participant to Address Book**.

The participant is added to the Address Book.

Alternatively, you could:

a Double-click the participant's icon or right-click the participant icon and click **Participant Properties**.

The *Participant Properties* window opens.



Participants (EP:2 L:0 T:3)

Name	Status	Role	IP Address/Phone	Alias Name/	Network	Dialing Di	Audio	Video	Encryptio	Ser
Marc_59446123 (3 partici										
Marc	Connected									IP
HDX	Connected									IP
XYZ	Disconnecte									IP

Ruth Properties

Name: Ruth

Dialing Direction: Dial out

Type: H.323

IP Address: 10.253.72.21

Alias Name / Type: H.323 ID

Website IP Address: 0.0.0.0

☐ Audio Only

Extension/Identifier String:

b Click the **Add to Address Book** button.

> If the participant name is already listed in the All Participants list, an error message is displayed. In such a case, change the name of the participant before adding the participant to the address book.

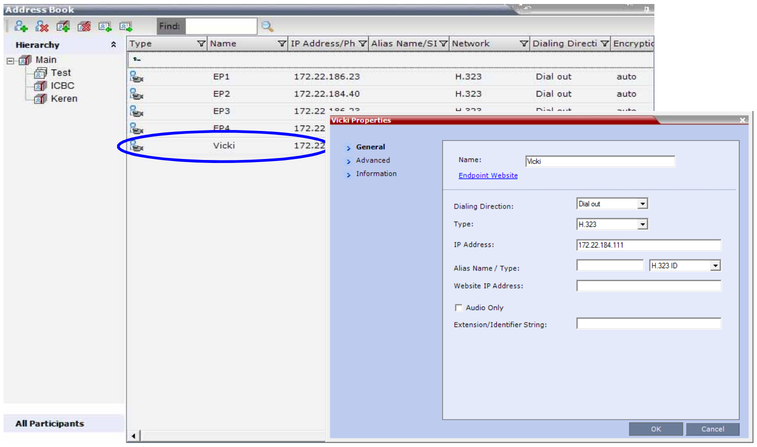
Modifying Participants in the Address Book

When required, you can modify the participant’s properties.

To modify participant properties in the Address Book:

1 In the *Address Book - Navigation* pane, select the group to where the participant to modify is listed.

2 In the *Address Book - List* pane, double-click the participant’s icon.
The *Participant’s Properties* window is displayed.



Type	Name	IP Address/Ph	Alias Name/SI	Network	Dialing Directi	Encryptic
	EP1	172.22.186.23		H.323	Dial out	auto
	EP2	172.22.184.40		H.323	Dial out	auto
	EP3	172.22				
	EP4	172.22				
	Vicki	172.22				

Name: Vicki

Dialing Direction: Dial out

Type: H.323

IP Address: 172.22.184.111

Alias Name / Type: H.323 ID

Website IP Address:

□ Audio Only

Extension/Identifier String:

3 Modify the necessary properties in the window, such as dialing direction, communication protocol type, and so on. You can modify any property in any of the three tabs: *General*, *Advanced* and *Info*.

4 Click **OK**.

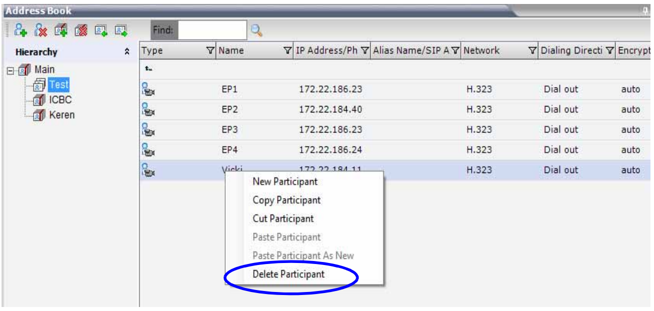
The changes to the participant’s properties are updated.

Deleting Participants from the Address Book

To delete participants from the Address Book:

1 In the *Address Book - Navigation* pane, select the group where the participant to delete is listed.

2 In the *Address Book - List* pane, either select the participant to delete and then click the **Delete Participant** (☒) button, or right-click the participant icon and then click the **Delete Participant** option.



Type	Name	IP Address/Ph	Alias Name/SIP A	Network	Dialing Directi	Encrypt
	EP1	172.22.186.23		H.323	Dial out	auto
	EP2	172.22.184.40		H.323	Dial out	auto
	EP3	172.22.186.23		H.323	Dial out	auto
	EP4	172.22.186.24		H.323	Dial out	auto
	Vicki	172.22.184.11		H.323	Dial out	auto

3 A confirmation message is displayed depending on the participant’s assignment to groups in the address book:

a When the participant belongs to only one group: click **Yes** to permanently delete the participant from the address book.

b When the participant belongs to multiple groups, a message is displayed requesting whether to delete the participant from the *Address Book* or from the current selected group. Select:

- **Current group** to delete the participant from the selected group
- **Address Book** to permanently delete the participant from the address book (all groups).

Click **OK** to perform the delete operation or **Cancel** to exit the delete operation.

Copying or Moving a Participant

You can copy or move a participant from one group to another group using the **Copy**, **Cut**, and **Paste** options. A participant can belong to multiple groups. However, there is only one entity per participant. Groups that contain the same participants refer to the same definition of the participant entity. Alternatively, you can drag a participant from one location in the *Address Book* to another location, moving the participant to its new location using the drag-and-drop operation.

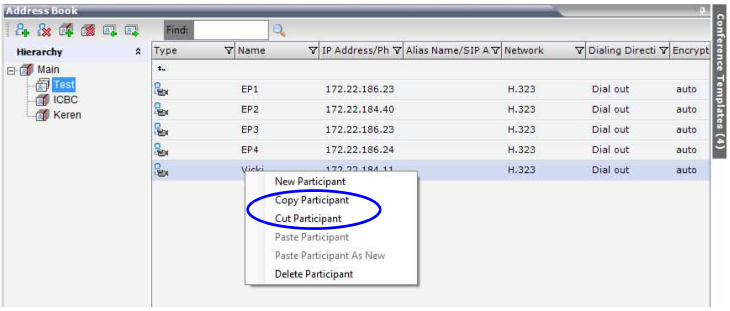
The cut and copy actions are not available when selecting multiple participants.

To copy or move a participant to another group:

1 In the *Address Book - Navigation* pane, select the group from where to copy the participant.

2 In the *Address Book - List* pane, select the participant you want to copy.

3 Right-click the selected participant and select one of the following functions from the drop-down menu:



Type	Name	IP Address/Ph	Alias Name/SIP A	Network	Dialing Directi	Encrypt
	EP1	172.22.186.23		H.323	Dial out	auto
	EP2	172.22.184.40		H.323	Dial out	auto
	EP3	172.22.186.23		H.323	Dial out	auto
	EP4	172.22.186.24		H.323	Dial out	auto
	Vicki	172.22.184.11		H.323	Dial out	auto

Table 8-5 Copy/Cut Functions

Function	Description
Copy Participant	Copies the participant to be pasted into an additional group.
Cut Participant	Moves the participant from the current group to a different group. Alternatively, you can move a participant to another location by dragging the participant to the new location.

4 In the *Address Book* navigation pane, navigate and select the group in which you want to paste the participant.

5 Right-click the selected group and click one of the following **Paste** functions from the drop-down menu:

Table 8-6 Paste functions

Function	Description
Paste Participant	Creates a link to the participant entity in the pasted location.
Paste Participant as New	Pastes as a new participant into the selected group. This paste action adds "*Copy*" to the end of the participant name.

The Paste functions are only available after a **Copy** or **Cut** action has been implemented.

To drag a participant from an address book group to another group:

1 Select the participant or participants you want to move.

2 Click and hold the left mouse button and drag the selection to the new group.

The participants are moved to the new address book group.

Searching the Address Book

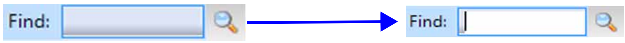
You can search the *Address Book* for a participant's name or a group name only on the currently selected group/level.

To search for participants or groups in the current selected level:

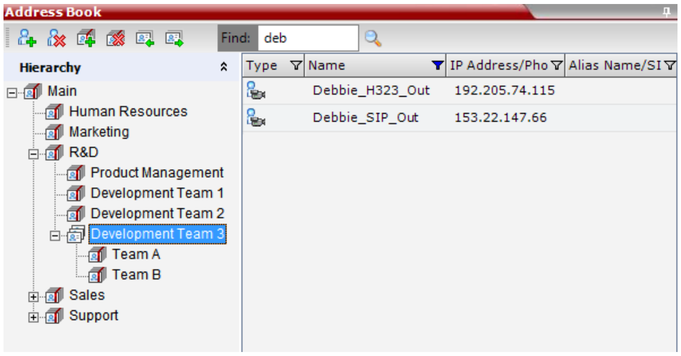
1 In the *Address Book Navigation* pane, select the group/level within to run the search.

2 In the *Address Book* toolbar, activate the search option by clicking the **Find** field.

The field clears and a cursor appears indicating that the field is active.



3 Type all or part of the participant's name or group name and click the search button.



Type	Name	IP Address/Pho	Alias Name/SI
	Debbie_H323_Out	192.205.74.115	
	Debbie_SIP_Out	153.22.147.66	

The closest matching participant entries are displayed and the Active Filter indicator turns on.

Filtering the Address Book

The entries in an address book group can be filtered to display only the entries (participants or groups) that meet criteria that you specify and hides entries that you do not want displayed. It enables you to select and work with a subset of *Address Book* entries.

You can filter by more than one column, by adding additional filters (columns).

The filter applies to the displayed group. If *All Participants* option is selected, it applies to all the listed participants.

Filtering can be done using:

- A predefined pattern
- Customized pattern

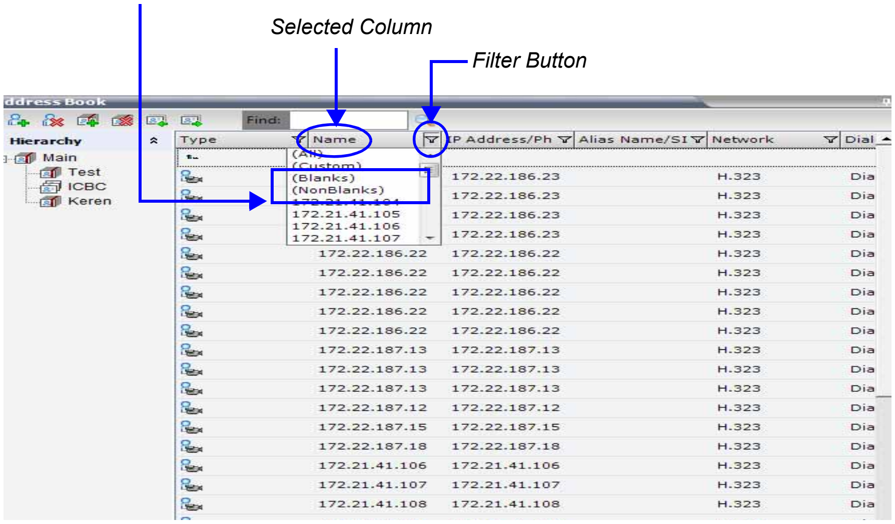
When you use the Find dialog box to search filtered data, only the data that is displayed is searched; data that is not displayed is not searched. To search all the data, clear all filters.

Filtering Address Book Data Using a Predefined Pattern

To filter the data in an address book group:

1. In the *Address Book - Navigation* pane, select the group to filter.
2. In the *Address Book - List* pane, in the *column* that you want to use for filtering, click the filter (▽) button.

   A drop-down menu is displayed containing all the matching patterns that can be applied to the selected field.

   Filtering Options

   Selected Column

   Filter Button

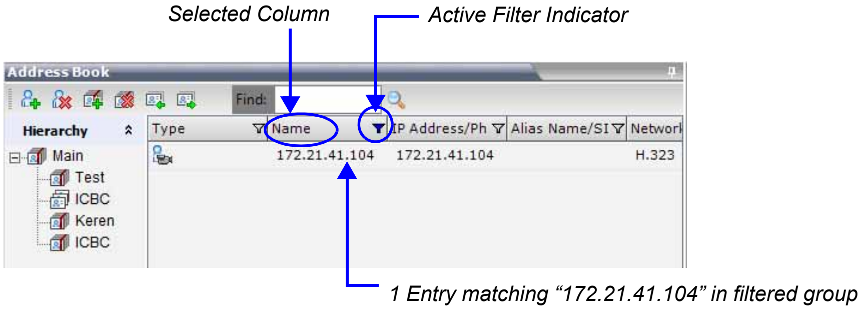
   (All)
   (Custom)
   (Blanks)
   (NonBlanks)
   172.21.41.104
   172.21.41.105
   172.21.41.106
   172.21.41.107

Type	Name	IP Address/Ph	Alias Name/SI	Network	Dial
		172.22.186.23		H.323	Dia
		172.22.186.23		H.323	Dia
		172.22.186.23		H.323	Dia
		172.22.186.23		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.187.13	172.22.187.13		H.323	Dia
	172.22.187.13	172.22.187.13		H.323	Dia
	172.22.187.13	172.22.187.13		H.323	Dia
	172.22.187.12	172.22.187.12		H.323	Dia
	172.22.187.15	172.22.187.15		H.323	Dia
	172.22.187.18	172.22.187.18		H.323	Dia
	172.21.41.106	172.21.41.106		H.323	Dia
	172.21.41.107	172.21.41.107		H.323	Dia
	172.21.41.108	172.21.41.108		H.323	Dia

3. Click the matching pattern to be applied.

   The filtered list is displayed with a filter indicator (▼) displayed in the selected column heading.

Example: If the user selects **172.21.41.104** as the matching pattern, the filtered group in the *Address Book* is displayed as follows:



Type	Name	IP Address/Ph	Alias Name/SI	Network
	172.21.41.104	172.21.41.104		H.323

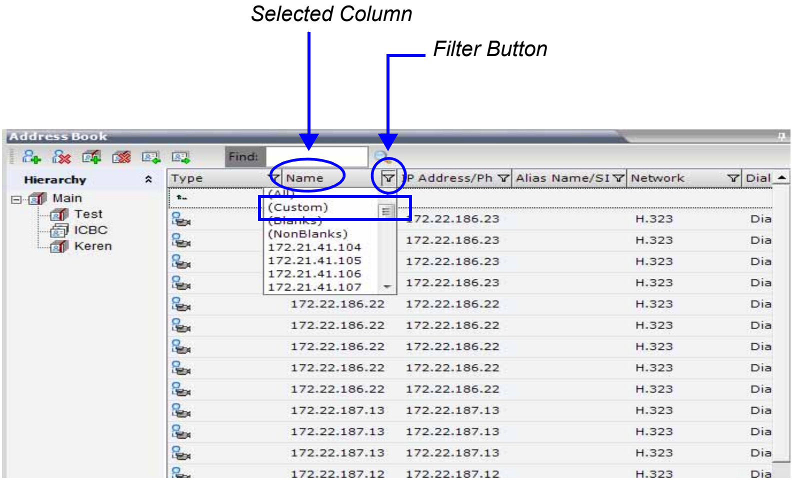
Filtering Address Book Data Using a Custom Pattern

To filter the data in an address book group:

1 In the *Address Book - Navigation* pane, select the group to filter.

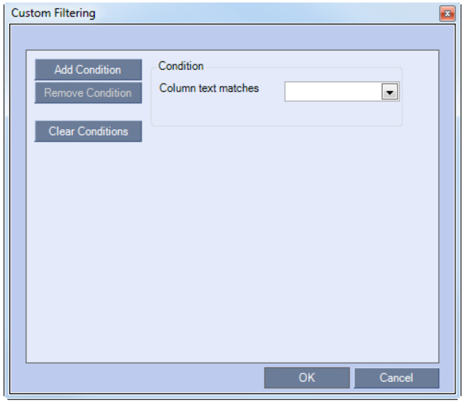
2 In the *Address Book - List* pane, in the *column* that you want to use for filtering, click the filter (Y) button.

3 Select the **(Custom)** option from the drop-down list.



Type	Name	IP Address/Ph	Alias Name/SI	Network	Dial
		172.22.186.23		H.323	Dia
		172.22.186.23		H.323	Dia
		172.22.186.23		H.323	Dia
		172.22.186.23		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.186.22	172.22.186.22		H.323	Dia
	172.22.187.13	172.22.187.13		H.323	Dia
	172.22.187.13	172.22.187.13		H.323	Dia
	172.22.187.13	172.22.187.13		H.323	Dia
	172.22.187.12	172.22.187.12		H.323	Dia

The *Custom Filtering* dialog box opens.

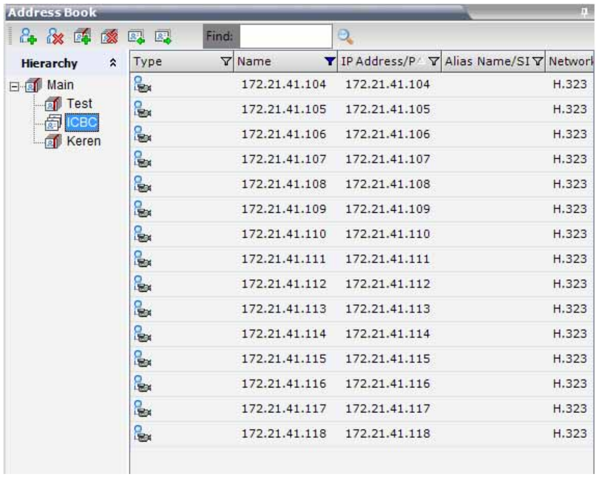


4 In the *Condition - Column text matches* field, enter the filtering pattern.
For example, to list only endpoints that include the numerals 41 in their name, enter 41.

5 **Optional.** Click the **Add Condition** button to define additional filtering patterns to further filter the list and fine tune your search.

To clear a filtering pattern, click the **Clear Condition** button.

The filtered list is displayed with an active filter (blue) indicator (▼) displayed in the selected column heading. For example, if the filtering pattern is 41, the participants list includes all the endpoints that contain the numerals 41 in their name.



Type	Name	IP Address/P	Alias Name/SI	Network
	172.21.41.104	172.21.41.104		H.323
	172.21.41.105	172.21.41.105		H.323
	172.21.41.106	172.21.41.106		H.323
	172.21.41.107	172.21.41.107		H.323
	172.21.41.108	172.21.41.108		H.323
	172.21.41.109	172.21.41.109		H.323
	172.21.41.110	172.21.41.110		H.323
	172.21.41.111	172.21.41.111		H.323
	172.21.41.112	172.21.41.112		H.323
	172.21.41.113	172.21.41.113		H.323
	172.21.41.114	172.21.41.114		H.323
	172.21.41.115	172.21.41.115		H.323
	172.21.41.116	172.21.41.116		H.323
	172.21.41.117	172.21.41.117		H.323
	172.21.41.118	172.21.41.118		H.323

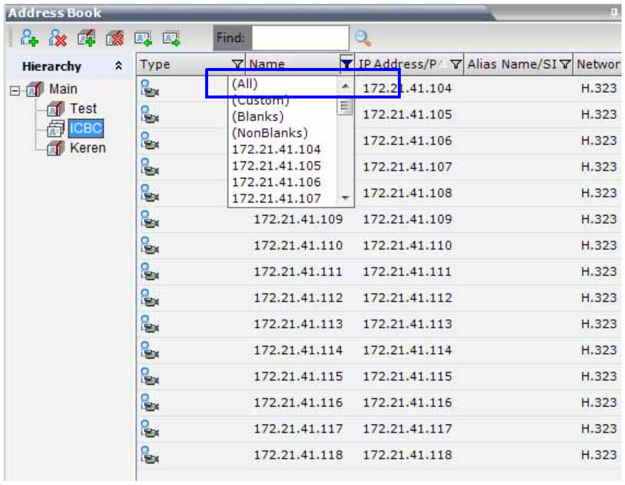
Clearing the Filter

To clear the filter and display all entries:

1 In the filtered *Address Book* column heading, click the *Active Filter* indicator.

The pattern matching options menu is displayed.

2 Click (**All)**.



Type	Name	IP Address/P	Alias Name/SI	Networ
		172.2 .41.104		H.323
		172.21.41.105		H.323
		172.21.41.106		H.323
		172.21.41.107		H.323
		172.21.41.108		H.323
	172.21.41.109	172.21.41.109		H.323
	172.21.41.110	172.21.41.110		H.323
	172.21.41.111	172.21.41.111		H.323
	172.21.41.112	172.21.41.112		H.323
	172.21.41.113	172.21.41.113		H.323
	172.21.41.114	172.21.41.114		H.323
	172.21.41.115	172.21.41.115		H.323
	172.21.41.116	172.21.41.116		H.323
	172.21.41.117	172.21.41.117		H.323
	172.21.41.118	172.21.41.118		H.323

The filter is deactivated and all the *group/level* entries are displayed.

Obtaining the Display Name from the Address Book

The MCU can be configured to replace the name of the dial-in participant as defined in the endpoint (site name) with the name defined in the Address Book.

In this process, the system retrieves the data (name, alias, number or IP address) of the dial-in participant and compares it first with the conference defined dial-in participants and if the endpoint is not found, it then searches for the endpoint with entries in the address book. After a match is found, the system displays the participant name as defined in the address book instead of the site name, in both the video layout and the Collaboration Server Web Client/Manager.

The system compares the following endpoint data with the address book entries:

- For H.323 participants, the system compares the IP address, Alias, or H.323 number.
- For SIP participants, the system compares the IP address or the SIP URI.

Guidelines

- Only Users with *Administrator* and *Operator* Authorization Levels are allowed to enable and disable the *Obtain Display Name from Address Book* feature.
- This feature is supported only for IPv4 participants.

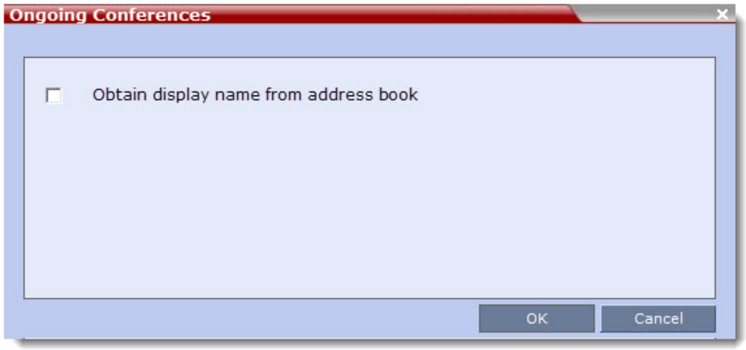
Enabling and Disabling the Obtain Display Name from Address Book Feature

The *Obtain Display Name from Address Book* option can be enabled for all participants connecting to the MCU if the name of the participants are defined in the Address Book.

To enable or disable the Obtain Display Name from Address Book option:

1 On the Collaboration Server main menu bar, click **Setup > Customize Display Settings > Ongoing Conferences**.

The *Ongoing Conferences* dialog box is displayed.



□ Obtain display name from address book

2 Select the **Obtain display name from address book** check box to enable the feature or clear the check box to disable the feature.

3 Click **OK**.

Importing and Exporting Address Books

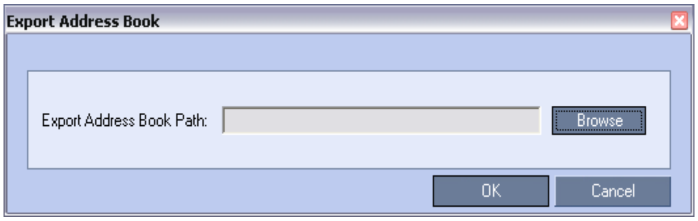
Address Books are proprietary Polycom data files that can only be distributed among Collaboration Server units. The Address Books are exported in XML format, which are editable offline. If no name is assigned to the exported Address Book, the default file name is: `EMA.DataObjects.OfflineTemplates.AddressbookContent_.xml`

Exporting an Address Book

To Export an Address Book:

1 In the *Address Book* pane, click the **Export Address Book** ( ) button or right-click an empty area in the pane and click **Export Address Book**.

The *Export Address Book* dialog box is displayed.



2 Enter the desired path or click the **Browse** button.

3 In the **Save Address Book** dialog box, select the directory to save the file. You may also rename the file in the *File Name* field.

4 Click **Save**.
You will return to the *Export File* dialog box.

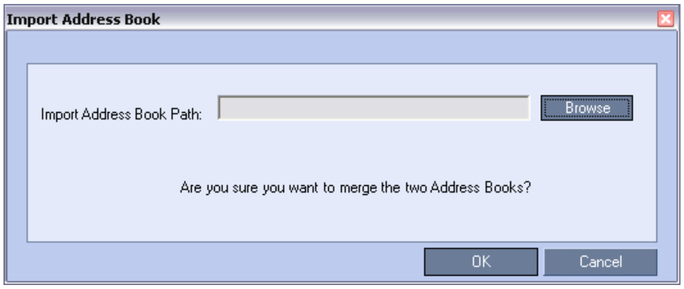
5 Click **OK**.
The exported Address Book is saved in the selected folder in XML format.

Importing an Address Book

To Import and Address Book:

1 In the *Address Book* pane, click the **Import Address Book** ( ) button or right-click an empty area in the pane and then click **Import Address Book**.

The *Import Address Book* dialog box is displayed.



Are you sure you want to merge the two Address Books?

2 Enter the path from which to import the Address Book or click the **Browse** button.

3 In the *Open* dialog box navigate to the desired Address Book file (in XML format) to import.

When importing an Address Book, participants with exact names in the current Address Book will be overwritten by participants defined in the imported Address Book.

4 Click **Open**.

You will return to the *Import File* dialog box.

5 Click **OK**.

The *Address Book* is imported and a confirmation message is displayed at the end of the process.

6 Click Close.

Integrating the Global Address Book (GAB) of Polycom RealPresence Resource Manager (XMA) or Polycom CMA™ with the Collaboration Server

The Polycom RealPresence Resource Manager (XMA) or Polycom CMA™ application includes a Global Address Book (GAB) with all registered endpoints. This address book can be used by the Collaboration Server users to add participants to conferences.

Guidelines for integrating with the Global Address Book of Polycom RealPresence Resource Manager (XMA) or Polycom CMA™:

- The Collaboration Server can use only one address book at a time. After you integrate the Polycom RealPresence Resource Manager (XMA) or Polycom CMA with the Polycom Collaboration Server, the XMA/CMA address book replaces the Collaboration Server internal address book.
- The Collaboration Server uses the RealPresence Resource Manager (XMA) or Polycom CMA address book in read-only mode. You can only add or modify XMA/CMA address book entries from the RealPresence Resource Manager (XMA) or Polycom CMA.

The Collaboration Server acts as a proxy to all address book requests between the Collaboration Server Web Client and the XMA/CMA. **Ensure that firewall and other network settings allow the Collaboration Server access to the XMA or CMA server.**

To Integrate the RealPresence Resource Manager (XMA) or Polycom CMA Global Address Book (GAB) with the Collaboration Server:

RealPresence Resource Manager (XMA) or Polycom CMA Side

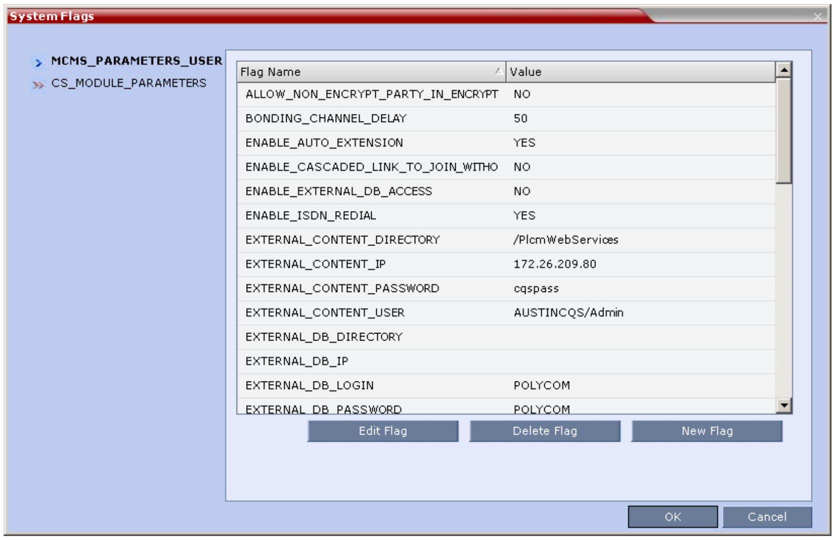
1 In the RealPresence Resource Manager (XMA) or Polycom CMA application, manually add the Polycom Collaboration Server system to the RealPresence Resource Manager (XMA) or Polycom CMA system as directed in the *PRealPresence Resource Manager (XMA) or Polycom CMA Operations Guide*.

2 In the RealPresence Resource Manager (XMA) or Polycom CMA application, add a user or use an existing user for Collaboration Server login as directed in the *RealPresence Resource Manager (XMA) or Polycom CMA Operations Guide*.
Write down the User Name and Password as they will be used later to define the Collaboration Server connection to the RealPresence Resource Manager (XMA) or Polycom CMA Global Address Book.

Collaboration Server Side

1 On the *Collaboration Server* menu, click **Setup > System Configuration**.

The *System Flags - MCMS_PARAMETERS_USER* dialog box opens.



Flag Name	Value
ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT	NO
BONDING_CHANNEL_DELAY	50
ENABLE_AUTO_EXTENSION	YES
ENABLE_CASCADED_LINK_TO_JOIN_WITHO	NO
ENABLE_EXTERNAL_DB_ACCESS	NO
ENABLE_ISDN_REDIAL	YES
EXTERNAL_CONTENT_DIRECTORY	/PlcmWebServices
EXTERNAL_CONTENT_IP	172.26.209.80
EXTERNAL_CONTENT_PASSWORD	cqspass
EXTERNAL_CONTENT_USER	AUSTINCQS/Admin
EXTERNAL_DB_DIRECTORY	
EXTERNAL_DB_IP	
EXTERNAL_DB_LOGIN	POLYCOM
EXTERNAL_DB_PASSWORD	POLYCOM

2 Modify the values of the following flags:

For more information, see "*Modifying System Flags"* on page **20-1**.

Table 8-7 System Flags for CMA Address Book Integration

Flag	Description
EXTERNAL_CONTENT_DIRECTORY	The Web Server folder name. Change this name if you have changed the default names used by the RealPresence Resource Manager (XMA) or Polycom CMA application. Default: /PlcmWebServices
EXTERNAL_CONTENT_IP	Enter the IP address of the RealPresence Resource Manager (XMA) or Polycom CMA server. For example: 172.22.185.89. This flag is also the trigger for replacing the internal Collaboration Server address book with the RealPresence Resource Manager (XMA) or Polycom CMA Global Address Book (GAB). Leave this flag blank to disable address book integration with the RealPresence Resource Manager (XMA) or Polycom CMA server.
EXTERNAL_CONTENT_PASSWORD	The password associated with the user name defined for the Collaboration Server in the RealPresence Resource Manager (XMA) or Polycom CMA server.
EXTERNAL_CONTENT_USER	The login name defined for the Collaboration Server in the RealPresence Resource Manager (XMA) or Polycom CMA server defined in the format: domain name/user name.

3 Click **OK** to complete the definitions.

When prompted, click **Yes** to reset the MCU and implement the changes to the system configuration.

9

Operator Assistance & Participant Move

> Operator conferences and participant move are supported in AVC CP Conferencing Mode only.

Users (operators) assistance to participants is available when:

- Participants have requested individual help (using *0 DTMF code) during the conference.
- Participants have requested help for the conference (using 00 DTMF code) during the conference.
- Participants have problems connecting to conferences, for example, when they enter the wrong conference ID or password.

In addition, the user (operator) can join the ongoing conference and assist all conference participants.

Operator assistance is available only when an *Operator conference* is running on the MCU.

The *Operator conference* offers additional conference management capabilities to the Collaboration Server users, enabling them to attend to participants with special requirements and acquire participant details for billing and statistics. This service is designed usually for large conferences that require the personal touch.

Operator Conferences

An *Operator conference* is a special conference that enables the Collaboration Server user acting as an operator to assist participants without disturbing the ongoing conferences and without being heard by other conference participants. The operator can move a participant from the Entry Queue or ongoing conference to a private, one-on-one conversation in the Operator conference.

In attended mode, the Collaboration Server user (operator) can perform one of the following actions:

- Participants connected to the Entry Queue who fail to enter the correct destination ID or conference password can be moved by the user to the Operator conference for assistance.
- After a short conversation, the operator can move the participant from the Operator conference to the appropriate destination conference (Home conference).
- The operator can connect participants belonging to the same destination conference to their conference simultaneously by selecting the appropriate participants and moving them to the Home conference (interactively or using the right-click menu).

- The operator can move one or several participants from an ongoing conference to the *Operator conference* for a private conversation.
- The operator can move participants between ongoing Continuous Presence conferences.

Operator Conference Guidelines

- An *Operator conference* can only run in Continuous Presence mode.
- *Operator conference* is defined in the Conference Profile. When enabled in Conference Profile, *High Definition Video Switching* option is disabled.
- An *Operator conference* can only be created by a User with Operator or Administrator *Authorization* level.
- *Operator conference* name is derived from the User Login Name and it cannot be modified.
- Only one *Operator conference* per User *Login Name* can be created.
- When created, the *Operator conference* must include one and only one participant - the Operator participant.
- Only a defined dial-out participant can be added to an *Operator conference* as an Operator participant
- Once running, the Collaboration Server user can add new participants or move participants from other conferences to this conference. The maximum number of participants in an *Operator conference* is the same as in standard conferences.
- Special icons are used to indicate an *Operator conference* in the Ongoing Conferences list and the operator participant in the Participants list.
- An *Operator conference* can be saved to a Conference Template. An ongoing *Operator conference* can be started from a Conference Template.
- The Operator participant cannot be deleted from the *Operator conference* or from any other conference to which she/he was moved to, but it can be disconnected from the conference.
- When deleting or terminating the *Operator conference*, the operator participant is automatically disconnected from the MCU, even if participating in a conference other than the *Operator conference*.
- Moving participants from/to an *Operator conference* follows the same guidelines as moving participants between conferences. For move guidelines, see "*Move Guidelines"* on page **9-14**.
- When a participant is moved from the Entry Queue to the *Operator conference*, the option to move back to the source (Home) conference is disabled as the Entry Queue is not considered as a source conference.
- The conference chairperson cannot be moved to the *Operator conference* following the individual help request if the *Auto Terminate When Chairperson Exits* option is enabled, to prevent the conference from automatically ending prematurely. In such a case, the assistance request is treated by the system as a conference assistance request, and the operator can join the conference.

Defining the Components Enabling Operator Assistance

To enable operator assistance for conferences, the following conferencing entities must be adjusted or created:

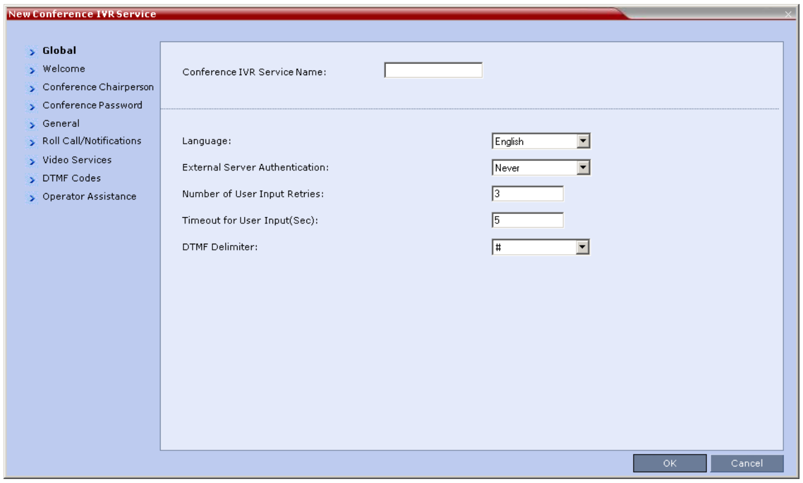
- IVR Service (Entry Queue and Conference) in which Operator Assistance options are enabled.
- A Conference Profile with the *Operator Conference* option enabled.
- An active Operator conference with a connected Operator participant.

Defining a Conference IVR Service with Operator Assistance Options

In the *RealPresence Collaboration Server Management* pane, expand the *Rarely Used* list and click the **IVR Services** () entry.

1 On the *IVR Services* toolbar, click the **New Conference IVR Service** (*)* button.

The *New Conference IVR Service - Global* dialog box opens.



Conference IVR Service Name:

Language: English

External Server Authentication: Never

Number of User Input Retries: 3

Timeout for User Input(Sec): 5

DTMF Delimiter: #

2 Enter the *Conference IVR Service Name*.

3 Define the *Conference IVR Service - Global* parameters. For more information, see *Table 16-3, "Conference IVR Service Properties - Global Parameters,"* on page **16-7**.

4 Click the **Welcome** tab.
The *New Conference IVR Service - Welcome* dialog box opens.

5 Define the system behavior when the participant enters the Conference IVR queue. For more information, see "*Defining a New Conference IVR Service"* on page **16-6**.

6 Click the **Conference Chairperson** tab.
The *New Conference IVR Service - Conference Chairperson* dialog box opens.

7 If required, enable the chairperson functionality and select the various voice messages and options for the chairperson connection. For more information, see *Table 16-4, "New Conference IVR Service Properties - Conference Chairperson Options and Messages,"* on page **16-9**.

8 Click the **Conference Password** tab.

The *New Conference IVR Service - Conference Password* dialog box opens.

9 If required, enable the request for conference password before moving the participant from the conference IVR queue to the conference and set the MCU behavior for

password request for *Dial-in* and *Dial-out* participant connections. For more information, see *Table 16-5, "New Conference IVR Service Properties - Conference Password Parameters,"* on page **16-10**.

10 Select the various audio messages that will be played in each case. For more information, see *Table 16-5, "New Conference IVR Service Properties - Conference Password Parameters,"* on page **16-10**.

11 Click the **General** tab.
The *New Conference IVR Service - General* dialog box opens.

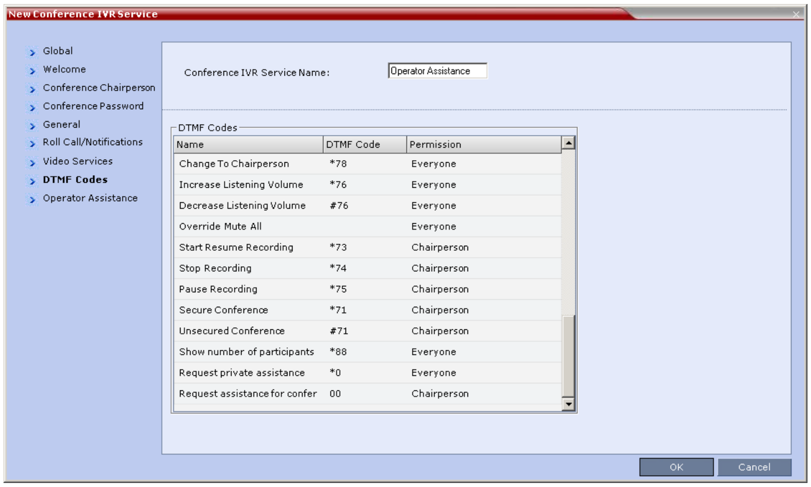
12 Select the messages that will be played during the conference. For more information, see *Table 16-6, "Conference IVR Service Properties - General Voice Messages,"* on page **16-11**.

13 Click the **Video Services** tab.
The *New Conference IVR Service - Video Services* dialog box opens.

14 Define the *Video Services* parameters. For more information, see *Table 16-8, "New Conference IVR Service Properties - Video Services Parameters,"* on page **16-15**.

15 Click the **DTMF Codes** tab.

The *New Conference IVR Service - DTMF Codes* dialog box opens.



Conference IVR Service Name: Operator Assistance

DTMF Codes

Name	DTMF Code	Permission
Change To Chairperson	*78	Everyone
Increase Listening Volume	*76	Everyone
Decrease Listening Volume	#76	Everyone
Override Mute All		Everyone
Start Resume Recording	*73	Chairperson
Stop Recording	*74	Chairperson
Pause Recording	*75	Chairperson
Secure Conference	*71	Chairperson
Unsecured Conference	#71	Chairperson
Show number of participants	*88	Everyone
Request private assistance	*0	Everyone
Request assistance for confer	00	Chairperson

The default DTMF codes for the various functions that can be performed during the conference by all participants or by the chairperson are listed. For the full list of the available DTMF codes, see *Table 16-9, "New Conference IVR Service Properties - DTMF Codes,"* on page **16-16**.

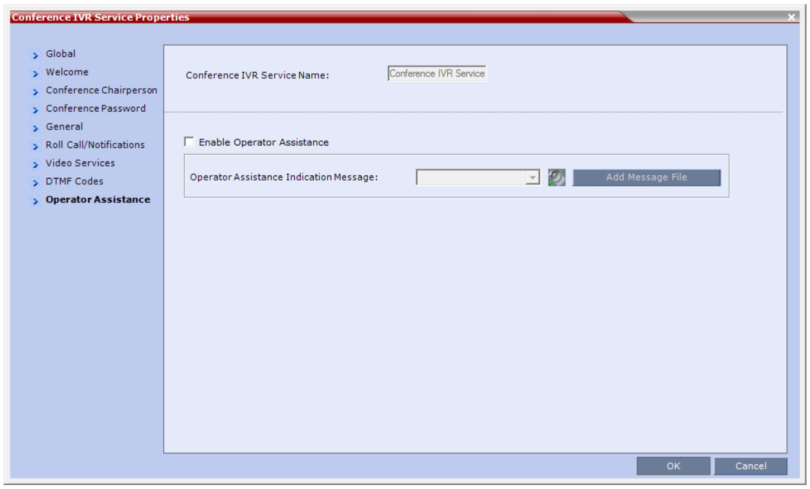
16 If required, modify the default DTMF codes and the permissions for various operations including Operator Assistance options:

— ***0** for individual help - the participant requested help for himself or herself. In such a case, the participant requesting help is moved to the Operator conference for one-on-one conversation. By default, all participants can use this code.

— **00** for conference help - the conference chairperson (default) can request help for the conference. In such a case, the operator joins the conference.

17 Click the **Operator Assistance** tab.

The *Operator Assistance* dialog box opens.



Conference IVR Service Name: Conference IVR Service

☐ Enable Operator Assistance

Operator Assistance Indication Message:

18 Select **Enable Operator Assistance** to enable operator assistance when the participant requires or requests help during the connection process to the conference or during the conference.

19 In the *Operator Assistance Indication Message* field, select the audio message to be played when the participant requests or is waiting for the operator's assistance.

> If the audio file was not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the Collaboration Server.

20 Click **OK** to complete the IVR Service definition.
The new Conference IVR Service is added to the *IVR Services* list.

Defining an Entry Queue IVR Service with Operator Assistance Options

1 In the *RealPresence Collaboration Server Management* pane, click **IVR Services** (▦).

2 In the *IVR Services* list, click the **New Entry Queue IVR Service** (✚) button.

The *New Entry Queue IVR Service - Global* dialog box opens.

3 Define the *Entry Queue Service Name.*

4 Define the Entry Queue IVR Service Global parameters. For more information, see Table 16-10, "*Entry Queue IVR Service Properties - Global Parameters,*" on page **16-19**.

5 Click the **Welcome** tab.
The *New Entry Queue IVR Service - Welcome* dialog box opens.

6 Define the system behavior when the participant enters the Entry Queue. This dialog box contains options that are identical to those in the *Conference IVR Service - Welcome Message* dialog box.

7 Click the **Conference ID** tab.
The *New Entry Queue IVR Service - Conference ID* dialog box opens.

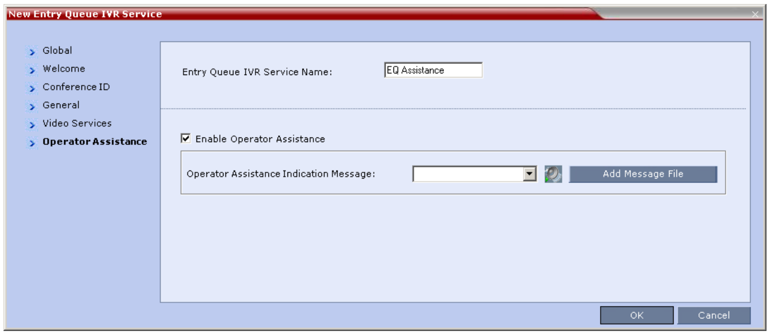
8 Select the required voice messages. For more information, see *Table 16-11, "Entry Queue IVR Service Properties - Conference ID,"* on page **16-21**.

9 Click the **Video Services** tab.
The *New Entry Queue IVR Service - Video Services* dialog box opens.

10 In the *Video Welcome Slide* list, select the video slide that will be displayed to participants connecting to the Entry Queue. The slide list includes the video slides that were previously uploaded to the MCU memory.

11 Click the **Operator Assistance** tab.

The *Operator Assistance* dialog box opens.



Entry Queue IVR Service Name: EQ Assistance

☑ Enable Operator Assistance

Operator Assistance Indication Message:

12 Select **Enable Operator Assistance** to enable operator assistance when the participant requires or requests help during the connection process.

13 In the *Operator Assistance Indication Message* field, select the audio message to be played when the participant requests or is waiting for operator's assistance.

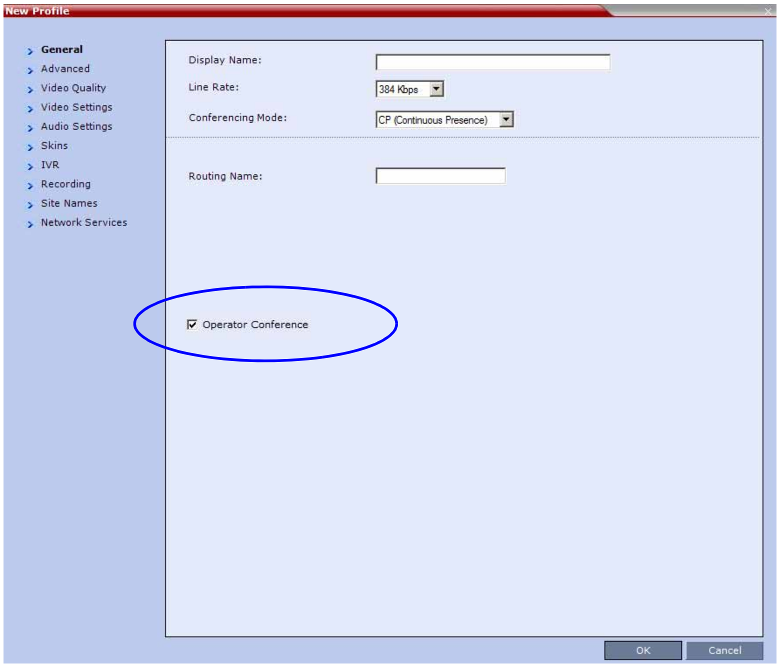
If the audio file was not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the Collaboration Server.

14 Click **OK** to complete the Entry Queue IVR Service definition.
The new Entry Queue IVR Service is added to the *IVR Services* list.

Defining a Conference Profile for an Operator Conference

1 In the *RealPresence Collaboration Server Management* pane, click **Conference Profiles**.

2 In the *Conference Profiles* pane, click the **New Profile** button.
The *New Profile – General* dialog box opens.



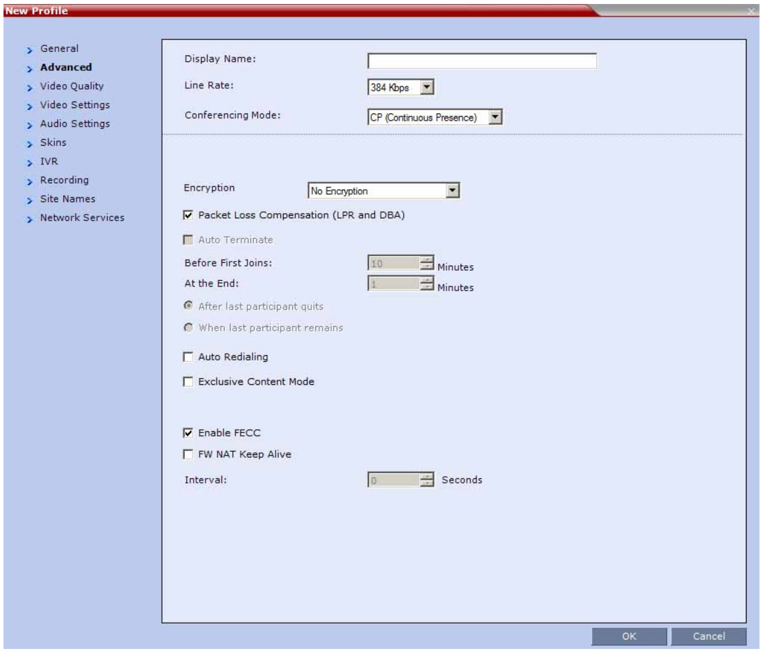
Display Name:
Line Rate: 384 Kbps
Conferencing Mode: CP (Continuous Presence)
Routing Name:
☑ Operator Conference

3 Define the Profile name and, if required, the Profile general parameters.
For more details, see Table 2-10, *"New AVC CP Profile - General Parameters,"* on page 2-21.

4 Click the **Operator Conference** check box.

5 Click the **Advanced** tab.

The *New Profile – Advanced* dialog box opens.



Display Name:
Line Rate: 384 Kbps
Conferencing Mode: CP (Continuous Presence)
Encryption No Encryption
☑ Packet Loss Compensation (LPR and DBA)
☐ Auto Terminate
Before First Joins: 10 Minutes
At the End: 1 Minutes
● After last participant quits
○ When last participant remains
☐ Auto Redialing
☐ Exclusive Content Mode
☑ Enable FECC
☐ FW NAT Keep Alive
Interval: 0 Seconds

6 Define the Profile *Advanced* parameters.

Note that when Operator Conference is selected, the **Auto Terminate** selection is automatically cleared and disabled and the Operator conference cannot automatically end unless it is terminated by the Collaboration Server User.

7 Click the **Video Quality** tab.

The *New Profile – Video Quality* dialog box opens.

8 Define the Video Quality parameters.

9 Click the **Video Settings** tab.

The *New Profile - Video Settings* dialog box opens.

10 Define the video display mode and layout. For more details, see *Table 2-13, "New AVC CP Profile - Video Settings Parameters,"* on page 2-28.

11 Define the remaining Profile parameters.

12 Click **OK** to complete the *Profile* definition.
A new *Profile* is created and added to the *Conference Profiles* list.

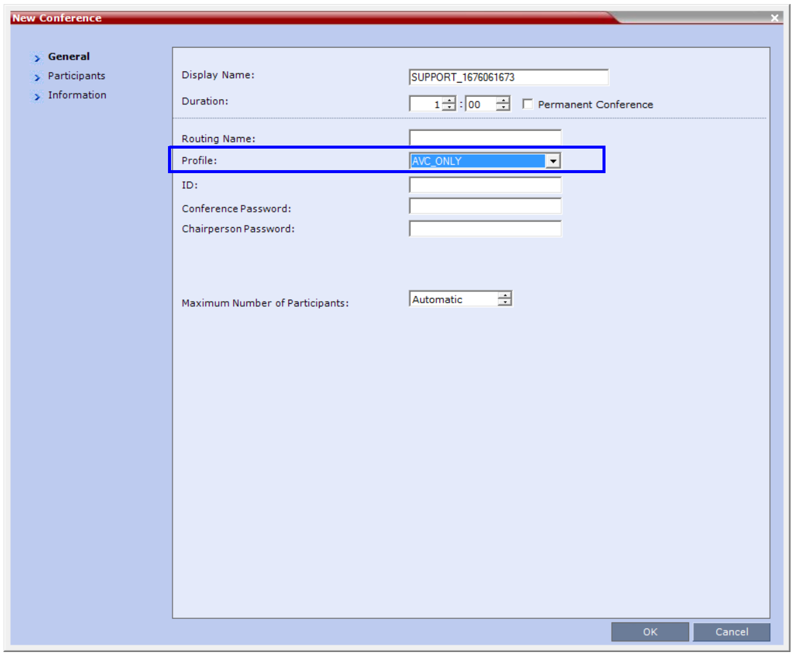
Starting an Ongoing Operator Conference

To start a conference from the Conference pane:

1 In the *Conferences* pane, click the **New Conference** (✚) button.

The *New Conference – General* dialog box opens.

2 In the *Profile* field, select a Profile in which the *Operator Conference* option is selected.



Display Name: SUPPORT_1676061673

Duration: 1 : 00 □ Permanent Conference

Routing Name:

Profile: AVC_ONLY

ID:

Conference Password:

Chairperson Password:

Maximum Number of Participants: Automatic

Upon selection of the *Operator Conference* Profile, the *Display Name* is automatically taken from the Collaboration Server User *Login Name*. This name cannot be modified.

Only one Operator conference can be created for each User Login name.

3 Define the following parameters:

Table 9-1 New Conference – General Options

Field	Description
Duration	Define the duration of the conference in hours using the format HH:MM (default 01:00). **Notes:** • The Operator conference is automatically extended up to a maximum of 168 hours. Therefore, the default duration can be used. • This field is displayed in all tabs.
Routing Name	*Routing Name* is the name with which ongoing conferences, Meeting Rooms, Entry Queues and SIP Factories register with various devices on the network such as gatekeepers and SIP servers. This name must be defined using ASCII characters. **Comma, colon and semicolon characters cannot be used in the *Routing Name*.** The *Routing Name* can be defined by the user or automatically generated by the system if no *Routing Name* is entered as follows: • If ASCII characters are entered as the *Display Name*, it is used also as the *Routing Name* • If a combination of Unicode and ASCII characters (or full Unicode text) is entered as the *Display Name*, the *ID* (such as Conference ID) is used as the *Routing Name.* If the same name is already used by another conference, Meeting Room or Entry Queue, the Collaboration Server displays an error message and requests that you to enter a different name.
ID	Enter the unique-per-MCU conference ID. If left blank, the MCU automatically assigns a number once the conference is launched. This ID must be communicated to conference participants to enable them to dial in to the conference.
Conference Password	Leave this field empty when defining an Operator conference.
Chairperson Password	Leave this field empty when defining an Operator conference.
Maximum Number of Participants	Enter the maximum number of participants that can connect to an Operator conference (you can have more than two), or leave the default selection (Automatic). Maximum number of participants that can connect to an Operator conference:

4 Click the **Participants** tab.
The *New Conference - Participants* dialog box opens.

You must define or add the Operator participant to the Operator conference.

This participant must be defined as a **dial-out** participant.

Define the parameters of the endpoint that will be used by the Collaboration Server User to connect to the Operator conference and to other conference to assist participants.

For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Participants Tab"* on page 3-17.

5 **Optional**. Click the **Information** tab**.**
The *Information* tab opens.

6 Enter the required information. For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Information Tab"* on page 3-20.

7 Click **OK**.

The new Operator conference is added to the ongoing *Conferences* list with a special icon .

The Operator participant is displayed in the *Participants* list with an Operator participant icon , and the system automatically dials out to the Operator participant.

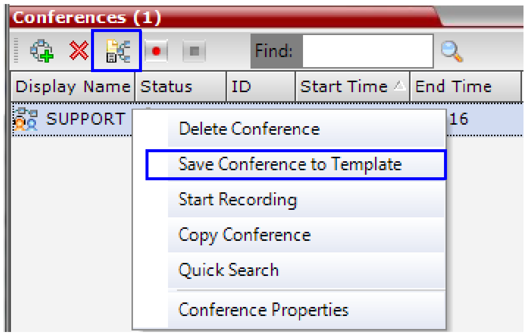
Saving an Operator Conference to a Template

The Operator conference that is ongoing can be saved as a template.

To save an ongoing Operator conference as a template:

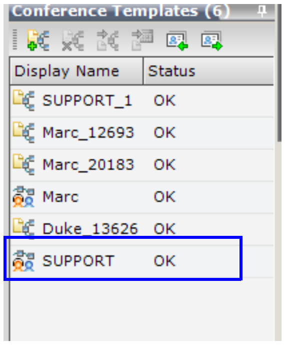
1 In the *Conferences List*, select the Operator conference you want to save as a Template.

2 Click the **Save Conference to Template** () button.
or
Right-click and select **Save Conference to Template**.



Display Name	Status	ID	Start Time	End Time
SUPPORT				16

The conference is saved to a template whose name is taken from the ongoing conference *Display Name* (the Login name of the Collaboration Server User). The Template is displayed with the Operator Conference icon.



Display Name	Status
SUPPORT_1	OK
Marc_12693	OK
Marc_20183	OK
Marc	OK
Duke_13626	OK
SUPPORT	OK

Starting an Operator Conference from a Template

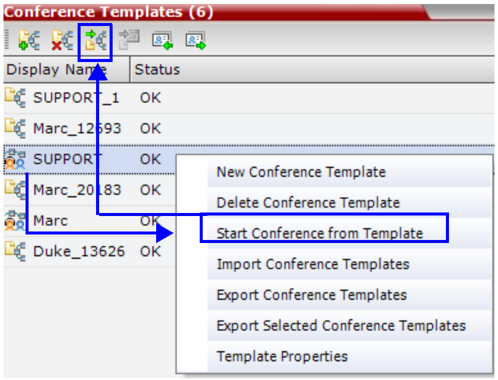
An ongoing Operator conference can be started from an Operator Template saved in the *Conference Templates* list.

To start an ongoing Operator conference from an Operator Template:

1 In the *Conference Templates* list, select the Operator Template to start as an ongoing Operator conference.

> - You can only start an Operator conference from a template whose name is identical to your Login Name. For example, if your Login name is Polycom, you can only start an Operator conference from a template whose name is Polycom.
> - If an ongoing Operator conference with the same name or any other conference with the same ID is already running, you cannot start another Operator conference with the same login name.

2 Click the **Start Conference from Template** ( ) button.
or
Right-click and select **Start Conference from Template**.



Conference Templates (6)

Display Name	Status
SUPPORT_1	OK
Marc_12693	OK
SUPPORT	OK
Marc_20183	OK
Marc	OK
Duke_13626	OK

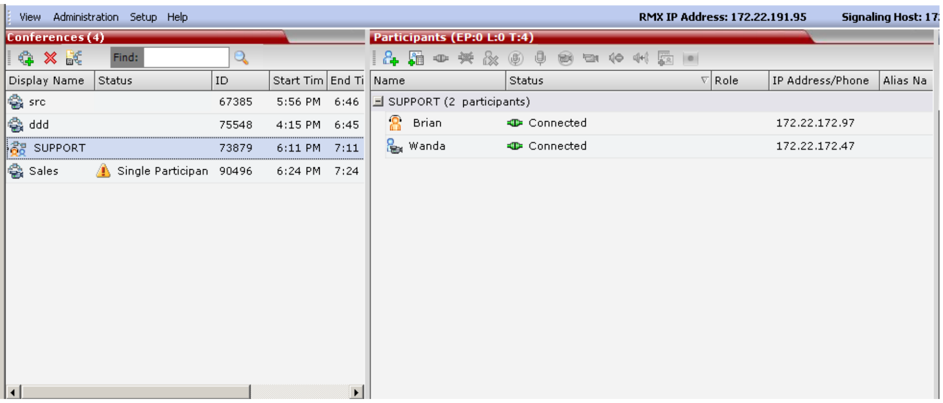
New Conference Template
Delete Conference Template
Start Conference from Template
Import Conference Templates
Export Conference Templates
Export Selected Conference Templates
Template Properties

The conference is started.

The name of the ongoing conference in the *Conferences* list is taken from the Conference Template *Display Name*.

Monitoring Operator Conferences and Participants Requiring Assistance

Operator conferences are monitored in the same way as standard ongoing conferences. Each Operator conference includes at least one participant - the Operator.


View Administration Setup Help
RMX IP Address: 172.22.191.95
Signaling Host: 17
Conferences (4)
Find:
Participants (EP:0 L:0 T:4)

Display Name	Status	ID	Start Tim	End Ti
src		67385	5:56 PM	6:46
ddd		75548	4:15 PM	6:45
SUPPORT		73879	6:11 PM	7:11
Sales	Single Participan	90496	6:24 PM	7:24

Name	Status	Role	IP Address/Phone	Alias Na
SUPPORT (2 participants)				
Brian	Connected		172.22.172.97	
Wanda	Connected		172.22.172.47	

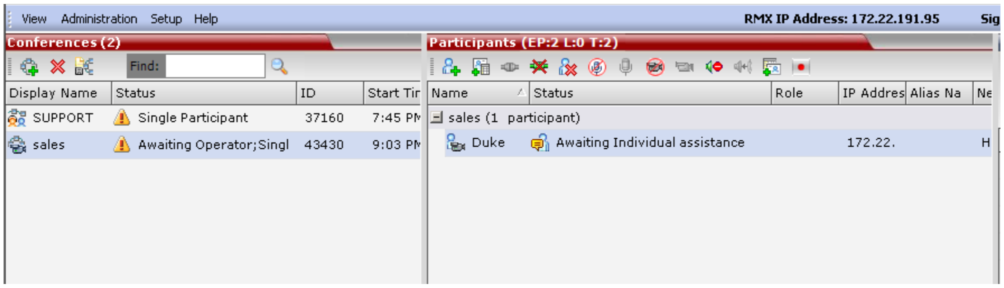
You can view the properties of the *Operator conference* by double-clicking the conference entry in the *Conferences* list or by right-clicking the conference entry and selecting **Conference Properties**. For more information, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Conference Level Monitoring*" on page **3-45**.

Requesting Help

A participant can request help using the appropriate DTMF code from his/her touch tone telephone or the endpoint's DTMF input device. The participant can request *Individual Assistance* (default DTMF code ***0**) or *Conference Assistance* (default DTMF code **00**).

Participants in Entry Queues who failed to enter the correct destination conference ID or the conference password will wait for operator assistance (provided that an Operator conference is active).

When requiring or requesting operator assistance, the Collaboration Server management application displays the following:


View Administration Setup Help
RMX IP Address: 172.22.191.95
Sig
Conferences (2)
Find:
Participants (EP:2 L:0 T:2)

Display Name	Status	ID	Start Tir
SUPPORT	Single Participant	37160	7:45 PM
sales	Awaiting Operator;Singl	43430	9:03 PM

Name	Status	Role	IP Addres	Alias Na	Ne
sales (1 participant)					
Duke	Awaiting Individual assistance		172.22.		H

- The participant's connection *Status* changes, reflecting the help request. For more information, see Table 9-2.
- The conference status changes and it is displayed with the exclamation point icon and the status "Awaiting Operator".
- The appropriate voice message is played to the relevant participants indicating that assistance will be provided shortly.

The following icons and statuses are displayed in the *Participant Status* column:

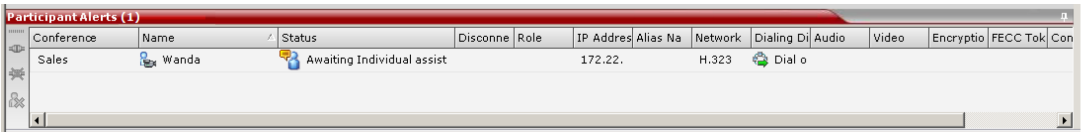
Table 9-2 Participants List Status Column Icons and Indications

Icon	Status Indication	Description
👤	*Awaiting Individual Assistance*	The participant has requested the operator's assistance for himself/herself.
👥	*Awaiting Conference Assistance*	The participant has requested the operator's assistance for the conference. Usually this means that the operator is requested to join the conference.

When the Operator moves the participant to the *Operator conference* for individual assistance the participant Status indications are cleared.

Participant Alerts List

The *Participant Alerts* list contains all the participants who are currently waiting for operator assistance.



Participant Alerts (1)

Conference	Name	Status	Disconne	Role	IP Addres	Alias Na	Network	Dialing Di	Audio	Video	Encryptio	FECC Tok	Con
Sales	Wanda	Awaiting Individual assist			172.22.		H.323	Dial o					

Participants are automatically added to the *Participants Alerts* list in the following circumstances:

- The participant fails to connect to the conference by entering the wrong conference ID or conference password and waits for the operator's assistance
- The participant requests Operator's Assistance during the ongoing conference

This list is used as reference only. Participants can be assisted and moved to the *Operator conference* or the destination conference only from the *Participants* list of the Entry Queues or ongoing conference where they are awaiting assistance.

The participants are automatically removed from the *Participant Alerts* list when moved to any conference (including the *Operator conference*).

Audible Alarms

In addition to the visual cues used to detect events occurring on the Collaboration Server, an audible alarm can be activated and played when participants request Operator Assistance.

Using Audible Alarms

The Audible Alarm functionality for Operator Assistance requests is enabled for each MCU in either the Collaboration Server Web Client or RMX Manager.

The Audible Alarm played when Operator Assistance is requested is enabled and selected in the **Setup > Audible Alarm > User Customization**. When the Audible Alarm is activated, the *.wav file selected in the *User Customization* is played, and it is repeated according to the number of repetitions defined in the *User Customization*.

If more than one Collaboration Server is monitored in the *RMX Manager*, the Audible Alarm must be enabled separately for each Collaboration Server installed in the site/configuration. A different *.wav file can be selected for each MCU.

When multiple Audible Alarms are activated in different conferences or by multiple MCUs, the Audible Alarms are synchronized and played one after the other. It is important to note that when *Stop Repeating Alarm* is selected from the toolbar from the Collaboration Server Web Client or RMX Manager, all activated Audible Alarms are immediately halted.

For more details on Audible alarms and their configuration, see "*Audible Alarms"* on page **19-31**.

Moving Participants Between Conferences

The Collaboration Server User can move participants between ongoing conferences, including the *Operator conference*, and from the Entry Queue to the destination conference if help is required.

When moving between conferences or when a participant is moved from an Entry Queue to a conference by the Collaboration Server user (after failure to enter the correct destination ID or conference password), the IVR messages and slide display are skipped.

Move Guidelines

- Move is available only between CP conferences. Move is unavailable from/to Video Switching conferences.
- Move between conferences can be performed without an active *Operator conference*.
- When moving the conference chairperson from his/her conference to another conference, the source conference will automatically end if the *Auto Terminate When Chairperson Exits* option is enabled and that participant is the only conference chairperson.
- When moving the Operator to any conference (following assistance request), the IVR messages and slide display are skipped.
- Participants cannot be moved from LPR-enabled conferences to non-LPR conferences. Move from non-LPR conferences to LPR-enabled conferences is available.
- Move between encrypted and non-encrypted conferences depends on the **ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF** flag setting, as described in Table 9-3:

Table 9-3 *Participant Move Capabilities vs. ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF flag setting*

Flag Setting	Source Conference/ EQ Encrypted	Destination Conference Encrypted	Move Enabled?
NO	Yes	Yes	Yes
NO	Yes	No	Yes
NO	No	Yes	No
NO	No	No	Yes
YES	Yes	Yes	Yes

Table 9-3 *Participant Move Capabilities vs. ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF flag setting (Continued)*

Flag Setting	Source Conference/ EQ Encrypted	Destination Conference Encrypted	Move Enabled?
YES	Yes	No	Yes
YES	No	Yes	Yes
YES	No	No	Yes

- When moving dial-out participants who are disconnected to another conference, the system automatically dials out to connect them to the destination conference.
- Cascaded links cannot be moved between conferences.
- Participants cannot be moved to a conference if the move will cause the number of participants to exceed the maximum number of participants allowed for the destination conference.

Moving Participants Options

Collaboration Server users can assist participants by performing the following operations:

- Move a participant to an *Operator conference* (Attend a participant).
- Move a participant to the Home (destination) conference.
- Move participant from one ongoing conference to another

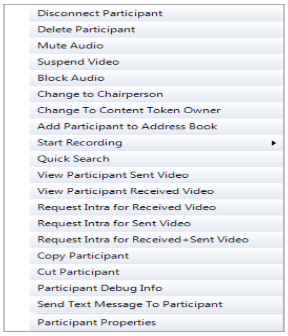
A move can be performed using the following methods:

- Using the participant right-click menu
- Using drag and drop

To move a participant from the ongoing conference using the right-click menu options:

1 In the *Conferences* list, click the conference where there are participants waiting for Operator's Assistance to display the list of participants.

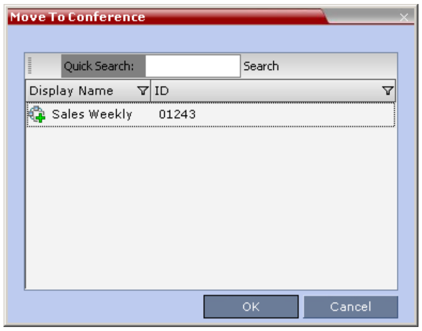
2 In the *Participants* list, right-click the icon of the participant to move and select one of the following options:


Disconnect Participant
Delete Participant
Mute Audio
Suspend Video
Block Audio
Change to Chairperson
Change To Content Token Owner
Add Participant to Address Book
Start Recording ▸
Quick Search
View Participant Sent Video
View Participant Received Video
Request Intra for Received Video
Request Intra for Sent Video
Request Intra for Received+Sent Video
Copy Participant
Cut Participant
Participant Debug Info
Send Text Message To Participant
Participant Properties

— **Move to Operator Conference** - to move the participant to the Operator conference.

— **Move to Conference** - to move the participant to any ongoing conference.

When selected, the *Move to Conference* dialog box opens, letting you select the name of the destination conference.



Display Name	ID
Sales Weekly	01243

— **Back to Home Conference** - if the participant was moved to another conference or to the *Operator conference*, this options moves the participant back to his/her source conference.

This option is not available if the participant was moved from the Entry Queue to the *Operator conference* or the destination conference.

Moving a Participant Interactively

You can drag and drop a participant from the Entry Queue or ongoing conference to the Operator or destination (Home) conference:

1 Display the participants list of the Entry Queue or the source conference by clicking its entry in the *Conferences* list.

2 In the Participants list, drag the icon of the participant to the *Conferences List* pane and drop it on the *Operator Conference* icon or another ongoing conference.

10

Conference Templates

Conference Templates enable administrators and operators to create, save, schedule and activate identical conferences.

A *Conference Template*:

- Saves the conference Profile.
- Saves all participant parameters including their *Personal Layout* and *Video Forcing* settings.

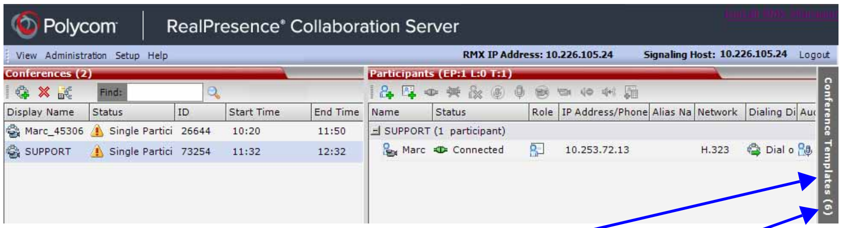
Guidelines

- The maximum number of templates is 100.

  Trying to start a *Conference Template* that exceeds the allowed maximum number of participants will result in participants being disconnected due to resource deficiency.
- If the Profile assigned to a conference is deleted while the conference is ongoing the conference cannot be saved as a template.
- A Profile assigned to a *Conference Template* cannot be deleted. The system does not permit such a deletion.
- Profile parameters are not embedded in the *Conference Template*, and are taken from the Profile when the *Conference Template* becomes an ongoing conference. Therefore, any changes to the Profile parameters between the time the *Conference Template* was created and the time that it is activated (and becomes an ongoing conference) will be applied to the conference.
- Only defined participants can be saved to the *Conference Template*. Before saving a conference to a template ensure that all undefined participants have disconnected.
- Undefined participants are not saved in *Conference Templates*.
- Participant properties are embedded in the *Conference Template* and therefore, if the participant properties are modified in the Address Book after the *Conference Template* has been created they are not applied to the participant whether the *Template* becomes an ongoing conference or not.
- The *Conference Template* display name, routing name or ID can be the same as an Ongoing Conference, reservation, Meeting Room or Entry Queue as it is not active. However, an ongoing conference cannot be launched from the *Conference Template* if an ongoing conference, Meeting Room or Entry Queue already has the same name or ID. Therefore, it is recommended to modify the template ID, display name, routing name to be unique.
- SIP Factories and Entry Queues cannot be saved as *Conference Templates*.
- The conference specified in the *Conference Template* can be designated as a *Permanent Conference.* For more information see "*Lecture Mode (AVC CP Only)"* on page **3-38**.

Using Conference Templates

The *Conference Templates* list is initially displayed as a closed tab in the *Collaboration Server Web Client* main window. The number of saved *Conference Templates* is indicated on the tab.


Polycom
RealPresence® Collaboration Server
View Administration Setup Help
RMX IP Address: 10.226.105.24
Signaling Host: 10.226.105.24
Logout
Conferences (2)
Find:
Participants (EP:1 L:0 T:1)
Conference Templates (6)

Display Name	Status	ID	Start Time	End Time
Marc_45306	Single Partici	26644	10:20	11:50
SUPPORT	Single Partici	73254	11:32	12:32

Name	Status	Role	IP Address/Phone	Alias Na	Network	Dialing Di	Auc
SUPPORT (1 participant)							
Marc	Connected		10.253.72.13		H.323	Dial o	

Conference Templates Tab

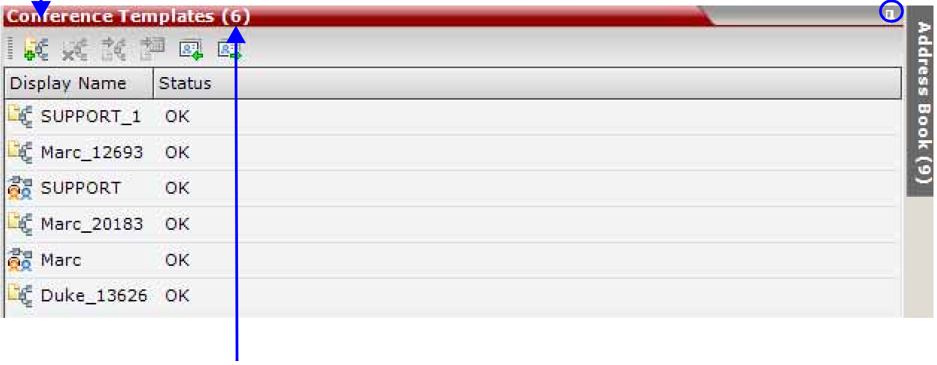
Number of Saved Conference Templates

Clicking the tab opens the *Conference Templates* list.

Toolbar buttons

Click to hide the Conference Templates List

List of Saved Templates



Display Name	Status
SUPPORT_1	OK
Marc_12693	OK
SUPPORT	OK
Marc_20183	OK
Marc	OK
Duke_13626	OK

Number of Saved Conference Templates

The *Conference Templates* are listed by *Conference Template Display Name* and *ID* and can be sorted by either field. The list can be customized by re-sizing the pane, adjusting the column widths or changing the order of the column headings.

For more information see *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Customizing the Main Screen"* on page 3-10.

Clicking the anchor pin (⊕) button hides the *Conference Templates* list as a closed tab.

Toolbar Buttons

The *Conference Template* toolbar includes the following buttons:

Table 1 *Conference Templates – Toolbar Buttons*

Button	Description
New Conference Template	Creates a new Conference Template.
Delete Conference Template	Deletes the Conference Template(s) that are selected in the list.

Table 1 *Conference Templates – Toolbar Buttons (Continued)*

Button	Description
Start Conference from Template	Starts an ongoing conference from the *Conference Template* that has an identical name, ID parameters and participants as the template.
Schedule Reservation from Template	Creates a conference Reservation from the Conference Template with the same name, ID, parameters and participants as the Template. Opens the *Scheduler* dialog box enabling you to modify the fields required to create a single or recurring *Reservation* based on the template. For more information see "*Reservations”* on page **9-1**.

The ***Conferences List*** toolbar includes the following button:

Table 2 *Conferences List – Toolbar Button*

Button	Description
Save Conference to Template	Saves the selected ongoing conference as a Conference Template.

Creating a New Conference Template

There are two methods to create a *Conference Template*:

- From scratch - defining the conference parameters and participants
- Saving an ongoing conference as Template

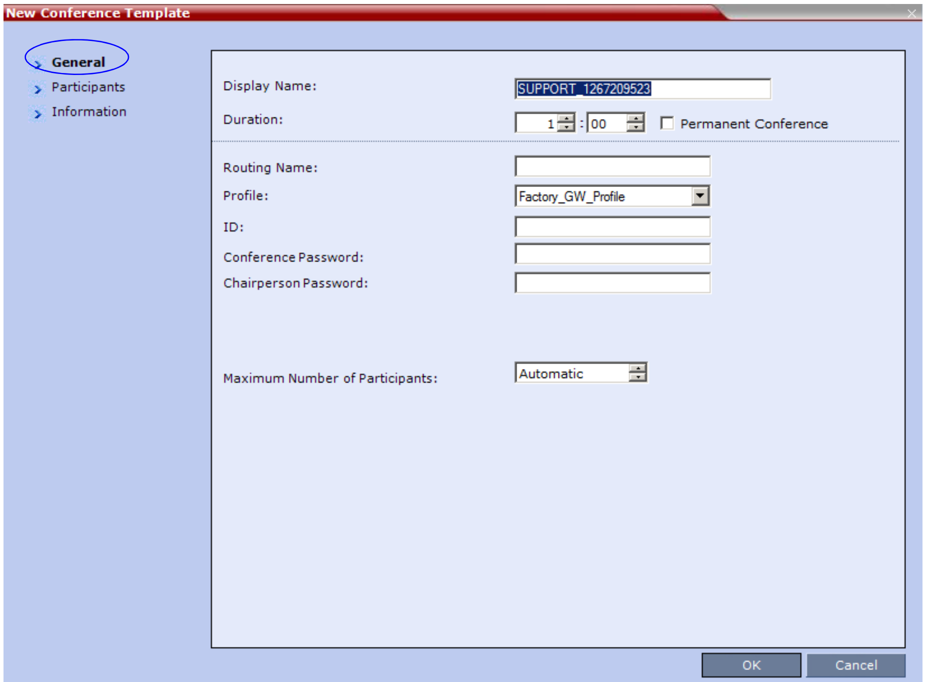
Creating a new Conference Template from Scratch

To create a new Conference Template:

1 In the *Collaboration Server Polycom® RealPresence Collaboration Server Virtual Edition Administrator's Guide*, click the **Conference Templates** tab.

2 Click the **New Conference Template** ( ) button.

The *New Conference Template - General* dialog box opens.



Display Name: SUPPORT_1267209523

Duration: 1 : 00 ☐ Permanent Conference

Routing Name:

Profile: Factory_GW_Profile

ID:

Conference Password:

Chairperson Password:

Maximum Number of Participants: Automatic

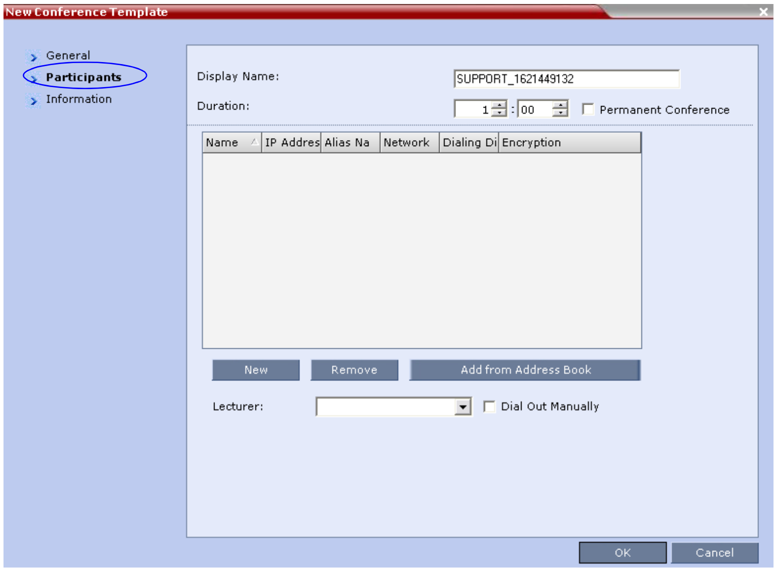
3 The fields of the *New Template – General* dialog box are identical to those of the *New Conference – General* dialog box. For a full description of the fields see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "General Tab"* on page **3-14**.

4 Modify the fields of the *General* tab.

> A unique dial-in number must be assigned to each conferencing entity. However, Conference Templates can be assigned dial-in numbers that are already assigned to other conferencing entities, but when the template is used to start an ongoing conference or schedule a reservation, it will not start if another ongoing conference, Meeting Room, or Entry Queue is using this number.

5 Click the **Participants** tab.

The *New Template – Participants* dialog box opens.



Display Name: SUPPORT_1621449132

Duration: 1 : 00 ☐ Permanent Conference

Lecturer: ☐ Dial Out Manually

The fields of the *New Template – Participants* dialog box are the same as those of the *New Conference – Participant* dialog box.

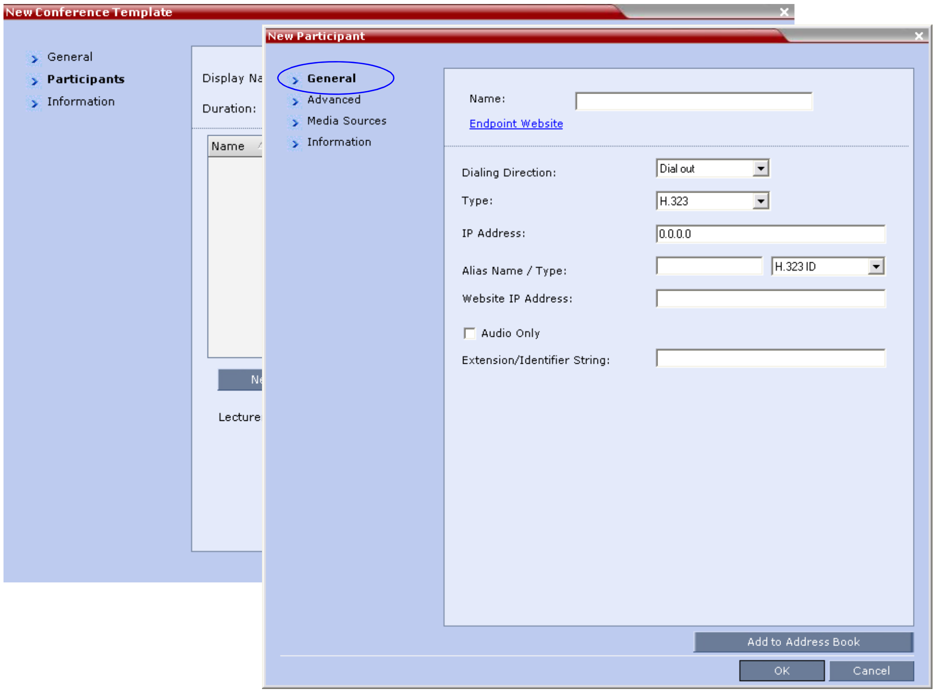
For a full description of these fields see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Participants Tab"* on page **3-17**.

6 **Optional.** Add participants to the template from the *Address Book*.

7 Click the **New** button.

The *New Participant – General* tab opens.

The *New Template – Participant* dialog box remains open in the background.



Name:

Dialing Direction: Dial out

Type: H.323

IP Address: 0.0.0.0

Alias Name / Type: H.323 ID

Website IP Address:

☐ Audio Only

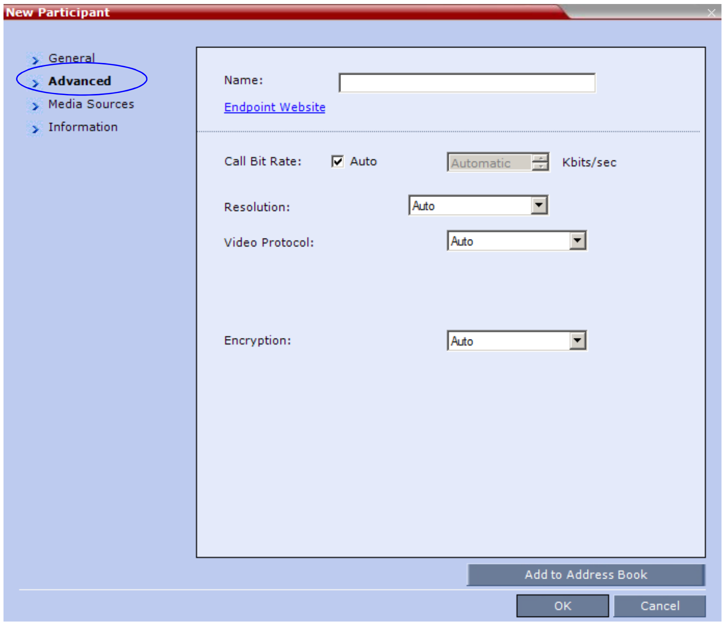
Extension/Identifier String:

For a full description of the *General* tab fields see "*Adding a New participant to the Address Book Directly*" on page 8-8.

8 Modify the fields of the *General* tab.

9 Click the **Advanced** tab.

The *New Participant – Advanced* tab opens.



Name:

Call Bit Rate: ☑ Auto Automatic Kbits/sec

Resolution: Auto

Video Protocol: Auto

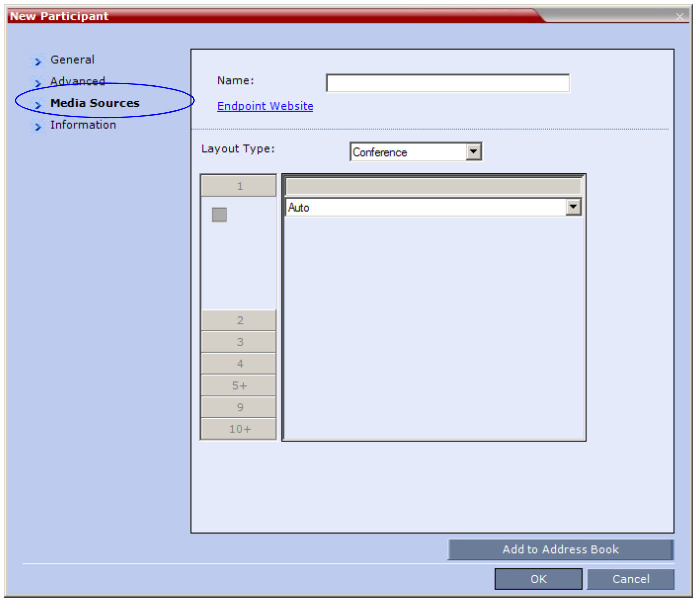
Encryption: Auto

For a full description of the *Advanced* tab fields see, "*New Participant – Advanced Properties*" on page 8-12.

10 Modify the fields of the *Advanced* tab.

11 Click the **Media Sources** tab.

The *Media Sources* tab opens.



Name:

Layout Type: Conference

Auto

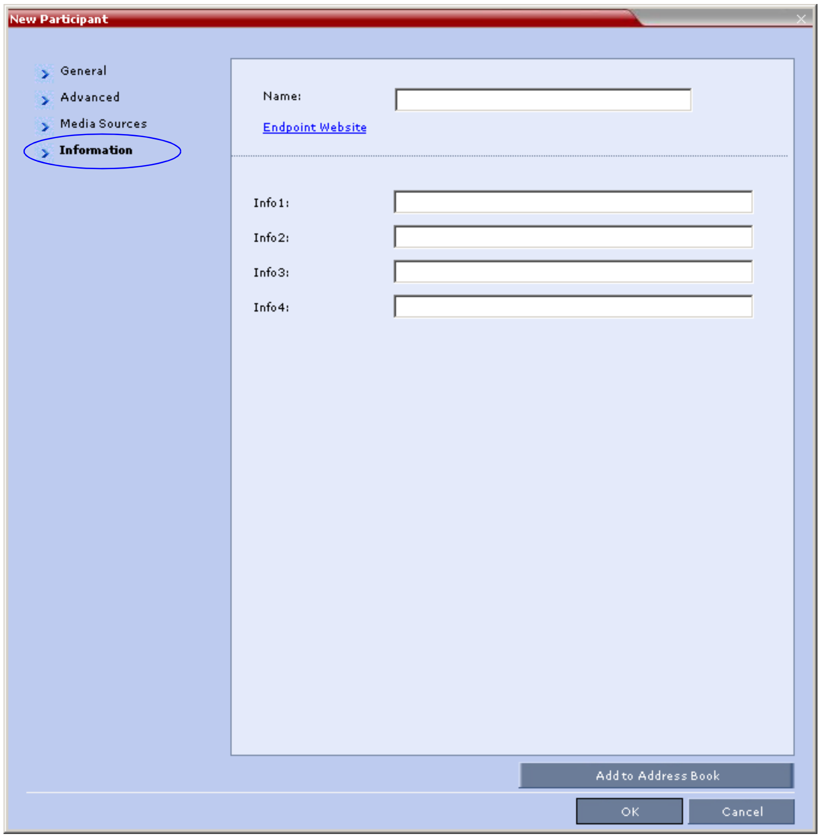
The *Media Sources* tab enables you to set up and save *Personal Layout* and *Video Forcing* settings for each participant.

For a full description of *Personal Layout* and *Video Forcing* settings see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Changing the Video Layout of a Conference (AVC-Based CP and Mixed CP and SVC Conferences)"* on page **3-59** and "*Video Forcing (AVC-Based CP and Mixed CP and SVC Conferences)"* on page **3-61**.

12 Modify the *Personal Layout* and *Video Forcing* settings for the participant.

13 **Optional.** Click the **Information** tab.

The *New Participant – Information* tab opens.



For a full description of the *Information* fields see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Information Tab"* on page **3-20**.

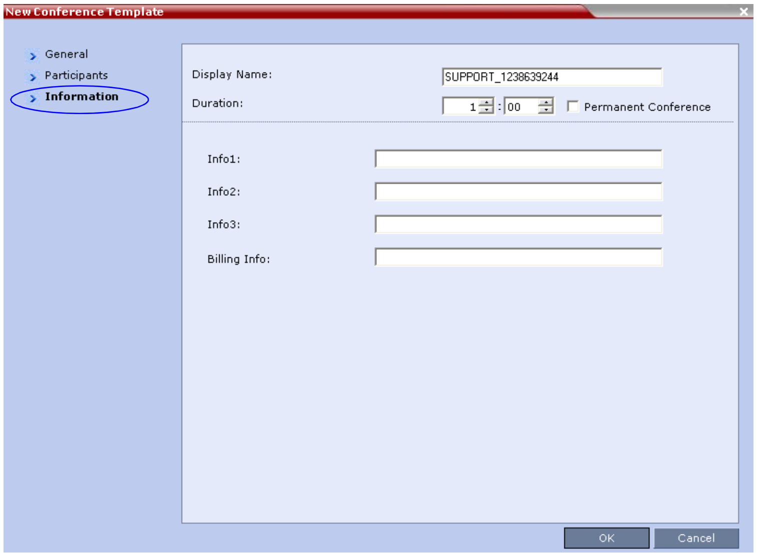
14 Click the **OK** button.

The participant you have defined is added to the *Participants List*.

The New Participant dialog box closes and you are returned to the *New Template – Participant* dialog box (which has remained open since Step 7).

15 **Optional.** In the *New Conference Template* dialog box, click the **Information** tab.

The *New Conference Template – Information* tab opens.



Display Name: SUPPORT_1238639244

Duration: 1 : 00 □ Permanent Conference

Info1:

Info2:

Info3:

Billing Info:

For a full description of the *Information* fields see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Information Tab"* on page 20.

16 Click the **OK** button.

The *New Conference Template* is created and its name is added to the *Conference Templates* list.

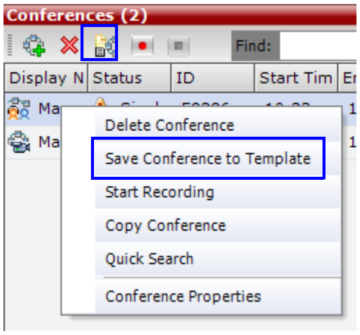
Saving an Ongoing or AVC-based CP Operator Conference as a Template

Any ongoing or AVC-based CP *Operator Conference* can be saved as a template.

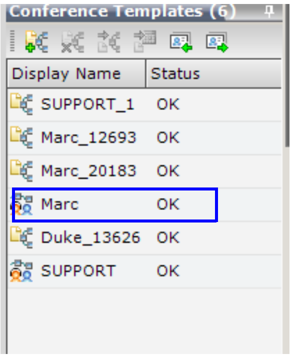
To save an ongoing or AVC-based CP Operator Conference as a template:

1 In the *Conferences List*, select the conference or *Operator Conference* to be saved as a Template.

2 Click the **Save Conference to Template** ( ) button.
or
Right-click and select **Save Conference to Template**.



The conference is saved to a template whose name is taken from the ongoing conference *Display Name* (the *Login* name of the *Collaboration Server User*). The *Template* is displayed with the *Operator Conference* icon.



Conference Templates (6)

Display Name	Status
SUPPORT_1	OK
Marc_12693	OK
Marc_20183	OK
Marc	OK
Duke_13626	OK
SUPPORT	OK

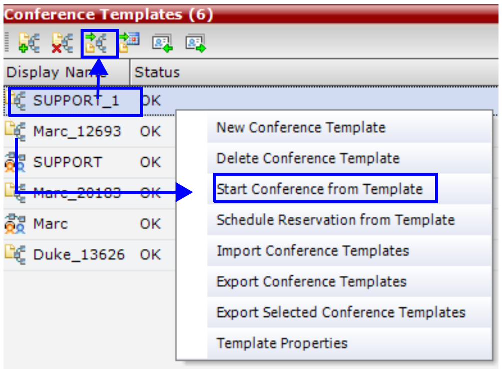
Starting an Ongoing Conference From a Template

An ongoing conference can be started from any Template saved in the *Conference Templates* list. In SVC-based templates, only defined dial-in participants may be part of the conference.

To start an ongoing conference from a Template:

1 In the *Conference Templates* list, select the Template you want to start as an ongoing conference.

2 Click the **Start Conference from Template** () button.
or
Right-click and select **Start Conference from Template**.



Conference Templates (6)

Display Name	Status
SUPPORT_1	OK
Marc_12693	OK
SUPPORT	OK
Marc_20183	OK
Marc	OK
Duke_13626	OK

New Conference Template
Delete Conference Template
Start Conference from Template
Schedule Reservation from Template
Import Conference Templates
Export Conference Templates
Export Selected Conference Templates
Template Properties

The conference is started.

The name of the ongoing conference in the *Conferences* list is taken from the Conference Template *Display Name*.

Participants that are connected to other ongoing conferences when the template becomes an ongoing conference are not connected.

If an ongoing conference, Meeting Room or Entry Queue with the same *Display Name, Routing Name* or *ID* already exists in the system, the conference will not be started.

Starting an Operator Conference from a Template (AVC Conferencing)

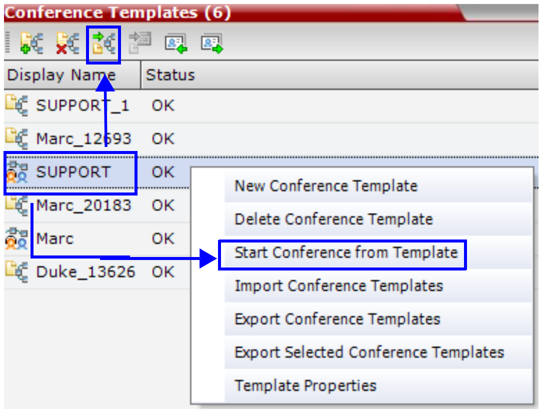
An ongoing Operator conference can be started from an Operator Template saved in the *Conference Templates* list.

To start an ongoing Operator conference from an Operator Template:

1 In the *Conference Templates* list, select the Operator Template to start as an ongoing Operator conference.

- You can only start an Operator conference from a template whose name is identical to your Login Name. For example, if your Login name is Polycom, you can only start an Operator conference from a template whose name is Polycom.
- If an ongoing Operator conference with the same name or any other conference with the same ID is already running, you cannot start another Operator conference with the same login name.

2 Click the **Start Conference from Template** ( ) button.
or
Right-click and select **Start Conference from Template**.


Conference Templates (6)
Display Name
Status
SUPPORT_1
OK
Marc_12693
OK
SUPPORT
OK
Marc_20183
OK
Marc
OK
Duke_13626
OK
New Conference Template
Delete Conference Template
Start Conference from Template
Import Conference Templates
Export Conference Templates
Export Selected Conference Templates
Template Properties

The conference is started.

The name of the ongoing conference in the *Conferences* list is taken from the Conference Template *Display Nam*e.

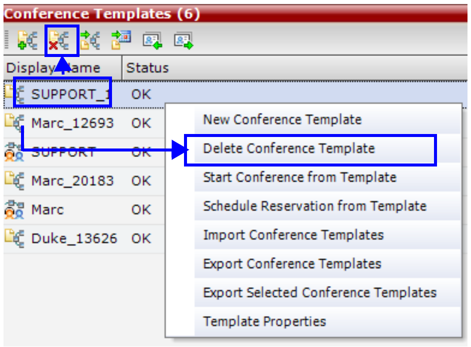
Deleting a Conference Template

One or several *Conference Templates* can be deleted at a time.

To delete Conference Templates:

1 In the *Conference Templates* list, select the *Template(s)* you want to delete.

2 Click the **Delete Conference Template** ( ) button.
or
Right-click and select **Delete Conference Template**.


Conference Templates (6)
New Conference Template
Delete Conference Template
Start Conference from Template
Schedule Reservation from Template
Import Conference Templates
Export Conference Templates
Export Selected Conference Templates
Template Properties

Display Name	Status
SUPPORT_1	OK
Marc_12693	OK
SUPPORT	OK
Marc_20183	OK
Marc	OK
Duke_13626	OK

A confirmation dialog box is displayed.

3 Click the **OK** button to delete the *Conference Template(s)*.

Exporting and Importing Conference Templates

Conference Templates can be exported from one MCU and imported to multiple MCUs in your environment. Additionally, you can export *Conference Templates* and their associated *Conference Profiles* simultaneously. Using this option can save configuration time and ensures that identical settings are used for conferences running on different MCUs. This is especially important in environments using cascading conferences that are running on different MCUs.

- Administrators can export and import *Conference Templates*. Operators are only allowed to export *Conference Templates*.
- You can select a single, multiple or all *Conference Templates* to be exported.
- Both *Conference Templates* and their associated *Conference Profiles* can be exported and imported simultaneously when enabling the **Export includes conference profiles** or **Import includes conference profiles** options.
- Exporting and importing *Conference Templates* only can be used when you want to export and import individual *Conference Templates* without their associated Conference Profiles. This option enables you to import *Conference Templates* when *Conference Profiles* already exist on an MCU.

Exporting Conference Templates

Conference Templates are exported to a single XML file that can be used to import the *Conference Templates* on multiple MCUs.

Using the *Export Conference Templates* option, you can:

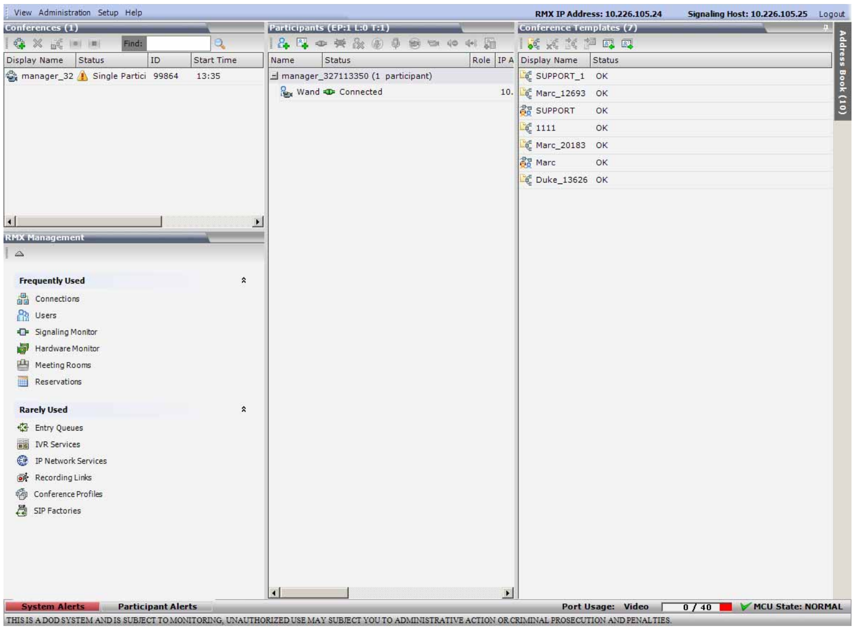
- Export all *Conference Templates* from an MCU
- Export selected *Conference Templates*

Exporting All Conference Templates from an MCU

To export all Conference Templates from an MCU:

1 In the *Collaboration Server Web Client* main window, click the *Conference Templates* tab.

The *Conference Templates* list pane is displayed.



Display Name	Status	ID	Start Time
manager_32	Single Partici	99864	13:35

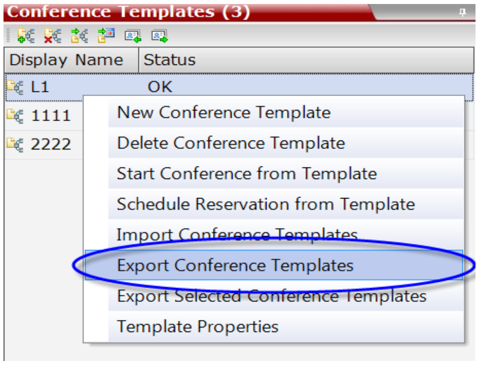
Name	Status	Role	IP A
manager_327113350 (1 participant)			
Wand	Connected		10.

Display Name	Status
SUPPORT_1	OK
Marc_12693	OK
SUPPORT	OK
1111	OK
Marc_20183	OK
Marc	OK
Duke_13626	OK

Port Usage: Video 0 / 40 MCU State: NORMAL

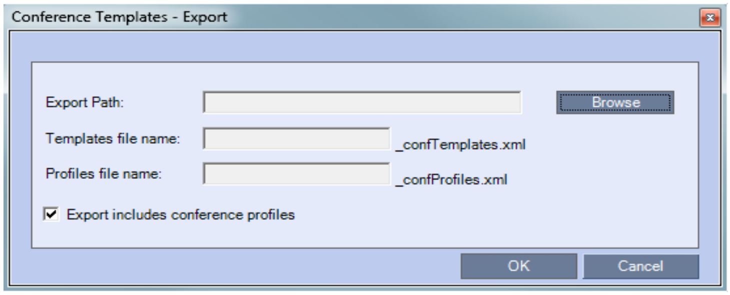
THIS IS A DOD SYSTEM AND IS SUBJECT TO MONITORING, UNAUTHORIZED USE MAY SUBJECT YOU TO ADMINISTRATIVE ACTION OR CRIMINAL PROSECUTION AND PENALTIES.

2 Click the **Export Conference Templates** button or right-click the *Conference Templates* list, and then click **Export Conference Templates**.


Conference Templates (3)
New Conference Template
Delete Conference Template
Start Conference from Template
Schedule Reservation from Template
Import Conference Templates
Export Conference Templates
Export Selected Conference Templates
Template Properties

Display Name	Status
L1	OK
1111	
2222	

The *Conference Templates - Export* dialog box is displayed.



Export Path:

Templates file name: _confTemplates.xml

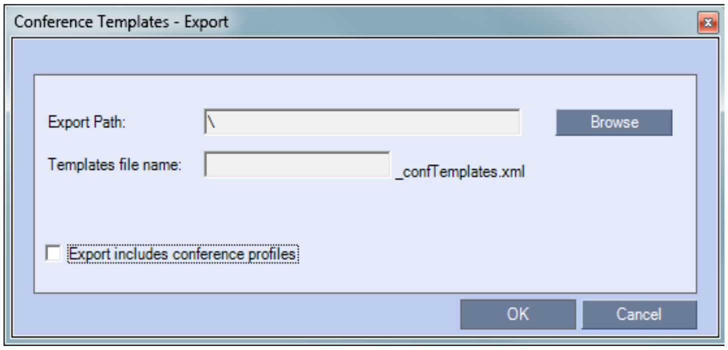
Profiles file name: _confProfiles.xml

☑ Export includes conference profiles

3 In the *Export Path* field, type the path name to the location where you want to save the exported file or click **Browse** to select the desired path.

4 Optional. Clear the **Export includes conference profiles** check box when you only want to export *Conference Templates*.

When this check box is cleared, the *Conference Templates - Export* dialog box is displayed without the *Profiles file name* field.


Conference Templates - Export

Export Path: \

Templates file name: _confTemplates.xml

□ Export includes conference profiles

5 In the *Templates file name* field, type the file name prefix. The file name suffix (_confTemplates.xml) is predefined by the system. For example, if you type *Templates01*, the exported file name is defined as *Templates01_confTemplates.xml*.

The system automatically defines the *Profiles file name* field with the same file name prefix as the *Templates file name* field. For example, if you type *Templates01* in the *Templates file name* field, the exported profiles file name is defined as *Templates01_confProfiles.xml*.

6 Click **OK** to export the *Conference Templates* and *Conference Profiles* to a file.

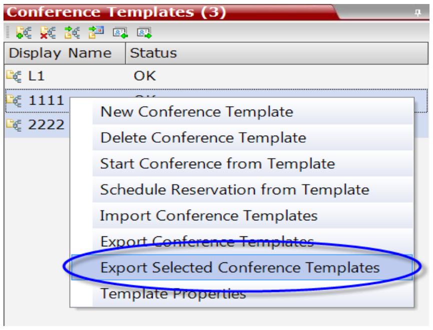
Exporting Selected Conference Templates

You can export a single *Conference Template* or multiple *Conference Templates* to other MCUs in your environment.

To export selected Conference Templates:

1 In the *Conference Templates* list, select the templates you want to export.

2 Right-click the *Conference Templates* to be exported, and then click **Export Selected Conference Templates**.

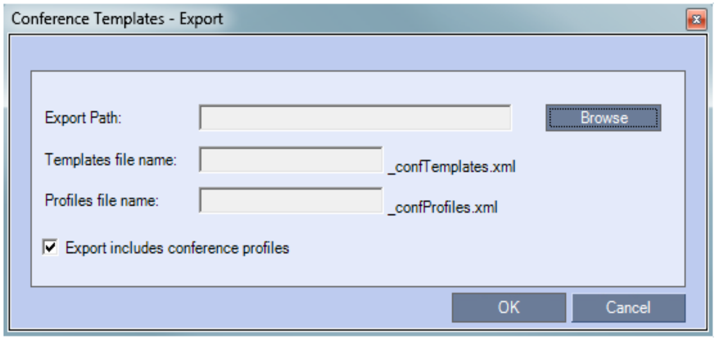

Conference Templates (3)

Display Name	Status
L1	OK
1111	
2222	

New Conference Template
Delete Conference Template
Start Conference from Template
Schedule Reservation from Template
Import Conference Templates
Export Conference Templates
Export Selected Conference Templates
Template Properties

.

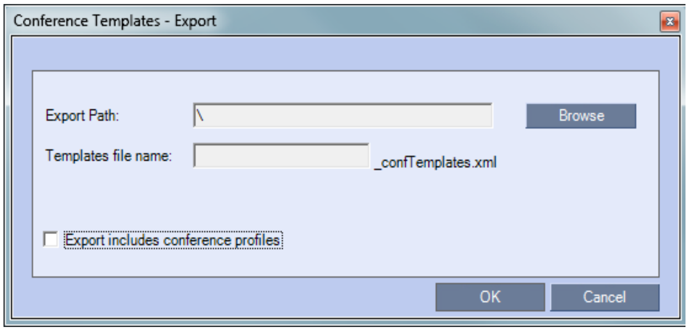
The *Conference Templates - Export* dialog box is displayed.



3 In the *Export Path* field, type the path name to the location where you want to save the exported file or click **Browse** to select the desired path.

4 Optional. Clear the **Export includes conference profiles** check box when you only want to export Conference Templates.

When this check box is cleared, the *Conference Templates - Export* dialog box is displayed without the *Profiles file name* field.


Conference Templates - Export
Export Path:
\
Browse
Templates file name:
_confTemplates.xml
□ Export includes conference profiles
OK
Cancel

5 In the *Templates file name* field, type the file name prefix. The file name suffix (_confTemplates.xml) is predefined by the system. For example, if you type, *Templates01*, the exported file name is defined as *Templates01_confTemplates.xml*.

The system automatically defines the *Profiles file name* field with the same file name prefix as the *Templates file name* field. For example, if you type *Templates01* in the *Templates file name* field, the exported profiles file name is defined as *Templates01_confProfiles.xml*.

6 Click **OK** to export the *Conference Templates* and *Conference Profiles* to a file.

Importing Conference Templates

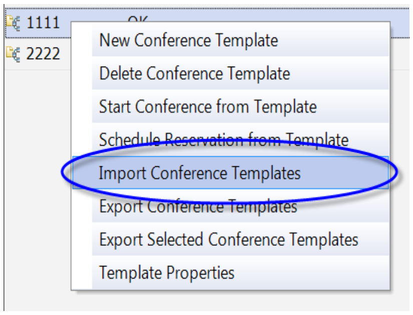
You can import *Conference Templates* and *Conference Profiles* from one MCU to multiple MCUs in your environment.

To import Conference Templates:

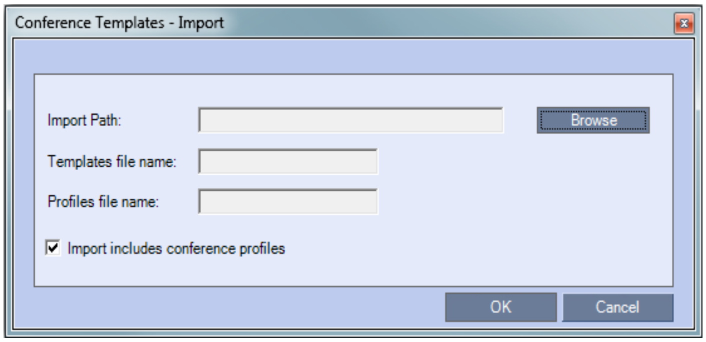
1 In the *Collaboration Server Web Client* main window, click the *Conference Templates* tab.

The *Conference Templates* are displayed.

2 Click the **Import Conference Templates** button or right-click the Conference Templates pane, and then click **Import Conference Templates.**


1111
2222
New Conference Template
Delete Conference Template
Start Conference from Template
Schedule Reservation from Template
Import Conference Templates
Export Conference Templates
Export Selected Conference Templates
Template Properties

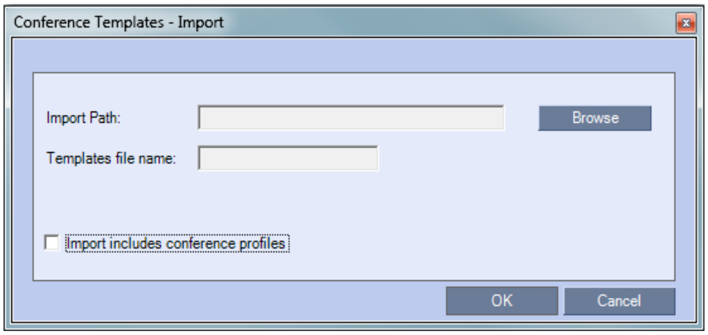
The *Conference Templates - Import* dialog box is displayed.



☑ Import includes conference profiles

3 Optional. Clear the **Import includes conference profiles** check box when you only want to import *Conference Templates*.

When this check box is cleared, the *Conference Templates - Import* dialog box is displayed without the *Profiles file name* field.



☐ Import includes conference profiles

4 In the *Import Path* field, click **Browse** to navigate to the path and file name of the *Conference Templates* you want to import.

When clicking the exported templates file you want to import, the system automatically displays the appropriate files in the *Templates file name* field and the *Profiles file name* field (when the **Import includes conference profiles** check box is selected).

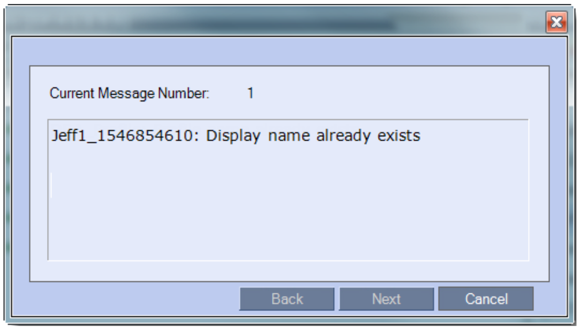
5 Click **OK** to import the *Conference Templates* and their associated *Conference Profiles,* if selected.

Conference Templates are not imported when:

— A *Conference Template* already exists

— An associated *Conference Profile* is not defined in the *Conference Profiles* list

When one or more *Conference Templates* are not imported, a Message Alert window is displayed with the templates that were not imported.



Current Message Number: 1

Jeff1_1546854610: Display name already exists

6 Click **Cancel** to exit the *Message Alerts* window.

The imported *Conference Templates* are added to the *Conference Templates* list. When the **Import includes conference profiles** check box is selected, the imported *Conference Profiles* are added to the *Conference Profiles* list.

11

Polycom Conferencing for Microsoft Outlook®

Polycom Conferencing for Microsoft Outlook is supported in AVC CP Conferencing Mode only.

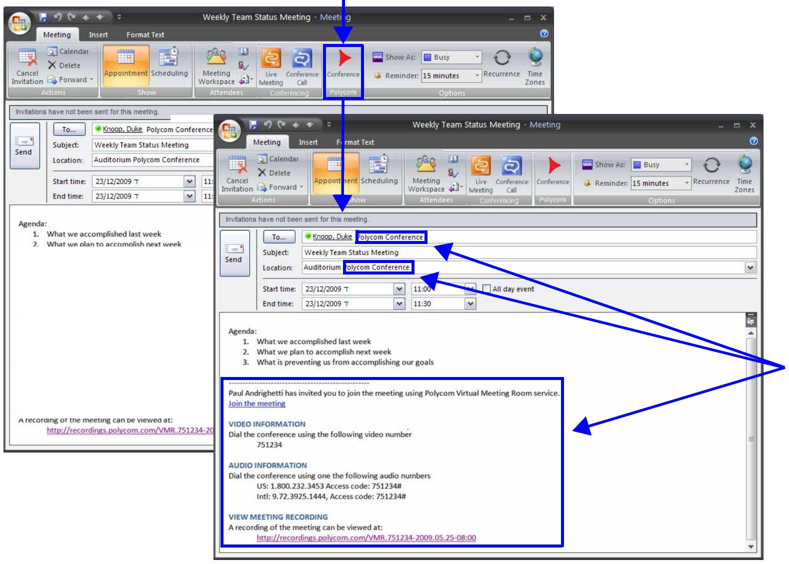
Polycom Conferencing for Microsoft Outlook is an add-in that enables users to easily organize and invite attendees to *Video Enabled* meetings via *Microsoft Outlook*®.

Polycom Conferencing for Microsoft Outlook is implemented by installing the *Polycom Conferencing Add-in for Microsoft Outlook* on *Microsoft Outlook*® e-mail clients. It enables meetings to be scheduled with video endpoints from within *Outlook*. The add-in also adds a *Polycom Conference* button in the *Meeting* tab of the *Microsoft Outlook* e-mail client ribbon.

The meeting organizer clicks the **Polycom Conference** button to add *Conference Information* to the meeting invitation.

Attendees call the meeting at the scheduled *Start Time* using the link or the dial-in number provided in the meeting invitation.

Polycom Conference Button



Invitations have not been sent for this meeting.

To...: Knoop, Duke Polycom Conference

Subject: Weekly Team Status Meeting

Location: Auditorium Polycom Conference

Start time: 23/12/2009 11:00 □ All day event

End time: 23/12/2009 11:30

Agenda:

1. What we accomplished last week
2. What we plan to accomplish next week
3. What is preventing us from accomplishing our goals

Paul Andrighetti has invited you to join the meeting using Polycom Virtual Meeting Room service.
Join the meeting

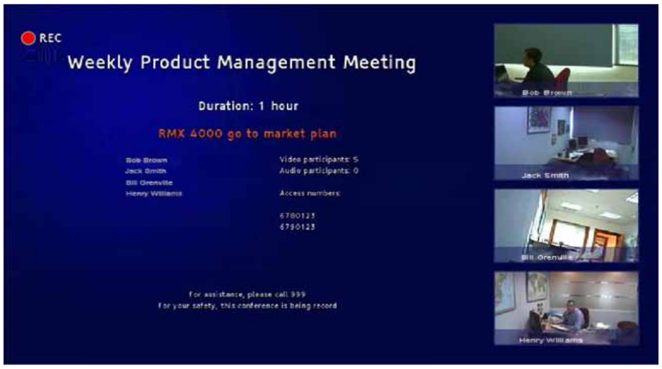
VIDEO INFORMATION
Dial the conference using the following video number
751234

AUDIO INFORMATION
Dial the conference using one the following audio numbers
US: 1.800.232.3453 Access code: 751234#
Intl: 9.72.3925.1444, Access code: 751234#

VIEW MEETING RECORDING
A recording of the meeting can be viewed at:
http://recordings.polycom.com/VMR.751234-2009.05.25-08:00

Conference Information Added

A *Gathering Slide* is displayed to connected participants until the conference starts.



● REC

Weekly Product Management Meeting

Duration: 1 hour

RMX 4000 go to market plan

Bob Brown
Jack Smith
Bill Grenville
Henry Williams

Video participants: 5
Audio participants: 0

Access numbers:

6780123
6790123

For assistance, please call 999
For your safety, this conference is being record

Gathering Slide:
Displays Meeting Information Until Conference Starts

The *Gathering Slide* displays live video along with information taken from the meeting invitation such as the subject, meeting organizer, duration, dial-in numbers etc. At the end of the *Gathering Phase*, the conference layout is displayed.

For more information see "*Video Preview (AVC Only Participants)"* on page **3-20**.

Setting up the Calendaring Solution

The following steps are performed to set up the Calendaring solution:

A The administrator installs the *Polycom Conferencing Add-in for Microsoft* for *Microsoft Outlook* e-mail clients. For more information, see the *Polycom Unified Communications Deployment Guide for Microsoft Environments*.

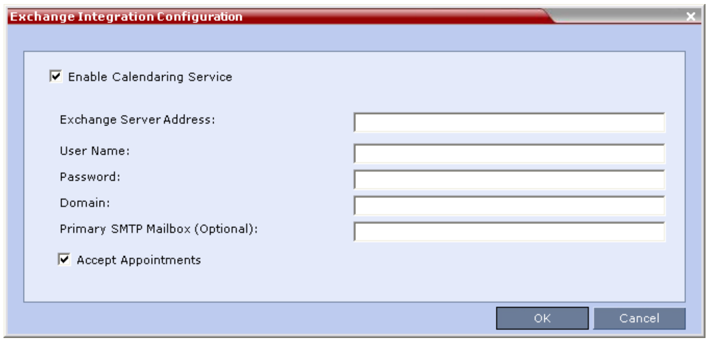
B The administrator creates an *Microsoft Outlook* e-mail-account for the Collaboration Server.
If included in the solution, *Polycom RealPresence DMA system* and calendaring-enabled endpoints share this e-mail account. For more information, see the *Polycom Unified Communications Deployment Guide for Microsoft Environments*.

C The administrator configures the Collaboration Server for *Calendaring* using the *Exchange Integration Configuration* dialog box, providing it with the *Microsoft* Exchange Server Name, User Name and Password and optional Primary SMTP Mail box information needed to access the e-mail account.

To configure the Collaboration Server's Exchange Integration Configuration:

1 On the Collaboration Server menu, click **Setup > Exchange Integration Configuration**.

The *Exchange Integration Configuration* dialog box is displayed.



Exchange Integration Configuration

☑ Enable Calendaring Service

Exchange Server Address:

User Name:

Password:

Domain:

Primary SMTP Mailbox (Optional):

☑ Accept Appointments

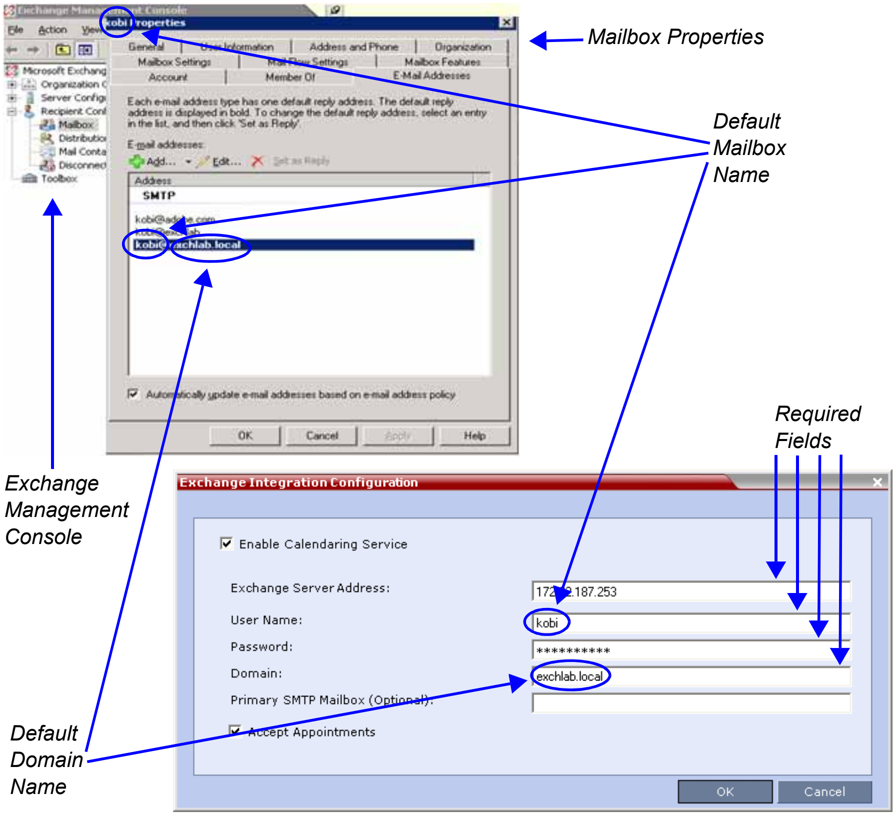
There are three options that can be used to configure the *Exchange Integration Configuration*. The option you choose will depend on the configuration of the mailbox in the *Exchange Server* and the configuration of the *Exchange Server* itself.

— **Option 1** - Use this option if the *Exchange Server* settings have been left at their default values.

— **Option 2** - Use this option if the *Primary SMTP Mailbox* is not the default mailbox.

— **Option 3** - Use this option if the *Exchange Server* settings have been modified by the administrator.

Option 1 - Using default Exchange Server settings



Each e-mail address type has one default reply address. The default reply address is displayed in bold. To change the default reply address, select an entry in the list, and then click 'Set as Reply'.

E-mail addresses:

kobi@adobe.com
kobi@exchlab
kobi@exchlab.local

☑ Automatically update e-mail addresses based on e-mail address policy

☑ Enable Calendaring Service

Exchange Server Address: 172.22.187.253
User Name: kobi
Password: **********
Domain: exchlab.local
Primary SMTP Mailbox (Optional):

☑ Accept Appointments

a Define the following fields:

Table 11-1 Exchange Integration Configuration - Option 1

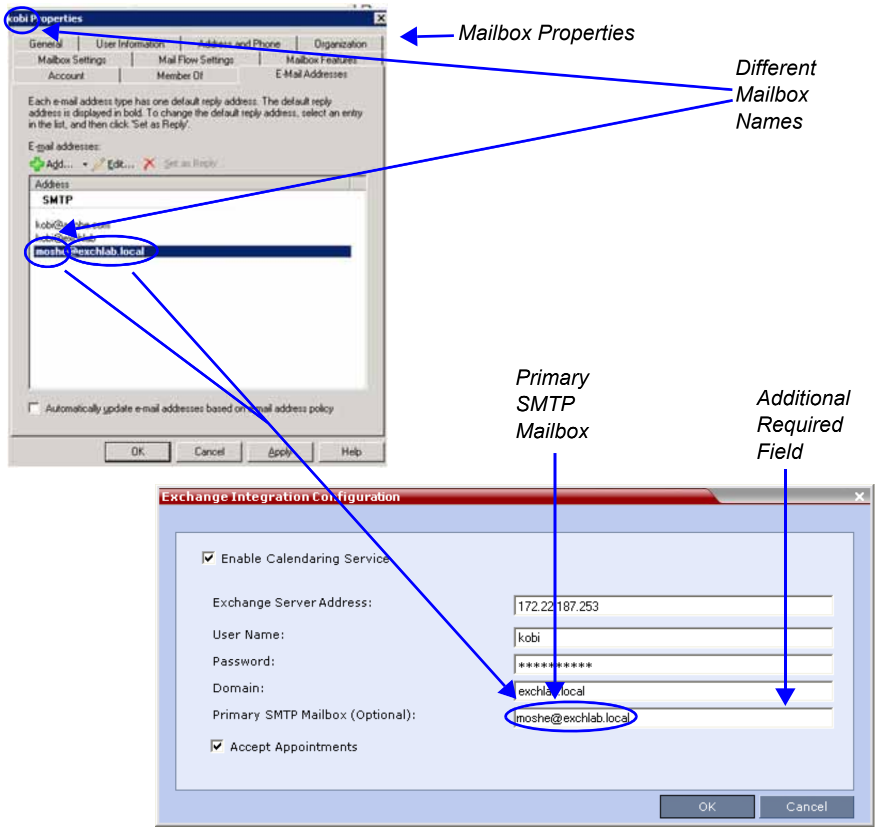
Field	Description
Enable Calendaring Service	Select or clear this check box to enable or disable the Calendaring Service using the Polycom Add-in for Microsoft Outlook. When this check box is cleared all fields in the dialog box are disabled.
Exchange Server Address	Enter the IP address of the Exchange Server.

Table 11-1 Exchange Integration Configuration - Option 1 (Continued)

Field	Description
User Name	Enter the User Name of the Collaboration Server, as registered in the Microsoft Exchange Server, that the Collaboration Server uses to login to its e-mail account. Field length: Up to 80 characters.
Password	Enter the Password the Collaboration Server uses to login to its e-mail account as registered in the Microsoft Exchange Server. Field length: Up to 80 characters.
Domain	Enter the name of the network domain where the Collaboration Server is installed as defined in the Microsoft Exchange Server.
Primary SMTP Mailbox (Optional)	This field is left empty.
Accept Appointments	Select this check box to enable the Collaboration Server to send replies to meeting invitations. Clear this check box when the Collaboration Server is part of a Unified Conferencing solution that includes a RealPresence DMA system, as the RealPresence DMA system will send a reply to the meeting invitation.

b Click the **OK** button.

Option 2 - Using an alternate Primary SMTP Mailbox



Each e-mail address type has one default reply address. The default reply address is displayed in bold. To change the default reply address, select an entry in the list, and then click 'Set as Reply'.

E-mail addresses:

SMTP

kobi@

moshe@exchlab.local

☐ Automatically update e-mail addresses based on e-mail address policy

☑ Enable Calendaring Service

Exchange Server Address: 172.22.187.253

User Name: kobi

Password: **********

Domain: exchlab.local

Primary SMTP Mailbox (Optional): moshe@exchlab.local

☑ Accept Appointments

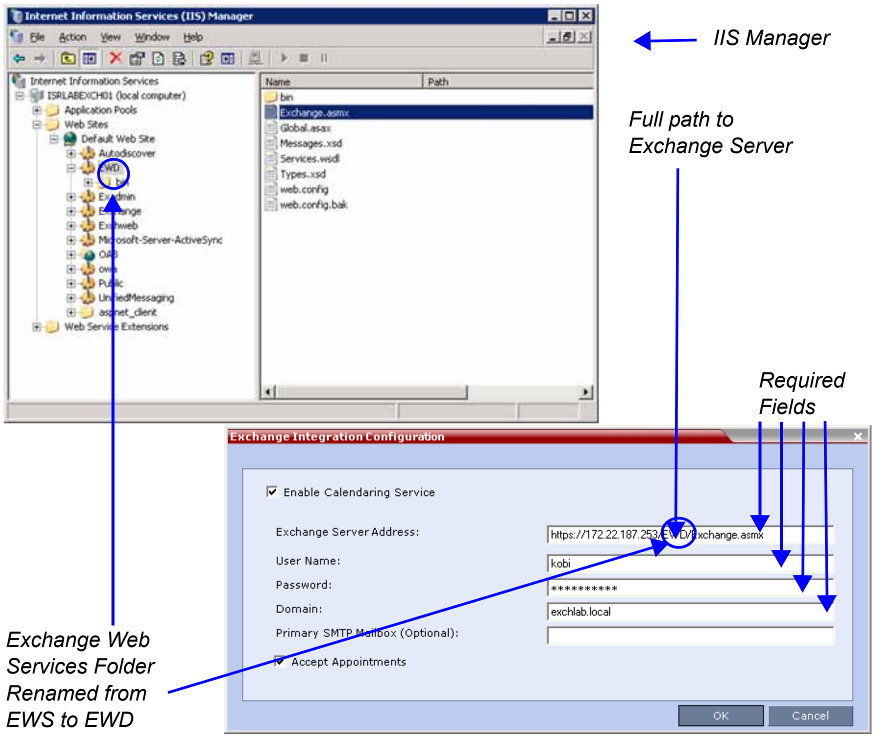
a Define the following fields:

Table 11-2 Exchange Integration Configuration - Option 2

Field	Description
Enable Calendaring Service	These fields are defined as for **Option 1** above.
Exchange Server Address	
User Name	
Password	
Domain	
Accept Appointments	
Primary SMTP Mailbox (Optional)	Enter the name of the SMTP Mailbox in the Microsoft Exchange Server to be monitored by the Collaboration Server. **Note:** Although several mailboxes can be assigned to each user in the Microsoft Exchange Server, only the Primary SMTP Mailbox is monitored. The Primary SMTP Mailbox name does not have to contain either the Collaboration Server's User Name or Domain name.

b Click the **OK** button.

Option 3 - Using modified Exchange Server settings



Exchange Integration Configuration

☑ Enable Calendaring Service

Exchange Server Address: https://172.22.187.253/EWD/Exchange.asmx

User Name: kobi

Password: **********

Domain: exchlab.local

Primary SMTP Mailbox (Optional):

☑ Accept Appointments

IIS Manager

Full path to Exchange Server

Required Fields

Exchange Web Services Folder Renamed from EWS to EWD

a Define the following fields:

Table 11-3 *Exchange Integration Configuration - Option 3*

Field	Description
Exchange Server Address	If Exchange Server settings have been modified, enter the full path to the Microsoft Exchange Server where the Collaboration Server’s Microsoft Outlook e-mail account is registered, for example if the EWS folder has been renamed *EWD*: `https://labexch01/`**`EWD`**`/Exchange.asmx` **Note:** If a server name is entered, the Collaboration Server and the Microsoft Exchange Server must be registered to the same Domain. (The Domain name entered in this dialog box must match the Local Domain Name entry in the Management Network - DNS Properties dialog box.) For more information see "*Modifying the Default IP Network Service”* on page **16-6**. Field length: Up to 80 characters.

Table 11-3 Exchange Integration Configuration - Option 3 (Continued)

Field	Description
Enable Calendaring Service	These fields are defined as for **Option 1** above.
User Name	
Password	
Domain	
Primary SMTP Mailbox (Optional)	
Accept Appointments	

b Click the **OK** button.

If applicable, *RSS, VMC, RealPresence DMA* system, and calendaring-enabled endpoints are configured with the *Exchange Server Name, User Names* and *Passwords* needed to access their accounts.

For more information see the *Polycom Unified Communications Deployment Guide for Microsoft Environments.*

2 The administrator configures the Collaboration Server to have a default *Ad-hoc Entry Queue* service enabled.
For more information see "*Defining a New Entry Queue"* on page **7-3**.

Calendaring Guidelines

- The Collaboration Server must have its *MCU* prefix registered in the gatekeeper.
  For more information see "*Modifying the Default IP Network Service"* on page **16-6**.
- The Collaboration Server must be configured as a *Static Route*.
  For more information see "*Modifying the Default IP Network Service"* on page **16-6**.
- The Collaboration Server's *Default Entry Queue* must be configured as an *Ad Hoc Entry Queue* and must be designated as the *Transit Entry Queue*.
  For more information see the "*Entry Queues"* on page **7-1**.
- The meeting organizer can enable recording and/or streaming of the meeting.
- If meeting is to be recorded, the *Ad Hoc Entry Queue* must have recording enabled in its *Profile*.
  For more information see "*Defining New Profiles"* on page **2-20**.
- Meetings can be single instance or have multiple occurrences.
- Attendees that do not have video devices may be invited to the meeting.
- Attendees using e-mail applications that use the *iCalendar* format may be invited to meetings via the *Calendaring Service*.
- Meeting invitations sent by *Polycom Conferencing for Microsoft Outlook* can be in a different language to the *Collaboration Server Web Client.* The following languages are supported:
  - — English
  - — French
  - — German

  - International Spanish
  - Korean
  - Japanese
  - Simplified Chinese
- Collaboration Server resource management is the responsibility of the system administrator:
  - Conferences initiated by Polycom Conferencing for Microsoft Outlook are ad hoc and therefore resources are not reserved in advance.
  - Polycom Conferencing for Microsoft Outlook Add-in assumes that sufficient resources are available and does not check resource availability. Sufficient resources are therefore not guaranteed.
  - A meeting invitation that is automatically accepted by the Collaboration Server is not guaranteed availability of resources.
  - If the Collaboration Server runs out of resources, attendees will not be able to connect to their conferences.
- By using RealPresence DMA system to load-balance resources between several Collaboration Servers, resource capacity can be increased, alleviating resource availability problems.

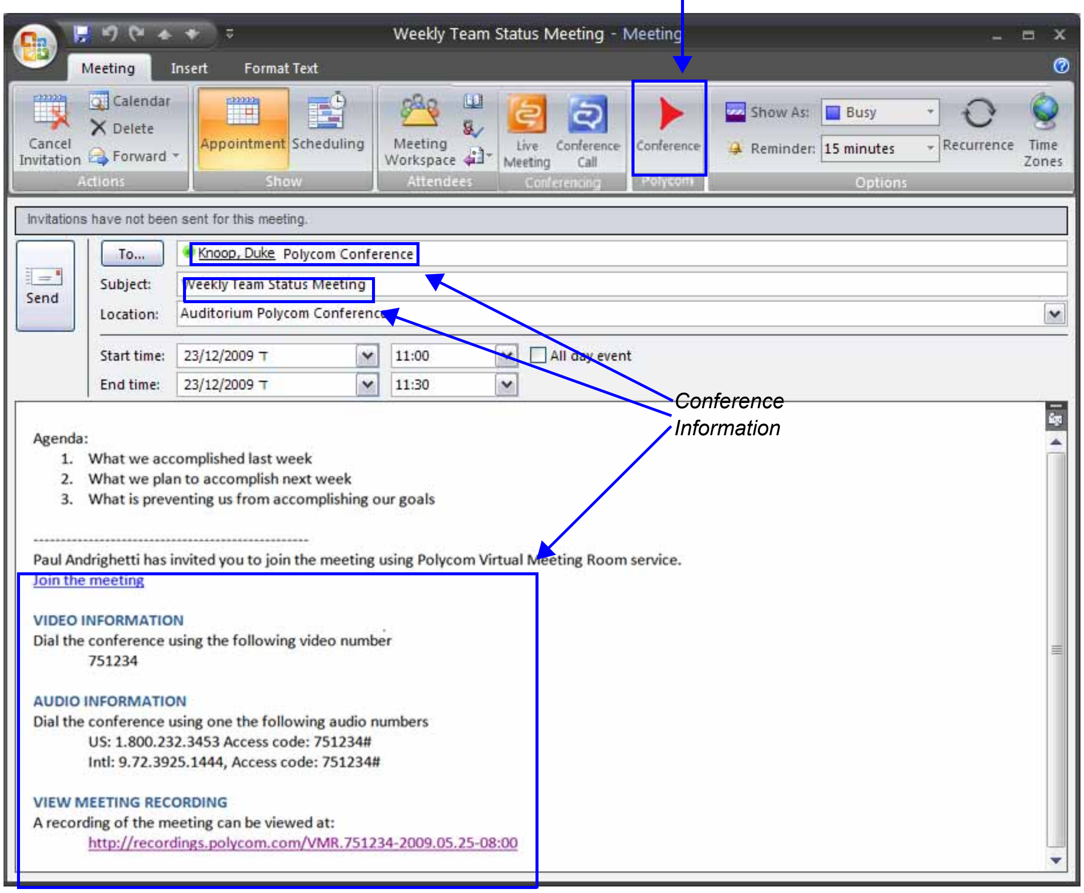
Creating and Connecting to a Conference

Creating a Conference

Meetings are organized using the *Microsoft Outlook* client in the normal manner.

If the meeting organizer decides that video participants are to be included in a multipoint video conference, he/she clicks the **Polycom Conference** button. *Conference Information* such as the *Meeting ID* and connection information is automatically added to the existing appointment information.

Polycom Conference Button



Invitations have not been sent for this meeting.

To... Knoop, Duke Polycom Conference

Subject: Weekly Team Status Meeting

Location: Auditorium Polycom Conference

Start time: 23/12/2009 Т 11:00 □ All day event

End time: 23/12/2009 Т 11:30

Agenda:

1. What we accomplished last week
2. What we plan to accomplish next week
3. What is preventing us from accomplishing our goals

Paul Andrighetti has invited you to join the meeting using Polycom Virtual Meeting Room service.

Join the meeting

VIDEO INFORMATION

Dial the conference using the following video number

751234

AUDIO INFORMATION

Dial the conference using one the following audio numbers

US: 1.800.232.3453 Access code: 751234#

Intl: 9.72.3925.1444, Access code: 751234#

VIEW MEETING RECORDING

A recording of the meeting can be viewed at:

http://recordings.polycom.com/VMR.751234-2009.05.25-08:00

Conference Information

The meeting organizer can add a meeting agenda or personal text to the invitation before it is sent. The meeting organizer can update or cancel the video enabled meeting in the same manner as for any other meeting.

When the meeting organizer sends the meeting invitation a meeting record is saved in the *Microsoft Exchange Server*, the RealPresence Collaboration Server, *RealPresence DMA* system, *RSS* and calendaring-enabled endpoints.

RealPresence Collaboration Servers, *RealPresence DMA* system, and calendaring-enabled endpoints poll the *Microsoft Exchange Server* to retrieve new meeting records and updates to existing meeting records.

Table 11-4 summarizes the Collaboration Server's usage of *Microsoft Outlook* data fields included in the meeting invitation.

Table 11-4 *Microsoft Outlook Field Usage*

Microsoft Outlook Field	**Usage by the Collaboration Server / RealPresence DMAsystem**	
	Conference / Meeting Room	**Gathering Slide**
Subject	Display Name of Conference / Meeting Room.	Meeting Name.
Start/End Time	Used to calculate the Conference's Duration.	
Record	Enable Recording in the Conference or Meeting Room Profile.	Display Recording option.
Video Access Number	Comprised of: `<MCU Prefix in Gatekeeper>` `<Conference Numeric ID>`. **Note:** It is important that *MCU Prefix in Gatekeeper* field in the Collaboration Server's *IP Network Service - Gatekeeper* tab and the *Dial-in prefix* field in the *Polycom Conferencing Add-in for Microsoft Outlook - Video Network* tab contain the same prefix information.	Displayed as the IP dial in number in the Access Number section of the Gathering Slide.
Video Access Number (Cont.)	If Recording and Streaming are enabled in the Conference Profile, this number is used as part of the recording file name.	
Streaming recording link	Enables the recording of the conference to the Polycom RSS using the recording link. Enables streaming of the recording of the conference from the Polycom RSS.	If recording is enabled, a REC indicator is displayed in the top left corner of the slide.

Connecting to a Conference

Participants can connect to the conference in the following ways:

- Participants with *Polycom CMA Desktop™* or a *Microsoft Office Communicator* client running on their PCs can click on a link in the meeting invitation to connect to the meeting.
- Participants with a *HDX* or a room system will receive a prompt from the endpoint's calendaring system along with a button that can be clicked in order to connect. Participants with endpoints that are not calendaring-enabled can connect to the meeting by dialing the meeting number manually.

Collaboration Server Standalone Deployment

When using a single Collaboration Server in a standalone deployment, connection is via an *Ad Hoc Entry Queue.* The meeting is started when the first participant connects to the Collaboration Server.

When the first participant connects, a conference is created and named according to the information contained in the dial string. Subsequent participants connecting with the same dial string are routed from the *Ad Hoc Entry Queue* to the conference.

After the conference has been created the *Conference Name, Organizer, Time, Duration* and *Password* (if enabled) are retrieved from the conference parameters for display during the *Gathering Phase*.

Collaboration Server and Polycom RealPresence DMA System Deployment

In a RealPresence *DMA* system deployment a *Virtual Meeting Room* is activated when the first participant connects to the *RealPresence DMA* system. The *RealPresence DMA* system receives the dial string to activate a *Virtual Meeting Room* on the Collaboration Server.

The *RealPresence DMA* system uses the *Meeting ID* contained in the dial-in string to access meeting information stored in the *Exchange Server* database.

When the meeting information is found on the *Exchange Server*, the *Conference Name, Organizer, Time, Duration* and *Password* (if enabled) are retrieved from the *Exchange Server* database for display during the *Gathering Phase*.

If enabled, automatically generated passwords are ignored.
For more information see "*Automatic Password Generation Flags"* on page **20-37**.

Polycom Solution Support

Polycom Implementation and Maintenance services provide support for Polycom solution components only. Additional services for supported third-party Unified Communications (UC) environments integrated with Polycom solutions are available from Polycom Global Services and its certified Partners. These additional services will help customers successfully design, deploy, optimize and manage Polycom visual communications within their UC environments.

Professional Services for Microsoft Integration is mandatory for Polycom Conferencing for Microsoft Outlook and Microsoft Office Communications Server integrations. For additional information and details please see http://www.polycom.com/services/professional_services/index.html or contact your local Polycom representative.

12

Conference and Participant Monitoring

Viewing Permissions			
Tab	Chairperson	Operator	Administrator
Skins	✓	✓	✓
IVR		✓	✓
Info	✓	✓	✓

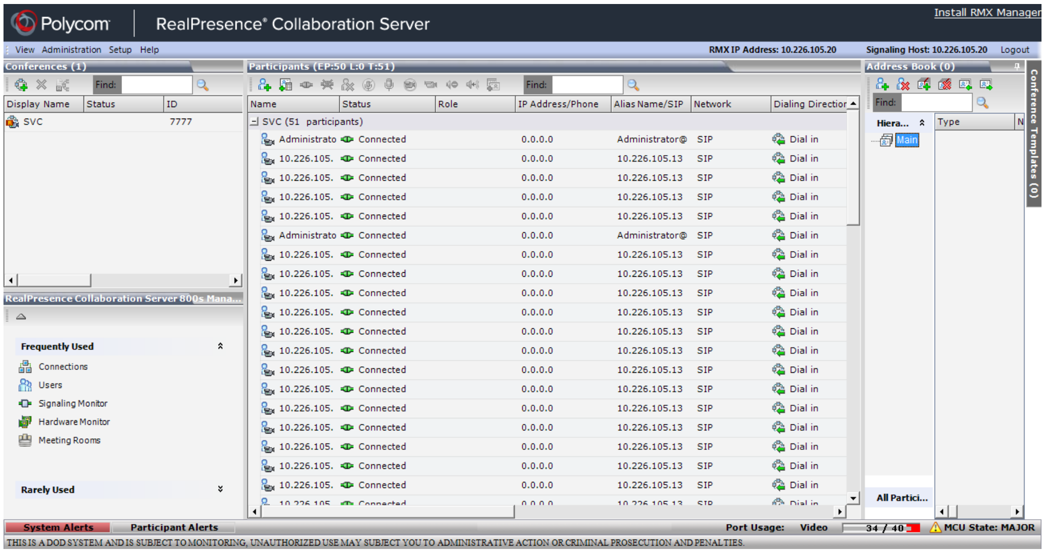
You can monitor ongoing conferences and perform various operations while conferences are running.

Three levels of monitoring are available with the Collaboration Server:

- *General Monitoring* - You can monitor the general status of all ongoing conferences and their participants in the main window.
- *Conference Level Monitoring* - You can view additional information regarding a specific conference and modify its parameters if required, using the *Conference Properties* option.
- *Participant Level Monitoring* - You can view detailed information on the participant's status, using the *Participant Properties* option.
- The maximum number of participants in a conference:
  - In SVC Only conference: up to 60 SVC-based participants
  - In CP Only conference: up to 20 AVC-based HD participants or 40 AVC-based SD participants
  - In a Mixed CP and SVC conference: any combination of SVC and AVC-based participants depending on the number of AVC and SVC participants.

General Monitoring

Users can monitor a conference or keep track of its particiRealPresence Collaboration Serverpants and progress. For more information, see *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Monitoring Ongoing Conferences"* on page **3-43**.



Display Name	Status	ID
SVC		7777

SVC (51 participants)

Name	Status	Role	IP Address/Phone	Alias Name/SIP	Network	Dialing Direction
Administrato	Connected		0.0.0.0	Administrator@	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
Administrato	Connected		0.0.0.0	Administrator@	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in

THIS IS A DOD SYSTEM AND IS SUBJECT TO MONITORING, UNAUTHORIZED USE MAY SUBJECT YOU TO ADMINISTRATIVE ACTION OR CRIMINAL PROSECUTION AND PENALTIES.

You can click the blinking **Participant Alerts** indication bar to view participants that require attention. For more information, see "*System and Participant Alerts"* on page **19-1**.

Conference Level Monitoring

In addition to the general conference information that is displayed in the *Conference* list pane, you can view the details of the conference's current status and setup parameters, using the *Conference Properties* dialog box.

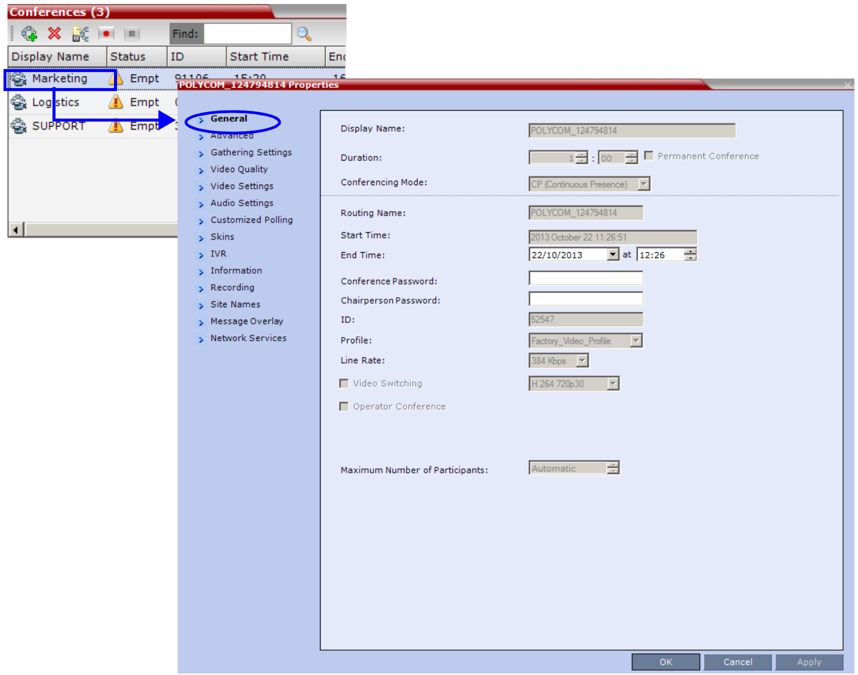
Viewing the Properties of an Ongoing AVC CP and Mixed CP and SVC Conference

To view the parameters of an ongoing AVC CP conference:

1 In the *Conference* list pane, double-click the **AVC CP** conference or right-click the **AVC CP** conference and then click **Conference Properties**.

Viewing Permissions			
Tab	Chairperson	Operator	Administrator
General	✔	✔	✔

The *Conference Properties - General* dialog box with the **General** tab opens.



Display Name: POLYCOM_124794814

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: CP (Continuous Presence)

Routing Name: POLYCOM_124794814

Start Time: 2013 October 22 11:26:51

End Time: 22/10/2013 at 12:26

Conference Password:

Chairperson Password:

ID: 52547

Profile: Factory_Video_Profile

Line Rate: 384 Kbps

□ Video Switching H.264 720p30

□ Operator Conference

Maximum Number of Participants: Automatic

The following information is displayed in the *General* tab:

Table 12-1 *Conference Properties - General*

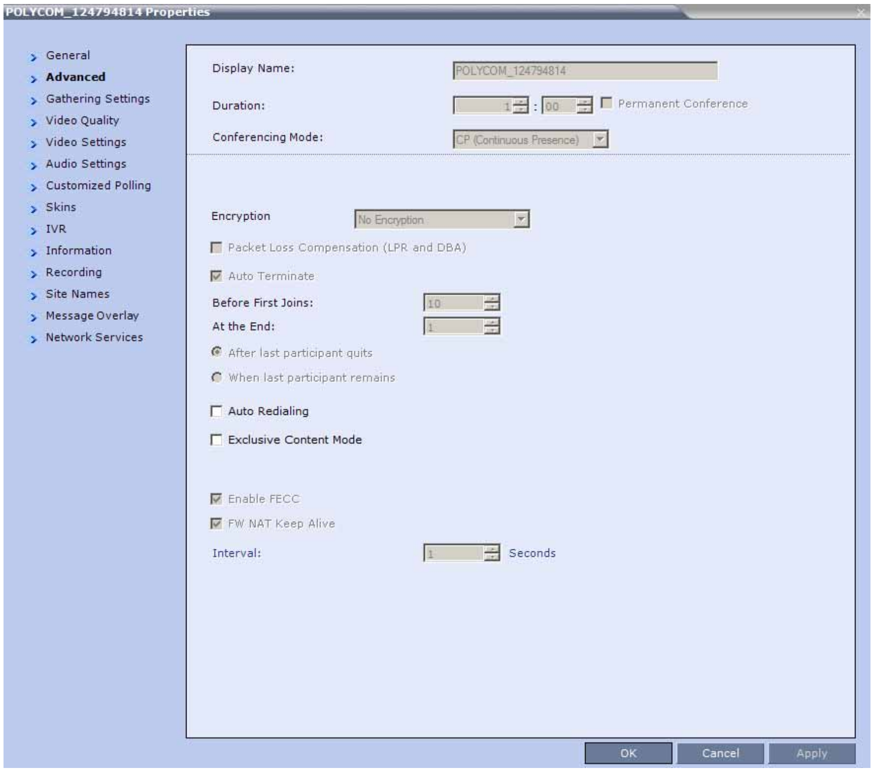
Field	Description
Display Name	The Display Name is the conference name in native language and Unicode character sets to be displayed in the *Collaboration Server Web Client*. **Note:** This field is displayed in all tabs.
Duration	The expected duration of the conference using the format HH:MM. **Note:** This field is displayed in all tabs.
Permanent Conference	Indicates whether the conference is set as a Permanent Conference, with no pre-determined End Time. This conference continues until it is terminated by an administrator, operator or chairperson. **Note:** This field is displayed in all tabs.
Routing Name	The ASCII name of the conference. It can be used by H.323 and SIP participants for dialing in directly to the conference. It is used to register the conference in the gatekeeper and the SIP server.
Conferencing Mode	The conferencing mode set for the conference: CP, VSW, SVC only or CP and SVC.
Start Time	The time the conference started.
End Time	The expected conference end time. **Note:** This field is not shown when the conference is set as a *Permanent Conference*.

Table 12-1 *Conference Properties - General (Continued)*

Field	Description
Conference Password	A numeric password for participants to access the conference.
Chairperson Password	A numeric password used by participants to identify themselves as the conference chairperson.
ID	The conference ID.
Profile	The name of the conference Profile from which conference parameters were taken.
Line Rate	The maximum transfer rate, in kilobytes per second (Kbps) of the call (video and audio streams).
Max Number of Participants	Indicates the total number of participants that can be connected to the conference. The Automatic setting indicates the maximum number of participants that can be connected to the MCU according to resource availability.

2 Click the **Advanced** tab.

The *Conference Properties - Advanced* dialog box opens.



Display Name: POLYCOM_124794814

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: CP (Continuous Presence)

Encryption: No Encryption

□ Packet Loss Compensation (LPR and DBA)

☑ Auto Terminate

Before First Joins: 10

At the End: 1

● After last participant quits

○ When last participant remains

□ Auto Redialing

□ Exclusive Content Mode

☑ Enable FECC

☑ FW NAT Keep Alive

Interval: 1 Seconds

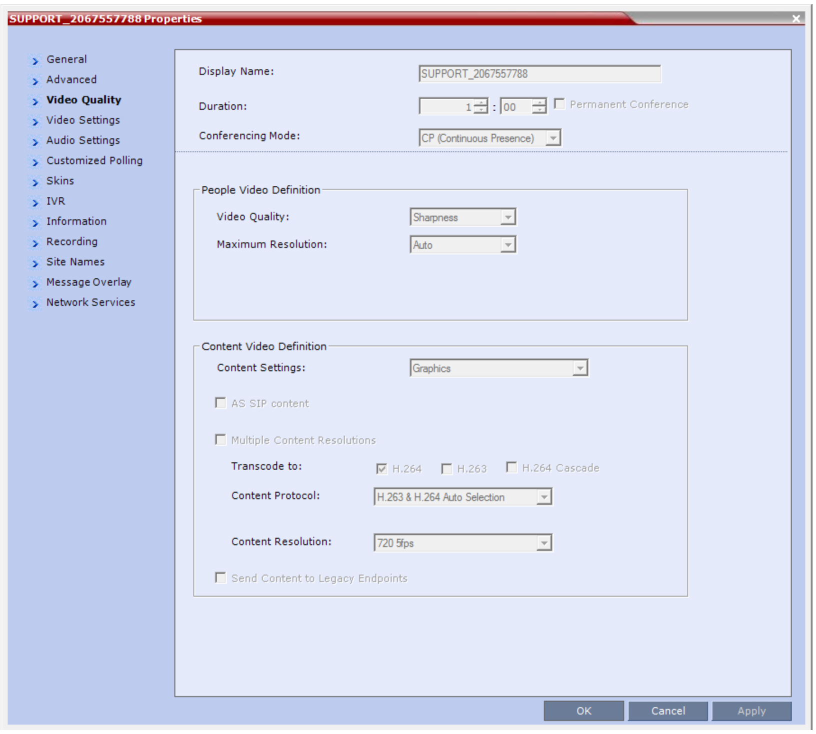
3 The following information is displayed in the *Advanced* tab:

Table 12-2 Conference Properties - Advanced Parameters

Field/Option	Description
Encryption	Indicates whether the conference is encrypted.
Packet Loss Compensation (LPR and DBA)	Indicates wether Packet Loss Compensation (LPR and DBA) is enabled for the conference.
Auto Terminate	When selected, indicates that the MCU will automatically terminate the conference when *Before First Joins*, *At the End-After Last Quits and At the End - When Last Participant Remains* parameters apply.
Auto Redialing	Indicates whether dial-out participants are automatically (when selected) or manually (when cleared) connected to the conference. This option is disabled in mixed CP and SVC conferences.
Exclusive Content Mode	When selected, *Content* is limited to one participant.
Enable FECC	When selected, Far End Camera Control is enabled.
FW NAT Keep Alive	When selected, sends a *FW NAT Keep Alive* message at specific Intervals for the RTP, UDP and BFCP channels. The interval specifies how often a *FW NAT Keep Alive* message is sent. For more information, see

4 Click the **Video Quality** tab.

The *Conference Properties - Video Quality* dialog box opens.



SUPPORT_2067557788 Properties

Display Name: SUPPORT_2067557788

Duration: 1 : 00 ☐ Permanent Conference

Conferencing Mode: CP (Continuous Presence)

People Video Definition

Video Quality: Sharpness

Maximum Resolution: Auto

Content Video Definition

Content Settings: Graphics

☐ AS SIP content

☐ Multiple Content Resolutions

Transcode to: ☑ H.264 ☐ H.263 ☐ H.264 Cascade

Content Protocol: H.263 & H.264 Auto Selection

Content Resolution: 720 5fps

☐ Send Content to Legacy Endpoints

The following information is displayed:

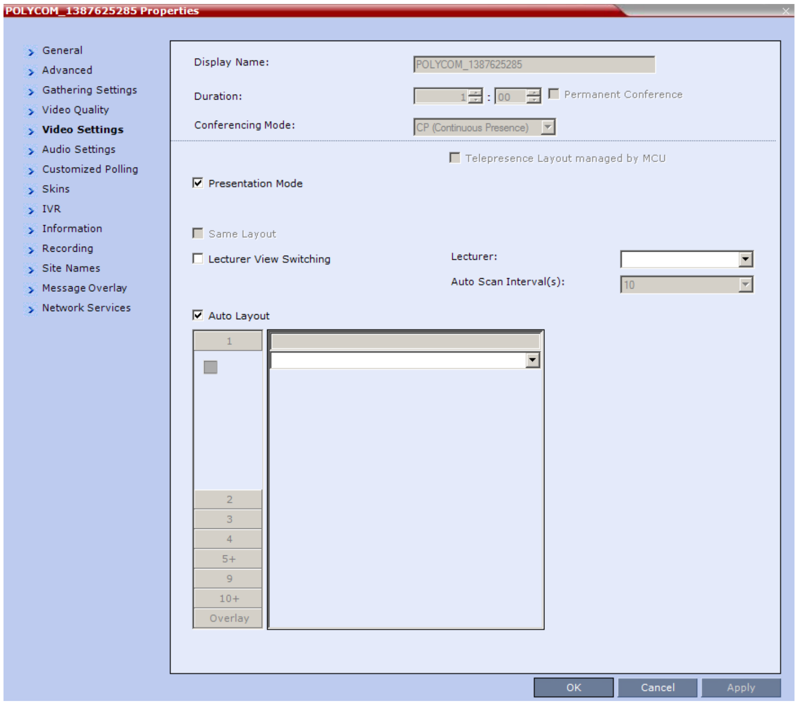
Table 12-3 Conference Properties - Video Quality Parameters

Field/Option	Description
People Video Definition	
Video Quality	Indicates the resolution and frame rate that determine the video quality set for the conference. This is always **Sharpness**. For more information, see "*Video Resolutions in AVC-based CP Conferencing"* on page **4-1**.
Maximum Resolution	Indicates the *Maximum Resolution* setting for the conference. - *Auto* (default) - indicates that the *Maximum Resolution* is as selected in the *Resolution Configuration* dialog box. The *Maximum Resolution* settings for conferences and participants cannot be changed during an ongoing conference.
Content Video Definition	
AS-SIP	This option is not supported with RealPresence Collaboration Server Virtual Edition.
Multiple Content Resolutions	This option is not supported with RealPresence Collaboration Server Virtual Edition.
Content Settings	Indicates the Content channel resolution set for the conference. Possible resolutions are: - **Graphics –** default mode - **Hi-res Graphics –** requiring a higher bit rate - **Live Video –** content channel is live video - **Customized Content Rate** - resolution is manually defined.
Content Protocol	Indicates the Content Protocol used for content sharing in Highest Common Content Sharing Mode. For more information, see "*Content Protocols"* on page **3-8**.
Content Resolution	Indicates the Content Resolution and frame rate according to the selected Content Sharing Mode (Highest common Content or Multiple Resolution Contents) and the video protocol. For more information, see "*Defining Content Sharing Parameters for a Conference"* on page **3-3**.
Send Content to Legacy Endpoints (CP only)	This Content Sharing option is not supported with RealPresence Collaboration Server Virtual Edition.

5 Click the **Video Settings** tab to list the video parameters.

Viewing Permissions

Tab	*Chairperson*	*Operator*	*Administrator*
Video Settings	✔	✔	✔



Display Name: POLYCOM_1387625285

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: CP (Continuous Presence)

□ Telepresence Layout managed by MCU

☑ Presentation Mode

□ Same Layout

□ Lecturer View Switching

Lecturer:

Auto Scan Interval(s): 10

☑ Auto Layout

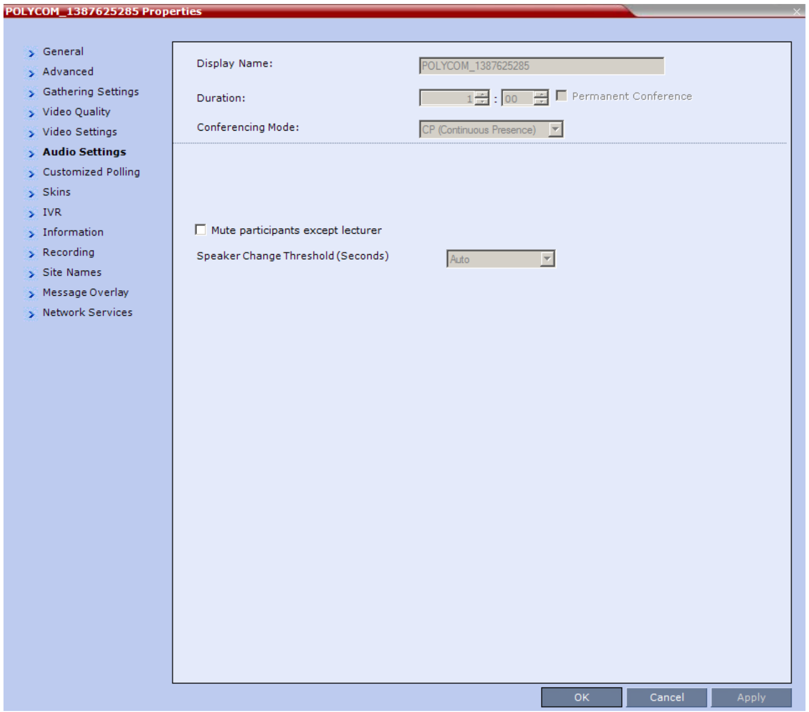
Table 12-4 *Conference Properties - Video Settings Parameters*

Field	Description
Presentation Mode	When checked, indicates that the Presentations Mode is active. This option is disabled in a mixed CP and SVC conference. For more information, see .
Lecturer View Switching	When checked, the *Lecturer View Switching* enables automatic random switching between the conference participants in the lecturer video window. This option is disabled in a mixed CP and SVC conference.
Same Layout	When checked, forces the selected layout on all conference participants, and the Personal Layout option is disabled. This option is disabled in a mixed CP and SVC conference.
Auto Layout	When enabled, the system automatically selects the conference layout based on the number of participants in the conference.
Lecturer	Indicates the name of the lecturer (if one is selected). Selecting a lecturer enables the Lecture Mode. This option is disabled in a mixed CP and SVC conference.
Auto Scan Interval(s)	The time interval, 10 - 300 seconds, that Auto Scan uses to cycle the display of participants that are not in the conference layout in the selected cell. This option is disabled in a mixed CP and SVC conference.

***Table 12-4** Conference Properties - Video Settings Parameters (Continued)*

Field	Description
Video Layouts (graphic)	Indicates the currently selected video layout.

6 Click the A**udio Settings** tab to view the audio setting for the conference.



Display Name: POLYCOM_1387625285

Duration: 1 : 00 □ Permanent Conference

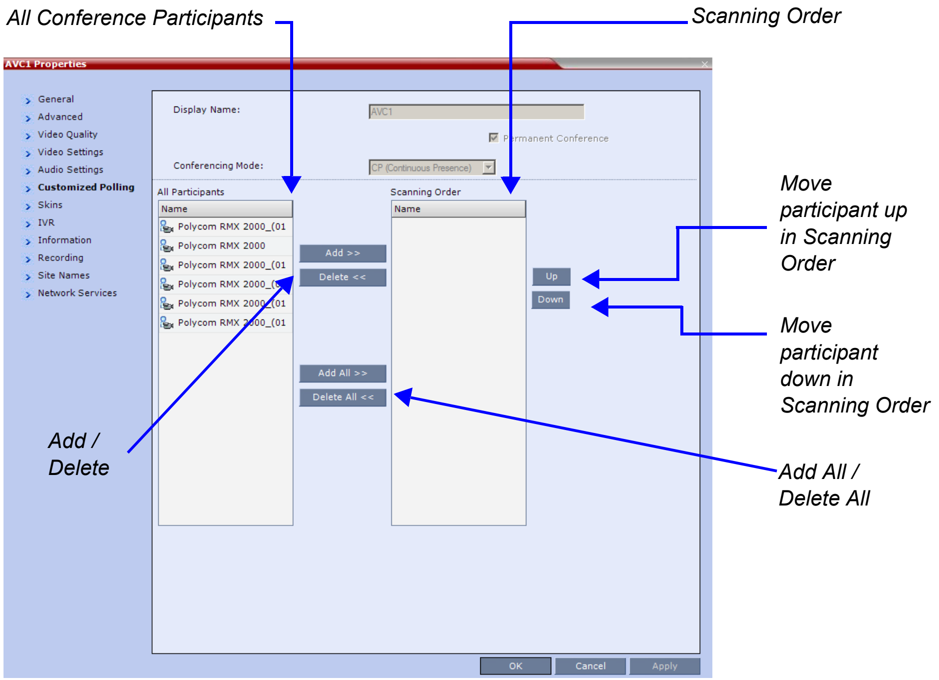
Conferencing Mode: CP (Continuous Presence)

□ Mute participants except lecturer

Speaker Change Threshold (Seconds) Auto

7 If needed, you can enable or disable the *Mute participants except lecturer* setting.

8 **CP Only Conferences:** Click the **Customized Polling** tab to view and modify the customized polling for the conference.



Display Name: AVC1

☑ Permanent Conference

Conferencing Mode: CP (Continuous Presence)

All Participants

Name
Polycom RMX 2000_(01
Polycom RMX 2000
Polycom RMX 2000_(01
Polycom RMX 2000_(0
Polycom RMX 2000_(01
Polycom RMX 2000_(01

Scanning Order

Name

All conference participants are listed in the left pane (*All Participants*) while the participants that are to be displayed in the Auto Scan enabled cell of the video layout are listed in the right pane (*Scanning Order*).

The dialog box buttons are summarized in Table 12-5.

Table 12-5 *Customized Polling - Buttons*

Button	Description
Add	Select a participant and click this button to *Add* a the participant to the list of participants to be Auto Scanned. The participants name is removed from the *All Participants* pane.
Delete	Select a participant and click this button to *Delete* the participant from the list of participants to be *Auto Scanned*. The participants name is moved back to the *All Participants* pane.
Add All	Add all participants to the list of participants to be *Auto Scanned*. All participants' names are removed from the *All Participants* pane.
Delete All	Delete all participant from the list of participants to be *Auto Scanned*. All participants' names are moved back to the *All Participants* pane.
Up	Select a participant and click this button to move the participant *Up* in the *Scanning Order*.
Down	Select a participant and click this button to move the participant *Down* in the *Scanning Order*.

9 **Optional.** Add a participant to the list of participants to be *Auto Scanned*:

a Click on the participant's name in the *All Participants* list.

b Click the **Add** button to move the participant to the *Scanning Order* pane.

10 **Optional.** Delete a participant from the list of participants to be *Auto Scanned*:

a Click on a participant's name in the *Scanning Order* list.

b Click the **Delete** button to move the participant back to the *All Participants* pane.

11 **Optional.** Add all participants to the list of participants to be *Auto Scanned*:

— Click the **Add All** button.

12 **Optional.** Delete all participant from the list of participants to be *Auto Scanned*:

— Click the **Delete All** button.

13 **Optional.** Move the participant up in the *Scanning Order*:

— Click the **Up** button.

14 **Optional.** Move the participant down in the *Scanning Order*:

— Click the **Down** button.

15 Click the **Apply** button to confirm and keep the *Conference Properties* dialog box open.

or

Click the **OK** the button to confirm and return to the *Collaboration Server Web Client Main Screen*.

16 Click the **Skins** tab to view the skin selected for the conference.

You cannot select another skin during an ongoing conference.

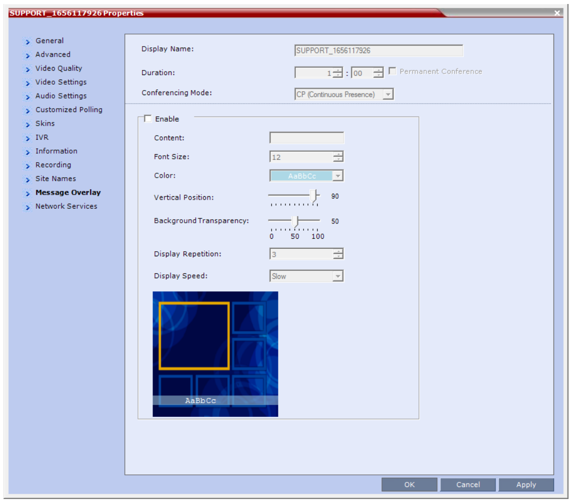
17 Click the **IVR** tab to view the IVR settings.

18 Click the **Information** tab to view general information defined for the conference. Changes made to this information once the conference is running are not saved to the CDR.

19 Click the **Recording** tab to review the recording settings for the conference.

20 Click the **Site Names** tab to enable or disable the display of site names during the conference, and adjust the display properties.

21 Click the **Message Overlay** tab to send text messages to the conference participants during the conference, and adjust the display properties of the text messages.



SUPPORT_1656117926 Properties

Display Name: SUPPORT_1656117926

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: CP (Continuous Presence)

□ Enable

Content:

Font Size: 12

Color: AaBbCc

Vertical Position: 90

Background Transparency: 50

Display Repetition: 3

Display Speed: Slow

For more information, see "*Message Overlay for Text Messaging*" on page 2-58.

22 Click the **Network Services** tab to verify the SIP registration for the conference.

23 Click **OK** to close the *Conference Properties* dialog box.

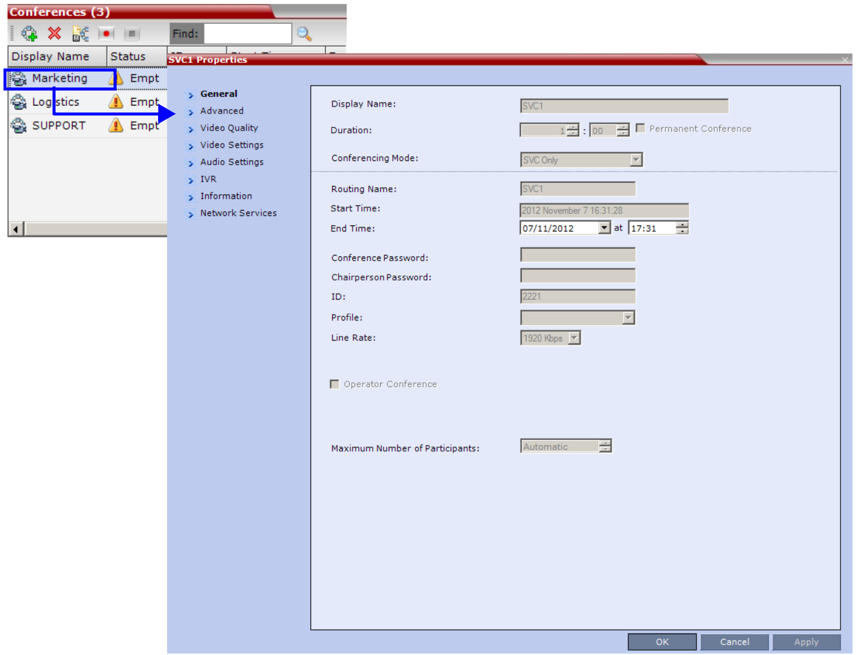
Viewing the Properties of an Ongoing SVC-based Conference

Viewing Permissions			
Tab	*Chairperson*	*Operator*	*Administrator*
General	✓	✓	✓

To view the parameters of an ongoing SVC conference:

1 In the *Conference* list pane, double-click the SVC conference or right-click the SVC conference and then click **Conference Properties**.

The *Conference Properties - General* dialog box with the **General** tab opens.



Display Name: SVC1
Duration: 1 : 00 □ Permanent Conference
Conferencing Mode: SVC Only
Routing Name: SVC1
Start Time: 2012 November 7 16:31:28
End Time: 07/11/2012 at 17:31
Conference Password:
Chairperson Password:
ID: 2221
Profile:
Line Rate: 1920 Kbps
□ Operator Conference
Maximum Number of Participants: Automatic

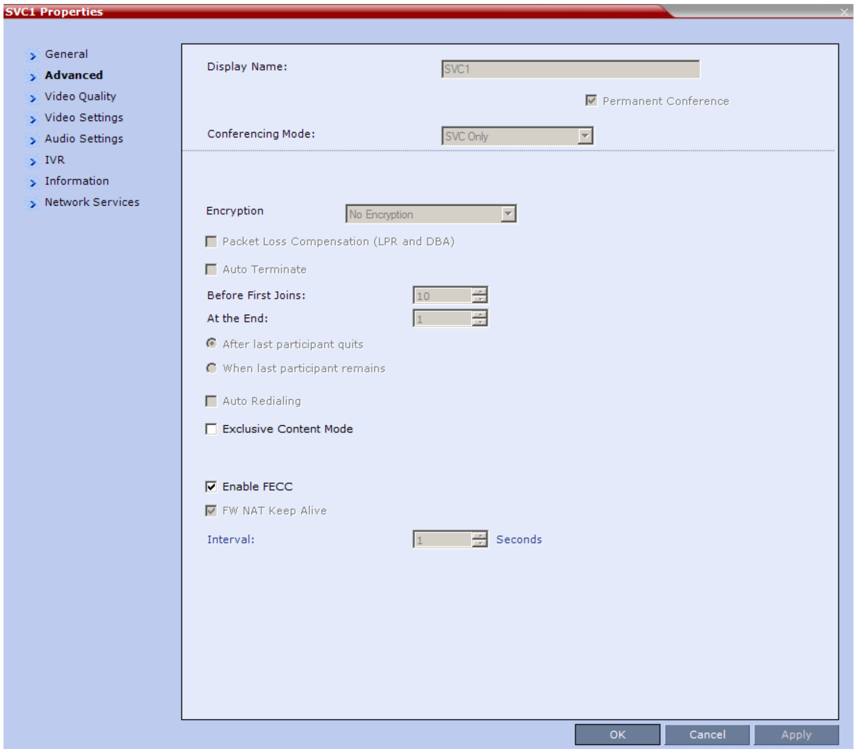
2 The following information is displayed in the *General* tab:

Field	Description
Display Name	The Display Name is the conference name in native language and Unicode character sets to be displayed in the *Collaboration Server Web Client*. **Note:** This field is displayed in all tabs.
Duration	The expected duration of the conference using the format HH:MM. **Note:** This field is displayed in all tabs.
Conferencing Mode	The conferencing mode for the conference.
Routing Name	The ASCII name of the conference. It can be used by H.323 and SIP participants for dialing in directly to the conference. It is used to register the conference in the gatekeeper and the SIP server.
Start Time	The time the conference started.
End Time	The expected conference end time.

Field	Description
Conference Password	A numeric password for participants to access the conference.A numeric password for participants to access the conference.
Chairperson Password	A numeric password used by participants to identify themselves as the conference chairperson.A numeric password used by participants to identify themselves as the conference chairperson.
ID	The conference ID.
Profile	The name of the conference Profile from which conference parameters were taken.
Line Rate	The maximum transfer rate, in kilobytes per second (Kbps) of the call (video and audio streams).
Max Number of Participants	Indicates the total number of participants that can be connected to the conference. The Automatic setting indicates the maximum number of participants that can be connected to the MCU according to resource availability.

3 Click the **Advanced** tab.

The *Conference Properties - Advanced* dialog box opens.



Display Name: SVC1

☑ Permanent Conference

Conferencing Mode: SVC Only

Encryption: No Encryption

□ Packet Loss Compensation (LPR and DBA)

□ Auto Terminate

Before First Joins: 10

At the End: 1

● After last participant quits

○ When last participant remains

□ Auto Redialing

□ Exclusive Content Mode

☑ Enable FECC

☑ FW NAT Keep Alive

Interval: 1 Seconds

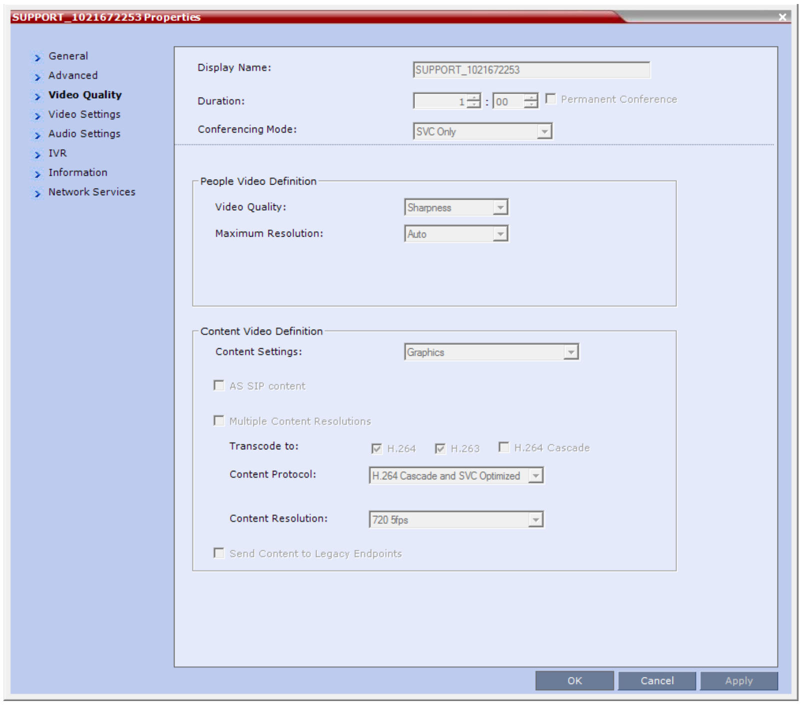
4 The following information is displayed in the *Advanced* tab:

Table 12-6 *Conference Properties - Advanced Parameters*

Field/Option	Description
Auto Terminate	When selected, indicates that the MCU will automatically terminate the conference when *Before First Joins*, *At the End-After Last Quits and At the End - When Last Participant Remains* parameters apply.
Auto Redialing	Dial-out is not supported in SVC conferences.
Exclusive Content Mode	When selected, *Content* is limited to one participant.
Enable FECC	Far End Camera Control is not supported in SVC conferences.
FW NAT Keep Alive	When selected, sends a *FW NAT Keep Alive* message at specific Intervals for the RTP, UDP and BFCP channels. The interval specifies how often a *FW NAT Keep Alive* message is sent. For more information, see "*RealPresence Collaboration Server Virtual Edition Network Port Usage"* on page **15-28**.

5 Click the **Video Quality** tab.

The *Conference Properties - Video Quality* dialog box opens.



Display Name: SUPPORT_1021672253

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: SVC Only

People Video Definition

Video Quality: Sharpness

Maximum Resolution: Auto

Content Video Definition

Content Settings: Graphics

□ AS SIP content

□ Multiple Content Resolutions

Transcode to: ☑ H.264 ☑ H.263 □ H.264 Cascade

Content Protocol: H.264 Cascade and SVC Optimized

Content Resolution: 720 5fps

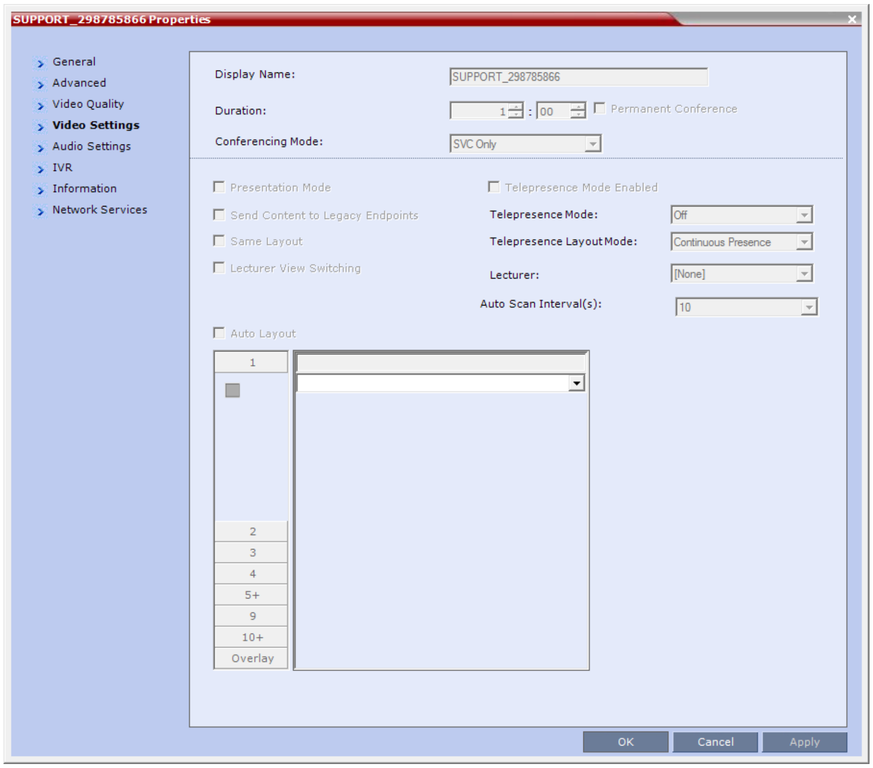
□ Send Content to Legacy Endpoints

The following information is displayed:

Table 12-7 Conference Properties - Video Quality Parameters

Field/Option	Description
People Video Definition	
Video Quality	Indicates the resolution and frame rate that determine the video quality set for the conference. Only Sharpness is supported.
Maximum Resolution	In *SVC conferencing*, this is always *Auto* (default) - The *Maximum Resolution* remains as selected in the *Resolution Configuration* dialog box.
Content Video Definition	
AS-SIP	*AS-SIP* This option is not supported with RealPresence Collaboration Server Virtual Edition.is not supported in SVC conferences.
Multiple Content Resolutions	*Multiple Content Resolutions* This option is not supported with RealPresence Collaboration Server Virtual Edition.This option is not supported with RealPresence Collaboration Server Virtual Edition.is not supported in SVC conferences.
Content Settings	In *SVC conferencing*, this is always set to **Graphics**
Content Protocol	In *SVC conferencing* this is always set to **H.264 Cascade and SVC Optimized**.
Content Resolution	Resolution is fixed in SVC conferences.

6 Click the **Video Settings** tab to view the video parameters defined for the conference.


SUPPORT_298785866 Properties

Display Name: SUPPORT_298785866

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: SVC Only

□ Presentation Mode

□ Send Content to Legacy Endpoints

□ Same Layout

□ Lecturer View Switching

□ Telepresence Mode Enabled

Telepresence Mode: Off

Telepresence Layout Mode: Continuous Presence

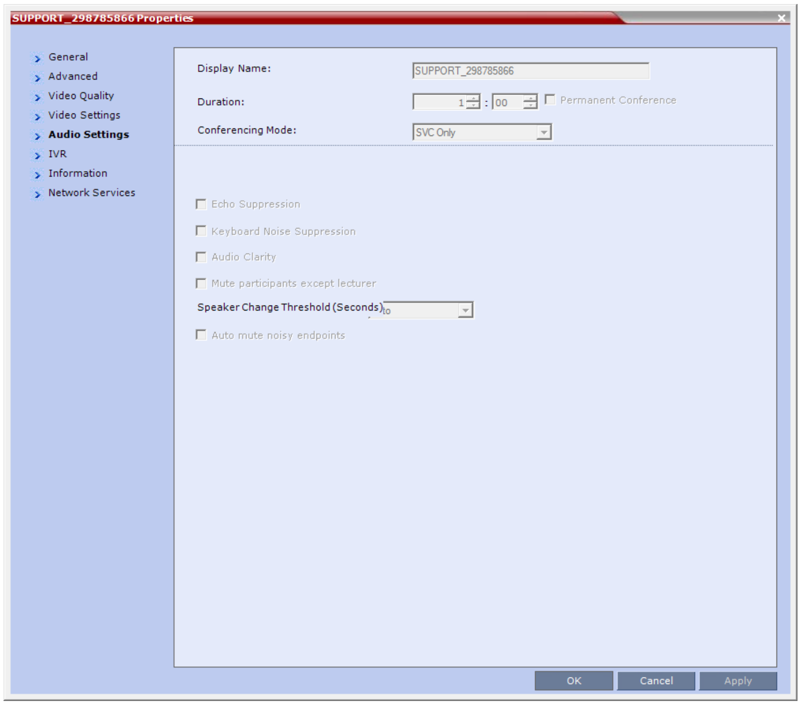
Lecturer: [None]

Auto Scan Interval(s): 10

□ Auto Layout

In SVC conferences, only Auto Layout is enabled and cannot be disabled. All other video settings are disabled.

7 Click the **Audio Settings** tab to view the audio parameters defined for the conference.


SUPPORT_298785866 Properties

Display Name: SUPPORT_298785866

Duration: 1 : 00 □ Permanent Conference

Conferencing Mode: SVC Only

□ Echo Suppression

□ Keyboard Noise Suppression

□ Audio Clarity

□ Mute participants except lecturer

Speaker Change Threshold (Seconds) to

□ Auto mute noisy endpoints

In SVC conferences, all Audio Settings options are disabled.

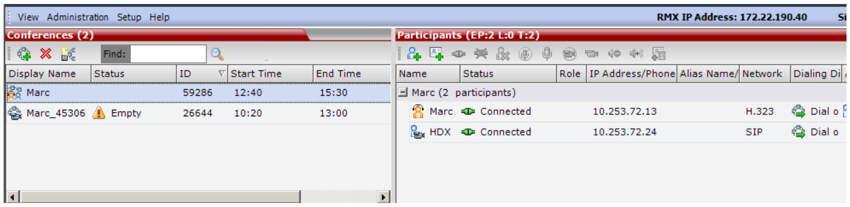
8 Click the **Information** tab to view general information defined for the conference. Changes made to this information once the conference is running are not saved to the CDR.

9 Click **OK** to close the *Conference Properties* dialog box.

Monitoring Operator Conferences and Participants Requiring Assistance (CP and Mixed CP and SVC Conferences)

Operator conferences are monitored in the same way as standard ongoing conferences.

Each Operator conference includes at least one participant - the Operator.


View Administration Setup Help
RMX IP Address: 172.22.190.40
Conferences (2)
Find:
Participants (EP:2 L:0 T:2)

Display Name	Status	ID	Start Time	End Time
Marc		59286	12:40	15:30
Marc_45306	Empty	26644	10:20	13:00

Name	Status	Role	IP Address/Phone	Alias Name/	Network	Dialing Di
Marc (2 participants)						
Marc	Connected		10.253.72.13		H.323	Dial o
HDX	Connected		10.253.72.24		SIP	Dial o

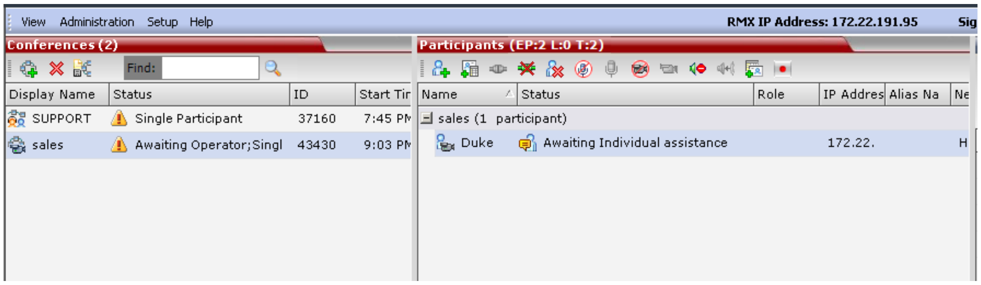
You can view the properties of the *Operator conference* by double-clicking the conference entry in the *Conferences* list or by right-clicking the conference entry and selecting **Conference Properties**. For more information, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Conference Level Monitoring*" on page 3-45.

Requesting Help

A participant can request help using the appropriate DTMF code from his/her touch tone telephone or the endpoint's DTMF input device. The participant can request *Individual Assistance* (default DTMF code ***0**) or *Conference Assistance* (default DTMF code **00**).

Participants in Entry Queues who failed to enter the correct destination conference ID or the conference password will wait for operator assistance (provided that an Operator conference is active).

When requiring or requesting operator assistance, the Collaboration Server management application displays the following:


View Administration Setup Help
RMX IP Address: 172.22.191.95
Sig
Conferences (2)
Find:
Participants (EP:2 L:0 T:2)

Display Name	Status	ID	Start Tir
SUPPORT	Single Participant	37160	7:45 PM
sales	Awaiting Operator;Singl	43430	9:03 PM

Name	Status	Role	IP Addres	Alias Na	Ne
sales (1 participant)					
Duke	Awaiting Individual assistance		172.22.		H

- The participant's connection *Status* changes, reflecting the help request. For details, see Table 12-8.
- The conference status changes and it is displayed with the exclamation point icon and the status "Awaiting Operator".
- The appropriate voice message is played to the relevant participants indicating that assistance will be provided shortly.

The following icons and statuses are displayed in the *Participant Status* column:

Table 12-8 *Participants List Status Column Icons and Indications*

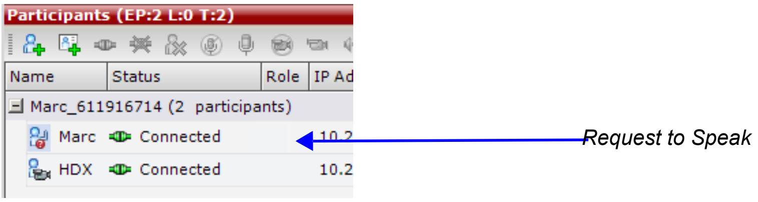
Icon	Status indication	Description
	Awaiting Individual Assistance	The participant has requested the operator's assistance for himself/herself.
	Awaiting Conference Assistance	The participant has requested the operator's assistance for the conference. Usually this means that the operator is requested to join the conference.

When the Operator moves the participant to the *Operator conference* for individual assistance the participant Status indications are cleared.

Request to Speak

Participants that were muted by the conference organizer/system operator can indicate that they want to be unmuted by entering the appropriate DTMF code.

An icon is displayed in the *Role* column of the *Participants* list for 30 seconds.



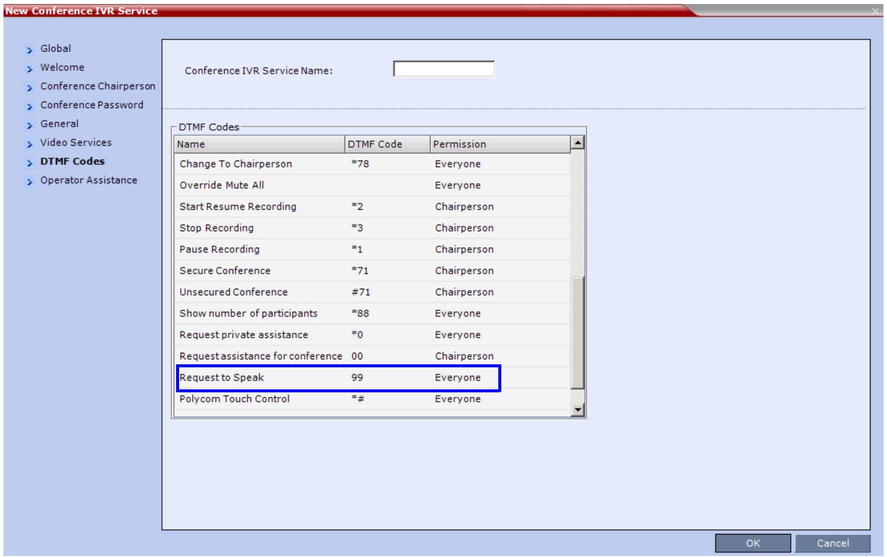
Participants (EP:2 L:0 T:2)

Name	Status	Role	IP Ad
Marc_611916714 (2 participants)			
Marc	Connected		10.2
HDX	Connected		10.2

Request to Speak is:

- Activated when the participant enters the appropriate DTMF code (default: **99)**.

  The DTMF code can be modified in the conference *IVR Service Properties - DTMF Codes* dialog box.



Conference IVR Service Name:

DTMF Codes

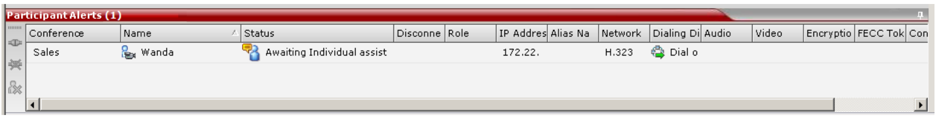
Name	DTMF Code	Permission
Change To Chairperson	*78	Everyone
Override Mute All		Everyone
Start Resume Recording	*2	Chairperson
Stop Recording	*3	Chairperson
Pause Recording	*1	Chairperson
Secure Conference	*71	Chairperson
Unsecured Conference	#71	Chairperson
Show number of participants	*88	Everyone
Request private assistance	*0	Everyone
Request assistance for conference	00	Chairperson
Request to Speak	99	Everyone
Polycom Touch Control	*#	Everyone

- Available for dial-in and dial-out participants.
- A participant can request to speak more than once during the conference.

- Supported in *all* conference types.
- Supported in H.323 and SIP environments.
- The duration of the icon display cannot be modified.

Participant Alerts List

The *Participant Alerts* list contains all the participants who are currently waiting for operator assistance.



Participant Alerts (1)

Conference	Name	Status	Disconne	Role	IP Addres	Alias Na	Network	Dialing Di	Audio	Video	Encryptio	FECC Tok	Con
Sales	Wanda	Awaiting Individual assist			172.22.		H.323	Dial o					

Participants are automatically added to the *Participants Alerts* list in the following circumstances:

- The participant fails to connect to the conference by entering the wrong conference ID or conference password and waits for the operator's assistance.
- The participant requests Operator's Assistance during the ongoing conference.

This list is used as reference only. Participants can be assisted and moved to the *Operator conference* or the destination conference only from the *Participants* list of the Entry Queues or ongoing conference where they are awaiting assistance.

The participants are automatically removed from the *Participant Alerts* list when moved to any conference (including the *Operator conference*).

Participant Level Monitoring

In addition to conference information, you can view detailed information regarding the status and parameters of each listed participant, using the *Participant Properties* dialog box. Participant properties can be displayed for all participants currently connected to a conference and for defined participants that have been disconnected.

SIP SVC-based participant properties are similar to SIP AVC-based participant properties.

Displaying Participants Properties

To display the participant Properties:

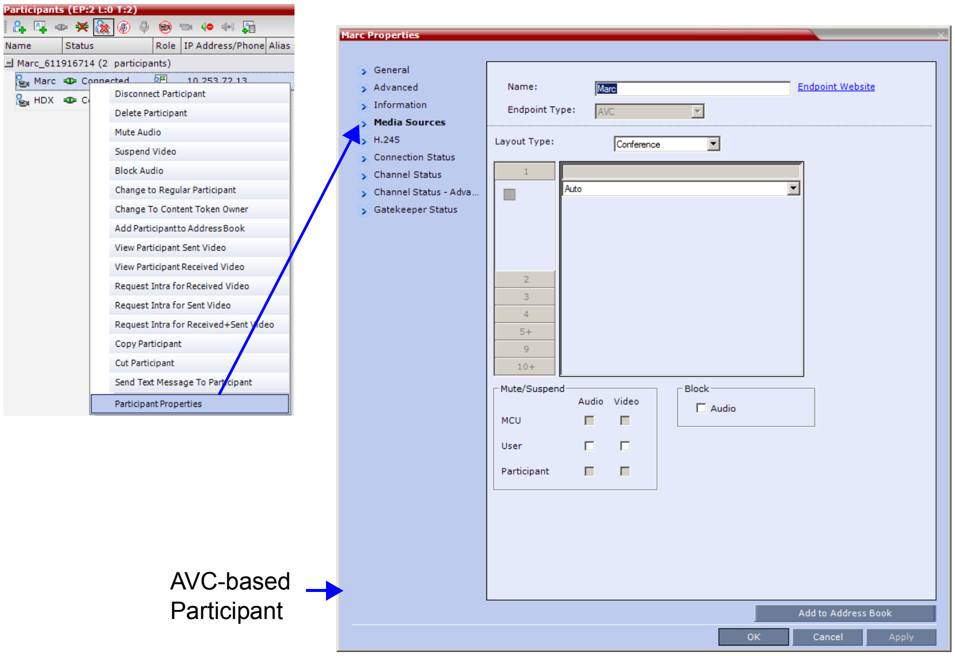
1 In the *Participant List* pane double-click the participant entry. Alternatively, right-click a participant and then click **Participant Properties**.

The *Participant Properties - Media Sources* dialog box opens.

Media Sources properties are not available for SVC participants.

Viewing Permissions

Tab	*Chairperson*	*Operator*	*Administrator*
Media Sources	✓	✓	✓



Name: Marc

Endpoint Type: AVC

Layout Type: Conference

Auto

The *Media Sources* dialog box enables you to mute participant's audio, suspend participant's video transmission and select a personal Video Layout for the participant.

IP Participant Properties

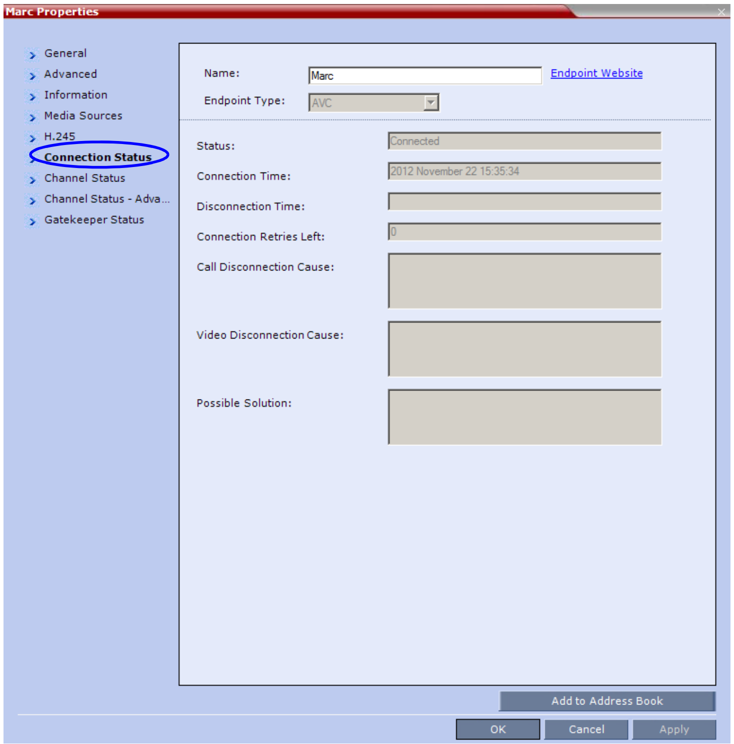
The following parameters are displayed for an IP participant

***Table 12-9** Participant Properties - Media Sources Parameters*

Field	Description
Name	Indicates the participant's name. **Note:** This field is displayed in all tabs.
Endpoint Website (link)	Click the Endpoint Website hyperlink to connect to the internal website of the participant's endpoint. It enables you to perform administrative, configuration and troubleshooting activities on the endpoint. The connection is available only if the IP address of the endpoint's internal site is filled in the *Website IP Address* field in the *Participant Properties - General* dialog box. **Note:** This field is displayed in all tabs**.**
Endpoint Type	Indicates whether the participant is using an AVC-based or SVC-based endpoint. Fields, tabs and options are enabled or disabled according to the endpoint type. **Note:** This field is displayed in all tabs.
Layout Type	Indicates whether the video layout currently viewed by the participant is the Conference or Personal Layout. If *Personal Layout* is selected, you can select a Video Layout that will be viewed only by this participant.
Video Layout	Indicates the video layout currently viewed by the participant. When *Personal Layout* is selected in the *Layout Type* you can force participants to the video windows in a layout that is specific to the participant. For more information*, see Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide,* "*Changing the Video Layout of a Conference (AVC-Based CP and Mixed CP and SVC Conferences)"* on page **3-59**.
Mute/Suspend	Indicates if the endpoint's audio and/or video channels have been muted/ suspended. The entity that initiated audio mute or video suspend is also indicated. • **MCU** – Audio or Video channel has been muted/suspended by the MCU. • **User** – Channels have been muted/suspended by the Collaboration Server user. • **Participant** – Channels have been muted/suspended by the participant from the endpoint. You can also cancel or perform mute and suspend operation using these check boxes. **Note:** If the participant muted his/her audio channel, the system displays the mute icon only for H.323. This icon is not displayed for SIP participant due to SIP standard limitation.
Block	When checked, the audio transmission from the conference to the participant's endpoint is blocked, but the participant will still be heard by other participants.

2 Click the **Connection Status** tab to view the connection status, and if disconnected the cause of the disconnection.

Viewing Permissions			
Tab	*Chairperson*	*Operator*	*Administrator*
Connection Status	✓	✓	✓



Name: Marc

Endpoint Type: AVC

Status: Connected

Connection Time: 2012 November 22 15:35:34

Disconnection Time:

Connection Retries Left: 0

Call Disconnection Cause:

Video Disconnection Cause:

Possible Solution:

This dialog box is the same for AVC-based and SVC-based participants.

The following parameters are displayed:

Table 12-10 Participant Properties - Connection Status Parameters

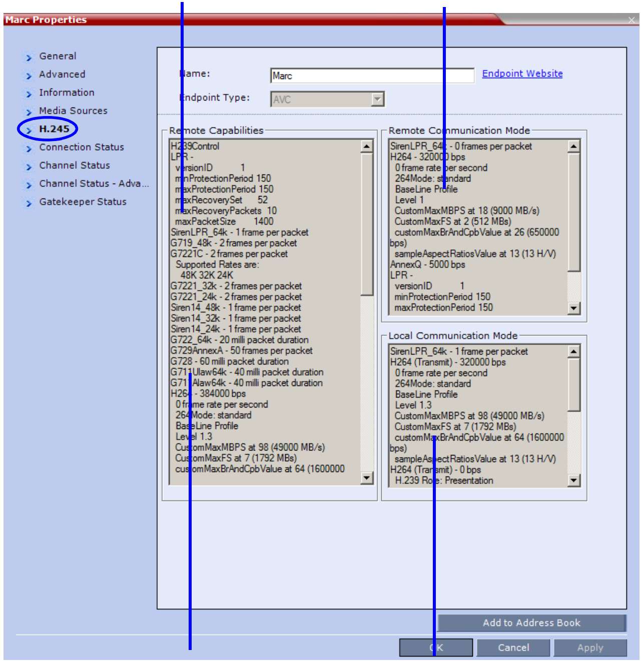
Field	Description
Participant Status	
Status	Indicates the connection status of the participant.
Connection Time	The date and time the participant connected to the conference. **Note:** The time format is derived from the MCU's operating system time format.
Disconnection Time	The date and time the defined participant disconnected from the conference.
Connection Retries Left	Indicates the number of retries left for the system to connect defined participant to the conference.
Call Disconnection Cause	Displays the cause for the defined participant's disconnection from the conference. See *Appendix A: "Disconnection Causes"* on page **A-1**.
Video Disconnection Cause	Displays the cause the video channel could not be connected. For more information, see *Appendix A: "Disconnection Causes"* on page **A-1**.

Table 12-10 *Participant Properties - Connection Status Parameters (Continued)*

Field	Description
Possible Solution	In some cases, a possible solution is indicated to the cause of the video disconnection.

3 Click the **H.245** (H.323) or **SDP** (SIP) tab during or after the participant's connection process to view information that can help in resolving connection issues.

H.323 Participant (AVC-based|)



Name: Marc

Endpoint Type: AVC

Remote Capabilities

```
H239Control
LPR -
  versionID         1
  minProtectionPeriod 150
  maxProtectionPeriod 150
  maxRecoverySet      52
  maxRecoveryPackets  10
  maxPacketSize       1400
SirenLPR_64k - 1 frame per packet
G719_48k - 2 frames per packet
G7221C - 2 frames per packet
  Supported Rates are:
   48K 32K 24K
G7221_32k - 2 frames per packet
G7221_24k - 2 frames per packet
Siren14_48k - 1 frame per packet
Siren14_32k - 1 frame per packet
Siren14_24k - 1 frame per packet
G722_64k - 20 milli packet duration
G729AnnexA - 50 frames per packet
G728 - 60 milli packet duration
G711Ulaw64k - 40 milli packet duration
G711Alaw64k - 40 milli packet duration
H264 - 384000 bps
  0 frame rate per second
  264Mode: standard
  BaseLine Profile
  Level 1.3
  CustomMaxMBPS at 98 (49000 MB/s)
  CustomMaxFS at 7 (1792 MBs)
  customMaxBrAndCpbValue at 64 (1600000
```

Remote Communication Mode

```
SirenLPR_64k - 0 frames per packet
H264 - 320000 bps
  0 frame rate per second
  264Mode: standard
  BaseLine Profile
  Level 1
  CustomMaxMBPS at 18 (9000 MB/s)
  CustomMaxFS at 2 (512 MBs)
  customMaxBrAndCpbValue at 26 (650000
bps)
  sampleAspectRatiosValue at 13 (13 H/V)
AnnexQ - 5000 bps
LPR -
  versionID         1
  minProtectionPeriod 150
  maxProtectionPeriod 150
```

Local Communication Mode

```
SirenLPR_64k - 1 frame per packet
H264 (Transmit) - 320000 bps
  0 frame rate per second
  264Mode: standard
  BaseLine Profile
  Level 1.3
  CustomMaxMBPS at 98 (49000 MB/s)
  CustomMaxFS at 7 (1792 MBs)
  customMaxBrAndCpbValue at 64 (1600000
bps)
  sampleAspectRatiosValue at 13 (13 H/V)
H264 (Transmit) - 0 bps
  H.239 Role: Presentation
```

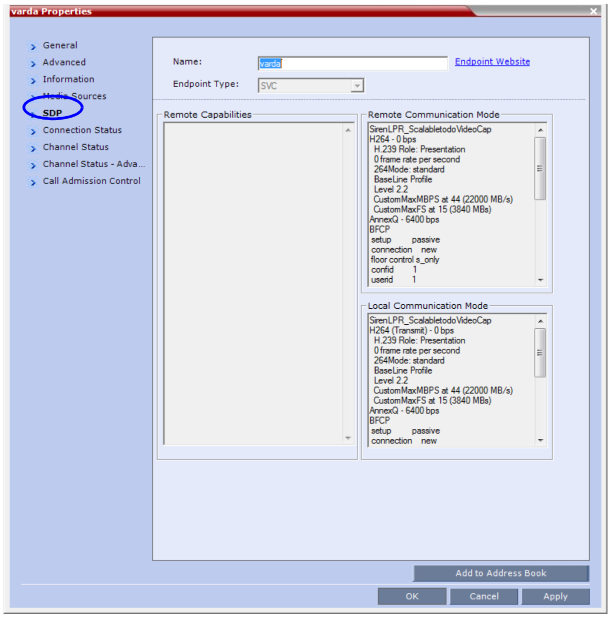
LPR activity (Displayed in all three panes)

Displays the endpoint's actual capabilities used for the connection

List's the endpoint's capabilities as retrieved from the remote site

Displays the MCU's capabilities used for connection with the participant

SIP Participant (AVC-based and SVC-based)



Name: varda

Endpoint Type: SVC

Remote Communication Mode

```
SirenLPR_ScalabletodoVideoCap
H264 - 0 bps
  H.239 Role: Presentation
  0 frame rate per second
  264Mode: standard
  BaseLine Profile
  Level 2.2
  CustomMaxMBPS at 44 (22000 MB/s)
  CustomMaxFS at 15 (3840 MBs)
AnnexQ - 6400 bps
BFCP
 setup        passive
 connection   new
 floor control s_only
 confid       1
 userid       1
```

Local Communication Mode

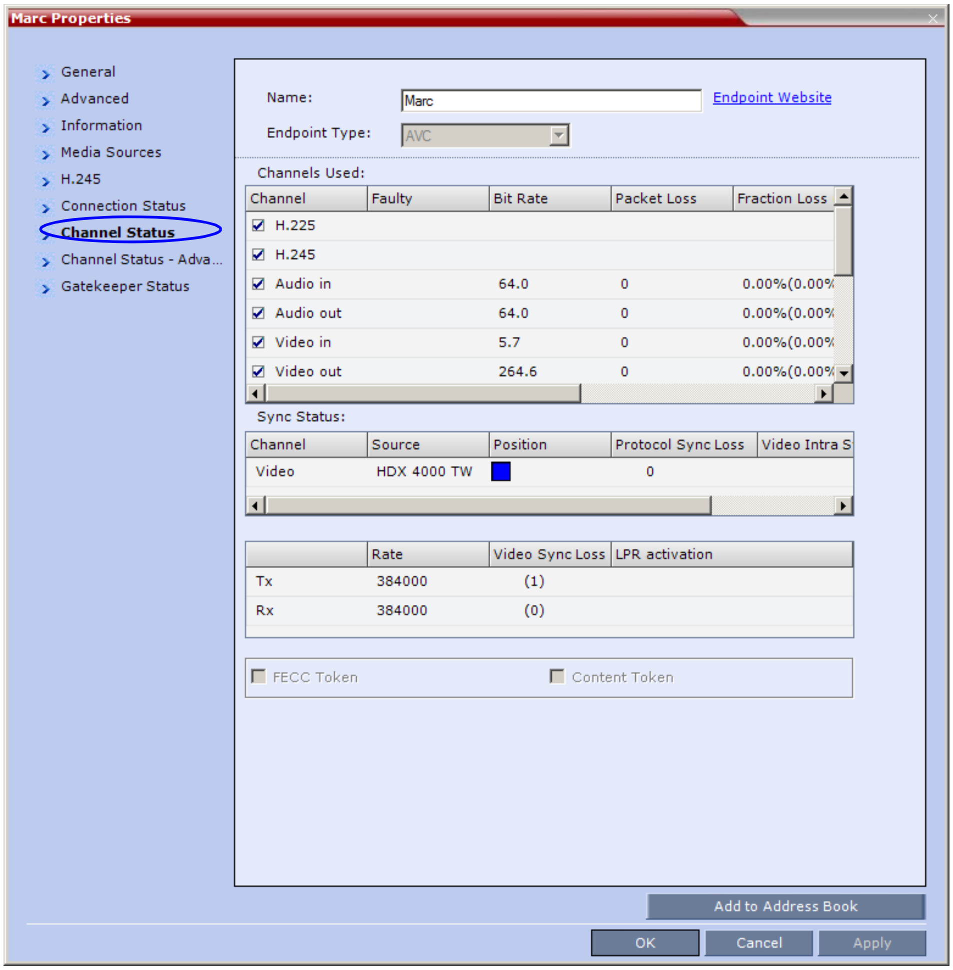
```
SirenLPR_ScalabletodoVideoCap
H264 (Transmit) - 0 bps
  H.239 Role: Presentation
  0 frame rate per second
  264Mode: standard
  BaseLine Profile
  Level 2.2
  CustomMaxMBPS at 44 (22000 MB/s)
  CustomMaxFS at 15 (3840 MBs)
AnnexQ - 6400 bps
BFCP
 setup        passive
 connection   new
```

Table 12-11 *Participant Properties - H.245/SDP Parameters*

Field	Description
Remote Capabilities	Lists the participant’s capabilities as declared by the endpoint.
Remote Communication Mode	Displays the actual capabilities used by the endpoint when establishing the connection with the MCU (Endpoint to MCU).
Local Communication Mode	Displays the actual capabilities used by the MCU when establishing the connection with the participant’s endpoint (MCU to Endpoint).

Viewing Permissions			
Tab	*Chairperson*	*Operator*	*Administrator*
Channel Status		✓	✓

4 Click on the **Channel Status** tab to view the status of the various channels.



Name: Marc

Endpoint Type: AVC

Channels Used:

Channel	Faulty	Bit Rate	Packet Loss	Fraction Loss
☑ H.225				
☑ H.245				
☑ Audio in		64.0	0	0.00%(0.00%
☑ Audio out		64.0	0	0.00%(0.00%
☑ Video in		5.7	0	0.00%(0.00%
☑ Video out		264.6	0	0.00%(0.00%

Sync Status:

Channel	Source	Position	Protocol Sync Loss	Video Intra S
Video	HDX 4000 TW	■	0	

	Rate	Video Sync Loss	LPR activation
Tx	384000	(1)	
Rx	384000	(0)	

☐ FECC Token ☐ Content Token

The following parameters are displayed:

Table 12-12 Participant Properties - Channel Status Parameters

Field	Description
Channels Used	When checked, indicates the channel type used by the participant to connect to the conference: Incoming channels are endpoint to MCU, Outgoing channels are from MCU to endpoint. **Channels:** • *H.225/Signaling* - The call-signaling channel. • *H.245/SDP* - The Control channel. • *Audio in - Incoming audio channel* • *Audio out - Outgoing audio channel* • *Video in - Incoming video channel* • *Video out - Outgoing video channel* • *Content in* - H.239/People+Content conferences • *Content out* - H.239/People+Content conferences • *FECC in* - The incoming FECC channel is open. • *FECC out* - The outgoing FECC channel is open. **Columns:** • **Faulty** – A red exclamation point indicates a faulty channel condition. This is a real-time indication; when resolved the indication disappears. An exclamation point indicates that further investigation may be required using additional parameters displayed in the *Advanced Channel Status* tab. • **Bit Rate** – The actual transfer rate for the channel. When channel is inactive, bit rate value is 0. For example, if the participant is connected without video, the bit rate for the video channel is 0. **Note:** The CTS Audio Auxiliary channel is used only for Content. In all other cases, the bit rate shown in this column for this channel is 0. • **Packet Loss** – The accumulated count of all packets that are missing according to the RTCP report since the channel was opened. This field is relevant only during the connection stage and does not display faulty indications. • **Fraction Loss (Peak)** – The ratio between the number of lost packets and the total number of transmitted packets since the last RTCP report. *Peak* (in parentheses) indicates the highest ratio recorded since the channel was opened. • **Number of Packets** – The number of received or transmitted packets since the channel has opened. This field does not cause the display of the faulty indicator. • **Jitter (Peak)** – Displays the network jitter (the deviation in time between the packets) as reported in the last RTCP report (in milliseconds). *Peak* (in parentheses) reflects the maximum network jitter since the channel was opened. • **Latency** – Indicates the time it takes a packet to travel from one end to another in milliseconds (derived from the RTCP report). High latency value may indicate that there is a problem in the network, or that the endpoint is sending an incorrect RTCP values.

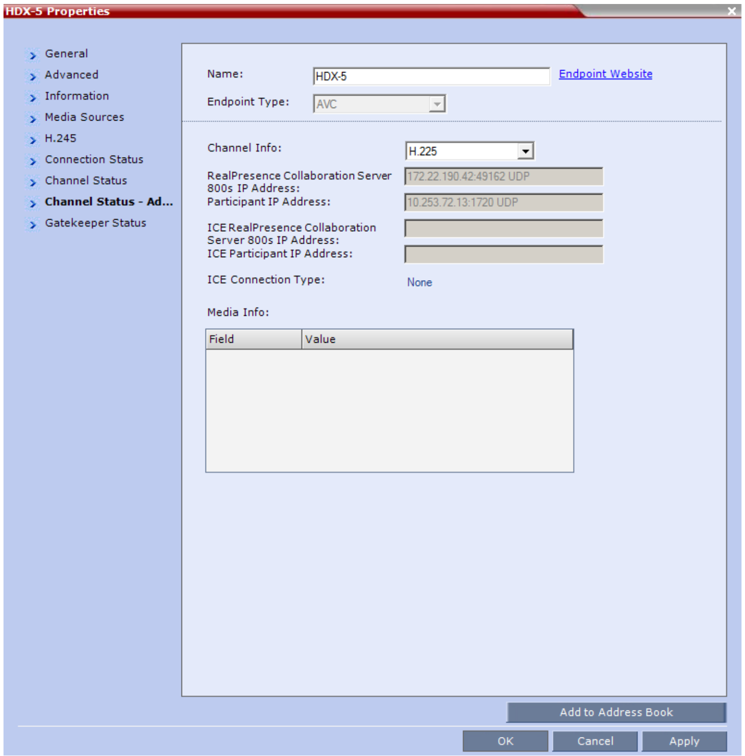
Table 12-12 Participant Properties - Channel Status Parameters (Continued)

Field	Description
Sync Status	*Channel* - The channel type: Video or Content. *Source* - The name of the participant currently viewed by this participant. ***Position*** - The video layout position indicating the place of each participant as they appear in a conference. ***Protocol Sync Loss*** - Indicates whether the system was able to synchronize the bits order according to the selected video protocol. ***Video Intra Sync*** - Indicates whether the synchronization on a video Intra frame was successful. ***Video Resolution*** - The video resolution of the participant.
Rx - Rate	The received line rate.
Tx - Rate	The transmitted line rate.
Tx - Video Sync Loss	When checked, indicates a video synchronization problem in the outgoing channel from the MCU. The counter indicates the sync-loss count.
Rx - Video Sync Loss	When checked, indicates a video synchronization problem in the incoming channel from the endpoint. The counter indicates the sync-loss count.
Tx - LPR Activation	When checked, indicates LPR activation in the outgoing channel.
Rx - LPR Activation	When checked, indicates LPR activation in the incoming channel.
FECC Token	When checked, indicates that the participant is the holder of the FECC Token.
Content Token	When checked, indicates that the participant is the holder of the Content Token.

5 Click the **Channel Status Advanced** tab to view additional information for selected audio and video channels.

In the *Channel Status - Advanced* tab, channels can be selected for viewing additional information:

Viewing Permissions			
Tab	*Chairperson*	*Operator*	*Administrator*
Channel Status Advanced			✔



HDX-5 Properties

Name: HDX-5

Endpoint Type: AVC

Channel Info: H.225

RealPresence Collaboration Server 800s IP Address: 172.22.190.42:49162 UDP

Participant IP Address: 10.253.72.13:1720 UDP

ICE RealPresence Collaboration Server 800s IP Address:

ICE Participant IP Address:

ICE Connection Type: None

Media Info:

Field	Value

***Table 12-13** Participant Properties - Channel Status Advanced Parameters*

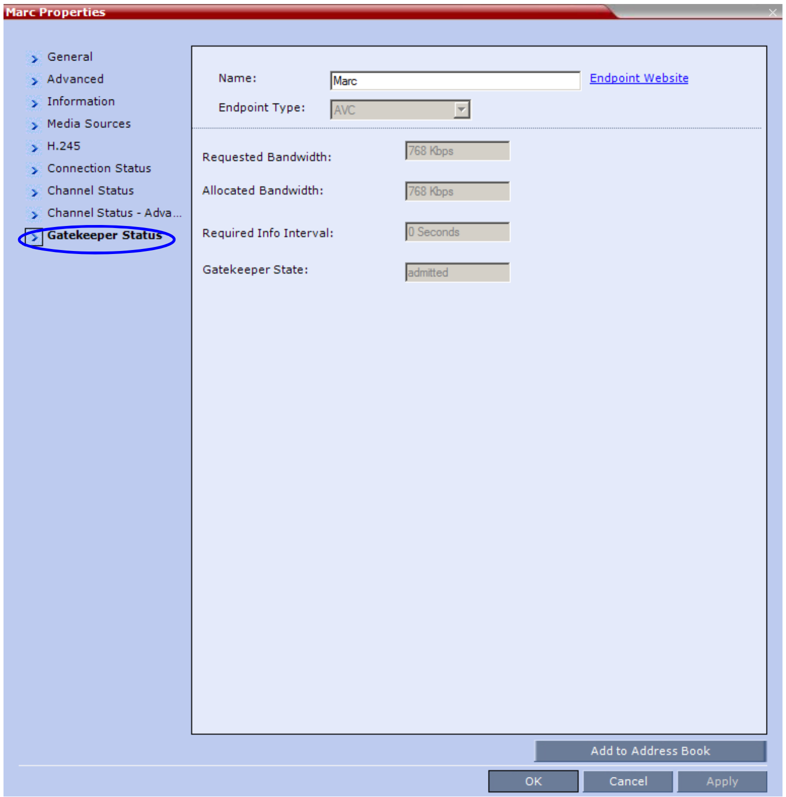
Field	Description
Channel Info	Select a channel to view its information: • H.225 • H.245 • Audio in • Audio out • Video in • Video out • Content in • Content Out • SIP BFCP TCP
Collaboration Server IP Address	The IP address and the transport protocol (TCP/UDP) of the MCU to which the participant is connected and the port number allocated to the participant incoming media stream on the MCU side.
Participant IP Address	The IP address and the transport protocol (TCP/UDP) of the participant and the port number allocated to the media stream on the participant side.

Table 12-13 Participant Properties - Channel Status Advanced Parameters (Continued)

Field	Description
ICE RealPresence Collaboration Server Virtual Edition IP Address	Not Supported with RealPresence Collaboration Server.
ICE Participant IP Address	Not Supported with RealPresence Collaboration Server..
ICE Connection Type	• Not Supported with RealPresence Collaboration Server.
Media Info	This table provides information about the audio and video parameters, such as video algorithm, resolution, etc. For more information, see *Appendix E:* "*Participant Properties Advanced Channel Information"* on page **E-1**.
RTP Statistics	This information may indicate problems with the network which can affect the audio and video quality. For more information, see *Appendix E:* "*Participant Properties Advanced Channel Information"* on page **E-1**.

6 **Optional for H.323 AVC-based participants.** Click the **Gatekeeper Status** tab to view its parameters.

Viewing Permissions			
Tab	*Chairperson*	*Operator*	*Administrator*
Gatekeeper Status	✓	✓	✓



Name: Marc

Endpoint Type: AVC

Requested Bandwidth: 768 Kbps

Allocated Bandwidth: 768 Kbps

Required Info Interval: 0 Seconds

Gatekeeper State: admitted

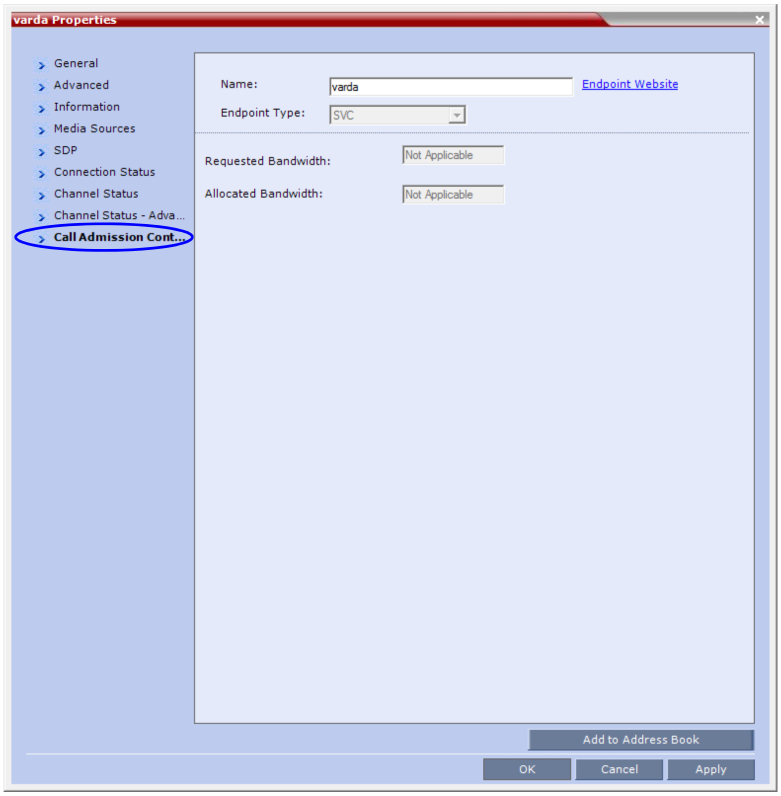
***Table 12-14** Participant Properties - Gatekeeper Status Parameters*

Field	**Description**
Requested Bandwidth	The bandwidth requested by the MCU from the gatekeeper.
Allocated Bandwidth	The actual bandwidth allocated by the gatekeeper to the MCU.
Required Info Interval	Indicates the interval, in seconds, between registration messages that the MCU sends to the gatekeeper to indicate that it is still connected.
Gatekeeper State	Indicates the status of the participant’s registration with the gatekeeper and the bandwidth allocated to the participant. The following statuses may be displayed: • ***ARQ*** – Admission Request - indicates that the participant has requested the gatekeeper to allocate the required bandwidth on the LAN. • ***Admitted*** – indicates that the gatekeeper has allocated the required bandwidth to the participant. • ***DRQ*** – Disengage Request – the endpoint informs the gatekeeper that the connection to the conference is terminated and requests to disconnect the call and free the resources. • ***None*** – indicates that there is no connection to the gatekeeper.

7 **Optional for SIP AVC-based and SVC-based participants.** Click the **Call Admission Control** tab to view its parameters.

Viewing Permissions

Tab	*Chairperson*	*Operator*	*Administrator*
Gatekeeper Status	✓	✓	✓



Name: varda

Endpoint Type: SVC

Requested Bandwidth: Not Applicable

Allocated Bandwidth: Not Applicable

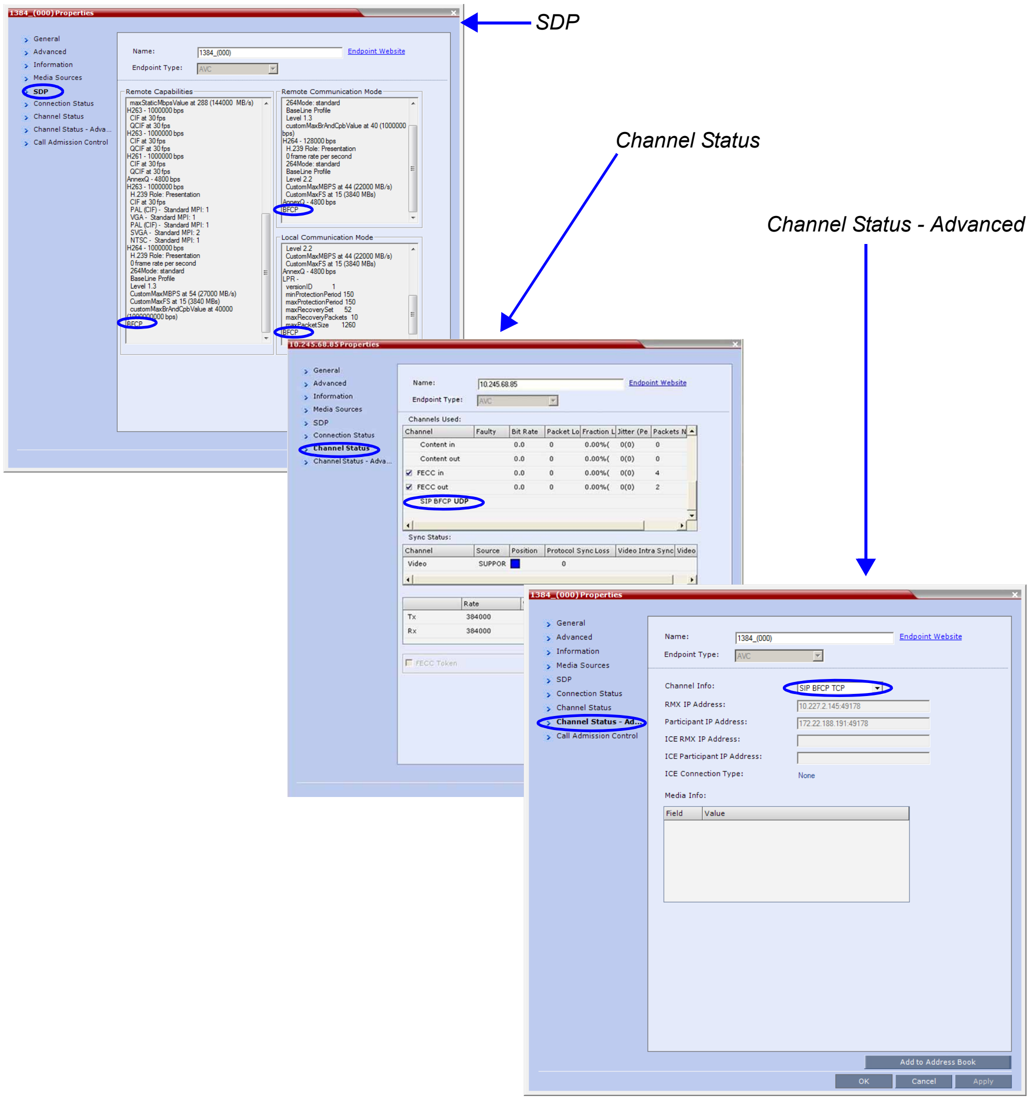
Table 12-15 Participant Properties - Gatekeeper Status Parameters

Field	Description
Requested Bandwidth	The bandwidth requested by the MCU from the SIP server.
Allocated Bandwidth	The actual bandwidth allocated by the SIP server to the MCU.

Monitoring SIP BFCP Content

In the SIP *Participant Properties* dialog box, *BFCP* status information appears in:

- All three panes of the *SDP* tab.
- The *Channel Status* tab.
- The *Channel Status -Advanced* tab.



Name: 1384_(000)
Endpoint Type: AVC

Remote Capabilities

```
 maxStaticMbpsValue at 288 (144000  MB/s)
H263 - 1000000 bps
 CIF at 30 fps
 QCIF at 30 fps
H263 - 1000000 bps
 CIF at 30 fps
 QCIF at 30 fps
H261 - 1000000 bps
 CIF at 30 fps
 QCIF at 30 fps
AnnexQ - 4800 bps
H263 - 1000000 bps
 H.239 Role: Presentation
 CIF at 30 fps
 PAL (CIF) -  Standard MPI: 1
 VGA -  Standard MPI: 1
 PAL (CIF) -  Standard MPI: 1
 SVGA -  Standard MPI: 2
 NTSC -  Standard MPI: 1
H264 - 1000000 bps
 H.239 Role: Presentation
 0 frame rate per second
 264Mode: standard
 BaseLine Profile
 Level 1.3
 CustomMaxMBPS at 54 (27000 MB/s)
 CustomMaxFS at 15 (3840 MBs)
 customMaxBrAndCpbValue at 40000
(1000000000 bps)
BFCP
```

Remote Communication Mode

```
 264Mode: standard
 BaseLine Profile
 Level 1.3
 customMaxBrAndCpbValue at 40 (1000000
bps)
H264 - 128000 bps
 H.239 Role: Presentation
 0 frame rate per second
 264Mode: standard
 BaseLine Profile
 Level 2.2
 CustomMaxMBPS at 44 (22000 MB/s)
 CustomMaxFS at 15 (3840 MBs)
AnnexQ - 4800 bps
BFCP
```

Local Communication Mode

```
 Level 2.2
 CustomMaxMBPS at 44 (22000 MB/s)
 CustomMaxFS at 15 (3840 MBs)
AnnexQ - 4800 bps
LPR -
 versionID          1
 minProtectionPeriod 150
 maxProtectionPeriod 150
 maxRecoverySet      52
 maxRecoveryPackets  10
 maxPacketSize       1260
BFCP
```

Name: 10.245.68.85
Endpoint Type: AVC

Channels Used:

Channel	Faulty	Bit Rate	Packet Lo	Fraction L	Jitter (Pe	Packets N
Content in		0.0	0	0.00%(	0(0)	0
Content out		0.0	0	0.00%(	0(0)	0
☑ FECC in		0.0	0	0.00%(	0(0)	4
☑ FECC out		0.0	0	0.00%(	0(0)	2
SIP BFCP UDP						

Sync Status:

Channel	Source	Position	Protocol Sync Loss	Video Intra Sync	Video
Video	SUPPOR	■	0		

	Rate
Tx	384000
Rx	384000

□ FECC Token

Name: 1384_(000)
Endpoint Type: AVC

Channel Info: SIP BFCP TCP
RMX IP Address: 10.227.2.145:49178
Participant IP Address: 172.22.188.191:49178
ICE RMX IP Address:
ICE Participant IP Address:
ICE Connection Type: None

Media Info:

Field	Value

For more information see "*Participant Level Monitoring"* on page **12-19**.

Detecting SIP Endpoint Disconnection

When an abnormal disconnection of SIP endpoints occurs because of network problems or client application failures, SIP endpoints remain connected to the conference causing connection disruptions. For example, the video freezes in the layout or blocks content for SIP endpoints when a quick re-connection is performed. It can take several minutes to detect the SIP endpoint disconnection using the SIP standard behavior.

In a normal SIP video call, audio and video (RTP and RTCP) messages are sent from the endpoints to the MCU to detect the signaling of connected endpoints. Conversely, SVC endpoints might not send video RTP messages to the MCU when a participant is not displayed in the video layout of any of the participants in the conference. For SVC endpoints, the MCU will only verify audio RTP and RTCP messages and video RTCP messages. Video RTP messages will not be checked.

To detect the disconnection of SIP endpoints in a reasonable amount of time, a new system flag can be defined to specify the amount of time that the MCU should wait for an RTCP or RTP message from the SIP endpoint before the endpoint starts the disconnection process. The system default value is automatically set to 20 seconds.

The system flag, DETECT_SIP_EP_DISCONNECT _TIMER, contains the amount of time in seconds to wait for an RTCP or RTP message to be received from the endpoint. When the time that was set in the system flag has elapsed and no RTCP or RTP audio or video message has been received on either the audio or the video channel, the MCU disconnects the SIP endpoint from the conference. A CDR event record is created with a Call Disconnection Cause of "SIP remote stopped responding".

The Microsoft Lync add-in endpoint opens audio and content channels. Lync endpoints can send RTCP/RTP messages and empty RTP audio messages. When the time that was set in the system flag has elapsed and no RTCP or RTP message has been received on the audio channel, the MCU disconnects the endpoint from the conference.

SIP audio only endpoints use the audio channel only. When the time that was set in the system flag has elapsed and no RTCP or RTP message has been received on the audio channel, the MCU disconnects the SIP audio endpoint from the conference.

Configuring the System Flag

When you want to change the system default value of 20 seconds, the system flag, DETECT_SIP_EP_DISCONNECT_TIMER, can be manually added to the *System Flags* configuration to detect the disconnection of SIP endpoints. For more information see "*Manually Adding and Deleting System Flags"* on page **20-12**.

The value range is from 0 to 300 seconds. When the value is set between 0 and 14, the feature is disabled and SIP endpoints are not detected for disconnection. When the value is set between 15 and 300, the feature is enabled.

Recording Conferences

Conference recording is not available in SVC Conferencing Mode.

Conferences running on the *Collaboration Server* can be recorded using a *Polycom® RSS™ Recording and Streaming Server* (*RSS*).

The recording system can be installed at the same site as the conferencing *MCU* or at a remote site. Several *MCU's* can share the same recording system.

Recording conferences is enabled via a *Recording Link*, which is a dial-out connection from the conference to the recording system.

Recording can start automatically, when the first participant connects to a conference, or on request, when the *Collaboration Server* user or conference chairperson initiates it.

Multiple Recording Links may be defined.

Conference Recording Links can be associated on the *Collaboration Server* with *Virtual Recording Rooms* (*VRR*), created and saved on the *Polycom® RSS™ 4000 Version 8.5 Recording and Streaming Server* (*RSS*).

Each *Recording Link* defined on the *Collaboration Server* can be given a descriptive name and can be associated with one *VRR* saved on the *Polycom RSS 4000*.

The following guidelines apply:

- A *Recording Link* that is being used by an ongoing conference cannot be deleted.
- A *Recording Link* that is assigned to a *Profile* cannot be deleted.
- The *Recording Link* supports H.264 High Profile with H.323 connections.
- While a *Profile* is being used in an ongoing conference, it cannot have a different *Recording Link* assigned to it.
- Up to 100 Recording Links can be listed for selection in the Conference Profile.
- *Multiple Recording Links* are supported in *Continuous Presence* and *Video Switched* conferences.
- The number of *Recording Links* available for selection is determined by the value of the **MAXIMUM_RECORDING_LINKS** *System Flag* in *system.cfg*. Default value is 20 Recording Links.
- The recording link can be encrypted when recording from an encrypted conference to the RSS that is set to encryption. For more details, see "*Recording Link Encryption"* on page **13-7**.

Creating Multiple Virtual Recording Rooms on the RSS

If the environment includes a *Polycom® RSS™ 4000 Version 8.5 Recording and Streaming Server (RSS)* and you want to associate *Recording Links* on the *Collaboration Server* with *Virtual Recording Rooms* (*VRR*), created and saved on the *Polycom® RSS™ 4000 Version 8.5* perform the following operations on the *RSS*:

1. Modify the parameters of a recording *Template* to meet the recording requirements.
2. Assign the modified recording *Template* to a *VRR*. The recording and streaming server will assign a number to the *VRR*.
3. Repeat Step 1 and Step 2 for each *VRR* to create additional *VRR*s.

   For more information see the *RSS 4000 User Guide.*

Configuring the Collaboration Server to Enable Recording

To make recording possible the following components you must be configured on the *Collaboration Server*:

- *Recording Link* – defines the connection between the conference and the recording system.
- *Recording-enabled Conference IVR Service* – recording *DTMF* codes and messages must be set in the *Conference IVR* Service to enable "recording-related" voice messages to be played and to allow the conference chairperson to control the recording process using *DTMF* codes.
- *Recording-enabled Profile* – recording must be enabled in the *Conference Profile* assigned to the recorded conference.

If *Multiple Recording Links* are being defined for *Virtual Recording Rooms* (*VRRs*), created and saved on the *Polycom® RSS™ 4000 Version 8.5*, the **MAXIMUM_RECORDING_LINKS** *System Flag* in *system.cfg* can be modified to determine the number of *Recording Links* available for selection.

- **Range:** 20 - 100
- **Default:** 20

The flag value can be modified by selecting the *System Configuration* option from the *Setup* menu. For more information, see "*Modifying System Flags"* on page **20-1**.

Defining the Recording Link

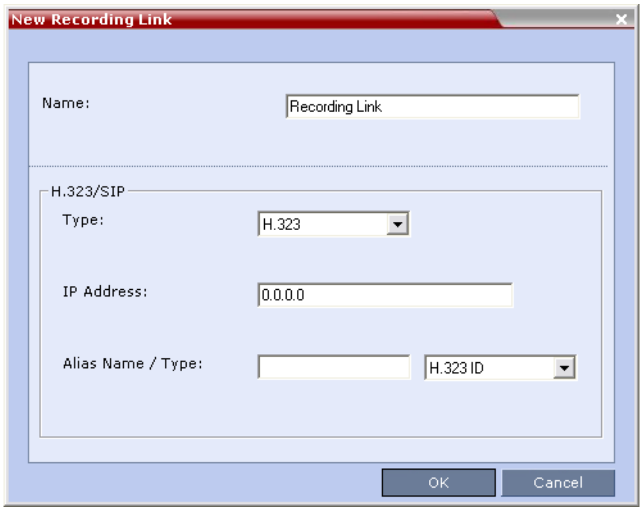
The *Recording Link* is defined once and can be updated when the *H.323* alias or the IP address (of the recording system) is changed. Only one *Recording Link* can be defined in the *Collaboration Server*. Its type must be *H.323*.

In *Multiple Networks* Configuration, Recording Links use the default Network Service to connect to conferences, therefore the recording system must be defined on the default IP Network Service to enable the recording.

To define a Recording Link:

1. In the *Collaboration Server Management* pane, click **Recording Links** ( ).
2. In the *Recording Links* list, click the **New Recording Link** ( ) button.

The *New Recording Link* dialog box is displayed.



New Recording Link

Name: Recording Link

H.323/SIP

Type: H.323

IP Address: 0.0.0.0

Alias Name / Type: H.323 ID

3 Define the following parameters:

Table 13-1 Recording Link Parameters

Parameter	Description
Name	Displays the default name that is assigned to the Recording Link. If multiple Recording Links are defined, it is recommended to use a descriptive name to be indicate the VRR to which it will be associated. Default: *Recording Link*
Type	Select the network environment: • H.323 • SIP
IP Address	• If no gatekeeper is configured, enter the IP Address of the RSS. Example: If the RSS IP address is 173.26.120.2 enter `173.26.120.2.` • If a gatekeeper is configured, you can either enter the IP address or an alias (see the alias description).
Alias Name	If using the endpoint's alias instead of IP address, first select the alias type and then enter the endpoint's alias. If you are associating this recording link to a VRR on the RSS, define the alias as follows: • If you are using the RSS IP address, enter the VRR number in the Alias field. For example, if the VRR number is 5555, enter **5555**. • Alternatively, if the *Alias Type* is set to **H.323 ID**, enter the RSS IP address and the VRR number in the format: **<RSS_IP_Address>##<VRR number>** For example: If the RSS IP is 173.26.120.2 and the VRR number is 5555, enter **173.26.120.2##5555**
Alias Type	Depending on the format used to enter the information in the IP address and Alias fields, select **H.323 ID** or **E.164** (for multiple Recording links). **E-mail ID** and **Participant Number** are also available.

4 Click **OK**.

The Recording Link is added to the *Collaboration Server* unit.

Enabling the Recording Features in a Conference IVR Service

To record a conference, a *Conference IVR Service* in which the recording messages and DTMF codes are activated must be assigned to the conference. The default *Conference IVR Service* shipped with the *Collaboration Server* includes the recording-related voice messages and default DTMF codes that enable the conference chairperson to control the recording process from the endpoint. You can modify these default settings.

To modify the default recording settings for an existing Conference IVR Service:

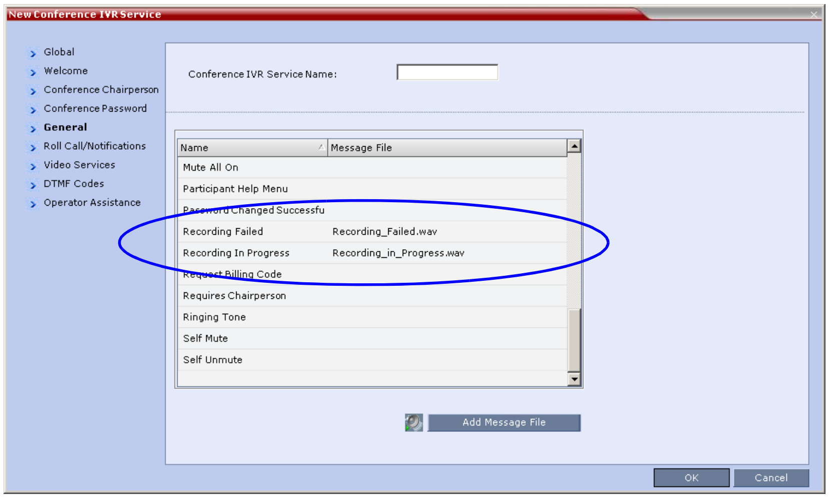
1 In the *Collaboration Server Management* pane, click the **IVR Services** (▦) button.

The IVR Services are listed in the *IVR Services* list pane.

2 To modify the default recording settings, double-click the Conference IVR Service or right-click and select **Properties**.

The *Conference IVR Service Properties* dialog box is displayed.

3 To assign voice messages other than the default, click the **General** tab and scroll down the list of messages to the recording messages.



Conference IVR Service Name:

Name	Message File
Mute All On	
Participant Help Menu	
Password Changed Successfu	
Recording Failed	Recording_Failed.wav
Recording In Progress	Recording_in_Progress.wav
Request Billing Code	
Requires Chairperson	
Ringing Tone	
Self Mute	
Self Unmute	

4 Select the *Recording In Progress* message, and then select the appropriate message file (by default, Recording_in_Progress.wav) from the file list to the right of the field.

5 Select the *Recording Failed* message, and then select the appropriate message file (by default, Recording_Failed.wav) from the file list to the right of the field.

6 To modify the default DTMF codes, click the **DTMF Codes** tab.

7 To modify the DTMF code or permission for a recording function:

a Select the desired DTMF name (Start, Stop or Pause Recording), click the DTMF code entry and type a new code.

Table 13-2 Default DTMF Codes assigned to the recording process

Recording Operation	DTMF Code	Permission
Start or Resume Recording	*3	Chairperson
Stop Recording	*2	Chairperson
Pause Recording	*1	Chairperson

b In the *Permission* entry, select whether this function can be used by all conference participants or only the chairperson.

8 Click **OK**.

Enabling the Recording in the Conference Profile

To be able to record a conference, the recording options must be enabled in the *Conference Profile* assigned to it. You can add recording to existing *Profiles* by modifying them.

To enable recording for a conference:

1 In the *Collaboration Server Management* pane, click the **Conference Profiles** ( ) button.

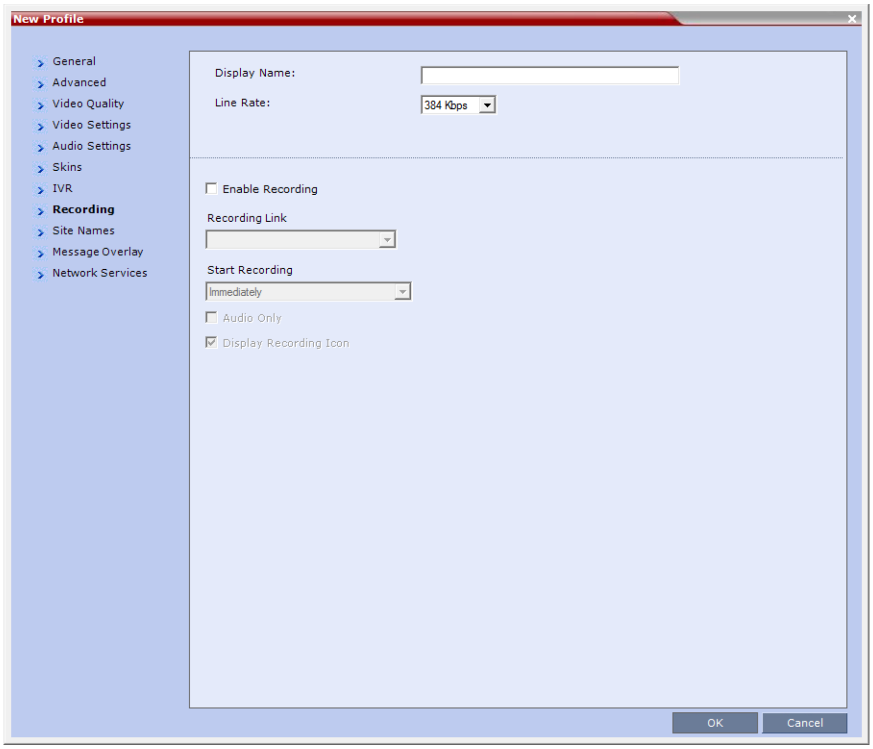
The *Conference Profiles* list is displayed.

2 Create a new profile by clicking the **New Profile** ( ) button or modify an existing profile by double-clicking or right-clicking an existing profile and then selecting **Profile Properties**.

If creating a new profile, complete the conference definition. For more information on creating Profiles see on "*Defining New Profiles”* on page 2-20.

3 In the *Profile Properties* dialog box, click the **Recording** tab.

4 Select the **Enable Recording** check box.



Display Name:

Line Rate: 384 Kbps

□ Enable Recording

Recording Link

Start Recording: Immediately

□ Audio Only

☑ Display Recording Icon

5 Define the following parameters:

Table 13-3 Conference Profile Recording Parameters

Parameter	Description
Enable Recording	Select to enable Recording Settings in the dialog box.
Recording Link	Select a recording link for the conference from the list.
Start recording	Select one of the following: • **Immediately** – conference recording is automatically started upon connection of the first participant. • **Upon Request** – the operator or chairperson must initiate the recording (manual).
Audio only	Select this option to record only the audio channel of the conference. **Note:** An *Audio Only* Recording Link cannot be used to record a conference if there are no Voice resources allocated in the *Video/Voice Port Configuration*.
Display Recording Icon	Select this option to display *Recording Indication* to all conference participants informing them that the conference is being recorded. The recording icon is replaced by a *Paused* icon when conference recording is paused.

6 Click **OK**.
Recording is enabled in the *Conference Profile*.

Recording Link Encryption

The Recording Link can be encrypted when recording an encrypted conference. The encryption of the *Recording Link* is enabled when *Encryption* is selected in the *Conference Profile* on the *Collaboration Server* and on the RSS, and the system flag **ALLOW_NON_ENCRYPT_RECORDING_LINK_IN_ENCRYPT_CONF** is set to **NO**.

Recording Link Encryption Guidelines:

- The *Recording Link* connection type must be H.323.
- The *Recording Link* uses the *AES* encryption format.
- The *RSS 4000* recorder must be set to support encryption. For more information see the *RSS 4000 User Manual*.
- Encryption must be selected in the *Conference Profile.*

Recording Link Encryption Flag Setting

Recording Links are treated as regular participants, however if the ALLOW_NON_ENCRYPT_RECORDING_LINK_IN_ENCRYPT_CONF *System Flag* is set to YES a non-encrypted *Recording Link* is to be allowed to connect to an encrypted conference.

Table 13-4 summarizes the connection possibilities for a *Recording Link* that is to be connected to a conference for each of the conference *profile* and *Entry Queue* encryption options.

Table 13-4 Connections by Recording Link and Conference Encryption Settings

Conference Profile Setting	Recording Link Connection Status according to flag: ALLOW_NON_ENCRYPT_RECORDING_ LINK_IN_ENCRYPT_CONF	
	YES	NO
Encrypt All	Connected encrypted if possible, otherwise connected non-encrypted.	Connected only if encrypted, otherwise disconnected.
No Encryption	Connected non-encrypted.	Connected non-encrypted.
Encrypt when possible	Connected encrypted if possible, otherwise connected non-encrypted.	Connected encrypted if possible, otherwise connected non-encrypted.

Recording Link Settings

The recording of encrypted conferences via an encrypted *Recording Link* is enabled in the *Conference Profile* by:

- Selecting the *Encryption* option (**Encrypt All** or **Encrypt when Possible**) in the *Advanced* tab.

  For more details, see "*Media Encryption"* on page **3-26**.
- Setting the Recording options in the *Recording* tab. For more details, see "*Enabling the Recording in the Conference Profile"* on page **13-5**.

Managing the Recording Process

When a conference is started and recording is enabled in its *Profile*, the system will automatically start the recording if the *Start Recording* parameter is set to *immediately*. If it is set to *Upon Request*, the system waits for the chairperson or *Collaboration Server* user's request. Once the recording is initiated for a conference, the MCU connects to the recording device (*RSS 2000/4000*) using the default *Recording Link*. The connection that is created between the conference and the recording device is represented as a special participant (Recording) whose name is the *Recording Link*. Once the recording has started, the recording process can be stopped and restarted from the Chairperson's endpoint (using DTMF codes) or from the *Collaboration Server Web Client*. After the recording process has finished, the recording can be identified in the *RSS 2000/4000* by its *Collaboration Server* conference name.

A conference participant and the *Recording Link* cannot have identical names, otherwise the recording process will fail.

Recording Link Layout

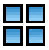
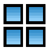
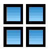
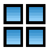
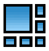
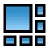
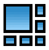
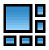
When the video layout of the conference is set to *Auto Layout*, the recording of the conference will now include all the conference participants and not n-1 participants as in previous versions.

In the new *Auto Layout* algorithm, the *Recording Link* is counted as a "participant" and therefore it is excluded from the layout display used for the recording. The layout used for the other participants will behave as in the "standard" *Auto Layout* behavior.

The *Recording Link Layout* can be changed during an ongoing conference in the same manner as for any other conference participant. For more information see the "*Participant Level Monitoring"* on page **12-19**.

The default settings for *Auto Layout* for the conference and the *Recording Link* are summarized in the following table:

Table 14 *Recording Link Default Layout Settings (Auto Layout Mode)*

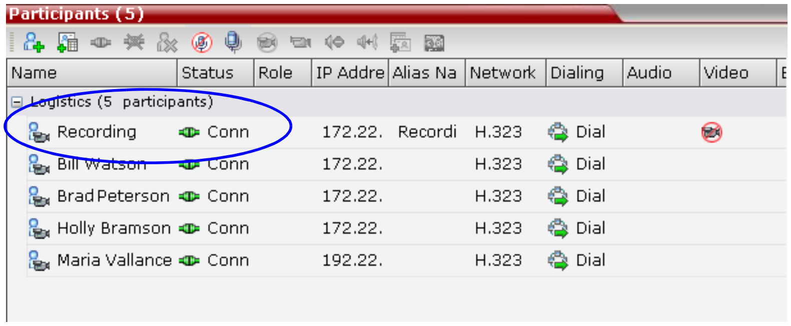
Participants	Conference Auto Layout Default Settings	Recording Link Auto Layout Settings
0	Not applicable	Not applicable
1		
2		
3		
4		
5		
6		
7		
8		
9		
10 or more		

The default settings for *Auto Layout* of the *Recording Link* cannot be changed, and the *Auto Layout* flags do not apply to the *Recording Link Auto Layout* default settings.

Using the Collaboration Server Web Client to Manage the Recording Process

To manage the recording process using the right-click menu:

>> Right-click the *Recording* participant in the conference and select from one of the following options:



Participants (5)

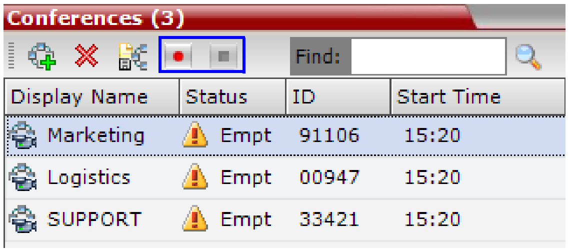
Name	Status	Role	IP Addre	Alias Na	Network	Dialing	Audio	Video	E
Logistics (5 participants)									
Recording	Conn		172.22.	Recordi	H.323	Dial			
Bill Watson	Conn		172.22.		H.323	Dial			
Brad Peterson	Conn		172.22.		H.323	Dial			
Holly Bramson	Conn		172.22.		H.323	Dial			
Maria Vallance	Conn		192.22.		H.323	Dial			

Table 13-1 Recording Participant Right-click Options

Name	Description
Start	Starts recording. When recording has started, this option toggles with the *Pause* option.
Pause	Pauses the recording of the conference without disconnecting. When the Recording is Paused, this option toggles with the *Start* option.
Resume	Resumes the recording of the conference. The Resume option toggles with the *Pause* option when it is used.
Stop	Stops the recording. **Note:** The Stop button is only enabled when the Recording is *Started* or *Paused*.
Suspend Video	The Suspend Video option prevents the incoming video of the recording link participant to be part of the conference layout. The Recording Link participant is set by default to Suspend Video. The Suspend Video option toggles with the Resume Video option.
Resume Video	The Resume Video option enables the incoming video of the recording link participant to be part of the conference layout. This feature may be used to play back previously recorded video or audio feeds in the conference layout. For more information, see the RSS 4000 User Guide.
Participant Properties	The Participant Properties option displays viewing only information for monitoring, e.g. communication capabilities and channels used to connect to the conference. Users will not be able to perform any functional requests from this window, i.e. disconnect, change layout and mute.

To manage the recording process using the Conference toolbar:

>> In the *Conferences* pane, click one of the following buttons in the Conference tool bar.



Display Name	Status	ID	Start Time
Marketing	Empt	91106	15:20
Logistics	Empt	00947	15:20
SUPPORT	Empt	33421	15:20

The recording buttons will only be displayed in the conference tool bar for a conference that is recording-enabled.

Table 13-2 Conferences List - Recording Tool bar buttons

Button	Description
●	Start/Resume recording. This button toggles with the *Pause* button.

Table 13-2 Conferences List - Recording Tool bar buttons (Continued)

Button	Description
■	Stop recording.
⏸	Pause recording. This button toggles with the *Start/Resume* button.

Using DTMF Codes to Manage the Recording Process

By entering the appropriate DTMF code on the endpoint, the chairperson can **Stop** the recording (*74), **Pause** it (*75), or **Start/Resume** the recording (*73). For more information on managing the recording process via DTMF codes, see the *RSS 2000 User's Guide*.

Recording between the *Collaboration Server* and the *Codian VCR* is enabled by adding an IP participant to the recorded conference that acts as a link between the conference and the recording device. This participant is identified as a recording link to the Codian *VCR* according to the product ID sent from the *VCR* during the connection phase, in the call setup parameters.

The video channel between the conference and the recording device is unidirectional where the video stream is sent from the conference to the recorder.

If the *Codian VCR* opens a video channel to the conference - this channel is excluded from the conference video mix.

To record a conference running on the Collaboration Server using Codian recorder:

>> In the conference, define or add a dial-out participant using the *Codian VCR* IP address as the address for dialing.

Once added to the conference, the *MCU* automatically connects the participant (the link to *Codian VCR*) and the recording is automatically started on the *Codian VCR*.

A connection can also be defined on the *Codian VCR*, dialing into the recorded conference using the *MCU* prefix and the *Conference ID* as for any other dial-in participant in the conference.

Monitoring the recording participant:

This connection is monitored as any other participant in the conference. The connection can also be monitored in the *Codian VCR* web client.

14

Users, Connections, and Notes

Users

Collaboration Server Web Client users are defined in the User's table and can connect to the MCU to perform various operations.

A maximum of 100 users can be defined per MCU.

User Types

The MCU supports the following user Authorization Levels:

- Administrator
- Operator
- Machine Account (Application-user)
- Administrator Read-only
- Chairperson
- Auditor

> Users with *Auditor* authorization level cannot connect to the *Collaboration Server* via the *RMX Manager* application and must use the *Collaboration Server* Web Client.

The authorization level determines a user's capabilities within the system.

Administrator

An administrator can define and delete other users, and perform all configuration and maintenance tasks.

Administrator Read-only

A user with Administrator permission with the same viewing and monitoring permissions of a regular Administrator. However, this user is limited to creating system backups and cannot perform any other configuration or conference related operation.

Operator

An Operator can manage Meeting Rooms, Profiles, Entry Queues, and SIP Factories, and can also view the *Collaboration Server* configurations, but cannot change them.

Administrator and Operator users can verify which users are defined in the system. Neither of them can view the user passwords, but an Administrator can change a password.

Chairperson

A Chairperson can only manage ongoing conferences and participants. The Chairperson does not have access to the *Collaboration Server* configurations and utilities.

Auditor

An **Auditor** can only view *Auditor Files* and audit the system.

Machine Account

User names can be associated with servers (machines) to ensure that all users are subject to the same account and password policies. For more details, see "*Machine Account"* on page **14-6**.

Listing Users

The *Users* pane lists the currently defined users in the system and their authorization levels. The pane also enables the administrators to add and delete users.

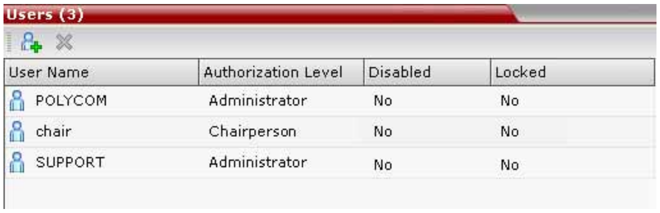
The system is shipped with a default Administrator user called POLYCOM, whose password is POLYCOM. However, once you have defined other authorized Administrator users, it is recommended to remove the default user.

You can view the list of users that are currently defined in the system.

To view the users currently defined in the system:

1 In the *Collaboration Server Management* pane, click the **Users** (👥) button.

The *Users* pane is displayed.



Users (3)

User Name	Authorization Level	Disabled	Locked
POLYCOM	Administrator	No	No
chair	Chairperson	No	No
SUPPORT	Administrator	No	No

The list includes three columns: User Name, Authorization Level and Disabled.

The *User Name* is the login name used by the user to connect to the MCU.

The *Authorization* indicates the Authorization Level assigned to the User: *Administrator, Administrator Read-only, Operator, Chairperson* or *Auditor*.

Disabled indicates whether the user is disabled and cannot access the system unless enabled by the administrator. For more details, see "*Disabling a User"* on page **14-4**. *Locked* indicates whether the user has been locked out and cannot access the system unless enabled by the administrator.

Adding a New User

Administrators can add new users to the system.

The User Name and Password must be in ASCII.

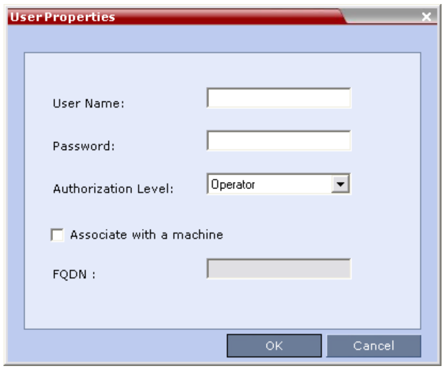
To add a new user to the system:

1 In the *Collaboration Server Management* pane, click the **Users** (👥) button.

2 The *Users* pane is displayed.

3 Click the **New User** () button or right-click anywhere in the pane and then click **New User**.

The *User Properties* dialog box opens.



User Name:

Password:

Authorization Level: Operator

□ Associate with a machine

FQDN :

4 In the *User Name* text box, enter the name of the new user. This is the login name used by the user when logging into the system.

5 In the *Password* text box, enter the new user's password. This will be the user's password when logging into the system.

6 In the *Authorization Level* list, select the user type: **Administrator, Administrator Read-Only, Operator, Chairperson** or **Auditor**.

7 **Optional. To associate a user with a machine**:

a In the *User Properties* dialog box, select the **Associate with a machine** check box.

b Enter the *FQDN* of the server that hosts the application who's application-user name is being added. Example: `cma1.polycom.com`

8 Click **OK**.
The *User Properties* dialog box closes and the new user is added to the system.

Deleting a User

> To delete a user, you must have Administrator authorization. The last remaining Administrator in the *Users* list cannot be deleted.

1 In the *Collaboration Server Management* pane, click the **Users** () button.

2 Select the user and click the **Delete** () button or right-click the user and then click **Delete User**.

The system displays a confirmation message.

3 In the *confirmation* dialog box, select **Yes** to confirm or **No** to cancel the operation.

If you select **Yes**, the user name and icon are removed from the system.

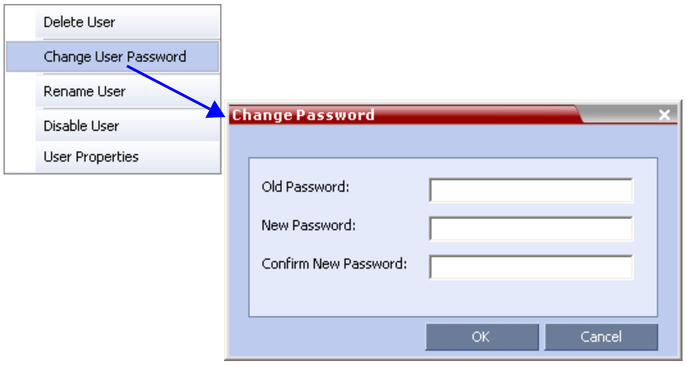
Changing a User's Password

Users with Administrator authorization can change their own password and other users' passwords. Users with Operator authorization can change their own password.

To change a user's password:

1. In the *Collaboration Server Management* pane, click the **Users** (👥) option.
2. Right-click the user and click **Change User Password**.

   The *Change Password* dialog box opens.



3. Enter the *Old Password* (current), *New Password* and *Confirm the New Password*.

> The Password must be in ASCII.

4. Click **OK**.

   The user's password is changed.

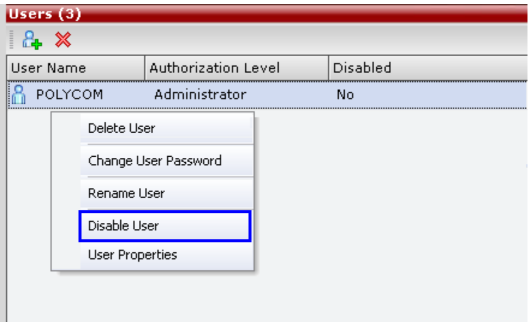
Disabling a User

An administrator can disable an enabled user. An indication is displayed in the Users List when the User is disabled. An administrator can enable a disabled User.

To disable a user:

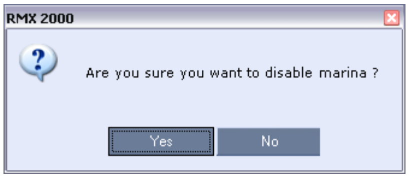
1. In the *Collaboration Server Management* pane, click the **Users** (👥) button.

   The Users pane is displayed.
2. In the *Users* pane, right-click the user to be disabled and select **Disable User** in the menu.



User Name	Authorization Level	Disabled
POLYCOM	Administrator	No

A confirmation box is displayed.



Are you sure you want to disable marina ?

3 Click **YES**.
The User status in the *Users* list - *Disabled* column changes to **Yes**.

Enabling a User

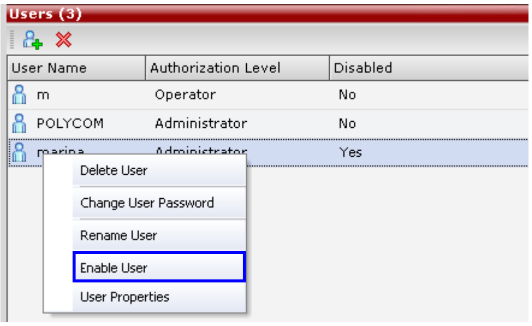
An administrator can enable a User who was disabled manually by the administrator.

To enable a user:

1 In the *Collaboration Server Management* pane, click the **Users** ( ) button.

The *Users* pane is displayed.

2 Right-click the user to be enabled and select **Enable User**.



User Name	Authorization Level	Disabled
m	Operator	No
POLYCOM	Administrator	No
marina	Administrator	Yes

A confirmation box is displayed.

3 Click **YES**.
The User status in the *Users* list - *Disabled* column changes to **NO**.

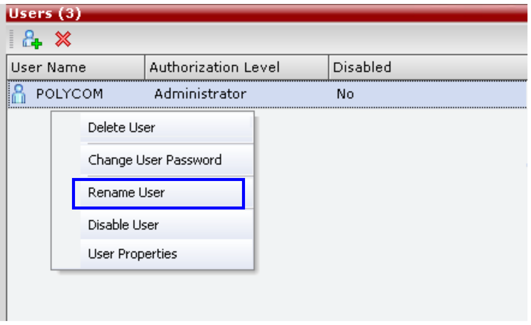
Renaming a User

To rename a user:

1 In the *Collaboration Server Management* pane, click the **Users** ( ) button.

The Users pane is displayed.

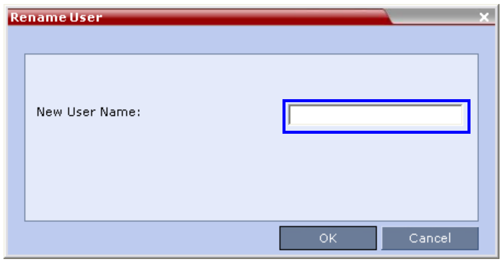
2 Right-click the user to be renamed and select **Rename User**.



Users (3)

User Name	Authorization Level	Disabled
POLYCOM	Administrator	No

The *Rename User* dialog box is displayed.



3 Enter the user's new name in the *New User Name* field and click **OK**.

The user is renamed and is forced to change his/her password.

Machine Account

User names can be associated with servers (machines) to ensure that all users are subject to the same account and password policies.

For enhanced security reasons it is necessary for the *Collaboration Server* to process user connection requests in the same manner, whether they be from regular users accessing the *Collaboration Server* via the *Collaboration Server Web Browser / RMX Manager* or from *application-users* representing applications such as *CMA* and *RealPresence DMA* system.

Regular users can connect from any workstation having a valid certificate while application-users representing applications can only connect from specific servers. This policy ensures that a regular user cannot impersonate an *application-user* to gain access to the *Collaboration Server* in order to initiate an attack that would result in a *Denial of Service* (*DoS*) to the impersonated application.

The connection process for an *application-user* connecting to the *Collaboration Server* is as follows:

1 The *application-user* sends a connection request, including its *TLS* certificate, to the *Collaboration Server*.

2 The *Collaboration Server* searches its records to find the *FQDN* that is associated with the *application-user's* name.

3 If the *FQDN* in the received certificate matches that associated with *application-user*, and the password is correct, the connection proceeds.

Guidelines

- *Application-users* are only supported when *TLS* security is enabled and *Request peer certificate* is selected. *TLS* security cannot be disabled until all *application-user* accounts have been deleted from the system.
- For *Secure Communications*, an administrator must set up on the *Collaboration Server* system a machine account for the *RealPresence CMA/DMA/XMA* system with which it interacts. This machine account must include a fully-qualified domain name (*FQDN*) for the *RealPresence CMA/DMA/XMA* system.
- *Application-user* names are the same as regular user names.
  Example: the *CMA* application could have an *application-user* name of `CMA1`.
- The *FQDN* can be used to associate all user types: *Administrator*, *Operator* with the *FQDN* of a server.
- Multiple *application-users* can be configured the same *FQDN* name if multiple applications are hosted on the same server
- If the system is downgraded the *application-user's FQDN* information is not deleted from the *Collaboration Server*'s user records.
- A *System Flag,* **PASS_EXP_DAYS_MACHINE,** enables the administrator to change the password expiration period of *application-user's* independently of regular users. The default flag value is 365 days.
- The server hosting an *application-user* whose password is about to expire will receive a login response stating the number of days until the *application-user's* password expires. This is determined by the value of the **PASSWORD_EXPIRATION_WARNING_DAYS** *System Flag*. The earliest warning can be displayed 14 days before the password is due to expire and the latest warning can be displayed 7 days before passwords are due to expire. An *Active Alarm* is created stating the number of days before the password is due to expire.
- The **MIN_PWD_CHANGE_FREQUENCY_IN_DAYS** *System Flag* does not effect *application-user* accounts. Applications typically manage their own password change frequency.
- If an *application-user* identifies itself with an incorrect *FQDN*, its account will not be locked, however the event is written to the *Auditor Event File*.
- If an *application-user* identifies itself with a correct *FQDN* and an incorrect password, its account will be locked and the event written to the *Auditor Event File*.
- An *application-user* cannot be the last administrator in the system. The last administrator must be regular user.
- User names are not case sensitive.

Monitoring

- An *application-user* and its connection is represented by a specific icon.

Active Directory

- When working with *Active Directory*, *CMA, RealPresence DMA* system, and *XMA* cannot be registered within *Active Directory* as regular users. *CMA* and *RealPresence DMA* system *application-users* must be manually.
- The only restriction is that TLS mode is enabled together with client certificate validation.
- If the above configuration are set off it will not be possible to add machine accounts.

- When setting the TLS mode off the system should check the existence of a machine account and block this operation until all machine accounts are removed.

Connections

The *Collaboration Server* enables you to list all connections that are currently logged into the MCU, e.g. users, servers or API users. The MCU issues an ID number for each login. The ID numbers are reset whenever the MCU is reset.

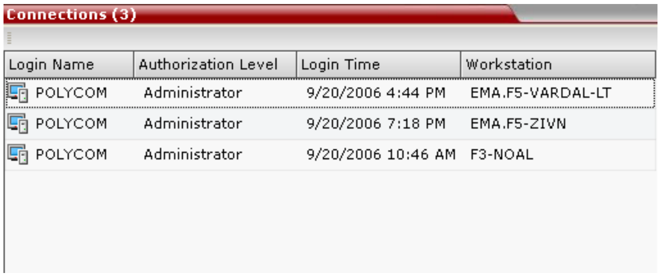
A maximum of 50 users can be concurrently logged in to the MCU.

Viewing the Connections List

To list the users who are currently connected to the MCU:

1 In the *Collaboration Server Management* pane, click the **Connections** () button.

A list of connected users is displayed in the *Connections* pane.



Connections (3)

Login Name	Authorization Level	Login Time	Workstation
POLYCOM	Administrator	9/20/2006 4:44 PM	EMA.F5-VARDAL-LT
POLYCOM	Administrator	9/20/2006 7:18 PM	EMA.F5-ZIVN
POLYCOM	Administrator	9/20/2006 10:46 AM	F3-NOAL

The information includes:

— The user's login name.

— The user's authorization level (Chairperson, Operator, Administrator or Auditor).

— The time the user logged in.

— The name/identification of the computer used for the user's connection.

Notes

Notes are the electronic equivalent of paper sticky notes. You can use notes to write down questions, important phone numbers, names of contact persons, ideas, reminders, and anything you would write on note paper. *Notes* can be left open on the screen while you work.

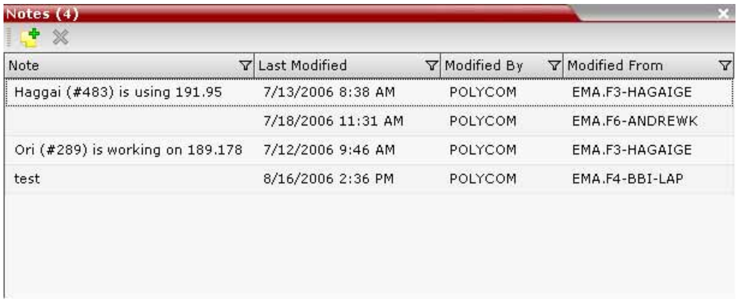
Notes can be read by all system Users concurrently connected to the MCU. Notes that are added to the *Notes* list are updated on all workstations by closing and re-opening the *Notes* window. Notes can be written in any Unicode language.

Using Notes

To create a note:

1 On the *Collaboration Server* menu, click **Administration** > **Notes**.

The *Notes* window opens.



Notes (4)

Note	Last Modified	Modified By	Modified From
Haggai (#483) is using 191.95	7/13/2006 8:38 AM	POLYCOM	EMA.F3-HAGAIGE
	7/18/2006 11:31 AM	POLYCOM	EMA.F6-ANDREWK
Ori (#289) is working on 189.178	7/12/2006 9:46 AM	POLYCOM	EMA.F3-HAGAIGE
test	8/16/2006 2:36 PM	POLYCOM	EMA.F4-BBI-LAP

2 In the *Notes* toolbar, click the **New Note** (✚) button, or right-click anywhere inside the *Notes* window and select **New Note**.

3 In the *Note* dialog box, type the required text and click **OK**.

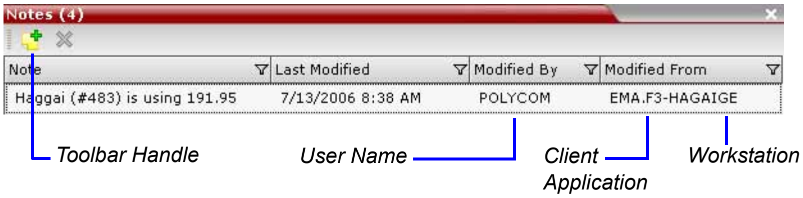
The new note is saved and closed. The *Notes* list is updated, listing the new note and its properties:

— **Note** – The beginning of the note's text.

— **Last Modified** – The date of creation or last modification.

— **Modified By** – The *Login Name* of the user who last modified the note.

— **Modified From** – The *Client Application* and *Workstation* from which the note was created or modified.



Notes (4)

Note	Last Modified	Modified By	Modified From
Haggai (#483) is using 191.95	7/13/2006 8:38 AM	POLYCOM	EMA.F3-HAGAIGE

To open or edit a note:

4 Double-click the entry to edit, or right-click the entry and select **Note Properties**.

The note opens for viewing or editing.

To delete a note:

1 In the *Notes* list, select the entry for the note to delete and click the **Delete Note** button (☒), or right-click the entry and select **Delete Note**.

A *delete confirmation* dialog box is displayed.

2 Click **OK** to delete the note, or click **Cancel** to keep the note.

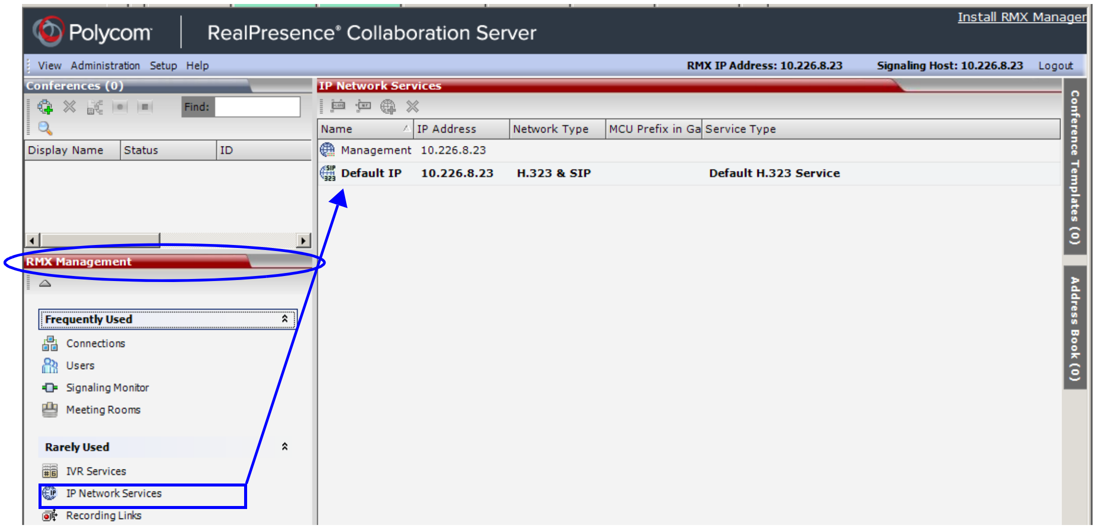
15

IP Network Services

To enable the RealPresence Collaboration Server Virtual Edition to function within IP network environments, network parameters must be defined for the *IP Network Services*.

> The RealPresence Collaboration Server Virtual Edition allows you only to view the parameters of the IP Network Services in theRealPresence Collaboration Server Web Client or the but you cannot define a new IP Network Service or modify the parameters of an existing Network Service. Attempting to do so may cause unexpected results, including complete inability to use or access the RealPresence Collaboration Server. These settings can be modified only using the text user interface. For more information, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide, "Manual IP Configuration"* on page **2-22**.

The configuration dialog boxes for the IP network services are accessed via the *Collaboration Server Management* pane of the *Collaboration Server Web Client*.



RMX IP Address: 10.226.8.23

Signaling Host: 10.226.8.23

Name	IP Address	Network Type	MCU Prefix in Ga	Service Type
Management	10.226.8.23			
Default IP	10.226.8.23	H.323 & SIP		Default H.323 Service

IP Network Services

Two *IP Services* are defined for the Collaboration Server:

- **Management Network**
- **Default IP Service (Conferencing Service)**

Dial in, dial out connections and Collaboration Server management are supported within IPv4 addressing environments. Both *IP Services* use the same IP. Therefore, only one IP is needed for the Collaboration Server.

Management Network (Primary)

The *Management Network* is used to control the Collaboration Server, mainly via the *RealPresence Collaboration Server Web Client* application. The *Management Network* contains the network parameters, such as the IP address of the *Control Unit,* needed for connection between the Collaboration Server unit and the *RP Collaboration Server Web Client.* This IP address can be used by the administrator or service personnel to connect to the *Control Unit* should the MCU become corrupted or inaccessible.

In a typical environment where DHCP is enabled, the *Management Network* parameters are automatically set during *First Time Power-up* and whenever the Collaboration Server is restarted. In an environment where DHCP is not enabled, the *Management Network* properties must be set manually. For more information, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Manual IP Configuration"* on page **2-22**.

Default IP Service (Conferencing Service - Media and signaling)

The *Default IP Service (media and signaling)* is used to configure and manage communications between the Collaboration Server and conferencing devices such as endpoints, gatekeepers, SIP servers, etc. The same IP is used for both the *Management Network* and the *Default IP Service*.

The *Default IP Service* contains parameters for:

- Signaling Host IP Address
- External conferencing devices

Calls from all external IP entities are made to the *Signaling Host,* which initiates call set-up. Conferencing related definitions such as environment (H.323 or SIP) are also defined in this service.

In a typical environment where DHCP is enabled, the *Default IP Service* is configured automatically during *First Time Power-up* and whenever the Collaboration Server is restarted, using the same parameters as are used for the *Management Network*. In an environment where DHCP is not enabled, the *Default IP Service* properties are set when the *Management Network* properties are set manually. For more information, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Manual IP Configuration"* on page **2-22**.

Changes made to any of these parameters only take effect when the Collaboration Server is reset. An *Active Alarm* is created when changes made to the system have not yet been implemented and the MCU must be reset.

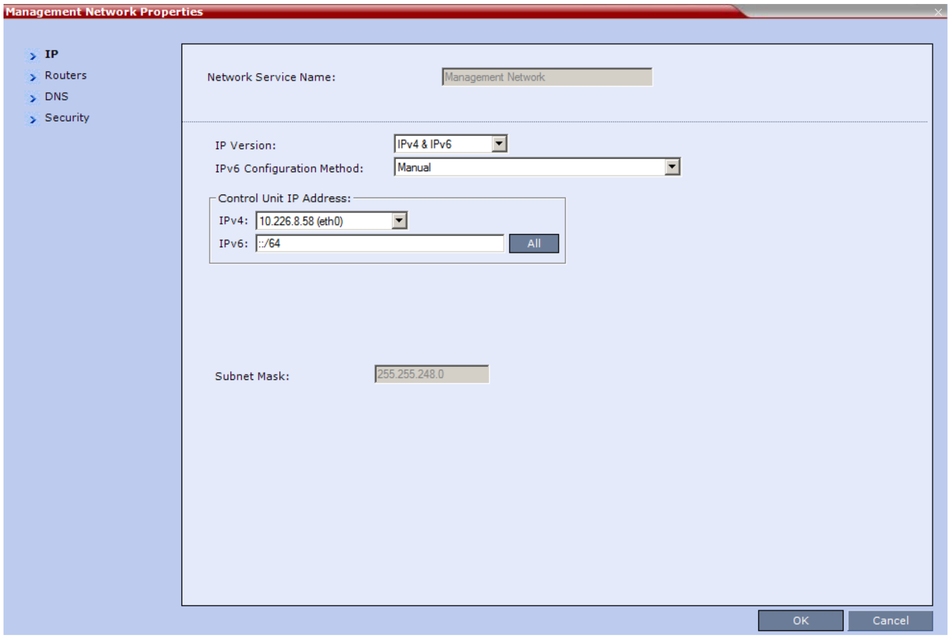
Modifying the Management Network

In the RealPresence Collaboration Server Virtual Edition, these settings can only be changed in the console Text User Interface. For more information, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Manual IP Configuration"* on page **2-22**.

To view the Management Network Service:

1. In the *Collaboration Server RMX Management* pane, click the **IP Network Services** ( ) button.
2. In the *IP Network Services* list pane, double-click the **Management Network** ( ) entry.

The *Management Network Properties - IP* dialog box opens.



Network Service Name: Management Network

IP Version: IPv4 & IPv6

IPv6 Configuration Method: Manual

Control Unit IP Address:

IPv4: 10.226.8.58 (eth0)

IPv6: ::/64

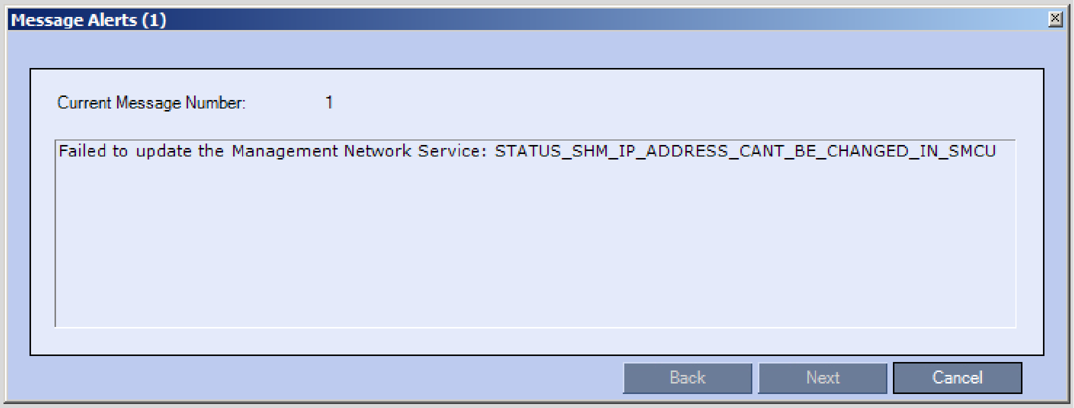
Subnet Mask: 255.255.248.0

The following fields can be viewed, but can not be modified:

Table 15-1 *Default Management Network Service – IP*

Field	Description	
Network Service Name	Displays the name of the Management Network. This name cannot be modified. **Note:** This field is displayed in all Management Network Properties tabs.	
Control Unit IP Address	IPv4	The IPv4 address of the Collaboration Server. This IP address is used by the *Collaboration Server Web Client* to connect to the Collaboration Server.
Subnet Mask	The subnet mask of the *Management Network Service.*	

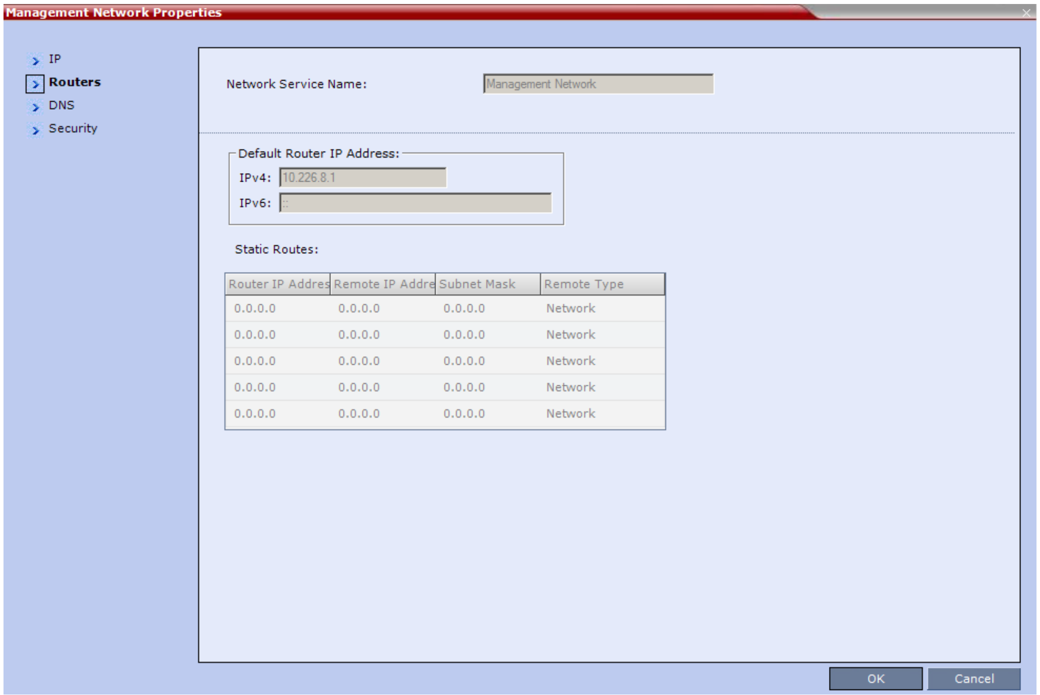
If an attempt is made to modify these settings, the message below will be displayed:



Current Message Number: 1

Failed to update the Management Network Service: STATUS_SHM_IP_ADDRESS_CANT_BE_CHANGED_IN_SMCU

3 Click the **Routers** tab.



Network Service Name: Management Network

Default Router IP Address:

IPv4: 10.226.8.1

IPv6: ::

Static Routes:

Router IP Addres	Remote IP Addre	Subnet Mask	Remote Type
0.0.0.0	0.0.0.0	0.0.0.0	Network
0.0.0.0	0.0.0.0	0.0.0.0	Network
0.0.0.0	0.0.0.0	0.0.0.0	Network
0.0.0.0	0.0.0.0	0.0.0.0	Network
0.0.0.0	0.0.0.0	0.0.0.0	Network

The following fields can be viewed but not modified.:

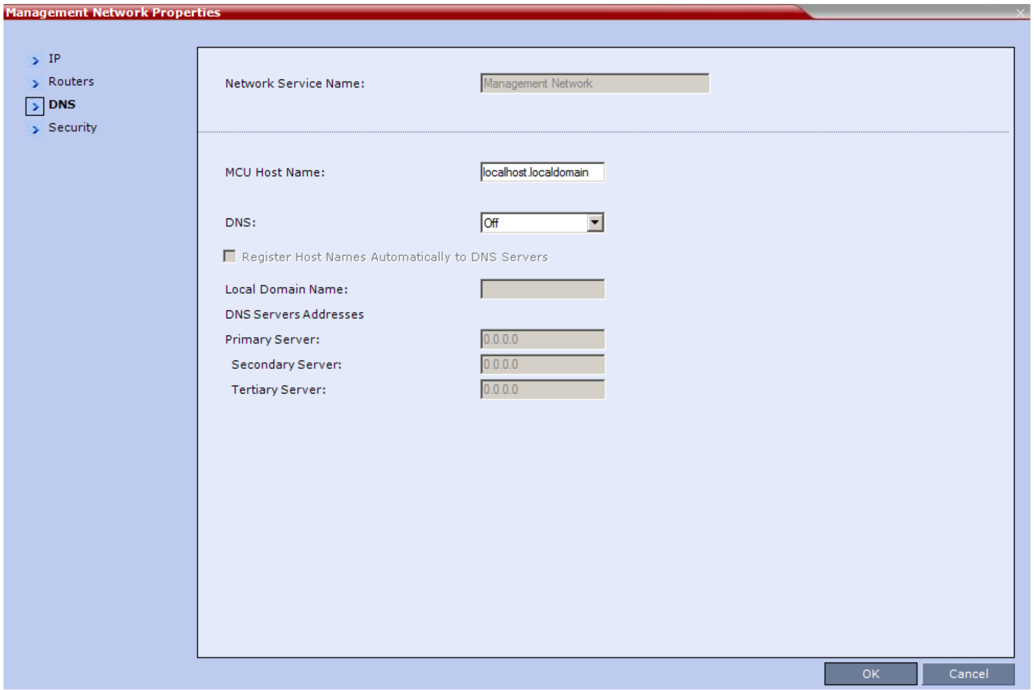
Table 15-2 Default Management Network Service – Routers

Field	Description	
Default Router IP Address	IPv4	The IP address of the default router. The default router is used whenever the defined static routers are not able to route packets to their destination. The default router is also used when host access is restricted to one default router.
	IPv6	

Table 15-2 *Default Management Network Service – Routers (Continued)*

Field	Description	
Static Routes		The system uses *Static Routes* to search other networks for endpoint addresses that are not found on the local LAN. Up to five routers can be defined in addition to the Default Router. The order in which the routers appear in the list determines the order in which the system looks for the endpoints on the various networks. If the address is in the local subnet, no router is used. To define a static route (starting with the first), click the appropriate column and enter the required value.
	Router IP Address	The IP address of the router.
	Remote IP Address	The IP address of the entity to be reached outside the local network. The *Remote Type* determines whether this entity is a specific component (Host) or a network. • If Host is selected in the *Remote Type* field, enter the IP address of the endpoint. • If Network is selected in the *Remote Type* field, enter of the segment of the other network.
	Remote Subnet Mask	The subnet mask of the remote network.
	Remote Type	The type of router connection: • **Network** – defines a connection to a router segment in another network. • **Host** – defines a direct connection to an endpoint found on another network.

4 Click the **DNS** tab.



Network Service Name: Management Network

MCU Host Name: localhost.localdomain

DNS: Off

□ Register Host Names Automatically to DNS Servers

Local Domain Name:

DNS Servers Addresses

Primary Server: 0.0.0.0

Secondary Server: 0.0.0.0

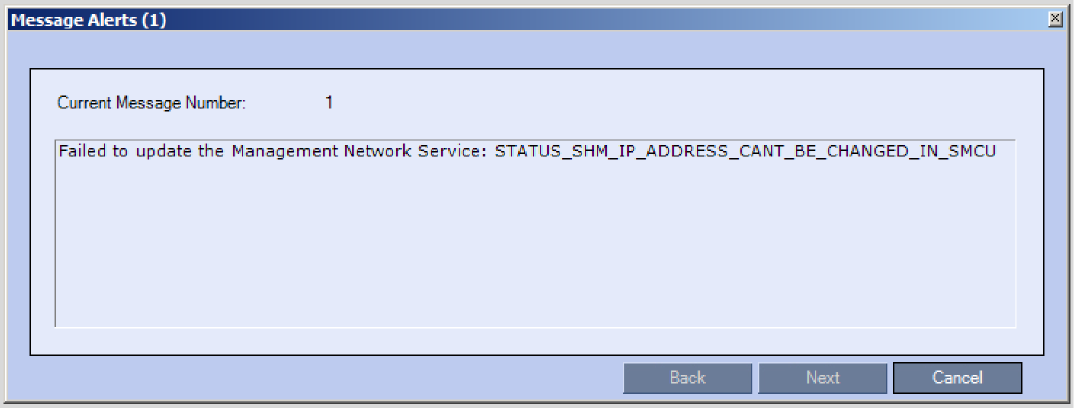
Tertiary Server: 0.0.0.0

The following fields can be modified, but their values will not be applied:

Table 15-3 Default Management Network Service – DNS

Field	Description
MCU Host Name	The name of the MCU on the network. Default name is PolycomMCU
DNS	• **Off** – if DNS servers are not used in the network. • **Specify** – to enter the IP addresses of the DNS servers. **Note:** The IP address fields are enabled only if **Specify** is selected.
Register Host Names Automatically to DNS Servers	Select this option to automatically register the MCU Signaling Host with the DNS server.
Local Domain Name	Enter the name of the domain where the MCU is installed.
DNS Servers Addresses:	
Primary Server	The static IP addresses of the DNS servers. A maximum of three servers can be defined.
Secondary Server	
Tertiary Server	

If an attempt is made to modify these settings, the message below will be displayed:



Current Message Number: 1

Failed to update the Management Network Service: STATUS_SHM_IP_ADDRESS_CANT_BE_CHANGED_IN_SMCU

5 Click **OK.**

6 If you have modified the *Management Network Properties*, reset the MCU.

Modifying the Default IP Network Service

In the RealPresence Collaboration Server Virtual Edition, both the Default IP Network Service and the Management Network Service use the same IP address and subnet mask. These settings can be modified only using the text user interface. To modify the IP address and subnet mask, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, *"Manual IP Configuration"* on page 2-22.

The *Default IP Service* parameters need to be modified if you want to change the:

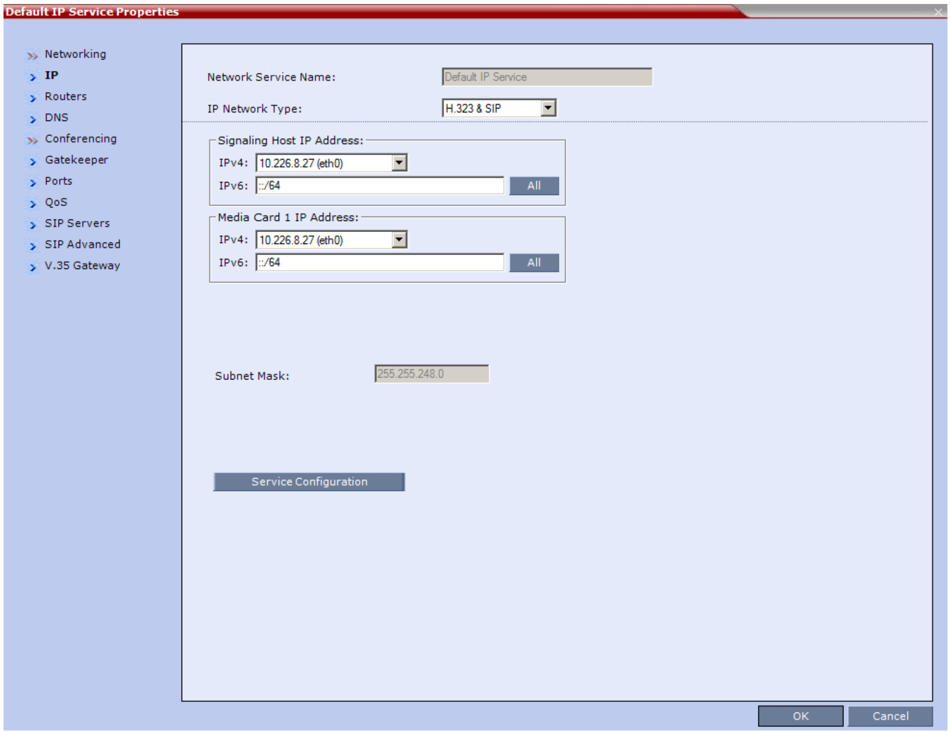
- Gatekeeper parameters or add gatekeepers to the Alternate Gatekeepers list
- SIP server parameters

To view or modify the Default IP Service:

1 In the *Collaboration Server Management* pane, click **IP Network Services** (IP).

2 In the *Network* list pane, double-click the **Default IP Service** (323, SIP, or SIP 323) entry.

The *Default IP Service - Networking IP* dialog box opens.



Network Service Name: Default IP Service

IP Network Type: H.323 & SIP

Signaling Host IP Address:

IPv4: 10.226.8.27 (eth0)

IPv6: ::/64

Media Card 1 IP Address:

IPv4: 10.226.8.27 (eth0)

IPv6: ::/64

Subnet Mask: 255.255.248.0

3 Modify the following fields:

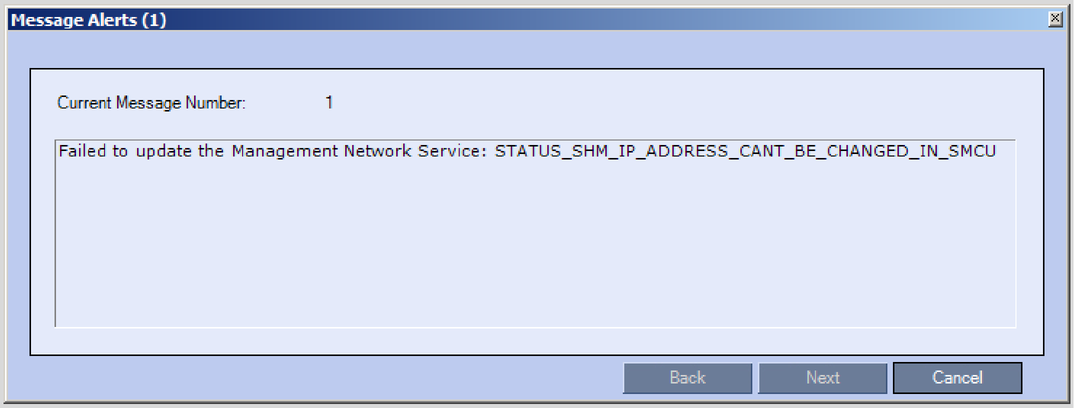
Table 15-4 Default IP Network Service – IP

Field	Description
Network Service Name	The name *Default IP Service* is assigned to the IP Network Service by the Fast Configuration Wizard. This field can not be changed. **Note:** This field is displayed in all IP Signaling dialog boxes and can contain character sets that use Unicode encoding.
IP Network Type	Displays the network type selected during the First Entry configuration. The Default IP Network icon indicates the selected environment. You can select: • **H.323:** For an H.323-only Network Service. • **SIP:** For a SIP-only Network Service. • **H.323 & SIP:** For an integrated IP Service. Both H.323 and SIP participants can connect to the MCU using this service. **Note:** This field is displayed in all Default IP Service tabs.
Signaling Host IP Address	This field can not be changed. One IP address is used for signaling and media transmission. Dial out calls from the Collaboration Server are initiated from this address. This address is used to register the RMX with a Gatekeeper or a SIP Proxy server.

Table 15-4 Default IP Network Service – IP (Continued)

Field	Description
Media Card 1 IP Address	This field can not be changed. One IP address is used for signaling and media transmission.
Subnet Mask	The subnet mask of the MCU. This field can not be changed. Default value: 255.255.255.0.

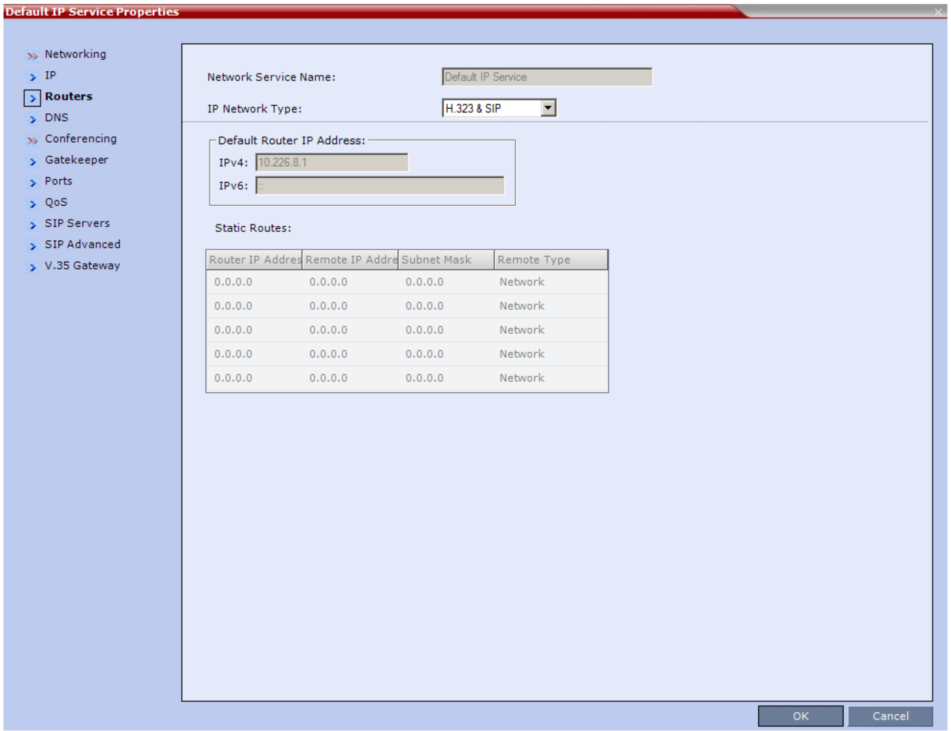
If an attempt is made to modify settings that can not be modified, the message below will be displayed after clicking **OK.**:



Current Message Number: 1

```
Failed to update the Management Network Service: STATUS_SHM_IP_ADDRESS_CANT_BE_CHANGED_IN_SMCU
```

4 Click the **Routers** tab.



Network Service Name: Default IP Service

IP Network Type: H.323 & SIP

Default Router IP Address:

IPv4: 10.226.8.1

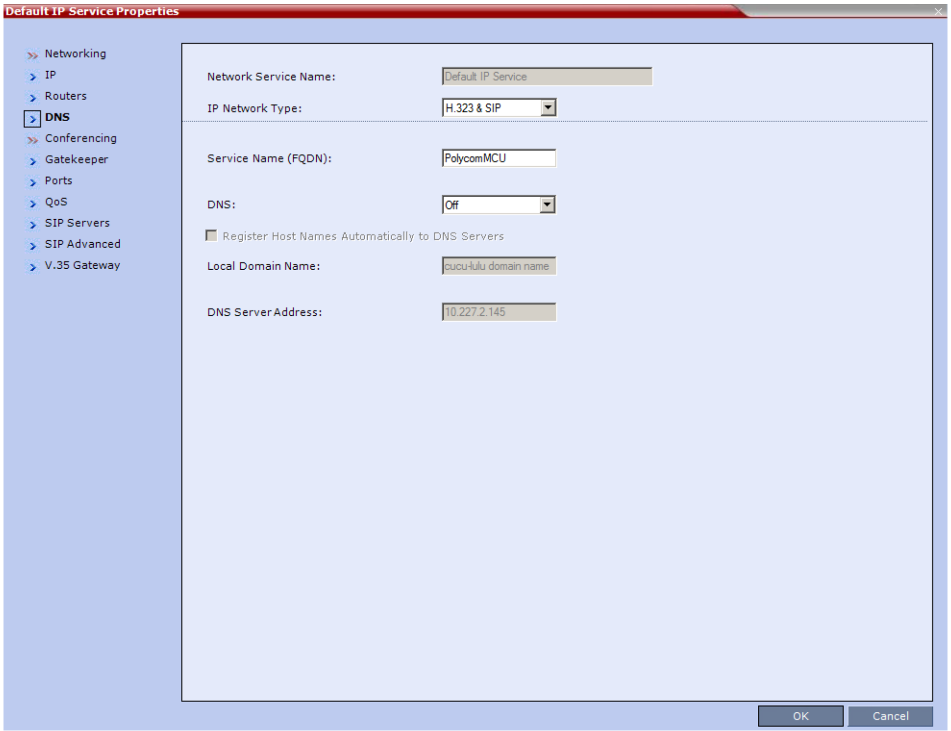
IPv6: ::

Static Routes:

Router IP Addres	Remote IP Addre	Subnet Mask	Remote Type
0.0.0.0	0.0.0.0	0.0.0.0	Network
0.0.0.0	0.0.0.0	0.0.0.0	Network
0.0.0.0	0.0.0.0	0.0.0.0	Network
0.0.0.0	0.0.0.0	0.0.0.0	Network
0.0.0.0	0.0.0.0	0.0.0.0	Network

With the exception of *IP Network Type*, the field definitions of the *Routers* tab are the same as for the *Default Management Network*. For more information see page 15-4.

5 **Optional.** Click the **DNS** tab.



Network Service Name: Default IP Service

IP Network Type: H.323 & SIP

Service Name (FQDN): PolycomMCU

DNS: Off

□ Register Host Names Automatically to DNS Servers

Local Domain Name: cucu-lulu domain name

DNS Server Address: 10.227.2.145

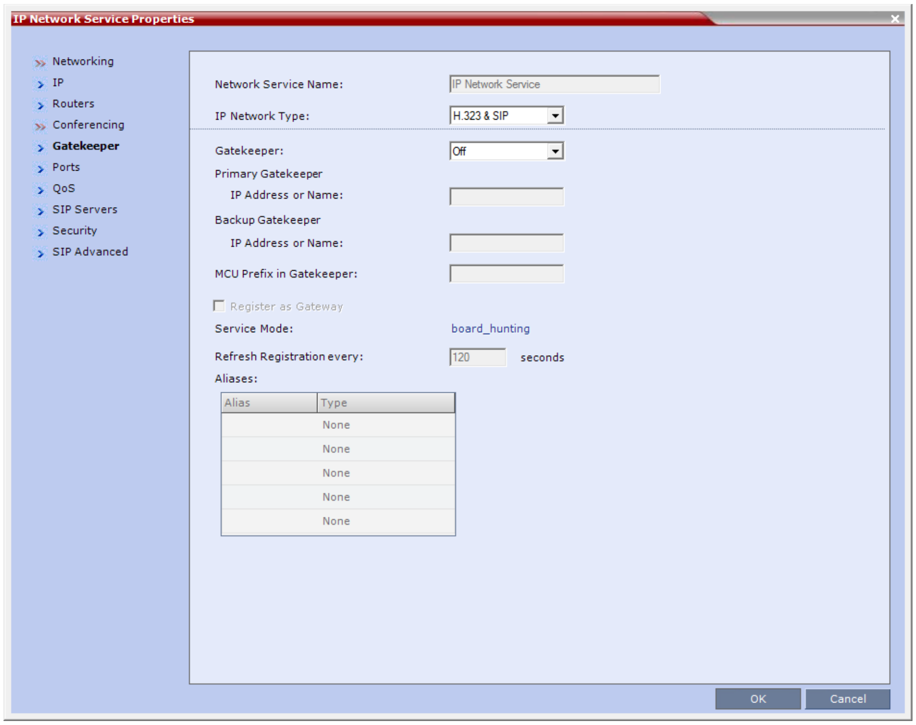
The following fields can be modified, but their values will not be applied:

Table 15-5 Default Management Network Service – DNS

Field	Description
Service Name (FQDN)	The name of the MCU on the network. Default name is PolycomMCU
DNS	• **Off** – if DNS servers are not used in the network. • **Specify** – to enter the IP addresses of the DNS servers. **Note:** The IP address fields are enabled only if **Specify** is selected.
Register Host Names Automatically to DNS Servers	Select this option to automatically register the MCU Signaling Host with the DNS server.
Local Domain Name	Enter the name of the domain where the MCU is installed.
DNS Server Address	The static IP addresses of the DNS server.

If an attempt is made to modify these settings, the MCU will prompt you to reset. Regardless, these settings will not be applied.

6 Click the **Gatekeeper** tab.



Network Service Name: IP Network Service

IP Network Type: H.323 & SIP

Gatekeeper: Off

Primary Gatekeeper

IP Address or Name:

Backup Gatekeeper

IP Address or Name:

MCU Prefix in Gatekeeper:

□ Register as Gateway

Service Mode: board_hunting

Refresh Registration every: 120 seconds

Aliases:

Alias	Type
	None
	None
	None
	None
	None

7 Modify the following fields:

Table 15-6 Default IP Service – Conferencing – Gatekeeper Parameters

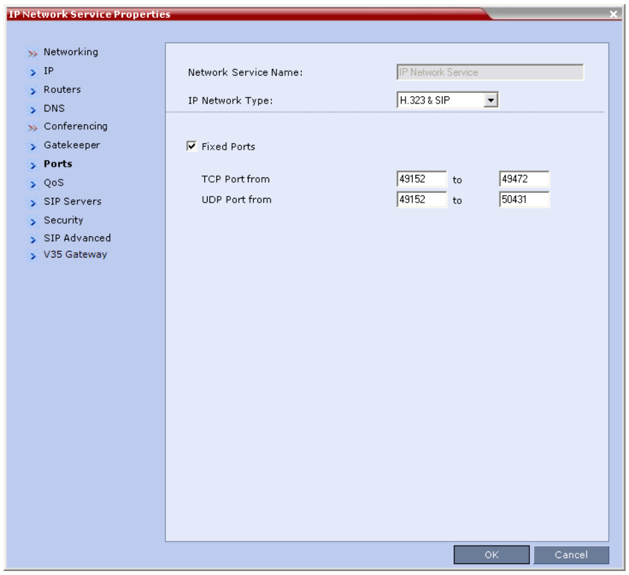
Field	Description
Gatekeeper	Select **Specify** to enable configuration of the gatekeeper IP address. When **Off** is selected, all gatekeeper options are disabled.
Primary Gatekeeper IP Address or Name	Enter either the gatekeeper's host name as registered in the DNS or IP address.
Backup Gatekeeper IP Address or Name	Enter the DNS host name or IP address of the gatekeeper used as a fallback gatekeeper used when the primary gatekeeper is not functioning properly.
MCU Prefix in Gatekeeper	Enter the number with which this Network Service registers in the gatekeeper. This number is used by H.323 endpoints as the first part of their dial-in string when dialing the MCU.
Register as Gateway	Select this check box if the Collaboration Server is to be seen as a gateway, for example, when using a Cisco gatekeeper.

Table 15-6 *Default IP Service – Conferencing – Gatekeeper Parameters (Continued)*

Field	Description
Refresh Registration every __ seconds	The frequency with which the system informs the gatekeeper that it is active by re-sending the IP address and aliases of the system to the gatekeeper. If the system does not register within the defined time interval, the gatekeeper will not refer calls to the system until it re-registers. If set to 0, re-registration is disabled. **Note:** • It is recommended to use default settings. • This is a re-registration and not a 'keep alive' operation – an alternate gatekeeper address may be returned.
Aliases:	
Alias	The alias that identifies the Collaboration Server's Signaling Host within the network. Up to five aliases can be defined for each Collaboration Server. **Note:** When a gatekeeper is specified, at least one alias must be entered in the table. Additional aliases or prefixes may also be entered.
Type	The type defines the format in which the system alias is sent to the gatekeeper. Each alias can be of a different type: • H.323 ID (alphanumeric ID) • E.164 (digits 0-9, * and #) • Email ID (email address format, e.g. abc@example.com) • Participant Number (digits 0-9, * and #) **Note:** Although all types are supported, the type of alias to be used depends on the gatekeeper's capabilities.

8 Click the **Ports** tab.

Settings in the *Ports* tab allow specific ports in the firewall to be allocated to multimedia conference calls.



Network Service Name: IP Network Service

IP Network Type: H.323 & SIP

☑ Fixed Ports

TCP Port from 49152 to 49472

UDP Port from 49152 to 50431

The port range recommended by IANA (Internet Assigned Numbers Authority) is 49152 to 65535. The Collaboration Server uses this recommendation along with the number of licensed ports to calculate the port range.

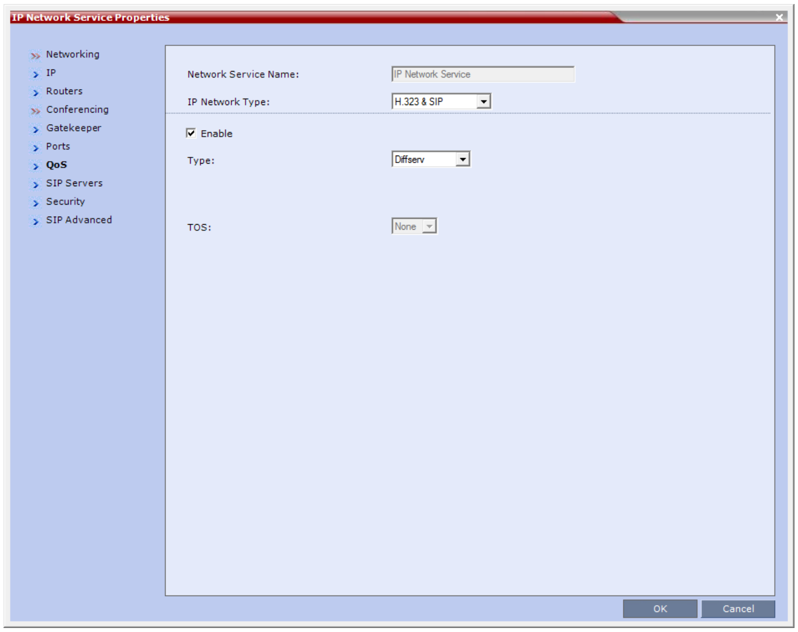
9 Modify the following fields:

Table 15-7 *Default IP Service – Conferencing – Ports Parameters*

Field	Description
Fixed Ports	Leave this check box cleared if you are defining a Network Service for local calls that do not require configuring the firewall to accept calls from external entities. When cleared, the system uses the default port range and allocates 4 RTP and 4 RTCP ports for media channels (Audio, Video, Content and FECC). Click this check box to manually define the port ranges or to limit the number of ports to be left open.
TCP Port from - to	Displays the default settings for port numbers used for signaling and control. To modify the number of TCP ports, enter the first and last port numbers in the range. The number of ports is calculated as follows: Number of simultaneous calls x 2 ports (1 signaling + 1 control).
UDP Port from - to	Displays the default settings for port numbers used for audio and video. To modify the number of UDP ports, enter the first and last port numbers in the range, and the range must be **1024** ports.

If the network administrator does not specify an adequate port range, the system will accept the settings and issue a warning. Calls will be rejected when the Collaboration Server's ports are exceeded.

10 If required, click the **QoS** tab.



Network Service Name: IP Network Service

IP Network Type: H.323 & SIP

☑ Enable

Type: Diffserv

TOS: None

Quality of Service (QoS) is important when transmitting high bandwidth audio and video information. *QoS* can be measured and guaranteed in terms of:

- Average delay between packets
- Variation in delay (jitter)
- Transmission error rate

DiffServ and *Precedence* are the two *QoS* methods supported by the Collaboration Server. These methods differ in the way the packet's priority is encoded in the packet header.

The Collaboration Server's implementation of *QoS* is defined per Network Service, not per endpoint.

> The routers must support QoS in order for IP packets to get higher priority.

11 View or modify the following fields:

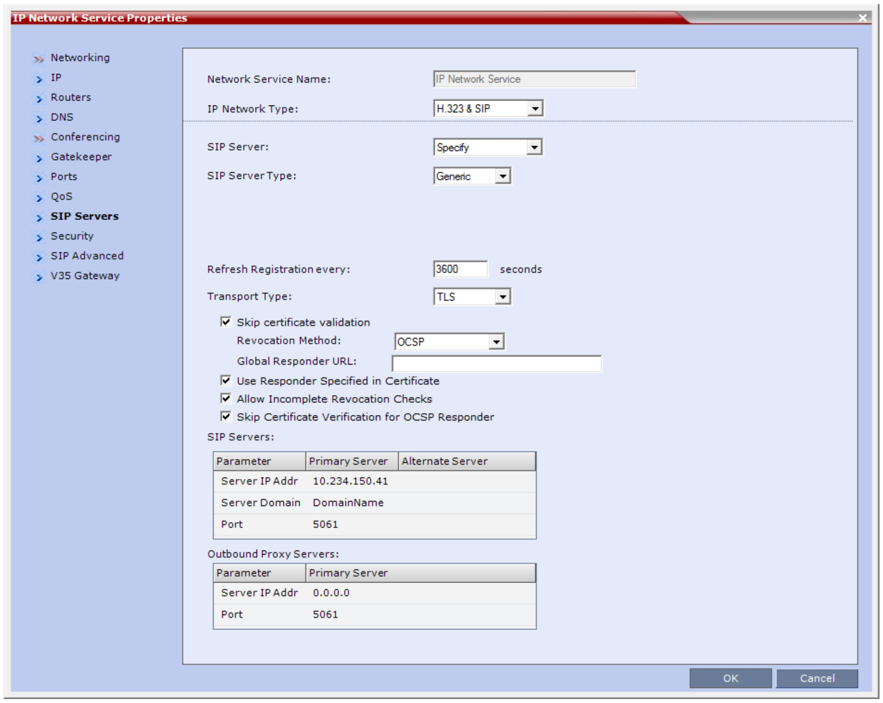
Table 15-8 Default IP Service – Conferencing – QoS Parameters

Field	Description
Enable	Select to enable the configuration and use of the QoS settings. When un-checked, the values of the DSCP (Differentiated Services Code Point) bits in the IP packet headers are zero.

Table 15-8 *Default IP Service – Conferencing – QoS Parameters (Continued)*

Field	Description
Type	DiffServ and Precedence are two methods for encoding packet priority. The priority set here for audio video and IP Signaling packets should match the priority set in the router. • **DiffServ**: Select when the network router uses DiffServ for priority encoding. The default priorities for both audio and video packets is 0x88. These values are determined by the QOS_IP_VIDEO and QOS_IP_AUDIO flags. The default priority for Signaling IP traffic is 0x00 and is determined by the QOS_IP_SIGNALING flag. For more information see "*Modifying System Flags”* on page **20-1**. • **Precedence**: Select when the network router uses Precedence for priority encoding, or when you are not sure which method is used by the router. Precedence should be combined with None in the TOS field. The default priority is 5 for audio and 4 for video packets. **Note**: Precedence is the default mode as it is capable of providing priority services to all types of routers, as well as being currently the most common mechanism.
Audio / Video	You can prioritize audio and video IP packets to ensure that all participants in the conference hear and see each other clearly. Select the desired priority. The scale is from 0 to 5, where 0 is the lowest priority and 5 is the highest. The recommended priority is 4 for audio and 4 for video to ensure that the delay for both packet types is the same and that audio and video packets are synchronized and to ensure lip sync.
TOS	Select the type of Service (TOS) that defines optimization tagging for routing the conferences audio and video packets. • **Delay:** The recommended default for video conferencing; prioritized audio and video packets tagged with this definition are delivered with minimal delay (the throughput of IP packets minimizes the queue sequence and the delay between packets). • **None:** No optimization definition is applied. This is a compatibility mode in which routing is based on Precedence priority settings only. Select None if you do not know which standard your router supports.

12 Click the **SIP Servers** tab.



Network Service Name: IP Network Service

IP Network Type: H.323 & SIP

SIP Server: Specify

SIP Server Type: Generic

Refresh Registration every: 3600 seconds

Transport Type: TLS

☑ Skip certificate validation

Revocation Method: OCSP

Global Responder URL:

☑ Use Responder Specified in Certificate

☑ Allow Incomplete Revocation Checks

☑ Skip Certificate Verification for OCSP Responder

SIP Servers:

Parameter	Primary Server	Alternate Server
Server IP Addr	10.234.150.41	
Server Domain	DomainName	
Port	5061	

Outbound Proxy Servers:

Parameter	Primary Server
Server IP Addr	0.0.0.0
Port	5061

13 Modify the following fields:

Table 15-9 Default IP Network Service – SIP Servers

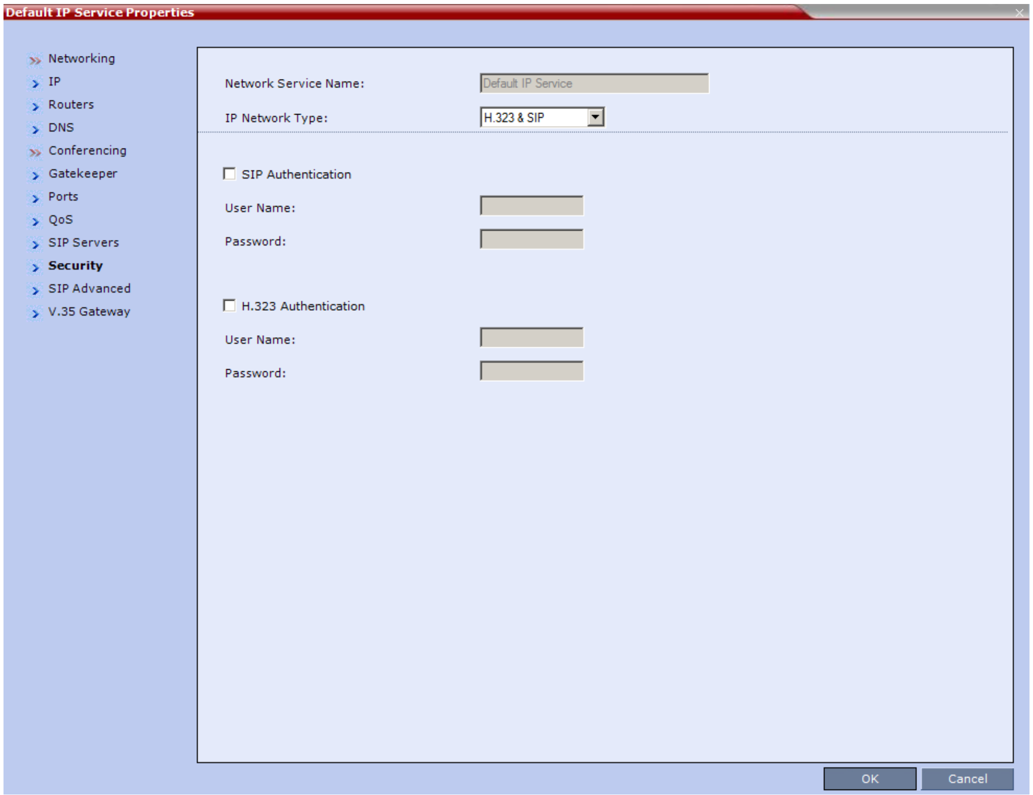
Field	Description
SIP Server	Select: • **Specify** – to manually configure SIP servers. • **Off** – if SIP servers are not present in the network.
SIP Server Type	Select: • Generic - for non Microsoft environments. • Microsoft - for Microsoft environments.
Refresh Registration	This defines the time in seconds, in which the Collaboration Server refreshes it's registration on the SIP server. For example, if "3600" is entered the Collaboration Server will refresh it's registration on the SIP server every 3600 seconds.
Transport Type	Select the protocol that is used for signaling between the Collaboration Server and the SIP Server or the endpoints according to the protocol supported by the SIP Server: **UDP –** Select this option to use UDP for signaling. **TCP –** Select this option to use TCP for signaling. **TLS –** The *Signaling Host* listens on secured port 5061 only and all outgoing connections are established on secured connections. Calls from SIP clients or servers to non secured ports are rejected. The following protocols are supported: TLS 1.0, SSL 2.0 and SSL 3.0.

Table 15-9 *Default IP Network Service – SIP Servers (Continued)*

Field	Description
Skip certificate validation	Not supported in the RealPresence Collaboration Server Virtual Edition.
Revocation Method	The *certificate revocation method*: • ***ONE*** (Default) - *Certificate Revocation* is not implemented. • **CRL** - Requires at least one CRL file be installed, failing which an error message, At least one CRL should be installed, is displayed.
Global Responder URL	Not supported in the RealPresence Collaboration Server Virtual Edition.
Use Responder Specified in Certificate	Not supported in the RealPresence Collaboration Server Virtual Edition.
Allow Incomplete Revocation Checks	• If the check box is checked and the *CRL* of the specific *CA* is not loaded, all *Certificates* are the *CA* are not considered revoked. • If the check box is unchecked and the *CRL* of the specific *CA* is not loaded, all *Certificates* are the *CA* are considered revoked.
Skip Certificate Validation for OSCP Responder	Not supported in the RealPresence Collaboration Server Virtual Edition.
SIP Servers: Primary / Backup Server Parameter	
Server IP Address	Enter the IP address of the preferred SIP server. If DNS is used, you can enter the SIP server name.
Server Domain Name	Enter the name of the domain that you are using for conferences, for example: `user_name@domain name` The domain name is used for identifying the SIP server in the appropriate domain according to the host part in the dialed string. For example, when a call to `EQ1@polycom.com` reaches its outbound proxy, this proxy looks for the SIP server in the `polycom.com` domain, to which it will forward the call. When this call arrives at the SIP server in `polycom.com`, the server looks for the registered user (EQ1) and forwards the call to this Entry Queue or conference.
Port	Enter the number of the TCP or UDP port used for listening. The port number must match the port number configured in the SIP server. Default port is 5060.
Outbound Proxy Servers: Primary / Alternate Server Parameter	
Server IP Address	By default, the Outbound Proxy Server is the same as the SIP Server. If they differ, modify the IP address of the Outbound Proxy and the listening port number (if required). If DNS is used, you can enter the SIP server name.
Port	Enter the port number the outbound proxy is listening to. The default port is 5060.

When updating the parameters of the SIP Server in the *IP Network Service - SIP Servers* dialog box, the Collaboration Server must be reset to implement the change.

14 Click the **Security** tab.



Network Service Name: Default IP Service

IP Network Type: H.323 & SIP

□ SIP Authentication

User Name:

Password:

□ H.323 Authentication

User Name:

Password:

15 Modify the following fields:

Table 15-10 Default IP Network Service – Security (SIP Digest)

Field	Description	
SIP Authentication	Click this check box to enable SIP proxy authentication. Select this check box only if the authentication is enabled on the SIP proxy, to enable the Collaboration Server to register with the SIP proxy. If the authentication is enabled on the SIP proxy and disabled on the RMX, calls will fail to connect to the conferences. Leave this check box cleared if the authentication option is disabled on the SIP proxy.	

Table 15-10 Default IP Network Service – Security (SIP Digest) (Continued)

Field		Description	
	User Name	Enter the user name the Collaboration Server will use to authenticate itself with the SIP proxy. This name must be defined in the SIP proxy.	These fields can contain up to 20 ASCII characters.
	Password	Enter the password the Collaboration Server will use to authenticate itself with the gatekeeper. This password must be defined in the SIP proxy.	
H.323 Authentication		Click this check box to enable H.323 server authentication. Select this check box only if the authentication is enabled on the gatekeeper, to enable the Collaboration Server to register with the gatekeeper. If the authentication is enabled on the gatekeeper and disabled on the RMX, calls will fail to connect to the conferences. Leave this check box cleared if the authentication option is disabled on the gatekeeper.	
	User Name	Enter the user name the Collaboration Server will use to authenticate itself with the gatekeeper. This name must be defined in the gatekeeper.	These fields can contain up to 64 ASCII characters.
	Password	Enter the password the Collaboration Server will use to authenticate itself with the gatekeeper. This password must be defined in the gatekeeper.	

If the *Authentication User Name* and *Authentication Password* fields are left empty, the SIP Digest authentication request is rejected. For registration without authentication, the Collaboration Server must be registered as a trusted entity on the SIP server.

> The fields in the SIP Advanced and V.35 Gateway tabs can be modified but their values will not be applied.

16 Click the **OK** button.

> When updating the parameters of the SIP Server in the *IP Network Service - SIP Servers* dialog box, the Collaboration Server must be reset to implement the change.

IP Network Monitoring

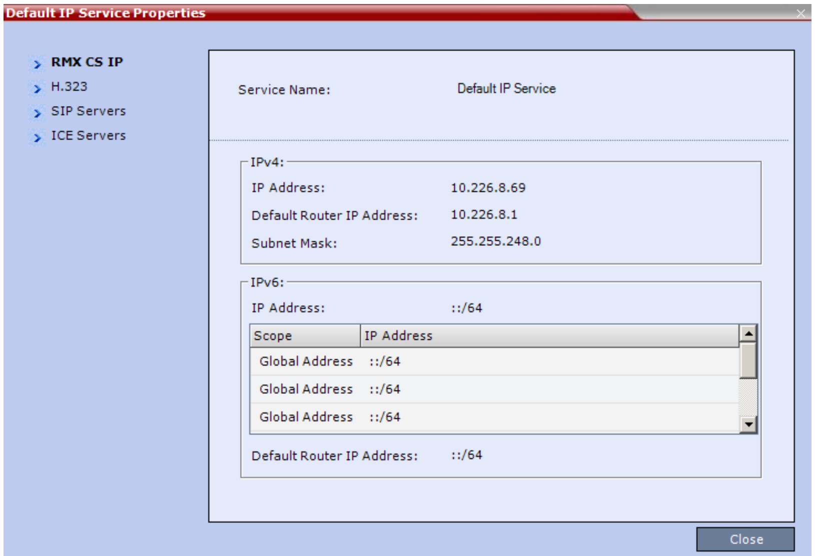
The *Signaling Monitor* is the Collaboration Server entity used for monitoring the status of external network entities such as the gatekeeper, DNS, SIP proxy and Outbound proxy and their interaction with the MCU.

To monitor signaling status:

1 In the *Collaboration Server Management* pane, click **Signaling Monitor** (▭).

2 In the *Signaling Monitor* pane, double-click **Default IP Service**.

The *IP Network Services Properties – RMX CS IP* tab opens:



Default IP Service Properties

- RMX CS IP
- H.323
- SIP Servers
- ICE Servers

Service Name: Default IP Service

IPv4:

IP Address:	10.226.8.69
Default Router IP Address:	10.226.8.1
Subnet Mask:	255.255.248.0

IPv6:

IP Address: ::/64

Scope	IP Address
Global Address	::/64
Global Address	::/64
Global Address	::/64

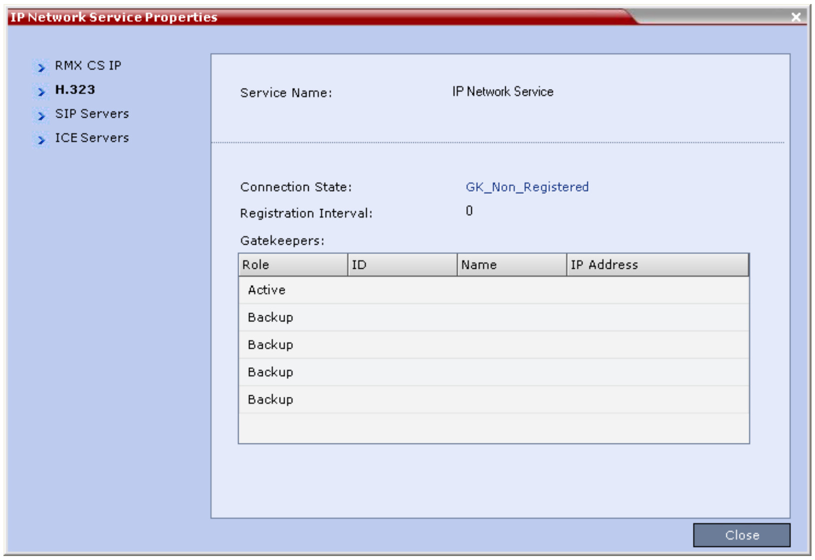
Default Router IP Address: ::/64

The *RMX CS IP* tab displays the following fields:

Table 15-11 IP Network Services Properties – RMX CS IP

<table>
<tr><th>Field</th><th colspan="2">Description</th></tr>
<tr><td><i>Service Name</i></td><td colspan="2">The name assigned to the <i>IP Network Service</i> by the <i>Fast Configuration Wizard</i>.
<b>Note:</b> This field is displayed in all tabs.</td></tr>
<tr><td rowspan="3"><i>IPv4</i></td><td colspan="2">IP Address</td></tr>
<tr><td>Default Router IP Address</td><td>The IP address of the default router. The default router is used whenever the defined static routers are not able to route packets to their destination. The default router is also used when host access is restricted to one default router.</td></tr>
<tr><td>Subnet Mask</td><td>The subnet mask of the MCU.
Default value: 255.255.255.0.</td></tr>
</table>

3 Click the **H.323** tab.



Service Name: IP Network Service

Connection State: GK_Non_Registered

Registration Interval: 0

Gatekeepers:

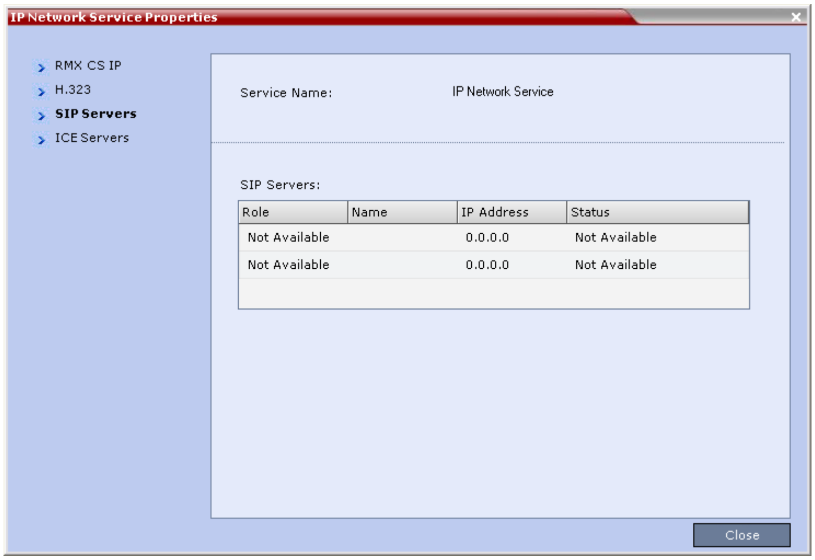
Role	ID	Name	IP Address
Active			
Backup			
Backup			
Backup			
Backup			

The *H.323* tab displays the following fields:

Table 15-12 *IP Network Services Properties – H.323*

Field	Description	
Connection State	The state of the connection between the Signaling Host and the gatekeeper: **Discovery** - The Signaling Host is attempting to locate the gatekeeper. **Registration** - The Signaling Host is in the process of registering with the gatekeeper. **Registered** - The Signaling Host is registered with the gatekeeper. **Not Registered** - The registration of the Signaling Host with the gatekeeper failed.	
Registration Interval	The interval in seconds between the Signaling Host's registration messages to the gatekeeper. This value is taken from either the IP Network Service or from the gatekeeper during registration. The lesser value of the two is chosen.	
	Role	**Active** - The active gatekeeper. **Backup** - The backup gatekeeper that can be used if the connection to the preferred gatekeeper fails.
	ID	The gatekeeper ID retrieved from the gatekeeper during the registration process.
	Name	The gatekeeper's host's name.
	IP Address	The gatekeeper's IP address.

4 Click the **SIP Servers** tab.



Service Name: IP Network Service

SIP Servers:

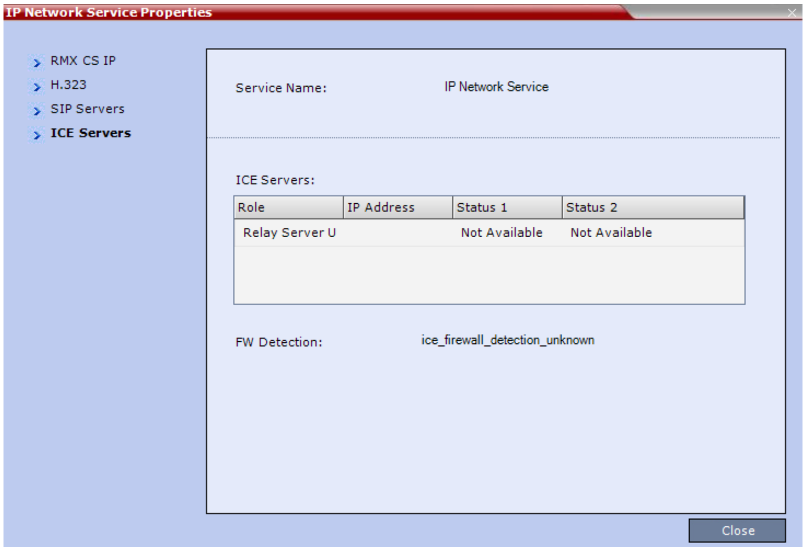
Role	Name	IP Address	Status
Not Available		0.0.0.0	Not Available
Not Available		0.0.0.0	Not Available

The *SIP Servers* tab displays the following fields:

Table 15-13 IP Network Services Properties – SIP Servers

Field	Description
Role	**Active** -The default SIP Server is used for SIP traffic. **Backup** -The SIP Server is used for SIP traffic if the preferred proxy fails.
Name	The name of the SIP Server.
IP Address	The SIP Server's IP address.
Status	The connection state between the SIP Server and the Signaling Host. **Not Available** - No SIP server is available. **Auto** - Gets information from DHCP, if used.

5 Click the **ICE Servers** tab.



Service Name: IP Network Service

ICE Servers:

Role	IP Address	Status 1	Status 2
Relay Server U		Not Available	Not Available

FW Detection: ice_firewall_detection_unknown

ICE is not supported with RealPresence Collaboration Server Virtual Edition.

The *ICE Servers* tab displays the following fields:

***Table 15-14** IP Network Services Properties – ICE Servers*

Field	Description
Role	The ICE Server's role is displayed: • STUN password server • STUN Server UDP • STUN Server TCP • Relay Server UDP • Relay Server TCP
IP Address	The ICE Server's IP Address.
Status 1/2/3/4	A status is displayed for each media card installed in the Collaboration Server: • Connection O.K. • MS – register fail • MS – subscribe fail • MS – service fail • Connection failed • User/password failed • Channel didn't receive any packets for 5 seconds • Channel exceeded allotted bandwidth • Unknown failure

Table 15-14 IP Network Services Properties – ICE Servers (Continued)

Field	Description
FW Detection	The Firewall Detection status is displayed: • Unknown • UDP enabled • TCP enabled • Proxy -TCP is possible only through proxy • Block – both UDP & TCP blocked • None

NAT (Network Address Translation) Traversal

NAT Traversal is a set of techniques enabling participants behind firewalls to connect to conferences, hosted on the Collaboration Server, remotely using the internet.

Session Border Controller (SBC)

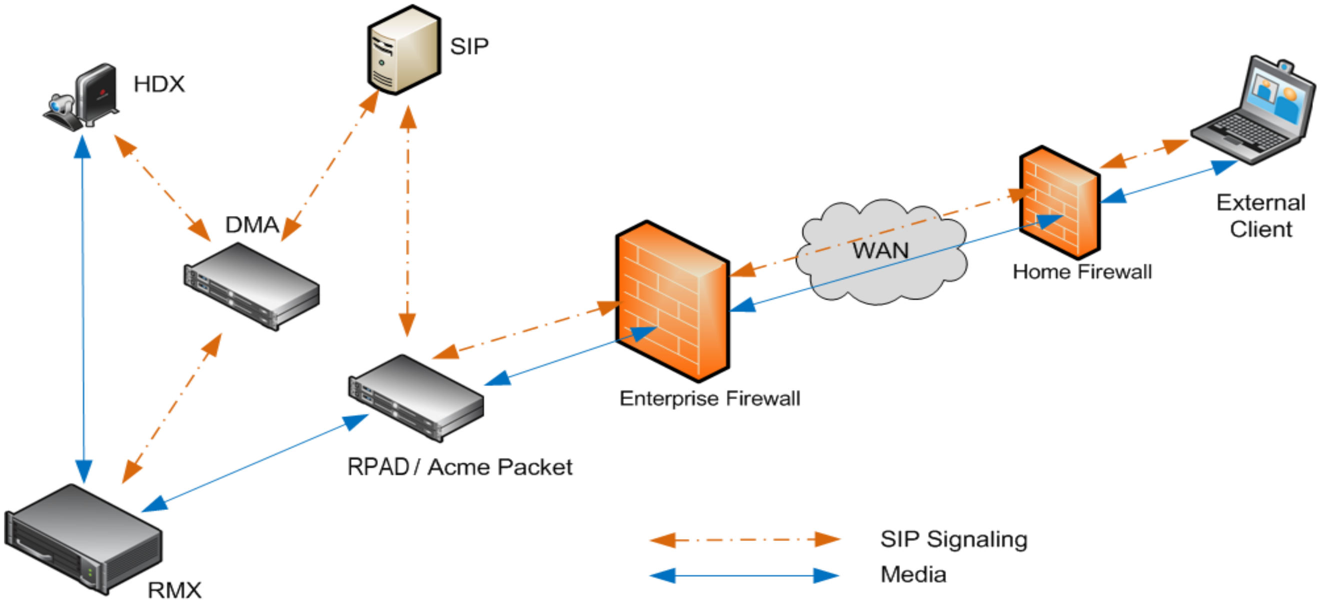
All signaling and media for both SIP and H.323 will be routed through an *SBC*.
The following *SBC* environments are supported:

- *RealPresence Access Director (RPAD)* - a *Polycom SBC*
- *Acme Packet* - a 3rd party *SBC*
- *VBP - Polycom Video Border Proxy*

Deployment Architectures

The following *NAT Traversal* topologies are given as examples. Actual deployments will depend on user requirements and available infrastructure:

Remote Connection Using the Internet



The following *Remote Connection* call flow options are supported:

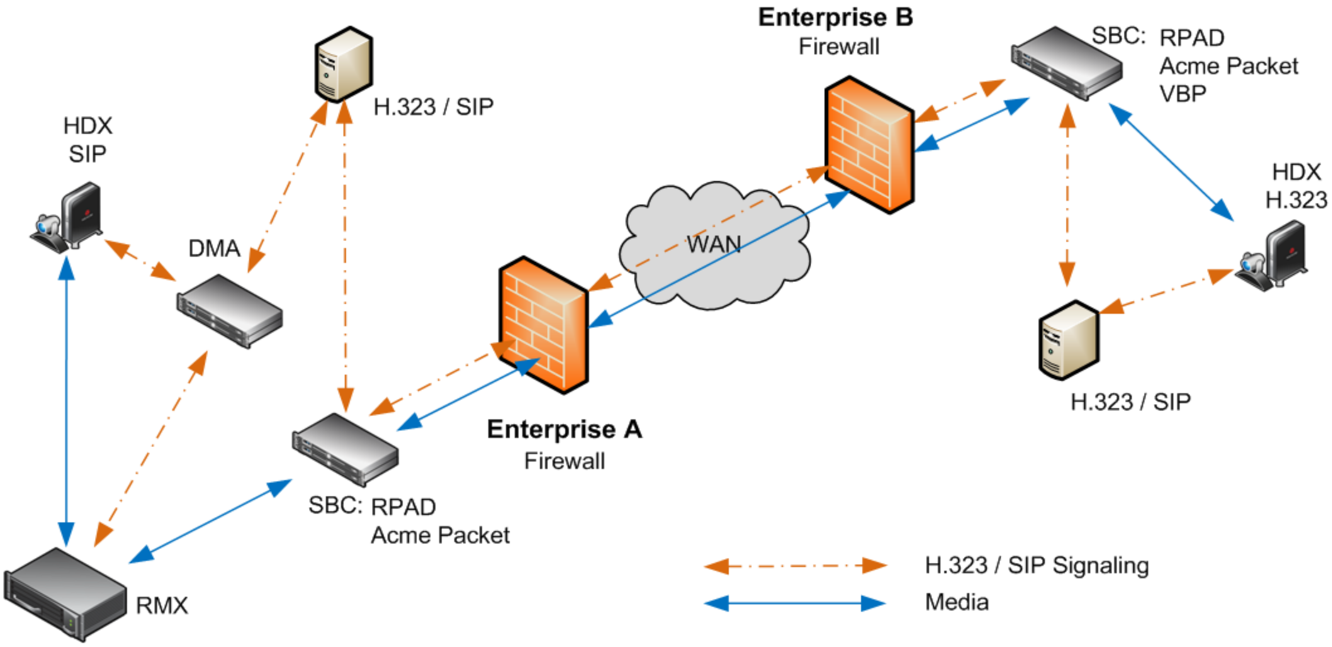
Table 15-15 *Remote Connection*

Enterprise Client				CMA Client	
Environment	**Registered**	**SBC**		**Registered**	**Environment**
SIP / H.323	Yes	SAM / Acme Packet	⇆	Yes	*SIP*
SIP / H.323	No	SAM / Acme Packet	⇆	No	*SIP*

***Table 15-15** Remote Connection (Continued)*

Enterprise Client				CMA Client	
Environment	Registered	SBC		Registered	Environment
SIP / H.323	No	SAM Only	⇆	No	*H.323*

Business to Business Connections



The following *Business to Business* connection call flow options are supported:

***Table 15-16** Business to Business Connection*

Enterprise A Client				Enterprise B Client		
Environment	Registered	SBC		SBC	Registered	Environment
H.323	Yes	RPAD	⇆	*RPAD*	Yes	*H.323*
H.323	Yes	RPAD	⇆	*VBP*	Yes	*H.323*
SIP	Yes	RPAD	⇆	*RPAD*	Yes	*H.323*
SIP	Yes	Acme Packet	⇆	Acme Packet	Yes	*H.323*

FW (Firewall) NAT Keep Alive

The Collaboration Server can be configured to send a *FW NAT keep alive* message at specific *Intervals* for the *RTP, UDP* and *BFCP* channels.

This is necessary because port mappings in the firewall are kept open only if there is network traffic in both directions. The firewall will only allow *UDP* packets into the network through ports that have been used to send packets out.

By default the Collaboration Server sends a *FW NAT Keep Alive* message every **30** seconds. As there is no traffic on the *Content* and *FECC* channels as a call begins, the firewall will not allow any incoming packets from the *Content* and *FECC* channels in until the Collaboration Server sends out the first of the *FW NAT Keep Alive* messages 30 seconds after the call starts.

If *Content* or *FECC* are required within the first 30 seconds of a call the *FW NAT Keep Alive Interval* should be modified to a lower value.

To enable and modify FW NAT Keep Alive:

FW NAT Keep Alive is enabled in the *New Profile - Advanced* dialog box.

>> Select the *FW NAT Keep Alive* check box and if required, modify the *Interval* field within the range of **1 - 86400** seconds.

System Configuration in SBC environments

In an environment that includes *SAM* (a *Polycom SBC*), to ensure that a *RealPresence Mobile* endpoint can send content to a conference the value of the system flag **NUM_OF_INITIATE_HELLO_MESSAGE_IN_CALL_ESTABLISHMENT** must be set to at least 3.

For more details on modifying the values of system flags, see "*Manually Adding and Deleting System Flags"* on page 20-12.

SIP Proxy Failover With Polycom® RealPresence Distributed Media Application™ (DMA™) 7000 System

Collaboration Server systems that are part of a *RealPresence DMA* system environment can benefit from the *RealPresence DMA* system's *SIP Proxy Failover* functionality.

SIP Proxy Failover is supported in the RealPresence DMA system's *Local Clustering* mode with redundancy achieved by configuring two *DMA* servers to share a single virtual *IP* address.

The virtual *IP* address is used by the *Collaboration Server* as the *IP* address of its *SIP Proxy*.

No additional configuration is needed on the *Collaboration Server*.

Should a SIP Proxy failure occur in one of the RealPresence **DMA system servers:**

- The other *RealPresence DMA* system server takes over as *SIP Proxy*.
- Ongoing calls may be disconnected.
- Previously ongoing calls will have to be re-connected using the original *IP* address, registration and connection parameters.

- New calls will connect using the original *IP* address, registration and connection parameters.

RealPresence Collaboration Server Virtual Edition Network Port Usage

The following table summarizes the port numbers and their usage in the RealPresence Collaboration Server Virtual Edition:

Table 15-17 Collaboration Server Network Port Usage Summary

Connection Type	Port Number	Protocol	Description	Configurable
HTTP	8080	TCP	Management between the Collaboration Server and *RealPresence Collaboration Server Web Client*.	No
HTTPS	4433	TCP	Secured Management between the Collaboration Server and *RealPresence Collaboration Server Web Client*.	No
DNS	53	TCP	Domain name server.	Can be disabled in the console Text User Interface.
DHCP	68	TCP	Dynamic Host Configuration Protocol.	Can be disabled in the console Text User Interface.
SSH	22	TCP	Secured shell. It is the Collaboration Server terminal.	Can be disabled in the console Text User Interface.
NTP	123	UDP	Network Time Protocol. Enables access to a time server on the network.	No
H.323 GK RAS	1719	UDP	Gatekeeper RAS messages traffic.	No
H.323 Q.931	1720 - incoming ; 49152-59999 - outgoing	TCP	H.323 Q.931 call signaling. Each outgoing call has a separate port. The port for each outgoing call is allocated dynamically.	Yes - for outgoing calls only. It is configured in the Fixed Ports section of the IP service.

Table 15-17 *Collaboration Server Network Port Usage Summary (Continued)*

Connection Type	Port Number	Protocol	Description	Configurable
H.323 H.245	49152 - 59999	TCP	H.245 control. Each outgoing call has a separate port. The port for each outgoing call is allocated dynamically. It can be avoided by tunneling.	Yes - for outgoing calls only. It is configured in the Fixed Ports section of the IP service.
SIP server	5060 60000	UDP, TCP	Connection to the SIP Server. Sometimes port 60000 is used when the system cannot reuse the TCP port. This port can be set in the Central signaling (CS) configuration file.	Yes - in the IP service.
Alternative SIP server	5060 60000	UDP, TCP	Connection to the alternate SIP Server. Sometimes port 60000 is used when the system cannot reuse the TCP port. This port can be set in the Central signaling (CS) configuration file.	Yes - in the IP service.
SIP Outbound proxy	5060 60000	UDP, TCP	Connection to the SIP outbound proxy. Sometimes port 60000 is used when the system cannot reuse the TCP port. This port can be set in the Central signaling (CS) configuration file.	Yes - in the IP service.
Alternative SIP Outbound proxy	5060 60000	UDP, TCP	Connection to the alternate SIP outbound proxy. Sometimes port 60000 is used when the system cannot reuse the TCP port. This port can be set in the Central signaling (CS) configuration file.	Yes - in the IP service.
SIP-TLS	60002	TCP	Required for Binary Floor Control Protocol (BFCP) functionality for SIP People+Content content sharing.	No - port is not opened if SIP People+Content is disabled.
RTP	49152 - 59999	UDP	RTP media packets. The ports are dynamically allocated.	Yes - It is configured in the Fixed Ports section of the IP service.
RTCP	49152 - 59999	UDP	RTP control. The ports are dynamically allocated.	Yes - It is configured in the Fixed Ports section of the IP service.

Table 15-17 *Collaboration Server Network Port Usage Summary (Continued)*

Connection Type	Port Number	Protocol	Description	Configurable
SIP -TLS	5061	TCP	SIP -TLS for SIP server, alternate SIP server, outbound proxy and alternate outbound proxy.	No

16

IVR Services

Interactive Voice Response (IVR) is an application that allows participants to communicate with the conferencing system via their endpoint's input device (such as a remote control). The IVR Service includes a set of voice prompts and a video slide used to automate the participants connection to a conference or Entry Queue. It allows customization of menu driven scripts and voice prompts to meet different needs and languages.

The IVR module includes two types of services:

- Conference IVR Service that is used with conferences
- Entry Queue IVR Service that is used with Entry Queues

The system is shipped with two default Conference IVR Services (one for the conferences and the other for gateway calls) and one default Entry Queue IVR Service. The default services include voice messages and video slides in English.

To customize the IVR messages and video slide perform the following operations:

- Record the required voice messages and create a new video slide. For more information, see "*Creating a Welcome Video Slide"* on page **16-30**.
- Optional. Add the language to the list of languages supported by the system.
- Upload the voice messages to the MCU (This can be done as part of the language definition or during the IVR Service definition).
- Create the Conference IVR Service and upload the video slide, and if required any additional voice messages.
- Optional. Create the Entry Queue IVR Service and upload the required video slide and voice messages.

IVR Services List

You can view the currently defined Conference IVR and Entry Queue IVR Services in the *IVR Services* list pane.

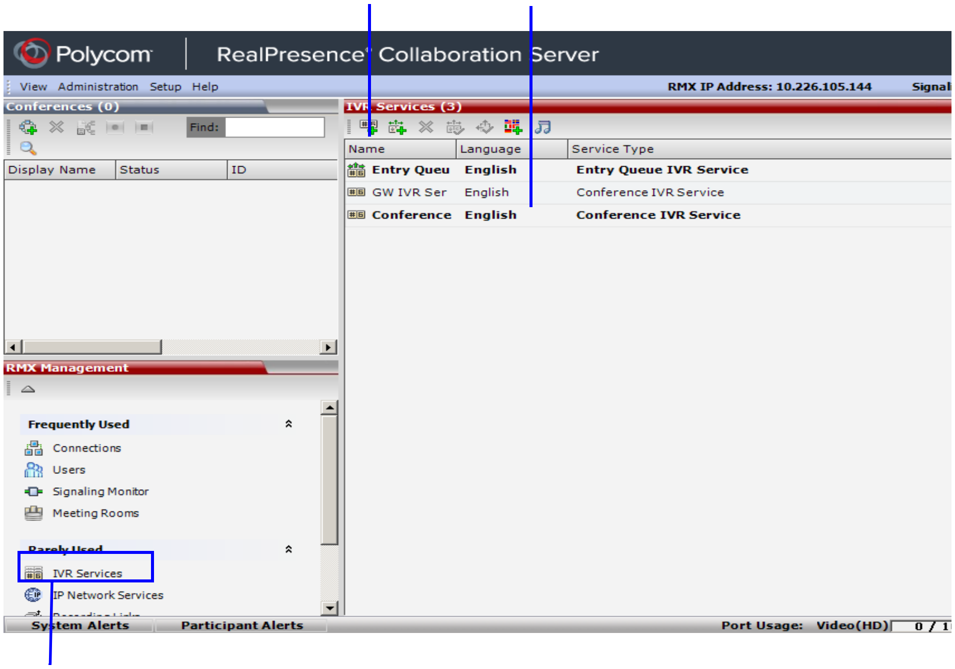
To view the IVR Services list:

1 In the *Collaboration Server Management* pane, expand the *Rarely Used* list.

2 Click the **IVR Services** (#) entry.

The list pane displays the *Conference IVR Services* list and the total number of IVR services currently defined in the system.

IVR Toolbar *IVR Services List Pane*



Name	Language	Service Type
Entry Queu	English	Entry Queue IVR Service
GW IVR Ser	English	Conference IVR Service
Conference	English	Conference IVR Service

Access to IVR Services list and customization

IVR Services Toolbar

The IVR Services toolbar provides quick access to the IVR Service definitions as follows:

Table 16-1 *IVR Toolbar buttons*

Button	Button Name	Descriptions
✚	*New Conference IVR Service*	To create a new Conference IVR Service.
✚	*New Entry Queue IVR Service*	To create a new Entry Queue IVR Service.
✕	*Delete Service*	Deletes the selected IVR service(s).
✓	*Set Default Conference IVR Service*	Sets the selected Conference IVR Service as default. When creating a new conference Profile the default IVR Service is automatically selected for the Profile (but can be modified).
✓	*Set Default Entry Queue Service*	Sets the selected Entry Queue IVR Service as default. When creating a new Entry Queue the default Entry Queue IVR Service is automatically selected.

Table 16-1 IVR Toolbar buttons

Button	Button Name	Descriptions
	Add Supported Languages	Adds languages to the IVR module, enabling you to download voice prompts and messages for various languages.
♫	*Replace/Change Music File*	To replace the currently loaded music file that is used to play background music, the MCU is shipped with a default music file.

Adding Languages

You can define different sets of audio prompts in different languages, allowing the participants to hear the messages in their preferred language.

The *Collaboration Server* is shipped with a default language (English) and all the prompts and messages required for the default IVR Services, conference and Entry Queues shipped with the system.

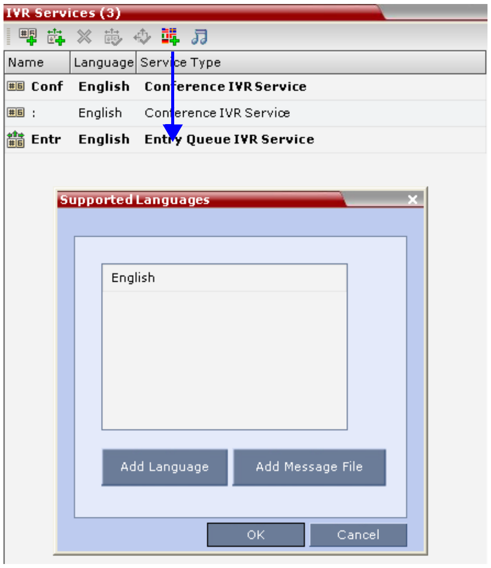
You can add languages to the list of languages for which different messages are downloaded to the MCU and IVR Services are created. This step is required before the creation of additional IVR messages using languages that are different from English, or if you want to download additional voice files to existing files in one operation and not during the IVR service definition.

To add a language:

1 In the *Collaboration Server Management* pane, expand the **Rarely Used** list.

2 Click the **IVR Services** () entry.

3 In the *Conference IVR Services* list, click the **Add Supported Languages** () button. The *Supported Languages* dialog box opens.



IVR Services (3)

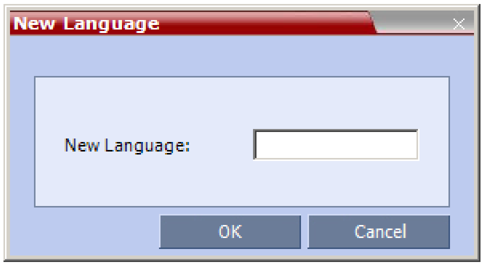
Name	Language	Service Type
Conf	English	Conference IVR Service
:	English	Conference IVR Service
Entr	English	Entry Queue IVR Service

Supported Languages

English

4 Click the **Add Language** button.

The *New Language* dialog box opens.



5 In the *New Language* box, enter the name of the new language. The language name can be typed in Unicode and cannot start with a digit. Maximum field length is 31 characters.

6 Click **OK**.
The new language is added to the list of *Supported Languages*.

Uploading a Message File to the Collaboration Server

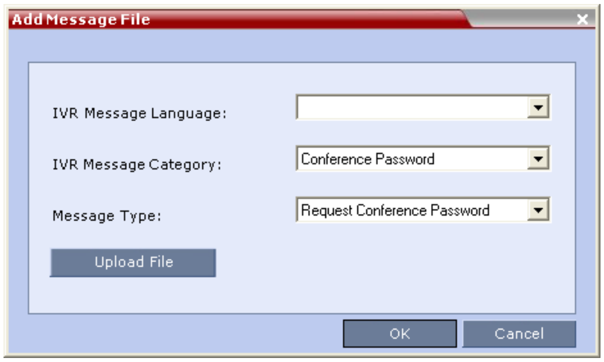
You can upload audio files for the new language or additional files for an existing language now, or you can do it during the definition of the IVR Service. In the latter case, you can skip the next steps.

- Voice messages should not exceed 3 minutes.
- It is not recommended to upload more than 1000 audio files to the MCU memory.

To upload messages to the MCU:

1 To upload the files to the MCU, in the *Supported Languages* dialog box, click the **Add Message File** button.

2 The *Add Message File* dialog box opens.



IVR Message Language:

IVR Message Category: Conference Password

Message Type: Request Conference Password

Audio files are uploaded to the MCU one-by-one.

3 In the *IVR Message Language* list, select the language for which the audio file will be uploaded to the MCU.

4 In the *IVR Message Category* list, select the category for which the audio file is uploaded.

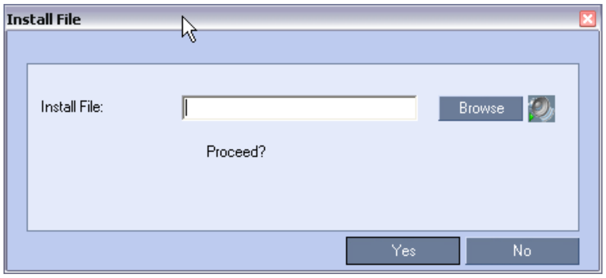
5 In the *Message Type* list, select the message type for which the uploaded message is to be played. You can upload several audio files for each Message Type. Each file is downloaded separately.

Table 16-2 lists the Message Types for each category:

Table 16-2 *IVR Message Types by Message Category*

Message Category	Message Type	Message
Conference Password	Request Conference Password	Requests the participant to enter the conference password.
	Request Conference Password Retry	A participant who enters an incorrect password is requested to enter it again.
	Request Digit	Requests the participant to enter any digit in order to connect to the conference. Used for dial-out participants to avoid answering machines in the conference.
Welcome Message	Welcome Message	The first message played when the participant connects to the conference or Entry Queue.
Conference Chairperson	Request Chairperson Identifier	Requests the participants to enter the chairperson identifier key.
	Request Chairperson Password	Requests the participant to enter the chairperson password.
	Request Chairperson Password Retry	When the participant enters an incorrect chairperson password, requests the participant to enter it again.
General	Messages played for system related event notifications, for example, notification that the conference is locked. Upload the files for the voice messages that are played when an event occurs during the conference. For more information, see "*Conference IVR Service Properties - General Voice Messages”* on page **16-11**.	
Billing Code	Requests the chairperson to enter the conference Billing Code.	
Roll Call	Not applicable.	
Conference ID	Requests the participant to enter the required Conference ID to be routed to the destination conference.	

6 Click **Upload File** to upload the appropriate audio file to the MCU.
The *Install File* dialog box opens.



7 Enter the file name or click the **Browse** button to select the audio file to upload.
The *Select Source File* dialog box opens.

8 Select the appropriate *.wav audio file, and then click the **Open** button.
The name of the selected file is displayed in the *Install* field in the *Install File* dialog box.

9 Optional. You can play a .wav file by selecting the *Play* button ( ).

10 Click **Yes** to upload the file to the MCU.
The system returns to the *Add Message File* dialog box.

11 Repeat step 6 to 10 for each additional audio file to be uploaded to the MCU.

12 Once all the audio files are uploaded to the MCU, close the *Add Message File* dialog box and return to the *Add Language* dialog box.

13 Click **OK**.

Defining a New Conference IVR Service

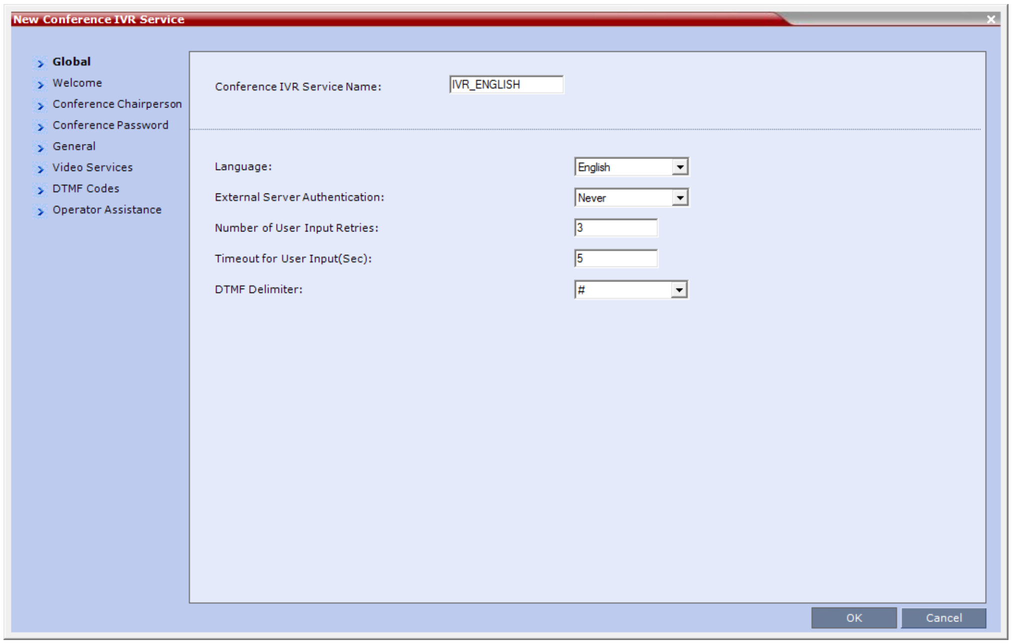
The *Collaboration Server* is shipped with two default Conference IVR Services and all its audio messages and video slide. You can define new Conference IVR Services or modify the default Conference IVR Service.

> Up to 40 IVR Services (Conference IVR Services and Entry Queue IVR Services) can be defined for a single *Collaboration Server* unit.

Defining a New Conference IVR Service

To define a new Conference IVR Service:

1 On the *IVR Services* toolbar, click the **New Conference IVR Service** ( ) button.
The *New Conference IVR Service - Global* dialog box opens.



Conference IVR Service Name: IVR_ENGLISH

Language: English

External Server Authentication: Never

Number of User Input Retries: 3

Timeout for User Input(Sec): 5

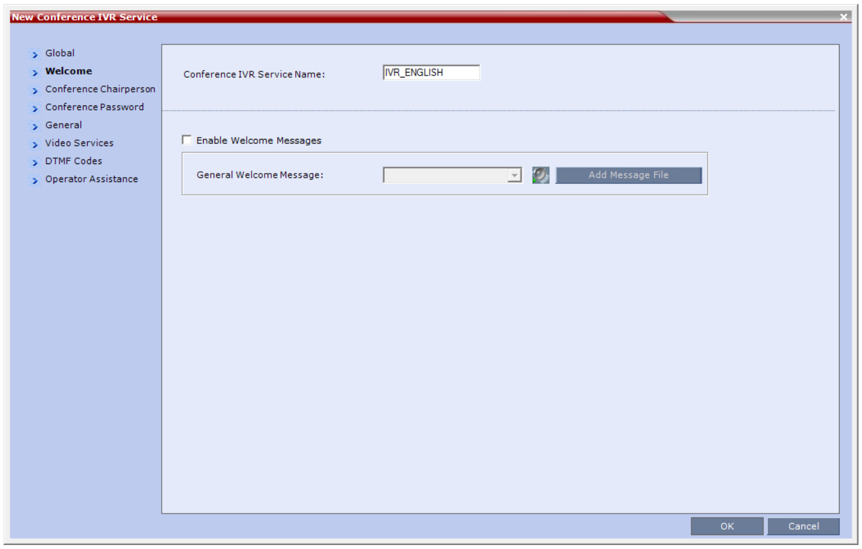
DTMF Delimiter: #

2 Define the following parameters:

Table 16-3 Conference IVR Service Properties - Global Parameters

Field/Option	Description
Conference IVR Service Name	Enter the name of the Conference IVR Service. The maximum field length is 20 characters and may be typed in Unicode.
Language For IVR	Select the language of the audio messages and prompts from the list of languages defined in the *Supported languages*. The default language is English. For more information, see "*Adding Languages”* on page 16-3.
External Server Authentication	
Number of User Input Retries	Enter the number of times the participant will be able to respond to each menu prompt before being disconnected from the conference. Range is between 1-4, and the default is 3.
Timeout for User Input (Sec)	Enter the duration in seconds that the system will wait for the participant’s input before prompting for another input. Range is between 1-10, and the default value is 5 seconds.
DTMF Delimiter	Enter the key that indicates the last input key. Possible values are the pound (#) and star (*) keys. The default is #.

3 Click the **Welcome** tab.
The *New Conference IVR Service - Welcome* dialog box opens.



Conference IVR Service Name: IVR_ENGLISH

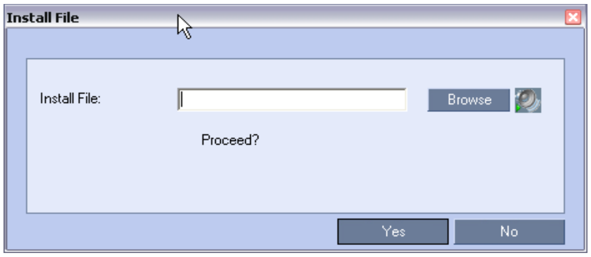
□ Enable Welcome Messages

General Welcome Message:

4 Select the **Enable Welcome Messages** check box to define the system behavior when the participant enters the Conference IVR queue. When participants access a conference through an Entry Queue, they hear messages included in both the Entry Queue Service and Conference IVR Service. To avoid playing the Welcome Message twice, disable the Welcome Message in the Conference IVR Service.

5 Select the **General Welcome Message,** to be played when the participant enters the conference IVR queue.

6 To upload an audio file for an IVR message, click **Add Message File**.
The *Install File* dialog box opens.



Install File

Install File:

Proceed?

The *Collaboration Server* unit is bundled with default audio IVR message files. To upload a customized audio file, see *"Creating Audio Prompts and Video Slides"* on page 16-27.

a Click the **Browse** button to select the audio file (*.wav) to upload.
The *Select Source File* dialog box opens.

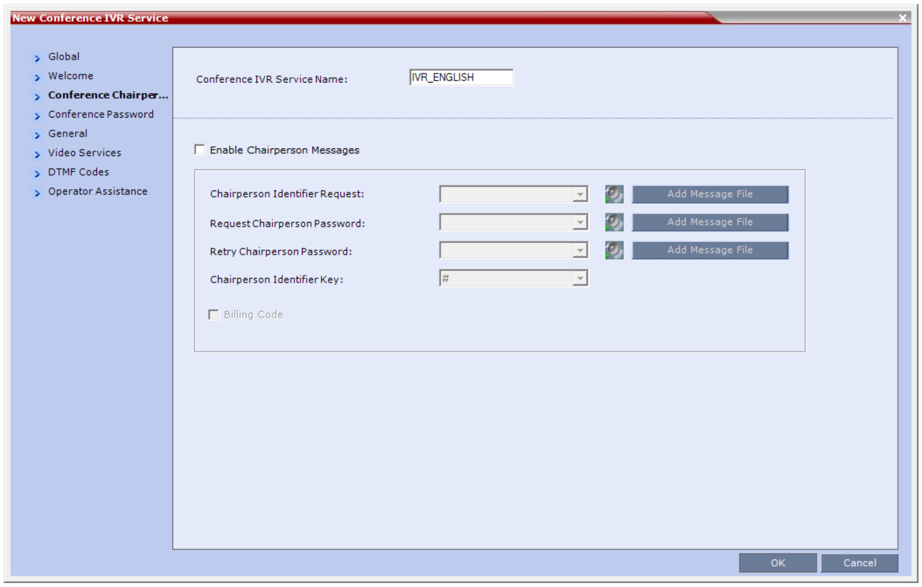
b Select the appropriate *.wav audio file and then click the **Open** button.

c Optional. You can play a .wav file by selecting the *Play* button (🔊).

d In the *Install File* dialog box, click **Yes** to upload the file to the MCU memory.
The *Done* dialog box opens.

e Once the upload is complete, click **OK** and return to the *IVR* dialog box. The new audio file can now be selected from the list of audio messages.

7 Click the **Conference Chairperson** tab.
The *New Conference IVR Service - Conference Chairperson* dialog box opens.



New Conference IVR Service

Conference IVR Service Name: IVR_ENGLISH

□ Enable Chairperson Messages

Chairperson Identifier Request:

Request Chairperson Password:

Retry Chairperson Password:

Chairperson Identifier Key: #

□ Billing Code

8 Select the **Enable Chairperson Messages** check box to enable the chairperson functionality. If this feature is disabled, participants are not able to connect as the chairperson.

When both Conference Password and Chairperson Password options are enabled and defined, the system first plays the prompt "Enter conference password". However, if the participant enters the chairperson password, the participant becomes the chairperson.
To play the prompt requesting the Chairperson password, "For conference chairperson services...", **do not select** the *Enable Password Messages* option.

9 Select the various voice messages and options for the chairperson connection.

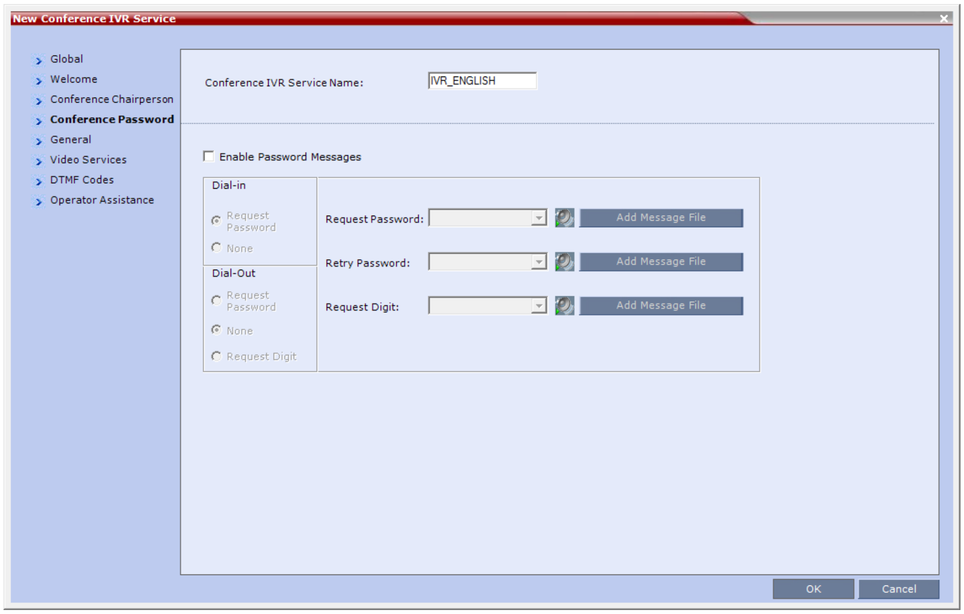
If the files were not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the *Collaboration Server*.

Table 16-4 New Conference IVR Service Properties - Conference Chairperson Options and Messages

Field/Option	Description
Chairperson Identifier Request	Select the audio file that requests the participants to enter the key that identifies them as the conference chairperson.
Request Chairperson Password	Select the audio file that prompts the participant for the chairperson password.
Retry Chairperson Password	Select the audio file that prompts participants to re-enter the chairperson password if they enter it incorrectly.
Chairperson Identifier Key	Enter the key to be used for identifying the participant as a chairperson. Possible keys are: pound key (#) or star (*).
Billing Code	The prompt requesting the chairperson billing code selected in the General tab.

10 Click the **Conference Password** tab.

The *New Conference IVR Service - Conference Password* dialog box opens.



Conference IVR Service Name: IVR_ENGLISH

☐ Enable Password Messages

Dial-in
● Request Password
○ None

Dial-Out
○ Request Password
● None
○ Request Digit

11 Select the **Enable Password Messages** check box to request the conference password before moving the participant from the conference IVR queue to the conference.

> When both Conference Password and Chairperson Password are enabled and defined, the system first plays the prompt "Enter conference password". However, if the participant enters the chairperson password, the participant becomes the chairperson.
> To play the prompt requesting the Chairperson password, "For conference chairperson services...", **do not select** the *Enable Password Messages* option.

12 Select the MCU behavior for password request for *Dial-in* and *Dial-out* participant connections.

Select the required system behavior as follows:

- **Request password** - The system requests the participant to enter the conference password.
- **None** - The participant is moved to the conference without any password request.
- **Request Digit** - The system requests the participant to enter any key. This option is used mainly for dial-out participants and to prevent an answering machine from entering the conference.

13 Select the various audio messages that will be played in each case.

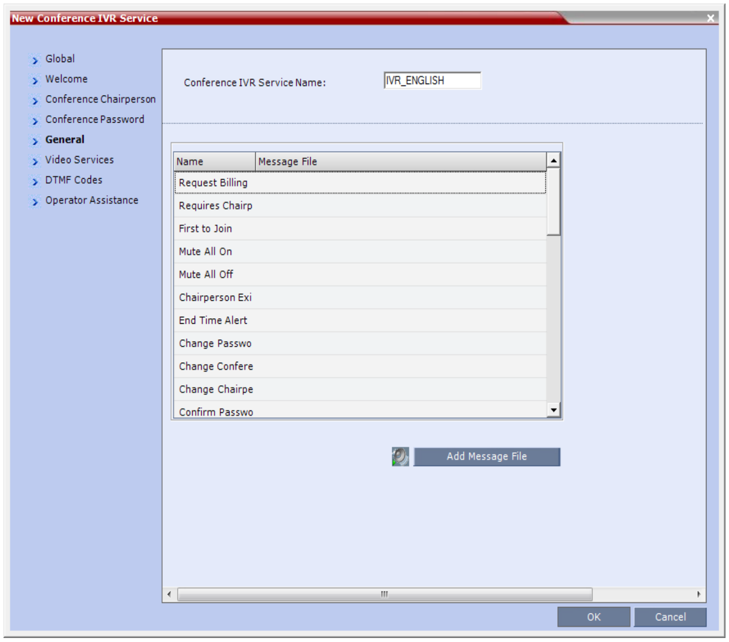
Table 16-5 New Conference IVR Service Properties - Conference Password Parameters

Option	Description
Request Password	Select the audio file that prompts the participant for the conference password.
Retry Password	Select the audio file that requests the participant to enter the conference password again when failing to enter the correct password.

Table 16-5 *New Conference IVR Service Properties - Conference Password Parameters*

Option	Description
Request Digit	Select the audio file that prompts the participant to press any key when the *Request Digit* option is selected.

14 Click the **General** tab.
The *New Conference IVR Service - General* dialog box opens.



Conference IVR Service Name: IVR_ENGLISH

Name	Message File
Request Billing	
Requires Chairp	
First to Join	
Mute All On	
Mute All Off	
Chairperson Exi	
End Time Alert	
Change Passwo	
Change Confere	
Change Chairpe	
Confirm Passwo	

The *General* dialog box lists messages that are played during the conference. These messages are played when participants or the conference chairperson perform various operations or when a change occurs.

15 To assign the appropriate audio file to the message type, click the appropriate table entry, in the *Message File* column. A drop-down list is enabled.

16 From the list, select the audio file to be assigned to the event/indication.

17 Repeat steps 15 and 16 to select the audio files for the required messages.
The following types of messages and prompts can be enabled:

Table 16-6 *Conference IVR Service Properties - General Voice Messages*

Message Type	Description
Blip on Cascade Link	Indicates that the link to the cascaded conference connected successfully.
Chairperson Exit	Informs all the conference participants that the chairperson has left the conference, causing the conference to automatically terminate after a short interval. **Note:** This message is played only when the *Requires Chairperson* option is selected in the *Conference Profile - IVR* dialog box.

Table 16-6 *Conference IVR Service Properties - General Voice Messages (Continued)*

Message Type	Description
Chairperson Help Menu	A voice menu is played upon a request from the chairperson, listing the operations and their respective DTMF codes that can be performed by the chairperson. The playback can be stopped any time. **Note:** If you modify the default DTMF codes used to perform various operations, the default voice files for the help menus must be replaced.
Change Chairperson Password	Requests the participant to enter a new chairperson password when the participant is attempting to modify the chairperson password.
Change Conference Password	Requests the participant to enter a new conference password when the participant is attempting to modify the conference password.
Change Password Failure	A message played when the participant enters an invalid password, for example when a password is already in use.
Change Passwords Menu	This voice menu is played when the participants requests to change the conference password. This message details the steps required to complete the procedure.
Conference is Locked	This message is played to participants attempting to join a Secured conference.
Conference is Secured	This message is played when the conference status changes to Secure as initiated by the conference chairperson or participant (using DTMF code *71).
Conference is unsecured	This message is played when the conference status changes to Unsecured as initiated by the conference chairperson or participant (using DTMF code #71).
Confirm Password Change	Requests the participant to re-enter the new password.
Enter Destination ID	Prompts the calling participant for the destination number. Default message prompts the participant for the conference ID (same message as in the Entry Queue IVR Service). **Note:** This option is not available in SVC conferences and for SVC participants in mixed CP and SVC conferences.
First to Join	Informs the participant that he or she is the first person to join the conference.
Incorrect Destination ID	If the participant entered an incorrect conference ID (in gateway calls it is the destination number), requests the participant to enter the number again. **Note:** This option is not available in SVC conferences and for SVC participants in mixed CP and SVC conferences.
Maximum Number of Participants Exceeded	Indicates the participant cannot join the destination conference as the maximum allowed number of participants will be exceeded.

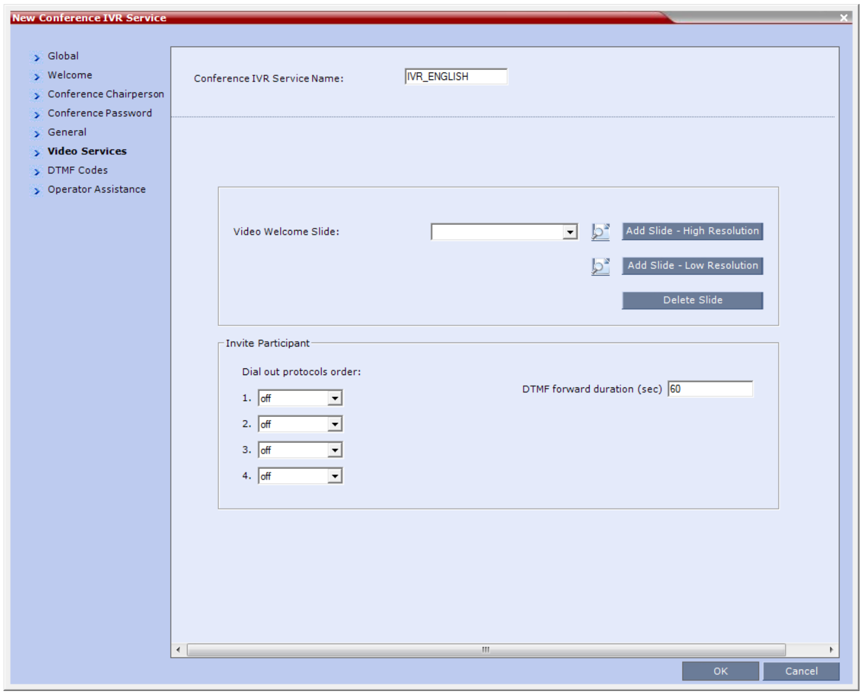
Table 16-6 *Conference IVR Service Properties - General Voice Messages (Continued)*

Message Type	Description
Mute All Off	This message is played to the conference to inform all participants that they are unmuted (when *Mute All* is cancelled).
Mute All On	Informs all participants that they are muted, with the exception of the conference chairperson. **Note:** This message is played only when the *Mute All Except Me* option is activated.
No Video Resources Audio Only.	Informs the participant of the lack of Video Resources in the *Collaboration Server* and that he/she is being connected as Audio Only.
Participant Help Menu	A voice menu that is played upon request from a participant, listing the operations and their DTMF codes that can be performed by any participant.
Password Changed Successfully	A message is played when the password was successfully changed.
Recording Failed	This message is played when the conference recording initiated by the chairperson or the participant (depending on the configuration) fails to start.
Recording in Progress	This message is played to participant joining a conference that is being recorded indicating the recording status of the conference.
Request Billing Code	Requests the participant to enter a code for billing purposes.
Requires Chairperson	The message is played when the conference is on hold and the chairperson joins the conference. For this message to be played the *Conference Requires Chairperson* option must be selected in the *Conference Profile - IVR* dialog box.
Self Mute	A confirmation message that is played when participants request to mute their line.
Self Unmute	A confirmation message that is played when participants request to unmute their line.

18 Click the **Video Services** tab.

The Roll Call and Tone Notification options are disabled in SVC and mixed CP and SVC conferences.

The *New Conference IVR Service - Video Services* dialog box opens.



Conference IVR Service Name: IVR_ENGLISH

Video Welcome Slide:

Invite Participant

Dial out protocols order:

1. off
2. off
3. off
4. off

DTMF forward duration (sec) 60

The Click&View and Invite Participants features are disabled in SVC and mixed CP and SVC conferences.

In addition to the low and high resolution slides included in the default slide set, customized low and high resolution slides are supported.

The following guidelines apply:

— Two customized slides can be loaded per *IVR Service*:

- A low resolution slide, to be used with low resolution endpoints.
- A high resolution slide, to be used with high resolution endpoints.

Table 16-7 summarizes the recommended input slide formats and the resulting slides that are generated:

Table 16-7 *IVR Slide - Input / Output Formats*

Slide Resolution	Format	
	Input Slides	Generated Slides
High	HD720p (16:9)	HD720p
Low	4CIF (4:3) or CIF (4:3)	4SIF SIF CIF

— The source images for the high resolution slides must be in ****.bmp*** or ****.jpg*** format.

- If the uploaded slides are not of the exact *SD* or *HD* resolution, an error message is displayed and the slides are automatically cropped or enlarged to the right size.
- If a slide that is selected in an *IVR Service* is deleted, a warning is displayed listing the *IVR Services* in which it is selected. If deleted, it will be replaced with a default *Collaboration Server* slide.
- The generated slides are not deleted if the system is downgraded to a lower software version.
- The first custom source file uploaded, whatever its format, is used to generate both high and low resolution custom slides. High resolution source files uploaded after the first upload will be used to generate and replace high resolution custom slides. Likewise, low resolution source files uploaded after the first upload will be used to generate and replace low resolution custom slides.
- If there are two custom source files in the folder, one high resolution, one low resolution, and a new high resolution custom source file is uploaded, new high resolution custom slides are created. The existing low resolution custom slides are not deleted.
- If there are two custom source files in the folder, one high resolution, one low resolution, and a new low resolution custom source file is uploaded, new low resolution custom slides are created. The existing high resolution custom slides are not deleted.

19 Define the following parameters:

Table 16-8 *New Conference IVR Service Properties - Video Services Parameters*

Video Services	Description
Video Welcome Slide	Select the *Low Resolution* and *High Resolution* video slides to be displayed when participants connect to the conference. To view any slide, click the **Preview Slide** (🔍) button. **Notes:** • When using one of the default Polycom slides, the slide will be displayed in the resolution defined in the profile, i.e. CIF, SD, HD 720p • Customized H.261 slides are not supported.
Invite Participant	Not applicable. **Note:** The Invite Participant feature is not available in SVC conferences and for SVC participants in mixed CP and SVC conferences.

20 If the video slide file was not uploaded to the MCU prior to the IVR Service definition, click the:

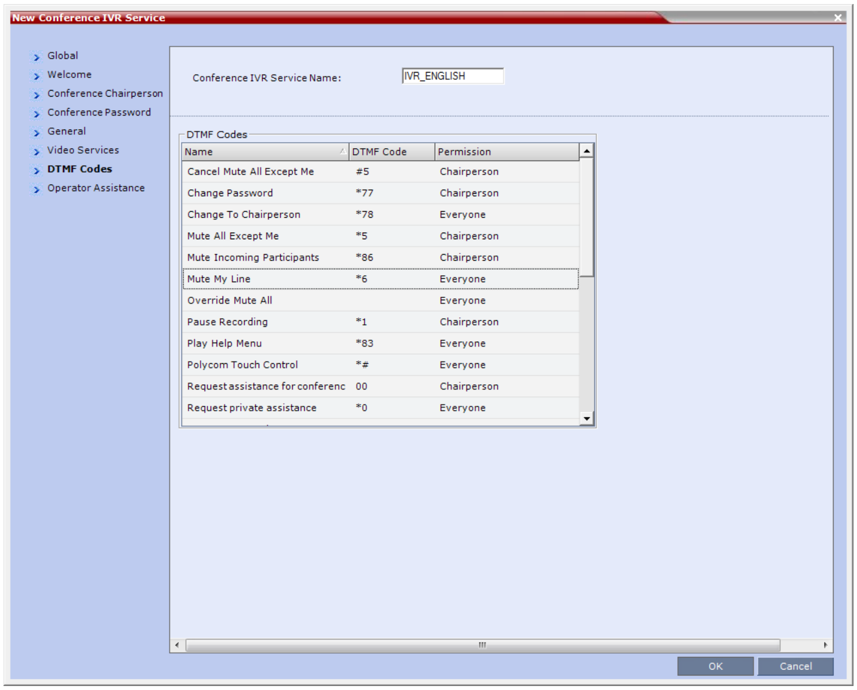
- **Add Slide - Low Resolution** button to upload a *Low Resolution Slide*.
- **Add Slide - High Resolution** button to upload a *High Resolution Slide*.

The *Install File* dialog box opens. The uploading process is similar to the uploading of audio files. For more information, see step **6** on page 16-8.

- The video slide must be in a .jpg or .bmp file format. For more information, see "*Creating a Welcome Video Slide"* on page 16-30.
- Customized H.261 slides are not supported.

21 Click the **DTMF Codes** tab.

The *New Conference IVR Service - DTMF Codes* dialog box opens.



Conference IVR Service Name: IVR_ENGLISH

DTMF Codes

Name	DTMF Code	Permission
Cancel Mute All Except Me	#5	Chairperson
Change Password	*77	Chairperson
Change To Chairperson	*78	Everyone
Mute All Except Me	*5	Chairperson
Mute Incoming Participants	*86	Chairperson
Mute My Line	*6	Everyone
Override Mute All		Everyone
Pause Recording	*1	Chairperson
Play Help Menu	*83	Everyone
Polycom Touch Control	*#	Everyone
Request assistance for conferenc	00	Chairperson
Request private assistance	*0	Everyone

- This dialog box lists the default DTMF codes for the various functions that can be performed during the conference by all participants or by the chairperson.

Table 16-9 New Conference IVR Service Properties - DTMF Codes

Operation	DTMF String	Permission
Mute My Line	*6	Everyone
Unmute My Line	#6	Everyone
Mute All Except Me	*5	Chairperson
Cancel Mute All Except Me	#5	Chairperson
Change Password	*77	Chairperson
Mute Incoming Participants	*86	Chairperson
Unmute Incoming Participants	#86	Chairperson
Play Help Menu	*83	Everyone
Terminate Conference	*87	Chairperson

Table 16-9 New Conference IVR Service Properties - DTMF Codes

Operation	DTMF String	Permission
Change To Chairperson	*78	Everyone
Override Mute All	Configurable	Everyone
Start Recording	*3	Chairperson
Stop Recording	*2	Chairperson
Pause Recording	*1	Chairperson
Secure Conference	*71	Chairperson
Unsecured Conference	#71	Chairperson
Request individual assistance **Note:** This option is not available for SVC participants.	*0	Everyone
Request assistance for conference **Note:** This option is not available for SVC participants.	00	Chairperson
Request to Speak	99	Everyone

22 To modify the DTMF code or permission:

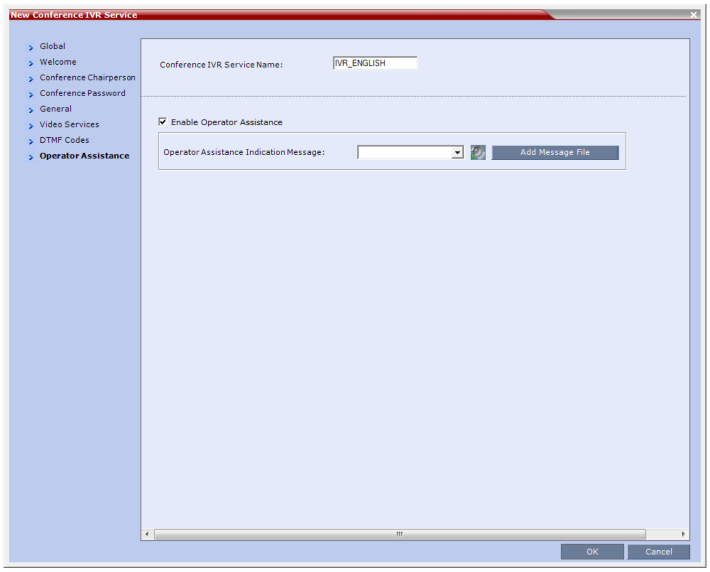
a In the *DTMF Code* column, in the appropriate entry enter the new code.

b In the *Permission* column, select from the list who can use this feature (Everyone or just the Chairperson).

By default, the Secure, Unsecure Conference and Show Number of Participants options are enabled in the Conference IVR Service. These options can be disabled by removing their codes from the Conference IVR Service.

- To disable the Text Indication option in the DTMF Code column, clear the DTMF code **(*88)** of *Show Number of Participants* from the table.
- To disable the Secure Conference options, in the *DTMF Code* column, clear the DTMF codes of both Secured Conference **(*71)** and Unsecured Conference **(#71)** from the table.

23 Click the **Operator Assistance** tab.
The *Operator Assistance* dialog box opens.



Conference IVR Service Name: IVR_ENGLISH

☑ Enable Operator Assistance

Operator Assistance Indication Message:

24 Select **Enable Operator Assistance** to enable operator assistance when the participant requires or requests help during the connection process to the conference or during the conference.

25 In the *Operator Assistance Indication Message* field, select the audio message to be played when the participant requests or is waiting for the operator's assistance.

> If the audio file was not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the *Collaboration Server*.

26 Click **OK** to complete the IVR Service definition.
The new Conference IVR Service is added to the *IVR Services* list.

Change to Chairperson

Regular participants can request to become the conference chairperson using the appropriate DTMF code (default: *78), which enabled them to perform operations designated for chairpersons only.

The Change to Chairperson via the DTMF code (default: *78) is executed only if the following settings were configured for the MCU and the conference:

- In the *Conference IVR Service - Conference Chairperson* dialog box, select the **Enable Chairperson Messages** check box, and select the appropriate voice messages.

  For more information, see *Polycom® RealPresence Collaboration Server Virtual Edition Administrator's Guide*, "*New Conference IVR Service Properties - Conference Chairperson Options and Messages"* on page **16-9**.

- When starting a new conference or defining a new Meeting Room, define the *Chairperson Password* in the conference General dialog box.

  For more information, see "*Creating a New Meeting Room"* on page **6-4**.

Entry Queue IVR Service

An Entry Queue (EQ) is a routing lobby for conferences. Participants are routed to the appropriate conference according to the conference ID they enter.

An Entry Queue IVR Service must be assigned to the Entry Queue to enable the voice prompts and video slide guiding the participants through the connection process.

An Entry Queue IVR Service is a subset of an IVR Service. You can create different Entry Queue Services for different languages and personalized voice messages.

The *Collaboration Server* is shipped with a default Entry Queue IVR Service and all its audio messages and video slide. You can define new Entry Queue IVR Services or modify the default Entry Queue IVR Service.

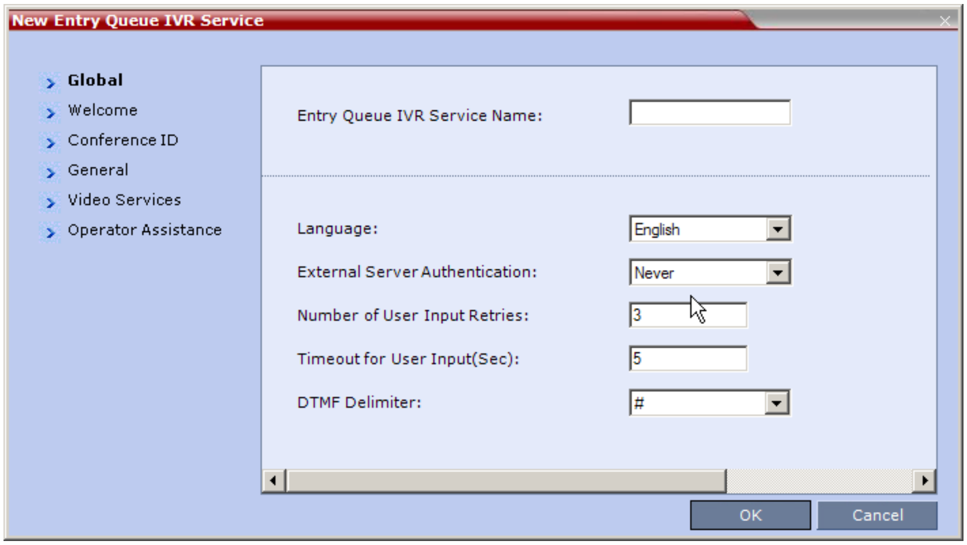
Defining a New Entry Queue IVR Service

To set up a new Entry Queue IVR Service:

1 In the *Collaboration Server Management* pane, click **IVR Services** ( ).

2 In the *IVR Services* list, click the **New Entry Queue IVR Service** ( ) button.

The *New Entry Queue IVR Service - Global* dialog box opens.



Entry Queue IVR Service Name:

Language: English

External Server Authentication: Never

Number of User Input Retries: 3

Timeout for User Input(Sec): 5

DTMF Delimiter: #

3 Fill in the following parameters:

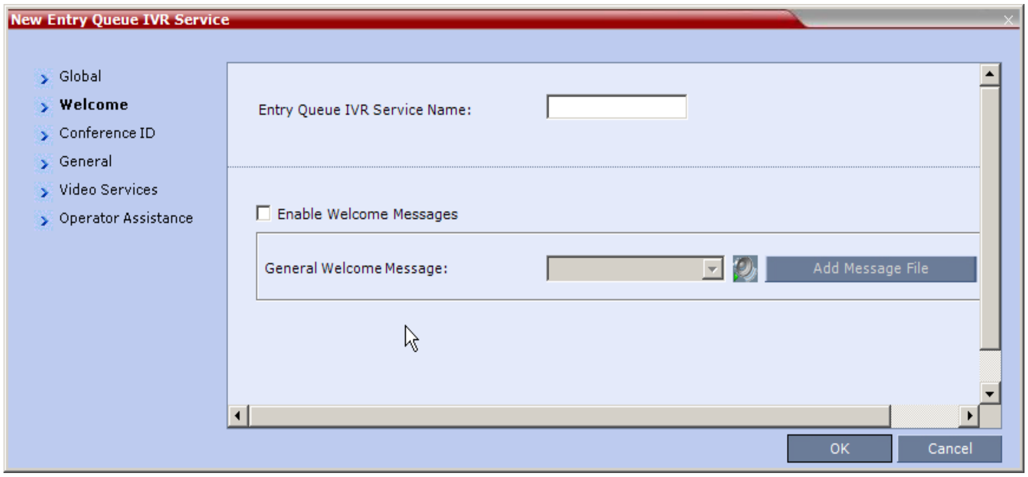
Table 16-10 *Entry Queue IVR Service Properties - Global Parameters*

Option	Description
Entry Queue Service Name	(Mandatory) Enter the name of the Entry Queue Service. The name can be typed in Unicode. Maximum field length is 80 ASCII characters.
Language	Select the language in which the Audio Messages and prompts will be heard. The languages are defined in the *Supported Languages* function.

Table 16-10 Entry Queue IVR Service Properties - Global Parameters (Continued)

Option	Description
External Server Authentication	This option is used for Ad Hoc conferencing, to verify the participant's permission to initiate a new conference. For a detailed description see *Appendix D: "Ad Hoc Conferencing and External Database Authentication"* on page **D-1**. Select one of the following options: • **None** to start a new conference without verifying with an external database the user right to start it. • **Conference ID** to verify the user's right to start a new conference with an external database application using the conference ID.
Number of User Input Retries	Enter the number of times the participant is able to respond to each menu prompt before the participant is disconnected from the MCU.
Timeout for User Input (Sec.)	Enter the duration in seconds that the system waits for input from the participant before it is considered as an input error.
DTMF Delimiter	The interaction between the caller and the system is done via touch-tone signals (DTMF codes). Enter the key that will be used to indicate a DTMF command sent by the participant or the conference chairperson. Possible keys are the pound key (#) or star (*).

4 Click the **Welcome** tab.
The *New Entry Queue IVR Service - Welcome* dialog box opens.



Entry Queue IVR Service Name:

☐ Enable Welcome Messages

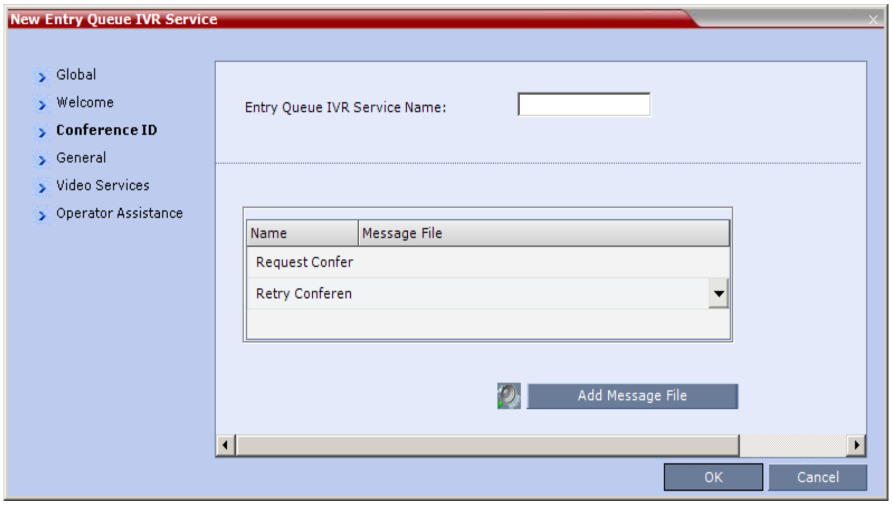
General Welcome Message:

If the files were not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the *Collaboration Server*.

5 Define the appropriate parameters. This dialog box contains options that are identical to those in the *Conference IVR Service - Welcome Message* dialog box. For more information about these parameters, see Table 16-4 on page **16-9**.

6 Click the **Conference ID** tab.

The *New Entry Queue IVR Service - Conference ID* dialog box opens.



Entry Queue IVR Service Name:

Name	Message File
Request Confer	
Retry Conferen	

7 Select the voice messages:

Table 16-11 Entry Queue IVR Service Properties - Conference ID

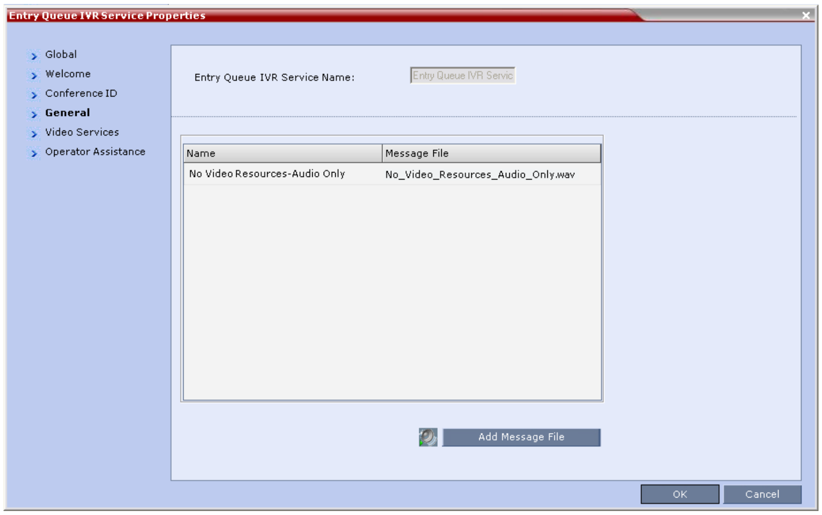
Field/Option	Description
Request Conference ID	Prompts the participant for the conference ID.
Retry Conference ID	When the participant entered an incorrect conference ID, requests the participant to enter the ID again.

8 Assign an audio file to each message type, as follows:

— In the *Message File* column, click the table entry, and then select the appropriate audio message.

9 Click the **General** tab.

The *New Entry Queue IVR Service - General* dialog box opens.



Entry Queue IVR Service Name: Entry Queue IVR Servic

Name	Message File
No Video Resources-Audio Only	No_Video_Resources_Audio_Only.wav

The administrator can enable an audio message that informs the participant of the lack of *Video Resources* in the *Collaboration Server* and that he/she is being connected as *Audio Only*. The message states: *All video resources are currently in use. Connecting using audio only*.

The following guidelines apply:

— The *IVR* message applies to video participants only. *Audio Only* participants will not receive the message.

— Only *H*.323 and *SIP* participants receive the audio message.

— The audio message is the first message after the call is connected, preceding all other *IVR* messages.

— The message is called *No Video Resources-Audio Only* and the message file (*.wav*) is called *No video resources audio only.wav*.

— The audio message must be added to the *Conference* and *Entry Queue IVR Services* separately.

— The IVR message can be enabled/disabled by the administrator using the **ENABLE_ NO_VIDEO_RESOURCES_ AUDIO_ONLY_MESSAGE** *System Flag* in *system.cfg*.

Possible values: **YES** / **NO,** default: **YES**

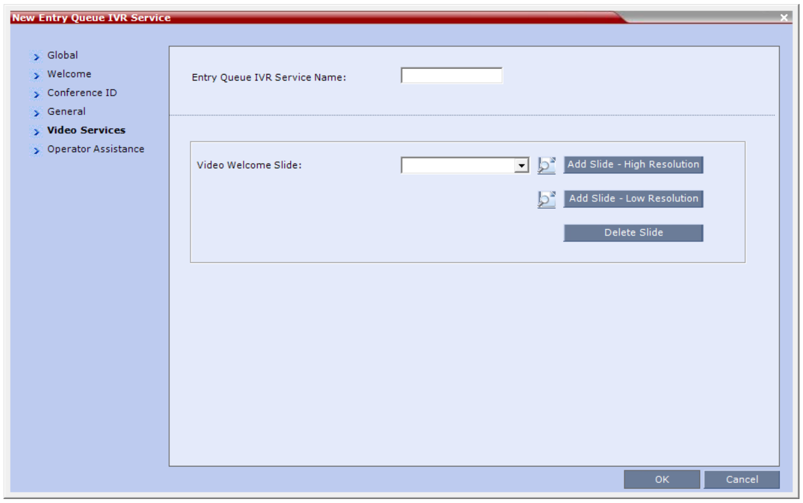
If you wish to modify the flag value, the flag must be added to the *System Configuration* file. For more information see the "*Modifying System Flags"* on page **20-1**.

10 Enter the message *Name* and *Message File* name for the *Audio Only* message:

— Message *Name*: **No Video Resources-Audio Only**

— *Message File* name: **No_Video_Resources_Audio_Only.wav**

11 Click the **Video Services** tab.
The *New Entry Queue IVR Service - Video Services* dialog box opens.


New Entry Queue IVR Service
Global
Welcome
Conference ID
General
Video Services
Operator Assistance
Entry Queue IVR Service Name:
Video Welcome Slide:
Add Slide - High Resolution
Add Slide - Low Resolution
Delete Slide
OK
Cancel

12 In the *Video Welcome Slide* list, select the video slide that will be displayed to participants connecting to the Entry Queue. The slide list includes the video slides that were previously uploaded to the MCU memory.

13 To view any slide, click the **Preview Slide** (🔍) button.

14 If the video slide file was not uploaded to the MCU prior to the IVR Service definition, click the:

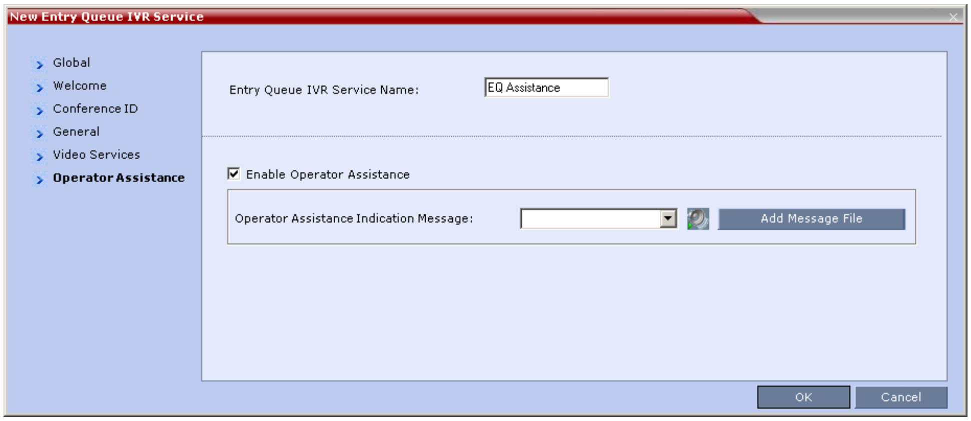
— **Add Slide - Low Resolution** button to upload a *Low Resolution Slide*.

— **Add Slide - High Resolution** button to upload a *High Resolution Slide*.

The *Install File* dialog box opens. The uploading process is similar to the uploading of audio files. For more information, see step **6** on page **16-8**.

> The video slide must be in a .jpg or .bmp file format. For more information, see "*Creating a Welcome Video Slide*" on page **16-30**.

15 Click the **Operator Assistance** tab.
The *Operator Assistance* dialog box opens.



Entry Queue IVR Service Name: EQ Assistance

☑ Enable Operator Assistance

Operator Assistance Indication Message:

16 Select **Enable Operator Assistance** to enable operator assistance when the participant requires or requests help during the connection process.

17 In the *Operator Assistance Indication Message* field, select the audio message to be played when the participant requests or is waiting for operator's assistance.

> If the audio file was not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the *Collaboration Server*.

18 Click **OK** to complete the Entry Queue Service definition.
The new Entry Queue IVR Service is added to the *IVR Services* list. For more information, see "*IVR Services List"* on page **16-1**.

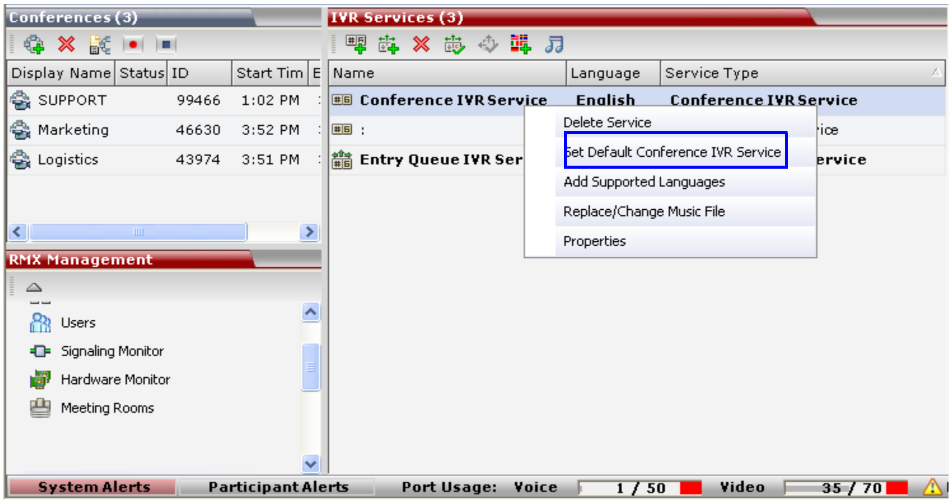
Setting a Conference IVR Service or Entry Queue IVR Service as the Default Service

The first Conference IVR Service and Entry Queue IVR Service are automatically selected by default. The IVR Services (Conference and Entry Queue) shipped with the system are also set as default. If additional Conference IVR Services and Entry Queue IVR Services are defined, you can set another service as the default for each service type.

To select the default Conference IVR Service:

>> In the *IVR Services* list, select the Conference IVR Service to be defined as the default, and then click the **Set Default Conference IVR Service** () button.

Alternatively, in the *IVR Services* list, right-click the Conference IVR Service and then select *Set Default Conference IVR Service*.



Display Name	Status	ID	Start Tim	E
SUPPORT		99466	1:02 PM	:
Marketing		46630	3:52 PM	:
Logistics		43974	3:51 PM	:

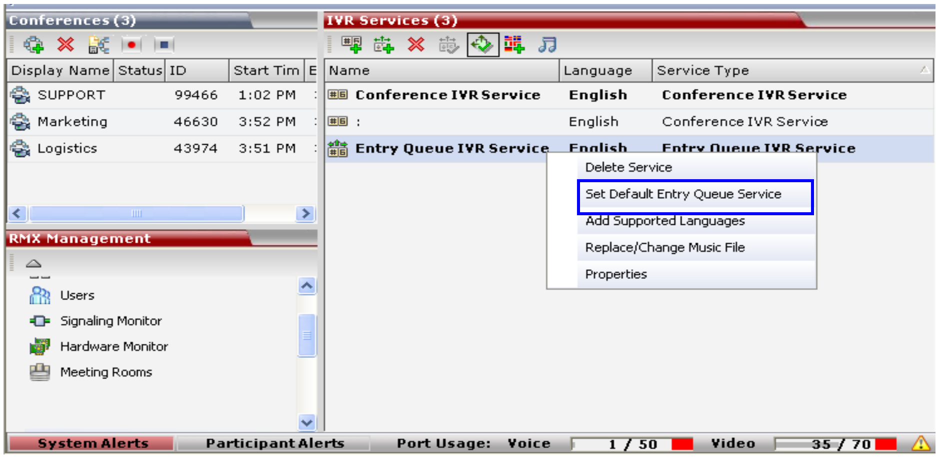
Name	Language	Service Type
Conference IVR Service	**English**	**Conference IVR Service**
:		ice
Entry Queue IVR Ser		**ervice**

The IVR Service is displayed in bold, indicating that it is the current default service.

To select the Default Entry Queue IVR Service:

>> In the *IVR Services* list, select the Entry Queue IVR Service to be defined as the default, and then click **Set Default Entry Queue IVR Service** ( ) button.

Alternatively, in the *Conference IVR Services* list, right-click the Entry Queue IVR Service and then select *Set Default Entry Queue IVR Service*.



Display Name	Status	ID	Start Tim	E
SUPPORT		99466	1:02 PM	:
Marketing		46630	3:52 PM	:
Logistics		43974	3:51 PM	:

Name	Language	Service Type
Conference IVR Service	**English**	**Conference IVR Service**
:	English	Conference IVR Service
Entry Queue IVR Service	**English**	**Entry Queue IVR Service**

The default Entry Queue IVR Service is displayed in bold, indicating that it is the current default service.

Modifying the Conference or Entry Queue IVR Service Properties

You can modify the properties of an existing IVR Service, except the service name and language.

To modify the properties of an IVR Service:

1. In the *Collaboration Server Management* pane, click **IVR Services**.
2. In the *IVR Services* list, Click the IVR Service to modify.
   For more information about the tabs and options of this dialog box, see "*Defining a New Conference IVR Service"* on page **16-6**.
3. Modify the required parameters or upload the required audio files.
4. Click **OK**.

Replacing the Music File

The *Collaboration Server* is shipped with a default music file that is played when participants are placed on hold, for example, while waiting for the chairperson to connect to the conference (if the conference requires a chairperson), or when a single participant is connected to the conference. You can replace the default music file with your own recorded music.

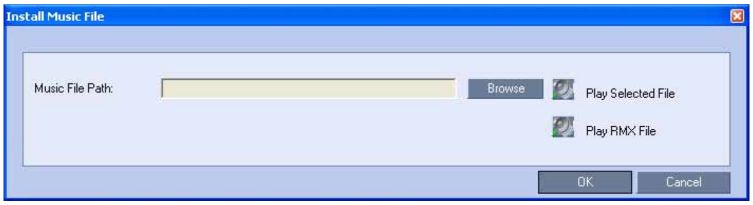
Music file guidelines:

- The file must be in *.wav format.
- Music length cannot exceed one hour.
- The music recording must be in the range of (-12dB) to (-9dB).

Adding a Music File

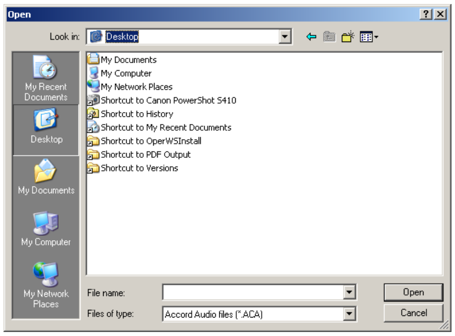
To replace the Music file:

1. In the *Collaboration Server Management* pane, click **IVR Services**.
2. In the *IVR Services* list toolbar, click the **Replace/Change Music File** (♫) button.
   The *Install Music File* window opens.



3. Click the **Browse** button to select the audio file (*.wav) to upload.

   The *Open* dialog box opens.



4. Select the appropriate audio *.wav file and then click the **Open** button.
   The selected file name is displayed in the *Install Music File* dialog box.
5. Optional. You can play the selected file by clicking the *Play* (🔊) button.
   a. Click **Play Selected File** to play a file on your computer.
   b. Click **Play *Collaboration Server* File** to play a file already uploaded on the *Collaboration Server*.
6. In the *Install Music File* dialog box, click **OK** to upload the file to the MCU.
   The new file replaces the previously uploaded file and this file is used for all background music played by the MCU.

Creating Audio Prompts and Video Slides

The *Collaboration Server* is shipped with default voice messages (in WAV format) and video slides that are used for the default IVR services. You can create your own video slides and record the voice messages for different languages or customize them to your needs.

Recording an Audio Message

To record audio messages, use any sound recording utility available in your computer or record them professionally in a recording studio. Make sure that recorded message can be saved as a Wave file (*.wav format) and that the recorded format settings are as defined in steps 4 and 5 on page **16-28**. The files are converted into the *Collaboration Server* internal format during the upload process.

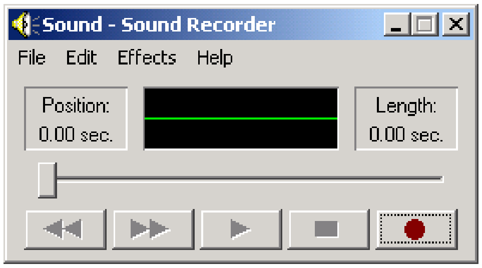
This section describes the use of the Sound Recorder utility delivered with Windows 95/98/2000/XP.

To define the format settings for audio messages:

- The format settings for audio messages need to be set only once. The settings will then be applied to any new audio messages recorded.
- The utility or facility used to record audio messages must be capable of producing audio files with the formats and attributes as shown in the following procedure, namely, **PCM**, **16.000kHz, 16Bit, Mono**.
  Windows® XP® Sound Recorder is one of the utilities that can be used.

1 On your PC, click **Start** > **Programs** > **Accessories** > **Entertainment** > **Sound Recorder**.

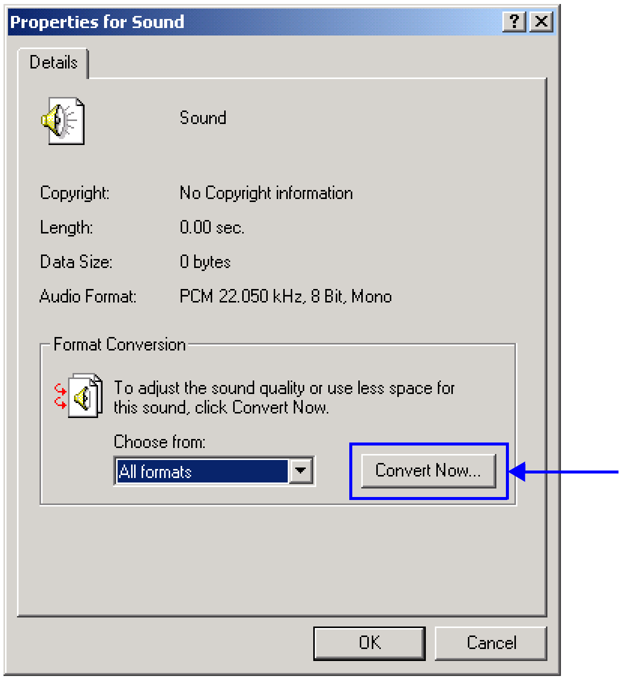
The *Sound–Sound Recorder* dialog box opens.



2 To define the recording format, click **File** > **Properties**.

The *Properties for Sound* dialog box opens.

3 Click the **Convert Now** button.

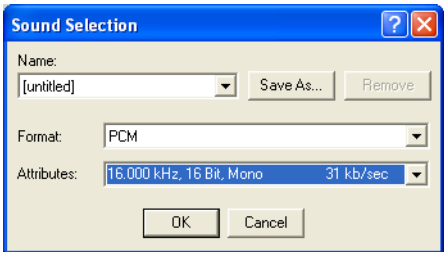


Properties for Sound
Details
Sound
Copyright: No Copyright information
Length: 0.00 sec.
Data Size: 0 bytes
Audio Format: PCM 22.050 kHz, 8 Bit, Mono
Format Conversion
To adjust the sound quality or use less space for this sound, click Convert Now.
Choose from:
All formats
Convert Now...
OK
Cancel

The *Sound Selection* dialog box opens.

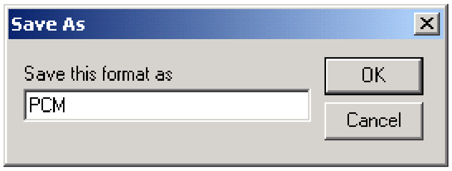
4 In the *Format* field, select **PCM**.

5 In the *Attributes* list, select **16.000 kHz, 16Bit, Mono.**



6 To save this format, click the **Save As** button.
The *Save As* dialog box opens.

7 Select the location where the format will reside, enter a name and then click **OK**.



The system returns to the *Sound Selection* dialog box.

8 Click **OK**.
The system returns to the *Properties for Sound* dialog box.

9 Click **OK**.
The system returns to the *Sound–Sound Recorder* dialog box. You are now ready to record your voice message.

To record a new audio message:

Regardless of the recording utility you are using, verify that any new audio message recorded adheres to the following format settings: **16.000kHz, 16Bit, Mono**.

Make sure that a microphone or a sound input device is connected to your PC.

1 On your PC, click **Start** > **Programs** > **Accessories** > **Entertainment** > **Sound Recorder**.

The *Sound–Sound Recorder* dialog box opens.

2 Click **File** > **New**.

3 Click the **Record** button.
The system starts recording.

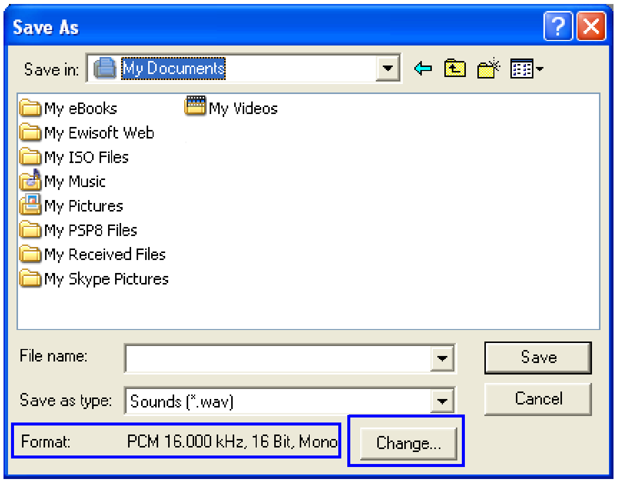
4 Start narrating the desired message.

For all audio IVR messages, stop the recording anytime up to 3 minutes (which is the maximum duration allowed for an IVR voice message). If the message exceeds 3 minutes it will be rejected by the *Collaboration Server* unit.

5 Click the **Stop Recording** button.

6 Save the recorded message as a wave file, click **File** > **Save As**.

The *Save As* dialog box opens.



Save in: My Documents

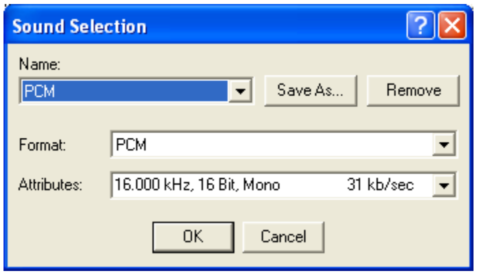
File name:

Save as type: Sounds (*.wav)

Format: PCM 16.000 kHz, 16 Bit, Mono

7 Verify that the *Format* reads: **PCM 16.000 kHz, 16Bit, Mono**. If the format is correct, continue with step 10. If the format is incorrect, click the **Change** button.

The *Sound Selection* dialog box is displayed.



Name: PCM

Format: PCM

Attributes: 16.000 kHz, 16 Bit, Mono 31 kb/sec

8 In the *Name* field, select the name of the format created in step 7 on page 16-28.

9 Click **OK**.

The system returns to the *Save As* dialog box.

10 In the *Save in* field, select the directory where the file will be stored.

11 In the *Save as Type* field, select the ***.wav** file format.

12 In the *File name* box, type a name for the message file, and then click the **Save** button.

13 To record additional messages, repeat steps 1 to 10.

> To upload your recorded *.wav file to the *Collaboration Server*, see step 6 on **page 16-8**.

Creating a Welcome Video Slide

The video slide is a still picture that can be created in any graphic application.

To create a welcome video slide:

1 Using any graphic application, save your image in either ***.jpg** or ***.bmp** file format.

2 For optimum quality, ensure that the image dimensions adhere to the *Collaboration Server* recommended values (width x height in pixels):

— 640 x 480

— 704 x 480

— 848 x 480

— 720 x 576

— 704 x 576

— 1024 x 576

— 960 x 720

— 1280 x 720

— 1440 x 1088

— 1920 x 1088

The *Collaboration Server* can accommodate small deviations from the recommended slide resolutions.

3 Save your file.

> Customized H.261 slides are not supported..

> If using a default Polycom slide, the slide's resolution will be as defined in the profile, i.e. SD, HD or CIF.
> If the display of the Welcome slide is cut in the upper area of the screen, change the settings of the endpoint's monitor to People "Stretch" instead of "Zoom".

> To upload your video slide to the *Collaboration Server*, see step 12 on **page 16-22**.

Default IVR Prompts and Messages

The system is shipped with the following audio prompts and messages:

Table 16-12 Default IVR Messages

Message Type	Message Text	When Played	File Name
General Welcome Message	"Welcome to unified conferencing."	The participant enters the conference IVR queue	General_Welcome.wav
Chairperson Identifier Request	"For conference Chairperson Services, Press the Pound Key. All other participants please wait..."	The participant is asked to self-identify as the chairperson	Chairperson_ Identifier.wav
Request Chairperson Password	"Please enter the Conference Chairperson Password. Press the pound key when complete."	The participant is asked for the chairperson password	Chairperson_Password.wav
Retry Chairperson Password	"Invalid chairperson password. Please try again."	A participant enters an incorrect Chairperson password	Chairperson_ Password_Failure.wav
Request Password	"Please enter the conference password. Press the pound key when complete."	A participant is requested to enter the conference password	Conference_ Password.wav
Retry Password	"Invalid conference password. Please try again."	An incorrect conference password is entered	Retry_ Conference_ Password.wav
Request Digit	"Press any key to enter the conference."	A participant is requested to press any key	Request_Digit.wav
Request Billing Code	"Please enter the Billing code. Press the pound key when complete."	A participant is asked to enter a billing code	Billing_Code.wav
Requires Chairperson	"Please wait for the chairperson to join the conference."	A participant attempts to join a conference prior to the Chairperson joining	Requires Chairperson.wav
Chairperson Exit	"The chairperson has left the conference." **Note:** The *TERMINATE_CONF_AFTER_CHAIR_ DROPPED* flag must be enabled to play this message.	The chairperson has left the conference.	Chairperson_Exit.wav

Table 16-12 *Default IVR Messages (Continued)*

Message Type	Message Text	When Played	File Name
First to Join	"You are the first person to join the conference."	The first participant joins a conference	First to Join.wav
Mute All On	"All conference participants are now muted."	When all participants are muted by the operator or chairperson.	Mute_All_On.wav
Mute All Off	"All conference participants are now unmuted."	When all participants are unmuted by the operator or chairperson.	Mute_All_Off.wav
End Time Alert	"The conference is about to end."	The conference is about it end	End_Time_Alert.wav
Change Password Menu	"Press one to change conference password. Press two to change chairperson password. Press nine to exit the menu."	A participant requests a conference password change	Change_Password_ Menu.wav
Change Conference Password	"Please enter the new conference password. Press the pound key when complete."	A participant presses two in the *Change Password* IVR menu.	Change_ Conference_ Password.wav
Change Chairperson Password	"Please enter the new chairperson password. Press the pound key when complete."	A participant presses one in the *Change Password* IVR menu.	*Change_ Chairperson_ Password.wav*
Confirm Password Change	"Please re-enter the new password. Press the pound key when complete."	A participant enters a new conference or chairperson password	*Confirm_ Password_ Change.wav*
Change Password Failure	"The new password is invalid."	A participant enters an invalid password	*Change_ Password_ Failure.wav*
Password Changed Successfully	"The password has been successfully changed."	A participant has confirmed a password change	*Password_ Changed_ Successfully.wav*
Self Mute	"You are now muted."	A participant mutes his or her audio	*Self_Mute.wav*
Self Unmute	"You are no longer muted."	A participant unmutes his or her audio	*Self_Unmute.wav*

Table 16-12 *Default IVR Messages (Continued)*

Message Type	Message Text	When Played	File Name
Chairperson Help Menu	"The available touch-tone keypad actions are as follows: • To exit this menu press any key. • To request private assistance, press star, zero. • To request operator's assistance for the conference, press zero, zero. • To mute your line, press star, six. • To unmute your line, press pound, six."	A chairperson requests the chairperson help menu	*Chairperson_ Help_Menu.wav*
Participant Help Menu	"The available touch-tone keypad actions are as follows: • To exit this menu press any key. • To request private assistance, press star, zero. • To mute your line, press star, six. • To unmute your line, press pound, six. • To increase your volume, press star, nine. • To decrease your volume, press pound, nine.	A participant requests the participant help menu	*Participant_Help_Menu.wav*
Maximum Participants Exceeded	"The conference is full. You cannot join at this time."	A participant attempts to join a full conference	*Maximum_ Participants_ Exceeded.wav*
Request Conference NID	"Please enter your conference NID. Press the pound key when complete."		*Request_ Conference_NID.wav*
Retry Conference NID	"Invalid conference NID. Please try again."	A participant enters an invalid conference NID	*Retry_Conference_NID.wav*
Secured Conference	"The conference is now secured."	A chairperson or participant secures a conference	*Conference_Secured.wav*
Unsecured Conference	"The conference is now in an unsecured mode"	A chairperson or participant unsecures a conference	*Conference_Unsecured.wav*
Locked Conference	"Conference you are trying to join is locked"		*Conference_Locked.wav*
Conference Recording	"The conference is being recorded"		*Recording_ in_Progress.wav*

Table 16-12 Default IVR Messages (Continued)

Message Type	Message Text	When Played	File Name
Conference Recording Failed	"The conference recording has failed"		*Recording_Failed.wav*
No Video Resources Audio Only.	"All video resources are currently in use. Connecting using audio only"		*No_Video_Resources_Audio_ Only.wav*

17

The Call Detail Record (CDR) Utility

The Call Detail Record (CDR) utility enables you to view summary information about conferences, and retrieve full conference information and archive it to a file. The file can be used to produce reports or can be exported to external billing programs.

> The value of the fields that support Unicode values, such as the info fields, will be stored in the CDR file in UTF8. The application that reads the CDR must support Unicode.

The *Collaboration Server* can store details of up to 2000 conferences. When this number is exceeded, the system overwrites conferences, starting with the earliest conference. To save the conferences' information, their data must be retrieved and archived. The frequency with which the archiving should be performed depends on the volume of conferences run by the MCU.

The *Collaboration Server* displays Active Alarms before overwriting the older files, enabling the users to backup the older files before they are deleted.

The display of Active Alarms is controlled by the ENABLE_CYCLIC_FILE_SYSTEM_ALARMS System Flag.

If the ENABLE_CYCLIC_FILE_SYSTEM_ALARMS is set to YES (default setting when ULTRA_SECURE_MODE System Flag is set to YES) and a Cyclic File reaches a file storage capacity limit, an Active Alarm is created: “Backup of CDR files is required”.

Each conference is a separate record in the MCU memory. Each conference is archived as a separate file. Each conference CDR file contains general information about the conference, such as the conference name, ID, start time and duration, as well as information about events occurring during the conference, such as adding a new participant, disconnecting a participant or extending the length of the conference.

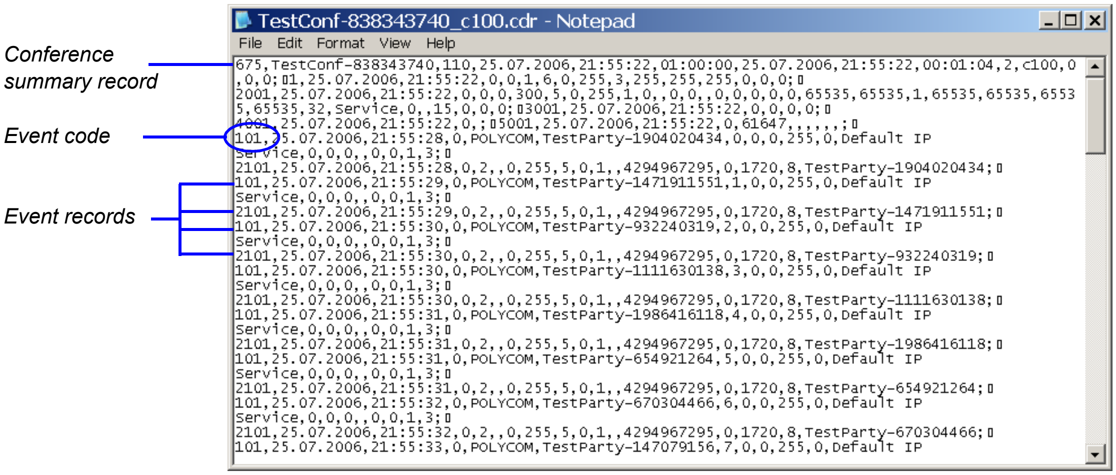
The CDR File

CDR File Formats

The conference CDR records can be retrieved and archived in the following two formats:

- **Unformatted data** – Unformatted CDR files contain multiple records in “raw data” format. The first record in each file contains general conference data. The remaining records contain event data, one record for each event. Each record contains field values separated by commas. This data can be transferred to an external program such as Microsoft Excel$^{©}$ for billing purposes.

The following is a sample of an unformatted CDR file:


TestConf-838343740_c100.cdr - Notepad
File Edit Format View Help
Conference summary record
Event code
Event records

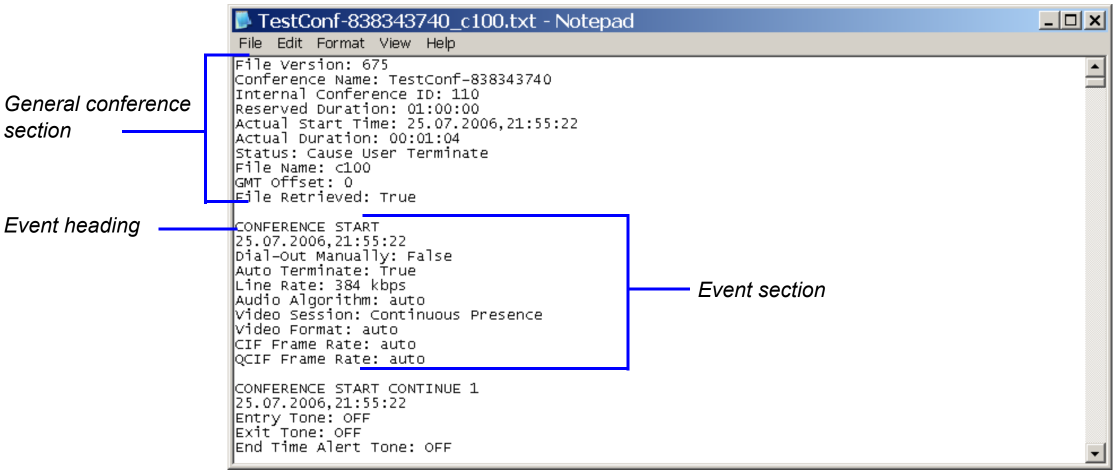
```
675,TestConf-838343740,110,25.07.2006,21:55:22,01:00:00,25.07.2006,21:55:22,00:01:04,2,c100,0
,0,0;▯1,25.07.2006,21:55:22,0,0,1,6,0,255,3,255,255,255,0,0,0;▯
2001,25.07.2006,21:55:22,0,0,0,300,5,0,255,1,0,,0,0,,0,0,0,0,0,65535,65535,1,65535,65535,6553
5,65535,32,Service,0,,15,0,0,0;▯3001,25.07.2006,21:55:22,0,0,0,0;▯
4001,25.07.2006,21:55:22,0,;▯5001,25.07.2006,21:55:22,0,61647,,,,,,;▯
101,25.07.2006,21:55:28,0,POLYCOM,TestParty-1904020434,0,0,0,255,0,Default IP
Service,0,0,0,,0,0,1,3;▯
2101,25.07.2006,21:55:28,0,2,,0,255,5,0,1,,4294967295,0,1720,8,TestParty-1904020434;▯
101,25.07.2006,21:55:29,0,POLYCOM,TestParty-1471911551,1,0,0,255,0,Default IP
Service,0,0,0,,0,0,1,3;▯
2101,25.07.2006,21:55:29,0,2,,0,255,5,0,1,,4294967295,0,1720,8,TestParty-1471911551;▯
101,25.07.2006,21:55:30,0,POLYCOM,TestParty-932240319,2,0,0,255,0,Default IP
Service,0,0,0,,0,0,1,3;▯
2101,25.07.2006,21:55:30,0,2,,0,255,5,0,1,,4294967295,0,1720,8,TestParty-932240319;▯
101,25.07.2006,21:55:30,0,POLYCOM,TestParty-1111630138,3,0,0,255,0,Default IP
Service,0,0,0,,0,0,1,3;▯
2101,25.07.2006,21:55:30,0,2,,0,255,5,0,1,,4294967295,0,1720,8,TestParty-1111630138;▯
101,25.07.2006,21:55:31,0,POLYCOM,TestParty-1986416118,4,0,0,255,0,Default IP
Service,0,0,0,,0,0,1,3;▯
2101,25.07.2006,21:55:31,0,2,,0,255,5,0,1,,4294967295,0,1720,8,TestParty-1986416118;▯
101,25.07.2006,21:55:31,0,POLYCOM,TestParty-654921264,5,0,0,255,0,Default IP
Service,0,0,0,,0,0,1,3;▯
2101,25.07.2006,21:55:31,0,2,,0,255,5,0,1,,4294967295,0,1720,8,TestParty-654921264;▯
101,25.07.2006,21:55:32,0,POLYCOM,TestParty-670304466,6,0,0,255,0,Default IP
Service,0,0,0,,0,0,1,3;▯
2101,25.07.2006,21:55:32,0,2,,0,255,5,0,1,,4294967295,0,1720,8,TestParty-670304466;▯
101,25.07.2006,21:55:33,0,POLYCOM,TestParty-147079156,7,0,0,255,0,Default IP
```

Figure 17-1 *Unformatted CDR File*

- **Formatted text** – Formatted CDR files contain multiple sections. The first section in each file contains general conference data. The remaining sections contain event data, one section for each event. Each field value is displayed in a separate line, together with its name. This data can be used to generate a summary report for a conference

> The field names and values in the formatted file will appear in the language being used for the *Collaboration Server Web Client* user interface at the time when the CDR information is retrieved.

The following is an example of a formatted CDR file:


TestConf-838343740_c100.txt - Notepad
File Edit Format View Help
General conference section
Event heading
Event section

```
File Version: 675
Conference Name: TestConf-838343740
Internal Conference ID: 110
Reserved Duration: 01:00:00
Actual Start Time: 25.07.2006,21:55:22
Actual Duration: 00:01:04
Status: Cause User Terminate
File Name: c100
GMT Offset: 0
File Retrieved: True

CONFERENCE START
25.07.2006,21:55:22
Dial-Out Manually: False
Auto Terminate: True
Line Rate: 384 kbps
Audio Algorithm: auto
Video Session: Continuous Presence
Video Format: auto
CIF Frame Rate: auto
QCIF Frame Rate: auto

CONFERENCE START CONTINUE 1
25.07.2006,21:55:22
Entry Tone: OFF
Exit Tone: OFF
End Time Alert Tone: OFF
```

Figure 17-2 *Formatted CDR File*

Multi-Part CDR Files

By default, the maximum CDR (Call Data Record) file size is limited to 1MB. When a *CDR* file reaches a size of 1MB the file is saved and further call data recording is stopped and the additional data is lost.

The *Collaboration Server* can be configured to keep recording the data in multiple CDR file set of 1MB each. *Multi-Part CDR* ensures that conference call data from long duration or permanent conferences is recorded and not lost.

Guidelines

- *Multi-Part CDR* is enabled by setting the value of the **ENABLE_MULTI_PART_CDR** *System Flag* to **YES**.

  The flag's default value is **NO**.

  When the flag value is **NO,** *CDR* file size is limited to one file of 1MB and further call data recording is stopped.

  To modify the default setting, the flag must be manually added to the *System Configuration*. For more information see, "*Modifying System Flags"* on page **20-1**.
- If the flag value is set to **YES**, when a *CDR* file reaches 1MB, an additional *CDR* file is created and added to the *CDR* file set for that conference.
- If the flag value is changed from **YES** to **NO** (or visa versa) all existing *CDR* files are retained.

CDR File Contents

The general conference section or record contains information such as the Routing Name and ID, and the conference starting date and time.

The event sections or records contain an event type heading or event type code, followed by event data. For example, an event type may be that a participant connects to the conference, and the event data will list the date and time the participant connects to the conference, the participant name and ID, and the participant capabilities used to connect to the conference.

To enable compatibility for applications that written for the MGC family, the *Collaboration Server* CDR file structure is based on the MGC CDR file structure.

The unformatted and formatted text files contain basically the same information. The following differences should be noted between the contents of the unformatted and formatted text files:

- In many cases a formatted text file field contains a textual value, whereas the equivalent unformatted file field contains a numeric value that represents the textual value.
- For reading clarity, in a few instances, a single field in the unformatted file is converted to multiple fields in the formatted text file, and in other cases, multiple fields in the unformatted file are combined into one field in the formatted file.
- To enable compatibility between MGC CDR files and *Collaboration Server* CDR files, the unformatted file contains fields that were applicable to the MGC MCUs, but are not supported by the *Collaboration Server* MCUs. These fields are omitted from the formatted text file.

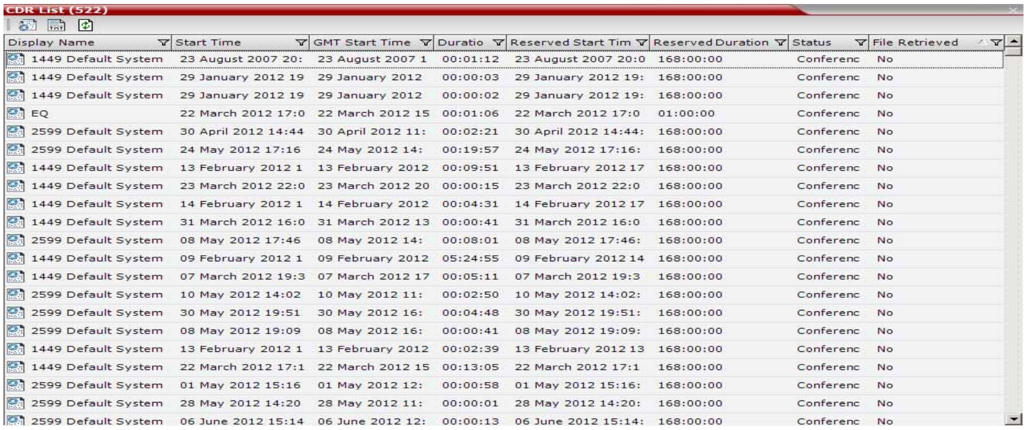
Appendix C: "*CDR Fields - Unformatted File"* on page **C-1**, contains a full list of the events, fields and values that appear in the unformatted file. This appendix can be referred to for information regarding the contents of fields in the unformatted text file, but does not reflect the exact contents of the formatted text file.

Viewing, Retrieving and Archiving Conference Information

Viewing the Conference Records

To open the CDR utility:

- On the *Collaboration Server Menu*, click **Administration > CDR**.
  The *CDR List* pane opens, displaying a list of the conference CDR records stored in the MCU memory.



CDR List (522)

Display Name	Start Time	GMT Start Time	Duratio	Reserved Start Tim	Reserved Duration	Status	File Retrieved
1449 Default System	23 August 2007 20:	23 August 2007 1	00:01:12	23 August 2007 20:0	168:00:00	Conferenc	No
1449 Default System	29 January 2012 19	29 January 2012	00:00:03	29 January 2012 19:	168:00:00	Conferenc	No
1449 Default System	29 January 2012 19	29 January 2012	00:00:02	29 January 2012 19:	168:00:00	Conferenc	No
EQ	22 March 2012 17:0	22 March 2012 15	00:01:06	22 March 2012 17:0	01:00:00	Conferenc	No
2599 Default System	30 April 2012 14:44	30 April 2012 11:	00:02:21	30 April 2012 14:44:	168:00:00	Conferenc	No
2599 Default System	24 May 2012 17:16	24 May 2012 14:	00:19:57	24 May 2012 17:16:	168:00:00	Conferenc	No
1449 Default System	13 February 2012 1	13 February 2012	00:09:51	13 February 2012 17	168:00:00	Conferenc	No
1449 Default System	23 March 2012 22:0	23 March 2012 20	00:00:15	23 March 2012 22:0	168:00:00	Conferenc	No
1449 Default System	14 February 2012 1	14 February 2012	00:04:31	14 February 2012 17	168:00:00	Conferenc	No
1449 Default System	31 March 2012 16:0	31 March 2012 13	00:00:41	31 March 2012 16:0	168:00:00	Conferenc	No
2599 Default System	08 May 2012 17:46	08 May 2012 14:	00:08:01	08 May 2012 17:46:	168:00:00	Conferenc	No
1449 Default System	09 February 2012 1	09 February 2012	05:24:55	09 February 2012 14	168:00:00	Conferenc	No
1449 Default System	07 March 2012 19:3	07 March 2012 17	00:05:11	07 March 2012 19:3	168:00:00	Conferenc	No
2599 Default System	10 May 2012 14:02	10 May 2012 11:	00:02:50	10 May 2012 14:02:	168:00:00	Conferenc	No
2599 Default System	30 May 2012 19:51	30 May 2012 16:	00:04:48	30 May 2012 19:51:	168:00:00	Conferenc	No
2599 Default System	08 May 2012 19:09	08 May 2012 16:	00:00:41	08 May 2012 19:09:	168:00:00	Conferenc	No
1449 Default System	13 February 2012 1	13 February 2012	00:02:39	13 February 2012 13	168:00:00	Conferenc	No
1449 Default System	22 March 2012 17:1	22 March 2012 15	00:13:05	22 March 2012 17:1	168:00:00	Conferenc	No
2599 Default System	01 May 2012 15:16	01 May 2012 12:	00:00:58	01 May 2012 15:16:	168:00:00	Conferenc	No
2599 Default System	28 May 2012 14:20	28 May 2012 11:	00:00:01	28 May 2012 14:20:	168:00:00	Conferenc	No
2599 Default System	06 June 2012 15:14	06 June 2012 12:	00:00:13	06 June 2012 15:14:	168:00:00	Conferenc	No

The following fields are displayed:

Table 17-1 Conference Record Fields

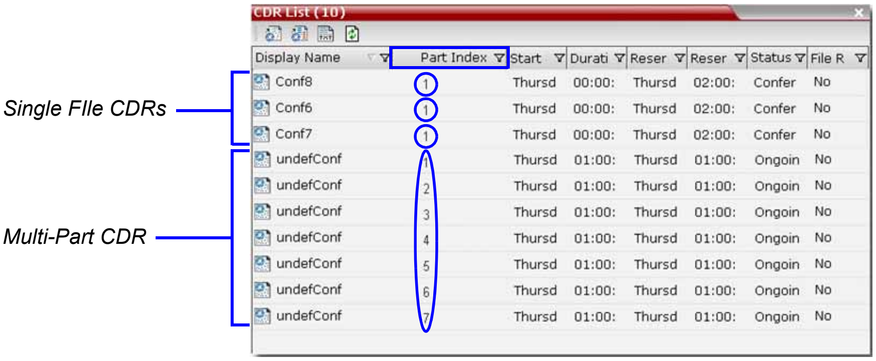
Field	Description
Display Name	The Display Name of the conference and an icon indicating whether or not the CDR record has been retrieved and saved to a formatted text file. The following icons are used: The CDR record has not been saved. The CDR record has been saved.
Start Time	The actual time the conference started.
GMT Start Time	The actual time the conference started according to Greenwich Mean Time (GMT).
Duration	The actual conference duration.
Reserved Start Time	The reserved start time of the conference. If the conference started immediately this is the same as the *Start Time.*
Reserved Duration	The time the conference was scheduled to last. Discrepancy between the scheduled and the actual duration may indicate that the conference duration was prolonged or shortened.

Table 17-1 Conference Record Fields (Continued)

Field	Description
Status	The conference status. The following values may be displayed: • ***Ongoing Conference*** • ***Terminated by User*** • ***Terminated when end time passed*** • ***Automatically terminated when conference was empty*** – The conference ended automatically because no participants joined the conference for a predefined time period, or all the participants disconnected from the conference and the conference was empty for a predefined time period. • ***Conference never became ongoing due to a problem*** • ***Unknown error*** **Note:** If the conference was terminated by an MCU reset, the status **Ongoing Conference** will be displayed.
File Retrieved	Indicates if the conference record was downloaded using any of the file retrieval buttons in the CDR List pane or the API. • **Yes** - when the conference record was retrieved to any file or using the API. • **No** - when the conference record was not retrieved at all. The File Retrieved field is updated whenever the record is downloaded.

Multi-part CDR File display

When the *Multi-Part CDR* is configured on the *Collaboration Server,* an additional column, *Part Index*, is added to the CDR list.



CDR List (10)

Display Name	Part Index	Start	Durati	Reser	Reser	Status	File R
Conf8	1	Thursd	00:00:	Thursd	02:00:	Confer	No
Conf6	1	Thursd	00:00:	Thursd	02:00:	Confer	No
Conf7	1	Thursd	00:00:	Thursd	02:00:	Confer	No
undefConf	1	Thursd	01:00:	Thursd	01:00:	Ongoin	No
undefConf	2	Thursd	01:00:	Thursd	01:00:	Ongoin	No
undefConf	3	Thursd	01:00:	Thursd	01:00:	Ongoin	No
undefConf	4	Thursd	01:00:	Thursd	01:00:	Ongoin	No
undefConf	5	Thursd	01:00:	Thursd	01:00:	Ongoin	No
undefConf	6	Thursd	01:00:	Thursd	01:00:	Ongoin	No
undefConf	7	Thursd	01:00:	Thursd	01:00:	Ongoin	No

The *Part Index* column displays the *CDR* file's sequence in the CDR file set:

- *CDRs* that are up to 1MB consist of a single file. Each file has a unique *Display Name* and a *Part Index* of **1**.
- Files included in a *Multi-Part CDR* file sets have the same *Display Name*. The first file of the set is numbered **1** with each additional *CDR* file numbered in an ascending numeric sequence.

Refreshing the CDR List

To refresh the CDR list:

- Click the **Refresh** button, or right-click on any record and then select **Refresh**. Updated conference CDR records are retrieved from the MCU memory.

Retrieving and Archiving Conference CDR Records

To retrieve and archive CDR records:

1 To retrieve a single CDR record, right-click the record to retrieve and then select the required format (as detailed in Table 17-2).
Alternatively, select the record to retrieve, and then click the appropriate button on the toolbar (as detailed in Table 17-2).

To retrieve multiple CDR records simultaneously, use standard Windows multi-selection methods.

Table 17-2 Conference Information Retrieval Options

Menu Option	Button	Action
Retrieve		Retrieves the conference information as unformatted data into a file whose extension is .cdr.
Retrieve Formatted XML		Retrieves the conference information as formatted text into a file whose extension is .xml. Note: Viewed when logged in as a special support user.
Retrieve Formatted	TXT	Retrieves the conference information as formatted text into a file whose extension is .txt.

The *Retrieve* dialog box opens.
The dialog box displays the names of the destination CDR files.

2 Select the destination folder for the CDR files and then click **OK**.

If the destination file already exists, you will be asked if you want to overwrite the file or specify a new name for the destination file.
The files are saved to the selected folder.

CDR files are not included in the backup process and should be backed up manually by saving the CDR files to a destination device.

18

RMX Manager Application

The *RMX Manager* is the Windows version of the *Collaboration Server Web Client*. It can be used instead of the *Collaboration Server Web Client* for routine *Collaboration Server* management and for *Collaboration Server* management.

Using the *RMX Manager* application, a single user can control a single or multiple MCU units as well as conferences from multiple MCUs. The RealPresence Collaboration Server Virtual Edition system can be managed and controlled by the RMX Manager application.

The *RMX Manager* can list and monitor:

- Up to 20 *Collaboration Server* systems in the MCUs pane
- Up to 800 conferences in the Conferences pane
- Up to 1600 participants in the Participants pane

The *RMX Manager* is faster than the *RMX Web Clien*t and can give added efficiency to *Collaboration Server* management tasks, especially when deployed on workstations affected by:

- Lack of performance due to bandwidth constraints within the LAN/WAN environment.
- Slow operation and disconnections that can be caused by the anti-phishing component of various antivirus applications.

> Users with *Auditor* authorization level cannot connect to the *RealPresence Collaboration Server* via the *RMX Manager* application and must use the *RMX Web Client*.

The *RMX Manager* application can be installed in your local workstation or accessed directly on the *RealPresence Collaboration Server* system without installing it in your workstation.

Accessing the RMX Manager Directly

To access the RMX Manager directly:

>> Start Internet Explorer and in your browser enter:
http://<Collaboration Server IP Address>:8080/RMXManager.html.

For example, if the *Collaboration Server* IP address is 10.226.10.46, enter in the browser the following address: ***http://10.226.10.46:8080/RMXManager.html.***

Installing the RMX Manager

The *RMX Manager* application can be downloaded from one of the *RealPresence Collaboration Server* systems installed in your site or from Polycom web site at
 http://www.polycom.com/support.

Upgrade Notes

- When upgrading the *RMX Manager* application, it is recommended to backup the MCU list using the **Export RMX Manager Configuration** option. For more details, see "*Import/Export RMX Manager Configuration"* on page **18-20**.
- When upgrading the *RMX Manager* from a major version (for example, version 8.0) to a maintenance version of that version (for example, 8.0.x), the installation must be performed from the same MCU (IP address) from which the major version (for example, version 8.0) was installed.
  If you are upgrading from another MCU (different IP address), you must first uninstall the *RMX Manager* application using **Control Panel > Add or Remove Programs**.

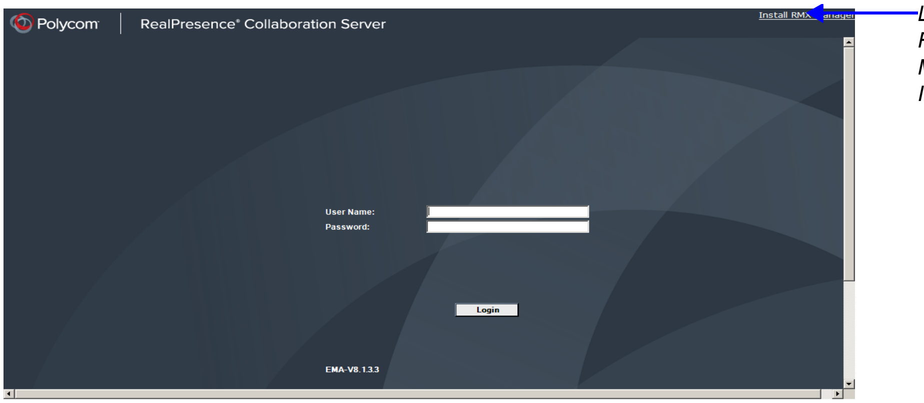
New *RealPresence Collaboration Server* Installation Note

When managing the RealPresence Collaboration Server, upgrade/install the latest MCU version and then install the latest RMX Manager application.
The *Collaboration Server Installation and First Entry Configuration* must be completed before installing the *RMX Manager* application. For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Software Installation"* on page **2-1**.
Once the connection to the *Collaboration Server* unit is established and the *Login* window is displayed, the *RMX Manager* application can be installed.

To install RMX Manager (downloading the application from the *RealPresence Collaboration Server*):

1 Start Internet Explorer and connect to one of the *Collaboration Server* units in your site. It is recommended to connect to the *Collaboration Server* installed with the latest software version.

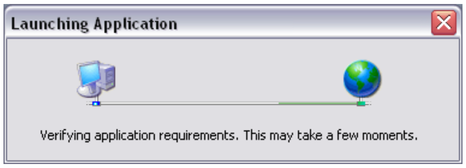
The *Login* screen is displayed. There is a link to the *RMX Manager Installer* at the top of the right edge of the
screen.



Link to RMX Manager Installer

2 Click the **Install RMX Manager** link.

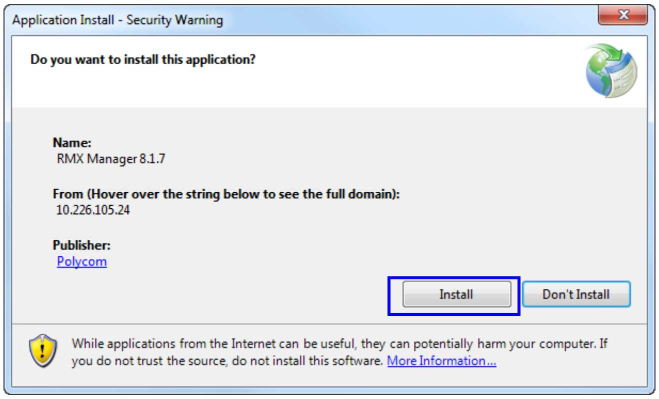
The installer verifies the application's requirements on the workstation.



Launching Application

Verifying application requirements. This may take a few moments.

The *Install* dialog box is displayed.



Application Install - Security Warning

Do you want to install this application?

Name:
RMX Manager 8.1.7

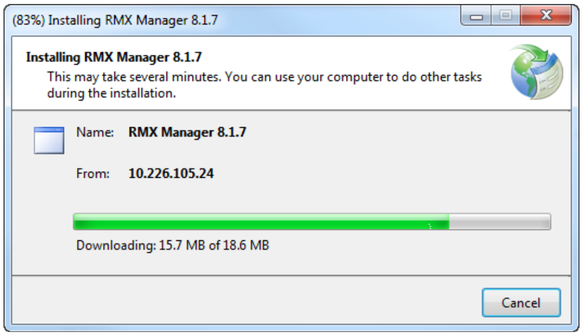
From (Hover over the string below to see the full domain):
10.226.105.24

Publisher:
Polycom

While applications from the Internet can be useful, they can potentially harm your computer. If you do not trust the source, do not install this software. More Information...

3 Click **Install**.

The installation proceeds.



(83%) Installing RMX Manager 8.1.7

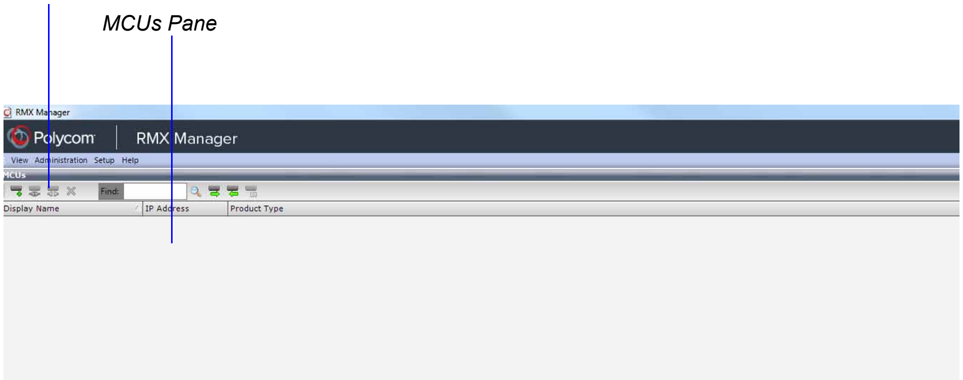
Installing RMX Manager 8.1.7
This may take several minutes. You can use your computer to do other tasks during the installation.

Name: **RMX Manager 8.1.7**

From: **10.226.105.24**

Downloading: 15.7 MB of 18.6 MB

The installation completes, the application loads and the *RMX Manager - MCUs* screen is displayed.



MCUs Toolbar

MCUs Pane

RMX Manager

Display Name	IP Address	Product Type

The first time you start the *RMX Manager* application, the *MCUs* pane is empty.

Starting the RMX Manager Application

Once installed, the *RMX Manager* can be run using the **`http://`** (non-secured) or **`https://`** (secured) command in the browser's address line or the Windows *Start* menu.

To use the browser:

1 In the browser's command line, enter:

```
http://<MCU Control Unit IP Address>:8080/RMXManager.html
```

or

```
https://<MCU Control Unit IP Address>:8080/RMXManager.html
```

2 Press **Enter**.

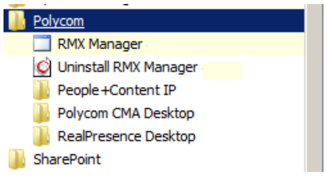
To use the Windows Start menu:

1 Click **Start > Programs.**

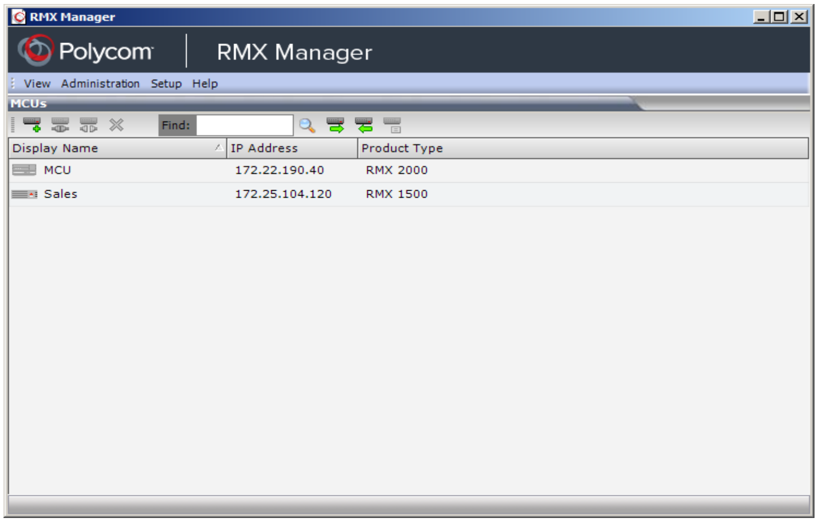
a If the *RMX Manager* is displayed in the recently used programs list, click **RMX Manager** in the list to start the application.

or

b Click **All Programs** > **Polycom** > **RMX Manager**.



The *MCUs* screen is displayed, listing the MCUs currently defined in the *RMX Manager*.



Display Name	IP Address	Product Type
MCU	172.22.190.40	RMX 2000
Sales	172.25.104.120	RMX 1500

This screen enables you to add additional MCUs or connect to any of the MCUs listed. For details on adding MCUs, see *"Adding MCUs to the MCUs List"* on page **11**.

For each listed MCU, the system displays the following information:

— MCU *Display Name* (as defined in the Add MCU dialog box).

— *IP Address* of the MCU's control unit

— *Product Type* - The MCU type: RealPresence Collaboration Server 800s, RMX 800VE, RealPresence Collaboration Server (RMX) 1500, RealPresence Collaboration

Server (RMX) 2000, or RealPresence Collaboration Server (RMX) 4000.
Before connecting to the MCU for the first time, the *Collaboration Server* type is unknown so "RMX" is displayed instead as a general indication.

To display the *RMX Manager* main screen you must connect to one of the listed *Collaboration Server*s. For more details, see "*Connecting to the MCU*" on page **18-5**.

Connecting to the MCU

Once an MCU is defined, the *RMX Manager* can be connected to it. This allows you to set up conferences, make reservations, monitor On Going Conferences and perform other activities on several MCUs.

> The first *Collaboration Server* unit that is connected to the *RMX Manager* dictates the Authorization Level of Users that can connect to the other MCUs on the list. For example, if the Authorization level of the User POLYCOM is Administrator, all Users connecting to the other MCUs on the list must be Administrators. Each user can have a different login name and password for each of the listed MCUs and they must be defined in the Users list of each of the listed MCUs.

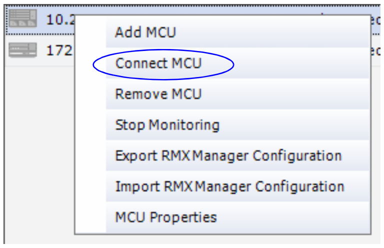
To connect the RMX Manager to an MCU:

1 In the *MCUs* pane or screen, use one of the following methods:

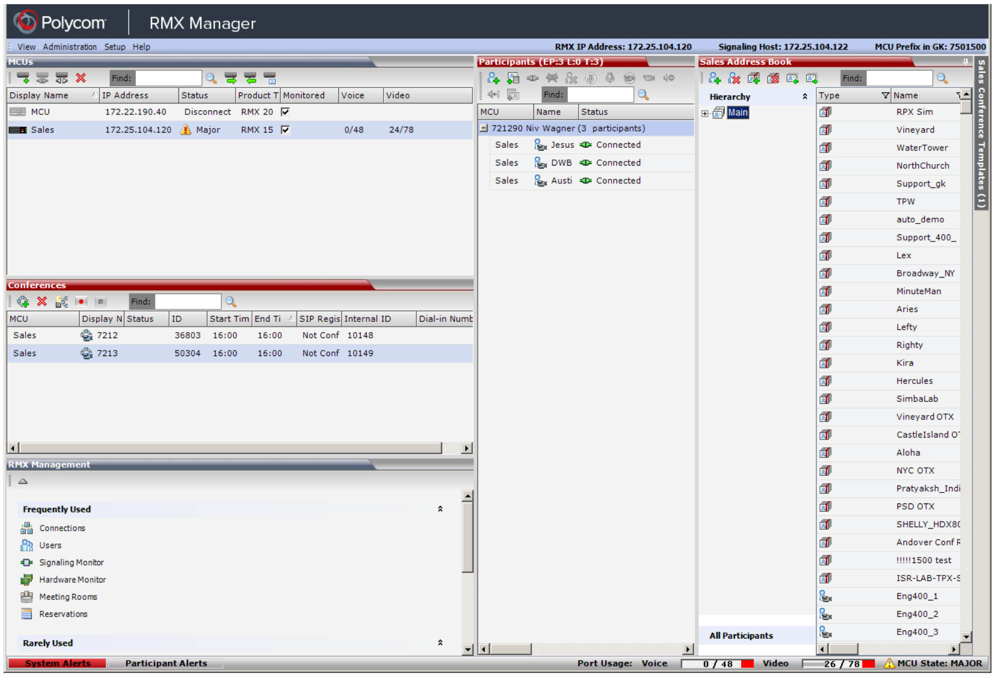
   a Double-click the MCU icon.

   b Select the *Collaboration Server* to connect and click the **Connect MCU** button.

   c Right-click the MCU icon and then click **Connect MCU**.


10.2
172
Add MCU
Connect MCU
Remove MCU
Stop Monitoring
Export RMX Manager Configuration
Import RMX Manager Configuration
MCU Properties

If you are connecting to the MCU from the *MCUs* opening screen and have defined the *Username* and *Password* for the connecting MCU, the system connects to the *Collaboration Server*, and the *RMX Manager Main Screen* is displayed.


Polycom
RMX Manager
View Administration Setup Help
Find:
Sales Conference Templates (1)

RMX IP Address: 172.25.104.120 Signaling Host: 172.25.104.122 MCU Prefix in GK: 7501500

MCUs

Display Name	IP Address	Status	Product T	Monitored	Voice	Video
MCU	172.22.190.40	Disconnect	RMX 20	☑		
Sales	172.25.104.120	Major	RMX 15	☑	0/48	24/78

Participants (EP:3 L:0 T:3)

MCU	Name	Status
721290 Niv Wagner (3 participants)		
Sales	Jesus	Connected
Sales	DWB	Connected
Sales	Austi	Connected

Sales Address Book

Hierarchy
Main
All Participants

Type	Name
	RPX Sim
	Vineyard
	WaterTower
	NorthChurch
	Support_gk
	TPW
	auto_demo
	Support_400_
	Lex
	Broadway_NY
	MinuteMan
	Aries
	Lefty
	Righty
	Kira
	Hercules
	SimbaLab
	Vineyard OTX
	CastleIsland O
	Aloha
	NYC OTX
	Pratyaksh_Indi
	PSD OTX
	SHELLY_HDX8(
	Andover Conf F
	!!!!!1500 test
	ISR-LAB-TPX-S
	Eng400_1
	Eng400_2
	Eng400_3

Conferences

MCU	Display N	Status	ID	Start Tim	End Ti	SIP Regis	Internal ID	Dial-in Numb
Sales	7212		36803	16:00	16:00	Not Conf	10148	
Sales	7213		50304	16:00	16:00	Not Conf	10149	

RMX Management

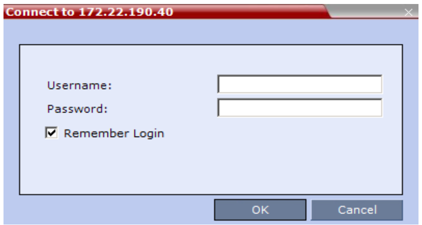
Frequently Used
- Connections
- Users
- Signaling Monitor
- Hardware Monitor
- Meeting Rooms
- Reservations

Rarely Used

System Alerts Participant Alerts Port Usage: Voice 0 / 48 Video 26 / 78 MCU State: MAJOR

If you are connecting to any MCU from the *MCUs* pane in the *RMX Manager Main Screen* and have defined the *Username* and *Password* for the connecting MCU, the MCU icon changes to connected and its status, type and number of audio and video resources are displayed in the *MCUs* pane.

If the *Username* and *Password* are missing from the MCU parameters, or if the *Remember Me* check box has been cleared, the *Connect* dialog box opens.



Connect to 172.22.190.40

Username:

Password:

☑ Remember Login

2 In the *Username* field, enter the user name with which you will login to the MCU.

3 In the *Password* field, enter the password as defined for the user name with which you will login to the MCU.

4 To add the user name and password to the MCU properties so you will not have to enter them each time you login to the MCU, make sure that the **Remember Login** check box is selected. Otherwise, clear the **Remember Login** check box.

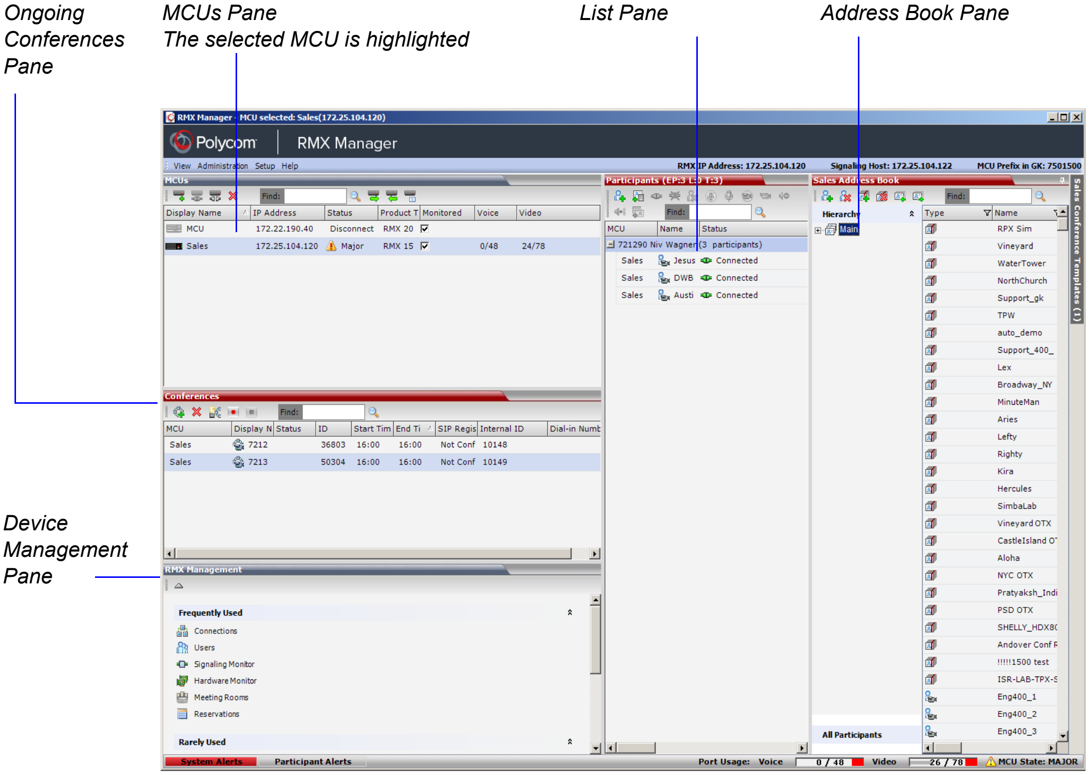
5 Click **OK**.
The system connects to the *Collaboration Server*, and the *RMX Manager Main Screen* is displayed.

If a User with the entered *Username* and *Password* is not defined in the *Collaboration Server*, an error message is displayed and the system lets you re-enter the *Username* and *Password.*

RMX Manager Main Screen

The *RMX Manager Main Screen* is displayed only when at least one MCU is connected.

This screen is similar to the *RMX Web Client Main Screen* with the addition of the *MCUs* pane. As in the *RMX Web Client*, the panes are displayed according to the *Authorization Level* of the logged in User. The *MCUs* pane is displayed to all users.



RMX Manager - MCU selected: Sales(172.25.104.120)

RMX IP Address: 172.25.104.120 Signaling Host: 172.25.104.122 MCU Prefix in GK: 7501500

MCUs

Display Name	IP Address	Status	Product T	Monitored	Voice	Video
MCU	172.22.190.40	Disconnect	RMX 20	☑		
Sales	172.25.104.120	Major	RMX 15	☑	0/48	24/78

Conferences

MCU	Display N	Status	ID	Start Tim	End Ti	SIP Regis	Internal ID	Dial-in Numb
Sales	7212		36803	16:00	16:00	Not Conf	10148	
Sales	7213		50304	16:00	16:00	Not Conf	10149	

Participants (EP:3 L:0 T:3)

MCU	Name	Status
721290 Niv Wagner (3 participants)		
Sales	Jesus	Connected
Sales	DWB	Connected
Sales	Austi	Connected

Sales Address Book

Type	Name
	RPX Sim
	Vineyard
	WaterTower
	NorthChurch
	Support_gk
	TPW
	auto_demo
	Support_400_
	Lex
	Broadway_NY
	MinuteMan
	Aries
	Lefty
	Righty
	Kira
	Hercules
	SimbaLab
	Vineyard OTX
	CastleIsland O
	Aloha
	NYC OTX
	Pratyaksh_Indi
	PSD OTX
	SHELLY_HDX80
	Andover Conf R
	!!!!!1500 test
	ISR-LAB-TPX-S
	Eng400_1
	Eng400_2
	Eng400_3

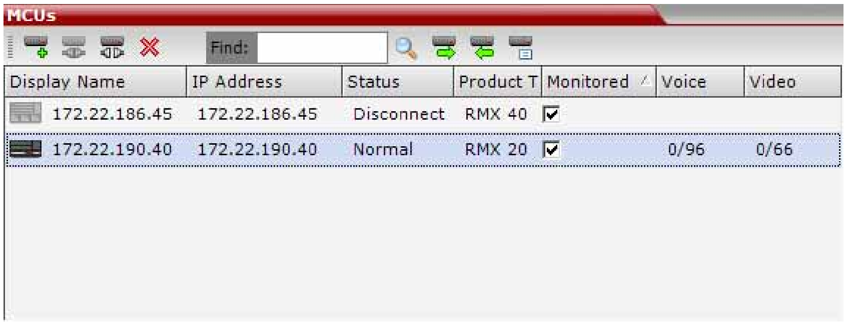
Port Usage: Voice 0 / 48 Video 26 / 78 MCU State: MAJOR

Only one MCU can be selected in the *MCUs* pane. If only one MCU is connected, it is automatically selected. The selected MCU is highlighted.

The menu items, the *Collaboration Server Management* features, the *Address Book* and the *Conference Templates* are all properties of the selected MCU and apply to it.

MCUs Pane

The MCUs pane includes a list of MCUs and a toolbar.



Display Name	IP Address	Status	Product T	Monitored	Voice	Video
172.22.186.45	172.22.186.45	Disconnect	RMX 40	☑		
172.22.190.40	172.22.190.40	Normal	RMX 20	☑	0/96	0/66

For each listed MCU, the system displays the following information:

- MCU *Display Name* - the name of the MCU and its icon according to its type and connection status. The following icons are available:

Table 18-1 MCU Icons and Statuses

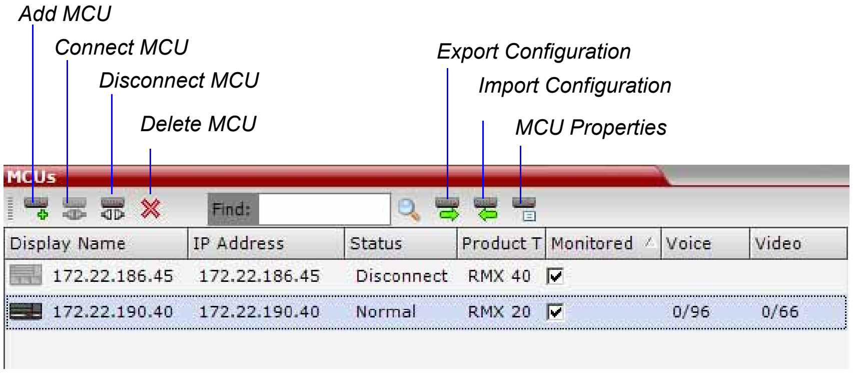
Icon	Description
▭	RealPresence Collaboration Server (RMX) 1500, disconnected.
▬	RealPresence Collaboration Server (RMX) 1500, connected.
▭	RealPresence Collaboration Server (RMX) 2000, disconnected.
▬	RealPresence Collaboration Server (RMX) 2000, connected.
▭	RealPresence Collaboration Server (RMX) 4000, disconnected.
▬	RealPresence Collaboration Server (RMX) 4000, connected.
▭	RealPresence Collaboration Server 800s, disconnected
▬	RealPresence Collaboration Server 800s, connected
▭	RealPresence Collaboration Server Virtual Edition, disconnected
▬	RealPresence Collaboration Server Virtual Edition, connected

- *IP Address* of the MCU's control unit.
- *Status* - The status of the MCU:
  - — *Connected* - the MCU is connected to the *RMX Manager* and can be managed by the *RMX Manager* user.
  - — *Disconnected* - The MCU is disconnected from the *RMX Manager*
  - — *Major* - The MCU has a major problem. MCU behavior could be affected and attention is required.

- *Product Type* - The MCU type: RealPresence Collaboration Server 1800, RealPresence Collaboration Server 1500/2000/4000, RMX 800VE.
  Before connecting to the MCU for the first time, the *Collaboration Server* type is unknown so *RMX* is displayed instead as a general indication.
- *Monitored* - When checked indicates that the conferences running on this MCU are automatically added to the *Conferences* list and monitored. To stop monitoring the conferences running on this MCU and their participants, clear the *Monitored* check box.
- *Video Resources* - The number of video resources that are available for conferencing.
- *Audio Resource*s - The number of audio resources that are available for conferencing.

MCUs Toolbar

The *MCUs* toolbar contains the following buttons:

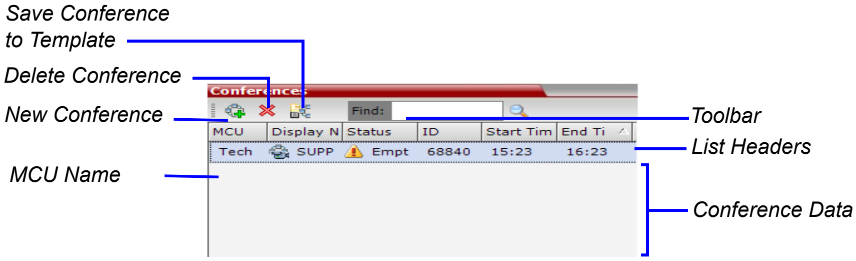


Display Name	IP Address	Status	Product T	Monitored	Voice	Video
172.22.186.45	172.22.186.45	Disconnect	RMX 40	☑		
172.22.190.40	172.22.190.40	Normal	RMX 20	☑	0/96	0/66

Conferences Pane

The *Conferences* pane lists all the ongoing conferences from all the MCUs that are connected and monitored along with their *MCU*, *Status*, *Conference ID*, *Start Time* and *End Time* data. The number of ongoing conferences is displayed in the pane's title.

The *Conferences* list toolbar contains the following buttons:



MCU	Display N	Status	ID	Start Tim	End Ti
Tech	SUPP	Empt	68840	15:23	16:23

Monitoring conferences

New conferences run on MCUs selected for *Monitoring* are automatically added to the *Conferences* list. You can sort the conferences by MCU by clicking the **MCU** column heading in the *Conferences* table. Conferences run on MCUs that are connected but not monitored are not listed.

Using Windows multiple selection methods to select conferences, participants from several conferences running on different MCUs can be listed in the *Participants* list pane.

Starting a new conference

When starting a new conference, you must first select the MCU to run the conference in the MCUs pane.

Collaboration Server Management

The *Collaboration Server Management* pane lists the entities **of the selected MCU** that need to be configured to enable the *Collaboration Server* to run conferences. Only users with Administrators permission can modify these parameters.

The *Collaboration Server Management* pane is divided into two sections:

- **Frequently Used** – parameters often configured monitored or modified.
- **Rarely Used** – parameters configured during initial system set-up and rarely modified afterward.

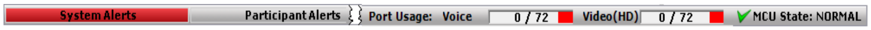
List Pane

The *List* pane displays details of the participants connected to the conferences selected in the *Conferences* pane or the item selected in *Collaboration Server Management* pane. The title of the pane changes according to the selected item.

When selecting an item in the *Collaboration Server Management* pane it applies only to the MCU selected in the MCUs list. In such a case, the system displays the name of the selected MCU in the List pane title.

Status Bar

The *Status Bar* at the bottom of the *RMX Web Client* contains *System* and *Participant Alerts* tabs as well as *Port Usage Gauges* and an *MCU State* indicator.

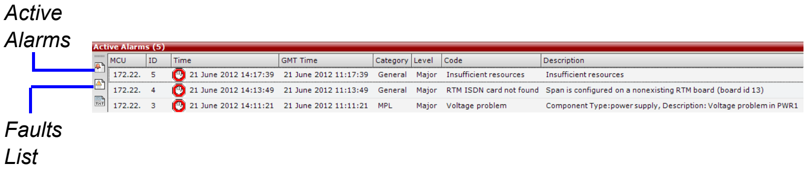


System Alerts

Lists system problems of all connected MCUs (even if the MCU is not monitored). The alert indicator flashes red when at least one system alert is active. The flashing continues until a user with Operator or Administrator permission reviews the list.

The *System Alerts* can be sorted by MCU by clicking the *MCU* header in the *System Alerts* table.

The *System Alerts* pane is opened and closed by clicking the **System Alerts** button in the left corner of the *Status Bar*.



MCU	ID	Time	GMT Time	Category	Level	Code	Description
172.22.	5	21 June 2012 14:17:39	21 June 2012 11:17:39	General	Major	Insufficient resources	Insufficient resources
172.22.	4	21 June 2012 14:13:49	21 June 2012 11:13:49	General	Major	RTM ISDN card not found	Span is configured on a nonexisting RTM board (board id 13)
172.22.	3	21 June 2012 14:11:21	21 June 2012 11:11:21	MPL	Major	Voltage problem	Component Type:power supply, Description: Voltage problem in PWR1

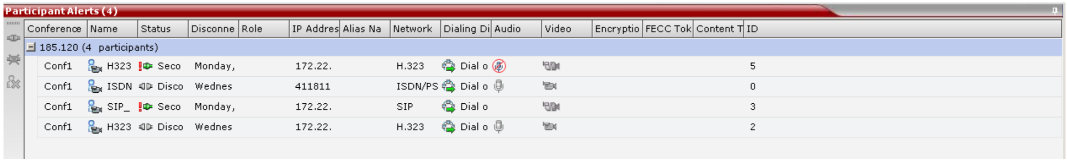
For more information about **Active Alarms** and **Faults List,** see "*System and Participant Alerts*" on page **19-1**.

Participant Alerts

Lists the participants of all monitored MCUs that are experiencing connection problems. The list is sorted by MCU and conference.

The *Participant Alerts* can be sorted by MCU by clicking the *MCU* header in the *Participant Alerts* table.

The *Participant Alerts* pane is opened and closed by clicking the **Participant Alerts** button in the left corner of the *Status Bar*.



Participant Alerts (4)

Conference	Name	Status	Disconne	Role	IP Addres	Alias Na	Network	Dialing Di	Audio	Video	Encryptio	FECC Tok	Content T	ID
185.120 (4 participants)														
Conf1	H323	Seco	Monday,		172.22.		H.323	Dial o						5
Conf1	ISDN	Disco	Wednes		411811		ISDN/PS	Dial o						0
Conf1	SIP_	Seco	Monday,		172.22.		SIP	Dial o						3
Conf1	H323	Disco	Wednes		172.22.		H.323	Dial o						2

Port Usage Gauges

The *Port Usage* gauges display for the selected MCU:

- The total number of *Video* ports in the system according.
- The number of *Video* ports in use.
- The *High Port Usage* threshold.

For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Port Usage Gauges*" on page **3-7**.

MCU State

The *MCU State* indicator displays the status of the selected MCU.

For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*MCU State*" on page **3-7**.

Address Book

Displays the *Address Book* of the **selected** MCU (regardless of its *Monitored* status). The *Address Book* is a list of *Participants* and *Groups* that have been defined on the **selected** *Collaboration Server*.

The information in the *Address Book* can be modified only by an administrator. All *Collaboration Server* users can, however, view and use the *Address Book* to assign participants to conferences.

The name of the selected *Collaboration Server* is displayed in the title of the Address Book pane. For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Address Book*" on page **3-8**.

Conference Templates

Conference Templates enable administrators and operators to create, save, schedule and activate identical conferences.

The *Conference Templates* pane lists the Conference Templates that have been defined on the **selected** *Collaboration Server* (regardless of its *Monitored* status).

The *Conference Templates* pane is initially displayed as a closed tab. The name of the selected *Collaboration Server* and the number of saved *Conference Templates* is indicated on the tab.

For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Conference Templates*" on page **3-9**.

Adding MCUs to the MCUs List

The *RMX Manager* can connect to one or several *Collaboration Server*s simultaneously. If the site's configuration includes more than one MCU, or when a new MCU is added to your configuration, and you want to monitor and control all MCUs from within the same window, you must add the MCU to the MCUs list.

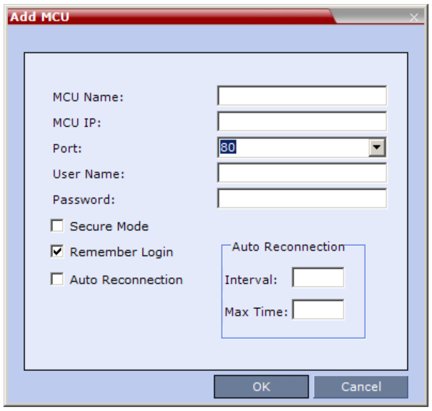
> The *Collaboration Server* must be installed and its IP addresses properly configured in the Management Network Service before defining its connection parameters in the *RMX Manager* application.

To add the MCU to the list of MCUs being managed, define the MCU's connection parameters.

To add a *Collaboration Server* unit:

1 On the *MCUs* toolbar, click the **Add MCU** button to add an MCU to the MCU list. The *Add MCU* dialog box opens.

2 Define the following parameters:



Add MCU

MCU Name:

MCU IP:

Port: 80

User Name:

Password:

☐ Secure Mode

☑ Remember Login

☐ Auto Reconnection

Auto Reconnection

Interval:

Max Time:

Table 19 *MCU Properties*

Field	Description
MCU Name	Enter the name of the MCU on the network.
MCU IP	Enter the IP address of the MCU's Control Unit. The IP address must be identical to the one configured in the MCU during first entry Configuration. For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*To obtain the IP address of the Virtual Machine:"* on page **2-21**.
Port	Enter the number of the port used for communication and data transactions between the *Collaboration Server* unit and the *RMX Manager.* For standard connection, enter **8080**. For a Secured connection (using TLS or SSL), enter **4433**. **Note**: When using the *RMX Manager* with other *Collaboration Server* models, the standard connection port is 80 and the secured connection port is 443.

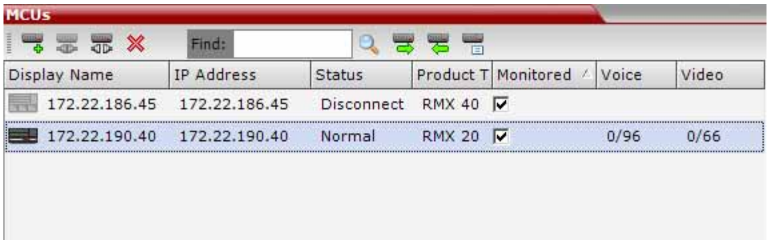
Table 19 MCU Properties (Continued)

Field	Description
Username	Enter the user name with which you will login to the MCU. A User with this name must be defined in the *Collaboration Server* Users list. The system is shipped with a default User whose name is POLYCOM.
Password	Enter the password as defined for the user name with which you will login to the MCU. The system is shipped with a default User whose password is POLYCOM.
Secure Mode	**Optional**. Select this check box to connect to the *Collaboration Server* with SSL and work in Secure Mode.
Remember Login	This check box is automatically selected, and it enables the usage of the user name and password entered in this dialog box when connecting to the *Collaboration Server*. If this check box is cleared, the user is prompted for the user name and password when connecting to this *Collaboration Server* unit.
Auto Reconnection	Select this check box to automatically reconnect to the *Collaboration Server* if the connection between the *RMX Manager* and the MCU is broken.
Interval	Enter time in seconds between reconnect ion attempts to the *Collaboration Server*. For example, if you enter 10, the system will wait 10 seconds between the connection attempts.
Max Time	Enter the maximum amount of time in seconds that the *Collaboration Server* is allowed to try to reconnect. If the *Collaboration Server* reconnects before the allotted time frame the count down timer is halted. For example, if you enter 100, the system will stop trying to reconnect if it has failed to do so within 100 seconds.

3 Click **OK**.
The MCU is added to the MCUs pane.

4 If required, repeat steps 1-3 to define additional *Collaboration Server* units.

The *MCUs* pane contains the list of all defined MCUs.



Display Name	IP Address	Status	Product T	Monitored	Voice	Video
172.22.186.45	172.22.186.45	Disconnect	RMX 40	☑		
172.22.190.40	172.22.190.40	Normal	RMX 20	☑	0/96	0/66

Starting a Conference

There are several ways to start a conference:

- Clicking the *New Conference* button in the *Conferences* pane. For more information, see *“Starting a Conference from the Conferences Pane”* on page 13.

- Dialing in to a Meeting Room defined on any of the MCUs.
  - — A Meeting Room is a conference that is saved on the MCU. It remains in passive mode until it is activated by the first participant, or the meeting organizer dialing in. For more information about Meeting Rooms, see "*Meeting Rooms"* on page **6-1**.
- Dialing in to an Ad Hoc Entry Queue defined on one of the MCUs which is used as the access point to the MCU.

  For a detailed description of Ad Hoc Entry Queues, see "*Entry Queues"* on page **7-1**.
- Start any *Conference Template* saved in the *Conference Templates* list.
  For more information, see "*Starting an Ongoing Conference From a Template"* on page **18-14**.

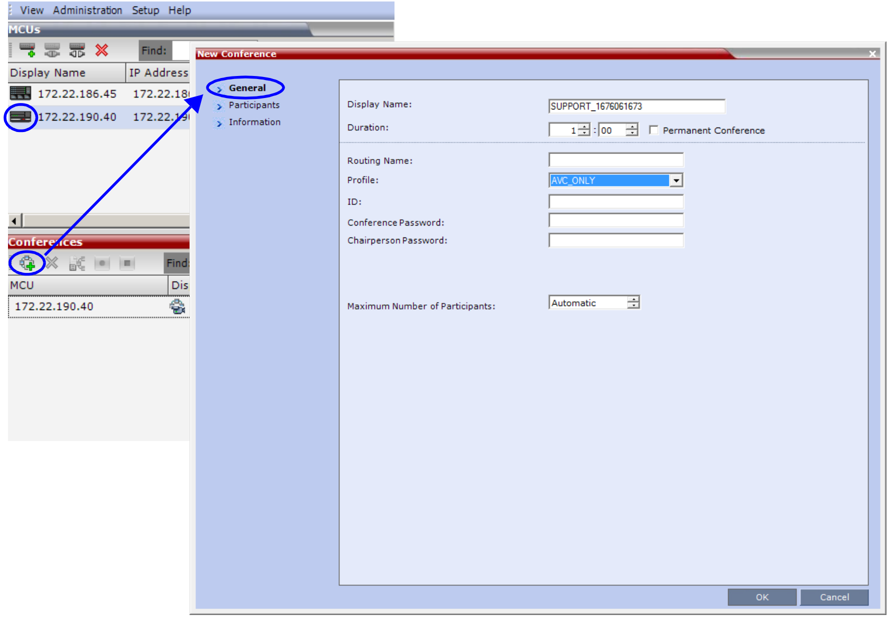
Starting a Conference from the Conferences Pane

To start a conference from the Conference pane:

1 In the *MCUs* pane, select the MCU to run the conference.

2 In the *Conferences* pane, click the **New Conference** (⊕) button.

The *New Conference – General* dialog box opens.



Display Name: SUPPORT_1676061673

Duration: 1 : 00 □ Permanent Conference

Routing Name:

Profile: AVC_ONLY

ID:

Conference Password:

Chairperson Password:

Maximum Number of Participants: Automatic

The system displays the conference's default *Name*, *Duration* and the default *Profile*, which contains the conference parameters and media settings.

The *Collaboration Server* automatically allocates the conference *ID*, when the conference starts.

In most cases, the default conference *ID* can be used and you can just click **OK** to launch the conference. If required, you can enter a conference *ID* before clicking **OK** to launch the conference.

If you are the meeting chairperson or organizer using the *RMX Web Client* to start your own meeting, you need to communicate the default conference ID (or the one you created) to the other conference participants so they can dial in.

You can use the *New Conference - General* dialog box to modify the conference parameters. If no defined participants are to be added to the conference, or you do not want to add additional information, click **OK**.

For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Starting an AVC CP Conference from the Conferences Pane"* on page **3-13**.

Starting an Ongoing Conference From a Template

An ongoing conference can be started from any Conference Template saved in the *Conference Templates* list of the selected MCU.

To start an ongoing conference from a Template:

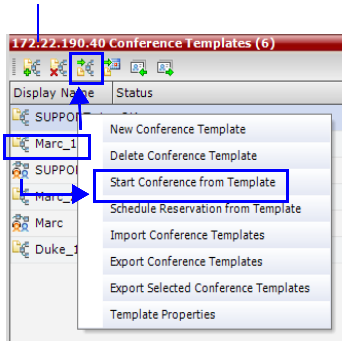
1 In the *MCUs* pane, select the MCU to run the conference.

1 In the *Conference Templates* list, select the Template you want to start as an ongoing conference.

2 Click the **Start Conference from Template** () button to start a conference or

Schedule Reservation from Template () button to schedule a reservation.
or
Right-click and select **Start Conference from Template** to start an ongoing conference or **Schedule Reservation from Template** to schedule a reservation.

Name of selected MCU


172.22.190.40 Conference Templates (6)
Display Name
Status
SUPPO
Marc_1
SUPPO
Marc_
Marc
Duke_1
New Conference Template
Delete Conference Template
Start Conference from Template
Schedule Reservation from Template
Import Conference Templates
Export Conference Templates
Export Selected Conference Templates
Template Properties

The conference is started.

For detailed description of *Conference Templates*, see "*Using Conference Templates"* on page **10-2**.

Monitoring Conferences

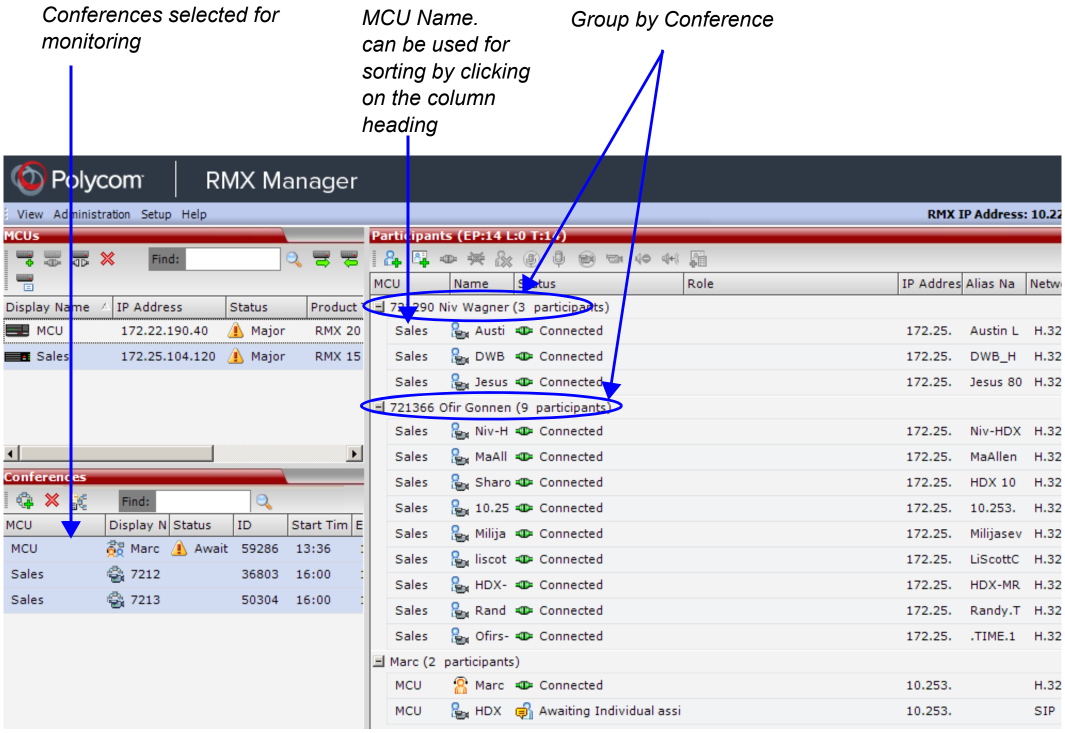
When MCUs are connected to the *RMX Manager* they are automatically monitored, that is, any ongoing conference that is started on that MCU is automatically added to the Conferences pane and its participants are monitored.

To list participants from several conferences (running on the same or different MCUs):

>> In the *Conferences* pane, using Windows multiple selection methods, select the conferences whose participants you want to list.

The participants are displayed in the *Participants* list pane.

By default, the participants are grouped by conferences, and the name of the MCU is displayed in the first column of the properties table, enabling sorting according to MCU name.



Conferences selected for monitoring

MCU Name. can be used for sorting by clicking on the column heading

Group by Conference

Display Name	IP Address	Status	Product
MCU	172.22.190.40	⚠ Major	RMX 20
Sales	172.25.104.120	⚠ Major	RMX 15

MCU	Display N	Status	ID	Start Tim	E
MCU	Marc	⚠ Await	59286	13:36	
Sales	7212		36803	16:00	
Sales	7213		50304	16:00	

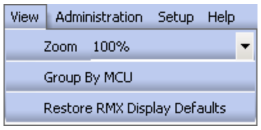
MCU	Name	Status	Role	IP Addres	Alias Na	Netw
⊟ 72[illegible]290 Niv Wagner (3 participants)						
Sales	Austi	Connected		172.25.	Austin L	H.32
Sales	DWB	Connected		172.25.	DWB_H	H.32
Sales	Jesus	Connected		172.25.	Jesus 80	H.32
⊟ 721366 Ofir Gonnen (9 participants)						
Sales	Niv-H	Connected		172.25.	Niv-HDX	H.32
Sales	MaAll	Connected		172.25.	MaAllen	H.32
Sales	Sharo	Connected		172.25.	HDX 10	H.32
Sales	10.25	Connected		172.25.	10.253.	H.32
Sales	Milija	Connected		172.25.	Milijasev	H.32
Sales	liscot	Connected		172.25.	LiScottC	H.32
Sales	HDX-	Connected		172.25.	HDX-MR	H.32
Sales	Rand	Connected		172.25.	Randy.T	H.32
Sales	Ofirs-	Connected		172.25.	.TIME.1	H.32
⊟ Marc (2 participants)						
MCU	Marc	Connected		10.253.		H.32
MCU	HDX	Awaiting Individual assi		10.253.		SIP

Grouping the Participants by MCU

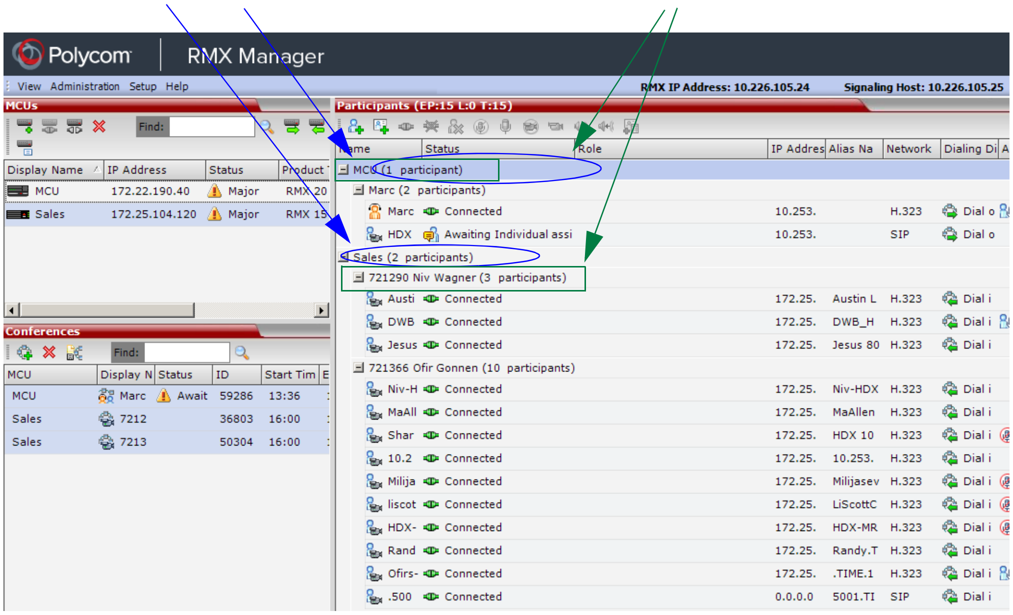
The Participants can be grouped by MCU and then by conferences.

To change the display mode for the Participants pane:

>> On the *Collaboration Server Menu*, click **View > Group by MCU**.



The *Participants* pane display changes accordingly.



Group by MCU

Group by Conference

Polycom
RMX Manager
View Administration Setup Help
RMX IP Address: 10.226.105.24
Signaling Host: 10.226.105.25
MCUs
Find:
Conferences
Find:
Participants (EP:15 L:0 T:15)

Display Name	IP Address	Status	Product
MCU	172.22.190.40	Major	RMX 20
Sales	172.25.104.120	Major	RMX 15

MCU	Display N	Status	ID	Start Tim	E
MCU	Marc	Await	59286	13:36	
Sales	7212		36803	16:00	
Sales	7213		50304	16:00	

Name	Status	Role	IP Addres	Alias Na	Network	Dialing Di	A
MCU (1 participant)							
Marc (2 participants)							
Marc	Connected		10.253.		H.323	Dial o	
HDX	Awaiting Individual assi		10.253.		SIP	Dial o	
Sales (2 participants)							
721290 Niv Wagner (3 participants)							
Austi	Connected		172.25.	Austin L	H.323	Dial i	
DWB	Connected		172.25.	DWB_H	H.323	Dial i	
Jesus	Connected		172.25.	Jesus 80	H.323	Dial i	
721366 Ofir Gonnen (10 participants)							
Niv-H	Connected		172.25.	Niv-HDX	H.323	Dial i	
MaAll	Connected		172.25.	MaAllen	H.323	Dial i	
Shar	Connected		172.25.	HDX 10	H.323	Dial i	
10.2	Connected		172.25.	10.253.	H.323	Dial i	
Milija	Connected		172.25.	Milijasev	H.323	Dial i	
liscot	Connected		172.25.	LiScottC	H.323	Dial i	
HDX-	Connected		172.25.	HDX-MR	H.323	Dial i	
Rand	Connected		172.25.	Randy.T	H.323	Dial i	
Ofirs-	Connected		172.25.	.TIME.1	H.323	Dial i	
.500	Connected		0.0.0.0	5001.TI	SIP	Dial i	

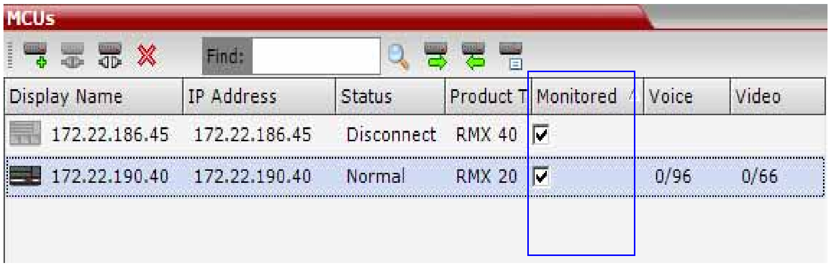
To toggle between the two display modes, click **View > Group by MCU.**

Start Monitoring/Stop Monitoring

By default, all conferences running on connected *Collaboration Servers* are monitored.

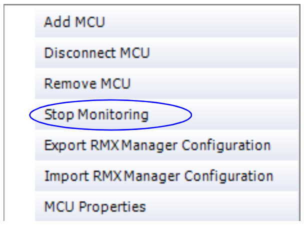
You can stop the automatic monitoring of conferences on a specific MCU in one of the following methods:

- By clearing the check box in the *Monitored* column in the *MCUs* pane.



Display Name	IP Address	Status	Product T	Monitored	Voice	Video
172.22.186.45	172.22.186.45	Disconnect	RMX 40	☑		
172.22.190.40	172.22.190.40	Normal	RMX 20	☑	0/96	0/66

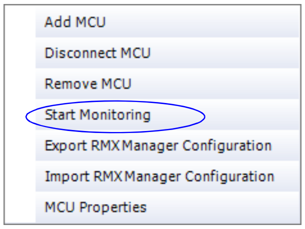
- Right-clicking the MCU icon and selecting **Stop Monitoring**.



Add MCU
Disconnect MCU
Remove MCU
Stop Monitoring
Export RMX Manager Configuration
Import RMX Manager Configuration
MCU Properties

The check box is cleared in the Monitored column.

To start monitoring again, click the check box in the *Monitored* column in the *MCUs* pane, or right-clicking the MCU icon and selecting **Start Monitoring**.


Add MCU
Disconnect MCU
Remove MCU
Start Monitoring
Export RMX Manager Configuration
Import RMX Manager Configuration
MCU Properties

Modifying the MCU Properties

You can view the currently defined MCU settings, and modify them when required, for example, change the MCU name, IP address or Secured mode.

Use this procedure to add the *Username* and *Password* to the properties of the MCU that was automatically added to the MCU list when installing the *RMX Manager*. This enables automatic login when connecting the MCU to the *RMX Manager*.

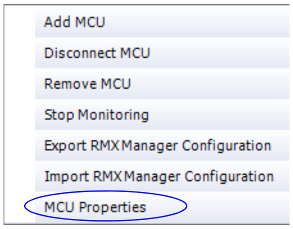
You can modify the MCU properties when the MCU is connected or disconnected.

To view and/or modify the MCU Properties:

1 Use one of the following methods:

a Select the MCU to disconnect and click the **MCU Properties** ▤ button.

b Right-click the MCU icon and then click **MCU Properties**.


Add MCU
Disconnect MCU
Remove MCU
Stop Monitoring
Export RMX Manager Configuration
Import RMX Manager Configuration
MCU Properties

The *MCU Properties* dialog box opens.

2 Define/modify the required parameters. For details, see *"MCU Properties"* on page 12.

3 Click **OK**.

Disconnecting an MCU

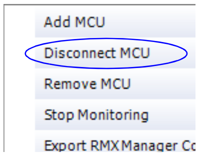
An MCU can be disconnected from the *RMX Manager*, without removing it from the *MCUs* list.

To disconnect an MCU:

1 Use one of the following methods:

a Select the MCU to disconnect and click the **Disconnect MCU** button.

b Right-click the MCU icon and then click **Disconnect MCU**.



The MCU icon changes to disconnected and any ongoing conference running on that MCU will not be monitored in this *RMX Manager*; they are removed from the *Conferences* pane. This MCU can still be monitored and controlled by other users.

Removing an MCU from the MCUs Pane

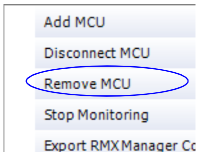
An MCU can be removed from the *RMX Manager*. This function should be used if the MCU hardware was disconnected and removed from the network.

To Remove an MCU from the list:

1 Use one of the following methods:

a Select the MCU to disconnect and click the **Delete** ✖ button.

b Right-click the MCU icon and then click **Remove MCU**.



A confirmation message is displayed.

2 Click **OK** to confirm or **Cancel** to abort the operation.

The MCU icon is removed from the MCUs pane.

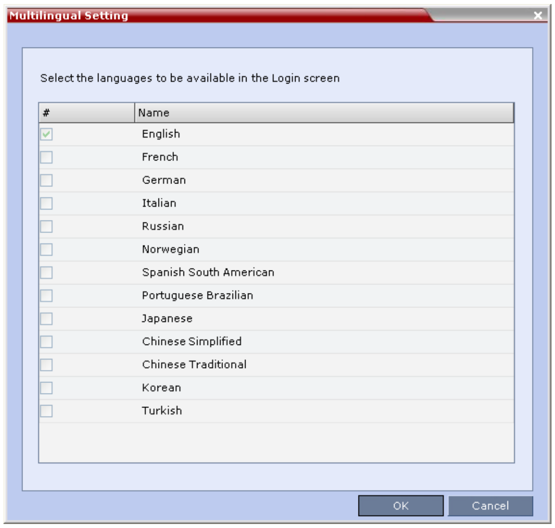
Changing the RMX Manager Language

You can change the language of the *RMX Manager* menus and dialog boxes. Only one language can be selected at a time and the *RMX Manager* application must be restarted after changing the display language.

To select a language:

1 On the *RMX Manager* menu, click **Setup > Customize Display Settings > Multilingual Settings**.

The *Multilingual Settings* dialog box opens, displaying the current language selection.



Multilingual Setting

Select the languages to be available in the Login screen

#	Name
☑	English
□	French
□	German
□	Italian
□	Russian
□	Norwegian
□	Spanish South American
□	Portuguese Brazilian
□	Japanese
□	Chinese Simplified
□	Chinese Traditional
□	Korean
□	Turkish

2 Click the check box of the required language. Only one language can be selected.

3 Click **OK**.

4 Restart the *RMX Manager* application to implement the language change.

Import/Export RMX Manager Configuration

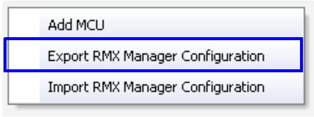
The *RMX Manager* configuration that includes the MCU list and the multilingual selection can be save to any workstation/PC on the network and imported to any *Multi-RMX Manager* installed in the network. This enables the creation of the MCUs list once and distributing it to all *RMX Manager* installations on the network.

In addition, when upgrading to a previous version, the MCU list is deleted, and can be imported after upgrade.

The exported file is save in XML format and can be edited in any text editor that can open XML files.

To Export the RMX Manager Configuration:

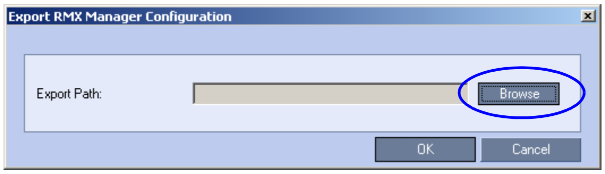
1 In the *Multi-RMX Manager*, click the **Export RMX Manager Configuration** button in the toolbar, or right-click anywhere in the MCUs pane and then click **Export RMX Manager Configuration**.



Add MCU
Export RMX Manager Configuration
Import RMX Manager Configuration

The *Export RMX Manager Configuration* dialog box opens.

2 Click the **Browse** button to select the location of the save file, or enter the required path in the *Export Path* box.

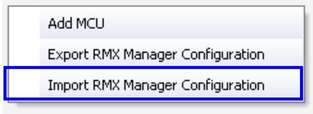


The selected file path is displayed in the *Export Path* box.

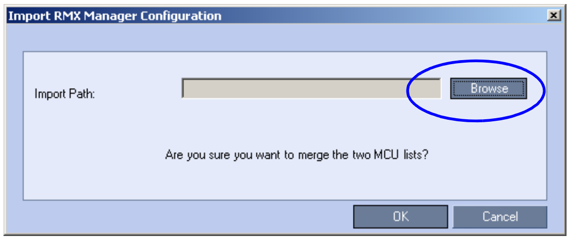
3 Click **OK** to export the *RMX Manager* configuration.

To Import the RMX Manager Configuration:

1 In the *Multi-RMX Manager*, click the **Import RMX Manager Configuration** button in the toolbar, or right-click anywhere in the MCUs pane and then click **Import RMX Manager Configuration**.

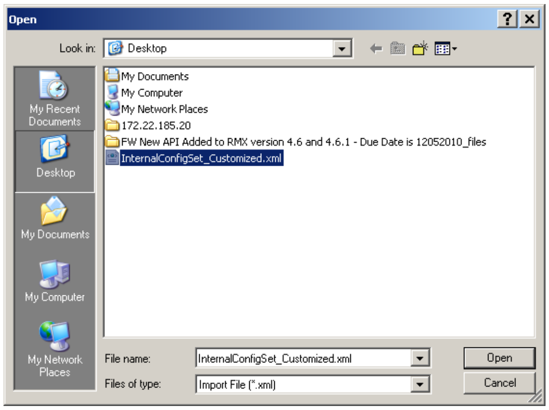

Add MCU
Export RMX Manager Configuration
Import RMX Manager Configuration

The *Import RMX Manager Configuration* dialog box opens.



Are you sure you want to merge the two MCU lists?

2 Click the **Browse** button to select the saved file, or enter the required path in the *Export Path* box.
The *Open* dialog box is displayed.



File name: InternalConfigSet_Customized.xml
Files of type: Import File (*.xml)

3 Select the XML file previously saved, and click the **Open** button.

The selected file path is displayed in the *Import Path* box.

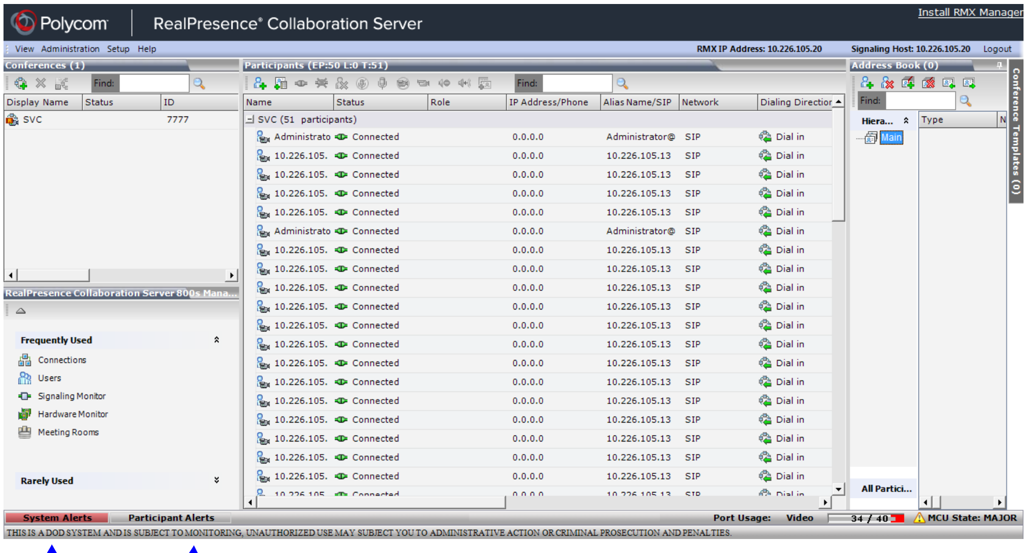
4 Click OK to import the file.

19

Collaboration Server Administration and Utilities

System and Participant Alerts

The MCU alerts users to any faults or errors the MCU encountered during operation. Two indication bars labeled System Alerts and Participant Alerts signal users of system errors by blinking red in the event of an alert.



SVC (51 participants)

Name	Status	Role	IP Address/Phone	Alias Name/SIP	Network	Dialing Direction
Administrato	Connected		0.0.0.0	Administrator@	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
Administrato	Connected		0.0.0.0	Administrator@	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in
10.226.105.	Connected		0.0.0.0	10.226.105.13	SIP	Dial in

THIS IS A DOD SYSTEM AND IS SUBJECT TO MONITORING, UNAUTHORIZED USE MAY SUBJECT YOU TO ADMINISTRATIVE ACTION OR CRIMINAL PROSECUTION AND PENALTIES.

System Alerts indication bar

Participant Alerts indication bar

The *System Alerts* indication bar blinks red prompting the user to view the active alarms. Once viewed, the *System Alerts* indication bar becomes statically red until the errors have been resolved in the MCU.

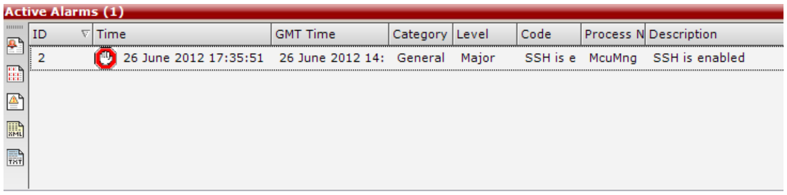
The *Participants Alerts* indication bar blinks red indicating participant connection difficulties in conferences. Once viewed, the *Participant Alerts* indication bar becomes statically red until the errors have been resolved in the MCU.

System Alerts

System Alerts are activated when the system encounters errors such as a general or card error. The system errors are recorded by the *Collaboration Server* and can be generated into a report that can be saved in *.txt format.

To view the System Alerts list:

1 Click the red blinking **System Alerts** indication bar.
The *Active Alarms* pane opens. This screen indicates what events have not been resolved.



Active Alarms (1)

ID	Time	GMT Time	Category	Level	Code	Process N	Description
2	⛔ 26 June 2012 17:35:51	26 June 2012 14:	General	Major	SSH is e	McuMng	SSH is enabled

The following columns appear in the *Active Alarms* pane:

***Table 19-1** Active Alarms Pane Columns*

Field	Description
ID	An identifying number assigned to the system alert.
Time	Lists the local date and time that the error occurred. This column also includes the icon indicating the error level (as listed in the level column).
GMT Time	Lists the date and time according to Greenwich Mean Time (GMT) that the error occurred.
Category	Lists the type of error. The following categories may be listed: • **File** – indicates a problem in one of the files stored on the MCU's hard disk. • **Card** – indicates problems with a card. • **Exception** – indicates software errors. • **General** – indicates a general error. • **Assert** – indicates internal software errors that are reported by the software program.
Category (cont.)	• **Startup** – indicates errors that occurred during system startup. • **Unit** – indicates problems with a unit.
Level	Indicates the severity of the problem, or the type of event. There are three fault level indicators: ⛔ – Major Error ❗ – System Message ⏳ – Startup Event
Code	Indicates the problem, as indicated by the error category.
Process Name	Lists the type of functional process involved.
Description	When applicable, displays a more detailed explanation of the cause of the problem.

For more information about the Active Alarms, see *Appendix B: "Active Alarms"* on page **B-1**.

2 Click one of the following two buttons to view its report in the *System Alerts* pane:

	Active Alarms (default) – this is the default reports list that is displayed when clicking the System Alerts indication bar. It displays the current system errors and is a quick indicator of the MCU status.
	Faults Full List - A list of all system faults. Note: Viewed when logged in as a special support user.
	Faults List – a list of faults that occurred previously (whether they were solved or not) for support or debugging purposes.

3 To save the *Active Alarms, Faults Full List* or *Faults* report:

— to a text file, click the **Save to Text** button

— to an XML file, click the **Save to XML** button

> The **Save to XML** button is only available when logged in as a special support user.

The *Save* dialog window opens.

4 Select a destination folder and enter the file name.

5 Click **Save**.

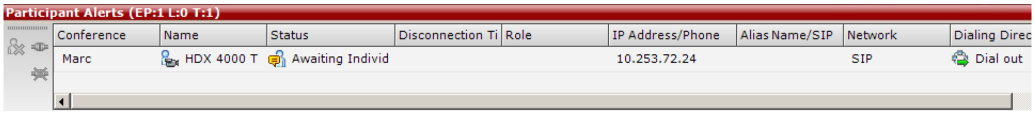
Participant Alerts

Participant Alerts enables users, participants and conferences to be prompted and currently connected. This includes all participants that are disconnected, idle, on standby or waiting for dial-in. Alerts are intended for users or administrators to quickly see all participants that need their attention.

To view the Participants Alerts list:

1 Click the red blinking **Participants Alerts** indication bar.

The *Participant Alerts* pane opens.



Participant Alerts (EP:1 L:0 T:1)

Conference	Name	Status	Disconnection Ti	Role	IP Address/Phone	Alias Name/SIP	Network	Dialing Direc
Marc	HDX 4000 T	Awaiting Individ			10.253.72.24		SIP	Dial out

> The *Participant Alerts* pane displays similar properties to that of the *Participant List* pane. For more information, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, *"Participant Level Monitoring"* on page 3-49.

2 To resolve participant issues that created the *Participant Alerts*, the administrator can either **Connect** , **Disconnect** or **Delete** a participant.

RMX Time

To ensure accurate conference scheduling, the MCU has an internal clock that can function in standalone mode, or in synchronization with up to three *Network Time Protocol* (*NTP*) servers.

Guidelines

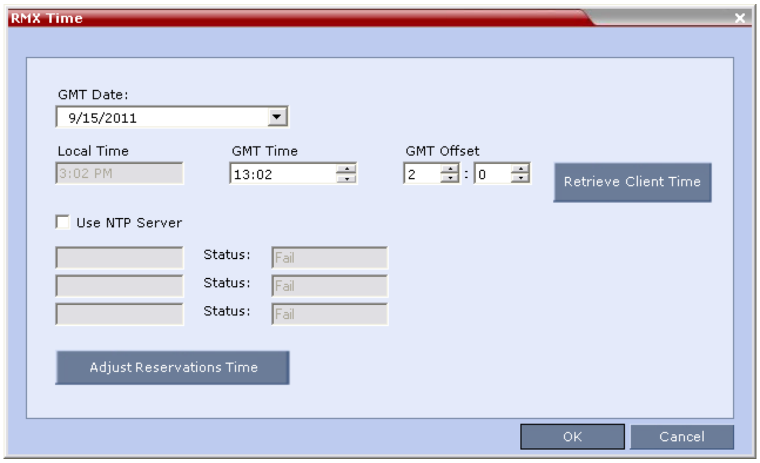
- *NTP Version 4* is the only supported protocol.
- If applicable, daylight saving adjustments must be implemented by the administrator whether the MCU is in standalone mode or synchronized with *NTP Servers*.

Altering the clock

The MCU's date and time can be set manually or enabled to synchronize with external *NTP* servers.

To Alter the MCU Time:

1 On the *Collaboration Server* menu, click **Setup > RMX Time** to open the *RMX Time* dialog box.



GMT Date: 9/15/2011

Local Time 3:02 PM

GMT Time 13:02

GMT Offset 2 : 0

□ Use NTP Server

Status: Fail

Status: Fail

Status: Fail

2 View or modify the following fields:

Table 19-2 RMX Time – Fields Properties

Field	Description
GMT Date	The date at Greenwich, UK.
Local Time	The MCU's local time settings, are calculated from the *GMT Time* and the *GMT Offset*.
GMT Time	The MCU's current *GMT Time* settings. Select the *Up* or *Down* arrows to alter the *GMT Time* on the MCU.

Table 19-2 *RMX Time – Fields Properties (Continued)*

Field	Description
GMT Offset	The time zone difference between Greenwich and the MCU's physical location in hours and minutes. Select the *Up* or *Down* arrows to alter the *GMT Offset* time on the MCU. To enter a negative offset either type a minus in the hour box or use the down arrow and decrease the offset below zero.
Retrieve Client Time	Click this button to automatically update the MCU's *GMT Date, Time* and *Offset* to match that of the workstation.
Use NTP Server	Select this check box to synchronize the time with up to three *NTP* servers. When selected, the manual *GMT Date* and *GMT Time* setting options are disabled. The *GMT Offset* fields are still active. To implement this mode an external connection to an *NTP* server must be enabled. Enter the IP addresses of the required *NTP* servers in order of precedence. The *Status* field indicates whether registration with the *NTP Server* failed or succeeded. Note: The*Collaboration Server* will not use a time source such as a Windows-based, W32Time service (SNTP) time service. Only full-featured (below Stratum 16) NTP Servers are considered sufficiently reliable for high-accuracy timing environments.
Adjust Reservations Time (*Button*)	Not supported in the RealPresence Collaboration Server.

After resetting the MCU a delay may occur when synchronizing with the external NTP server.

Resource Management

Resource Capacity

The following table describes the resource capacity allocations for the*RealPresence Collaboration Server Virtual Edition* per Resource type.

Table 19-4 *System Resource Capacity Per Resource Type*

Resource Type	Number of Resources
VoIP Ports	120
CIF Ports	40
SD Ports (4CIF)	40
HD 720p30 Ports	20

Table 19-4 System Resource Capacity Per Resource Type

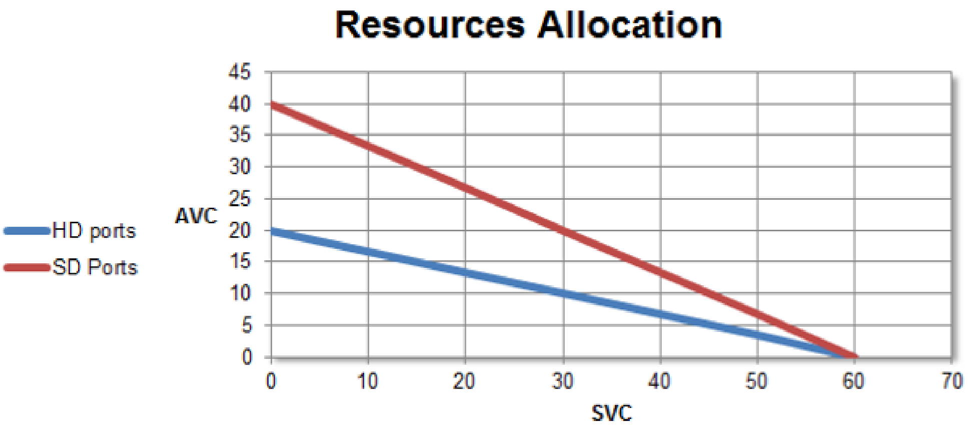
Resource Type	Number of Resources
VGA RTV Ports	20
SVC Only Ports	60

One SVC resource is equivalent to one AVC CIF resource.

In a mixed CP and SVC conference, video resources are used according to the amount of both AVC and SVC participants in the conference. The ratio of resources in a mixed conference is one AVC HD (720p30) video resource to three SVC video resources, meaning for each AVC HD video resource, three SVC video resources can be allocated.

For example, in a mixed AVC/SVC conference, 10 HD AVC ports and 30 SVC ports can be used, maintaining ratio of one HD port to three SVC ports.

The following diagram illustrates the amount of AVC to SVC port resources that are used in a mixed AVC/SVC conference:



Resource Usage

SVC Conferencing

During a *SVC* conference, each SVC-endpoint uses one video port. One SVC resource is equivalent to one AVC CIF resource. When sharing content an additional video resource is used.

AVC Conferencing - Continuous Presence

Video resources usage varies according to the video resolution used by the endpoints. The higher the video resolution (quality), the greater the amount of video resources consumed by the MCU. Table 19-7 shows the number of video resources used for each resolution.

Table 19-5 CP Video Resource Usage vs. Resolution

Resolution/fps	HD Video Resources Used
CIF/30	1/3
QCIF/30	
SIF/30	
WCIF/25	1/3
WSIF/30	
432X336/30	
SD/15	
WSD/15	
4CIF/15	1/3
WSD/30	2/3
4CIF/30	
4SIF/30	
WVGA/30	
WVGA/25	
480X352/30	
SD/30	
WSD/60	
HD720p/30	

Table 19-6 HD Video Resource Usage vs. Resolution

Resolution/fps	HD Video Resources Used
H.261 CIF 30fps	2/3
H.263 CIF 30fps	1/3
H.263 4CIF 30fps	1/3
H.264 CIF 30fps	1/3
H.264 CIF 60fps	1/3
H.264 4CIF 30fps	1/3
H.264 4CIF 60fps	1/3

Table 19-6 HD Video Resource Usage vs. Resolution

Resolution/fps	HD Video Resources Used
H.264 720p 30fps	1/3
H.264 720p 60fps	2/3
H.264 1080p 25/30fps	2/3
H.264 1080p 50/60fps	4/3

AVC Conferencing - Voice

One CIF video resource equals 3 voice resources. All resources are taken from the same pool of video resources.

Forcing Video Resource Allocation to CIF Resolution

You can set the MCU to allocate one CIF video resource to an endpoint, regardless of the resolution determined by the Conference Profile parameters. This forcing saves resources and enables more endpoints to connect to conferences.

The forcing is done by modifying the system configuration and it applies to all conferences running on the MCU.

You can specify the endpoint types for which resource allocation can be forced to CIF resource, enabling other types of endpoints to use higher resolutions in the same conference. For example, you can force the system to allocate one CIF video resource to CMAD and VSX endpoints while HDX endpoints can connect using SD or HD video resources.

Once the endpoint connects to the conference, its type is identified by the Collaboration Server and, if applicable, the Collaboration Server will connect it using one CIF resource, even if a higher resolution can be used.

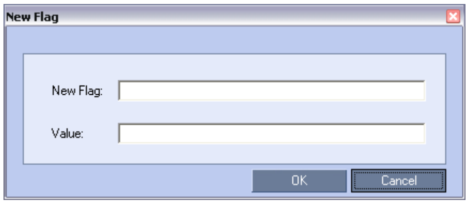
To force CIF resource:

1 On the Collaboration Server menu, click **Setup > System Configuration**.

The *System Flags* dialog box opens.

2 In the *MCMS_PARAMETERS* tab, click the **New Flag** button.

The *New Flag* dialog box is displayed.



3 In the *New Flag* field enter the flag name: **FORCE_CIF_PORT_ALLOCATION**

4 In the *Value* field enter the product type to which the CIF resource should be allocated. Possible values are:

— **CMA Desktop** for CMA desktop client

— **VSX nnnn** where nnnn represents the model number for example, VSX 8000.

You can define several endpoint types, listing them one after the other separated by semicolon (;).
For example, CMA Desktop;VSX 8000.

5 Click **OK**.
The new flag is added to the flags list.

Reset the MCU for changes to take effect. For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Administrator's Guide*, *"Resetting the Collaboration Server"* on page **19-57**.

To cancel the forcing of CIF resource:

1 On the Collaboration Server menu, click **Setup > System Configuration**.

The *System Flags* dialog box opens.

2 In the *MCMS_PARAMETERS* tab, double-click or select the flag **FORCE_CIF_PORT_ALLOCATION** and click the **Edit Flag** button.

3 In the *New Value* field, clear the value entries.

4 Click **OK**.

Reset the MCU for changes to take effect. For more details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Administrator's Guide*, *"Resetting the Collaboration Server"* on page **19-57**.

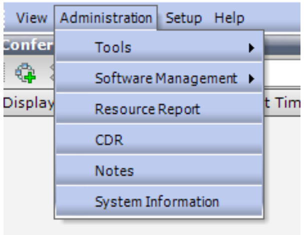
Resource Report

When viewing the Collaboration Server resource report, the resource allocations are described in AVC HD units. A port ratio of 1 AVC HD port will equal 2 AVC SD ports, which equals 3 SVC ports. This signifies that when the Collaboration Server is reporting the available capacity, it will appropriately round up the remaining capacity to the nearest whole value of available ports.

The *Resource Report* displays the real time resource usage. The *Resource Report* includes a graphic representation of the resource usage. One resource report is available for all resource usage including SVC-based endpoints.

Displaying the Resource Report

1 In the main toolbar, click **Administration > Resource Report**.



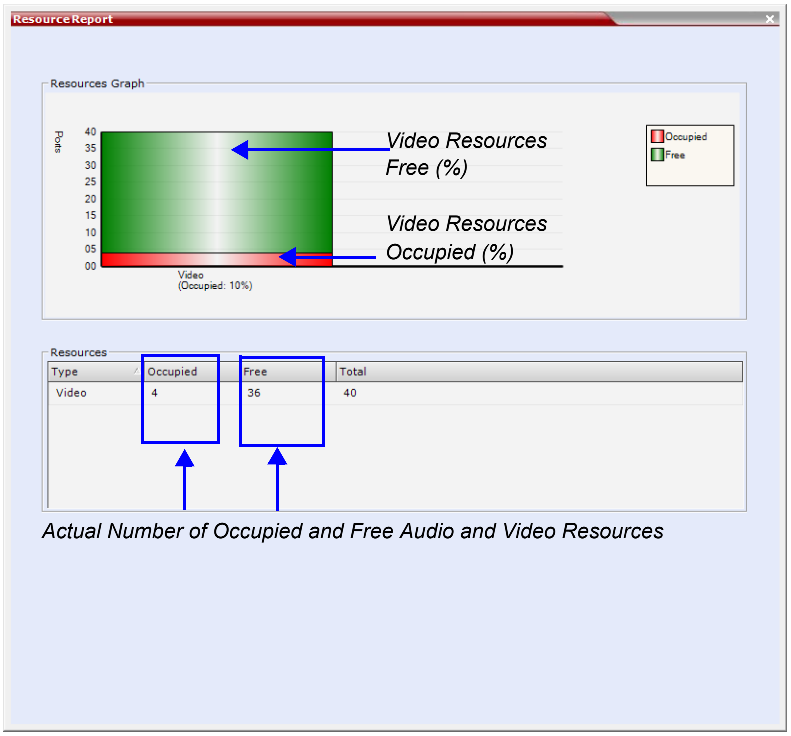
For each resource type, the Resource Report includes the following columns:

***Table 19-8** Resource Report Fields Parameters*

Column	Description
Type	This is always **Video**. This applies to both AVC and SVC-based endpoints (and resources).
Occupied	The number of MCU resources that are used by connected AVC and SVC-based participants or reserved for defined participants.
Free	The number of MCU resources available for connecting AVC and SVC-based endpoints.
Total	The *Total* column displays the total number of resources of that type (*Occupied* and *Free*).

•

The *Resource Report* dialog box is displayed, showing the resource usage according to the *Resource Capacity Mode*.

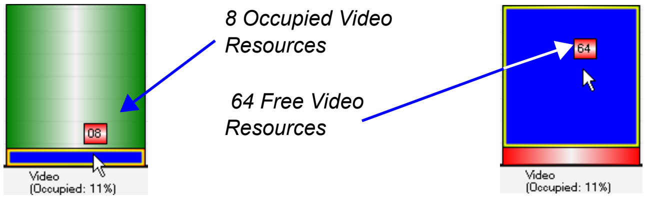


Type	Occupied	Free	Total
Video	4	36	40

Actual Number of Occupied and Free Audio and Video Resources

Resource usage is displayed for video resources only. They are displayed as percentages of the total resources.

The actual number of occupied or free resources can also be displayed by moving the cursor over the columns of the bar graph. Moving the cursor over the *Video* bar displays the following:



When viewing the Collaboration Server resource report for mixed CP and SVC conferences, the resource allocations are described in AVC HD720p30 units. A port ratio of 1 AVC HD port will equal 2 AVC SD ports, which equals 3 SVC ports (in a non-mixed conference).

When the Collaboration Server is reporting the available capacity, it will appropriately round up the remaining capacity to the nearest whole value of available ports. For example, one SVC endpoint in a conference is equal to *1/3* of the resource value. The resource report displays this as one full resource used. Two SVC endpoints is equal to *2/3* of the resource value. Therefore, the resource report displays this as one full resource used, and so forth.

The following table explains the actual resource capacity utilization for both CP only and mixed CP and SVC conferences:

Table 19-9 *Resource Capacity Allocation Per Port Type*

Port Type	Non-Mixed Conferences	Mixed CP and SVC Conferences
AVC HD	1	1.5 *
AVC SD	0.5	0.75 *
AVC CIF	0.333	0.75 *
SVC	0.333	0.333

* Resources are consumed at this rate only **after** the conference contains mixed endpoints.

T

MCU Resource Management by RealPresence Resource Manager (XMA), Polycom CMA and Polycom RealPresence DMA System

When the RealPresence Resource Manager (XMA), Polycom CMA and Polycom RealPresence DMA system are part of the solution, following a request by the RealPresence Resource Manager (XMA), Polycom CMA or Polycom RealPresence DMA system, the *MCU* will send updates on resource usage to both *CMA* and *DMA*, with each application updating its own resource usage for the *MCU*. This provides better management of the *Collaboration Server* resources by the RealPresence Resource Manager (XMA), Polycom CMA and Polycom RealPresence DMA system.

Guidelines

- Following requests sent by *CMA* and *RealPresence DMA* system, the *Collaboration Server* will send the number of occupied resources for a conference or total for the MCU.
- Occupied resources are resources that are connected to ongoing conferences. Disconnected endpoints in an ongoing conference are not counted as occupied resources.

- The *Collaboration Server* is unaware of the resource usage split between the *CMA* and *RealPresence DMA* system.

Port Usage Threshold

The *Collaboration Server* can be set to alert the administrator to potential port capacity shortages. A capacity usage threshold can be set as a percentage of the total number of licensed ports in the system.

When the threshold is exceeded, a *System Alert* is generated.

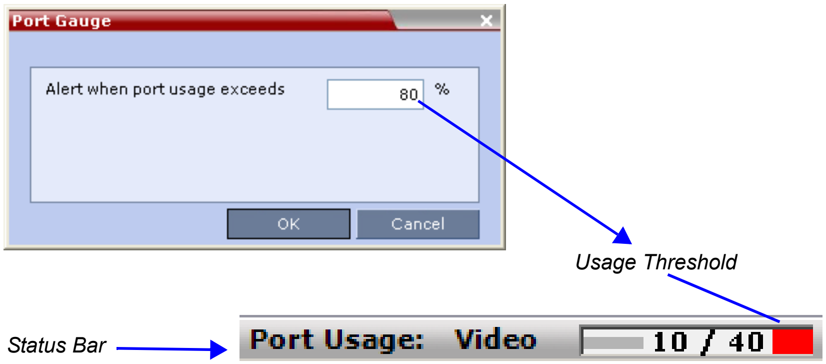
The default port capacity usage threshold is 80%.

The administrator can monitor the MCU's port capacity usage via the *Port Gauge* in the *Status Bar* of the *Collaboration Server Web Client*.

Setting the Port Usage Threshold

To Set the Port Usage Threshold:

1 In the *Setup* menu, click **Port Gauge** to open the *Port Gauge* dialog box.



Alert when port usage exceeds 80 %

Port Usage: Video 10 / 40

Enter the value for the percentage capacity usage threshold.
The high Port Usage threshold represents a percentage of the total number of video available. It is set to indicate when resource usage is approaching its maximum, resulting in no free resources to run additional conferences. When port usage reaches or exceeds the threshold, the red area of the gauge flashes. The default port usage threshold is 80%.

2 Click **OK.**

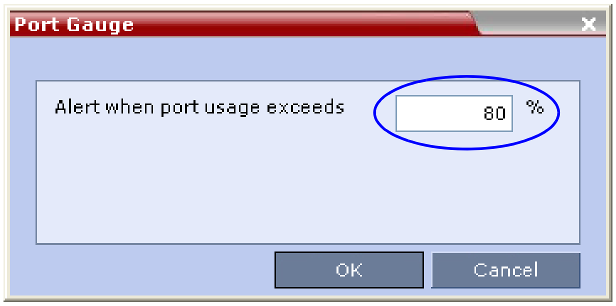
SIP Dial-in Busy Notification

When the system flag SEND_SIP_BUSY_UPON_RESOURCE_THRESHOLD is set to YES (NO is the default), it enables the MCU to send a busy notification to a SIP audio endpoint or a SIP device when dialing in to the MCU whose audio resource usage exceeded the Port Usage threshold.

The *Collaboration Server* will send a SIP busy response to SIP audio endpoints when:

- The system flag SEND_SIP_BUSY_UPON_RESOURCE_THRESHOLD is set to YES (NO is the default)

- The port usage threshold for Audio resources is exceeded. The threshold is defined in the **Setup > Port Gauge** dialog box.



Alert when port usage exceeds 80 %

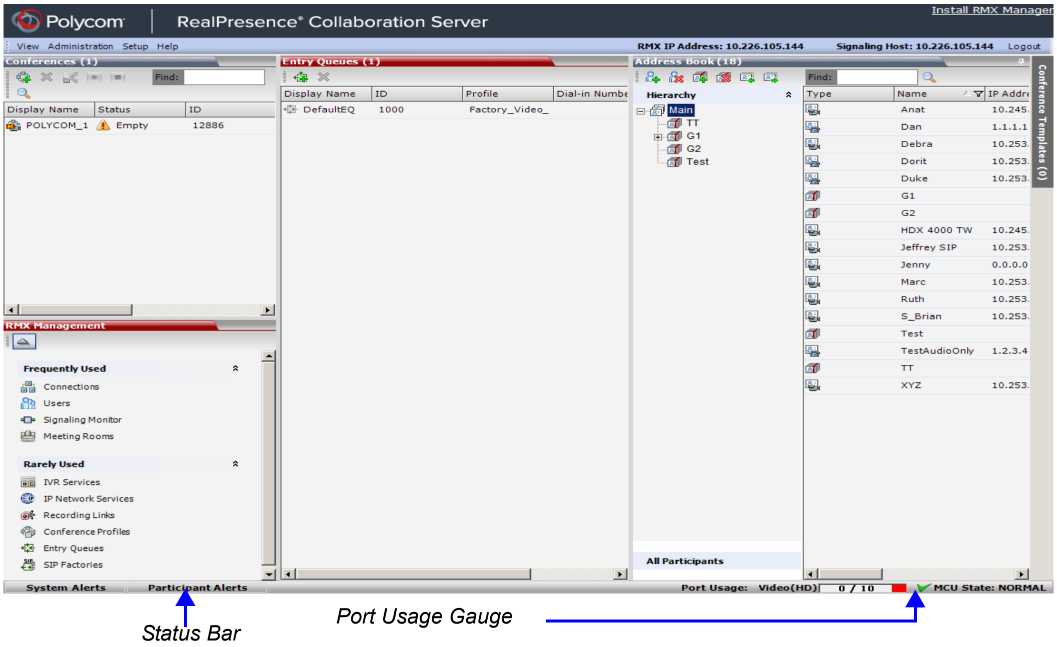
When the flag is set to YES, the system will allow SIP audio endpoints to connect to the MCU until the Port Usage threshold is reached. Once this threshold is exceeded, the SIP audio endpoints will not be able to connect, ensuring that the remaining system resources can be used by all other connections, including SIP video, and H.323 cascaded links. When the call is rejected by the MCU because of lack of resources, the appropriate indication will be sent by the MCU to the SIP audio endpoint.

For example, if the *Port Gauge* threshold is set to 80%, when 80% of the **Audio resources** are used, the system will not allow additional SIP audio endpoints to connect and will send a busy notification to the endpoint.

This does not affect the video resources usage.

Port Usage Gauge

The *Port Usage Gauge* is displayed in the *Status Bar* at the bottom of the *Collaboration Server Web Client* screen.



RMX IP Address: 10.226.105.144 Signaling Host: 10.226.105.144

Display Name	Status	ID
POLYCOM_1	Empty	12886

Display Name	ID	Profile	Dial-in Numbe
DefaultEQ	1000	Factory_Video_	

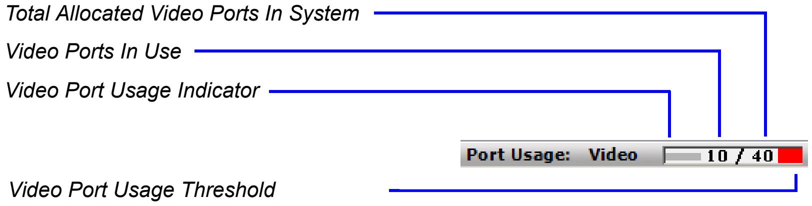
Type	Name	IP Addre
	Anat	10.245.
	Dan	1.1.1.1
	Debra	10.253.
	Dorit	10.253.
	Duke	10.253.
	G1	
	G2	
	HDX 4000 TW	10.245.
	Jeffrey SIP	10.253.
	Jenny	0.0.0.0
	Marc	10.253.
	Ruth	10.253.
	S_Brian	10.253.
	Test	
	TestAudioOnly	1.2.3.4
	TT	
	XYZ	10.253.

Port Usage: Video(HD) 0 / 10 MCU State: NORMAL

The *Port Usage* gauge indicates:

- The total number of *Video* ports in the system

- The number of *Video* ports in use.
- The *High Port Usage* threshold.


Total Allocated Video Ports In System
Video Ports In Use
Video Port Usage Indicator
Port Usage: Video
10 / 40
Video Port Usage Threshold

The basic unit used for reporting resource usage in the Port Gauges is HD720p30. Results are rounded to the nearest integer.

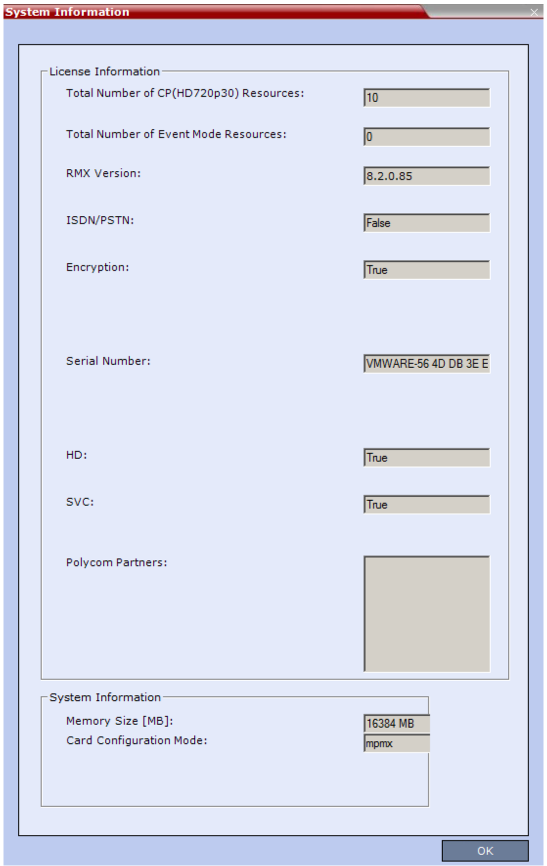
System Information

System Information includes *License Information* and general system information.

To view the System Information properties box:

>> On the menu, click **Administration > System Information**.

The *System Information* properties box is displayed.



License Information

Total Number of CP(HD720p30) Resources: 10

Total Number of Event Mode Resources: 0

RMX Version: 8.2.0.85

ISDN/PSTN: False

Encryption: True

Serial Number: VMWARE-56 4D DB 3E E

HD: True

SVC: True

Polycom Partners:

System Information

Memory Size [MB]: 16384 MB

Card Configuration Mode: mpmx

The *System Information* properties box displays the following information:

Table 19-11 *System Information*

Field	Description
Total Number of CP(HD720p30) Resources	Displays the number of HD720p30 video participants licensed for the system. Each HD720p30 resource represents one 3 CIF video resources. Each SVC resource is equivalent to one CIF video resource.
Total Number of Event Mode Resources	Displays the number of video/voice participants licensed for a system in Event Mode Licensing. It also determines the conference type that is available on the system. 0 - indicates that this Licensing mode is disabled for this system.
RMX™ Version	Displays the *System Software Version* of the RMX™.

Table 19-11 *System Information (Continued)*

Field	Description
Encryption	Indicates whether *Encryption* is included in the MCU license. Encryption is not available in all countries. Range: True / False
Serial Number	Displays the *Serial Number* of the Collaboration Server unit.
HD	Indicates if the MCU is licensed to connect endpoints at *HD* resolutions in *Continuous Presence* conferences.
SVC	license.
Polycom Partners	Indicates that the *System Software* contains features for the support of specific *Polycom Partner* environments.

SNMP (Simple Network Management Protocol)

SNMP enables managing and monitoring of the MCU status by **external** managing systems, such as HP OpenView or through web applications.

The Collaboration Server's implementation of SNMPv3 is FIPS 140 compliant.

MIBs (Management Information Base)

MIBs are a collection of definitions, which define the properties of the managed object within the device to be managed. Every managed device keeps a database of values for each of the definitions written in the MIB.

The SNMP systems poll the MCU according to the MIB definitions.

Traps

The MCU is able to send Traps to different managers. Traps are messages that are sent by the MCU to the SNMP Manager when an event such as MCU Reset occurs.

Guidelines

- *Version 1, Version 2* and *Version 3* traps are supported.
- When *SNMPv3* is selected only *SNMPv3 Queries* and *Traps* receive responses.
- A mixture of *Version 1, Version 2* and *Version 3* traps is not permitted.

MIB Files

The H.341 standard defines the MIBs that H.320 and H.323 MCUs must comply with. In addition, other MIBs should also be supported, such as MIB-II and the ENTITY MIB, which are common to all network entities.

The MIBs are contained in files in the *SNMP MIBS* sub-directory of the Collaboration Server root directory. The files should be loaded to the SNMP external system and compiled within that application. Only then can the SNMP external application perform the required monitoring tasks.

> The MULTI-MEDIA_MIB_TC must be compiled before compiling the other MIBs.

Private MIBs

- *RMX-MIB (RMX-MIB.MIB)*
  - — Contains the statuses of the Collaboration Server: Startup, Normal and Major.
  - — Contains all the Alarms of the Collaboration Server that are sent to the SNMP Manager.

Support for MIB-II Sections

The following table details the MIB-II sections that are supported:

***Table 19-12** Supported MIB-II Sections*

Section	Object Identifier
system	mib-2 1
interfaces	mib-2 2
ip	mib-2 4

The Alarm-MIB

MIB used to send alarms. When a trap is sent, the Alarm-MIB is used to send it.

H.341-MIB (H.341 – H.323)

- Gives the address of the gatekeeper.
- Supports H.341-MIB of SNMP events of H.323.

Standard MIBs

This section describes the MIBs that are included with the Collaboration Server. These MIBs define the various parameters that can be monitored, and their acceptable values.

***Table 19-13** Standard MIBs*

MIB Name	Description
MULTI-MEDIA-MIB-TC (MULTIMTC.MIB)	Defines a set of textual conventions used within the set of Multi Media MIB modules.
H.320ENTITY-MIB (H320-ENT.MIB)	This is a collection of common objects, which can be used in an H.320 terminal, an H.320 MCU and an H.320/H.323 gateway. These objects are arranged in three groups: Capability, Call Status, and H.221 Statistics.
H.320MCU-MIB (H320-MCU.MIB)	Used to identify managed objects for an H.320 MCU. It consists of four groups: System, Conference, Terminal, and Controls. The *Conference* group consists of the active conferences. The *Terminal* group is used to describe terminals in active MCU conferences. The *Controls* group enables remote management of the MCU.
H323MC-MIB (H323-MC.MIB)	Used to identify objects defined for an H.323 Multipoint Controller. It consists of six groups: System, Configuration, Conference, Statistics, Controls and Notifications. The *Conference* group is used to identify the active conferences in the MCU. The *Notifications* group allows an MCU, if enabled, to inform a remote management client of its operational status. **Note:** The Collaboration Server supports only one field in H.341-H323MC MIB. The Collaboration Server reports the Gatekeeper address using H.341-H323MC MIB – 323McConfigGatekeeperAddress (0.0.8.341.1.1.4.2.1.1.4) in response to a query from a manager.

Table 19-13 Standard MIBs

MIB Name	Description
MP-MIB (H323-MP.MIB)	Used to identify objects defined for an H.323 Multipoint Processor, and consists of two groups: Configuration and Conference. The *Configuration* group is used to identify audio/video mix configuration counts. The *Conference* group describes the audio and video multi-processing operation.
MIB-II/RFC1213-MIB (RFC1213.MIB)	Holds basic network information and statistics about the following protocols: TCP, UDP, IP, ICMP and SNMP. In addition, it holds a table of interfaces that the Agent has. MIB-II also contains basic identification information for the system, such as, Product Name, Description, Location and Contact Person.
ENTITY-MIB (ENTITY.MIB)	Describes the unit physically: Number of slots, type of board in each slot, and number of ports in each slot.

Unified MIB

The Collaboration Server uses the Polycom Unified MIB, in addition to the RMX specific MIB. The Polycom Unified MIB is an MIB that is used by many Polycom products. The following table describes the information provided by the Collaboration Server in the Unified MIB.

Table 19-14 Unified MIB SNMP Fields

Name	Type	Description
Debug	Boolean	Indicates whether the unit is in a debugging state.
IncomingCallsReqrGK	Boolean	Indicates whether a gatekeeper is required to receive incoming H.323 calls.
OutgoingCallsReqrGK	Boolean	Indicates whether a gatekeeper is required to make outgoing H.323 calls.
HDBitrateThrshld	Integer	The minimum bit rate required by endpoints in order to connect to an HD conference.
MaxCPRstln	Integer	Maximum resolution of a CP conference.
MaxCPRstlnCfg	Integer	Configured resolution for a CP conference.
EndpointDispayName	String	The name of the MCU that is displayed on the screen of endpoints that are connecting to the conference.
PALNTSC	NTSC/PAL/ AUTO	The video encoding of the RMX.
SeparateMgmtNet	Boolean	Indicates whether management network separation is enabled.
NumPorts	Integer	Total number of ports.
NumVideoPorts	Integer	Number of ports configured for video.

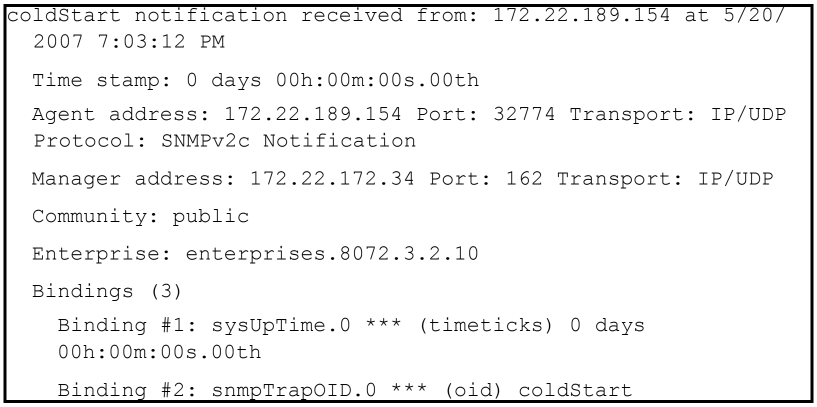
Table 19-14 *Unified MIB SNMP Fields (Continued)*

Name	Type	Description
ServiceH323	Integer	Indicates the status of H.323 capabilities: 1 - The service is enabled and operational. 2 - The service is enabled but is not operational. 3 - The service is disabled.
ServiceSIP	Integer	Indicates the status of SIP capabilities: 1 - The service is enabled and operational. 2 - The service is enabled but is not operational. 3 - The service is disabled.
ServiceISDN	Integer	Indicates the status of SIP capabilities: 1 - The service is enabled and operational. 2 - The service is enabled but is not operational. 3 - The service is disabled.
RsrcAllocMode	Fixed/ Flexible	The resource allocation method which determines how the system resources are allocated to the connecting endpoints.
McuSystemStatus	Integer	System State.
FanStatus	Boolean	Status of the hardware fan.
PowerSupplyStatus	Boolean	Status of the power supply.
IntegratedBoardsStatus	Boolean	Status of the integrated boards.
UltraSecureMode	Boolean	Indicates whether the RMX is operating in Ultra Secure Mode.
ChassisTemp	Integer	The temperature of the chasis.
NumPortsUsed	Integer	Number of ports currently in use.
NewCallsPerMinute	Integer	New calls in the last minute.
ScsfNewCallsPerMinute	Integer	Successful new calls in the last minute.
FldNewCallsPerMinute	Integer	Failed new calls in the last minute.
PctScsflNewCalls	Integer	Percentage of new calls in the last minute which were successful.
CallsEndedScsflPerMin	Integer	Number of calls in the last minute which ended with a success code.
CallsEndedFailedPerMin	Integer	Number of calls in the last minute which ended with a failure code.
CallsEndedScsfl	Integer	Number of calls in the last minute which ended with a success code.
CallsEndedFailed	Integer	Number of calls in the last minute which ended with a failure code.
NumActvCnfrncs	Integer	Number of active conferences.

Traps

Three types of traps are sent as follows:

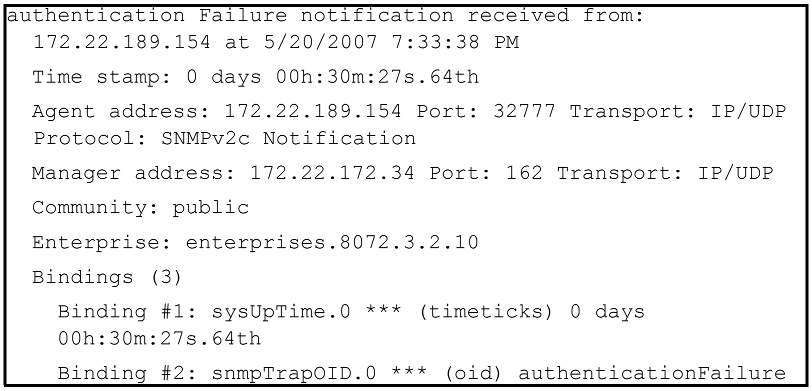
1 ColdStart trap. This is a standard trap which is sent when the MCU is reset.



```
coldStart notification received from: 172.22.189.154 at 5/20/
  2007 7:03:12 PM

  Time stamp: 0 days 00h:00m:00s.00th

  Agent address: 172.22.189.154 Port: 32774 Transport: IP/UDP
  Protocol: SNMPv2c Notification

  Manager address: 172.22.172.34 Port: 162 Transport: IP/UDP

  Community: public

  Enterprise: enterprises.8072.3.2.10

  Bindings (3)

    Binding #1: sysUpTime.0 *** (timeticks) 0 days
    00h:00m:00s.00th

    Binding #2: snmpTrapOID.0 *** (oid) coldStart
```

Figure 1 An Example of a ColdStart Trap

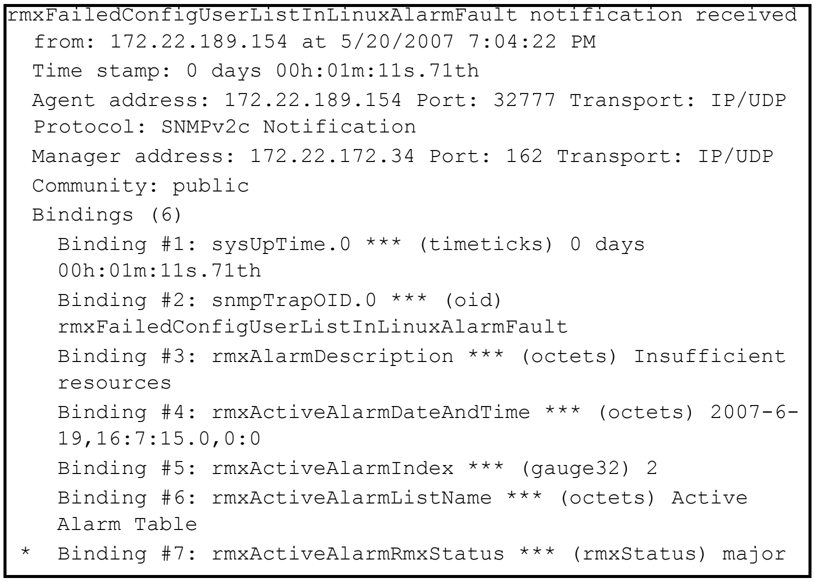
2 Authentication failure trap. This is a standard trap which is sent when an unauthorized community tries to enter.



```
authentication Failure notification received from:
  172.22.189.154 at 5/20/2007 7:33:38 PM

  Time stamp: 0 days 00h:30m:27s.64th

  Agent address: 172.22.189.154 Port: 32777 Transport: IP/UDP
  Protocol: SNMPv2c Notification

  Manager address: 172.22.172.34 Port: 162 Transport: IP/UDP

  Community: public

  Enterprise: enterprises.8072.3.2.10

  Bindings (3)

    Binding #1: sysUpTime.0 *** (timeticks) 0 days
    00h:30m:27s.64th

    Binding #2: snmpTrapOID.0 *** (oid) authenticationFailure
```

Figure 2 An Example of an Authentication Failure Trap

3 Alarm Fault trap. The third trap type is a family of traps defined in the POLYCOM-RMX-MIB file, these traps are associated with the Collaboration Server active alarm and clearance (proprietary SNMP trap).



```
rmxFailedConfigUserListInLinuxAlarmFault notification received
  from: 172.22.189.154 at 5/20/2007 7:04:22 PM
  Time stamp: 0 days 00h:01m:11s.71th
  Agent address: 172.22.189.154 Port: 32777 Transport: IP/UDP
  Protocol: SNMPv2c Notification
  Manager address: 172.22.172.34 Port: 162 Transport: IP/UDP
  Community: public
  Bindings (6)
    Binding #1: sysUpTime.0 *** (timeticks) 0 days
    00h:01m:11s.71th
    Binding #2: snmpTrapOID.0 *** (oid)
    rmxFailedConfigUserListInLinuxAlarmFault
    Binding #3: rmxAlarmDescription *** (octets) Insufficient
    resources
    Binding #4: rmxActiveAlarmDateAndTime *** (octets) 2007-6-
    19,16:7:15.0,0:0
    Binding #5: rmxActiveAlarmIndex *** (gauge32) 2
    Binding #6: rmxActiveAlarmListName *** (octets) Active
    Alarm Table
 *  Binding #7: rmxActiveAlarmRmxStatus *** (rmxStatus) major
```

Figure 3 An Example of an Alarm Fault Trap

Each trap is sent with a time stamp, the agent address, and the manager address.

Status Trap

The MCU sends status traps for the status **MAJOR** - a trap is sent when the card/MCU status is MAJOR.

All traps are considered "MAJOR".

RMX MIB entities that do not generate traps.

The following table lists the entities that appear in the RMX MIB of the SNMP that do not generate traps. These traps will be displayed as Faults in the System Alerts pane (at the bottom of the Collaboration Server (RMX) Web Client screen.

Table 19-15 SNMP MIB entities that do not generate traps

Key	Description	Comment
5002	Resource process did not receive the Meeting Room list during startup.	
5004	Task terminated	
5008	Low Processing Memory	
5009	Low system Memory	
5010	High system CPU usage	
5014	High CPU utilization	

***Table 19-15** SNMP MIB entities that do not generate traps (Continued)*

Key	Description	Comment
5016	Process idle	
5107	Failed to open Apache server configuration file	
5108	Failed to save Apache server configuration file	
5110	A private version is loaded	
5111	NTP synchronization failure	
5112	Invalid date and time	
5116	Incorrect Ethernet Settings	
5117	Smart Report found errors on hard disk	
5118	Invalid MCU Version	
5150	Music file error	
5205	Unspecified problem	
5207	Unit not responding	
5209	Failed to mount Card folder	
5401	The Log file system is disabled	
5450	Action redirection failure	
5601	Process terminated	
5602	Terminal initiated MCU reset	
5603	User initiated MCU reset	
5604	Internal MCU reset	
5605	MCU reset	
5606	MCU Reset to enable Diagnostics mode	
5607	Startup process failure	
5801	Polycom default User exists. For security reasons, it is recommended to delete this User and create your own User.	Only in non-Ultra Secure Mode
5904	Single clock source	
5950	MCU is not configured for AVF gatekeeper mode	
5652	Hard disk error /AA_HARD_DISK_FAILURE	Not in use
5551	Port configuration modified	Not in use
5011	Used for testing the Active Alarms mechanism	Not in use

Table 19-15 *SNMP MIB entities that do not generate traps (Continued)*

Key	Description	Comment
5001	License not found	Not in use (Product activation failure is trapped)

Defining the SNMP Parameters in the Collaboration Server

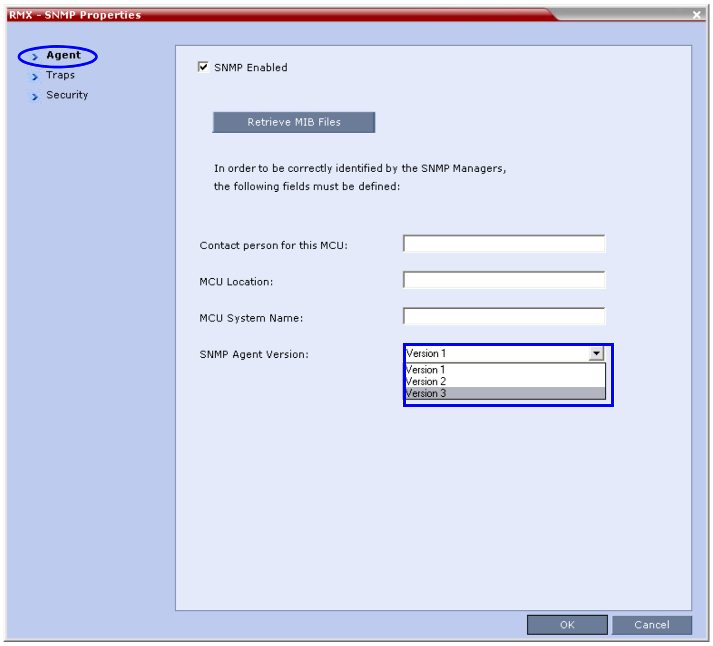
The SNMP option is enabled via the *Collaboration Server Web Client* application.

The addresses of the Managers monitoring the MCU and other security information are defined in the *Collaboration Server Web Client* application and are saved on the MCU's hard disk. Only users defined as Administrator can define or modify the SNMP security parameters in the *Collaboration Server Web Client* application.

To enable SNMP option:

1 In the *Collaboration Server Web Client* menu bar, click **Setup > SNMP**.

The *Collaboration Server-SNMP Properties - Agent* dialog box is displayed.



☑ SNMP Enabled

In order to be correctly identified by the SNMP Managers, the following fields must be defined:

Contact person for this MCU:

MCU Location:

MCU System Name:

SNMP Agent Version: Version 1 (Version 1, Version 2, Version 3)

This dialog box is used to define the basic information for this MCU that will be used by the SNMP system to identify it.

2 In the *Agent* dialog box, click the **SNMP Enabled** check box.

3 Click the **Retrieve MIB Files** button to obtain a file that lists the MIBs that define the properties of the object being managed.

The *Retrieve MIB Files* dialog box is displayed.

4 Click the **Browse** button and navigate to the desired directory to save the MIB files.

5 Click **OK**.
The path of the selected directory is displayed in the *Retrieve MIB Files* dialog box.

6 Click the **Save** button.
The MIB files are saved to the selected directory.

7 Click **Close** to exit the *Retrieve MIB Files* dialog box.

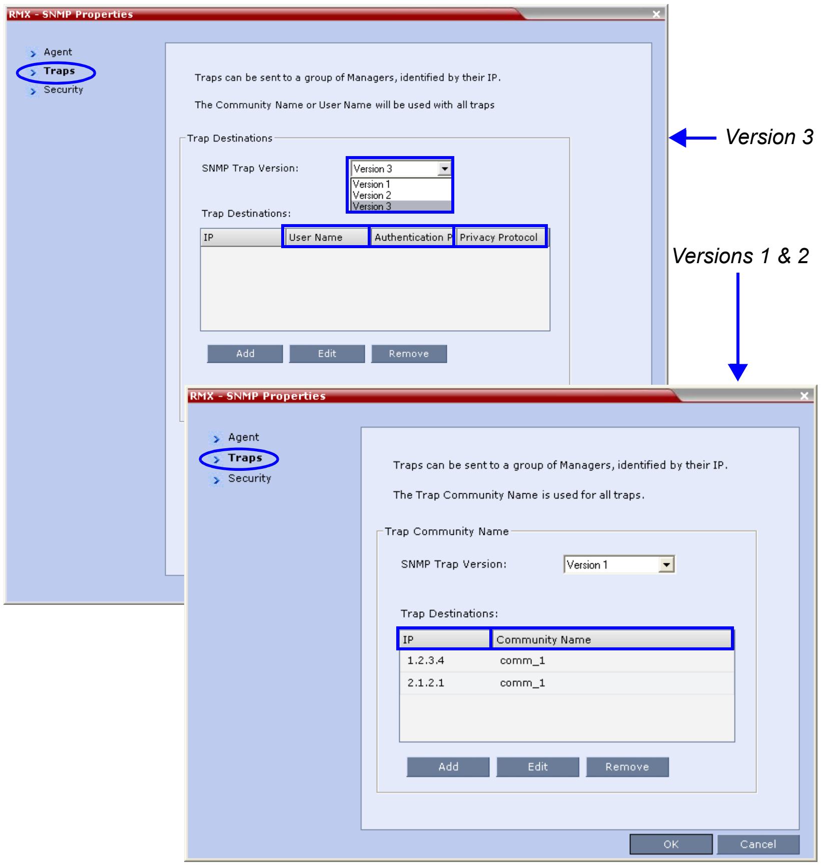
8 In the *Agent* dialog box, define the parameters that allow the SNMP Management System and its user to easily identify the MCU.

Table 19-16 *Collaboration Server-SNMP Properties - Agent Options*

Field	Description
Contact person for this MCU	Type the name of the person to be contacted in the event of problems with the MCU.
MCU Location	Type the location of the MCU (address or any description).
MCU System Name	Type the MCU's system name.

9 Click the **Traps** tab.

The *Collaboration Server-SNMP Properties – Traps* dialog box opens.



Traps can be sent to a group of Managers, identified by their IP.

The Community Name or User Name will be used with all traps

Trap Destinations

SNMP Trap Version: Version 3 (Version 1, Version 2, Version 3)

Trap Destinations:

IP	User Name	Authentication P	Privacy Protocol

Traps can be sent to a group of Managers, identified by their IP.

The Trap Community Name is used for all traps.

Trap Community Name

SNMP Trap Version: Version 1

Trap Destinations:

IP	Community Name
1.2.3.4	comm_1
2.1.2.1	comm_1

Traps are messages sent by the MCU to the SNMP Managers when events such as MCU Startup or Shutdown occur. Traps may be sent to several SNMP Managers whose IP addresses are specified in the *Trap Destinations* box.

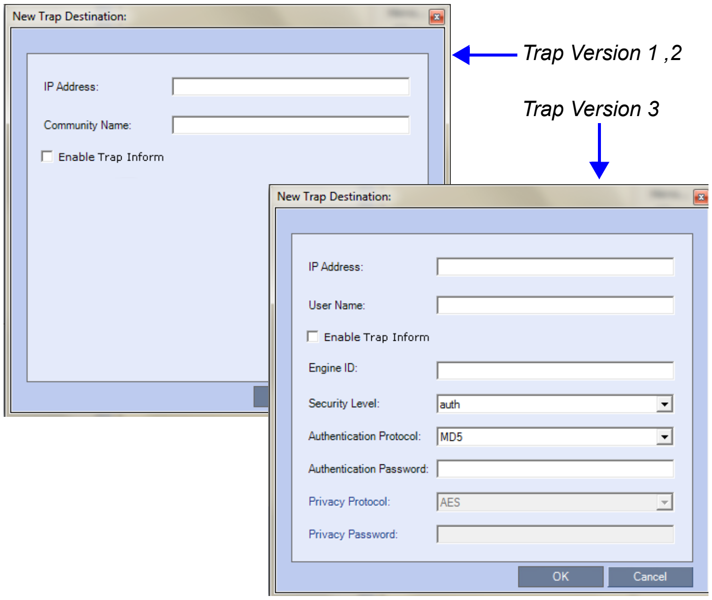
10 Define the following parameters:

Table 19-17 SNMPv3 - Traps

Field	Description		
SNMP Trap Version	Specifies the version, either Version 1 2 or 3 of the traps being sent to the IP Host. Polycom software supports the standard SNMP version 1 and 2 traps, which are taken from RFC 1215, convention for defining traps for use with SNMP. **Note:** The SNMP Trap Version parameters must be defined identically in the external SNMP application.		
Trap Destination	This box lists the currently defined IP addresses of the Manager terminals to which the message (trap) is sent.		
	IP	Enter the IP address of the SNMP trap recipient.	All Versions
	Community Name	Enter the Community Name of the manager terminal used to monitor the MCU activity	Version 1 and Version 2
	User Name	Enter the name of the user who is to have access to the trap.	Version 3
	Authentication Protocol	Enter the authentication protocol: MD5 or SHA.	
	Privacy Protocol	Enter the privacy protocol: DES or AES.	
	Engine ID	Enter an Engine ID to be used for both the Agent and the Trap. Default: Empty	

11 Click the **Add** button to add a new Manager terminal.

Depending on the *SNMP Trap Version* selected, one of the two following *New Trap Destination* dialog boxes opens.



New Trap Destination:

IP Address:

Community Name:

□ Enable Trap Inform

New Trap Destination:

IP Address:

User Name:

□ Enable Trap Inform

Engine ID:

Security Level: auth

Authentication Protocol: MD5

Authentication Password:

Privacy Protocol: AES

Privacy Password:

12 Define the following parameters:

Table 19-18 *SNMPv3 - Traps*

Field	Description	Version
IP Address	Enter the IP address of the SNMP trap recipient.	1,2,3
Enable Trap Inform	An Inform is a *Trap* that requires receipt confirmation from the entity receiving the *Trap*. If the *Engine ID* field (*Version 3*) is empty when *Enable Trap Inform* has been selected, the *Engine ID* is set by the *Client*.	
Community Name	Enter the Community Name of the manager terminal used to monitor the MCU activity	1, 2

***Table 19-18** SNMPv3 - Traps (Continued)*

Field	Description	Version
User Name	Enter the name of the user who is to have access to the trap.	3
Engine ID	Enter an *Engine ID* to be used for the *Trap*. This field is enabled when the *Enable Trap Inform* check box is selected. If the *Enable Trap Inform* check box is cleared the *Engine ID* of the *Agent* is used. The *Engine ID* is comprised of up to 64 Hexadecimal characters. Default: Empty	
Security Level	Select a *Security Level* from the drop-down menu. **Range:** *No Auth, No Priv; Auth, No Priv; Auth, Priv* **Default:** Auth, Priv	
Authentication Protocol	Enter the authentication protocol: MD5 or SHA. The availability of the MD5 Authentication Protocol as a selectable option is controlled by adding the SNMP_FIPS_MODE System Flag to system.cfg and setting its value. A value of YES means that MD5 will neither be displayed as selectable option nor supported. Range: YES/NO. Default: NO.	
Authentication Password		
Privacy Protocol	Enter the privacy protocol: DES or AES. The availability of the DES Privacy Protocol as a selectable option is controlled by adding the SNMP_FIPS_MODE System Flag to system.cfg and setting its value. A value of YES means that DES will neither be displayed as a selectable option nor supported. Range: YES/NO. Default: NO.	
Privacy Password		

13 Type the **IP Address** and the **Community name** of the manager terminal used to monitor the MCU activity, and then click **OK**.

The *Community name* is a string of characters that will be added to the message that is sent to the external Manager terminals. This string is used to identify the message source by the external Manager terminal.

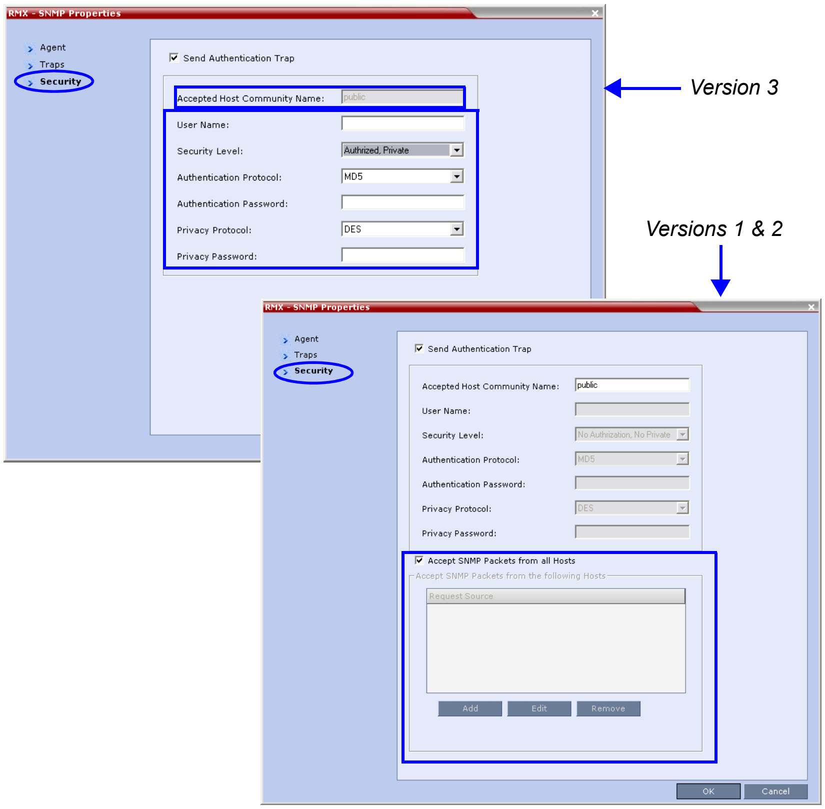
The new *IP Address* and *Community name* is added to the *Trap Destinations* box.

a To delete the IP Address of a Manager terminal, select the address that you wish to delete, and then click the **Remove** button.

The IP address in the *Trap Destinations* box is removed.

14 Click the **Security** tab.

The *Collaboration Server-SNMP Properties – Security* dialog box opens.



Version 3:

☑ Send Authentication Trap

Accepted Host Community Name: public

User Name:

Security Level: Authrized, Private

Authentication Protocol: MD5

Authentication Password:

Privacy Protocol: DES

Privacy Password:

Versions 1 & 2:

☑ Send Authentication Trap

Accepted Host Community Name: public

User Name:

Security Level: No Authrization, No Private

Authentication Protocol: MD5

Authentication Password:

Privacy Protocol: DES

Privacy Password:

☑ Accept SNMP Packets from all Hosts

Accept SNMP Packets from the following Hosts

Request Source

Add Edit Remove

This dialog box is used to define whether the query sent to the MCU is sent from an authorized source. When the *"Accept SNMP packets from all Hosts"* is disabled, a valid query must contain the appropriate community string and must be sent from one of the Manager terminals whose IP address is listed in this dialog box.

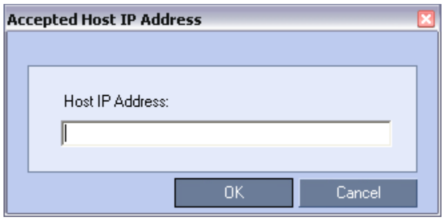
15 Define the following parameters:

Table 19-19 *SNMP - Security*

Field	Description		
Send Authentication Trap	Select this check box to send a message to the SNMP Manager when an unauthorized query is sent to the MCU. When cleared, no indication will be sent to the SNMP Manager.		Versions 1 & 2
Accept Host Community Name	Enter the string added to queries that are sent from the SNMP Manager to indicate that they were sent from an authorized source. **Note:** Queries sent with different strings will be regarded as a violation of security, and, if the Send Authentication Trap check box is selected, an appropriate message will be sent to the SNMP Manager.		
Accept SNMP Packets from all Host	Select this option if a query sent from any Manager terminal is valid. When selected, the Accept SNMP Packets from These Hosts option is disabled.		
Accept SNMP Packets from the following Hosts	Lists specific Manager terminals whose queries will be considered as valid. This option is enabled when the Accept SNMP Packets from any Host option is cleared.		
User Name	Enter a *User Name* of up to 48 characters **Default:** Empty		Version 3
Security Level	Select a *Security Level* from the drop-down menu. **Range:** *No Auth, No Priv; Auth, No Priv; Auth, Priv* **Default:** Auth, Priv		
Authentication Protocol	Select the authentication protocol **Range:** *MD5, SHA* **Default:** *MD5*	These fields are enabled if *Authentication* is selected in the *Security Level* field.	
Authentication Password	Enter an *Authentication Password.* **Range:** 8 - 48 characters **Default:** Empty		
Privacy Protocol	Select a *Privacy Protocol*. **Range:** DES, AES **Default:** DES	These fields are enabled if *Privacy* is selected in the *Security Level* field.	
Privacy Password	Enter a *Privacy Password*. **Range:** 8 - 48 characters **Default:** Empty		
Engine ID	Enter an *Engine ID* to be used for both the *Agent* and the *Trap*. **Default:** Empty		

16 To specifically define one or more valid terminals, ensure that the *Accept SNMP Packets from any Host* option is cleared and then click the **Add** button.

The *Accepted Host IP Address* dialog box opens.



17 Enter the *IP Address* of the Manager terminal from which valid queries may be sent to the MCU, and then click **OK**.
Click the **Add** button to define additional *IP Addresses*.
The *IP Address* or *Addresses* are displayed in the *Accept SNMP Packets from These Hosts* box.

Queries sent from terminals not listed in the *Accept SNMP Packets from These Hosts* box are regarded as a violation of the MCU security, and if the *Send Authentication Trap* check box is selected, an appropriate message will be sent to all the terminals listed in the *SNMP Properties – Traps* dialog box.

18 In the *Collaboration Server - SNMP Properties - Security* dialog box, click **OK**.

Audible Alarms

In addition to the visual cues used to detect events occurring on the Collaboration Server, an audible alarm can be activated and played when participants request Operator Assistance.

Using Audible Alarms

The Audible Alarm functionality for Operator Assistance requests is enabled for each MCU in either the *Collaboration Server Web Client* or *RMX Manager*.

The Audible Alarm played when Operator Assistance is requested is enabled and selected in the **Setup > Audible Alarm > User Customization**. When the Audible Alarm is activated, the *.wav file selected in the *User Customization* is played, and it is repeated according to the number of repetitions defined in the *User Customization*.

If more than one Collaboration Server is monitored in the *RMX Manager*, the Audible Alarm must be enabled separately for each Collaboration Server installed in the site/configuration. A different *.wav file can be selected for each MCU.

When multiple Audible Alarms are activated in different conferences or by multiple MCUs, the Audible Alarms are synchronized and played one after the other. It is important to note that when *Stop Repeating Alarm* is selected from the toolbar from the *Collaboration Server Web Client* or *RMX Manager*, all activated Audible Alarms are immediately halted.

Audible Alarm Permissions

An operator/administrator can configure the Request Operator Assistance audible alarm, however Users with different authorization level have different configuration capabilities as shown in Table 19-20.

Table 19-20 Audible Alarm Permissions

Option	Operator	Administrator
User Customization	✓	✓
Download Audible Alarm File		✓
Stop Repeating Alarms	✓	✓

Stop Repeating Message

The Collaboration Server User can stop playing the audible alarm at any time. If more than one audible alarm has been activated, all activated alarms are immediately stopped.

If after stopping the Audible Alarms a new Operator Assistance request event occurs, the audible alarm is re-activated.

To stop the Audible Alarm on the Collaboration Server Client or RMX Manager:

>> On the Collaboration Server menu, click **Setup** > **Audible Alarms** >**Stop Repeating Alarm**.

When selected all audible alarms are immediately stopped.

Configuring the Audible Alarms

User Customization

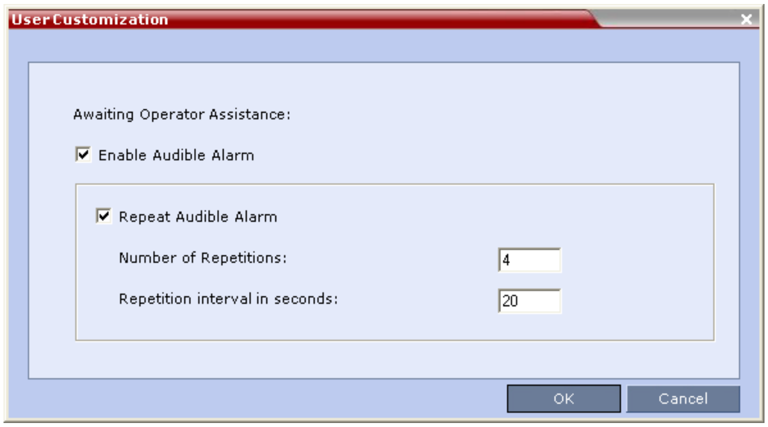
The operators and administrators can:

- Enable/Disable the Audible Alarm.
- Select whether to repeat the Audible Alarm.
- Define the number of repetitions and the interval between the repetitions.

To Customize the Audio Alert on the Collaboration Server Client or RMX Manager:

1 On the Collaboration Server menu, click **Setup** > **Audible Alarms** > **User Customization**.

The *User Customization* window opens.



User Customization

Awaiting Operator Assistance:

☑ Enable Audible Alarm

☑ Repeat Audible Alarm

Number of Repetitions: 4

Repetition interval in seconds: 20

2. Define the following parameters:

Table 19-21 Audible Alarm - User Customization Options

Option	Description
Enable Audible Alarm	Select this check box to enable the Audible Alarm feature and to define its properties. When this check box is cleared, the Audible Alarm functionality is disabled.
Repeat Audible Alarm	Select this check box to play the Audible Alarm repeatedly. When selected, it enables the definition of the number of repetitions and the interval between repetitions. When cleared, the Audible Alarm will not be repeated and will be played only once.
Number of Repetitions	Define the number of times the audible alarm will be played. Default number of repetitions is 4.
Repetition interval in seconds	Define the number of seconds that the system will wait before playing the Audible Alarm again. Default interval is 20 seconds.

3. Click **OK**.

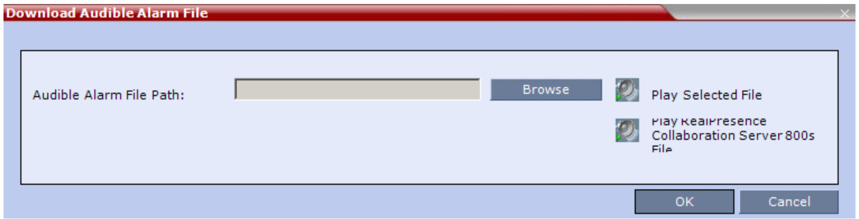
Replacing the Audible Alarm File

Each Collaboration Server is shipped with a default tone file in *.wav format that plays a specific tone when participants request Operator Assistance. This file can be replaced by a *.wav file with your own recording. The file must be in *.wav format and its length cannot exceed one hour.

Only the User with Administrator permission can download the Audible Alarm file.

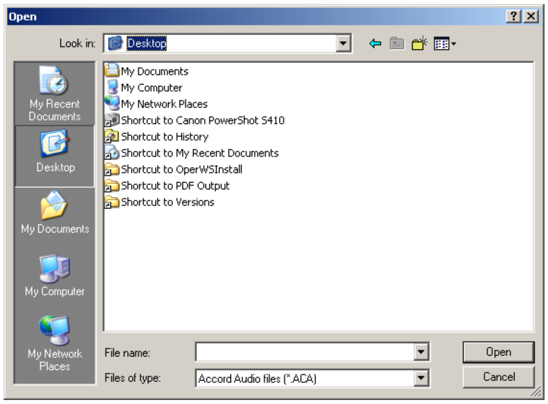
To replace the Audio file on the Collaboration Server Client or RMX Manager:

1 On the Collaboration Server menu, click **Setup** > **Audible Alarms** > **Download Audible Alarm File**.
The *Download Audible Alarm File* window opens.


Download Audible Alarm File
Audible Alarm File Path:
Browse
Play Selected File
Play RealPresence Collaboration Server 800s File
OK
Cancel

2 Click the **Browse** button to select the audio file (*.wav) to download.

The *Open* dialog box opens.



Look in: Desktop

Files of type: Accord Audio files (*.ACA)

3 Select the appropriate *.wav file and then click the **Open** button.
The selected file name is displayed in the *Install Audible Alarm File* dialog box.

4 **Optional**. You can play the selected file or the currently used file by clicking the *Play* ( ) button as follows:

a Click **Play Selected File** to play a file saved on your computer.

b Click **Play Collaboration Server File** to play the file currently saved on the Collaboration Server.

5 In the *Download Audible Alarm File* dialog box, click **OK** to download the file to the MCU.

The new file replaces the file stored on the MCU. If multiple Collaboration Servers are configured in the *RMX Manager*, the file must be downloaded to each of the required MCUs separately.

Multilingual Setting

Each supported language is represented by a country flag in the *Welcome Screen* and can be selected as the language for the *Collaboration Server Web Client.*

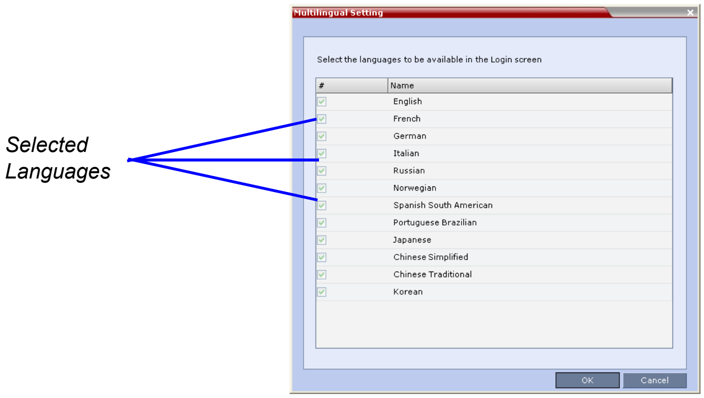
Customizing the Multilingual Setting

The languages available for selection in the *Login* screen of the *Collaboration Server Web Client* can be modified using the *Multilingual Setting* option.

To customize the Multilingual Setting:

1 On the Collaboration Server menu, click **Setup > Customize Display Settings > Multilingual Setting**.

The *Multilingual Setting* dialog box is displayed.



Select the languages to be available in the Login screen

#	Name
☑	English
☑	French
☑	German
☑	Italian
☑	Russian
☑	Norwegian
☑	Spanish South American
☑	Portuguese Brazilian
☑	Japanese
☑	Chinese Simplified
☑	Chinese Traditional
☑	Korean

2 Click the check boxes of the languages to be available for selection.

3 Click **OK**.

4 **Log out** from the *Collaboration Server Web Client* and **Log in** for the customization to take effect**.**

Banner Display and Customization

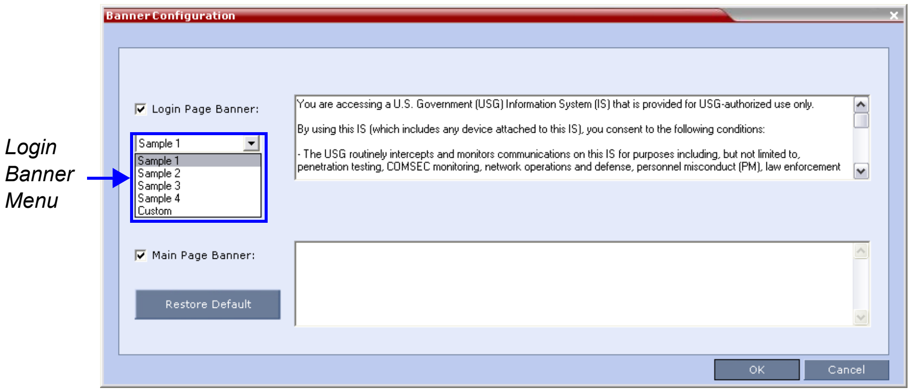
The *Login Screen* and *Main Screen* of the *Collaboration Server Web Client* and the *Collaboration Server Manager can* display informative or warning text banners. These banners can include general information or they can be cautioning users to the terms and conditions under which they may log into and access the system, as required in many secured environments.

Banner display is enabled in the *Setup* > *Customize Display Settings* > *Banners Configuration*.

The administrator can choose one of four alternative login banners to be displayed. The four alternative banners cannot be modified. A *Custom* banner (default) can also be defined.

The *Main Page Banner* is blank and can be defined.

The *Banner Configuration* dialog box allows the administrator to select a *Login Banner* from a drop-down menu.



You are accessing a U.S. Government (USG) Information System (IS) that is provided for USG-authorized use only.

By using this IS (which includes any device attached to this IS), you consent to the following conditions:

- The USG routinely intercepts and monitors communications on this IS for purposes including, but not limited to, penetration testing, COMSEC monitoring, network operations and defense, personnel misconduct (PM), law enforcement

One of the the following *Login Banners* can be selected:

- **Non-Modifiable Banners**
  — *Sample 1*
  — *Sample 2*
  — *Sample 3*
  — *Sample 4*
- **Modifiable Banner**
  — *Custom* (Default)

Guidelines

- The *Login Banner* must be acknowledged before the user is permitted to log in to the system.
- If a *Custom* banner has been created, and the user selects one of the alternative, non-modifiable banners the *Custom* banner not deleted.
- The *Custom Login Banner* banner may contain up to 1300 characters.
- An empty *Login Banner* is not allowed.

- Any attempt to modify a non-modifiable banner results in it automatically being copied to the *Custom* banner.

Non-Modifiable Banner Text

Sample 1 Banner

You are accessing a U.S. Government (USG) Information System (IS) that is provided for USG-authorized use only.

By using this IS (which includes any device attached to this IS), you consent to the following conditions:

- The USG routinely intercepts and monitors communications on this IS for purposes including, but not limited to, penetration testing, COMSEC monitoring, network operations and defense, personnel misconduct (PM), law enforcement (LE), and counterintelligence (CI) investigations.
- At any time, the USG may inspect and seize data stored on this IS.
- Communications using, or data stored on, this IS are not private, are subject to routine monitoring, interception, and search, and may be disclosed or used for any USG authorized purpose.
- This IS includes security measures (e.g., authentication and access controls) to protect USG interests--not for your personal benefit or privacy.
- Notwithstanding the above, using this IS does not constitute consent to PM, LE or CI investigative searching or monitoring of the content of privileged communications, or work product, related to personal representation or services by attorneys, psychotherapists, or clergy, and their assistants. Such communications and work product are private and confidential. See User Agreement for details.

Sample 2 Banner

This system is for the use of authorized users only. Individuals using this computer system without authority, or in excess of their authority, are subject to having all of their activities on this system monitored and recorded by systems personnel. In the course of monitoring individuals improperly using this system, or in the course of system maintenance, the activities of authorized users also may be monitored. Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible criminal activity, system personnel may provide the evidence of such monitoring to law enforcement officials.

Sample 3 Banner

You are about to access a system that is intended for authorized users only. You should have no expectation of privacy in your use of this system. Use of this system constitutes consent to monitoring, retrieval, and disclosure of any information stored within the system for any purpose including criminal prosecution.

Sample 4 Banner

This computer system including all related equipment, network devices (specifically including Internet access), is provided only for authorized use. All computer systems may be monitored for all lawful purposes, including ensuring that their use is authorized, for management of the system, to facilitate protection against unauthorized access, and to verify security procedures, survivability and operational security. Monitoring

```
includes active attacks by authorized personnel and their entities to test
or verify the security of the system. During monitoring, information may be
examined, recorded, copied and used for authorized purposes. All information
including personal information, placed on or sent over this system may be
monitored. Use of this system, authorized or unauthorized, constitutes
consent to monitoring of this system. Unauthorized use may subject you to
criminal prosecution. Evidence of any such unauthorized use collected during
monitoring may be used for administrative, criminal or other adverse action.
Use of this system constitutes consent to monitoring for these purposes.
```

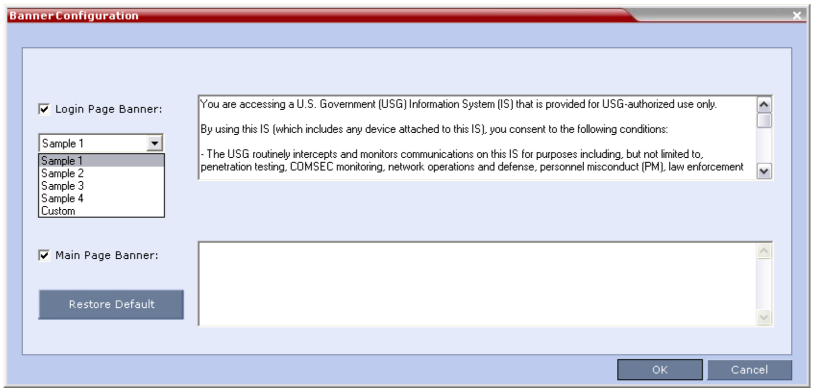
Customizing Banners

The *Login* and *Main Screen* banners can be customized to display conference information, assistance information or warning text as required in the *Ultra Secure Mode*.

To customize the banners:

1 In the *Collaboration Server* menu, click **Setup** > **Customize Display Settings > Banners Configuration**.

The *Banners Configuration* dialog box opens.



☑ Login Page Banner: Sample 1

You are accessing a U.S. Government (USG) Information System (IS) that is provided for USG-authorized use only.

By using this IS (which includes any device attached to this IS), you consent to the following conditions:

- The USG routinely intercepts and monitors communications on this IS for purposes including, but not limited to, penetration testing, COMSEC monitoring, network operations and defense, personnel misconduct (PM), law enforcement

☑ Main Page Banner:

2 Customize the banners by modifying the following fields:

Table 19-22 Banner Configuration

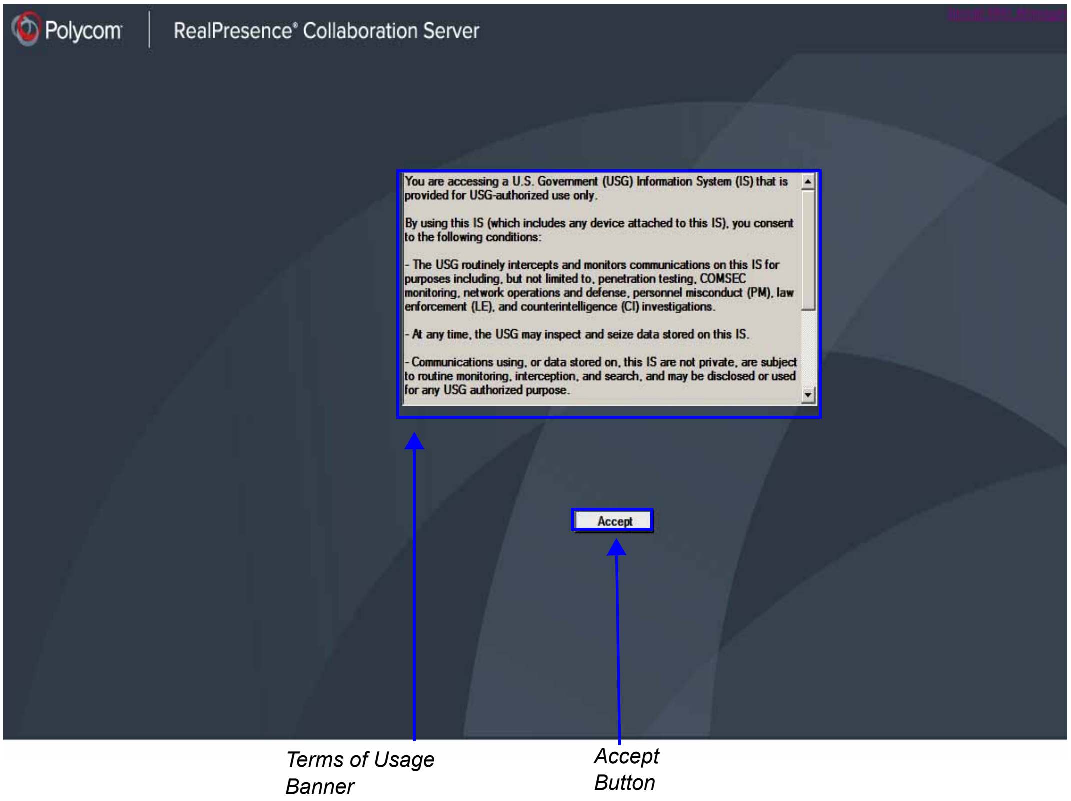
Field	Description		
	Check Box	Text Field	Restore Default Button
Login Page Banner	Select or clear the check box to enable or disable the display of the banner.	Edit the text in this field to meet local requirements: • Banner content is multilingual and uses Unicode, UTF-8 encoding. All text and special characters can be used. • Maximum banner size is 100KB.	Click the button to restore the default text to the banner
Main Page Banner			

3 Click the **OK** button.

Banner Display

Login Screen Banner

The *Login* screen banner can display any text, for example the terms and conditions for system usage. The user must acknowledge that the information was read and click the **Accept** button to proceed to the *Login* screen as shown in the following screen:



You are accessing a U.S. Government (USG) Information System (IS) that is provided for USG-authorized use only.

By using this IS (which includes any device attached to this IS), you consent to the following conditions:

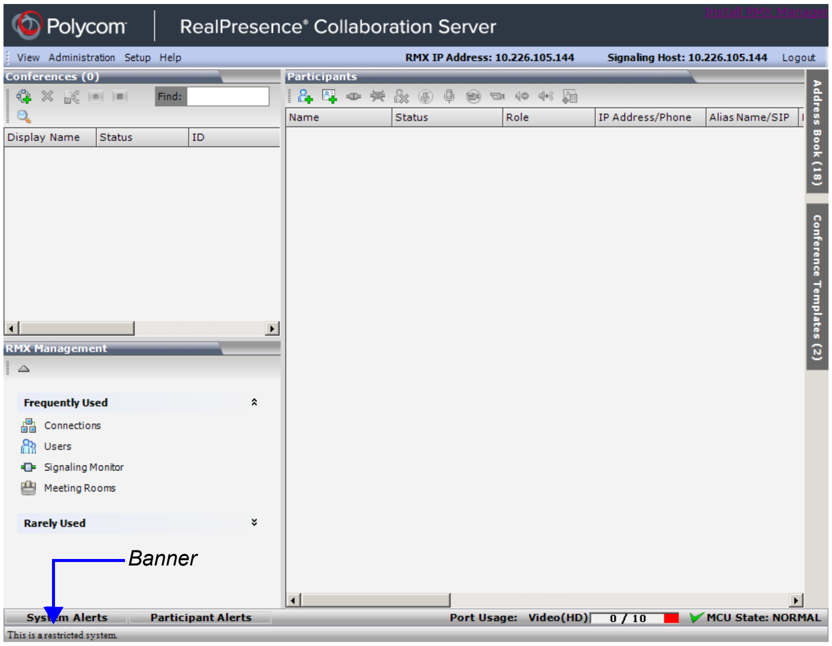
- The USG routinely intercepts and monitors communications on this IS for purposes including, but not limited to, penetration testing, COMSEC monitoring, network operations and defense, personnel misconduct (PM), law enforcement (LE), and counterintelligence (CI) investigations.

- At any time, the USG may inspect and seize data stored on this IS.

- Communications using, or data stored on, this IS are not private, are subject to routine monitoring, interception, and search, and may be disclosed or used for any USG authorized purpose.

Main Screen Banner

The *Main Screen* banner is displayed at the bottom of the screen, as follows:


RMX IP Address: 10.226.105.144

Signaling Host: 10.226.105.144

Port Usage: Video(HD) 0 / 10

MCU State: NORMAL

This is a restricted system.

Software Management

The *Software Management* menu is used to backup and restore the Collaboration Server's configuration files and to download MCU software.

Backup and Restore Guidelines

- *System Backup* can only be performed by an administrator.
- The *System Backup* procedure creates a single backup file that can be viewed or modified only by developers.
- A *System Backup* file from one system can be restored on another system.

This applies only to one RealPresence Collaboration Server Virtual Edition system to another. Do not use a backup file from the RealPresence Collaboration Server Virtual Edition on any other model.

- To ensure file system consistency, do not perform any configuration changes as the system does not suspended them during the backup procedure.
- The following parameters, settings and files are backed up:
  - — MCMS configuration files (/mcms/Cfg):
  - — Network and service configurations,

- — Rooms,
- — Profiles
- — System Flags
- — Resource Allocation
- — IVR messages, music
- — *Collaboration Server Web Client* user setting - fonts, windows
- — *Collaboration Server Web Client* global settings – notes, address book, language
- — Private keys and certificates (TLS)
- — Conference participant settings
- — Operation DB (administrator list)
- — SNMP settings
- — Time configuration

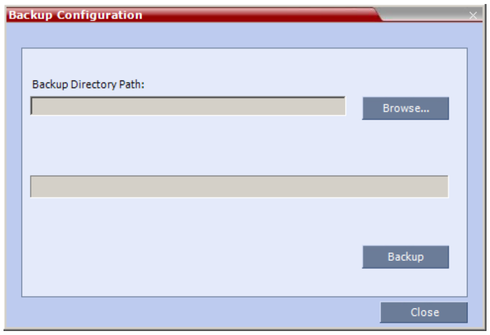
- CDR files are not included in the backup process and should be backed up manually by saving the CDR files to a destination device.

Using Software Management

To backup configuration files:

1 On the *Collaboration Server* menu, click **Administration > Software Management > Backup Configuration**.

The *Backup Configuration* dialog box opens.



2 **Click the Browse** button.
The Browse To File dialog box opens.

3 Select the *Backup Directory Path* and then click **Backup**.

> When the Collaboration Server system backs up the current configuration, if any changes occur immediately or during the request, then additional changes are not registered.

To restore configuration files:

1 On the *Collaboration Server* menu, click **Administration** > **Software Management** > **Restore Configuration**.

2 **Browse** to the *Restore Directory Path* where the backed up configuration files are stored and then click **Restore**.

To download MCU software files:

1 On the *Collaboration Server* menu, click **Administration** > **Software Management** > **Software Download**.

2 **Browse** to the *Install Path* and then click **Install**.

Ping Collaboration Server

The *Ping* administration tool enables the *Collaboration Server Signaling Host* to test network connectivity by *Pinging* IP addresses.

Guidelines

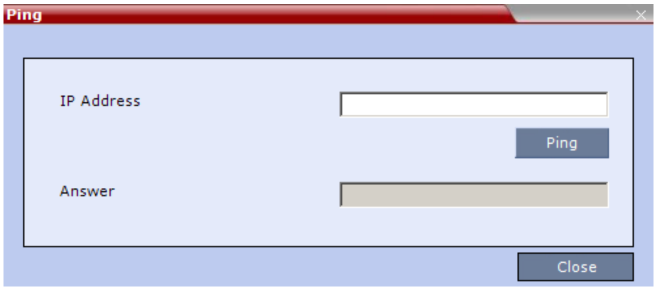
- Both explicit IP addresses and *Host Names* are supported.
- The *Collaboration Server Web Client* blocks any attempt to issue another *Ping* command before the current *Ping* command has completed. Multiple *Ping* commands issued simultaneously from multiple *Collaboration Server Web Client*s are also blocked.

Using Ping

To Ping a network entity from the *Collaboration Server*:

1 On the *Collaboration Server* menu, click **Administration > Tools > Ping**.

The *Ping* dialog box is displayed:



2 Modify or complete the following field:

Table 19-23 *Ping*

Field	Description
Host Name or Address	Enter the *Host Name* or *IP Address* of the *network* entity to be *Pinged*.

3 Click the **Ping** button.

The *Ping* request is sent to the *Host Name* or *IP Address* of the *Collaboration Server* entity.

The *Answer* is either:

— *OK, or*

— *FAILED*

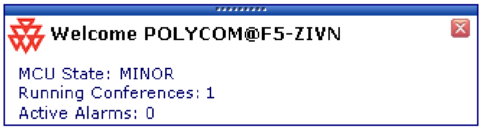
Notification Settings

The Collaboration Server can display notifications when:

- A new Collaboration Server user connects to the MCU.
- A new conference is started.

- Not all defined participants are connected to the conference or when a single participant is connected.
- A change in the MCU status occurs and an alarm is added to the alarm's list.

A welcome message is displayed to the Collaboration Server user upon connection.



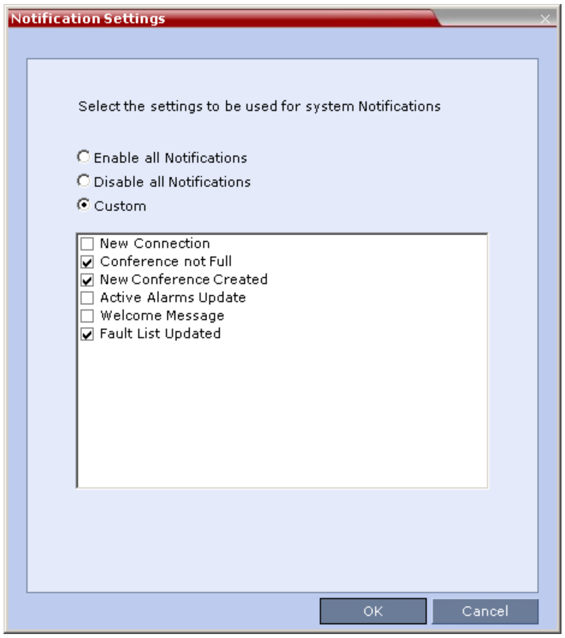
Welcome POLYCOM@F5-ZIVN

MCU State: MINOR
Running Conferences: 1
Active Alarms: 0

To configure the notifications:

1 On the *Collaboration Server* menu, select **Setup** > **Notification Settings**.

The *Notification Settings* dialog box is displayed.



Notification Settings

Select the settings to be used for system Notifications

○ Enable all Notifications
○ Disable all Notifications
● Custom

□ New Connection
☑ Conference not Full
☑ New Conference Created
□ Active Alarms Update
□ Welcome Message
☑ Fault List Updated

The following notification options are displayed.

Table 19-24 Notification Settings Parameters

Field	Description
New Connection	Notification of a new user/administrator connecting to the Collaboration Server.
New Conference Created	New conference has been created.
Conference Not Full	The conference is not full and additional participants are defined for the conference.
Welcome Message	A welcome message after user/administrator logon.
Active Alarms Update	Updates you of any new alarm that occurred.
Fault List Updated	Updates you when the faults list is updated (new faults are added or existing faults are removed).

2 **Enable/Disable All Notifications** or **Custom** to select specific notifications to display.

3 Click **OK**.

Logger Diagnostic Files

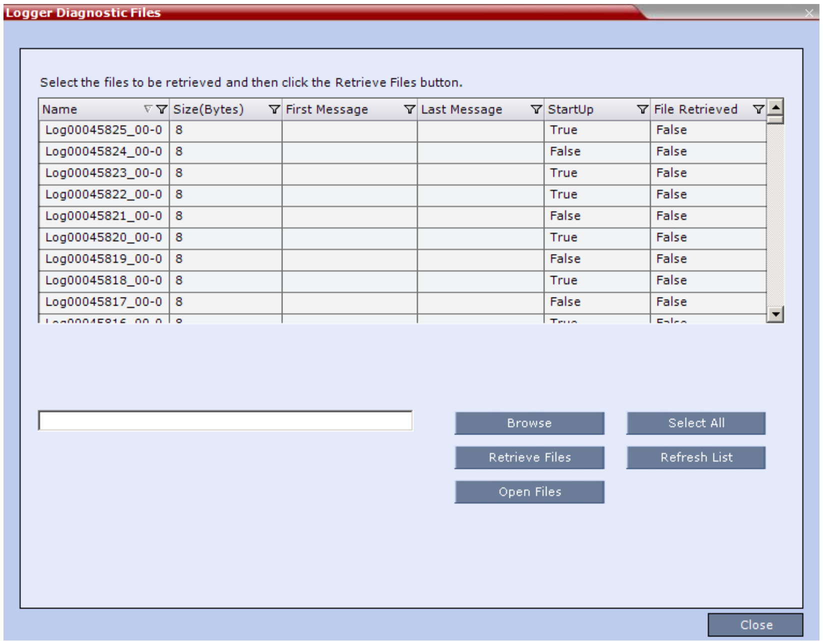
The Logger utility is a troubleshooting tool that continually records MCU system messages and saves them to files in the MCU hard drive. For each time interval defined in the system, a different data file is created. The files may be retrieved from the hard drive for off-line analysis and debugging purposes.

The Logger utility is activated at the MCU startup. The Logger is disabled when the MCU is reset manually or when there is a problem with the Logger utility, e.g. errors on the hard drive where files are saved. In such cases, data cannot be retrieved.

When the MCU is reset via the Collaboration Server, the files are saved on the MCU hard drive.

To access the Logger Diagnostic Files:

>> On the *Collaboration Server* menu, click **Administration > Tools > Logger Diagnostic Files**.



Select the files to be retrieved and then click the Retrieve Files button.

Name	Size(Bytes)	First Message	Last Message	StartUp	File Retrieved
Log00045825_00-0	8			True	False
Log00045824_00-0	8			False	False
Log00045823_00-0	8			True	False
Log00045822_00-0	8			True	False
Log00045821_00-0	8			False	False
Log00045820_00-0	8			True	False
Log00045819_00-0	8			False	False
Log00045818_00-0	8			True	False
Log00045817_00-0	8			False	False
Log00045816_00-0	8			True	False

The following tasks can be performed:

Table 19-25 Diagnostic File Button Options

Button	Description
Refresh List	Refreshes the list and adds newly generated logger files.
Select All	Selects all the logger files listed.
Browse	Selects the destination folder for download.
Retrieve Files	Saves files to the destination folder.

When retrieved, the log file name structure is as follows:

- Sequence number (starting with 1)
- Date and Time of first message
- Date and Time of last message
- File size
- Special information about the data, such as Startup

File name structure:

Log_SNxxxxxxxxxx_FMDddmmyyy_FMThhmm_LMDddmmyyyy_LMThhmm_SZxxxxxxxxxx_SUY.log

File name format:

- SN = Sequence Number
- FM = First Message, date and time
- LM = Last Message, date and time
- SZ = Size
- SU = Startup (Y/N) during the log file duration

Example:

Log_SN0000000002_FMD06032007_FMT083933_LMD06032007_LMT084356_SZ184951_SUY.log.

Retrieving the Logger Files:

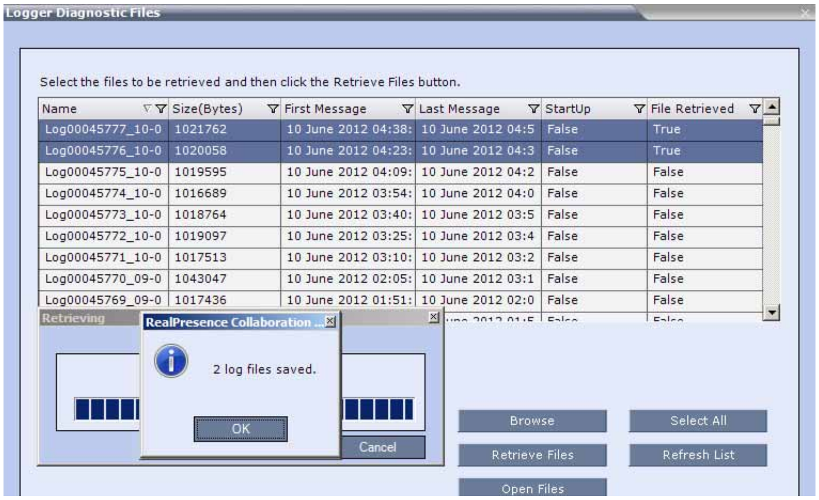
1 Select the log files to retrieve. Multiple selections of files are enabled using standard Windows conventions.

2 In the *Logger Diagnostic Files* dialog box, click the **Browse** button.

3 In the *Browse for Folder* window, select the directory location to save the Logger files and click **OK**.

You will return to the *Logger Diagnostic Files* dialog box.

4 Click the **Retrieve Files** button.



Logger Diagnostic Files

Select the files to be retrieved and then click the Retrieve Files button.

Name	Size(Bytes)	First Message	Last Message	StartUp	File Retrieved
Log00045777_10-0	1021762	10 June 2012 04:38:	10 June 2012 04:5	False	True
Log00045776_10-0	1020058	10 June 2012 04:23:	10 June 2012 04:3	False	True
Log00045775_10-0	1019595	10 June 2012 04:09:	10 June 2012 04:2	False	False
Log00045774_10-0	1016689	10 June 2012 03:54:	10 June 2012 04:0	False	False
Log00045773_10-0	1018764	10 June 2012 03:40:	10 June 2012 03:5	False	False
Log00045772_10-0	1019097	10 June 2012 03:25:	10 June 2012 03:4	False	False
Log00045771_10-0	1017513	10 June 2012 03:10:	10 June 2012 03:2	False	False
Log00045770_09-0	1043047	10 June 2012 02:05:	10 June 2012 03:1	False	False
Log00045769_09-0	1017436	10 June 2012 01:51:	10 June 2012 02:0	False	False

Retrieving

RealPresence Collaboration ...

2 log files saved.

The log files (in *.txt format) are saved to the defined directory and a confirmation caption box is displayed indicating a successful retrieval of the log files.

Viewing the Logger Files:

To analyze the log files generated by the system, open the retrieved *.txt files in any text editor application, i.e. Notepad, Textpad or MS Word.

1. Using Windows Explorer, browse to the directory containing the retrieved log files.
2. Use any text editor application to open the log file(s).

Information Collector

Standard Security Mode

The Information Collector comprehensively attains all information from all the MCU internal entities for data analysis. That data, stored in a central repository, is logged from the following system components:

- System Log Files
- CDR
- OS (Core dumps, CFG - DNS, DHCP, NTP, kernal state, event logs
- Signaling Trace files (H.323 & SIP)
- Central Signaling logs
- Processes internal state and statistics
- Full faults
- Apache logs
- CFG directory (without IVR)
- Cards info: HW version, state and status
- SW version number

The data collected is saved into a single compressed file containing all the information from each system component in its relative format (.txt, .xml, etc...). In case the disk is malfunctioning, the file will be written to the RAM (involves only a small amount of information where the RAM size is 1/2 a gigabyte). The zipped file (info.tgz) can be opened with the following applications: WinRAR and WinZip. The entire zipped file is then sent to Polycom's Network Systems Division for analysis and troubleshooting.

Using the Information Collector

When the *Information Collector* is used the following steps are performed:

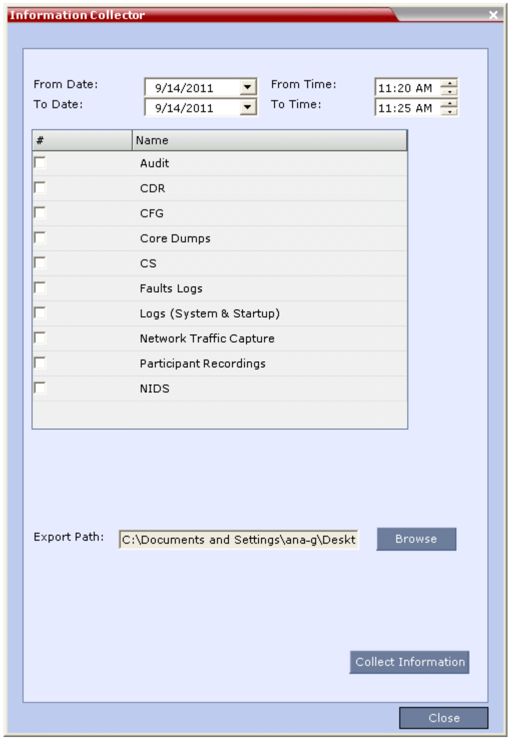
- **Step 1: Creating** the *Information Collector* file.
- **Step 2**: **Saving** the *Information Collector* file.
- **Step 3: Viewing** the information in the *Information Collector* file.

Step 1: Creating the Information Collector Compressed File

To create the compressed file:

1 In the Collaboration Server menu, click **Administration > Tools > Information Collector**.
The *Information Collector* dialog box is displayed.

Standard Security Mode



Information Collector

From Date: 9/14/2011 From Time: 11:20 AM
To Date: 9/14/2011 To Time: 11:25 AM

#	Name
□	Audit
□	CDR
□	CFG
□	Core Dumps
□	CS
□	Faults Logs
□	Logs (System & Startup)
□	Network Traffic Capture
□	Participant Recordings
□	NIDS

Export Path: C:\Documents and Settings\ana-g\Deskt

2 In the *From Date* and *Until Date* fields, use the arrow keys to define the date range of the data files to be included in the compressed file.

3 In the *From Time* and *Until Time* fields, use the arrow keys to define the time range of the data files to be included in the compressed file.

If logs are being collected in order to troubleshoot a specific issue, it is important that the date and time range include the time and date in which the issue occurred. The default date and time ranges may not be sufficient.
For example, if a specific issue occurred on October 1, 2013 at 12:15, the *From Date* and *Until Date* should be October 1, 2013, the *From Time* should be around 12:10, and the *Until Time* should be around 12:20.

4 Select check boxes of the information to be collected.

5 In the *Export Path* field, click the **Browse** button and navigate to the directory path where the compressed file is to be saved.

6 Click the **Collect Information** button.

A progress indicator is displayed in the *Information Collector* dialog box while the file is being created.

Step 2: Saving the Compressed File

1. The compressed file is automatically saved in the directory selected in the *Information Collector* dialog box. The file is named **info.tgz**.

   A success information box is displayed.
2. Click the **OK** button.

Step 3: Viewing the Compressed File

The compressed file is saved in *.tgz* format and can be viewed with any utility that can open files of that format, for example *WinRAR® 3.80*.

To view the compressed file:

1. Navigate to the directory on the workstation in which the file was saved.
2. Double click the **info.tgz** file to view the downloaded information.

> Some browsers save the file as *info.gz* due to a browser bug. If this occurs, the file must be manually renamed to *info.tgz* before it can be viewed.

Auditor

An *Auditor* is a user who can view *Auditor* and *CDR* files for system auditing purposes.

> The *Auditor* user must connect to the *Collaboration Server* using the *Collaboration Server Web Client* only.

The *Event Auditor* enables administrators and auditors to analyze configuration changes and unusual or malicious activities in the *Collaboration Server* system.

Auditor operates in real time, recording all administration activities and login attempts from the following *Collaboration Server* modules:

- Control Unit
- Shelf Manager

For a full list of monitored activities, see Table 19-27 on page **19-52** and Table 19-28 on page **19-54**.

The *Auditor* must always be active in the system. A *System Alert* is displayed if it becomes inactive for any reason.

The *Auditor* tool is composed of the *Auditor Files* and the *Auditor File Viewer* that enables you to view the *Auditor Files*.

> Time stamps of *Audit Events* are GMT.

Auditor Files

Auditor Event History File Storage

All audit events are saved to a buffer file on hard disk in real time and then written to a file on hard disk in XML in an uncompressed format.

A new current auditor event file is created when:

- the system is started
- the size of the current auditor event file exceeds 2 MB
- the current auditor event file’s age exceeds 24 hours

Up to 1000 auditor event files are stored per *Collaboration Server*. These files are retained for at least one year and require 1.05 GB of disk space. The files are automatically deleted by the system (oldest first) when the system reaches the auditor event file limit of 1000.

A *System Alert* is displayed with *Can't store data* displayed in its *Description* field if:

- the system cannot store 1000 files
- the *Collaboration Server* does not have available disk space to retain files for one year

Audit Event Files are retained by the *Collaboration Server* for at least 1 year. Any attempt to delete an audit event file that is less than one year old raises a *System Alert* with *File was removed* listed in the *Description* field.

Using the *Restore Factory Defaults* of the *System Restore* procedure erases *Audit Files*.

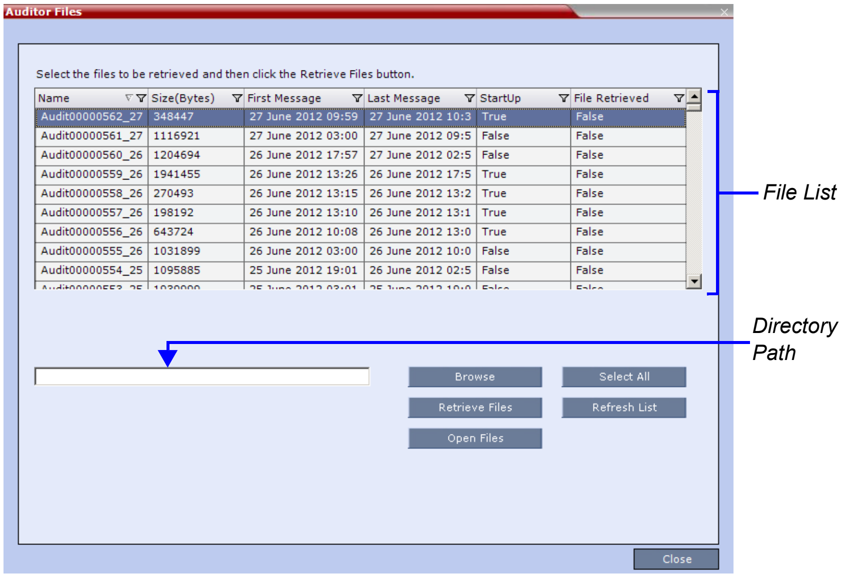
Retrieving Auditor Files

You can open the *Auditor* file directly from the *Auditor Files* list or you can retrieve the files and save them to a local workstation.

To access Auditor Files:

>> On the *Collaboration Server* menu, click **Administration > Tools > Auditor Files**.

The *Auditor Files* dialog box is displayed.



Select the files to be retrieved and then click the Retrieve Files button.

Name	Size(Bytes)	First Message	Last Message	StartUp	File Retrieved
Audit00000562_27	348447	27 June 2012 09:59	27 June 2012 10:3	True	False
Audit00000561_27	1116921	27 June 2012 03:00	27 June 2012 09:5	False	False
Audit00000560_26	1204694	26 June 2012 17:57	27 June 2012 02:5	False	False
Audit00000559_26	1941455	26 June 2012 13:26	26 June 2012 17:5	True	False
Audit00000558_26	270493	26 June 2012 13:15	26 June 2012 13:2	True	False
Audit00000557_26	198192	26 June 2012 13:10	26 June 2012 13:1	True	False
Audit00000556_26	643724	26 June 2012 10:08	26 June 2012 13:0	True	False
Audit00000555_26	1031899	26 June 2012 03:00	26 June 2012 10:0	False	False
Audit00000554_25	1095885	25 June 2012 19:01	26 June 2012 02:5	False	False
Audit00000553_25	1030000	25 June 2012 03:01	25 June 2012 10:0	False	False

The *Auditor Files* dialogue box displays a file list containing the following file information:

— *Name*

— Size (Bytes)

— *First Message* – date and time of the first audit event in the file

— *Last Message* – date and time of the last audit event in the file

— StartUp:

- *True* – file was created when the system was started

  - *False* – file was created when previous audit event file reached a size of 2 MB or was more than 24 hours old

— File Retrieved:

  - *True* - file was previously retrieved.
  - *False* - file was never previously retrieved.

The order of the *Auditor Files* dialog box field header columns can be changed and the fields can be filtered to enable searching.

For more information, see "*Auditor File Viewer"* on page **19-50**.

To retrieve files for storage on a workstation:

1 Click **Browse** and select the folder on the workstation to receive the files and then click **OK**.

The folder name is displayed in the directory path field.

2 Select the file(s) to be retrieved by clicking their names in the file list or click **Select All** to retrieve all the files. (Windows multiple selection techniques can be used.)

3 Click Retrieve Files.

The selected files are copied to the selected directory on the workstation.

To open the file in the Auditor File Viewer:

>> Double-click the file.

Auditor File Viewer

The *Auditor File Viewer* enables *Auditors* and *Administrators* to view the content of and perform detailed analysis on auditor event data in a selected *Auditor Event File*.

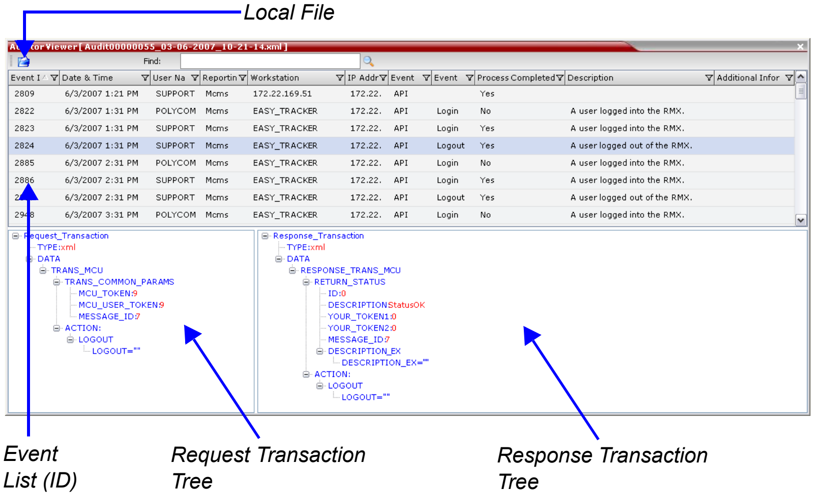
You can view an *Auditor Event File* directly from the *Auditor Files* list or by opening the file from the *Auditor File Viewer*.

To open the Auditor File Viewer from the Administration Menu:

1 On the *Collaboration Server* menu, click **Administration > Tools > Auditor File Viewer**.

The *Auditor File Viewer* is displayed.

If you previously double clicked an *Auditor Event File* in the *Auditor Files* list, that file is automatically opened.


Local File
Au[illegible]tor Viewer [Audit00000055_03-06-2007_10-21-14.xml]
Find:
Event List (ID)
Request Transaction Tree
Response Transaction Tree

Event I	Date & Time	User Na	Reportin	Workstation	IP Addr	Event	Event	Process Completed	Description	Additional Infor
2809	6/3/2007 1:21 PM	SUPPORT	Mcms	172.22.169.51	172.22.	API		Yes		
2822	6/3/2007 1:31 PM	POLYCOM	Mcms	EASY_TRACKER	172.22.	API	Login	No	A user logged into the RMX.	
2823	6/3/2007 1:31 PM	SUPPORT	Mcms	EASY_TRACKER	172.22.	API	Login	Yes	A user logged into the RMX.	
2824	6/3/2007 1:31 PM	SUPPORT	Mcms	EASY_TRACKER	172.22.	API	Logout	Yes	A user logged out of the RMX.	
2885	6/3/2007 2:31 PM	POLYCOM	Mcms	EASY_TRACKER	172.22.	API	Login	No	A user logged into the RMX.	
2886	6/3/2007 2:31 PM	SUPPORT	Mcms	EASY_TRACKER	172.22.	API	Login	Yes	A user logged into the RMX.	
2[illegible]	6/3/2007 2:31 PM	SUPPORT	Mcms	EASY_TRACKER	172.22.	API	Logout	Yes	A user logged out of the RMX.	
2948	6/3/2007 3:31 PM	POLYCOM	Mcms	EASY_TRACKER	172.22.	API	Login	No	A user logged into the RMX.	

```
Request_Transaction
  TYPE:xml
  DATA
    TRANS_MCU
      TRANS_COMMON_PARAMS
        MCU_TOKEN:9
        MCU_USER_TOKEN:9
        MESSAGE_ID:7
      ACTION:
        LOGOUT
          LOGOUT=""
```

```
Response_Transaction
  TYPE:xml
  DATA
    RESPONSE_TRANS_MCU
      RETURN_STATUS
        ID:0
        DESCRIPTIONStatusOK
        YOUR_TOKEN1:0
        YOUR_TOKEN2:0
        MESSAGE_ID:7
        DESCRIPTION_EX
          DESCRIPTION_EX=""
      ACTION:
        LOGOUT
          LOGOUT=""
```

The following fields are displayed for each event:

Table 19-26 *Auditor Event Columns*

Field	Description
Event ID	The sequence number of the event generated by the *Collaboration Server*.
Date & Time	The date and time of the event taken from the *Collaboration Server*'s *Local Time* setting.
User Name	The *Username* (Login Name) of the user who triggered the event.
Reporting Module	The *Collaboration Server* system internal module that reported the event: • MCMS • MPL • Central Signaling • MPL Simulation • *Collaboration Server* Web Client • CM Switch • ART • Video • MUX
Workstation	The name (alias) of the workstation used to send the request that triggered the event.
IP Address (Workstation)	The IP address of the workstation used to send the request that triggered the event.
Event Type	Auditor events can be triggered by: • API • HTTP • *Collaboration Server* Internal Event
Event	The process, action, request or transaction that was performed or rejected. • POST:SET transactions (API) • Configuration changes via XML (API) • Login/Logout (API) • GET (HTTP) • PUT (HTTP) • MKDIR (HTTP) • RMDIR (HTTP) • Startup (*Collaboration Server* Internal Event) • Shutdown (*Collaboration Server* Internal Event) • Reset (*Collaboration Server* Internal Event) • Enter Diagnostic Mode (*Collaboration Server* Internal Event) • IP address changes via USB (*Collaboration Server* Internal Event)

Table 19-26 Auditor Event Columns (Continued)

Field	Description
Process Completed	Status of the process, action, request or transaction returned by the system: • Yes – performed by the system. • No – rejected by the system.
Description	A text string describing the process, action, request or transaction.
Additional Information	An optional text string describing the process, action, request or transaction in additional detail.

The order of the *Auditor File Viewer* field header columns can be changed and the fields can be sorted and filtered to facilitate different analysis methods.

2 In the event list, click the events or use the keyboard's Up-arrow and Down-arrow keys to display the *Request Transaction* and *Response Transaction* XML trees for each audit event.

The transaction XML trees can be expanded and collapsed by clicking the expand (⊞) and collapse (⊟) buttons.

To open an auditor event file stored on the workstation:

1 Click the **Local File** button (📂) to open the *Open* dialogue box.

2 Navigate to the folder on the workstation that contains the audit event file.

3 Select the audit event file to be opened.

4 Click **Open**.

The selected file is opened in the *Auditor Viewer*.

Audit Events

Alerts and Faults

Table 1 lists *Alerts* and *Faults* that are recorded by the *Auditor*.

Table 19-27 Alerts and Faults

Event
Attempt to exceed the maximum number of management session per user
Attempt to exceed the maximum number of management sessions per system
Central Signaling indicating Recovery status.
Failed login attempt
Failed to open Apache server configuration file.
Failed to save Apache server configuration file.
Fallback version is being used.

Table 19-27 Alerts and Faults (Continued)

Event
File system scan failure.
File system space shortage.
Internal MCU reset.
Internal System configuration during startup.
Invalid date and time.
Invalid MCU Version.
IP addresses of Signaling Host and Control Unit are the same.
IP Network Service configuration modified.
IP Network Service deleted.
Login
Logout
Management Session Time Out
MCU Reset to enable Diagnostics mode.
MCU reset.
Music file error.
New activation key was loaded.
New version was installed.
NTP synchronization failure.
Polycom default User exists.
Private version is loaded.
Restoring Factory Defaults.
Secured SIP communication failed.
Session disconnected without logout
SSH is enabled.
System Configuration modified.
System is starting.
System Resets.
TCP disconnection
Terminal initiated MCU reset.
The Log file system is disabled.
The software contains patch(es).

***Table 19-27** Alerts and Faults (Continued)*

Event
USB key used to change system configuration.
User closed the browser
User initiated MCU reset.

Transactions

Table 2 lists *Transactions* that are recorded by the *Auditor*.

***Table 19-28** Transactions*

Transaction
TRANS_CFG:SET_CFG
TRANS_IP_SERVICE:DEL_IP_SERVICE
TRANS_IP_SERVICE:NEW_IP_SERVICE
TRANS_IP_SERVICE:SET_DEFAULT_H323_SERVICE
TRANS_IP_SERVICE:SET_DEFAULT_SIP_SERVICE
TRANS_IP_SERVICE:UPDATE_IP_SERVICE
TRANS_IP_SERVICE:UPDATE_MANAGEMENT_NETWORK
TRANS_MCU:BEGIN_RECEIVING_VERSION
TRANS_MCU:COLLECT_INFO
TRANS_MCU:CREATE_DIRECTORY
TRANS_MCU:FINISHED_TRANSFER_VERSION
TRANS_MCU:LOGIN
TRANS_MCU:LOGOUT
TRANS_MCU:REMOVE_DIRECTORY
TRANS_MCU:REMOVE_DIRECTORY_CONTENT
TRANS_MCU:RENAME
TRANS_MCU:RESET
TRANS_MCU:SET_PORT_CONFIGURATION
TRANS_MCU:SET_RESTORE_TYPE
TRANS_MCU:SET_TIME
TRANS_MCU:TURN_SSH
TRANS_MCU:UPDATE_KEY_CODE
TRANS_OPERATOR:CHANGE_PASSWORD

***Table 19-28** Transactions (Continued)*

Transaction
TRANS_OPERATOR:DELETE_OPERATOR
TRANS_OPERATOR:NEW_OPERATOR
TRANS_SNMP:UPDATE

ActiveX Bypass

At sites that, for security reasons, do not permit Microsoft® ActiveX® to be installed, the MSI (Windows Installer File) utility can be used to install .NET Framework and .NET Security Settings components on workstations throughout the network.

All workstation that connect to *Collaboration Server* systems must have both.NET Framework and .NET Security Settings running locally. These components are used for communication with the Collaboration Server and can only be installed on workstations by users with administrator privileges.

The MSI utility requires the IP addresses of all the Collaboration Server systems (both control unit and Shelf Management IP addresses) that each workstation is to connect to. If the IP address of the any of the target Collaboration Servers is changed, the ActiveX components must be reinstalled.

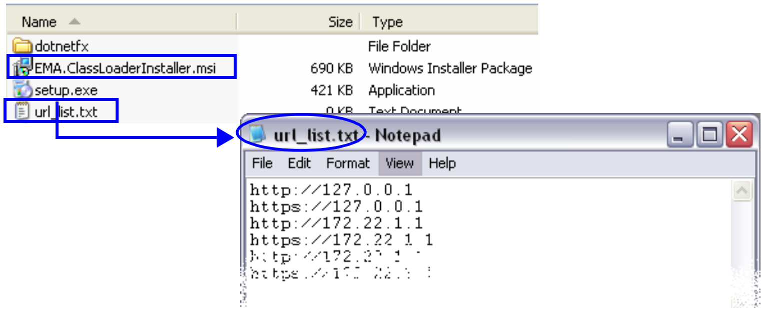
Installing ActiveX

To install ActiveX components on all workstations in the network:

1 Download the MSI file **EMA.ClassLoaderInstaller.msi** from the Polycom Resource Center.
The MSI file contains installation scripts for both .NET Framework and .NET Security Settings.

2 Create a text file to be used during the installation containing the IP addresses of all the Collaboration Server systems (both control unit and Shelf Management IP addresses) that each workstation in the network is to connect to.

The file must be named **url_list.txt** and must be saved in the same folder as the downloaded MSI file.



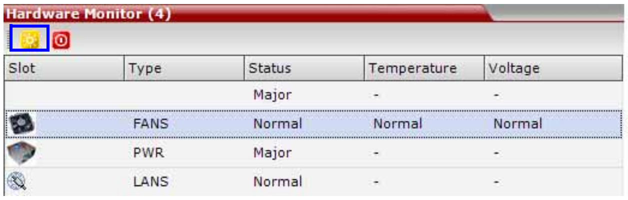
Name	Size	Type
dotnetfx		File Folder
EMA.ClassLoaderInstaller.msi	690 KB	Windows Installer Package
setup.exe	421 KB	Application
url_list.txt	0 KB	Text Document

url_list.txt - Notepad

```
http://127.0.0.1
https://127.0.0.1
http://172.22.1.1
https://172.22.1.1
http://172.2[illegible]
https://1[illegible]
```

3 Install the ActiveX components on all workstations on the network that connect to Collaboration Server systems.

The installation is done by the network administrator using a 3rd party network software installation utility and is transparent to all other users.



Slot	Type	Status	Temperature	Voltage
		Major	-	-
	FANS	Normal	Normal	Normal
	PWR	Major	-	-
	LANS	Normal	-	-

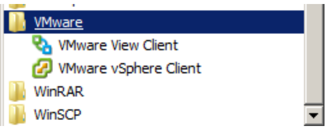
Resetting the Collaboration Server

To restart the MCU instance:

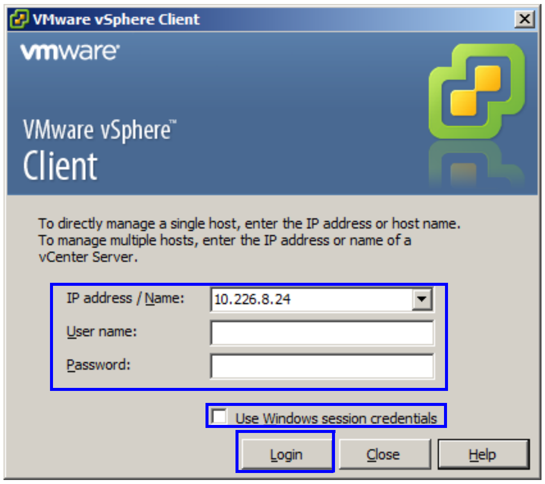
1 Click **Start > Programs.**

a If the **VMware vSphere Client** is displayed in the recently used programs list, click **VMware vSphere Client** in the list to start the application.
or

b Click **All Programs > VMware > VMware vSphere Client.**



The *VMware vSphere Client* login window opens.



To directly manage a single host, enter the IP address or host name.
To manage multiple hosts, enter the IP address or name of a vCenter Server.

IP address / Name: 10.226.8.24

User name:

Password:

2 In the *IP address / Name* field, enter the IP Address or the name of the **vSphere** host.

3 Enter the User Name and password by either:

a In the *User name* field, enter the user name of the with which you will log in to the **vSphere** host.

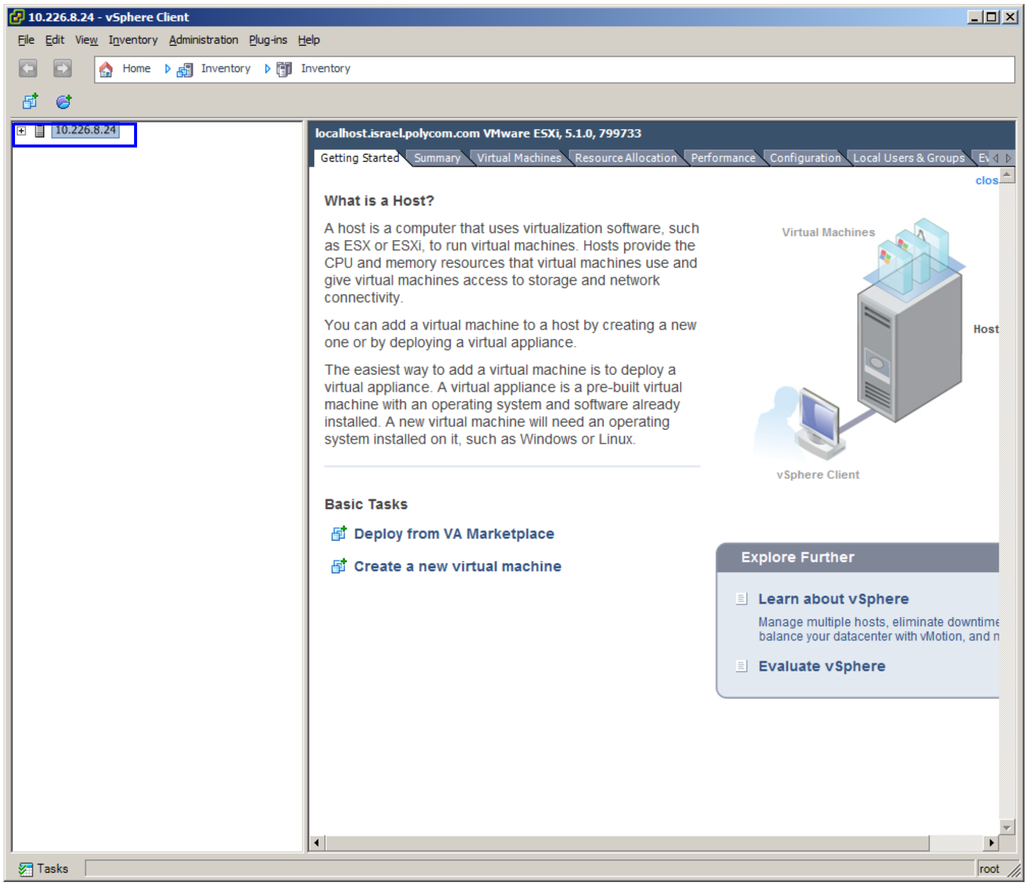
In the *Password* field, enter the password as defined for the user name with which you will log in to the **vSphere** host.

or

b Click the **Use Windows session credentials** check box.

4 Click **Login.**

The *VMware vSphere Client* opens.



localhost.israel.polycom.com VMware ESXi, 5.1.0, 799733

What is a Host?

A host is a computer that uses virtualization software, such as ESX or ESXi, to run virtual machines. Hosts provide the CPU and memory resources that virtual machines use and give virtual machines access to storage and network connectivity.

You can add a virtual machine to a host by creating a new one or by deploying a virtual appliance.

The easiest way to add a virtual machine is to deploy a virtual appliance. A virtual appliance is a pre-built virtual machine with an operating system and software already installed. A new virtual machine will need an operating system installed on it, such as Windows or Linux.

Basic Tasks

Deploy from VA Marketplace

Create a new virtual machine

Explore Further

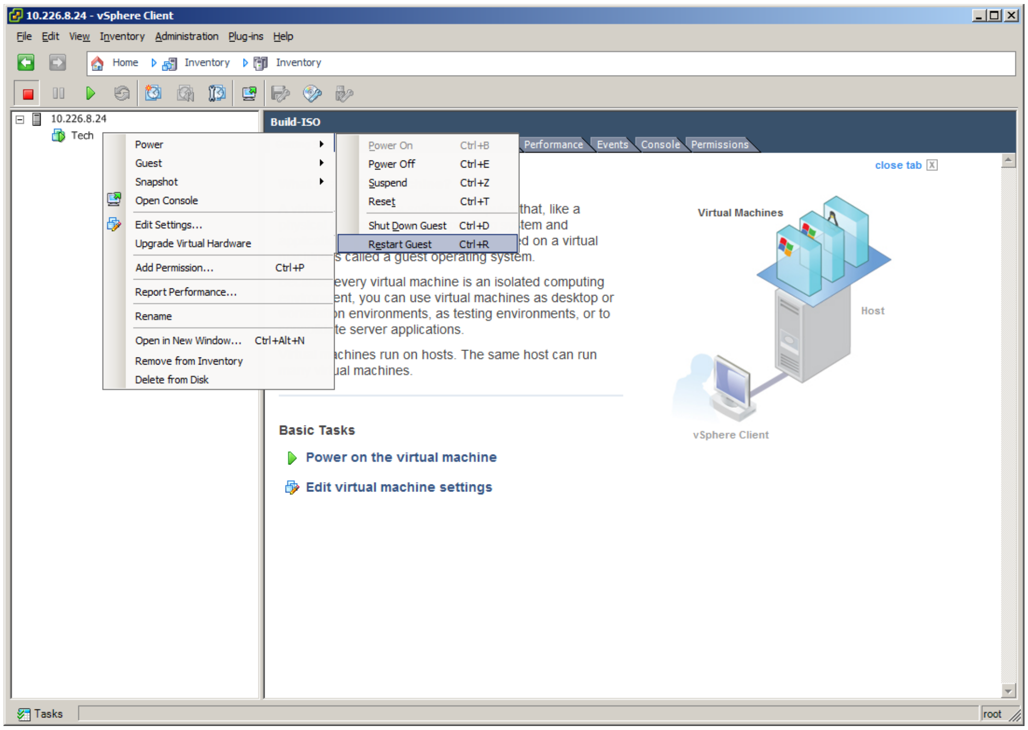
Learn about vSphere

Manage multiple hosts, eliminate downtime balance your datacenter with vMotion, and n

Evaluate vSphere

5 In the *Inventory Panel*, click the *Datastore* that houses the MCU.

6 Right-click on the MCU virtual machine, then click **Power > Restart Guest**



DO NOT click Reset. Doing so may corrupt the Virtual Machine.

The virtual machine and the MCU instance restart.

Upgrading and Downgrading

This procedure allows an administrator to update the MCU instance without requiring the administrator to reregister the product.

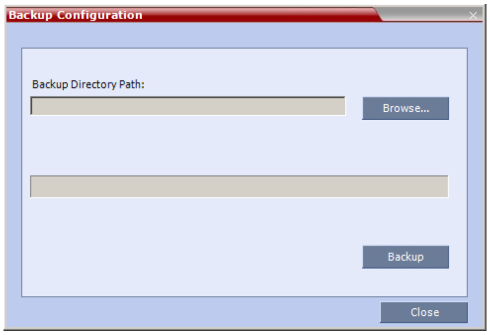
Updating the MCU instance requires the previously used activation key. If you no longer have the activation key, contact support before starting this procedure.

See http://kb.vmware.com/selfservice/microsites/search.do?language=en_US&cmd=displayKC&externalId=1714 before proceeding.

To update the MCU instance:

1 On the *RealPresence Collaboration Server* menu, click **Administration > Software Management > Backup Configuration**.

The *Backup Configuration* dialog box opens.



2 **Click the Browse** button.
The Browse To File dialog box opens.

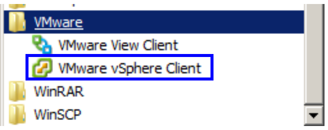
3 Select the *Backup Directory Path* and then click **Backup**.

> When the *RealPresence Collaboration Server* backs up the current configuration, if any changes occur immediately or during the request, then additional changes are not registered.

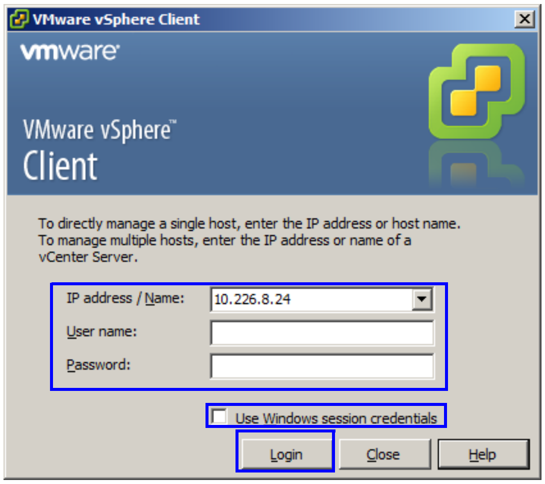
4 On the Windows taskbar, click the **Start > Programs.**

a If the *VMware vSphere Client* is displayed in the recently used programs list, click **VMware vSphere Client** in the list to start the application.
or

b Select **All Programs > VMware > VMware vSphere Client.**



The *VMware vSphere Client* login window is displayed.



To directly manage a single host, enter the IP address or host name.
To manage multiple hosts, enter the IP address or name of a vCenter Server.

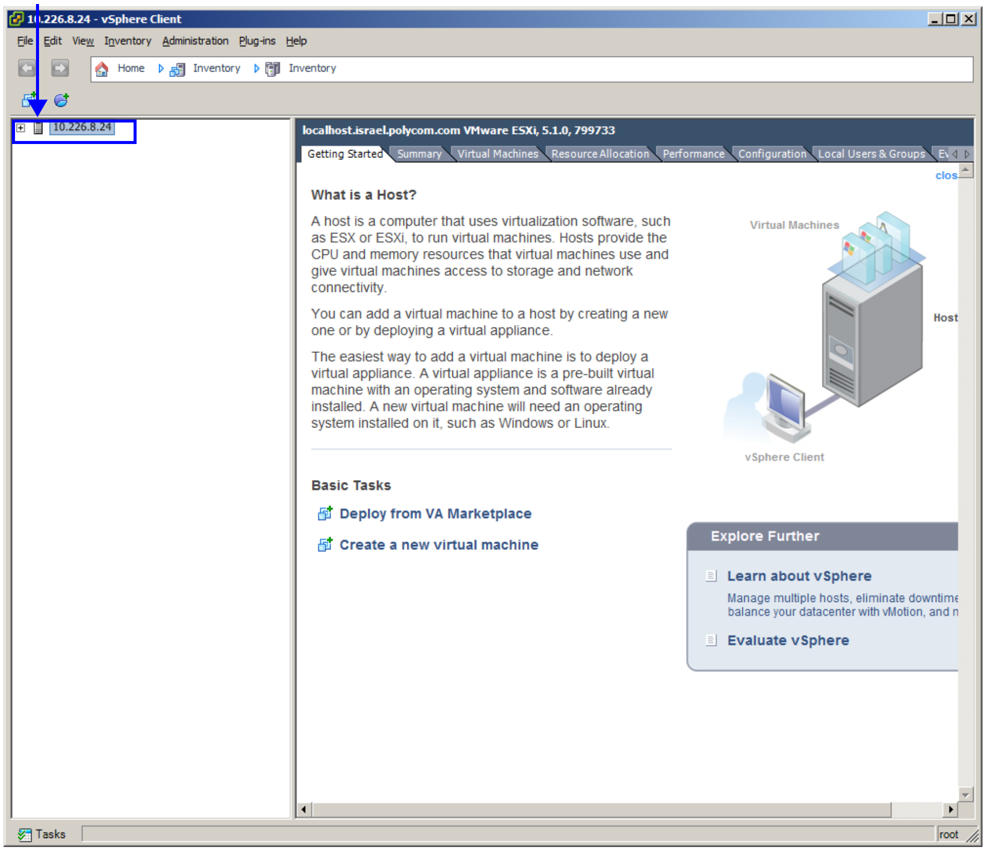
IP address / Name: 10.226.8.24
User name:
Password:

5 In the *IP address / Name* field, enter the IP Address or the name of the **vSphere** host.

6 Either type your **vSphere** *User Name* and *Password* or select **Use Windows sessions credentials**.

7 Click **Login.**

The *VMware vSphere Client* is displayed.



Inventory Panel

What is a Host?

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The easiest way to add a virtual machine is to deploy a virtual appliance. A virtual appliance is a pre-built virtual machine with an operating system and software already installed. A new virtual machine will need an operating system installed on it, such as Windows or Linux.

Basic Tasks

Deploy from VA Marketplace

Create a new virtual machine

Explore Further

Learn about vSphere

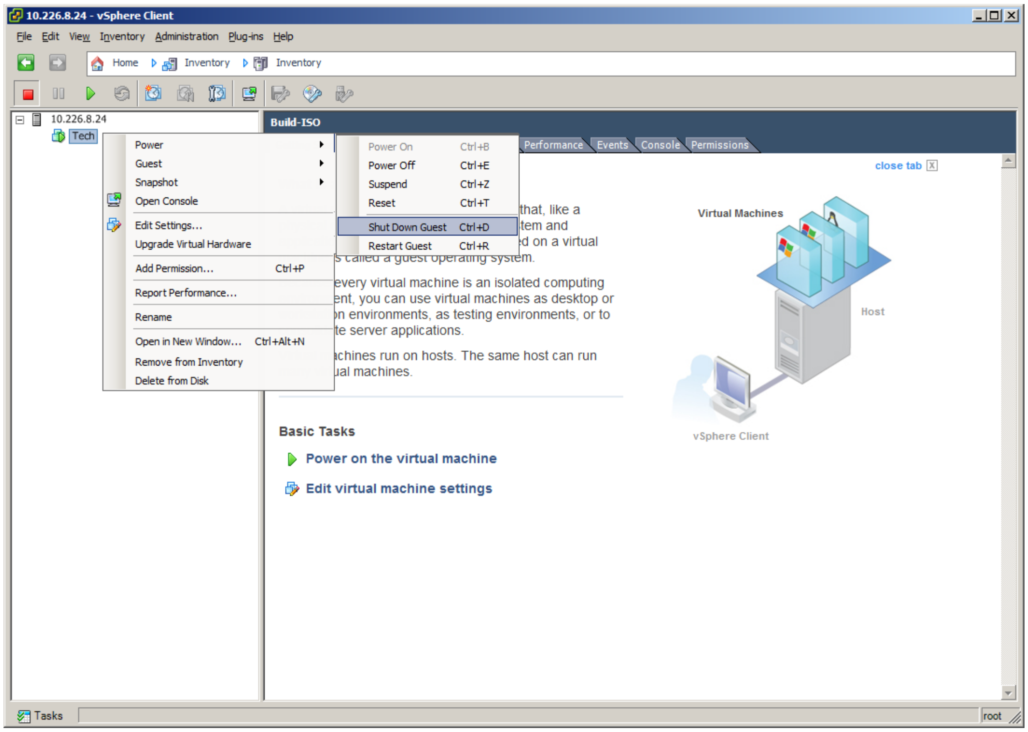
Manage multiple hosts, eliminate downtime
balance your datacenter with vMotion, and n

Evaluate vSphere

8 In the *Inventory Panel*, select the *Datastore* that houses the MCU.

The inventory of the Datastore appears.

9 Right-click the MCU virtual machine, then click **Power > Shut Down Guest**.



that, like a
tem and
d on a virtual
s called a guest operating system.

every virtual machine is an isolated computing
ent, you can use virtual machines as desktop or
n environments, as testing environments, or to
te server applications.

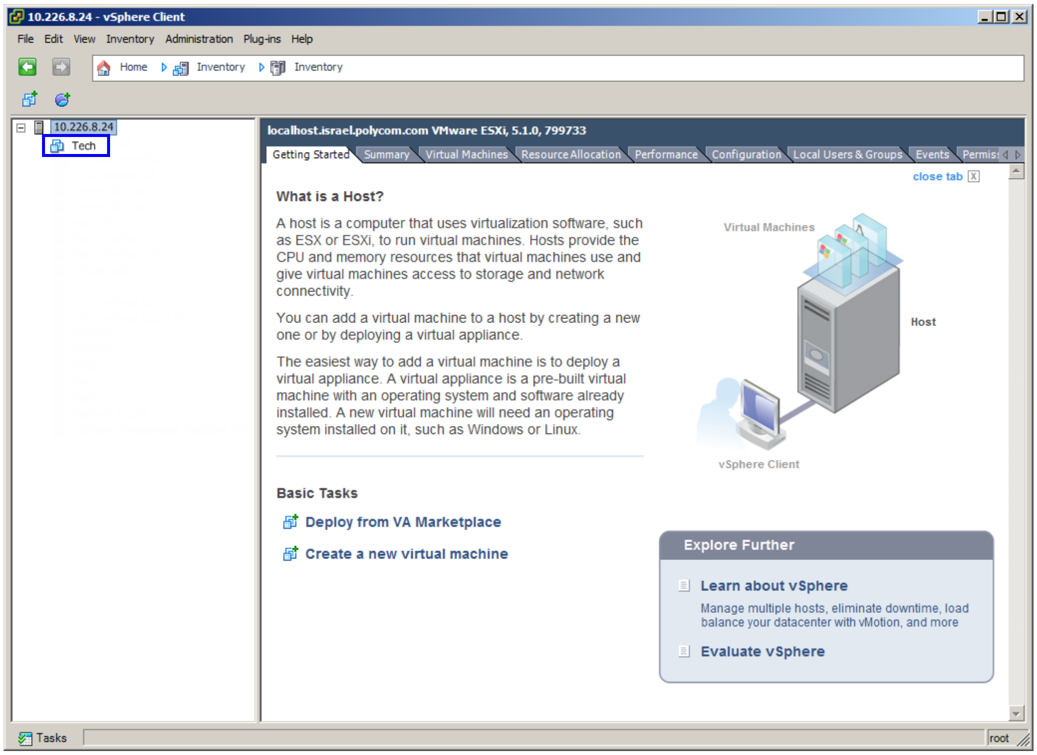
chines run on hosts. The same host can run
ual machines.

Basic Tasks

Power on the virtual machine

Edit virtual machine settings

When VM turns blue, the virtual machine has shut down.


10.226.8.24 - vSphere Client
File Edit View Inventory Administration Plug-ins Help
Home
Inventory
Inventory
10.226.8.24
Tech
localhost.israel.polycom.com VMware ESXi, 5.1.0, 799733
Getting Started Summary Virtual Machines Resource Allocation Performance Configuration Local Users & Groups Events Permis
close tab
Virtual Machines
Host
vSphere Client
Tasks
root

What is a Host?

A host is a computer that uses virtualization software, such as ESX or ESXi, to run virtual machines. Hosts provide the CPU and memory resources that virtual machines use and give virtual machines access to storage and network connectivity.

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The easiest way to add a virtual machine is to deploy a virtual appliance. A virtual appliance is a pre-built virtual machine with an operating system and software already installed. A new virtual machine will need an operating system installed on it, such as Windows or Linux.

Basic Tasks

Deploy from VA Marketplace

Create a new virtual machine

Explore Further

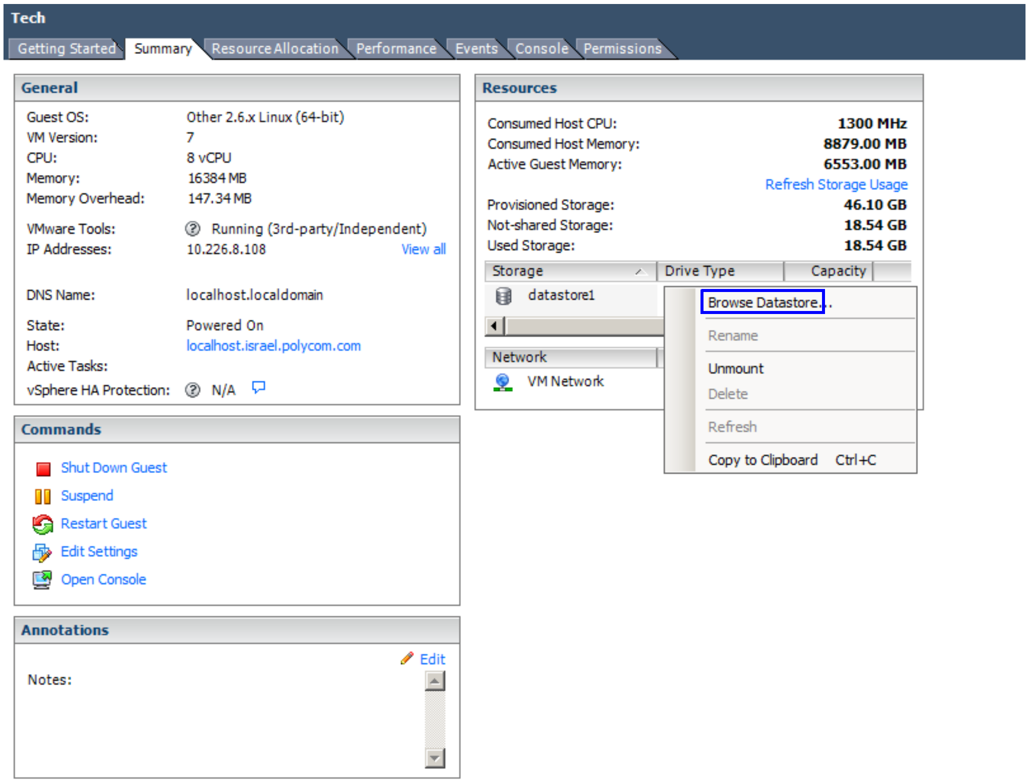
Learn about vSphere

Manage multiple hosts, eliminate downtime, load balance your datacenter with vMotion, and more

Evaluate vSphere

10 When the MCU has shut down, click the **Summary** tab.

11 Under *Resources*, right-click the datastore, and click **Browse Datastore**.



Guest OS:	Other 2.6.x Linux (64-bit)
VM Version:	7
CPU:	8 vCPU
Memory:	16384 MB
Memory Overhead:	147.34 MB
VMware Tools:	Running (3rd-party/Independent)
IP Addresses:	10.226.8.108 View all
DNS Name:	localhost.localdomain
State:	Powered On
Host:	localhost.israel.polycom.com
Active Tasks:	
vSphere HA Protection:	N/A

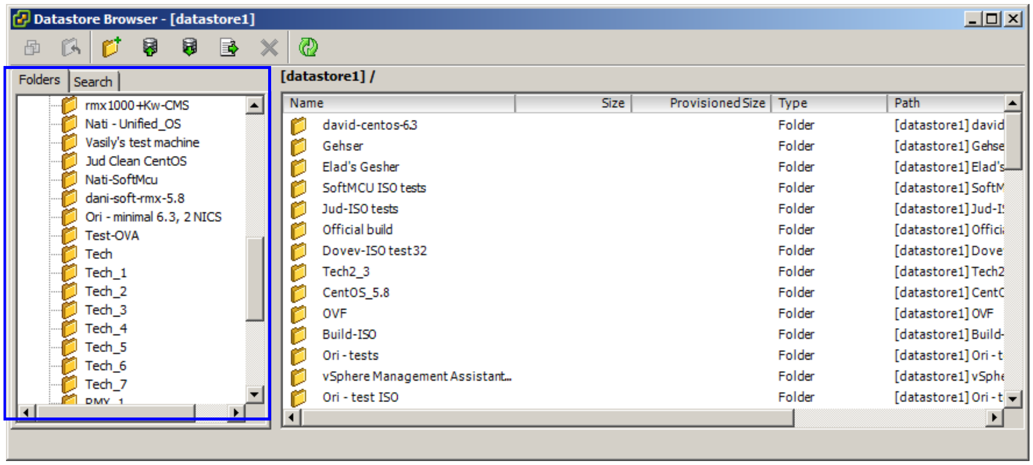
Consumed Host CPU:	1300 MHz
Consumed Host Memory:	8879.00 MB
Active Guest Memory:	6553.00 MB
	Refresh Storage Usage
Provisioned Storage:	46.10 GB
Not-shared Storage:	18.54 GB
Used Storage:	18.54 GB

Storage	Drive Type	Capacity
datastore1		

Network
VM Network

Notes:

The *Browse Datastore* window appears.



- rmx1000+Kw-CMS
- Nati - Unified_OS
- Vasily's test machine
- Jud Clean CentOS
- Nati-SoftMcu
- dani-soft-rmx-5.8
- Ori - minimal 6.3, 2 NICS
- Test-OVA
- Tech
- Tech_1
- Tech_2
- Tech_3
- Tech_4
- Tech_5
- Tech_6
- Tech_7
- DMX_1

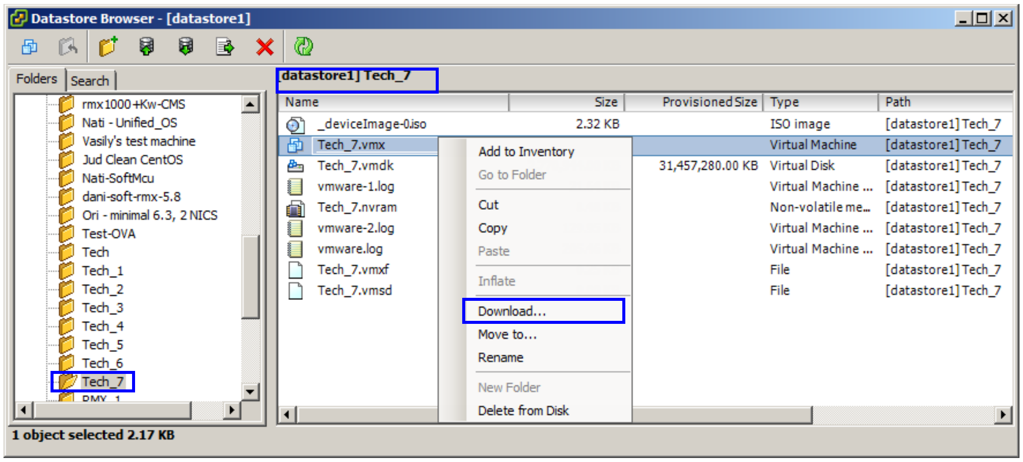
Name	Size	Provisioned Size	Type	Path
david-centos-6.3			Folder	[datastore1] david
Gehser			Folder	[datastore1] Gehse
Elad's Gesher			Folder	[datastore1] Elad's
SoftMCU ISO tests			Folder	[datastore1] SoftM
Jud-ISO tests			Folder	[datastore1] Jud-I:
Official build			Folder	[datastore1] Offici
Dovev-ISO test32			Folder	[datastore1] Dove
Tech2_3			Folder	[datastore1] Tech2
CentOS_5.8			Folder	[datastore1] CentC
OVF			Folder	[datastore1] OVF
Build-ISO			Folder	[datastore1] Build-
Ori - tests			Folder	[datastore1] Ori - t
vSphere Management Assistant...			Folder	[datastore1] vSphe
Ori - test ISO			Folder	[datastore1] Ori - t

12 In the **Folders** tab, select the folder whose name matches that of the MCU.

If the same name has been used multiple times, there will be multiple folders with an underscore and a number appended to the name. In such a case, select the folder with the name of the MCU which ends with the highest number.

The contents of the folder are displayed.

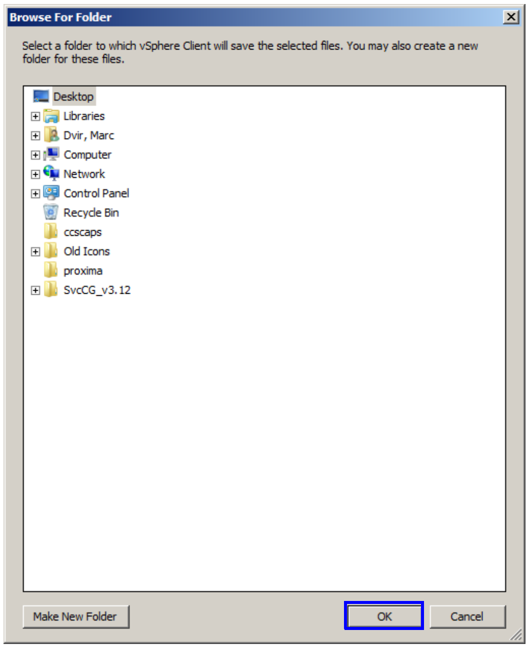
13 Right-click the file ending with “.vmx” and click **Download.**



Name	Size	Provisioned Size	Type	Path
_deviceImage-0.iso	2.32 KB		ISO image	[datastore1] Tech_7
Tech_7.vmx			Virtual Machine	[datastore1] Tech_7
Tech_7.vmdk		31,457,280.00 KB	Virtual Disk	[datastore1] Tech_7
vmware-1.log			Virtual Machine ...	[datastore1] Tech_7
Tech_7.nvram			Non-volatile me...	[datastore1] Tech_7
vmware-2.log			Virtual Machine ...	[datastore1] Tech_7
vmware.log			Virtual Machine ...	[datastore1] Tech_7
Tech_7.vmxf			File	[datastore1] Tech_7
Tech_7.vmsd			File	[datastore1] Tech_7

The *Browse For Folder* window appears.

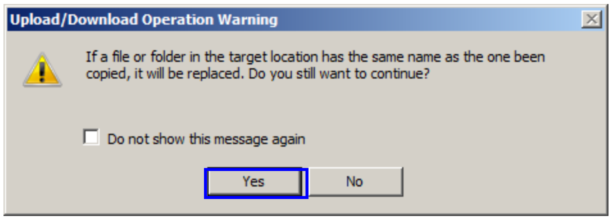
14 Browse to a location and click **OK**.



Select a folder to which vSphere Client will save the selected files. You may also create a new folder for these files.

The *Upload/Download Operation Warning* window may appear.

15 If the *Upload/Download Operation Warning* window appears, click **Yes**.



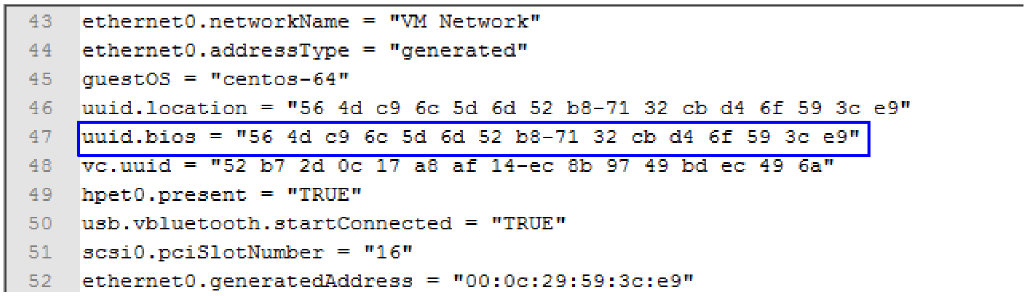
If a file or folder in the target location has the same name as the one been copied, it will be replaced. Do you still want to continue?

□ Do not show this message again

If it does not appear, proceed to Step 16.

The file downloads.

16 Open the file in any plain text editor.



```
ethernet0.networkName = "VM Network"
ethernet0.addressType = "generated"
guestOS = "centos-64"
uuid.location = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
uuid.bios = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
vc.uuid = "52 b7 2d 0c 17 a8 af 14-ec 8b 97 49 bd ec 49 6a"
hpet0.present = "TRUE"
usb.vbluetooth.startConnected = "TRUE"
scsi0.pciSlotNumber = "16"
ethernet0.generatedAddress = "00:0c:29:59:3c:e9"
```

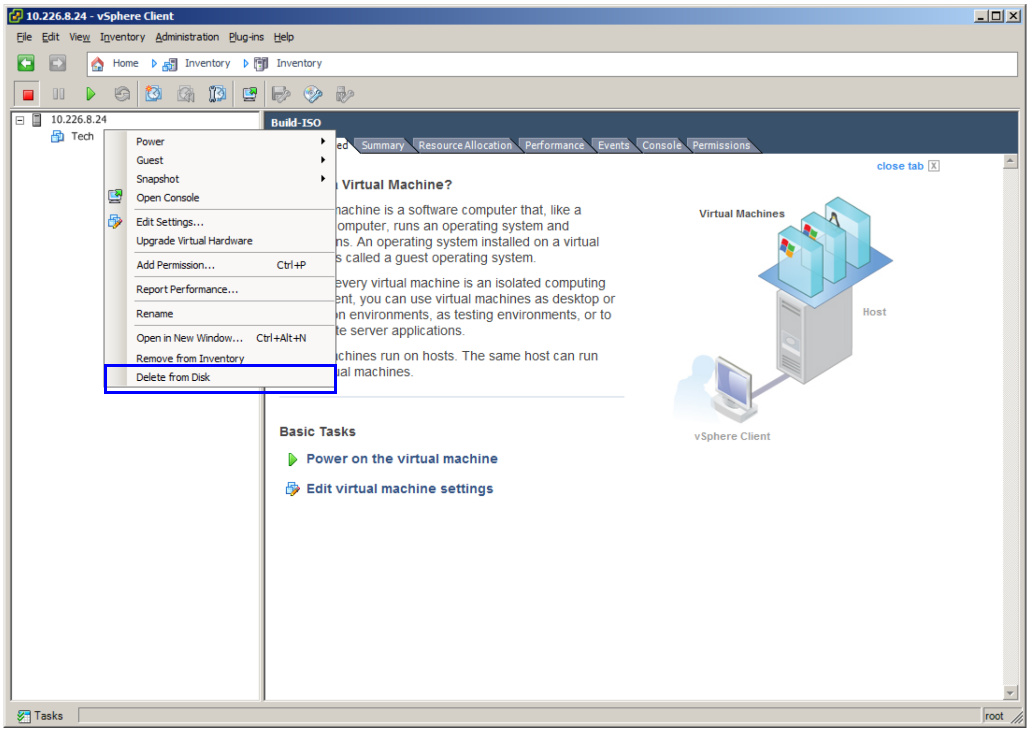
17 Locate the line that starts with *uuid.bios.*

18 **Copy** the entire line and paste it into another text file.

19 **Save** the text file.

20 In the *Inventory Panel*, click the Datastore that houses the MCU.

21 Right-click the MCU, and select **Delete from Disk.**



Virtual Machine?

achine is a software computer that, like a omputer, runs an operating system and ns. An operating system installed on a virtual s called a guest operating system.

very virtual machine is an isolated computing ent, you can use virtual machines as desktop or n environments, as testing environments, or to te server applications.

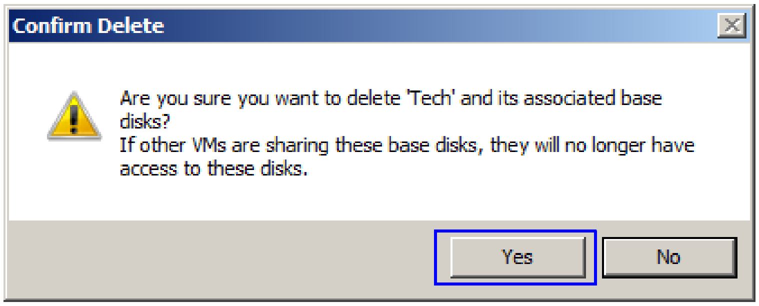
chines run on hosts. The same host can run al machines.

Basic Tasks

Power on the virtual machine

Edit virtual machine settings

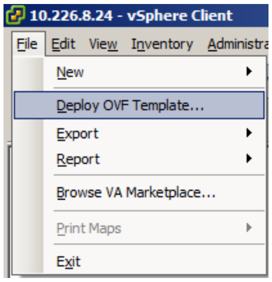
The **Confirm Delete** window appears.



Are you sure you want to delete 'Tech' and its associated base disks?
If other VMs are sharing these base disks, they will no longer have access to these disks.

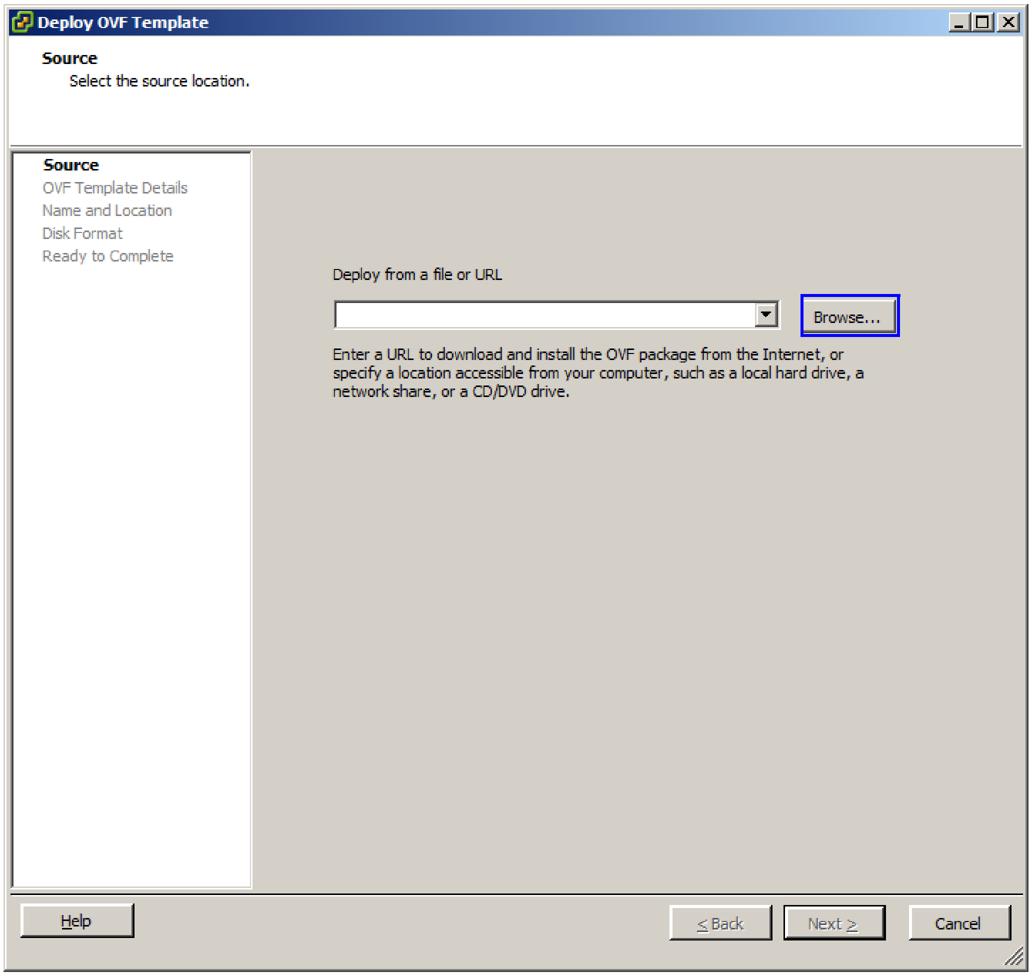
22 Click **Yes**.

23 On the *vSphere Client* menu, select **File > Deploy OVF Template.**



The *Deploy OVF Template* wizard opens to the *Source* page.

24 Click **Browse.**



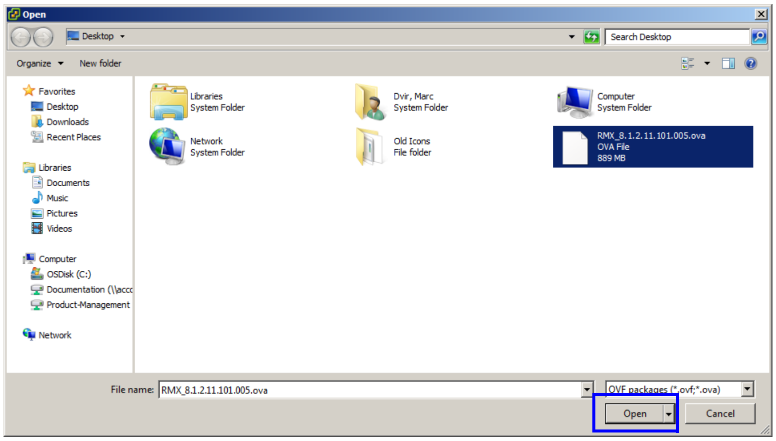
Select the source location.

Deploy from a file or URL

Enter a URL to download and install the OVF package from the Internet, or specify a location accessible from your computer, such as a local hard drive, a network share, or a CD/DVD drive.

The **Open** dialog box appears.

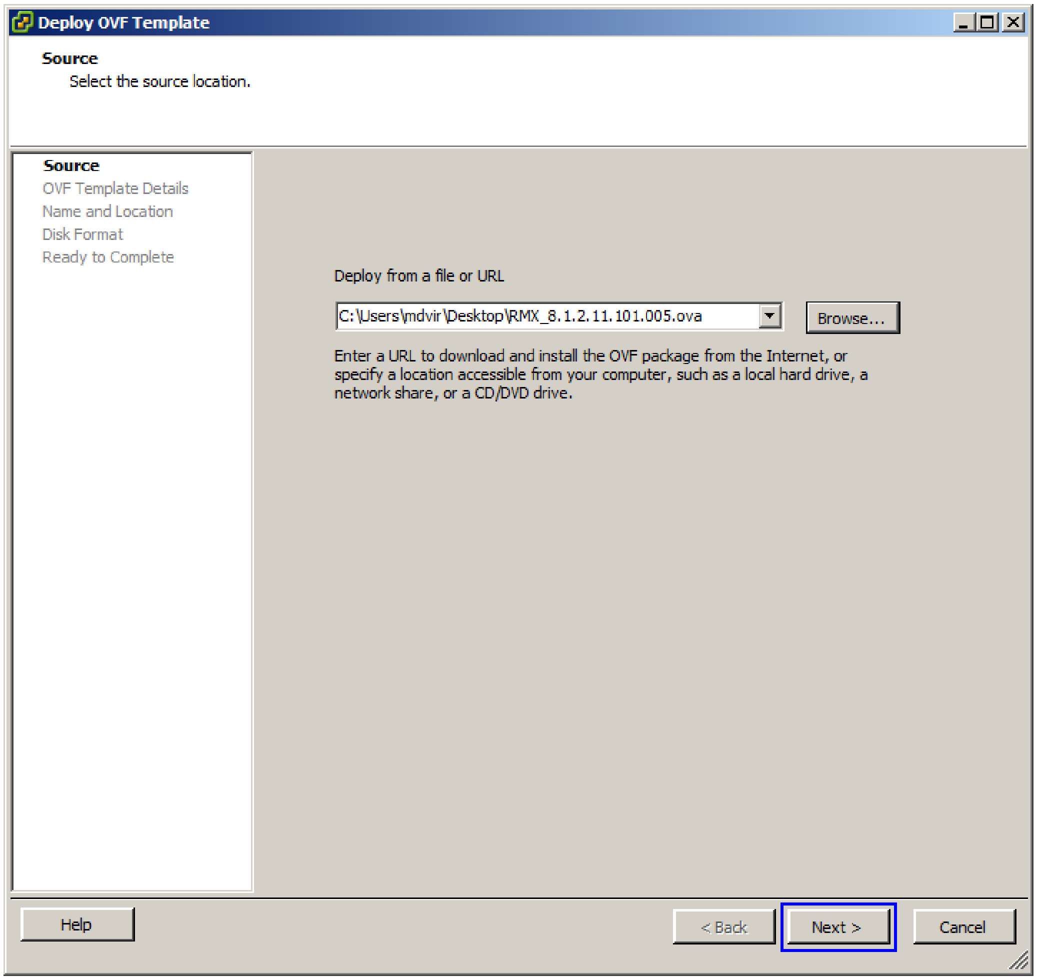
25 Browse to the new OVA file.



File name: RMX_8.1.2.11.101.005.ova

OVF packages (*.ovf;*.ova)

26 Either double-click on the OVA file or click on the file, then click **Open.**



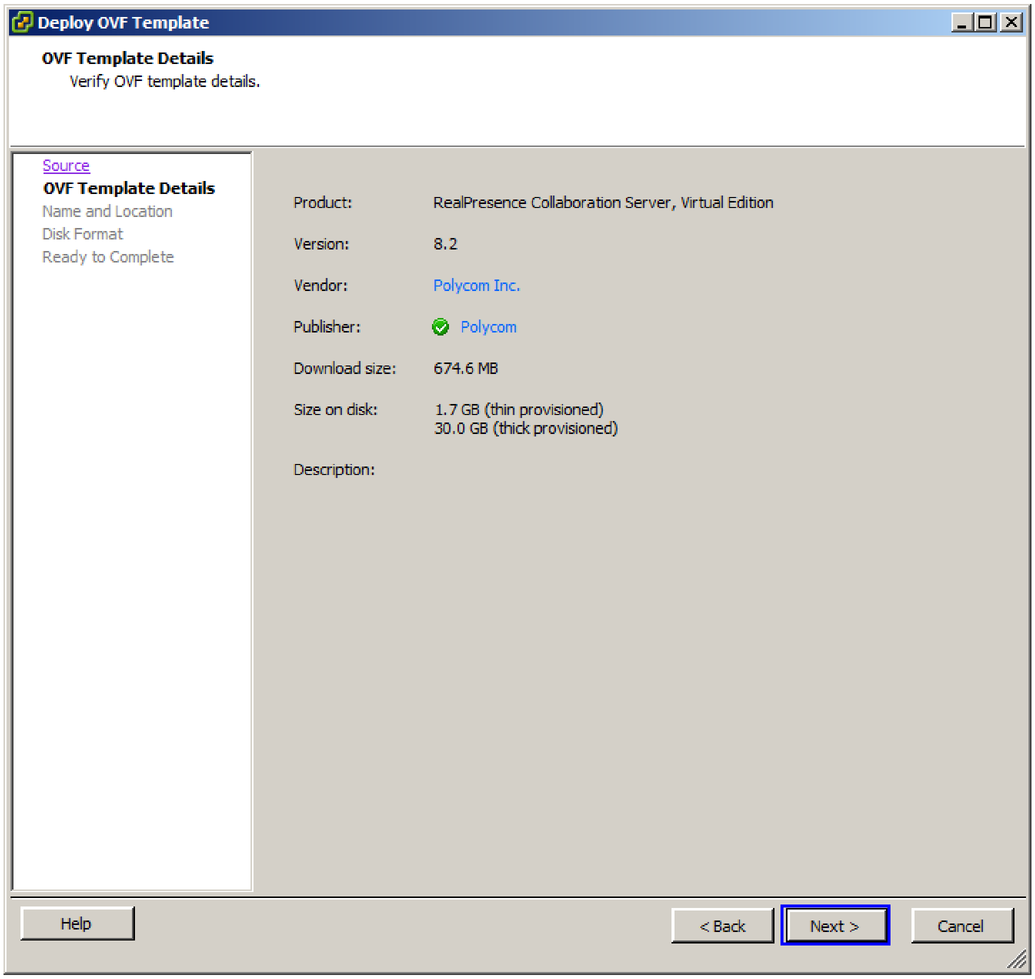
Deploy from a file or URL

C:\Users\mdvir\Desktop\RMX_8.1.2.11.101.005.ova

Enter a URL to download and install the OVF package from the Internet, or specify a location accessible from your computer, such as a local hard drive, a network share, or a CD/DVD drive.

27 Click **Next**.

The *OVF Template Details* page is displayed.



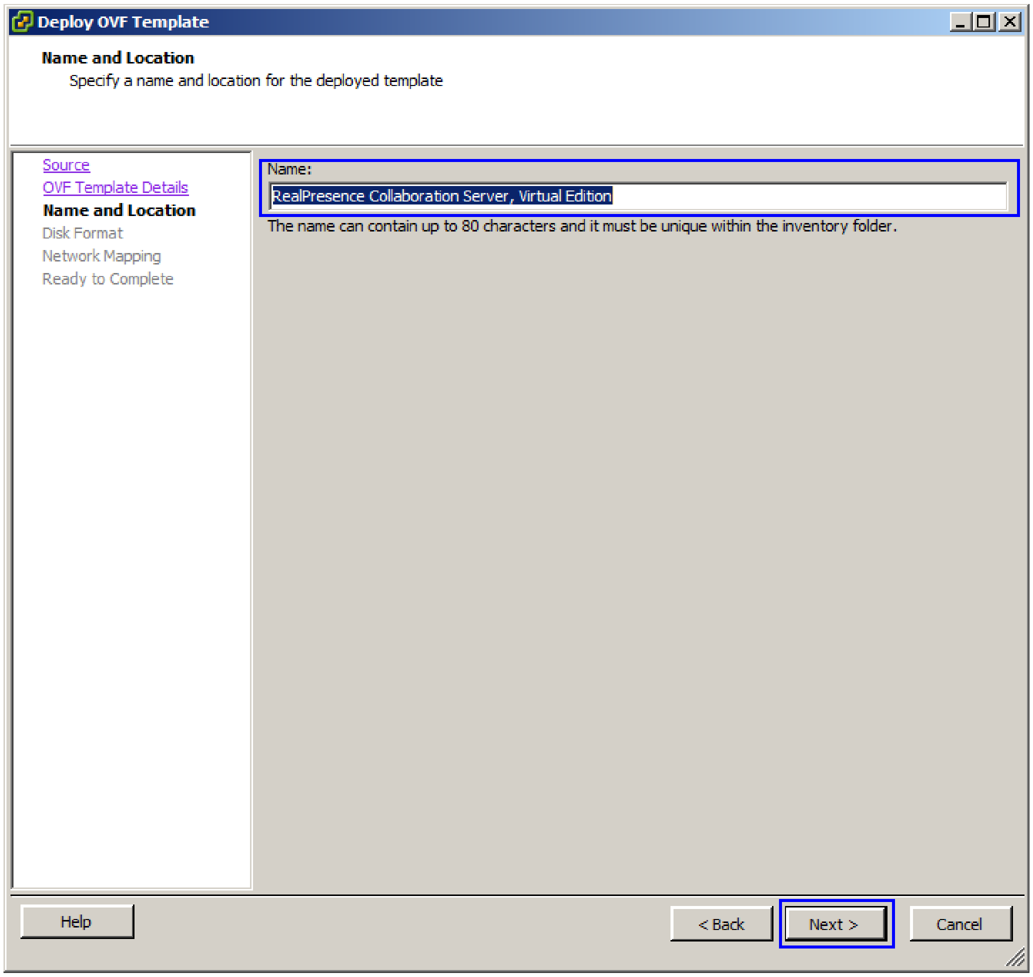
OVF Template Details
Verify OVF template details.

Product:	RealPresence Collaboration Server, Virtual Edition
Version:	8.2
Vendor:	Polycom Inc.
Publisher:	✓ Polycom
Download size:	674.6 MB
Size on disk:	1.7 GB (thin provisioned) 30.0 GB (thick provisioned)
Description:	

28 Click **Next**.

The *Name and Location* page is displayed.

29 In the *Name* field, type the same name previously used for the MCU.



Deploy OVF Template

Name and Location

Specify a name and location for the deployed template

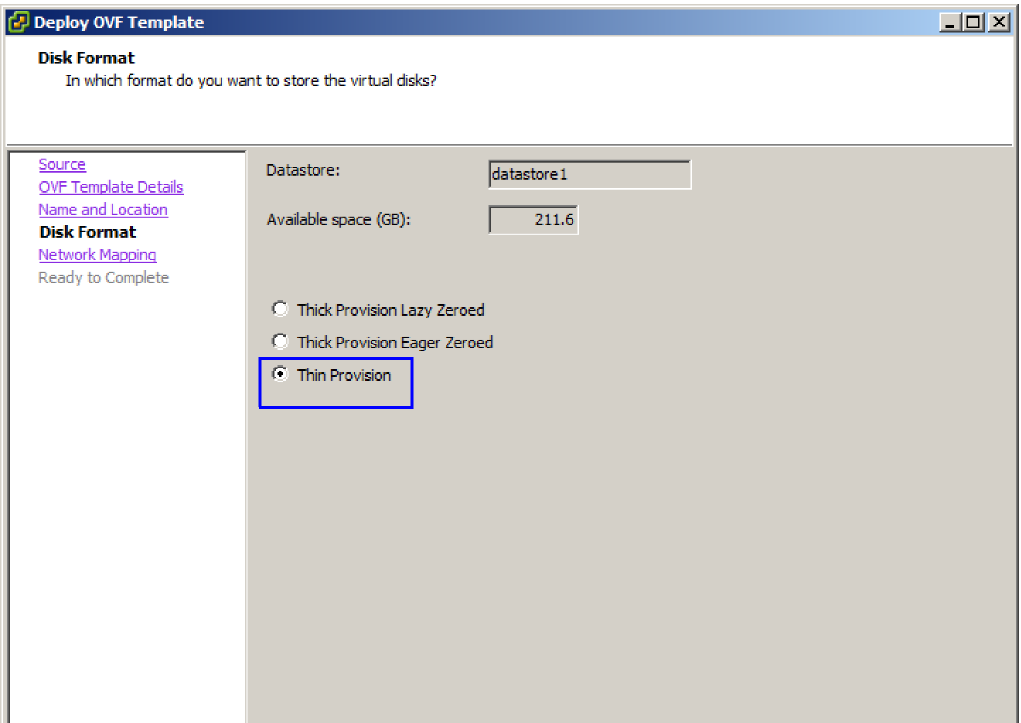
Name: RealPresence Collaboration Server, Virtual Edition

The name can contain up to 80 characters and it must be unique within the inventory folder.

30 Click **Next.**

The **Disk Format** page is displayed.

31 Select **Thin Provision**, then click **Next**.



Deploy OVF Template

Disk Format

In which format do you want to store the virtual disks?

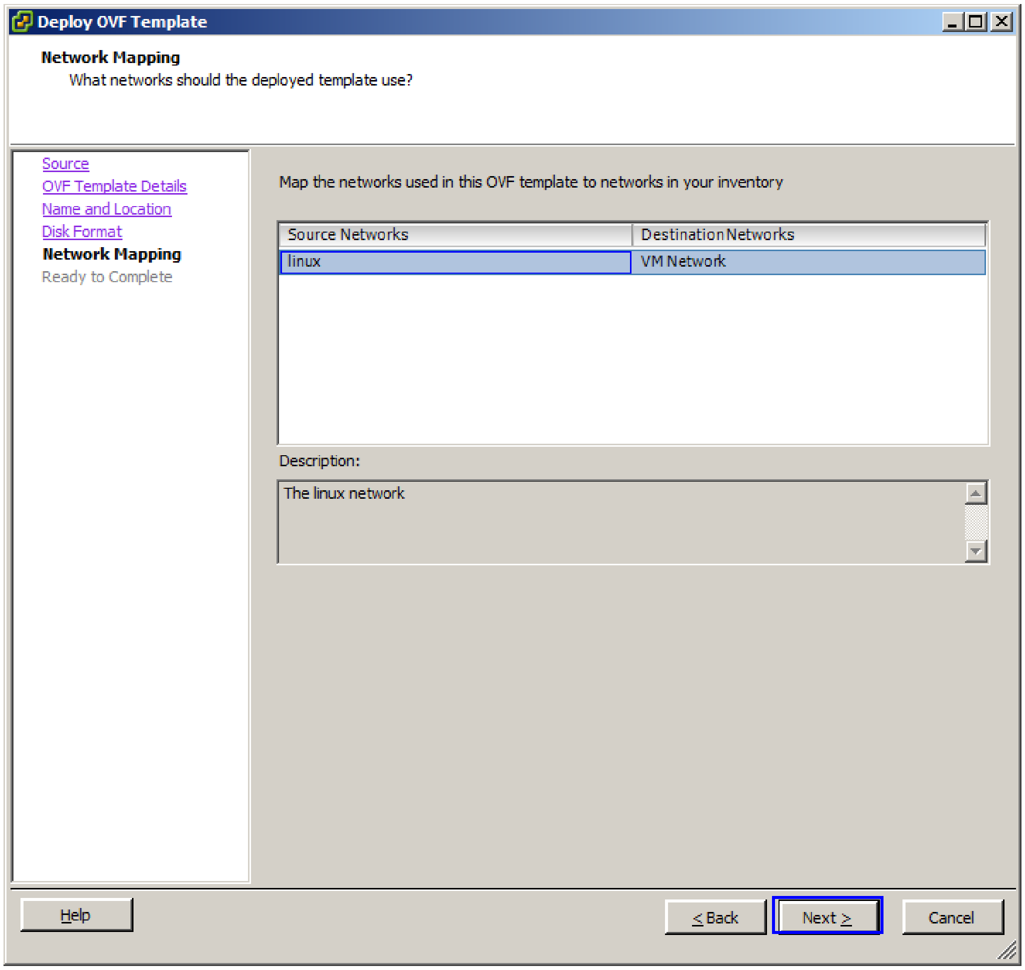
Datastore: datastore1

Available space (GB): 211.6

○ Thick Provision Lazy Zeroed
○ Thick Provision Eager Zeroed
● Thin Provision

The *Network Mapping* page is displayed.

32 Select the appropriate network mappings, then click **Next.**



Network Mapping

What networks should the deployed template use?

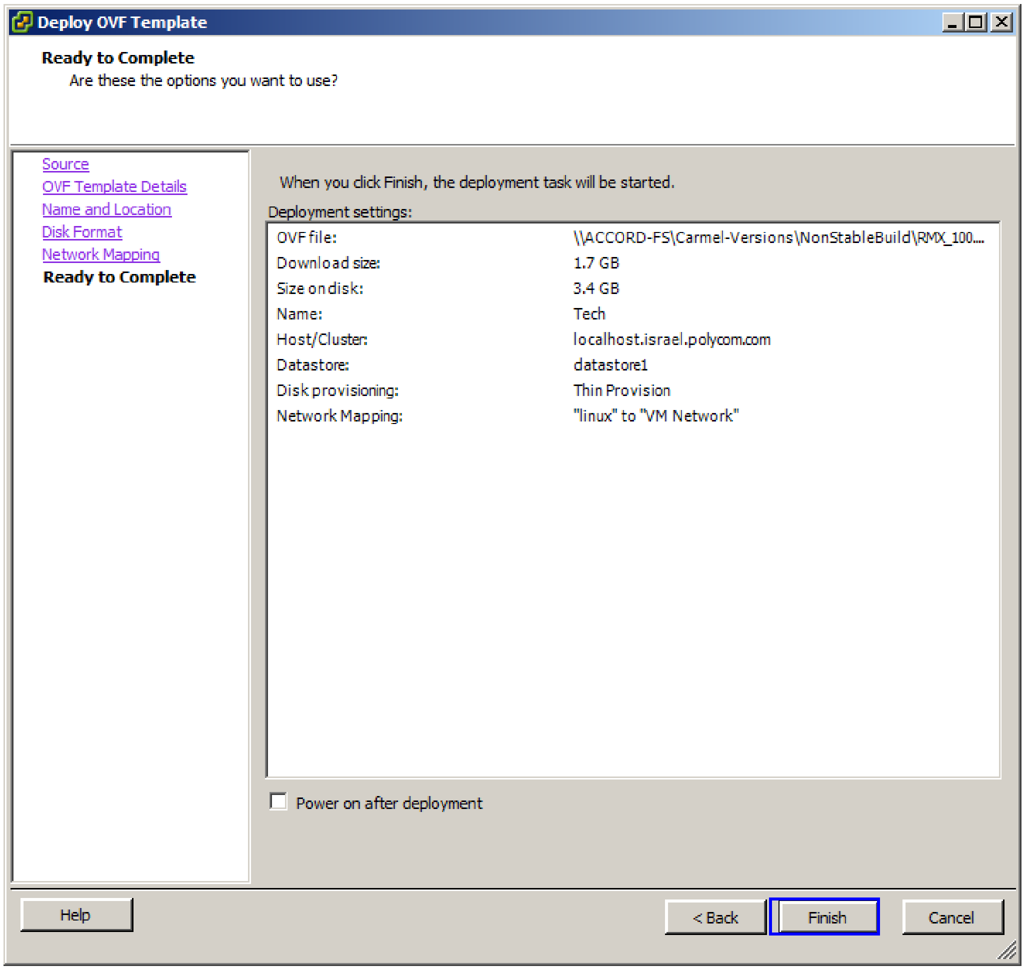
Map the networks used in this OVF template to networks in your inventory

Source Networks	Destination Networks
linux	VM Network

Description:

The linux network

The *Ready to Complete* page is displayed.



Deploy OVF Template

Ready to Complete

Are these the options you want to use?

When you click Finish, the deployment task will be started.

Deployment settings:

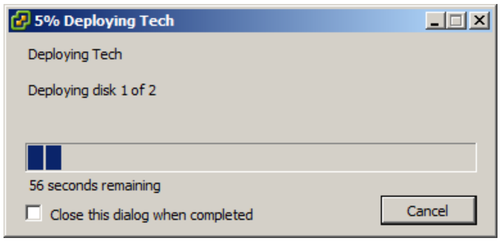
OVF file:	\\ACCORD-FS\Carmel-Versions\NonStableBuild\RMX_100....
Download size:	1.7 GB
Size on disk:	3.4 GB
Name:	Tech
Host/Cluster:	localhost.israel.polycom.com
Datastore:	datastore1
Disk provisioning:	Thin Provision
Network Mapping:	"linux" to "VM Network"

☐ Power on after deployment

33 Verify that **Power on after deployment** is not selected.

34 Confirm that all the settings are correct, then click **Finish.**

The *vSphere Client* deploys the OVF file.



5% Deploying Tech

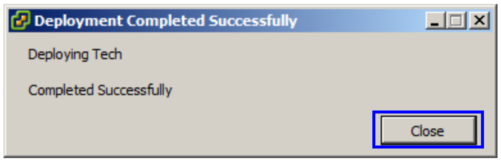
Deploying Tech

Deploying disk 1 of 2

56 seconds remaining

☐ Close this dialog when completed

When the deployment is complete the following window appears:



Deployment Completed Successfully

Deploying Tech

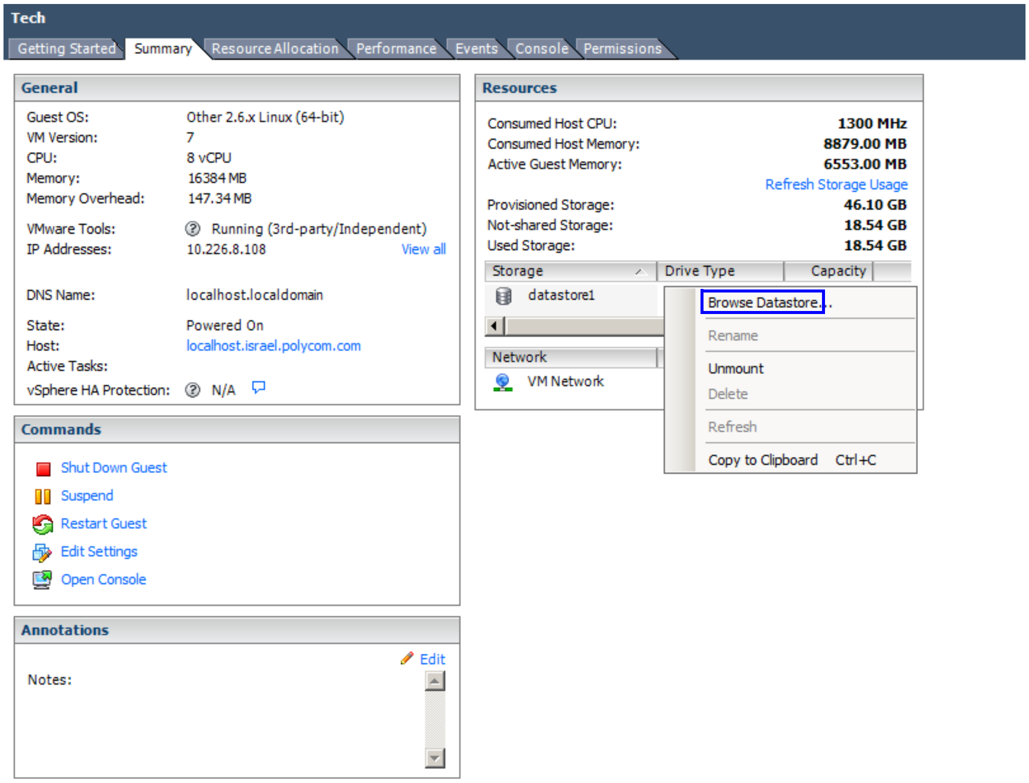
Completed Successfully

35 Click **Close.**

36 In the *Inventory Panel*, select the Datastore that used to house the MCU.

37 Click the **Summary** tab.

38 Under *Resources*, right-click the datastore, and click **Browse Datastore**.



General

Guest OS:	Other 2.6.x Linux (64-bit)
VM Version:	7
CPU:	8 vCPU
Memory:	16384 MB
Memory Overhead:	147.34 MB
VMware Tools:	Running (3rd-party/Independent)
IP Addresses:	10.226.8.108 View all
DNS Name:	localhost.localdomain
State:	Powered On
Host:	localhost.israel.polycom.com
Active Tasks:	
vSphere HA Protection:	N/A

Notes:

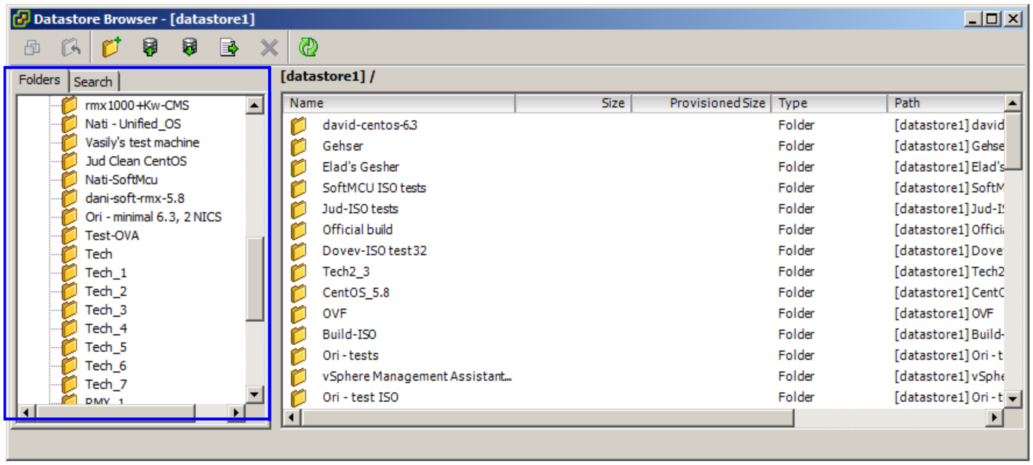
Resources

Consumed Host CPU:	1300 MHz
Consumed Host Memory:	8879.00 MB
Active Guest Memory:	6553.00 MB
	Refresh Storage Usage
Provisioned Storage:	46.10 GB
Not-shared Storage:	18.54 GB
Used Storage:	18.54 GB

Storage	Drive Type	Capacity
datastore1		

Network
VM Network

The *Browse Datastore* window appears.



rmx1000+Kw-CMS
Nati - Unified_OS
Vasily's test machine
Jud Clean CentOS
Nati-SoftMcu
dani-soft-rmx-5.8
Ori - minimal 6.3, 2 NICS
Test-OVA
Tech
Tech_1
Tech_2
Tech_3
Tech_4
Tech_5
Tech_6
Tech_7
RMX_1

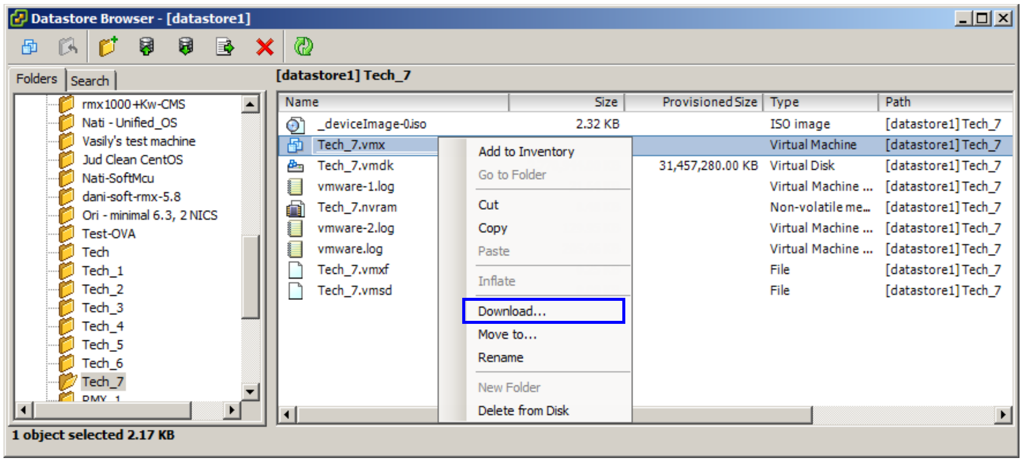
[datastore1] /

Name	Size	Provisioned Size	Type	Path
david-centos-6.3			Folder	[datastore1] david
Gehser			Folder	[datastore1] Gehse
Elad's Gesher			Folder	[datastore1] Elad's
SoftMCU ISO tests			Folder	[datastore1] SoftM
Jud-ISO tests			Folder	[datastore1] Jud-I
Official build			Folder	[datastore1] Offici
Dovev-ISO test32			Folder	[datastore1] Dove
Tech2_3			Folder	[datastore1] Tech2
CentOS_5.8			Folder	[datastore1] CentC
OVF			Folder	[datastore1] OVF
Build-ISO			Folder	[datastore1] Build-
Ori - tests			Folder	[datastore1] Ori - t
vSphere Management Assistant...			Folder	[datastore1] vSphe
Ori - test ISO			Folder	[datastore1] Ori - t

39 In the **Folders** tab, select the folder whose name matches that of the MCU. The folder name will have an underscore and a number at the end.

The contents of the folder are displayed.

40 Right-click the file ending with “.vmx” and click **Download.**

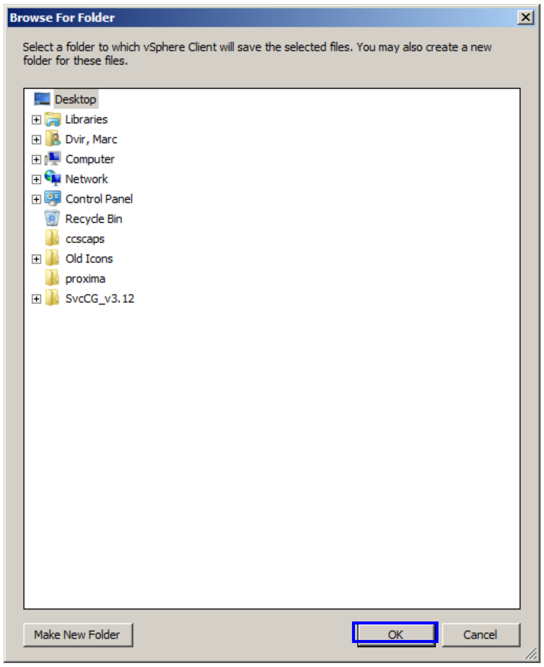


[datastore1] Tech_7

Name	Size	Provisioned Size	Type	Path
_deviceImage-0.iso	2.32 KB		ISO image	[datastore1] Tech_7
Tech_7.vmx			Virtual Machine	[datastore1] Tech_7
Tech_7.vmdk		31,457,280.00 KB	Virtual Disk	[datastore1] Tech_7
vmware-1.log			Virtual Machine ...	[datastore1] Tech_7
Tech_7.nvram			Non-volatile me...	[datastore1] Tech_7
vmware-2.log			Virtual Machine ...	[datastore1] Tech_7
vmware.log			Virtual Machine ...	[datastore1] Tech_7
Tech_7.vmxf			File	[datastore1] Tech_7
Tech_7.vmsd			File	[datastore1] Tech_7

1 object selected 2.17 KB

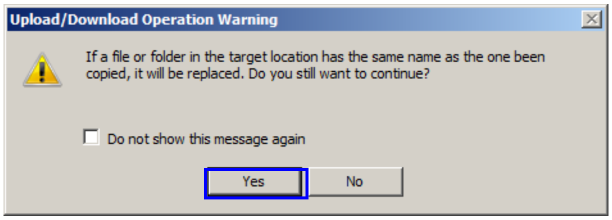
The *Browse For Folder* window appears.



Select a folder to which vSphere Client will save the selected files. You may also create a new folder for these files.

41 Browse to a location and click **OK**.

The *Upload/Download Operation Warning* window may appear.



If a file or folder in the target location has the same name as the one been copied, it will be replaced. Do you still want to continue?

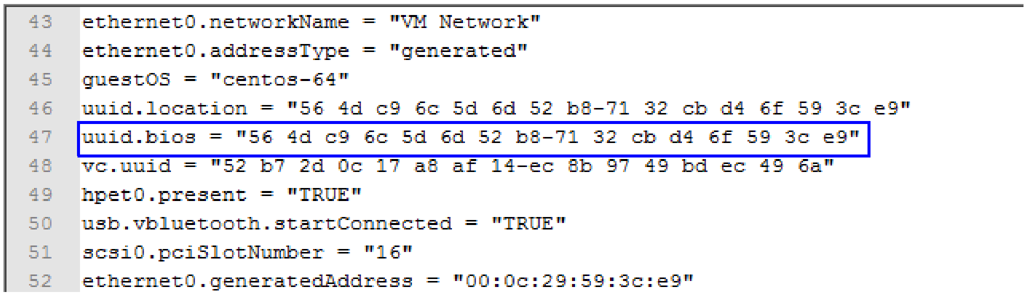
□ Do not show this message again

42 If the *Upload/Download Operation Warning* window appears, click **Yes**. If it does not appear, proceed to Step 43.

The file downloads.

43 Open the file created in Step 18.

44 Open the file in any plain text editor.

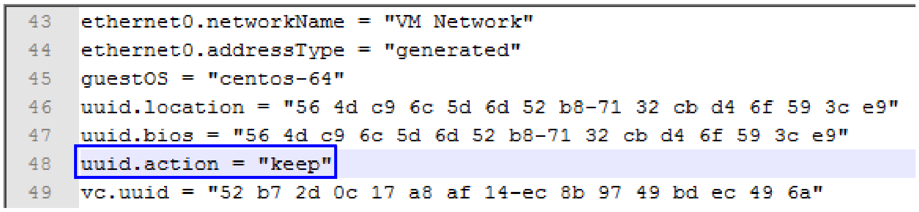


```
ethernet0.networkName = "VM Network"
ethernet0.addressType = "generated"
guestOS = "centos-64"
uuid.location = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
uuid.bios = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
vc.uuid = "52 b7 2d 0c 17 a8 af 14-ec 8b 97 49 bd ec 49 6a"
hpet0.present = "TRUE"
usb.vbluetooth.startConnected = "TRUE"
scsi0.pciSlotNumber = "16"
ethernet0.generatedAddress = "00:0c:29:59:3c:e9"
```

45 Locate the line that starts with, *uuid.bios*.

46 Replace that line with the line saved in the other text file.

47 Add the following as a separate line to the file, including the quotation marks: *uuid.action = "keep"*

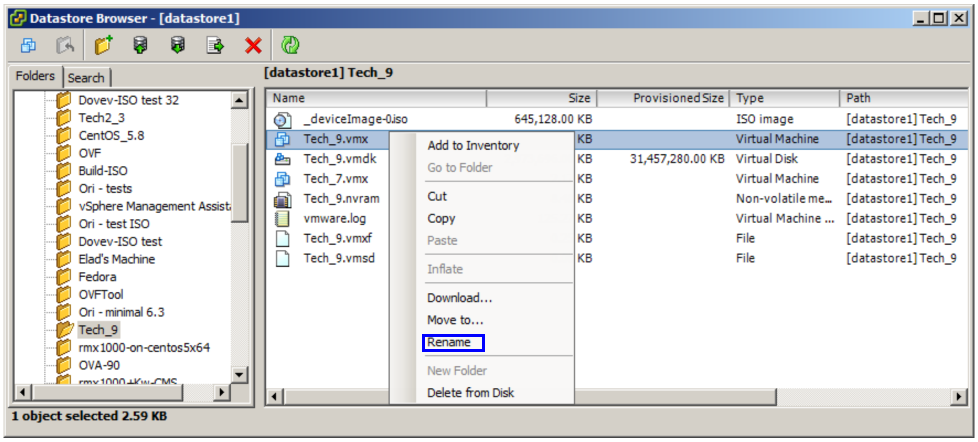


```
ethernet0.networkName = "VM Network"
ethernet0.addressType = "generated"
guestOS = "centos-64"
uuid.location = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
uuid.bios = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
uuid.action = "keep"
vc.uuid = "52 b7 2d 0c 17 a8 af 14-ec 8b 97 49 bd ec 49 6a"
```

48 Save and close the file.

49 **Optional**. To back up the previous configuration:

a In the *Datastore Browser* window, right-click the .vmx file, then select **Rename**.



Datastore Browser - [datastore1]

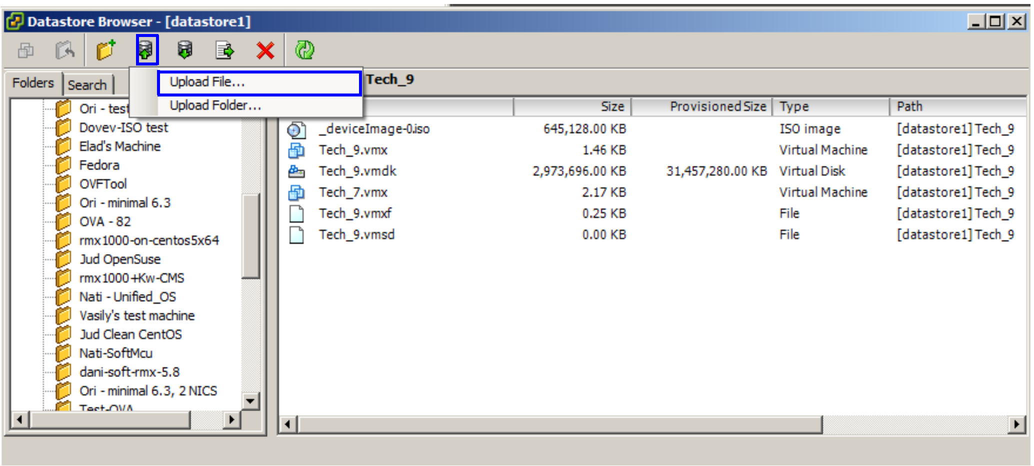
[datastore1] Tech_9

Name	Size	Provisioned Size	Type	Path
_deviceImage-0.iso	645,128.00 KB		ISO image	[datastore1] Tech_9
Tech_9.vmx	KB		Virtual Machine	[datastore1] Tech_9
Tech_9.vmdk	KB	31,457,280.00 KB	Virtual Disk	[datastore1] Tech_9
Tech_7.vmx	KB		Virtual Machine	[datastore1] Tech_9
Tech_9.nvram	KB		Non-volatile me...	[datastore1] Tech_9
vmware.log	KB		Virtual Machine ...	[datastore1] Tech_9
Tech_9.vmxf	KB		File	[datastore1] Tech_9
Tech_9.vmsd	KB		File	[datastore1] Tech_9

1 object selected 2.59 KB

b Change the file extension to ".bak".

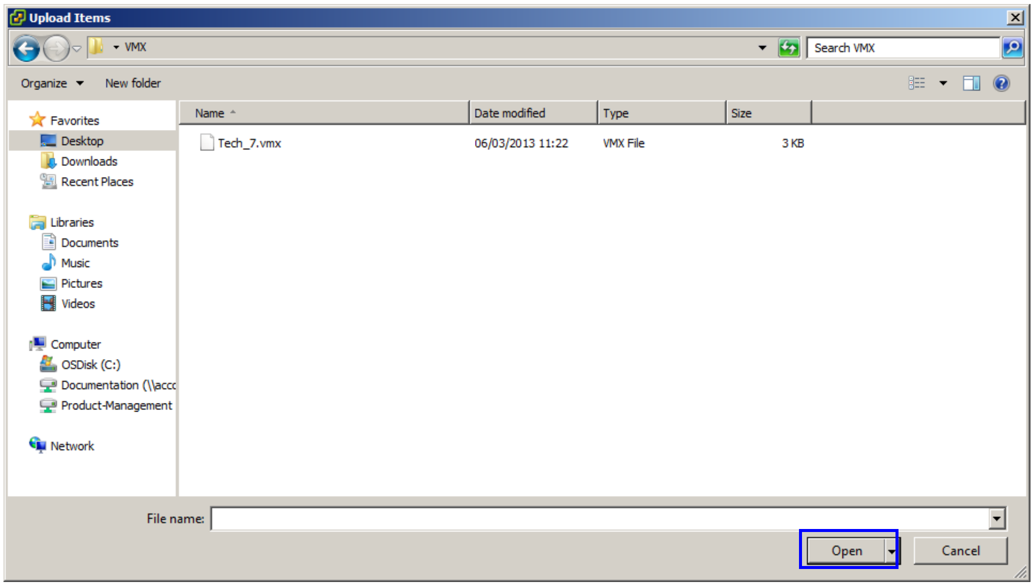
50 In the tool bar of the *Datastore Browser* window, click the **Upload files to this datastore** button.



	Size	Provisioned Size	Type	Path
_deviceImage-0.iso	645,128.00 KB		ISO image	[datastore1] Tech_9
Tech_9.vmx	1.46 KB		Virtual Machine	[datastore1] Tech_9
Tech_9.vmdk	2,973,696.00 KB	31,457,280.00 KB	Virtual Disk	[datastore1] Tech_9
Tech_7.vmx	2.17 KB		Virtual Machine	[datastore1] Tech_9
Tech_9.vmxf	0.25 KB		File	[datastore1] Tech_9
Tech_9.vmsd	0.00 KB		File	[datastore1] Tech_9

51 Click **Upload File.**

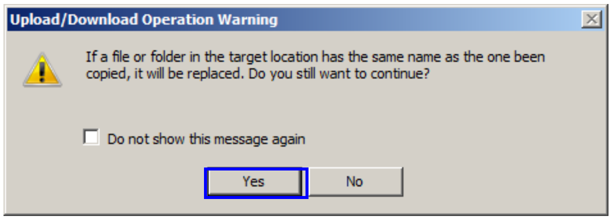
The *Upload Items* window appears.



Name	Date modified	Type	Size
Tech_7.vmx	06/03/2013 11:22	VMX File	3 KB

52 Navigate to where you saved the ".vmx" file in Step 48, select it, then click **Open**.

The *Upload/Download Operation Warning* window may appear.



If a file or folder in the target location has the same name as the one been copied, it will be replaced. Do you still want to continue?

□ Do not show this message again

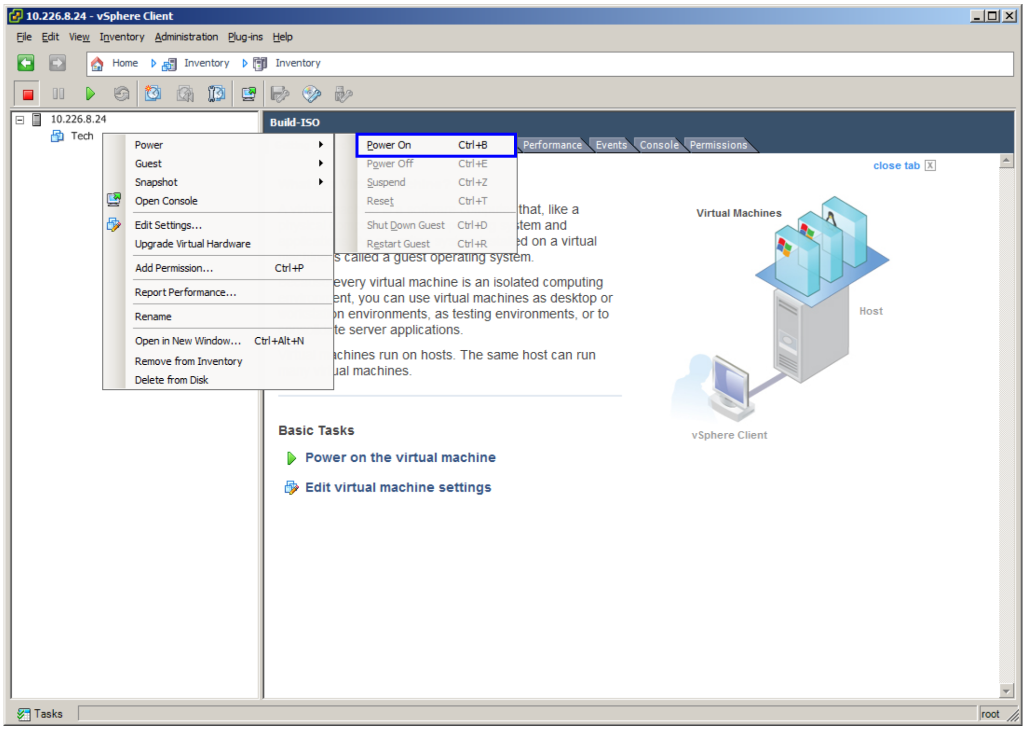
53 If the *Upload/Download Operation Warning* window appears, click **Yes**.

The file is uploaded.

54 Close the *Datastore Browser*.

55 In the *Inventory Panel*, click the Datastore that houses the MCU.

56 Right-click the MCU virtual machine, then click **Power > Power On**.



that, like a
tem and
d on a virtual
s called a guest operating system.

every virtual machine is an isolated computing
ent, you can use virtual machines as desktop or
n environments, as testing environments, or to
te server applications.

chines run on hosts. The same host can run
ual machines.

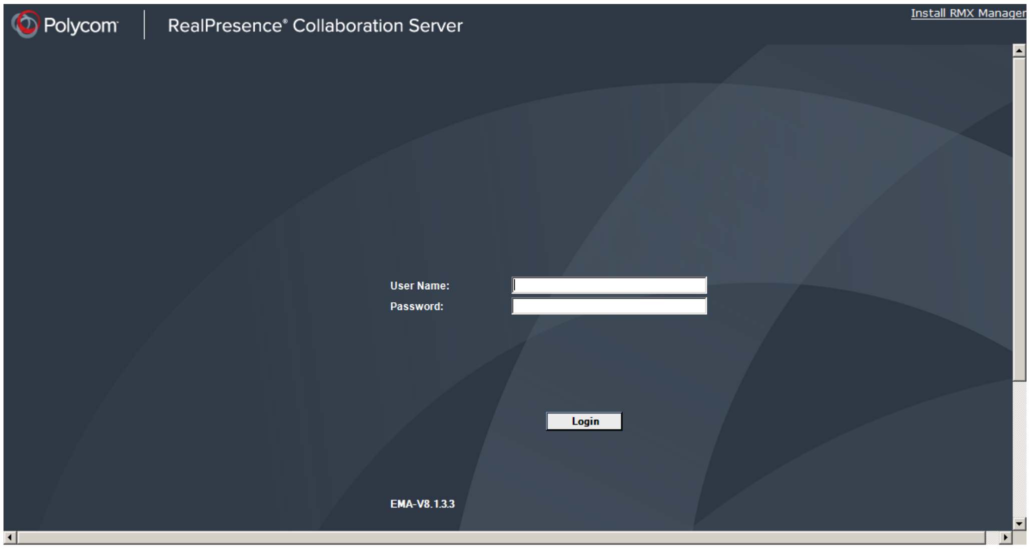
After a few minutes, the MCU turns on.

57 Start the *Collaboration Server Web Client* application on the workstation.

a In the browser's address line, enter the IP address of the *Control Unit* in the format: **`http://<Control Unit IP Address>:8080`**.

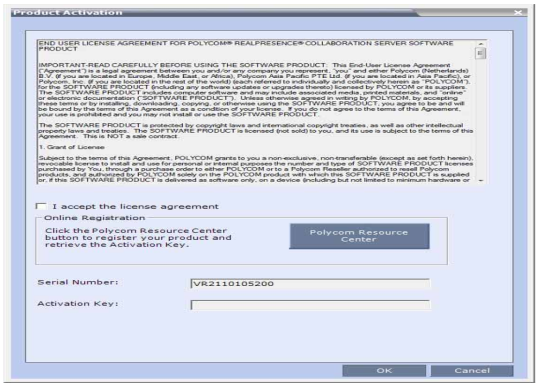
b Click **Enter**.

The *Collaboration Server Web Client* Login screen is displayed.



58 In the *Collaboration Server Web Client* Login screen, enter the default *Username* (**`POLYCOM`**) and *Password* (**`POLYCOM`**) and click **Login**.

The *Collaboration Server Web Client* opens and the *Product Activation* dialog box appears with the serial number filled in:



Product Activation

END USER LICENSE AGREEMENT FOR POLYCOM® REALPRESENCE® COLLABORATION SERVER SOFTWARE PRODUCT

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☐ I accept the license agreement

Online Registration

Click the Polycom Resource Center button to register your product and retrieve the Activation Key.

Polycom Resource Center

Serial Number: VR2110105200

Activation Key:

OK Cancel

59 In the *Activation Key* field, enter or **paste** the *Product Activation Key* that was used on the previous MCU.

60 Click **OK.**
A message indicating that the *Product Activation Key* was loaded successfully appears.
If the *Product Activation Key* fails to load, please contact your vendor.

61 Click **OK**.

> If the *Product Activation* dialog box does not appear, go to **Setup --> Product Activation** to display the dialog box.

62 On the *RealPresence Collaboration Server* menu, click **Administration** > **Software Management** > **Restore Configuration**.

63 **Browse** to the *Restore Directory Path* where the backed up configuration files are stored and then click **Restore**.

20

System Configuration Flags

The system's overall behavior can be configured by modifying the default values of the System Flags.

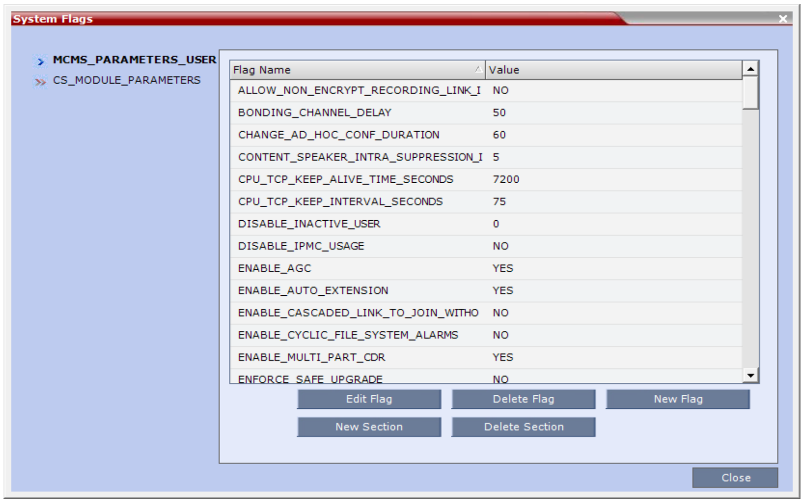
> For flag changes (including deletion) to take effect, the MCU must be reset. For more information, see 2.
>
> The following *System Flags* do not require an MCU reset:
>
> - IVR_MESSAGE_VOLUME
> - IVR_MUSIC_VOLUME
> - IVR_ROLL_CALL_VOLUME
> - ENABLE_SELECTIVE_MIXING

Modifying System Flags

To modify system flags:

1 On the *Collaboration Server* menu, click **Setup > System Configuration**.

The *System Flags* dialog box opens.



Flag Name	Value
ALLOW_NON_ENCRYPT_RECORDING_LINK_I	NO
BONDING_CHANNEL_DELAY	50
CHANGE_AD_HOC_CONF_DURATION	60
CONTENT_SPEAKER_INTRA_SUPPRESSION_I	5
CPU_TCP_KEEP_ALIVE_TIME_SECONDS	7200
CPU_TCP_KEEP_INTERVAL_SECONDS	75
DISABLE_INACTIVE_USER	0
DISABLE_IPMC_USAGE	NO
ENABLE_AGC	YES
ENABLE_AUTO_EXTENSION	YES
ENABLE_CASCADED_LINK_TO_JOIN_WITHO	NO
ENABLE_CYCLIC_FILE_SYSTEM_ALARMS	NO
ENABLE_MULTI_PART_CDR	YES
ENFORCE_SAFE_UPGRADE	NO

2 In the *MCMS_PARAMETERS* tab, the following flags can be modified:

Table 20-1 *System Flags – MCMS_PARAMETERS*

Flag	Description
ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF	If YES, allows non-encrypted participants to connect to encrypted conferences. Default: No

Table 20-1 *System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
ALLOW_NON_ENCRYPT_RECORDING_LINK_IN_ENCRYPT_CONF	When set to **NO** (default), the Recording Link inherits the encryption settings of the conference. If the conference is encrypted, the recording link will be encrypted. When set to **YES**, it disables the encryption of the recording link, regardless of the Encryption settings of the conference and RSS recorder.
BONDING_CHANNEL_DELAY (ISDN)	When connecting a bonding group, this is the delay (number of 1/100 seconds) between dialing attempts to connect sequential channels. The channel per second connection performance of ISDN switches can vary and can cause timing issues that result in bonding channel disconnection. Default: 6
CHANGE_AD_HOC_CONF_DURATION	The duration of an ad-hoc conference* can be configured on a system level by setting the flag to one of the following values (in minutes): **60** (default), **90**, **180** and **270**. * An ad-hoc conference is automatically created when the participant dials into an Ad-hoc Entry Queue and enters a conference ID that is not being used by any other conferencing entity. It is based on the Conference Profile assigned to the EQ.
CONTENT_SLAVE_LINKS_INTRA_SUPPRESSION_IN_SECONDS	Defines the interval, in seconds, during which the Collaboration Server is allowed to forward an *Intra Request* received from any of the *Slave Cascading Links.* The *Slave Cascading Link* can be connected to the local Collaboration Server, to an MCU on a higher cascade level or to the *Content* sharer. The first *Intra* request that is received from any of the *Slave MCUs* connected to the Collaboration Server starts the interval counter and is forwarded to the next level *MCU* or to the *Content* sharer. All other *Intra* requests that are received within this interval are registered but ignored. After an interval of <flag value> seconds, the system checks if during the last interval any additional *Intra* requests were registered. If there is at least one *Intra* request it will be forwarded. If there is no additional *Intra* request not no action is taken other than to wait for the next cycle. This filtering process is repeated every <flag value> seconds. **Default:** 30

Table 20-1 *System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
CONTENT_SPEAKER_INTRA_SUPPRESSION_IN_SECONDS	This flag controls the requests to refresh (intra) the content sent from the Collaboration Server system to the content sender as a result of refresh requests initiated by other conference participants. Enter the interval in seconds between the Intra requests sent from the Collaboration Server to the endpoint sending the content to refresh the content display. Refresh requests that will be received from endpoints within the defined interval will be postponed to the next interval. Default setting: 5
CPU_TCP_KEEP_ALIVE_TIME_SECONDS	This flag indicates when to send the first KeepAlive indication to check the TCP connection. Default value: **7200** second (120 minutes) Range: 600-18000 seconds When there are NAT problems, this default may be too long and the TCP connection is lost. In such a case, the default value should be changed to 3600 seconds (60 minutes) or less.
CPU_TCP_KEEP_INTERVAL_SECONDS	This flag indicates the interval in seconds between the KeepAlive requests. Default value: **75** second Range: 10-720 seconds.
DISABLE_INACTIVE_USER	Users can be automatically disabled by the system when they do not log into the Collaboration Server application for a predefined period. Possible Values: **0** - **90** days. Default: **0** (disables this option).
ENABLE_ACCEPTING_ICMP_REDIRECT	When set to YES, allows the RMX to accept *ICMP Redirect Messages (ICMP* message type #5). Possible values: YES / NO • Default: YES
ENABLE_AGC	Set this flag to YES to enable the AGC option. (Default setting is NO.) When disabled, selecting the AGC option in the *Participant Properties* has not effect on the participant audio. For more information see "*Managing the Address Book"* on page 8-7. The Auto Gain Control mechanism regulates noise and audio volume by keeping the received audio signals of all participants balanced. **Note:** Enabling AGC may result in amplification of background noise.
ENABLE_CASCADED_LINK_TO_JOIN_WITHOUT_PASSWORD	Enables a cascaded link to enter a conference without a password. Default: NO, for security reasons.

Table 20-1 System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
ENABLE_CYCLIC_FILE_SYSTEM_ALARMS	Enables or disables the display of Active Alarms before overwriting the older CDR/Auditor/Log files, enabling the users to backup the older files before they are deleted. Default: NO
ENFORCE_SAFE_UPGRADE	When set to YES this flag enables the Collaboration Server system to notify users when an incorrect version upgrade/downgrade or upgrade/downgrade path is selected. When set to NO, after initiating an upgrade or downgrade software installation, the Collaboration Server activates a fault alert in the Faults List: "Warning: Upgrade started and SAFE Upgrade protection is turned OFF" and the upgrade/downgrade process continues. Range: YES / NO Default: YES
ENABLE_SENDING_ICMP_DESTINATION_UNREACHABLE	Not supported with RealPresence Collaboration Server Virtual Edition.
EXT_DB_IVR_PROV_TIME_SECONDS	When an Entry Queue is set as IVR Service Provider for the RealPresence DMA system, the value here indicates the time interval in seconds in which the database is accesses for the ID. Default: 300
FORCE_CIF_PORT_ALLOCATION	Sets the MCU to allocate one CIF video resource to an endpoint, regardless of the resolution determined by the Conference Profile parameters. You can specify the endpoint types for which resource allocation can be forced to CIF resource, enabling other types of endpoints to use higher resolutions in the same conference. Enter the product type to which the CIF resource should be allocated. Possible values are: • **CMA Desktop -** for CMA desktop client • **VSX nnnn** - where nnnn represents the model number for example, VSX 8000.
FORCE_STRONG_PASSWORD_POLICY	When set to YES , implements the Strong Password rules. For more details, see "*Changing a User's Password"* on page **14-3**. Default: NO
FORCE_SYSTEM_BROADCAST_VOLUME	If set to YES, the level of broadcasting volume of the connected participant is value taken from the system flag SYSTEM_BROADCAST_VOLUME. If set to NO (default), the broadcasting volume level is 5.
FORCE_SYSTEM_LISTENING_VOLUME	If set to YES, the level of listening volume of the connected participant is value taken from the system flag SYSTEM_LISTENING_VOLUME. If set to NO (default), the listening volume level is 5.

Table 20-1 *System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
GK_MANDATORY_FOR_CALLS_IN	If set to **YES**, a gatekeeper is required to receive incoming H.323 calls. If a gatekeeper is not configure in the Collaboration Server, the calls will fail. If set to **NO** (default), gatekeeper is not required to process H.323 incoming calls and H.323 participants can dial in with or without a gatekeeper.
GK_MANDATORY_FOR_CALLS_OUT	If set to **YES**, a gatekeeper is required to perform H.323 outgoing calls. If a gatekeeper is not configure on the Collaboration Server, the calls will fail. If set to **NO** (default), gatekeeper is not required to dial out to H.323 participants and calls can be dialed out with or without a gatekeeper.
H263_ANNEX_T	Set to NO to send the content stream without Annex T and enable Aethra and Tandberg endpoints, that do not support Annex T, to process the content. Default: YES
HD_THRESHOLD_BITRATE	Sets the minimum bit rate required by endpoints to connect to an HD Conference. Endpoints that cannot support this bit rate are connected as audio only. Range: 384kbps - 4Mbs (Default: 768)
IVR_MESSAGE_VOLUME	The volume of IVR messages varies according to the value of this flag. Possible value range: 0-10 (Default: 6). 0 – disables playing the IVR messages 1 – lowest volume 10 – highest volume **Notes:** • It is not recommended to disable IVR messages by setting the flag value to 0. • System reset is not required for flag changes to take effect.
IVR_MUSIC_VOLUME	The volume of the IVR music played when a single participant is connected to the conference varies according to the value of this flag. Possible value range: 0-10 (Default: 5). 0 – disables playing the music 1 – lowest volume 10 – highest volume **Note:** System reset is not required for flag changes to take effect.

Table 20-1 *System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
IVR_ROLL_CALL_VOLUME	The volume of the Roll Call varies according to the value of this flag. Possible value range: 0-10 (Default: 6). 0 – disables playing the Roll Call 1 – lowest volume 10 – highest volume **Note:** • It is not recommended to disable the Roll Call by setting the flag value to 0. • System reset is not required for flag changes to take effect.
LAST_LOGIN_ATTEMPTS	If YES, the system displays a record of the last Login of the user. Default: NO. .
LEGACY_EP_CONTENT_DEFAULT_LAYOUT	Defines the video layout to be displayed on the screen of the legacy endpoints when switching to Content mode. Default value: **CP_LAYOUT_1P7** (1+7).
MAX_CONF_PASSWORD_REPEATED_CHAR	Allows the administrator to configure the maximum number of consecutive repeating characters that are to be allowed in a conference password. Range: 1 - 4 Default: 2
MAX_CP_RESOLUTION	The MAX_CP_RESOLUTION flag value is applied to the system during *First Time Power-on* and after a system upgrade. The default value is HD720. All subsequent changes to the Maximum CP Resolution of the system are made using the *Resolution Configuration* dialog box. Possible flag values: • MPM+ / **HD1080** - High Definition at 60 fps • **HD720** – High Definition at 60 fps • **HD** – High Definition at 30 fps • **SD30** – Standard Definition at 30 fps • **SD15** – Standard Definition at 15 fps • **CIF** – CIF resolution Default: HD1080 For more information see "*Video Resolutions in AVC-based CP Conferencing"* on page **4-1**. **Note:** From Version 8.1, MPM+ media card is not supported.

Table 20-1 *System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
*MAX_INTRA_REQUESTS_PER_INTERVAL_*	Enter the maximum number of refresh (intra) requests for the Content channel sent by the participant's endpoint in a 10 seconds interval that will be dealt by the Collaboration Server system. When this number is exceeded, the Content sent by this participant will be identified as noisy and his/her requests to refresh the Content display will be suspended. Default setting: 3
*MAX_INTRA_SUPPRESSION_DURATION_IN_SECONDS_*	Enter the duration in seconds to ignore the participant's requests to refresh the Content display. Default setting: 10
MAX_NUMBER_OF_MANAGEMENT_SESSIONS_PER_SYSTEM	Defines the maximum number of concurrent management sessions (http and https connections) per system. Value: 4 - 80 Default: 80
MAX_NUMBER_OF_MANAGEMENT_SESSIONS_PER_USER	Defines the maximum number of concurrent management sessions (http and https connections) per user. Value: 4 - 80 Default: 10
MAX_PASSWORD_REPEAPED_CHAR	Allows the administrator to configure the maximum number of consecutive repeating characters to be allowed in a user password. Range: 1 - 4 Default: 2
MCU_DISPLAY_NAME	The name of the MCU that is displayed on the endpoint's screen when connecting to the conference. Default: POLYCOM RealPresence Collaboration Server 800sPOLYCOM
MIN_PASSWORD_LENGTH	The length of passwords. Possible value: between 0 and 20. **0** means this rule is not enforced.
MIN_PWD_CHANGE_FREQUENCY_IN_DAYS	Defines the frequency with which a user can change a password. Values: 0 -7. **0** (standard default) - users do not have to change their passwords.
MIN_SYSTEM_DISK_SPACE_TO_ALERT	Defines a minimum remaining Collaboration Server disk capacity in megabytes. If the remaining disk capacity falls below this level an active alarm is raised. **Default:** 2048
MS_ENVIRONMENT	If YES, sets the Collaboration Server SIP environment to integrate with Microsoft OCS solution. Default: NO

Table 20-1 System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
MULTIPLE_SERVICES	Determines whether the Multiple Services option is be activated once the appropriate license is installed. Possible Values: YES / NO Default: NO
NUMERIC_CHAIR_PASS_DEFAULT_LEN	This flag enables or disables the automatic generation of chairperson passwords and determines the number of digits in the chairperson passwords assigned by the MCU. Possible values are: • **0** disables the automatic password generation. Any value other than 0 enables the automatic generation of chairperson passwords if the flag HIDE_CONFERENCE_PASSWORD is set to NO. • **1 – 16**, default: 6 (Standard Security Mode) If the default is used, in non-secured mode the system will automatically generate chairperson passwords that contain 6 characters.
NUMERIC_CHAIR_PASS_MAX_LEN	The maximum number of digits that the user can enter when manually assigning a password to the chairperson. Range: **0 – 16** Default: 16
NUMERIC_CHAIR_PASS_MIN_LEN	Defines the minimum length required for the Chairperson password. Value: 0-16 Default: **0 -** this rule is not enforced.
NUMERIC_CONF_ID_LEN	Defines the number of digits in the Conference ID that will be assigned by the MCU. Enter 0 to disable the automatic assignment of IDs by the MCU and let the Collaboration Server user manually assign them. Range: 2-16 (Default: 4).
NUMERIC_CONF_ID_MAX_LEN	The maximum number of digits that the user can enter when manually assigning an ID to a conference. Range: 2-16 (Default: 8) **Note:** Selecting 2 limits the number of simultaneous ongoing conferences to 99.
NUMERIC_CONF_ID_MIN_LEN	The minimum number of digits that the user must enter when manually assigning an ID to a conference. Range: 2-16 (Default: 4) **Note:** Selecting 2 limits the number of simultaneous ongoing conferences to 99.

Table 20-1 *System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
NUMERIC_CONF_PASS_DEFAULT_LEN	This flag enables or disables the automatic generation of conference passwords and determines the number of digits in the conference passwords assigned by the MCU. Possible values are: • **0** disables the automatic password generation. Any value other than 0 enables the automatic generation of conference passwords if the flag HIDE_CONFERENCE_PASSWORD is set to NO. • **1 – 16**, default: 6 If the default is used, in non-secured mode the system will automatically generate conference passwords that contain 6 characters.
NUMERIC_CONF_PASS_MAX_LEN	The maximum number of digits that the user can enter when manually assigning a password to the conference. Range: **0 – 16** Default (both Modes): 16
NUMERIC_CONF_PASS_MIN_LEN	Defines the minimum length required for the Conference password. Value: 0-16 • Default: **0** - this rule is not enforced.
PAL_NTSC_VIDEO_OUTPUT	When set to AUTO (default), the video output sent by the Collaboration Server is either PAL or NTSC format, depending on the current speaker in the layout. This ensures full synchronization between the frame rate of the speaker and the video encoder, ensuring smoother video. In environments where the majority of endpoints are configured to either NTSC or PAL, the flag can be set accordingly to change the video encoding of the Collaboration Server to be compatible with the majority of endpoints in the call. Possible Values: AUTO, PAL, NTSC
PASSWORD_EXPIRATION_DAYS	Determines the duration of password validity. Value: between 0 and 90 days. 0 - user passwords do not expire.
PASSWORD_EXPIRATION_DAYS_MACHINE	Enables the administrator to change the password expiration period of *Application-user*'s independently of regular users. Default: 365 (days).
PASSWORD_EXPIRATION_WARNING_DAYS	Determines the display of a warning to the user of the number of days until password expiration. Value: between 0 and 14 days. 0 - password expiry warnings are not displayed.
PASSWORD_HISTORY_SIZE	The number of passwords that are recorded to prevent users from re-using their previous passwords. Values are between 0 and 16. 0 (standard default) - the rule is not enforced.

Table 20-1 System Flags – MCMS_PARAMETERS (Continued)

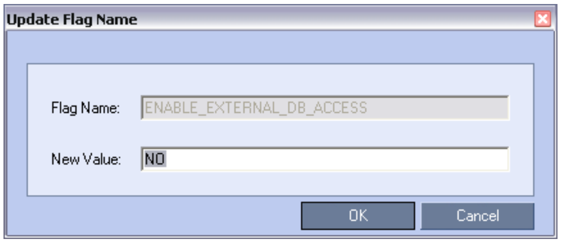
Flag	Description
RESTRICT_CONTENT_BROADCAST_TO_LECTURER	If set to YES, only the conference lecturer may send content to the conference. If set to NO, any conference participant can send content. Default: YES
RRQ_WITHOUT_GRQ	To enable registration, some gatekeepers require sending first RRQ and not GRQ. Set flag to **YES**, if this behavior is required by the gatekeeper in your environment. Default: NO. *GRQ (Gatekeeper Request)* - Gatekeeper discovery is the process an endpoint uses to determine which gatekeeper to register with. *RRQ* - registration request sent to the gatekeeper.
SEPARATE_MANAGEMENT_NETWORK	Enables/disables the Network Separation Default: NO.
SESSION_TIMEOUT_IN_MINUTES	If there is no input from the user or if the connection is idle for longer than the number of minutes specified by this flag, the connection to the Collaboration Server is terminated. Value: 0-999 0 - Session Timeout is disabled. Default: 0
SIP_AUTO_SUFFIX_EXTENSION	Used to automatically add a suffix to a SIP address (To Address) instead of adding it manually in the *Collaboration Server Web Client* (SIP address) when the SIP call is direct-dial and not through a Proxy. **Example:** Participant Name = john.smith Company Domain = maincorp.com SIP_AUTO_SUFFIX_EXTENSION flag value = @maincorp.com Entering john.smith will generate a SIP URI = john.smith@maincorp.com
STAR_DELIMITER_ALLOWED	When set to YES, an asterisk "*" can be used as a delimiter in Conference and Meeting Room dial strings. The dial string is first searched for "'#' first followed by "*". Default: NO

Table 20-1 System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
SYSTEM_BROADCAST_VOLUME	This value is used when the system flag FORCE_SYSTEM_BROADCAST_VOLUME is set to YES. Determines the default audio level with which the participants connects and sends audio to the conference. The volume scale is from 1 to 10, where 1 is the weakest and 10 is the strongest. The default connection value is 5. Each unit change represents an increase or decrease of 3 dB (decibel). Range: 1-10 Default: 5
SYSTEM_LISTENING_VOLUME	This value is used when the system flag FORCE_SYSTEM_LISTENING_VOLUME is set to YES. Determines the default audio level with which the participants connects and receives audio from the conference. The volume scale is from 1 to 10, where 1 is the weakest and 10 is the strongest. The default value is 5. Each unit change represents an increase or decrease of 3 dB (decibel). Range: 1-10 Default: 5
TERMINATE_CONF_AFTER_CHAIR_DROPPED	From Version 8.1, this flag's functionality is replaced by the **Terminate Conference after Chairperson Drops** check box in the *Profile - IVR* dialog box. In versions prior to 8.1, if YES, sets conferences to automatically terminate if the Chairperson disconnects from the conference. This takes effect only if the *Conference Requires Chairperson* check box in the Conference Profile Properties, IVR Tab, is selected. Default: YES **Note:** In order for the "Chairperson Exit" message to be played this flag must be set to YES.
USER_LOCKOUT	If YES, a user is locked out of the system after three consecutive Login failures with same User Name. The user is disabled and only the administrator can enable the user within the system. Default: NO
USER_LOCKOUT_DURATION_IN_MINUTES	Defines the duration of the Lockout of the user. Value: 0 - 480 0 means permanent User Lockout until the administrator re-enables the user within the system. Default: 0
USER_LOCKOUT_WINDOW_IN_MINUTES	Defines the time period during which the three consecutive Login failures occur. Value: 0 - 45000 0 means that three consecutive Login failures in any time period will result in User Lockout. Default: 60

3 To modify a flag value, double-click or select the flag and click the **Edit Flag** button.

4 In the *New Value* field, enter the flag's new value.



Update Flag Name

Flag Name: ENABLE_EXTERNAL_DB_ACCESS

New Value: NO

5 Click **OK** to close the *Update Flag* dialog box.

6 Repeat steps 2–4 to modify additional flags.

7 Click **OK** to close the *System Flags* dialog box

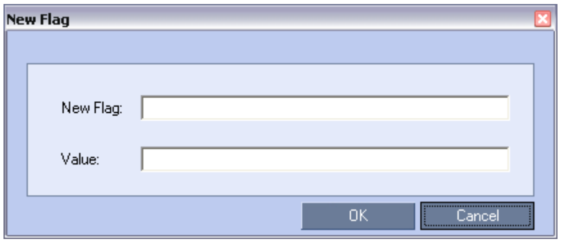
For flag changes (including deletion) to take effect, reset the MCU. For more information see "*Resetting the Collaboration Server"* on page **19-57**.

Manually Adding and Deleting System Flags

To add a flag:

1 In the *System Flags* dialog box, click the **New Flag** (New Flag) button.

The *New Flag* dialog box is displayed.



New Flag

New Flag:

Value:

2 In the *New Flag* field enter the flag name.

3 In the *Value* field enter the flag value.

The following flags can be manually added to the *MCMS_PARAMETERS* tab:

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS*

Flag	Description
802_1X_CERTIFICATE_MODE	Not supported with RealPresence Collaboration Server Virtual Edition.
802_1X_SKIP_CERTIFICATE_VALIDATION	Not supported with RealPresence Collaboration Server Virtual Edition.
802_1X_CRL_MODE	Not supported with RealPresence Collaboration Server Virtual Edition.

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
802_FIPS_MODE	Not supported with RealPresence Collaboration Server Virtual Edition.
ACCEPT_VOIP_DTMF_TYPE	Defines the type of *DTMF* tones *(inband)* or digits (*outband)* that the Collaboration Server will accept in *VOIP* calls. **Range:** • **0** - Auto (default): *Inband* or *outband DTMF* tones/digits are accepted depending on the endpoint's current setting. If the endpoint switches from *inband* to *outband* or visa versa the value of the SET_DTMF_SOURCE_DIFF_IN_SEC flag determines the time interval after which both *inband* and *outband* tones/digits will be accepted. • **1** - *Outband* (H.245) only • **2** - *Inband* only
ANAT_IP_PROTOCOL	If YES, enables *Alternative Network Address Types*. Range: DISABLED, AUTO, PREFER_IPv4, PREFER_IPv6 • Default: **YES**
APACHE_KEEP_ALIVE_TIMEOUT	If the connection is idle for longer than the number of seconds specified by this flag, the connection to the Collaboration Server is terminated. Value: 0 - 999 Default: 15 **Note:** A value of 0 results in an unlimited keep-alive duration.
AVOID_VIDEO_LOOP_BACK_IN_CASCADE	When set to YES the current speaker’s image is not sent back through the participant link in cascaded conferences with conference layouts other than 1x1. Default: YES Range: YES / NO
BLOCK_CONTENT_LEGACY_FOR_LYNC	This flag is used to control the system behavior in an environment where some Lync clients use the Polycom CCS plug-in and some do not. When set to **NO** (default), Content is sent to all Lync clients over the video channel, including those with the plug-in installed, even when the *Send Content to Legacy Endpoints* is disabled. Other, non-Lync legacy endpoints will not be affected by this flag and will receive content according to the *Send Content to Legacy Endpoints* settings in the conference Profile. When set to **YES**, Content is not sent to Lync clients over the video channel including those with the Polycom CCS plug-in installed, even when the *Send Content to Legacy Endpoints* is enabled. Other, non-Lync legacy endpoints will not be affected by this flag and will receive content according to the *Send Content to Legacy Endpoints* settings in the conference Profile.

Table 20-2 Manually Added System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
BURN_BIOS	Although not recommended, setting this flag's value to NO will prevent BIOS upgrade. Default: YES.
CAC_ENABLE	When set to YES, enables the Call Admission Control implementation in the Collaboration Server. Default: NO (CAC is disabled)
CASCADE_LINK_PLAY_TONE_ON_CONNECTION	When set to **YES,** the Collaboration Server plays a tone when a cascading link between conferences is established. The tone is played in both conferences. This tone is not played when the cascading link disconnects from the conferences. The tone used to notify that the cascading link connection has been established cannot be customized. Default value: **NO**. The tone volume is controlled by the same flag as the IVR messages and tones: IVR_MESSAGE_VOLUME.
CELL_IND_LOCATION	Change the location of the display of *Network Quality Indicators* displayed in the cells of the conference *Video Layout*. Default: TOP_RIGHT Range: • BOTTOM_LEFT • BOTTOM_RIGHT • TOP_LEFT • TOP_RIGHT
CFG_KEY_ENABLE_FLOW_CONTROL_REINVITE	Used to enable or disable sending a *re-INVITE* to endpoints to adjust their data rate. When set to YES, *re*-INVITE is used for endpoints that do not support *flow control* in SIP using either the *Information* or *RTCP Feedback* mechanisms. Default: NO.
CONF_GATHERING_DURATION_SECONDS	The value of this *System Flag* sets the duration of the *Gathering Phase* in seconds. The *Gathering Phase* duration of the conference is measured from the scheduled start time of the conference. Range: 0 - 3600 Default: 180 For more information see "*Video Preview (AVC Only Participants)"* on page 3-20.

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
CP_REGARD_TO_INCOMING_SETUP_RATE	For use in the Avaya Environment. If set to YES, the Collaboration Server calculates the line rate for incoming calls in CP conferences, according to the line rate which is declared by the endpoint in the H.225 setup message. If set to NO, the rate is calculated according to the conference line rate regardless of the rate in the H.225 setup message. Default: YES.
CPU_BONDING_LINK_MONITORING_FREQUENCY	Used when using the *MII Monitor* for troubleshooting networks.This flag sets the *MII Polling Interval* in milliseconds. A value of zero disables *MII* monitoring. Default: 100
CPU_BONDING_MODE	Sets the *Bonding Mode* of the *Signalling* and *Management* network interface controllers. *Mode=6, balance-alb,* *(Adaptive Load Balancing)* includes *balance-tlb, (Transmit Load Balancing*) and *balance-rlb* (*Receive Load Balancing*) for *IPV4* traffic. No special switch support is required. *Receive Load Balancing* is achieved by *ARP* negotiation. Outbound ARP Replies are intercepted and their source hardware address is overwritten with the unique hardware address of one of the slaves in the bond. In this way different peers will use different hardware addresses for the server. **Note:** *balance-alb* is the only supported value. All other possible values are for troubleshooting purposes only. **Default:** *balance-alb* **Possible values:** • balance-alb • balance-rr • active-backup • balance-xor • broadcast • 802.3ad • balance-tlb
DETECT_SIP_EP_DISCONNECT _TIMER	The flag value indicates the amount of time in seconds to wait for an RTCP or RTP message to be received from the endpoint. When the time that was set in the system flag has elapsed and no RTCP or RTP audio or video message has been received on either the audio or the video channel, the MCU disconnects the SIP endpoint from the conference. Default: 20 (seconds) Range: 0 - 300 For more information see "*Detecting SIP Endpoint Disconnection”* on page **12-32**.

Table 20-2 Manually Added System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
DISABLE_CELLS_NETWORK_IND	Disable the display of *Network Quality Indicators* displayed in the cells of the conference *Video Layout*. Default: YES Range: YES / NO
DISABLE_DUMMY_REGISTRATION	Enables or disables SIP dummy registration on the domain. Possible Values: NO (Default) - Disables SIP dummy registration. YES - Enables SIP dummy registration. **Note:** For homologation and certification testing, the flag must be set to YES.
DISABLE_GW_OVERLAY_INDICATION	When set to **NO** (default), displays progress indication during the connection phase of a gateway call. Set the value to **YES** to hide the connection indications displayed on the participant's screen during the connection phase of a gateway call.
DISABLE_SELF_NETWORK_IND	Disable the display of the *Network Quality Indicator* of the participant's own endpoint. Default: NO Range: YES / NO
DISABLE_WIDE_RES_TO_SIP_DIAL_OUT	When set to **NO** (default), the Collaboration Server sends wide screen resolution to dial-out SIP endpoints. Endpoint types that do not support wide screen resolutions are automatically identified by the Collaboration Server according to their product type and version and will not receive the wide resolution even if the flag is set to YES. When manually added and set to **YES**, the Collaboration Server does not send wide screen. Default: NO.
DTMF_FORWARD_ANY_DIGIT_TIMER_SECONDS	Used for DTMF code suppression in cascading conferences. Determines the time period (in seconds) that MCU A will forward DTMF inputs from conference A participants to MCU B. Flag range (in seconds): 0 - 360000 This flag is defined on MCU A (the calling MCU). For more information, see"*Video Layout in Cascading conferences (CP and mixed CP and SVC)"* on page **5-1**.
ENABLE_CISCO_GK	When set to YES, it enables the use of an identical prefix for different Collaboration Servers when registering with a Cisco MCM Gatekeeper. Default: NO.

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
ENABLE_CLOSED_CAPTION	Enables or disables the Closed Captions option that allow endpoints to endpoints to provide real-time text transcriptions or language translations of the video conference. When set to NO (default), Closed Captions are disabled. When set to YES, Closed Captions are enabled.
ENABLE_EPC	When set to YES (default), enables Polycom proprietary People+. When set to NO, disables this feature for all conferences and participants.
ENABLE_EXTERNAL_DB_ACCESS	If YES, the Collaboration Server connects to an external database application, to validate the participant’s right to start a new conference or access a conference. Default: NO
ENABLE_H239	When set to YES, Content is sent via a separate Content channel. Endpoints that do not support H.239 Content sharing will not be able to receive When set to NO, the Content channel is closed. In such a case, H.239 Content is sent via the video channel (“people” video) enabling endpoints that do not support H.239 Content sharing to receive the Content in their video channel. Default: YES.
ENABLE_H239_ANNEX_T	In H.239-enabled MIH Cascading, when MGC is on level 1, enables sending Content using Annex T.
ENABLE_LYNC_RTCP_INTRA	When set to YES, *RTCP FIR* is used for sending *Intra Requests*. When set to NO *Intra Requests* are sent using *SIP INFO Messages*. **Range:** YES / NO **Default:** NO
ENABLE_MS_FEC	Enables the Microsoft FEC (Forward Error Correction) support for RTV. Range: Auto/No Default: Auto When set to **Auto**, FEC support is enabled. FEC uses the DV00 option (DV=00 - one FEC per frame using XOR). When set to **No**, FEC support is disabled.
ENABLE_NO_VIDEO_RESOURCES_AUDIO_ONLY_MESSAGE	Enables playing a voice message that Informs the participant of the lack of Video Resources in the Collaboration Server and that he/she is being connected as Audio Only. Default: YES

Table 20-2 Manually Added System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
ENABLE_SELECTIVE_MIXING	Enables (default) or disables the Automatic muting of noisy AVC endpoints. For more details, see "*Permanent Conference"* on page **3-49**. When set to YES, the automatic muting of noisy endpoints can be enabled or disabled at the conference level in the *Conference Profile - Audio Settings* dialog box. When set to NO, the automatic muting of noisy endpoints is disabled at the conference level and cannot be enabled in the Conference *Profile - Audio Settings* dialog box. Default: YES **Note:** MCU reset is not required when changing the flag value.
ENABLE_SIP_PEOPLE_PLUS_CONTENT	If security is of higher priority than SIP Content sharing, SIP People+Content can be disabled by setting this System Flag to NO. (The content management control (BFCP) utilizes an unsecured channel (60002/TCP) even when SIP TLS is enabled.) Default: YES
ENABLE_SIP_PPC_FOR_ALL_USER_AGENT	When set to YES, SIP People+Content and BFCP capabilities are declared with all vendors' endpoints. Default: YES Range: YES / NO
ENABLE_SIRENLPR	Enable / disable SirenLPR Audio Algorithm for use in IP (H.323, SIP) calls in both CP and VSW conferences. Range: YES / NO Default: YES
ENABLE_SIRENLPR_SIP_ENCRYPTION	Enables the *SirenLPR* audio algorithm when using encryption with the *SIP* protocol. Range: YES / NO Default: NO
ENABLE_TC_PACKAGE	Enables or disables Network Traffic Control. Range: YES / NO Default: NO
ENABLE_TEXTUAL_CONFERENCE_STATUS	Set the value of this flag to NO to disable *Text Indication.* This setting is recommended for MCUs running Telepresence conferences. Default: YES.
ENABLE_VIDEO_PREVIEW	Enables the Video Preview feature. Default: YES. For more details, see "*Video Preview (AVC Only Participants)"* on page **3-20**.
EXTERNAL_CONTENT_DIRECTORY	The Web Server folder name. Change this name if you have changed the default names used by the CMA/XMA application. Default: /PlcmWebServices

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
EXTERNAL_CONTENT_IP	Enter the IP address of the CMA/XMA server in the format: **http://[IP address of the CMA server]**. For example, http://172.22.185.89. This flag is also the trigger for replacing the internal Collaboration Server address book with the CMA global Address Book. When empty, the integration of the CMA address book with the Collaboration Server is disabled.
EXTERNAL_CONTENT_PASSWORD	The password associated with the user name defined for the Collaboration Server in the CMA/XMA server.
EXTERNAL_CONTENT_PORT	The CMA/XMA port used by the Collaboration Server to send and receive XML requests/responses. Default: 80.
EXTERNAL_CONTENT_USER	The login name defined for the Collaboration Server in the CMA/XMA server defined in the format: domain name/user name.
EXTERNAL_DB_DIRECTORY	The URL of the external database application. For the sample script application, the URL is: *<virtual directory>/SubmitQuery.asp*
EXTERNAL_DB_IP	The IP address of the external database server, if one is used. Default: 0.0.0.0
EXTERNAL_DB_LOGIN	The login name defined for the Collaboration Server in the external database server. Default: POLYCOM
EXTERNAL_DB_PASSWORD	The password associated with the user name defined for the Collaboration Server on the external database server. Default: POLYCOM
EXTERNAL_DB_PORT	The external database server port used by the Collaboration Server to send and receive XML requests/responses. For secure communications set the value to 4433. Default: 5005.
FADE_IN_FADE_OUT	Enables or disables the transition format between speakers in a Continuous Presence conference. When set to YES (default), the system fades in the current speaker while fading out the previous speaker. When set to NO, the transition is sharp and immediate.

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
FORCE_1X1_LAYOUT_ON_CASCADED_LINK_CONNECTION	When set to **YES**, the cascaded link is automatically set to Full Screen (1x1) in CP conferences forcing the speaker in one cascaded conference to display in full window in the video layout of the other conference. Set this flag to **NO** when connecting to an MGC using a cascaded link, if the MGC is functioning as a Gateway and participant layouts on the other network are not to be forced to 1X1. Default: YES
FORCE_AUDIO_CODEC_FOR_MS_SINGLE_CORE	This flag is used to force the use of a specific Audio algorithm when a Microsoft Office Communicator R2 or Lync Client is hosted on a workstation with a single core processor. The flag value overrides the default audio algorithm selection (G.722.1) that may cause audio quality problems when G.722.1 is used by Microsoft Clients running on single processor workstations. This flag can be set to: • **AUTO** – No forcing occurs and the Collaboration Server negotiates a full set of Audio algorithm during capabilities exchange. • **G711A/U** or **G722** – Set this flag value according to the hosting workstation capabilities. If the Collaboration Server detects single core host during capabilities exchange it will assign a G.711 or G.722 Audio algorithm according to the flag value. Possible values: AUTO, G711A, G711U, G722 Default: G711A
FORCE_ENCRYPTION_FOR_UNDEFINED_PARTICIPANT_IN_WHEN_AVAILABLE_MODE	When set to **YES**, *Undefined participants* must connect encrypted, otherwise they are disconnected. When set to **NO** (default) and the conference *Encryption* in the *Profile* is set to “Encrypt When Possible”, both Encrypted and Non-encrypted *Undefined participants* can connect to the same conferences, where encryption is the preferred setting. Default: NO
FORCE_G711A	Setting this flag forces the use of the *G711A Audio Codec.* **Possible values:** YES / NO **Default:** NO
FORCE_RESOLUTION	Use this flag to specify IP (H.323 and SIP) endpoint types that cannot receive wide screen resolution and that were not automatically identified as such by the Collaboration Server. Possible values are endpoint types, each type followed by a semicolon. For example, when disabling Wide screen resolution in an HDX endpoint enter the following string: **HDX;** **Note:** Use this flag when the flag SEND_WIDE_RES_TO_IP is set to YES.

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
FORCE_STATIC_MB_ENCODING	This flag supports Tandberg MXP mode of sending and receiving video by IP endpoint in HD 720p resolution and Video Quality set to Motion. Default value: **Tandberg MXP**. To disable this flag, enter **NONE**.
G728_IP	Enables or disables declaration of G.728 Audio Algorithm capabilities in IP calls. Range: YES / NO Default: NO
H239_FORCE_CAPABILITIES	When the flag is set to NO, the Collaboration Server only verifies that the endpoint supports the Content protocols: Up to H.264 or H.263. When set to YES, the Collaboration Server checks frame rate, resolution and all other parameters of the Content mode as declared by an endpoint before receiving or transmitting Content. Default: NO.
H264_HD_GRAPHICS_MIN_CONTENT_RATE	Determines the minimum content rate (in kbps) required for endpoints to share H.264 high quality content via the Content channel When Content Setting is Graphics. Range: 0-1536 Default: 128
H264_HD_HIGHRES_MIN_CONTENT_RATE	Determines the minimum content rate (in kbps) required for endpoints to share H.264 high quality content via the Content channel When Content Setting is Hi Resolution Graphics. Range: 0-1536 Default: 256
H264_HD_LIVEVIDEO_MIN_CONTENT_RATE	Determines the minimum content rate (in kbps) required for endpoints to share H.264 high quality content via the Content channel When Content Setting is Live Video. Range: 0-1536 Default: 384
H323_FREE_VIDEO_RESOURCES	For use in the Avaya Environment. In the Avaya Environment there are features that involve converting undefined dial-in participants' connections from video to audio (or vice versa). To ensure that the participants' video resources remain available for them, and are not released for use by Audio Only calls, set this flag to **NO**. If set to YES, the Collaboration Server will release video resources for *Audio Only* calls. Default: YES.

***Table 20-2** Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
HIDE_CONFERENCE_PASSWORD	If set to YES: • Conference and Chairperson Passwords that are displayed in the *Collaboration Server Web Client* or *RMX Manager* are hidden when viewing the properties of the conference. • Automatic generation of passwords (both conference and chairperson passwords) is disabled, regardless of the settings of the flags: • NUMERIC_CONF_PASS_DEFAULT _LEN • NUMERIC_CHAIR_PASS_ DEFAULT_LEN. For more information see "*Automatic Password Generation Flags"* on page **20-37** Default: NO.
IP_LINK_ENVIRONMENT	In H.239-enabled MIH Cascading, when MGC is on level 1, setting this flag to YES will adjust the line rate of HD Video Switching conferences run on the RealPresence Collaboration Server Virtual Edition from 1920Kbps to 18432, 100bits/sec to match the actual rate of the IP Only HD Video Switching conference running on the MGC. **Note:** If the flag MIX_LINK_ENVIRONMENT is set to NO, the *IP_ENVIRONMENT_LINK* flag must be set to YES.
IP_RESPONSE_ECHO	When the *System Flag* value is **YES**, the Collaboration Server will respond to *ping (IPv4)* commands. When set to **NO**, the Collaboration Server will not respond to *ping* commands.
ITP_CERTIFICATION	When set to **NO** (default), this flag disables the telepresence features in the Conference Profile. Set the flag to **YES** to enable the telepresence features in the Conference Profile (provided that the appropriate License is installed).
LAN_REDUNDANCY	Enables Local Area Network port redundancy. Default: NO Range: YES / NO Note: If the flag value is set to YES and either of the LAN connections (LAN1 or LAN2) experiences a problem, an active alarm is raised stating that there is no LAN connection, specifying both the card and port number.

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
MANAGE_TELEPRESENCE_ROOM_SWITCH_LAYOUTS	Determines whether the *MLA* or the *RMX* controls the *Room Switch Telepresence Layouts*. • When set to NO, the *RMX* does not manage *Telepresence Room Switch Layouts* and they continue to be managed by the MLA. • When set to YES, the *RMX* manages *Telepresence Room Switch Layouts*. Default: NO Range: YES / NO **Note:** System re-start is not required for this flag's settings to take effect. For more information see "*Lecture Mode (AVC CP Only)"* on page **3-38**.
MAX_ALLOWED_RTV_HD_FRAME_RATE	Defines the threshold Frame Rate (fps) in which RTV Video Protocol initiates HD resolutions. Flag values are as follows: Range: 0-30 (fps) Default: 0 (fps) - Implements any Frame Rate based on Lync RTV Client capabilities
MAX_RTV_RESOLUTION	Enables you to override the Collaboration Server resolution selection and limit it to a lower resolution, hence minimizing the resource usage to 1 or 1.5 video resources per call instead of 3 resources. Possible flag values are: AUTO (default), QCIF, CIF, VGA or HD720.
MAX_TRACE_LEVEL	This flag indicates the debugging level for system support. **Possible values:** TRACE = t, DEBUG = d, INFO_NORMAL = n, INFO_HIGH = i, WARN = w, ERROR = e, FATAL = f, OFF = o. **Default:** n
MAXIMUM_RECORDING_LINKS	The maximum number of Recording Links available for selection in the Recording Links list and the Conference Profile - Recording dialog box. Range: 1 - 100 Default: 20
MINIMUM_FRAME_RATE_THRESHOLD_FOR_SD	Low quality, low frame rate video is prevented from being sent to endpoints by ensuring that an SD channel is not opened at frame rates below the specified value. Range: 0 -30 Default: 15

Table 20-2 Manually Added System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
MIX_LINK_ENVIRONMENT	In H.239-enabled MIH Cascading, when MGC is on level 1, setting this flag to YES will adjust the line rate of HD Video Switching conferences run on the RealPresence Collaboration Server Virtual Edition from 1920Kbps to 17897, 100bits/sec to match the actual rate of the HD Video Switching conference running on the MGC. **Note:** If the flag MIX_LINK_ENVIRONMENT is set to YES, the IP_ENVIRONMENT_LINK flag must be set to NO.
MS_CAC_AUDIO_MIN_BR	The minimum bit rate for audio using the Microsoft CAC (Call Admission Control) protocol. When the bit rate is lower than the MS_CAC_AUDIO_MIN_BR, the call is not connected. Range: 0 - 384 Default: 30
MS_CAC_VTDEO_MIN_BR	The minimum bit rate for video using the Microsoft CAC (Call Admission Control) protocol. When the bit rate is lower than the MS_CAC_VIDEO_MIN_BR, the call is not connected as a video call.. Range: 0 - 384 Default: 40
MS_PROXY_REPLACE	Enables the *proxy=replace* parameter in the *SIP Header*. When set to YES the outbound proxy to replaces the contact information in the contact header with its own enabling other clients and servers to reach the client using the proxy's *IP* address, even if the client is behind a firewall. **Possible Values:** YES / NO **Default:** YES
NETWORK_IND_CRITICAL_PERCENTAGE	The percentage degradation due to packet loss required to change the indicator from *Major* to *Critical*. Default: 5
NETWORK_IND_MAJOR_PERCENTAGE	The percentage degradation due to packet loss required to change the indicator from *Normal* to *Major*. Default: 1
NUM_OF_INITIATE_HELLO_MESSAGE_IN_CALL_ESTABLISHMENT	Indicates how many times the Hello (keep alive) message is sent from the Collaboration Server to the endpoint in an environment that includes a Session Border Controller (SBC) with a 3-second interval between messages. Range: 1 to 10. Default:3
NUMBER_OF_REDIAL	Enter the number re dialing attempts required. Dialing may continue until the conference is terminated. Default: 3

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
OCSP_RESPONDER_TIMEOUT	Determines the number of seconds the RMX is to wait for an OCSP response from the OCSP Responder before failing the connection. Network latency or slow WAN links can cause login problems when logging in to the RMX's Management Network.This System Flag's value determines the number of seconds the MCU is to wait for an OCSP response from the OCSP Responder before failing the connection. Default: 3 (seconds) Range: 1-20 (seconds)
PARTY_GATHERING_DURATION_SECONDS	The value of this *System Flag* sets the duration, in seconds, of the display of the *Gathering* slide for participants that connect to the conference after the conference start time. Range: 0 - 3600 Default: 15 For more information see "*Video Preview (AVC Only Participants)"* on page **3-20**.
PASSWORD_FAILURE_LIMIT	The number of unsuccessful Logins permitted in Ultra Secure Mode. Default: 3
PCM_FECC	Determines whether the DTMF Code, ##, the Far/Arrow Keys (FECC) or both will activate the PCM interface. This flag can be also be used in combination with DTMF code definitions to disable PCM. Possible Values: YES / NO Default: YES.
PCM_LANGUAGE	Determines the language of the PCM interface. Possible Values are: ENGLISH, CHINESE_SIMPLIFIED, CHINESE_TRADITIONAL, JAPANESE, GERMAN, FRENCH, SPANISH, KOREAN, PORTUGUESE, ITALIAN, RUSSIAN, NORWEGIAN Default: Current Collaboration Server Web Client language.
PORT_GAUGE_ALARM	When set to YES, if system resource usage reaches the High Port Usage Threshold as defined for the Port Gauges, System Alerts in the form of an Active Alarm and an SNMP trap are generated.
PRESERVE_ICE_CHANNEL_IN_CASE_OF_LOCAL_MODE	When set to NO (default), local the ICE channel is closed after applying CAC bandwidth management when Call Admission Control is enabled in the local network. When set to YES, the ICE channel is preserved open throughout the call. Default: NO

Table 20-2 Manually Added System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
PRESERVE_PARTY_CELL_ON_FORCE_LAYOUT	Used to prevent reassignment of cells in a forced layout that were assigned to endpoints that have disconnected, paused their video, or have been removed from the conference. The cell will remain black until the endpoint reconnects or a new layout is used, or the conference ends. Range: YES / NO Default: NO • NO - Cells of dropped endpoints are reassigned. Endpoints that reconnect will be treated as new endpoints. • YES - Cells of dropped endpoints are not reassigned, but will be reserved until the endpoint reconnects. For information see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*, "*Force Layout and Preserve Participant Call"* on page **3-62**.
QOS_IP_AUDIO	Used to select the priority of audio packets when *DiffServ is* the is the selected method for packet priority encoding. Default: 0x31
QOS_IP_VIDEO	Used to select the priority of video packets when *DiffServ is* the is the selected method for packet priority encoding. Default: 0x31
QOS_MANAGEMENT_NETWORK	Enter the *DSCP* value for the *RMX Management Network*. Default: 0x10 Range: 0x00 - 0x3F
REDUCE_CAPS_FOR_REDCOM_SIP	To accommodate deployments where some devices have limits on the size of the SDP payload in SIP messages (such as LSCs from Redcom running older software versions), when the flag value = YES, the SDP size is less than 2kb and includes only one audio and one video media line. Default: NO
REDIAL_INTERVAL_IN_SECONDS	Enter the number of seconds that the Collaboration Server should wait before successive re dialing attempts. Range: 0-30 (Default: 10)
REDUCE_CAPS_FOR_REDCOM_SIP	To accommodate Redcom's SDP size limit, when the flag value = YES, the SDP size is less than 2kb and includes only one audio and one video media line. Default: NO
REJECT_INCORRECT_PRECEDENCE_DOMAIN_NAME	When set to YES, when the Precedence Domain of a SIP dial-in call does not match the Precedence Domain of the RMX, the call is rejected. Possible values: YES/NO Default: No

Table 20-2 Manually Added System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
REMOVE_H323_EPC_CAP_TO_NON_POLYCOM_VENDOR	Used to disable *EPC* protocol. Use of *Polycom's* proprietary protocol, *High Profile*, *EPC*, may result in interoperability issues when used with other vendors' endpoints. **Possible values:** YES / NO **Default:** NO
REMOVE_H323_HIGH_PROFILE_CAP_TO_NON_POLYCOM_VENDOR	Used to disable *High Profile* protocol. Use of *Polycom's* proprietary protocol, *High Profile*, may result in interoperability issues when used with other vendors' endpoints. **Possible values:** YES / NO **Default:** NO
REMOVE_H323_HIGH_QUALITY_AUDIO_CAP_TO_NON_POLYCOM_VENDOR	Used to disable the following *Audio Codecs*: • G7221C • G7221 • Siren22 • Siren14 **Possible values:** YES / NO **Default:** NO
REMOVE_H323_LPR_CAP_TO_NON_POLYCOM_VENDOR	Used to disable *H.323 LPR* protocol. Use of *Polycom's* proprietary protocol, *H.323 LPR*, may result in interoperability issues when used with other vendors' endpoints. **Possible values:** YES / NO **Default:** NO
REMOVE_IP_IF_NUMBER_EXISTS	Between the time a conference is scheduled and when it becomes active, the IP of an endpoint may change, especially in an environment that uses DHCP. This flag determines if the *E.164* number is to be substituted for the IP address in the dial string. Range: YES / NO Default: YES - The IP address will be substituted with the E.164 number.
RMX_MANAGEMENT_SECURITY_PROTOCOL	Enter the protocol to be used for secure communications. **Default: TLSV1_SSLV3 (both).**
RTCP_FIR_ENABLE	When set to YES, the *Full Intra Request* (*FIR*) is sent as *INFO* (and not *RTCP*). Default = YES
RTCP_FLOW_CONTROL_TMMBR_ENABLE	Enables/disables the SIP RTCP flow control parameter. Default: YES
RTCP_FLOW_CONTROL_TMMBR_INTERVAL	Modifies the interval (in seconds) of the TMMBR (Temporary Maximum Media Stream Bit Rate) parameter for SIP RTCP flow control. Range: 5 - 999 (seconds) Default: 180

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
RTCP_PLI_ENABLE	When set to YES, the (Picture Loss Indication (*PLI*) is sent as *INFO* (and not *RTCP*). Default = YES
RTCP_QOS_IS_EQUAL_TO _RTP	Range: YES/NO Default: YES
SELF_IND_LOCATION	Change the location of the display of the *Network Quality Indicator* of the participant's own endpoint. Default: BOTTOM_RIGHT Range: • TOP_ LEFT • TOP • TOP_RIGHT • BOTTOM_ LEFT • BOTTOM • BOTTOM_RIGHT
SEND_SIP_BUSY_UPON_R ESOURCE_THRESHOLD	When set to **YES**, it enables the Collaboration Server to send a busy notification to a SIP audio endpoint or a SIP device when dialing in to the Collaboration Server whose audio resource usage exceeded the Port Usage threshold. When set to **NO**, the system does limit the SIP audio endpoint connections to a certain capacity and will not send a busy notification when the resource capacity threshold is exceeded. Default: NO

Table 20-2 Manually Added System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
SEND_SRTP_MKI	Enables or disables the inclusion of the *MKI* field in *SRTP* packets sent by the Collaboration Server. Setting the value to NO to disables the inclusion of the *MKI* field in *SRTP* packets sent by the Collaboration Server. Set this flag to: • **NO** • When all conferences on the *RMX* will not have *MS-Lync* clients participating and will have 3rd party endpoints participating. • When using endpoints (eg. *CounterPath Bria 3.2 Softphone*) that cannot decrypt *SRTP*-based audio and video streams if the *MKI* (*Master Key Identifier*) field is included in *SRTP* packets sent by the Collaboration Server. This setting is recommended for *Maximum Security Environments.* • **YES** • When any conferences on the *RMX* will have both *MS-Lync* clients and *Polycom* endpoints participating. • Some 3rd party endpoints may be unsuccessful in participating in conferences with this setting. **Notes:** • This *System Flag* must be added and set to YES (default) when *Microsoft Office Communicator* and *Lync Clients* are used as they all support *SRTP* with *MKI*. • The system flag must be added and set to NO when Siemens phones (*Openstage* and *ODC WE*) are used in the environment as they do not support *SRTP* with *MKI*. • *Polycom* endpoints function normally regardless of the setting of this flag. Default: **YES**
SEND_WIDE_RES_TO_IP	When set to **YES** (default), the Collaboration Server sends wide screen resolution to IP endpoints. Endpoint types that do not support wide screen resolutions are automatically identified by the Collaboration Server according to their product type and version and will not receive the wide resolution even when the flag is set to YES. When manually added and set to **NO**, the Collaboration Server does not send wide screen resolution to all IP endpoints. Default: YES.
SET_DTMF_SOURCE_DIFF_IN_SEC	If the ACCEPT_VOIP_DTMF_TYPE flag is set to 0 (Auto) this flag determines the interval, in seconds after which the Collaboration Server will accept both *DTMF* tones *(inband)* and digits (*outband*). **Default:** 120

Table 20-2 Manually Added System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
SIP_BFCP_DIAL_OUT_MODE	Controls *BFCP's* use of *UDP* and *TCP* protocols for dial-out *SIP Client* connections according to its value: • **AUTO** (Default) *If SIP Client* supports *UDP, TCP* or *UDP* and *TCP*: - BFCP/UDP is selected as *Content* sharing protocol. • **UDP** *If SIP Client* supports *UDP* or UDP and TCP: - *BFCP/UDP* selected as *Content* sharing protocol. *If SIP Client* supports *TCP* - Cannot share *Content*. • **TCP** *If SIP Client* supports *TCP* or *UDP* and *TCP* - BFCP/TCP selected as Content sharing protocol. *If SIP Client* supports UDP - Cannot share Content.
SIP_DUAL_DIRECTION_TCP_CON	In environments set to integration with Microsoft, if set to YES the system sends a new request on the same TCP connection (instead of opening a new one).
SIP_ENABLE_FECC	By default, FECC support for SIP endpoints is enabled at the MCU level. You can disable it by manually adding this flag and setting it to NO.
SIP_FAST_UPDATE_INTERVAL_ENV	Default setting is **0** to prevent the Collaboration Server from automatically sending an Intra request to all SIP endpoints. Enter **n** (where n is any number of seconds other than 0) to let the Collaboration Server automatically send an Intra request to all SIP endpoints every n seconds. It is recommended to set the flag to 0 and modify the frequency in which the request is sent at the endpoint level (as defined in the next flag).
SIP_FAST_UPDATE_INTERVAL_EP	Default setting is 6 to let the Collaboration Server automatically send an Intra request to Microsoft OC endpoints only, every 6 seconds. Enter any other number of seconds to change the frequency in which the Collaboration Server send the Intra request to Microsoft OC endpoints only. Enter 0 to disable this behavior at the endpoint level (not recommended).

Table 20-2 *Manually Added System Flags – MCMS_PARAMETERS (Continued)*

Flag	Description
SIP_FORMAT_GW_HEADERS_FOR_REDCOM	Controls whether the *RMX* adds special gateway prefix and postfix characters to the user portion of the *SIP URI* expressed in the "*From*" and "*Contact*" headers of *SIP* messages sent during calls involving *Gateway Services*. The addition of these characters can result in call failures with some *SIP* call servers. It is recommended to set this flag to YES whenever the *RMX* is deployed such that it registers its conferences to a *SIP* call server. Range: YES, NO Default: NO
SIP_FREE_VIDEO_RESOURCES	For use in Avaya and Microsoft Environments. When set to **NO** (required for Avaya and Microsoft environments), video resources that were allocated to participants remain allocated to the participants as long as they are connected to the conference even if the call was changed to audio only. The system allocates the resources according to the participant's endpoint capabilities, with a minimum of 1 CIF video resource. Enter YES to enable the system to free the video resources for allocation to other conference participants. The call becomes an audio only call and video resources are not guaranteed to participants if they want to add video again. Default value in Microsoft environment: NO.
SIP_TCP_PORT_ADDR_STRATEGY	Setting the flag to 1 prevents the use of two sockets for one SIP call - one for inbound traffic, one for outbound traffic. This is done by inserting port "5060/5061" into the Route[0] header. **Possible values:** • 0 - Inbound traffic on port 5060/5061 outbound traffic on port 60000 • 1 - Both inbound and outbound traffic on port 5060/5061 **Default**: 1
SOCKET_ACTIVITY_TIMEOUT	For use in Microsoft environments. When the MS_KEEP_ALIVE *System Flag* is set to YES, the value of this flag is used as the *MS Keep-Alive Timer* value.
SUPPORT_HIGH _PROFILE	Enables or disables the support of *High Profile* video protocol in CP conferences. This flag is specific to CP conferences and has no effect on VSW conferences. Range: YES / NO Default: YES
TC_BURST_SIZE	This flag regulates the Traffic Control buffer or maxburst size as a percentage of the participant line rate. Range: 1-30.
TC_LATENCY_SIZE	This flag limits the latency (in milliseconds) or the number of bytes that can be present in a queue. Range: 1-1000 (in milliseconds).

Table 20-2 Manually Added System Flags – MCMS_PARAMETERS (Continued)

Flag	Description
TCP_RETRANSMISSION_TIMEOUT	The number of seconds the server will wait for a *TCP* client to answer a call before closing the connection. Default = 5 (seconds)
V35_MULTIPLE_SERVICES	If the connection of multiple Serial Gateways to RTM-LAN cards is required: The V35_MULTIPLE_SERVICES System Flag must be set to YES. The default value of the V35_MULTIPLE_SERVICES System Flag is NO, enabling only one Serial Gateway to be supported per RTM-LAN card.
V35_ULTRA_SECURED_SUPPORT	This flag must be set to YES when deploying a *Serial Gateway S4GW* in *Ultra Secure Mode*..
VSW_CIF_HP_THRESHOLD_BITRATE	Controls the *Minimum Threshold Line Rate* (kbps) for *CIF* resolution for *High Profile-enabled VSW conferences*. Default: 64
VSW_HD1080p_HP_THRESHOLD_BITRATE	Controls the *Minimum Threshold Line Rate* (kbps) for *HD1080p* resolution for *High Profile-enabled VSW conferences*. Default: 1024
VSW_HD720p30_HP_THRESHOLD_BITRATE	Controls the *Minimum Threshold Line Rate* (kbps) for *HD720p30* resolution for *High Profile-enabled VSW conferences*. Default: 512
VSW_HD720p50-60_HP_THRESHOLD_BITRATE	Controls the *Minimum Threshold Line Rate* (kbps) for *HD720p50* and *HD720p50* resolutions for *High Profile-enabled VSW conferences*. Default: 832
VSW_RATE_TOLERANCE_PERECENT	Determines the percentage of bandwidth that can be deducted from the required bandwidth to allow participants to connect to the conference. For example, a value of 20 will allow a participant to connect to the conference if the allocated line rate is up to 20% lower than the conference line rate (or between 80% to 100% of the required bandwidth). Range: 0 - 75 Default: 0
VSW_SD_HP_THRESHOLD_BITRATE	Controls the *Minimum Threshold Line Rate* (kbps) for *SD* resolution for *High Profile-enabled VSW conferences*. Default: 128
WRONG_NUMBER_DIAL_RETRIES	The number of re-dial attempts for a wrong destination number or a wrong destination number time-out. Range: 0 - 5 Default: 3 A flag value of 0 means that no redials are attempted.

4 Click **OK** to close the *New Flag* dialog box.
The new flag is added to the flags list.

5 Click **OK** to close the *System Flags* dialog box.

For flag changes (including deletion) to take effect, reset the MCU. For more information see "*Resetting the Collaboration Server”* on page **19-57**.

Manually Adding Flags to the CS_MODULE_PARAMETERS Tab

Using the procedure to manually add flags to the System Configuration, the following flags can be manually added to the **CS_MODULE_PARAMETERS** tab:

Table 20-3 *Manually Added Flags - CS_MODULE_PARAMETERS Tab*

Flag	Description
CS_ENABLE_EPC	Add this flag with the value **YES** (default value is NO) to enable endpoints that support People+Content and require a different signaling (for example, FX endpoints) to receive Content.
H245_TUNNELING	For use in the Avaya Environment. In the Avaya Environment, set the flag to YES to ensure that H.245 is tunneled through H.225. Both H.245 and H.225 will use the same signaling port. Default: NO.
H323_TIMERS_SET_INDEX	Enables or disables H.323 index timer according to standard or proprietary H.323 protocol. Possible values: 0 (Default) - Sets the H.323 index timer to Polycom proprietary. 1 - Sets the H.323 index timer based on the H.323 Standard recommendation. **Note:** For homologation and certification testing, this flag must be set to 1.
MS_UPDATE_CONTACT_REMOVE	When the flag value is set to: • **YES** - The *Contact Header* is removed from the *UPDATE* message that is sent periodically to the endpoints. This is required when the *SIP Server Type* field of the *IP Network Service* is set as **Microsoft.** Removal of the *Contact Header* from the UPDATE message is required specifically by *OCS R2*. • **NO** - The *Contact Header* is included in the *UPDATE* message. This is the system behavior when the *SIP Server Type* is set as **Generic**. This is required when the Collaboration Server is configured to accept calls from both *Microsoft LYNC* and *Cisco CUCM* as *CUCM* requires the *Contact Header*.
QOS_IP_SIGNALING	Used to select the priority of IP packets when *DiffServ is* the is the selected method for packet priority encoding. Range: 0x## Default: 0x28

Table 20-3 Manually Added Flags - CS_MODULE_PARAMETERS Tab

Flag	Description
SIP_DUAL_DIRECTION_TCP_CON	For use in Microsoft environments. When set to YES, sends a new request on the same TCP connection instead of opening a new connection. Range: YES/NO Default: NO
SIP_ST_ENFORCE_VAL	For use in Microsoft environments. Session timer interval in seconds. Default = YES
SIP_TCP_TLS_TIMERS	Determines the timeout characteristics of SIP TCP TLS connections. Format: SIP_TCP_TLS_TIMERS = <string> The string contains the following parameters: Ct - Timeout of *TCP CONNECT* operation (seconds) Cs - Timeout of *TLS CONNECT* operation (seconds) A - Timeout of *accept* operation (seconds) D - Timeout of *disconnect* operation (nanoseconds) H - Timeout of *handshake* operation (seconds) Default: <1,5, 4,500000,5>
SIP_TIMERS_SET_INDEX	SIP Timer type timeout settings according to standard or proprietary protocol. Possible values are: 0 - Default 1 - SIP Standard recommendation. **Note:** For homologation and certification testing, this flag must be set to 1.
SIP_TO_TAG_CONFLICT	For use in Microsoft environments. In case of forking, a tag conflict will be resolved when Status 200 OK is received from an answering UA. Default: YES

Deleting a Flag

To delete a flag:

1 In the *System Flags* dialog box, select the flag to delete and click the **Delete Flag** button.

2 In the confirmation message box, click **Yes** to confirm.

3 Click **OK to** close the *System Flags* dialog box.

Auto Layout Configuration

The *Auto Layout* option lets the Collaboration Server automatically select the conference video layout based on the number of participants currently connected to the conference. You can modify the default selection of the conference video layout to customize it to your conferencing preferences.

Customizing the Default Auto Layout

The default *Auto Layout* is controlled by 13 flags:

PREDEFINED_AUTO_LAYOUT_0, ... , PREDEFINED_AUTO_LAYOUT_12

Each of the 11 *Auto Layout* flags can be left at its default value, or set to any of the *Possible Values* listed in Table 20-4.

The flag that controls the *Auto Layout* you wish to modify must be added to the *System Configuration* file. For more information see "*Modifying System Flags"* on page 20-1.

Table 20-4 *Flags: PREDEFINED_AUTO_LAYOUT_0,...,10*

Flag Name: PREDEFINED_AUTO_LAYOUT_n (n = Number of Participants)		
n	**Default Value**	**Possible Values**
0	CP_LAYOUT_1X1	CP_LAYOUT_1X1
1	CP_LAYOUT_1X1	CP_LAYOUT_1X2
2	CP_LAYOUT_1X1	CP_LAYOUT_1X2HOR
3	CP_LAYOUT_1x2VER	CP_LAYOUT_1X2VER
4	CP_LAYOUT_2X2	CP_LAYOUT_2X1
5	CP_LAYOUT_2X2	CP_LAYOUT_1P2HOR
6	CP_LAYOUT_1P5	CP_LAYOUT_1P2HOR_UP
7	CP_LAYOUT_1P5	CP_LAYOUT_1P2VER
8	CP_LAYOUT_1P7	CP_LAYOUT_2X2
9	CP_LAYOUT_1P7	CP_LAYOUT_1P3HOR_UP
10	CP_LAYOUT_1P7	CP_LAYOUT_1P3VER
11	CP_LAYOUT_2P8	CP_LAYOUT_1P4HOR
12	CP_LAYOUT_1P12	CP_LAYOUT_1P4HOR_UP

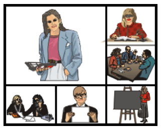
Table 20-4 *Flags: PREDEFINED_AUTO_LAYOUT_0,...,10 (Continued)*

Flag Name: PREDEFINED_AUTO_LAYOUT_n (n = Number of Participants)		
n	**Default Value**	**Possible Values**
		CP_LAYOUT_1P4VER CP_LAYOUT_1P5 CP_LAYOUT_1P7 CP_LAYOUT_1P8UP CP_LAYOUT_1P8CENT CP_LAYOUT_1P8HOR_UP CP_LAYOUT_3X3 CP_LAYOUT_2P8 CP_LAYOUT_1P12 CP_LAYOUT_4X4

Example:

Table 20-5 illustrates the effect of modifying the `PREDEFINED_AUTO_LAYOUT_5` flag in conferences with fewer or more participants than the number of windows selected in the default layout.

***Table 20-5** Example: Modifying PREDEFINED_AUTO_LAYOUT_5 Flag*

Flag	Set to Possible Value	Number of Participants	Participant's View
PREDEFINED _AUTO_LAYOUT_5 *Default =* ⊞	CP_LAYOUT _1x2VER	3	 Voice activated switching displays the current speaker in the left window of the video layout and only the two last speakers are displayed.
		7	
	CP_LAYOUT _1P5	3	 Voice activated switching displays the current speaker in the large (top left) window of the video layout.
		7	 Voice activated switching displays the current speaker in the top left window of the video layout.

CS_ENABLE_EPC Flag

Endpoints that support *People+* may require a different signaling (for example, FX endpoints). For these endpoints, manually add the flag **CS_ENABLE_EPC** with the value **YES** (default value is NO) to the **CS_MODULE_PARAMETERS** tab.

Automatic Password Generation Flags

The Collaboration Server can be configured to automatically generate conference and chairperson passwords when the *Conference Password* and *Chairperson Password* fields are left blank.

Guidelines

- If the flag **HIDE_CONFERENCE_PASSWORD** is set to **YES**, the automatic generation of passwords (both conference and chairperson passwords) is disabled, regardless of the settings of the flags NUMERIC_CONF_PASS_DEFAULT_LEN and NUMERIC_CHAIR_PASS_ DEFAULT _LEN.
- The automatic generation of conference passwords is enabled/disabled by the flag NUMERIC_CONF_PASS_DEFAULT_LEN.
- The automatic generation of chairperson passwords is enabled/disabled by the flag NUMERIC_CHAIR_PASS_ DEFAULT _LEN.
- The automatically generated passwords will be numeric and random.
- The passwords are automatically assigned to ongoing conferences, and Meeting Rooms at the end of the creation process (once they are added to the Collaboration Server).
- Automatically assigned passwords can be manually changed through the *Conference/ Meeting Room/Reservation Properties* dialog boxes.
- Deleting an automatically created password will not cause the system to generate a new password and the new password must be added manually or the field can be left blank.
- If a password was assigned to the conference via Microsoft Outlook using the PCO add-in, the system does not change these passwords and additional passwords will not be generated (for example, if only the conference password was assigned a chairperson password will not be assigned).
- If the flag values (i.e. the password lengths) are changed, passwords that were already assigned to conferencesand Meeting Rooms will not change and they can be activated using the existing passwords. Only new conferencing entities will be affected by the change.

Do not enable this option in an environment that includes a *Polycom DMA* system.

Enabling the Automatic Generation of Passwords

To enable the automatic generation of passwords, the following flags have to be defined:

Table 20-6 *Automatic Password Generation Flags*

Flag	Description
HIDE_CONFERENCE_PASSWORD	**NO (default)** - Conference and chairperson passwords are displayed when viewing the Conference/Meeting Room/ Reservation properties. It also enables the automatic generation of passwords in general. **Yes** - Conference and Chairperson Passwords are hidden (they are replaced by asterisks). It also disables the automatic generation of passwords.
NUMERIC_CONF_PASS_MIN_LEN	Enter the minimum number of characters required for conference passwords. Possible values: **0 – 16**. **0 (default)** means no minimum length.

Table 20-6 *Automatic Password Generation Flags (Continued)*

Flag	Description
NUMERIC_CHAIR_PASS_MIN_LEN	Enter the minimum number of characters required for chairperson passwords. Possible values: **0 – 16**. **0 (default)** means no minimum length. However this setting cannot be applied when the Collaboration Server is in *Ultra Secure Mode*.
NUMERIC_CONF_PASS_MAX_LEN	Enter the maximum number of characters permitted for conference passwords. Possible values: **0 – 16** **16 (default)** - Conference password maximum length is 16 characters.
NUMERIC_CHAIR_PASS_MAX_LEN	Enter the maximum number of characters permitted for chairperson passwords. Possible values: **0 – 16** **16 (default)** - chairperson password maximum length is 16 characters.
NUMERIC_CONF_PASS_DEFAULT_LEN	This flag enables or disables the automatic generation of conference passwords. The length of the automatically generated passwords is determined by the flag value. Possible values: **0 – 16, 6 default** Enter **0** to disable the automatic generation of passwords. Any value other than 0 enables the automatic generation of conference passwords provided the flag *HIDE_CONFERENCE_PASSWORD is set to NO.* If the default is used, in non-secured mode the system will automatically generate conference passwords that contain 6 characters.
NUMERIC_CHAIR_PASS_ DEFAULT _LEN	This flag enables or disables the automatic generation of chairperson passwords. The length of the automatically generated passwords is determined by the flag value. Possible values: **0 – 16, 6 default** Enter **0** to disable the automatic generation of passwords. Any value other than 0 enables the automatic generation of chairperson passwords provided the flag *HIDE_CONFERENCE_PASSWORD is set to NO.* If the default is used, in non-secured mode the system will automatically generate chairperson passwords that contain 6 characters.

If the default password length defined by the NUMERIC_CONF_PASS_DEFAULT_LEN or NUMERIC_CHAIR_PASS_ DEFAULT LEN does not fall within the range defined by the minimum and maximum length an appropriate fault is added to the Faults list.

Appendix A

Disconnection Causes

If a participant was unable to connect to a conference or was disconnected from a conference, the **Connection Status** tab in the *Participant Properties* dialog box indicates the call disconnection cause. In some cases, a possible solution may be displayed.

A video participant who is unable to connect the video channels, but is able to connect as an audio only participant, is referred to as a Secondary participant. For Secondary participants, the **Connection Status** tab in the *Participant Properties* dialog box indicates the video disconnection cause. In some cases, a possible solution may be indicated.

The table below lists the call disconnection causes that can be displayed in the Call Disconnection Cause field and provides an explanation of each message

IP Disconnection Causes

Table A-1 *Call Disconnection Causes*

Disconnection Cause	Description
Disconnected by User	The user disconnected the endpoint from the conference.
Remote device did not open the encryption signaling channel	The endpoint did not open the encryption signaling channel.
Remote devices selected encryption algorithm does not match the local selected encryption algorithm	The encryption algorithm selected by the endpoint does not match the MCU's encryption algorithm.
Resources deficiency	Insufficient resources available.
Call close. Call closed by MCU	The MCU disconnected the call.
H323 call close. No port left for audio	Insufficient audio ports.
H323 call close. No port left for video	The required video ports exceed the number of ports allocated to video in fixed ports.
H323 call close. No port left for FECC	The required data ports exceed the number of ports allocated to data in fixed ports.
H323 call close. No control port left	The required control ports exceed the number of ports allocated to control data in fixed ports.
H323 call close. No port left for videocont	The required video content ports exceed the number of ports allocated to video content in fixed ports.

Table A-1 Call Disconnection Causes (Continued)

Disconnection Cause	Description
H323 call closed. Small bandwidth	The gatekeeper allocated insufficient bandwidth to the connection with the endpoint.
H323 call closed. No port left	There are no free ports left in the IP card.
Caller not registered	The calling endpoint is not registered in the gatekeeper.
H323 call closed. ARQ timeout	The endpoint sent an ARQ message to the gatekeeper, but the gatekeeper did not respond before timeout.
H323 call closed. DRQ timeout	The endpoint sent a DRQ message to the gatekeeper, but the gatekeeper did not respond before timeout.
H323 call closed. Alt Gatekeeper failure	An alternate gatekeeper failure occurred.
H323 call closed. Gatekeeper failure	A gatekeeper failure occurred.
H323 call closed. Remote busy	The endpoint was busy. (Applicable only to dial-out)
H323 call closed. Normal	The call ended normally, for example, the endpoint disconnected.
H323 call closed. Remote reject	The endpoint rejected the call.
H323 call closed. Remote unreachable	The call remained idle for more than 30 seconds and was disconnected because the destination device did not answer. Possible causes can be due to network problems, the gatekeeper could not find the endpoint's address, or the endpoint was busy or unavailable (for example, the "do not disturb" status is selected).
H323 call closed. Unknown reason	The reason for the disconnection is unknown, for example, the endpoint disconnected without giving a reason.
H323 call closed. Faulty destination address	Incorrect address format.
H323 call closed. Small bandwidth	The gatekeeper allocated insufficient bandwidth to the connection with the endpoint.
H323 call closed. Gatekeeper reject ARQ	The gatekeeper rejected the endpoint's ARQ.
H323 call closed. No port left	There are no ports left in the IP card.
H323 call closed. Gatekeeper DRQ	The gatekeeper sent a DRQ.
H323 call closed. No destination IP address	For internal use.
H323 call. Call failed prior or during the capabilities negotiation stage	The endpoint did not send its capabilities to the gatekeeper.
H323 call closed. Audio channels didn't open before timeout	The endpoint did not open the audio channel.

Table A-1 Call Disconnection Causes (Continued)

Disconnection Cause	Description
H323 call closed. Remote sent bad capability	There was a problem in the capabilities sent by the endpoint.
H323 call closed. Local capability wasn't accepted by remote	The endpoint did not accept the capabilities sent by the gatekeeper.
H323 failure	Internal error occurred.
H323 call closed. Remote stop responding	The endpoint stopped responding.
H323 call closed. Master slave problem	A People + Content cascading failure occurred.
SIP bad name	The conference name is incompatible with SIP standards.
SIP bad status	A general IP card error occurred.
SIP busy everywhere	The participant's endpoints were contacted successfully, but the participant is busy and does not wish to take the call at this time.
SIP busy here	The participant's endpoint was contacted successfully, but the participant is currently not willing or able to take additional calls.
SIP capabilities don't match	The remote device capabilities are not compatible with the conference settings.
SIP card rejected channels	The IP card could not open the media channels.
SIP client error 400	The endpoint sent a SIP Client Error 400 (Bad Request) response. The request could not be understood due to malformed syntax.
SIP client error 402	The endpoint sent a SIP Client Error 402 (Payment Required) response.
SIP client error 405	The endpoint sent a SIP Client Error 405 (Method Not Allowed) response. The method specified in the Request-Line is understood, but not allowed for the address identified by the Request-URI.
SIP client error 406	The endpoint sent a SIP Client Error 406 (Not Acceptable) resources. The remote endpoint cannot accept the call because it does not have the necessary responses. The resource identified by the request is only capable of generating response entities that have content characteristics not acceptable according to the Accept header field sent in the request.

Table A-1 Call Disconnection Causes (Continued)

Disconnection Cause	Description
SIP client error 407	The endpoint sent a SIP Client Error 407 (Proxy Authentication Required) response. The client must first authenticate itself with the proxy.
SIP client error 409	The endpoint sent a SIP Client Error 409 (Conflict) response. The request could not be completed due to a conflict with the current state of the resource.
SIP client error 411	The endpoint sent a SIP Client Error 411 (Length Required) response. The server refuses to accept the request without a defined Content Length.
SIP client error 413	The endpoint sent a SIP Client Error 413 (Request Entity Too Large) response. The server is refusing to process a request because the request entity is larger than the server is willing or able to process.
SIP client error 414	The endpoint sent a SIP Client Error 414 (Request-URI Too Long) response. The server is refusing to service the request because the Request-URI is longer than the server is willing to interpret.
SIP client error 420	The endpoint sent a SIP Client Error 420 (Bad Extension) response. The server did not understand the protocol extension specified in a Require header field.
SIP client error 481	The endpoint sent a SIP Client Error 481 (Call/ Transaction Does Not Exist) response.
SIP client error 482	The endpoint sent a SIP Client Error 482 (Loop Detected) response.
SIP client error 483	The endpoint sent a SIP Client Error 483 (Too Many Hops) response.
SIP client error 484	The endpoint sent a SIP Client Error 484 (Address Incomplete) response. The server received a request with a To address or Request-URI that was incomplete.
SIP client error 485	The endpoint sent a SIP Client Error 485 (Ambiguous) response. The address provided in the request (Request-URI) was ambiguous.
SIP client error 488	The endpoint sent a SIP Client Error 488 (Not Acceptable Here) response.

Table A-1 Call Disconnection Causes (Continued)

Disconnection Cause	Description
SIP forbidden	The SIP server rejected the request. The server understood the request, but is refusing to fulfill it.
SIP global failure 603	A SIP Global Failure 603 (Decline) response was returned. The participant's endpoint was successfully contacted, but the participant explicitly does not wish to or cannot participate.
SIP global failure 604	A SIP Global Failure 604 (Does Not Exist Anywhere) response was returned. The server has authoritative information that the user indicated in the Request-URI does not exist anywhere.
SIP global failure 606	A SIP Global Failure 606 (Not Acceptable) response was returned.
SIP gone	The requested resource is no longer available at the Server and no forwarding address is known.
SIP moved permanently	The endpoint moved permanently. The user can no longer be found at the address in the Request-URI.
SIP moved temporarily	The remote endpoint moved temporarily.
SIP not found	The endpoint was not found. The server has definitive information that the user does not exist at the domain specified in the Request-URI.
SIP redirection 300	A SIP Redirection 300 (Multiple Choices) response was returned.
SIP redirection 305	A SIP Redirection 305 (Use Proxy) response was returned. The requested resource MUST be accessed through the proxy given by the Contact field.
SIP redirection 380	A SIP Redirection 380 (Alternative Service) response was returned. The call was not successful, but alternative services are possible.
SIP remote cancelled call	The endpoint canceled the call.
SIP remote closed call	The endpoint ended the call.
SIP remote stopped responding	The endpoint is not responding.
SIP remote unreachable	The endpoint could not be reached.
SIP request terminated	The endpoint terminated the request. The request was terminated by a BYE or CANCEL request.
SIP request timeout	The request was timed out.

Table A-1 Call Disconnection Causes (Continued)

Disconnection Cause	Description
SIP server error 500	The SIP server sent a SIP Server Error 500 (Server Internal Error) response. The server encountered an unexpected condition that prevented it from fulfilling the request.
SIP server error 501	The SIP server sent a SIP Server Error 501 (Not Implemented) response. The server does not support the functionality required to fulfill the request.
SIP server error 502	The SIP server sent a SIP Server Error 502 (Bad Gateway) response. The server, while acting as a gateway or proxy, received an invalid response from the downstream server it accessed in attempting to fulfill the request.
SIP server error 503	The SIP server sent a SIP Server Error 503 (Service Unavailable) response. The server is temporarily unable to process the request due to a temporary overloading or maintenance of the server.
SIP server error 504	The SIP server sent a SIP Server Error 504 (Server Time-out) response. The server did not receive a timely response from an external server it accessed in attempting to process the request.
SIP server error 505	The SIP server sent a SIP Server Error 505 (Version Not Supported) response. The server does not support, or refuses to support, the SIP protocol version that was used in the request.
SIP temporarily not available	The participant's endpoint was contacted successfully but the participant is currently unavailable (e.g., not logged in or logged in such a manner as to preclude communication with the participant).
SIP remote device did not respond in the given time frame	The endpoint did not respond in the given time frame.
SIP trans error TCP Invite	A SIP Invite was sent via TCP, but the endpoint was not found.
SIP transport error	Unable to initiate connection with the endpoint.
SIP unauthorized	The request requires user authentication.
SIP unsupported media type	The server is refusing to service the request because the message body of the request is in a format not supported by the requested resource for the requested method.

Appendix B

Active Alarms

Table B-1 Active Alarms

Alarm Code	Alarm Description
A matching activation key is required. To cancel the upgrade process, reset the Collaboration Server	The system upgrade requires that a valid activation key be entered. If none is available, resetting the Collaboration Server will cancel the upgrade and return the Collaboration Server to the previous version.
A new activation key was loaded. Reset the system.	A new activation key was loaded: Reset the MCU.
A new version was installed. Reset the system.	A new version was installed: Reset the MCU.
Alarm generated by a Central Signaling component	A system alert was generated by a component of the Central Signaling.
Alarm generated by an internal component	A system alert was generated by an internal system component.
Allocation mode was modified	
Automatic reset is unavailable in Safe Mode	The system switches to safe mode if many resets occur during startup. To prevent additional resets, and allow the system to complete the startup process the automatic system resets are blocked.
Backup of audit files is required	If the ENABLE_CYCLIC_FILE_SYSTEM_ALARMS is set to YES and a Cyclic File reaches a file retention time or file storage capacity limit, the user is alerted that audit files need to be backed up.
Backup of CDR files is required	If the ENABLE_CYCLIC_FILE_SYSTEM_ALARMS is set to YES and a Cyclic File reaches a file retention time or file storage capacity limit, the user is alerted that CDR files need to be backed up.
Backup of log files is required	If the ENABLE_CYCLIC_FILE_SYSTEM_ALARMS is set to YES and a Cyclic File reaches a file retention time or file storage capacity limit, the user is alerted that log files need to be backed up.
Central signaling component failure	Possible explanations: • Central signaling component failure; unit type: [NonComponent\CSMngnt\CSH323\CSSIP] • Central signaling component failure; unit type: (invalid: [NonComponent\CSMngnt\CSH323\CSSIP]) • Central signaling component failure - Invalid failure type. Unit id: [id], Type: [NonComponent\CSMngnt\CSH323\CSSIP], Status: [Ok\Failed\Recovered] • Central signaling component failure - Invalid failure type
Central Signaling indicating Faulty status	Central signaling failure detected in IP Network Service.

Table B-1 *Active Alarms (Continued)*

Alarm Code	Alarm Description
Central Signaling indicating Recovery status	
Central Signaling startup failure	Central Signaling component is down.
Conference Encryption Error	
Configuration of external database did not complete.	Check the configuration of the external DB.
CPU IPMC software was not updated.	Turn off the MCU and then turn it on.
CPU slot ID not identified	The CPU slot ID required for Ethernet Settings was not provided by the Shelf Management.
D channel cannot be established	
DEBUG mode enabled	Possible explanations: • System is running in DEBUG mode. • System DEBUG mode initiated. In this mode, additional prints are added and Startup and Recovery Conditions are different then Non Debug Mode. Change the DEBUG_MODE flag value to NO and reset the Collaboration Server.
DEBUG mode flags in use	The system is using the DEBUG CFG flags.
DMA not supported by IDE device	Possible explanations: • DMA (direct memory access) not supported by IDE device: Incompatible flash card / hard disk being used. • Flash card / hard drive are not properly connected to the board / one of the IDE channels is disconnected. • DMA was manually disabled for testing.
DNS configuration error	Check the DNS configuration.
DNS not configured in IP Network Service	Configure the DNS in the IP Network Services.
Encryption Server Error. Failed to generate the encryption key	FIPS 140 test failed while generating the new encryption key.
Error in external database certificate	
Error reading MCU time	Failed to read MCU time configuration file ([status]). Manually configure the MCU Time in the Collaboration Server Web Client or RMX Manager Manager application.
eUserMsgCode_Cs_EdgeServerDnsFailed	
eUserMsgCode_Cs_SipTLS_CertificateHasExpired	
eUserMsgCode_Cs_SipTLS_CertificateSubjNameIsNotValid_Or_DnsFailed	

Table B-1 *Active Alarms (Continued)*

Alarm Code	Alarm Description
eUserMsgCode_Cs_SipTLS_CertificateWillExpireInLessThanAWeek	
eUserMsgCode_Cs_SipTLS_FailedToLoadOrVerifyCertificateFiles	
eUserMsgCode_Cs_SipTLS_RegistrationHandshakeFailure	
eUserMsgCode_Cs_SipTLS_RegistrationServerNotResponding	
Event Mode Conferencing resources deficiency due to inappropriate license. Please install a new license	
External NTP servers failure	The MCU could not connect to any of the defined NTP server for synchronization due to the remote server error or network error or configuration error. Change the configuration of the NTP server.
Failed to access DNS server	Failed to access DNS server.
Failed to configure the Media card IP address	Possible reasons for the failure: • Failure type: [OK Or Not supported. • Does not exist Or IP failure. • Duplicate IP Or DHCP failure. • VLAN failure Or Invalid: [status_Number].
Failed to configure the Users list in Linux	The authentication process did not start. Use the Restore to factory Defaults to recover.
Failed to connect to application server	Possible reasons for the failure: • Failed to connect to application server: • Failed to establish connection to server, url = [url].
Failed to connect to recording device	The MCU could not connect to the defined recording device due to configuration error or network error.
Failed to connect to SIP registrar	Cannot establish connection with SIP registrar.
Failed to create Default Profile	Possible reasons for the failure: • Failed to validate the default Profile. • Failed to add the default Profile. Possible action: • Restore the Collaboration Server configuration from the Backup. • Use the Non-Comprehensive Restore To Factory Defaults operation.
Failed to initialize system base mode	

Table B-1 *Active Alarms (Continued)*

Alarm Code	Alarm Description
Failed to initialize the file system	Possible reasons for the failure: • Failed to initialize the file system. • Failed to initialize the file system and create the CDR index. Reset the MCU.
Failed to open Users list file	Restore the MCU configuration or re-define the user.
Failed to register with DNS server	Check the DNS configuration.
Failed to subscribe with the OCS, therefore the A/V Edge Server URI was not received	
Failure in initialization of SNMP agent.	
Fallback version is being used	Fallback version is being used. Restore current version. Version being used: [running version]; Current version: [current version].
Fan Problem Level Critical	
Fan Problem Level Major	
File error	Possible reasons for the file error: • XML file does not exist [file name]; Error no: [error number]. • Not authorized to open XML file [file name]; Error no: [error number]. • Unknown problem in opening XML file [file name]; Error no: [error number]. • Failed to parse XML file [file name].
File system scan failure	File system scan failure: Failed to scan [file system path]. Multiple occurrences may point to a hardware problem. System is functioning.
File system space shortage	File system space shortage: Out of file system space in [file system path]; Free space: [free space percentage]% ([free space] Blocks) - Minimum free space required: [minimum free space percentage]% ([minimum free space] Blocks).
FIPS 140 failure	
FIPS 140 test result not received	

Table B-1 *Active Alarms (Continued)*

Alarm Code	Alarm Description
Gatekeeper failure	Possible reasons for the Gatekeeper failure: • Failed to register to alternate Gatekeeper. • Gatekeeper discovery state. - Check GK IP address (GUI, ping) • Gatekeeper DNS Host name not found. • Gatekeeper Registration Timeout. • Gatekeeper rejected GRQ due to invalid revision. • Gatekeeper rejected GRQ due to resource unavailability. • Gatekeeper rejected GRQ due to Terminal Exclusion. • Gatekeeper rejected GRQ due to unsupported feature. • Gatekeeper rejected GRQ. Reason 18. • Gatekeeper rejected RRQ due to Discovery Required. • Gatekeeper rejected RRQ due to duplicate alias. - Check duplicate in aliases or in prefixes • Gatekeeper rejected RRQ due to Generic Data. • Gatekeeper rejected RRQ due to invalid alias. • Gatekeeper rejected RRQ due to invalid call signaling address. • Gatekeeper rejected RRQ due to invalid endpoint ID. • Gatekeeper rejected RRQ due to invalid RAS address. • Gatekeeper rejected RRQ due to invalid revision. • Gatekeeper rejected RRQ due to invalid state. • Gatekeeper rejected RRQ due to invalid terminal alias. • Gatekeeper rejected RRQ due to resource unavailability. • Gatekeeper rejected RRQ due to Security Denial. • Gatekeeper rejected RRQ due to terminal type. • Gatekeeper rejected RRQ due to unsupported Additive Registration. • Gatekeeper rejected RRQ due to unsupported feature. • Gatekeeper rejected RRQ due to unsupported QOS transport. • Gatekeeper rejected RRQ due to unsupported transport. • Gatekeeper rejected RRQ. Full registration required. • Gatekeeper rejected RRQ. Reason 18. • Gatekeeper Unregistration State. • Registration succeeded. Check the Gatekeeper configuration.
GUI System configuration file is invalid xml file	The XML format of the system configuration file that contains the user interface settings is invalid.
Hard disk error	Hard disk not responding.
Hot Backup: Master-Slave configuration conflict.	Possible reasons: • When both the MCUs are configured as Master or as Slave • The slave Collaboration Server is defined with the same IP as the Master.
Hot backup: Network issue	

Table B-1 Active Alarms (Continued)

Alarm Code	Alarm Description
Hot Backup: Paired MCU is unreachable.	
Initialization of ice stack failed	
Insufficient resources	The number of resources in the license is higher than the actual system resources. Check to make sure sufficient CPU cores are allocated in the Virtual Machine.
Insufficient UDP Ports	When defining fixed port, the number of defined UDP ports is lower than the required ports. Configure additional ports.
Internal System configuration during startup	System configuration during startup. Wait until Collaboration Server startup is completed.
Invalid System Configuration	
IP addresses of Signaling Host and Control Unit are the same	IP addresses of Signaling Host and Control Unit are identical. Assign different IP addresses to the Signaling Host and Control Unit.
IP Network Service added	
IP Network Service configuration modified	IP Network Service was modified. Reset the MCU.
IP Network Service deleted	IP Network Service was deleted. Reset the MCU.
IP Network Service not found	IP Service not found in the Network Services list. Configure the IP Network Service.
IPMC software upgrade in component	
IPS 140 test result not received	
LDAP TLS: Failed to connect to OSCP responder	
License not found	Possible causes: • The Central Signaling component could not find the IP Services after startup. • During Startup, the resources did not get the License required to utilize their Units. Possible action: • Configure IP service if not configured. • Reset the MCU. • Change the license
Management Network not configured	Configure the Management Network.
Missing Central Signaling configuration	Configure the central signaling.
Missing Central Signaling IP configuration	

Table B-1 *Active Alarms (Continued)*

Alarm Code	**Alarm Description**
MPL startup failure. Authentication not received.	Authentication was not received from Switch. Check the switch card.
MPL startup failure. Management Network configuration not received.	Management Network message was not received. Check the Switch card.
Network interface is not configured. New interface need to be chosen	
Network traffic capture is on	
New certificate for CS need Collaboration Server reset to take effect	
No default IVR Service in IVR Services list	No default IVR Service in IVR Services list. Ensure that one conference IVR Service and one EQ IVR Service are set as default.
No IP Network Services defined	IP Network Service parameters missing. Configure the IP Network Service.
No LAN connection	
No response from Central Signaling	No connection with central signaling.
No RTM-LAN or RTM-ISDN installed. One of these cards must be installed in the RealPresence Collaboration Server (RMX) 4000	
No usable unit for audio controller	No media card is installed, or the media card installed is not functioning. Install the appropriate media card.
OCS Registration failed	
Password expiration warning	
Please install a newer version	
Port configuration was modified	
Power off	
Power Problem Level Critical	
Power Problem Level Major	
Product activation failure	Assign a new activation key.
Product Type mismatch. System is restarting.	The user is alerted to a mismatch between the product type that is stored in MCU software and the product type received from another system component. In such a case the system is automatically restarted.
Received Notification failed	
Recording device has disconnected unexpectedly	

Table B-1 *Active Alarms (Continued)*

Alarm Code	Alarm Description
Requested changes to the certification repository were not completed. Repository must be updated to implement these changes.	
Resource process failed to request the Meeting Room list during startup.	Without the Meeting Rooms list, the system cannot allocate the appropriate dial numbers, Conference ID etc. and therefore cannot run conferences.
Restore Failed	Restoring the system configuration has failed as the system could not locate the configuration file in the selected path, or could not open the file.
Restore Succeeded	Restoring the system configuration has succeeded. Reset the MCU.
Restoring Factory Defaults. Default system settings will be restored once Reset is completed	Default system settings will be restored once Reset is completed.
Collaboration Server fails to connect to Active Directory server.	
Collaboration Server is uploading the version file. To cancel the upload and the upgrade, reset the Collaboration Server	
Collaboration Server user/password list will be reset	
Secured SIP communication failed	Error status (408) received from SIP proxy.
Security mode failed. Certificate has expired.	
Security mode failed. Certificate host name does not match the Collaboration Server host name.	
Security mode failed. Certificate is about to expire.	
Security mode failed. Certificate not yet valid.	
Security mode failed. Error in certificate file.	
Service Request failed	
SIP registrations limit reached	SIP registrations limit reached.
SIP TLS: Certificate has expired	The current TLS certificate files have expired and must be replaced with new files.
SIP TLS: Certificate is about to expire	The current TLS certificate files will expire shortly and will have to be replaced to ensure the communication with the OCS.
SIP TLS: Certificate subject name is not valid or DNS failed to resolve this name	This alarm is displayed if the name of the Collaboration Server in the certificate file is different from the FQDN name defined in the OCS.

Table B-1 *Active Alarms (Continued)*

Alarm Code	Alarm Description
SIP TLS: Failed to load or verify certificate files	This alarm indicates that the certificate files required for SIP TLS could not be loaded to the Collaboration Server. Possible causes are: • Incorrect certificate file name. Only files with the following names can be loaded to the system: rootCA.pem, pkey.pem, cert.pem and certPassword.txt • Wrong certificate file type. Only files of the following types can be loaded to the system: rootCA.pem, pkey.pem and cert.pem and certPassword.txt • The contents of the certificate file does not match the system parameters
SIP TLS: Registration handshake failure	This alarm indicates a mismatch between the security protocols of the OCS and the Collaboration Server, preventing the Registration of the Collaboration Server to the OCS.
SIP TLS: Registration server not responding	This alarm is displayed when the Collaboration Server does not receive a response from the OCS to the registration request in the expected time frame. Possible causes are: • The Collaboration Server FQDN name is not defined in the OCS pool, or is defined incorrectly. • The time frame for the expected response was too short and it will be updated with the next data refresh. The alarm may be cleared automatically the next time the data is refreshed. • The Collaboration Server FQDN name is not defined in the DNS server. Ping the DNS using the Collaboration Server FQDN name to ensure that the Collaboration Server is correctly registered to the DNS.
SIP TLS: Registration transport error	This alarm indicates that the communication with the SIP server cannot be established. Possible causes are: • Incorrect IP address of the SIP server • The SIP server listening port is other than the one defined in the system • The OCS services are stopped
Software upgrade in component	
SSH is enabled	
SWITCH not responding	Check the Switch card.
System Cards MPM Plus mode are not supported in Event mode	
System configuration changed. Please reset the MCU	
System Configuration modified	System configuration flags were modified. Reset the MCU.
System resources of Audio ports usage has exceeded Port Gauge threshold	
System resources of Video ports usage has exceeded Port Gauge threshold	

Table B-1 *Active Alarms (Continued)*

Alarm Code	Alarm Description
System resources usage has exceeded Port Gauge threshold	
Temperature Level - Critical	Possible explanations: • Temperature has reached a critical level.
Temperature Level - Major	Possible explanations: • Temperature has reached a problematic level and requires attention.
The Log file system is disabled because of high system CPU usage	
The MCCF channel is not connected	
The software contains patch(es)	The software contains patch(es).
Unable to connect to Exchange Server.	
User Name SUPPORT cannot be used in Enhanced Security Mode	
Version upgrade is in progress	
Voltage problem	Possible reasons for the problem: • Card voltage problem. • Voltage problem
Warning: Upgrade started and SAFE Upgrade protection is turned OFF	
Yellow Alarm	Problem sending/receiving data from/to network. Check the cables.

Appendix C

CDR Fields - Unformatted File

The CDR (Call Detail Records) utility is used to retrieve conference information to a file. The CDR utility can retrieve conference information to a file in both formatted and unformatted formats.

Unformatted CDR files contain multiple records. The first record in each file contains information about the conference in general, such as the conference name and start time. The remaining records each contain information about one event that occurred during the conference, such as a participant connecting to the conference, or a user extending the length of the conference. The first field in each record identifies the event type, and this is followed by values containing information about the event. The fields are separated by commas.

Formatted files contain basically the same information as unformatted files, but with the field values replaced by descriptions. Formatted files are divided into sections, each containing information about one conference event. The first line in each section is a title describing the type of event, and this is followed by multiple lines, each containing information about the event in the form of a descriptive field name and value.

The field names and values in the formatted file will appear in the language being used for the *Collaboration Server Web Client* user interface at the time when the CDR information is retrieved. The value of the fields that support Unicode values, such as the info fields, will be stored in the CDR file in UTF8. The application that reads the CDR file must support Unicode.

The MCU sends the entire CDR file via API or HTTP, and the Collaboration Server or external application does the processing and sorting. The Collaboration Server ignores events that it does not recognize, that is, events written in a higher version that do not exist in the current version. Therefore, to enable compatibility between versions, instead of adding new fields to existing events, new fields are added as separate events, so as not to affect the events from older versions. This allows users with lower versions to retrieve CDR files that were created in higher versions.

This appendix describes the fields and values in the unformatted CDR records.
Although the formatted files contain basically the same information, in a few instances a single field in the unformatted file is converted to multiple lines in the formatted file, and in other cases, multiple fields in the unformatted file are combined into one line in the formatted file.

In addition, to enable compatibility for applications that were written for the MGC family, the unformatted file contains fields that were supported by the MGC family, but are not supported by the Collaboration Server, whereas these fields are omitted from the formatted file.

The Conference Summary Record

The conference summary record (the first record in the unformatted CDR file) contains the following fields:

Table C-1 *Conference Summary Record Fields*

Field	Description
File Version	The version of the CDR utility that created the file.
Conference Routing Name	The Routing Name of the conference.
Internal Conference ID	The conference identification number as assigned by the system.
Reserved Start Time	The time the conference was scheduled to start in Greenwich Mean Time (GMT). The reservation time of a reservation that was started immediately or of an ongoing conference is the same as the *Actual Start Time*.
Reserved Duration	The amount of time the conference was scheduled to last.
Actual Start Time	The actual time the conference started in GMT.
Actual Duration	The actual conference duration.
Status	The conference status code as follows: **1** - The conference is an ongoing conference. **2** - The conference was terminated by a user. **3** - The conference ended at the scheduled end time. **4** - The conference ended automatically because no participants joined the conference for a predefined time period, or all the participants disconnected from the conference and the conference was empty for a predefined time period. **5** - The conference never started. **6** - The conference could not start due to a problem. **8** - An unknown error occurred. **9** - The conference was terminated by a participant using DTMF codes. **Note:** If the conference was terminated by an MCU reset, this field will contain the value **1** (ongoing conference).
File Name	The name of the conference log file.
GMT Offset Sign	Indicates whether the *GMT Offset* is positive or negative. The possible values are: **0** - Offset is negative. GMT Offset will be subtracted from the GMT Time. **1** - Offset is positive. GMT Offset will be added to the GMT Time.

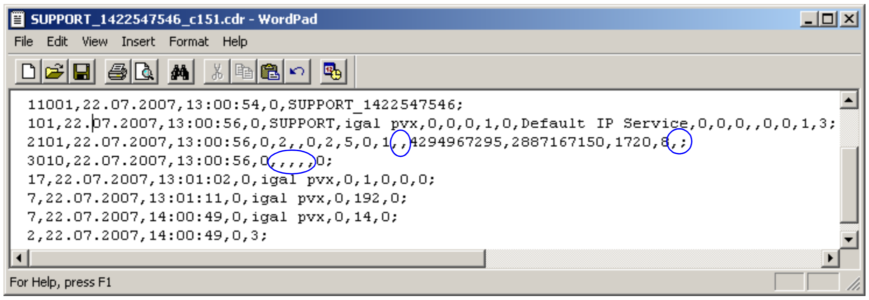
Table C-1 Conference Summary Record Fields (Continued)

Field	Description
GMT Offset	The time zone difference between Greenwich and the Collaboration Server's physical location in hours and minutes. Together with the *GMT Offset Sign* field the *GMT Offset* field is used to define the Collaboration Server local time. For example, if the *GMT Offset Sign* is 0 and *GMT Offset* is 3 hours then the time zone of the Collaboration Server's physical location is -3, which will be subtracted from the GMT time to determine the local time. However, if the *GMT Offset Sign* is 1 and *GMT Offset* is 4 hours then the time zone of the Collaboration Server's physical location is +4, which will be added to the GMT time to determine the local time.
File Retrieved	Indicates if the file has been retrieved and saved to a formatted file, as follows: **0** - No **1** - Yes

Event Records

The event records, that is, all records in the unformatted file except the first record, contain standard fields, such as the event type code and the time stamp, followed by fields that are event specific.

The event fields are separated by commas. Two consecutive commas with nothing between them (,,), or a comma followed immediately by a semi-colon (,;), indicates an empty field, as in the example below:



```
11001,22.07.2007,13:00:54,0,SUPPORT_1422547546;
101,22.07.2007,13:00:56,0,SUPPORT,igal pvx,0,0,0,1,0,Default IP Service,0,0,0,,0,0,1,3;
2101,22.07.2007,13:00:56,0,2,,0,2,5,0,1,,4294967295,2887167150,1720,8,;
3010,22.07.2007,13:00:56,0,,,,,0;
17,22.07.2007,13:01:02,0,igal pvx,0,1,0,0,0;
7,22.07.2007,13:01:11,0,igal pvx,0,192,0;
7,22.07.2007,14:00:49,0,igal pvx,0,14,0;
2,22.07.2007,14:00:49,0,3;
```

Standard Event Record Fields

All event records start with the following fields:

- The CDR event type code. For a list of event type codes and descriptions, refer to Table C-2, "*CDR Event Types,*" on page **C-4**.
- The event date.
- The event time.
- The structure length. This field is required for compatibility purposes, and always contains the value **0**.

Event Types

The table below contains a list of the events that can be logged in the CDR file, and indicates where to find details of the fields that are specific to that type of event.

> The event code identifies the event in the unformatted CDR file, and the event name identifies the event in the formatted CDR file.

Table C-2 *CDR Event Types*

Event Code	Event Name	Description
1	*CONFERENCE START*	The conference started. For more information about the fields, see Table C-3, "*Event Fields for Event 1 - CONFERENCE START,"* on page **C-10**. **Note:** There is one CONFERENCE START event per conference. It is always the first event in the file, after the conference summary record. It contains conference details, but not participant details.
2	*CONFERENCE END*	The conference ended. For more information about the fields, see Table C-8, "*Event Fields for Event 2 - CONFERENCE END,"* on page **C-15**. **Note:** There is one CONFERENCE END event per conference, and it is always the last event in the file.
7	*PARTICIPANT DISCONNECTED*	A participant disconnected from the conference. For more information about the fields, see Table C-11, "*Event Fields for Event 7 - PARTICIPANT DISCONNECTED,"* on page **C-15**.
10	*DEFINED PARTICIPANT*	Information about a defined participant, that is, a participant who was added to the conference before the conference started. For more information about the fields, see Table C-13, "*Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT,"* on page **C-16**. **Note:** There is one event for each participant defined before the conference started.
15	*H323 CALL SETUP*	Information about the IP address of the participant. For more information about the fields, see Table C-16, "*Event fields for Event 15 - H323 CALL SETUP,"* on page **C-19**.
17	*H323 PARTICIPANT CONNECTED*	An H.323 participant connected to the conference. For more information about the fields, see Table C-17, "*Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED,"* on page **C-20**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
18	*NEW UNDEFINED PARTICIPANT*	A new undefined participant joined the conference. For more information about the fields, see Table C-18, "*Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT,"* on page **C-22**.
20	*BILLING CODE*	A billing code was entered by a participant using DTMF codes. For more information about the fields, see Table C-20, "*Event Fields for Event 20 - BILLING CODE,"* on page **C-24**.
21	*SET PARTICIPANT DISPLAY NAME*	A user assigned a new name to a participant, or an end point sent its name. For more information about the fields, see Table C-21, "*Event Fields for Event 21 - SET PARTICIPANT DISPLAY NAME,"* on page **C-25**.
22	*DTMF CODE FAILURE*	An error occurred when a participant entered a DTMF code. For more information about the fields, see Table C-22, "*Event Fields for Event 22 - DTMF CODE FAILURE,"* on page **C-25**.
23	*SIP PARTICIPANT CONNECTED*	A SIP participant connected to the conference. For more information about the fields, see Table C-17, "*Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED,"* on page **C-20**.
26	*RECORDING LINK*	A recording event, such as recording started or recording resumed, occurred. For more information about the fields, see Table C-23, "*Event fields for Event 26 - RECORDING LINK,"* on page **C-25**.
28	*SIP PRIVATE EXTENSIONS*	Contains SIP Private Extensions information. For more information about the fields, see Table C-24, "*Event Fields for Event 28 - SIP PRIVATE EXTENSIONS,"* on page **C-26**.
30	*GATEKEEPER INFORMATION*	Contains the gatekeeper caller ID, which makes it possible to match the CDR in the gatekeeper and in the MCU. For more information about the fields, see Table C-25, "*Event Fields for Event 30 - GATEKEEPER INFORMATION,"* on page **C-26**.
31	*PARTICIPANT CONNECTION RATE*	Information about the line rate of the participant connection. This event is added to the CDR file each time the endpoint changes its connection bit rate. For more information about the fields, see Table C-26, "*Event fields for Event 31 - PARTICIPANT CONNECTION RATE,"* on page **C-26**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
33	*PARTY CHAIR UPDATE*	Participants connect to the conferences as standard participants and they are designated as chairpersons either by entering the chairperson password during the IVR session upon connection, or while participating in the conference using the appropriate DTM code. For more information about the fields, see see "*Event fields for Event 33 - PARTY CHAIR UPDATE"* on page **C-27**.
34	*PARTICIPANT MAXIMUM USAGE INFORMATION*	This event includes information of the maximum line rate, maximum resolution and maximum frame rate used by H.323 or SIP participant during the conference.
35	*SVC SIP PARTICIPANT CONNECTED*	An SVC user connected over SIP. For more information about the fields, see Table C-30, "*Event Fields for Event 35 - SVC SIP PARTICIPANT CONNECTED,"* on page **C-27**.
100	*USER TERMINATE CONFERENCE*	A user terminated the conference. For more information about the fields, see Table C-31, "*Event Fields for Event 100 - USER TERMINATE CONFERENCE,"* on page **C-28**.
101	*USER ADD PARTICIPANT*	A user added a participant to the conference during the conference. For more information about the fields, see Table C-13, "*Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT,"* on page **C-16**.
102	*USER DELETE PARTICIPANT*	A user deleted a participant from the conference. For more information about the fields, see Table C-32, "*Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT,"* on page **C-28**.
103	*USER DISCONNECT PARTICIPANT*	A user disconnected a participant. For more information about the fields, see Table C-32, "*Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT,"* on page **C-28**.
104	*USER RECONNECT PARTICIPANT*	A user reconnected a participant who was disconnected from the conference. For more information about the fields, see Table C-32, "*Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT,"* on page **C-28**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
105	*USER UPDATE PARTICIPANT*	A user updated the properties of a participant during the conference. For more information about the fields, see Table C-13, "*Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT,"* on page **C-16**.
106	*USER SET END TIME*	A user modified the conference end time. For more information about the fields, see Table C-33, "*Event Fields for Event 106 - USER SET END TIME,"* on page **C-29**.
107	*OPERATOR MOVE PARTY FROM CONFERENCE*	The participant moved from an Entry Queue to the destination conference or between conferences. For more information about the fields, see Table C-34, "*Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY,"* on page **C-29**.
108	*OPERATOR MOVE PARTY TO CONFERENCE*	The Collaboration Server User moved the participant from an ongoing conference to another conference. For more information, see Table C-35, "*Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE,"* on page **C-29**.
109	*OPERATOR ATTEND PARTY*	The Collaboration Server User moved the participant to the Operator conference. For more information, see Table C-34, "*Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY,"* on page **C-29**.
111	*OPERATOR BACK TO CONFERENCE PARTY*	The Collaboration Server User moved the participant back to his Home (source) conference. For more information, see Table C-36, "*Event Fields for Event 111 - OPERATOR BACK TO CONFERENCE PARTY,"* on page **C-33**.
112	*OPERATOR ATTEND PARTY TO CONFERENCE*	The Collaboration Server User moved the participant from the Operator conference to another conference. For more information, see Table C-35, "*Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE,"* on page **C-29**.
1001	*NEW UNDEFINED PARTICIPANT CONTINUE 1*	Additional information about a NEW UNDEFINED PARTICIPANT event. For more information about the fields, see Table C-19, "*Event Fields for Event 1001 - NEW UNDEFINED PARTY CONTINUE 1,"* on page **C-24**.

Table C-2 CDR Event Types (Continued)

Event Code	Event Name	Description
2001	*CONFERENCE START CONTINUE 1*	Additional information about a CONFERENCE START event. For more information about the fields, see Table C-4, "*Event Fields for Event 2001 - CONFERENCE START CONTINUE 1,"* on page **C-11**.
2007	*PARTICIPANT DISCONNECTED CONTINUE 1*	Additional information about a PARTICIPANT DISCONNECTED event. For more information about the fields, see Table C-12, "*Event Fields for Event 2007 - PARTICIPANT DISCONNECTED CONTINUE 1,"* on page **C-15**.
2010	*DEFINED PARTICIPANT CONTINUE 1*	Additional information about a DEFINED PARTICIPANT event. For more information about the fields, see Table C-14, "*Event Fields for Events 2010, 2011, 2015 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1,"* on page **C-18**.
2011	*RESERVED PARTICIPANT CONTINUE PV6 ADDRESS*	Additional information about a DEFINED PARTICIPANT event that includes the IPv6 addressing of the defined participant. For more details, see "*Event Fields for Events 2011, 2012, and 2016"* on page **C-33**.
2012	*RESERVED PARTICIPANT CONTINUE 2*	Additional information about a DEFINED PARTICIPANT event. For more information about the fields, see Table C-15, "*Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2012 - USER ADD PARTICIPANT CONTINUE 2, Event 2016 - USER UPDATE PARTICIPANT CONTINUE 2,"* on page **C-19**.
2101	*USER ADD PARTICIPANT CONTINUE 1*	Additional information about a USER ADD PARTICIPANT event. For more information about the fields, see Table C-14, "*Event Fields for Events 2010, 2011, 2015 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1,"* on page **C-18**.
2102	*USER ADD PARTICIPANT CONTINUE 2*	Additional information about a USER ADD PARTICIPANT event. For more information about the fields, see Table C-15, "*Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2012 - USER ADD PARTICIPANT CONTINUE 2, Event 2016 - USER UPDATE PARTICIPANT CONTINUE 2,"* on page **C-19**.
2105	*USER UPDATE PARTICIPANT CONTINUE 1*	Additional information about a USER UPDATE PARTICIPANT event. For more information about the fields, see Table C-14, "*Event Fields for Events 2010, 2011, 2015 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1,"* on page **C-18**.

Table C-2 *CDR Event Types (Continued)*

Event Code	Event Name	Description
2106	*USER UPDATE PARTICIPANT CONTINUE 2*	Additional information about a USER UPDATE PARTICIPANT event. For more information about the fields, see Table C-15, "*Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2012 - USER ADD PARTICIPANT CONTINUE 2, Event 2016 - USER UPDATE PARTICIPANT CONTINUE 2,"* on page **C-19**.
3010	*PARTICIPANT INFORMATION*	The contents of the participant information fields. For more information about the fields, see Table C-38, "*Event Fields for Event 3010 - PARTICIPANT INFORMATION,"* on page **C-33**.
5001	*CONFERENCE START CONTINUE 4*	Additional information about a CONFERENCE START event. For more information about the fields, see Table C-5, "*Event Fields for Event 5001 - CONFERENCE START CONTINUE 4,"* on page **C-13**. **Note:** An additional CONFERENCE START CONTINUE 4 event will be written to the CDR each time the value of one of the following conference fields is modified: • Conference Password • Chairperson Password • Info1, Info2 or Info3 • Billing Info These additional events will only contain the value of the modified field.
6001	*CONFERENCE START CONTINUE 5*	Additional information about a CONFERENCE START event. For more information about the fields, see Table C-6, "*Event Fields for Event 6001 - CONFERENCE START CONTINUE 5,"* on page **C-14**.
11001	*CONFERENCE START CONTINUE 10*	Additional information about a CONFERENCE START event. This event contains the Display Name. For more information about the fields, see Table C-7, "*Event Fields for Event 11001 - CONFERENCE START CONTINUE 10,"* on page **C-14**.

This list only includes events that are supported by the Collaboration Server. For a list of MGC Manager events that are not supported by the Collaboration Server, see "*MGC Manager Events that are not Supported by the Collaboration Server"* on page **C-36**.

Event Specific Fields

The following tables describe the fields which are specific to each type of event.

> Some fields that were supported by the MGC Manager, are not supported by the Collaboration Server. In addition, for some fields the Collaboration Server has a fixed value, whereas the MGC Manager supported multiple values. For more information about the MGC Manager fields and values, see the *MGC Manager User's Guide Volume II*, *Appendix A*.

Table C-3 Event Fields for Event 1 - CONFERENCE START

Field	Description
Dial-Out Manually	Indicates whether the conference was a dial-out manually conference or not. Currently the only value is: **0** - The conference was *not* a dial-out manually conference, that is, the MCU initiates the communication with dial-out participants, and the user does not need to connect them manually.
Auto Terminate	Indicates whether the conference was set to end automatically if no participant joins the conference for a predefined time period after the conference starts, or if all participants disconnect from the conference and the conference is empty for a predefined time period. Possible values are: **0** - The conference was *not* set to end automatically. **1** - The conference was set to end automatically.
Line Rate	The conference line rate, as follows: **0** - 64 kbps **6** - 384 kbps **12** - 1920 kbps **13** - 128 kbps **15** - 256 kbps **23** - 512 kbps **24** - 768 kbps **26** - 1152 kbps **29** - 1472 kbps **32** - 96 kbps
Line Rate (cont.)	**33** - 1024 kbps **34** - 4096 kbps
Restrict Mode	Not supported. Always contains the value **0**.
Audio Algorithm	The audio algorithm. Currently the only value is: **255** - Auto
Video Session	The video session type. Currently the only value is: **3** - Continuous Presence
Video Format	The video format. Currently the only value is: **255** - Auto

Table C-3 Event Fields for Event 1 - CONFERENCE START (Continued)

Field	Description
CIF Frame Rate	The CIF frame rate. Currently the only value is: **255** -Auto
QCIF Frame Rate	The QCIF frame rate: Currently the only value is: **255** - Auto
LSD Rate	Not supported. Always contains the value **0**.
HSD Rate	Not supported. Always contains the value **0**.
T120 Rate	Not supported. Always contains the value **0**.

Table C-4 Event Fields for Event 2001 - CONFERENCE START CONTINUE 1

Field	Description
Audio Tones	Not supported. Always contains the value **0**.
Alert Tone	Not supported. Always contains the value **0**.
Talk Hold Time	The minimum time that a speaker has to speak to become the video source. The value is in units of 0.01 seconds. Currently the only value is **150**, which indicates a talk hold time of 1.5 seconds.
Audio Mix Depth	The maximum number of participants whose audio can be mixed. Currently the only value is **5**.
Operator Conference	Not supported. Always contains the value **0**.
Video Protocol	The video protocol. Currently the only value is: **255** - Auto
Meet Me Per Conference	Indicates the Meet Me Per Conference setting. Currently the only value is: **1** - The Meet Me Per Conference option is enabled, and dial-in participants can join the conference by dialing the dial-in number.
Number of Network Services	Not supported. Always contains the value **0**.

Table C-4 *Event Fields for Event 2001 - CONFERENCE START CONTINUE 1 (Continued)*

Field	Description
Chairperson Password	The chairperson password for the conference. An empty field "" means that no chairperson password was assigned to the conference.
Chair Mode	Not supported. Always contains the value **0**.
Cascade Mode	The cascading mode. Currently the only value is: **0** - None
Master Name	Not supported. This field remains empty.
Minimum Number of Participants	The number of participants for which the system reserved resources. Additional participants may join the conference without prior reservation until all the resources are utilized. Currently the only value is **0**.
Allow Undefined Participants	Indicates whether or not undefined dial-in participants can connect to the conference. Currently the only value is: **1** - Undefined participants can connect to the conference
Time Before First Participant Joins	**Note:** This field is only relevant if the Auto Terminate option is enabled. Indicates the number of minutes that should elapse from the time the conference starts, without any participant connecting to the conference, before the conference is automatically terminated by the MCU.
Time After Last Participant Quits	**Note:** This field is only relevant if the Auto Terminate option is enabled. Indicates the number of minutes that should elapse after the last participant has disconnected from the conference, before the conference is automatically terminated by the MCU.
Conference Lock Flag	Not supported. Always contains the value **0**.
Maximum Number of Participants	The maximum number of participants that can connect to the conference at one time. The value **65535** (auto) indicates that as many participants as the MCU's resources allow can connect to the conference, up to the maximum possible for the type of conference.
Audio Board ID	Not supported. Always contains the value **65535**.
Audio Unit ID	Not supported. Always contains the value **65535**.
Video Board ID	Not supported. Always contains the value **65535**.

Table C-4 Event Fields for Event 2001 - CONFERENCE START CONTINUE 1 (Continued)

Field	Description
Video Unit ID	Not supported. Always contains the value **65535**.
Data Board ID	Not supported. Always contains the value **65535**.
Data Unit ID	Not supported. Always contains the value **65535**.
Message Service Type	The Message Service type. Currently the only value is: **3** - IVR
Conference IVR Service	The name of the IVR Service assigned to the conference. **Note:** If the name of the IVR Service contains more than 20 characters, it will be truncated to 20 characters.
Lecture Mode Type	Indicates the type of Lecture Mode, as follows: **0** - None **1** - Lecture Mode **3** - Presentation Mode
Lecturer	**Note:** This field is only relevant if the Lecture Mode Type is Lecture Mode. The name of the participant selected as the conference lecturer.
Time Interval	**Note:** This field is only relevant if Lecturer View Switching is enabled. The number of seconds a participant is to be displayed in the lecturer window before switching to the next participant. Currently the only value is **15**.
Lecturer View Switching	**Note:** This field is only relevant when Lecture Mode is enabled. Indicates the lecturer view switching setting, as follows: **0** - Automatic switching between participants is disabled. **1** - Automatic switching between participants is enabled.
Audio Activated	Not supported. Always contains the value **0**.
Lecturer ID	Not supported. Always contains the value **4294967295**.

Table C-5 Event Fields for Event 5001 - CONFERENCE START CONTINUE 4

Field	Description
Note: When this event occurs as the result of a change to the value of one of the event fields, the event will only contain the value of the modified field. All other fields will be empty.	
Conference ID	The conference ID.

Table C-5 *Event Fields for Event 5001 - CONFERENCE START CONTINUE 4 (Continued)*

Field	Description
Conference Password	The conference password. An empty field “” means that no conference password was assigned to the conference.
Chairperson Password	The chairperson password. An empty field “” means that no chairperson password was assigned to the conference.
Info1 *Info2* *Info3*	The contents of the conference information fields. These fields enable users to enter general information for the conference, such as the company name, and the contact person's name and telephone number. The maximum length of each field is 80 characters.
Billing Info	The billing code.

Table C-6 *Event Fields for Event 6001 - CONFERENCE START CONTINUE 5*

Field	Description
Encryption	Indicates the conference encryption setting, as follows: **0** - The conference is *not* encrypted. **1** - The conference is encrypted.

Table C-7 *Event Fields for Event 11001 - CONFERENCE START CONTINUE 10*

Field	Description
Display Name	The Display Name of the conference.

Table C-8 Event Fields for Event 2 - CONFERENCE END

Field	Description
Conference End Cause	Indicates the reason for the termination of the conference, as follows: **1** - The conference is an ongoing conference or the conference was terminated by an MCU reset. **2** - The conference was terminated by a user. **3** - The conference ended at the scheduled end time. **4** - The conference ended automatically because no participants joined the conference for a predefined time period, or all the participants disconnected from the conference and the conference was empty for a predefined time period. **5** - The conference never started. **6** - The conference could not start due to a problem. **8** - An unknown error occurred. **9** - The conference was terminated by a participant using DTMF codes.

Table C-11 Event Fields for Event 7 - PARTICIPANT DISCONNECTED

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Call Disconnection Cause	The disconnection cause. For more information about possible values, see Table C-39, "*Disconnection Cause Values,"* on page **C-33**.
Q931 Disconnect Cause	If the disconnection cause is "No Network Connection" or "Participant Hang Up", then this field indicates the Q931 disconnect cause.

Table C-12 Event Fields for Event 2007 - PARTICIPANT DISCONNECTED CONTINUE 1

Field	Description
Rx Synchronization Loss	The number of times that the general synchronization of the MCU was lost.
Tx Synchronization Loss	The number of times that the general synchronization of the participant was lost.
Rx Video Synchronization Loss	The number of times that the synchronization of the MCU video unit was lost.
Tx Video Synchronization Loss	The number of times that the synchronization of the participant video was lost.
Mux Board ID	Not supported. Always contains the value **0**.

Table C-12 Event Fields for Event 2007 - PARTICIPANT DISCONNECTED CONTINUE 1

Field	Description
Mux Unit ID	Not supported. Always contains the value **0**.
Audio Codec Board ID	Not supported. Always contains the value **0**.
Audio Codec Unit ID	Not supported. Always contains the value **0**.
Audio Bridge Board ID	Not supported. Always contains the value **0**.
Audio Bridge Unit ID	Not supported. Always contains the value **0**.
Video Board ID	Not supported. Always contains the value **0**.
Video Unit ID	Not supported. Always contains the value **0**.
T.120 Board ID	Not supported. Always contains the value **0**.
T.120 Unit ID	Not supported. Always contains the value **0**.
T.120 MCS Board ID	Not supported. Always contains the value **0**.
T.120 MCS Unit ID	Not supported. Always contains the value **0**.
H.323 Board ID	Not supported. Always contains the value **0**.
H323 Unit ID	Not supported. Always contains the value **0**.

Table C-13 Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT

Field	Description
User Name	The login name of the user who added the participant to the conference, or updated the participant properties.
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Dialing Direction	The dialing direction, as follows: **0** - Dial-out **5** - Dial-in

Table C-13 Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT (Continued)

Field	Description
Bonding Mode	Not supported. Always contains the value **0**.
Number Of Channels	Not applicable.
Net Channel Width	Not supported. Always contains the value **0**.
Network Service Name	The name of the Network Service. An empty field “” indicates the default Network Service.
Restrict	Not supported. Always contains the value **0**.
Audio Only	Indicates the participant’s Audio Only setting, as follows: **0** - The participant is *not* an Audio Only participant **1** - The participant is an Audio Only participant **255** - Unknown
Default Number Type	**Note:** This field is only relevant to ISDN/PSTN participants. The type of telephone number, as follows: **0** - Unknown **1** - International **2** - National **3** - Network specific **4** - Subscriber **6** - Abbreviated **255** - Taken from Network Service, default **Note:** For dial-in participants, the only possible value is: **255** - Taken from Network Service
Net Sub-Service Name	Not supported. This field remains empty.
Number of Participant Phone Numbers	Not applicable.
Number of MCU Phone Numbers	Not applicable.
Party and MCU Phone Numbers	Not applicable.

Table C-13 Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT (Continued)

Field	Description
Identification Method	**Note:** This field is only relevant to dial-in participants. The method by which the destination conference is identified, as follows: **1** - Called IP address or alias **2** - Calling IP address or alias
Meet Me Method	**Note:** This field is only relevant to dial-in participants. The meet-me per method. Currently the only value is: **3** - Meet-me per participant

Table C-14 Event Fields for Events 2010, 2011, 2015 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1

Field	Description
Network Type	The type of network between the participant and the MCU, as follows: **2** - H.323 **5** - SIP
H.243 Password	Not supported. This field remains empty.
Chair	Not supported. Always contains the value **0**.
Video Protocol	The video protocol used by the participant, as follows: **1** - H.261 **2** - H.263 **4** - H.264 **255** - Auto
Broadcasting Volume	The broadcasting volume assigned to the participant. The value is between **1** (lowest) and **10** (loudest). Each unit movement increases or decreases the volume by **3 dB**.
Undefined Participant	Indicates whether are not the participant is an undefined participant, as follows: **0** - The participant is *not* an undefined participant. **2** - The participant is an undefined participant.
Node Type	The node type, as follows: **0** - MCU **1** - Terminal
Bonding Phone Number	Not applicable.
Video Bit Rate	The video bit rate in units of kilobits per second. A value of **4294967295** denotes auto, and in this case, the rate is computed by the MCU.

Table C-14 *Event Fields for Events 2010, 2011, 2015 - DEFINED PARTICIPANT CONTINUE 1, USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1*

Field	Description
IP Address	The IP address of the participant. An address of **4294967295** indicates that no IP address was specified for the participant, and the gatekeeper is used for routing. In all other cases the address overrides the gatekeeper.
Signaling Port	The signaling port used for participant connection.
H.323 Participant Alias Type/SIP Participant Address Type	For H.323 participants, the alias type, as follows: **7** - E164 **8** - H.323 ID **13** - Email ID **14** - Participant number For SIP participants, the address type, as follows: **1** - SIP URI **2** - Tel URL
H.323 Participant Alias Name/SIP Participant Address	For H.323 participants: The participant alias. The alias may contain up to 512 characters. For SIP participants: The participant address. The address may contain up to 80 characters.

Table C-15 *Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2012 - USER ADD PARTICIPANT CONTINUE 2, Event 2016 - USER UPDATE PARTICIPANT CONTINUE 2*

Field	Description
Encryption	Indicates the participant's encryption setting as follows: **0** - The participant is *not* encrypted. **1** - The participant is encrypted. **2** - Auto. The conference encryption setting is applied to the participant.
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.

Table C-16 *Event fields for Event 15 - H323 CALL SETUP*

Field	Description
Participant Name	The name of the participant.

Table C-16 Event fields for Event 15 - H323 CALL SETUP (Continued)

Field	Description
Participant ID	The identification number assigned to the participant by the MCU.
Connect Initiator	Indicates who initiated the connection, as follows: **0** - MCU **1** - Remote participant **Any other number** - Unknown
Min Rate	The minimum line rate used by the participant. The data in this field should be ignored. For accurate rate information, see CDR event 31.
Max Rate	The maximum line rate achieved by the participant. The data in this field should be ignored. For accurate rate information, see CDR event 31.
Source Party Address	The IP address of the calling participant. A string of up to 255 characters.
Destination Party Address	The IP address of the called participant. A string of up to 255 characters.
Endpoint Type	The endpoint type, as follows: **0** - Terminal **1** - Gateway **2** - MCU **3** - Gatekeeper **4** - Undefined

Table C-17 Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED

Field	Description
Participant Name	The name of the participant. An empty field “” denotes an unidentified participant or a participant whose name is unspecified.
Participant ID	The identification number assigned to the participant by the MCU.

Table C-17 *Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED (Continued)*

Field	Description
Participant Status	The participant status, as follows: **0** - Idle **1** - Connected **2** - Disconnected **3** - Waiting for dial-in **4** – Connecting **5** - Disconnecting **6** - Partially connected. Party has completed H.221 capability exchange **7** - Deleted by a user **8** - Secondary. The participant could not connect the video channels and is connected via audio only **10** - Connected with problem **11** - Redialing
Capabilities	Not supported. Always contains the value **0**.
Remote Communication Mode	Not supported. Always contains the value **0**.
Secondary Cause	**Note:** This field is only relevant if the Participant Status is Secondary. The cause for the secondary connection (not being able to connect the video channels), as follows: **0** - Default **11** - The incoming video parameters are not compatible with the conference video parameters **13** - The conference video settings are not compatible with the endpoint capabilities **14** - The new conference settings are not compatible with the endpoint capabilities **15** - Video stream violation due to incompatible annexes or other discrepancy **16** - Inadequate video resources **17** - When moved to a Transcoding or Video Switching conference, the participant’s video capabilities are not supported by the video cards **18** - Video connection could not be established **24** - The endpoint closed its video channels **25** - The participant video settings are not compatible with the conference protocol **26** - The endpoint could not re-open the video channel after the conference video mode was changed **27** - The gatekeeper approved a lower bandwidth than requested **28** - Video connection for the SIP participant is temporarily unavailable **255** - Other

Table C-18 Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Dialing Direction	The dialing direction, as follows **0** - Dial-out **5** - Dial-in
Bonding Mode	Not supported. Always contains the value **0**.
Number of Channels	Not applicable
Net Channel Width	Not supported. Always contains the value **0**.
Network Service Name	The name of the Network Service. An empty field “” indicates the default Network Service.
Restrict	Not supported. Always contains the value **0**.
Audio Only	Indicates the participant's Audio Only setting, as follows: **0** - The participant is *not* an Audio Only participant **1** - The participant is an Audio Only participant **255** - Unknown
Default Number Type	Not applicable.
Net Sub-Service Name	Not supported. This field remains empty.
Number of Participant Phone Numbers	Not applicable.
Number of MCU Phone Numbers	Not applicable.
Party and MCU Phone Numbers	Not applicable.
Identification Method	**Note:** This field is only relevant to dial-in participants. The method by which the destination conference is identified, as follows: **1** - Called IP address or alias **2** - Calling IP address or alias
Meet Me Method	**Note:** This field is only relevant to dial-in participants. The meet-me per method, as follows: **3** - Meet-me per participant

Table C-18 *Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT (Continued)*

Field	Description
Network Type	The type of network between the participant and the MCU, as follows: **2** - H.323 **5** - SIP
H.243 Password	Not supported. This field remains empty.
Chair	Not supported. Always contains the value **0**.
Video Protocol	The video protocol, as follows: **1** - H.261 **2** - H.263 **4** - H.264 **255** - Auto
Broadcasting Volume	The broadcasting volume assigned to the participant. The value is between **1** (lowest) and **10** (loudest). Each unit movement increases or decreases the volume by **3 dB**.
Undefined Participant	Indicates whether are not the participant is an undefined participant, as follows: **0** - The participant is *not* an undefined participant. **2** - The participant is an undefined participant.
Node Type	The node type, as follows: **0** - MCU **1** - Terminal
Bonding Phone Number	**Note:** This field is only relevant to ISDN/PSTN participants. The phone number for Bonding dial-out calls. Bonding is a communication protocol that aggregates from two up to thirty 64 Kbps B channels together, to look like one large bandwidth channel.
Video Bit Rate	The video bit rate in units of kilobits per second. A value of **4294967295** denotes auto, and in this case, the rate is computed by the MCU.
IP Address	**Note:** This field is only relevant to IP participants. The IP address of the participant. An address of **4294967295** indicates that no IP address was specified for the participant, and the gatekeeper is used for routing. In all other cases the address overrides the gatekeeper.
Signaling Port	**Note:** This field is only relevant to IP participants. The signaling port used for participant connection. A value of **65535** is ignored by MCU.

Table C-18 *Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT (Continued)*

Field	Description
H.323 Participant Alias Type/SIP Participant Address Type	For H.323 participants, the alias type, as follows: **7** - E164 **8** - H.323 ID **13** - Email ID **14** - Participant number For SIP participants, the address type, as follows: **1** - SIP URI **2** - Tel URL
H.323 Participant Alias Name/SIP Participant Address	For H.323 participants: The participant alias. The alias may contain up to 512 characters. For SIP participants: The participant address. The address may contain up to 80 characters.

Table C-19 *Event Fields for Event 1001 - NEW UNDEFINED PARTY CONTINUE 1*

Field	Description
Encryption	Indicates the participant's encryption setting as follows: **0** - The participant is *not* encrypted. **1** - The participant is encrypted. **2** - Auto. The conference encryption setting is applied to the participant.
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.

Table C-20 *Event Fields for Event 20 - BILLING CODE*

Field	Description
Participant Name	The name of the participant who added the billing code.
Participant ID	The identification number, as assigned by the MCU, of the participant who added the billing code.
Billing Info	The numeric billing code that was added (32 characters).

Table C-21 *Event Fields for Event 21 - SET PARTICIPANT DISPLAY NAME*

Field	**Description**
Participant Name	The original name of the participant, for example, the name automatically assigned to an undefined participant, such as, “<conference name>_(000)”.
Participant ID	The identification number assigned to the participant by the MCU.
Display Name	The new name assigned to the participant by the user, or the name sent by the end point.

Table C-22 *Event Fields for Event 22 - DTMF CODE FAILURE*

Field	**Description**
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Incorrect Data	The incorrect DTMF code entered by the participant, or an empty field “” if the participant did not press any key.
Correct Data	The correct DTMF code, if known.
Failure Type	The type of DTMF failure, as follows: **2** - The participant did not enter the correct conference password. **6** - The participant did not enter the correct chairperson password. **12** - The participant did not enter the correct Conference ID.

Table C-23 *Event fields for Event 26 - RECORDING LINK*

Field	**Description**
Participant Name	The name of the Recording Link participant.
Participant ID	The identification number assigned to the Recording Link participant by the MCU.
Recording Operation	The type of recording operation, as follows: **0** - Start recording **1** - Stop recording **2** - Pause recording **3** - Resume recording **4** - Recording ended **5** - Recording failed
Initiator	Not supported.

***Table C-23** Event fields for Event 26 - RECORDING LINK (Continued)*

Field	Description
Recording Link Name	The name of the Recording Link.
Recording Link ID	The Recording Link ID.
Start Recording Policy	The start recording policy, as follows: **1** - Start recording automatically as soon as the first participant connects to the conference. **2** - Start recording when requested by the conference chairperson via DTMF codes or from the *Collaboration Server Web Client*, or when the operator starts recording from the *Collaboration Server Web Client*.

***Table C-24** Event Fields for Event 28 - SIP PRIVATE EXTENSIONS*

Field	Description
Participant Name	The name of the participant.
Participant ID	The participant’s identification number as assigned by the system.
Called Participant ID	The called participant ID.
Asserted Identity	The identity of the user sending a SIP message as it was verified by authentication.
Charging Vector	A collection of charging information.
Preferred Identity	The identity the user sending the SIP message wishes to be used for the P-Asserted-Header field that the trusted element will insert.

***Table C-25** Event Fields for Event 30 - GATEKEEPER INFORMATION*

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Gatekeeper Caller ID	The caller ID in the gatekeeper records. This value makes it possible to match the CDR in the gatekeeper and in the MCU.

***Table C-26** Event fields for Event 31 - PARTICIPANT CONNECTION RATE*

Field	Description
Participant Name	The participant name.

Table C-26 Event fields for Event 31 - PARTICIPANT CONNECTION RATE (Continued)

Field	Description
Participant ID	The identification number assigned to the participant by the MCU.
Participant Current Rate	The participant line rate in Kbps.

Table C-28 Event fields for Event 33 - PARTY CHAIR UPDATE

Field	Description
Participant Name	The participant name.
Participant ID	The identification number assigned to the participant by the MCU.
Chairperson	Possible values: • True - participant is a chairperson • False - Participant is not a chairperson participant (is a standard participant)

Table C-29 Event fields for Event 34 - PARTICIPANT MAXIMUM USAGE INFORMATION

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.
Maximum Bit Rate	The maximum bit rate used by the participant during the call.
Maximum Resolution	The maximum resolution used by the participant during the call. **Note:** The reported resolutions are: CIF, SD, HD720, and HD1080. Other resolutions are roundED up to the nearest resolution. For example, 2SIF is reported as SD resolution.
Maximum Frame Rate	The maximum frame rate used by the participant during the call.
Participant Address	For H.323 participants, the participant alias. The alias may contain up to 512 characters. For SIP participants, the participant address. The address may contain up to 80 characters.

Table C-30 Event Fields for Event 35 - SVC SIP PARTICIPANT CONNECTED

Field	Description
Participant Name	The name of the participant. An empty field "" denotes an unidentified participant or a participant whose name is unspecified

Table C-30 Event Fields for Event 35 - SVC SIP PARTICIPANT CONNECTED (Continued)

Field	Description
Participant ID	The identification number assigned to the participant by the MCU.
Participant Status	The participant status, as follows: 0 - Idle 1 - Connected 2 - Disconnected 3 - Waiting for dial-in 4 - Connecting 5 - Disconnecting 6 - Partially connected. Party has completed H.221 capability exchange 7 - Deleted by a user 8 - Secondary. The participant could not connect the video channels and is connected via audio only 10 - Connected with problem 11 - Redialing
Receive line rate	Negotiated reception line rate
Transmit line rate	Negotiated transmission line rate
Uplink Video Capabilities	a.Number of uplink streams b.Video stream (multiple streams) i.Resolution width ii.resolution height iii.max frame rate iv.max line rate
Audio Codec	SAC, Other
Secondary Cause	

Table C-31 Event Fields for Event 100 - USER TERMINATE CONFERENCE

Field	Description
Terminated By	The login name of the user who terminated the conference.

Table C-32 Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT

Field	Description
User Name	The login name of the user who reconnected the participant to the conference, or disconnected or deleted the participant from the conference.
Participant Name	The name of the participant reconnected to the conference, or disconnected or deleted from the conference.
Participant ID	The identification number assigned to the participant by the MCU.

Table C-33 Event Fields for Event 106 - USER SET END TIME

Field	Description
New End Time	The new conference end time set by the user, in GMT time.
User Name	The login name of the user who changed the conference end time.

Table C-34 Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY

Field	Description
Operator Name	The login name of the user who moved the participant.
Party Name	The name of the participant who was moved.
Party ID	The identification number of the participant who was moved, as assigned by the MCU.
Destination Conf Name	The name of the conference to which the participant was moved.
Destination Conf ID	The identification number of the conference to which the participant was moved.

Table C-35 Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE

Field	Description
Operator Name	The login name of the operator who moved the participant to the conference.
Source Conf Name	The name of the source conference.
Source Conf ID	The identification number of the source conference, as assigned by the MCU.
Party Name	The name of the participant who was moved.
Party ID	The identification number assigned to the participant by the MCU.
Connection Type	The connection type, as follows: **0** - Dial-out **5** - Dial-in
Bonding Mode	Not applicable.
Number Of Channels	**Note:** This field is only relevant to ISDN/PSTN participants. The number of channels, as follows: **255** - Auto Otherwise, in range of **1** - **30**

Table C-35 *Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE (Continued)*

Field	Description
Net Channel Width	The bandwidth of each channel. This value is always **0**, which represents a bandwidth of **1B**, which is the only bandwidth that is currently supported.
Net Service Name	The name of the Network Service. An empty field "" indicates the default Network Service.
Restrict	Indicates whether or not the line is restricted, as follows: **27** - Restricted line **28** - Non restricted line **255** - Unknown or not relevant
Voice Mode	Indicates whether or not the participant is an Audio Only participant, as follows: **0** - The participant is *not* an Audio Only participant **1** - The participant is an Audio Only participant **255** - Unknown
Number Type	**Note:** This field is only relevant to dial-out, ISDN/PSTN participants. The type of telephone number, as follows: **0** - Unknown **1** - International **2** - National **3** - Network specific **4** - Subscriber **6** - Abbreviated **255** - Taken from Network Service, default
Net SubService Name	**Note:** This field is only relevant to dial-out, ISDN/PSTN participants. The network sub-service name. An empty field "" means that MCU selects the default sub-service.
Number of Party Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The number of participant phone numbers. In a dial-in connection, the participant phone number is the CLI (Calling Line Identification) as identified by the MCU. In a dial-out connection, participant phone numbers are the phone numbers dialed by the MCU for each participant channel.
Number of MCU Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The number of MCU phone numbers. In a dial-in connection, the MCU phone number is the number dialed by the participant to connect to the MCU. In a dial-out connection, the MCU phone number is the MCU (CLI) number as seen by the participant.

Table C-35 *Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE (Continued)*

Field	Description
Party and MCU Phone Numbers	**Note:** This field is only relevant to ISDN/PSTN participants. The participant phone numbers are listed first, followed by the MCU phone numbers.
Ident. Method	**Note:** This field is only relevant to dial-in participants. The method by which the destination conference is identified, as follows: **0** - Password **1** - Called phone number, or IP address, or alias **2** - Calling phone number, or IP address, or alias
Meet Method	**Note:** This field is only relevant to dial-in participants. The meet-me per method, as follows: **1** - Meet-me per MCU-Conference **3** - Meet-me per participant **4** - Meet-me per channel
Net Interface Type	The type of network interface between the participant and the MCU, as follows: **0** - ISDN **2** - H.323 **5** - SIP
H243 Password	The H.243 password, or an empty field “” if there is no password.
Chair	Not supported. Always contains the value **0**.
Video Protocol	The video protocol, as follows: **1** - H.261 **2** - H.263 **3** - H.264* **4** - H.264 **255** - Auto
Audio Volume	The broadcasting volume assigned to the participant. The value is between **1** (lowest) and **10** (loudest).
Undefined Type	The participant type, as follows: **0** - Defined participant. (The value in the formatted text file is “default”.) **2** - Undefined participant. (The value in the formatted text file is “Unreserved participant “.)
Node Type	The node type, as follows: **0** - MCU **1** - Terminal
Bonding Phone Number	**Note:** This field is only relevant to ISDN/PSTN participants. The phone number for Bonding dial-out calls.

Table C-35 Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE, OPERATOR ATTEND PARTY TO CONFERENCE (Continued)

Field	Description
Video Rate	**Note:** This field is only relevant to IP participants. The video rate in units of kilobits per second. A value of **4294967295** denotes auto, and in this case, the rate is computed by the MCU.
IP Address	**Note:** This field is only relevant to IP participants. The IP address of the participant. An address of **4294967295** indicates that no IP address was specified for the participant, and the gatekeeper is used for routing. In all other cases the address overrides the gatekeeper.
Call Signaling Port	**Note:** This field is only relevant to IP participants. The signaling port used for participant connection. A value of **65535** is ignored by MCU.
H.323 Party Alias Type/SIP Party Address Type	**Note:** This field is only relevant to IP participants. For H.323 participants, the alias type, as follows: **7** - E164 **8** - H.323 ID **11** - URL ID alias type **12** - Transport ID **13** - Email ID **14** - Participant number For SIP participants, the address type, as follows: **1** - SIP URI **2** - Tel URL
H.323 Party Alias/ SIP Party Address	**Note:** This field is only relevant to IP participants. For H.323 participants, the participant alias. The alias may contain up to 512 characters. For SIP participants, the participant address. The address may contain up to 80 characters.

Table C-36 Event Fields for Event 111 - OPERATOR BACK TO CONFERENCE PARTY

Field	Description
Operator Name	The login name of the operator moving the participant back to the conference.
Party Name	The name of the participant being moved.
Party ID	The identification number, as assigned by the MCU, of the participant being moved.

Table C-37 Event Fields for Events 2011, 2012, and 2016

Field	Description
IP V6	IPv6 address of the participant's endpoint.

Table C-38 Event Fields for Event 3010 - PARTICIPANT INFORMATION

Field	Description
Info1 *Info2* *Info3* *Info4*	The participant information fields. These fields enable users to enter general information about the participant, such as the participant's e-mail address. The maximum length of each field is 80 characters.
VIP	Not supported. Always contains the value **0**.

Disconnection Cause Values

For an explanation of the disconnection causes, see *Appendix A: "Disconnection Causes"* on page **A-1**.

Table C-39 Disconnection Cause Values

Value	Call Disconnection Cause
0	Unknown
1	Participant hung up
2	Disconnected by User
5	Resources deficiency
6	Password failure
20	H323 call close. No port left for audio

Table C-39 Disconnection Cause Values (Continued)

Value	Call Disconnection Cause
21	H323 call close. No port left for video
22	H323 call close. No port left for FECC
23	H323 call close. No control port left
25	H323 call close. No port left for video content
51	A common key exchange algorithm could not be established between the MCU and the remote device
53	Remote device did not open the encryption signaling channel
59	The remote devices' selected encryption algorithm does not match the local selected encryption algorithm
141	Called party not registered
145	Caller not registered
152	H323 call close. ARQ timeout
153	H323 call close. DRQ timeout
154	H323 call close. Alt Gatekeeper failure
191	H323 call close. Remote busy
192	H323 call close. Normal
193	H323 call close. Remote reject
194	H323 call close. Remote unreachable
195	H323 call close. Unknown reason
198	H323 call close. Small bandwidth
199	H323 call close. Gatekeeper failure
200	H323 call close. Gatekeeper reject ARQ
201	H323 call close. No port left
202	H323 call close. Gatekeeper DRQ
203	H323 call close. No destination IP value
204	H323 call close. Remote has not sent capability
205	H323 call close. Audio channels not open
207	H323 call close. Bad remote cap
208	H323 call close. Capabilities not accepted by remote
209	H323 failure
210	H323 call close. Remote stop responding

Table C-39 Disconnection Cause Values (Continued)

Value	Call Disconnection Cause
213	H323 call close. Master slave problem
251	SIP timer popped out
252	SIP card rejected channels
253	SIP capabilities don’t match
254	SIP remote closed call
255	SIP remote cancelled call
256	SIP bad status
257	SIP remote stopped responding
258	SIP remote unreachable
259	SIP transport error
260	SIP bad name
261	SIP trans error TCP invite
300	SIP redirection 300
301	SIP moved permanently
302	SIP moved temporarily
305	SIP redirection 305
380	SIP redirection 380
400	SIP client error 400
401	SIP unauthorized
402	SIP client error 402
403	SIP forbidden
404	SIP not found
405	SIP client error 405
406	SIP client error 406
407	SIP client error 407
408	SIP request timeout
409	SIP client error 409
410	SIP gone
411	SIP client error 411
413	SIP client error 413
414	SIP client error 414

Table C-39 Disconnection Cause Values (Continued)

Value	Call Disconnection Cause
415	SIP unsupported media type
420	SIP client error 420
480	SIP temporarily not available
481	SIP client error 481
482	SIP client error 482
483	SIP client error 483
484	SIP client error 484
485	SIP client error 485
486	SIP busy here
487	SIP request terminated
488	SIP client error 488
500	SIP server error 500
501	SIP server error 501
502	SIP server error 502
503	SIP server error 503
504	SIP server error 504
505	SIP server error 505
600	SIP busy everywhere
603	SIP global failure 603
604	SIP global failure 604
606	SIP global failure 606

MGC Manager Events that are not Supported by the Collaboration Server

The following MGC Manager events are not supported by the Collaboration Server:

For details of these events see the M*GC Manager User's Guide Volume II*, *Appendix A*.

- Event 8 - REMOTE COM MODE
- Event 11 - ATM CHANNEL CONNECTED
- Event 12 - ATM CHANNEL DISCONNECTED

- Event 13 - MPI CHANNEL CONNECTED
- Event 14 - MPI CHANNEL DISCONNECTED
- Event 15 - H323 CALL SETUP
- Event 16 - H323 CLEAR INDICATION
- Event 24 - SIP CALL SETUP
- Event 25 - SIP CLEAR INDICATION
- Event 27 - RECORDING SYSTEM LINK
- Event 110 - OPERATOR ON HOLD PARTY
- Event 113 - CONFERENCE REMARKS
- Event 2108 - OPERATOR MOVE PARTY TO CONFERENCE CONTINUE 1
- Event 3001 - CONFERENCE START CONTINUE 2
- Event 3108 - OPERATOR MOVE PARTY TO CONFERENCE CONTINUE 2
- Event 4001 - CONFERENCE START CONTINUE 3
- Event 4108 - OPERATOR MOVE PARTY TO CONFERENCE CONTINUE 3

Appendix D

Ad Hoc Conferencing and External Database Authentication

The *RealPresence Collaboration Server Virtual Edition* Ad Hoc conferencing feature enables participants to start ongoing conferences on-the-fly, without prior definition when dialing an Ad Hoc-enabled Entry Queue. The created conference parameters are taken from the Profile assigned to the Ad Hoc-enabled Entry Queue.

Ad Hoc conferencing is available in two the following modes:

- Ad Hoc Conferencing without Authentication

  Any participant can dial into an Entry Queue and initiate a new conference if the conference does not exist. This mode is usually used for the organization's internal Ad Hoc conferencing.

- **Ad Hoc Conferencing with External Database Authentication**

  In this mode, the participant's right to start a new conference is validated against a database.

The external database application can also be used to validate the participant's right to join an ongoing conference. Conference access authentication can be:

- Part of the Ad Hoc conferencing flow where the participants must be authorized before they can enter the conference created in the Ad Hoc flow.
- Independent of Ad Hoc conferencing where conference access is validated for all conferences running on the MCU regardless of the method in which the conference was started.

Ad Hoc Conferencing without Authentication

A participant dials in to an Ad Hoc-enabled Entry Queue and starts a new conference based on the Profile assigned to the Entry Queue. In this configuration, any participant connecting to the Entry Queue can start a new conference, and no security mechanism is applied. This mode is usually used in organizations where Ad Hoc conferences are started from within the network and without security breach.

A conference is started using one of the following method:

1. The participant dials in to the Ad Hoc-enabled Entry Queue.
2. The Conference ID is requested by the system.
3. The participant inputs a Conference ID via his/her endpoint remote control using DTMF codes.

4 The MCU checks whether a conference with the same Conference ID is running on the MCU. If there is such a conference, the participant is moved to that conference. If there is no ongoing conference with that Conference ID, the system creates a new conference, based on the Profile assigned to the Entry Queue, and connects this participant as the conference chairperson.

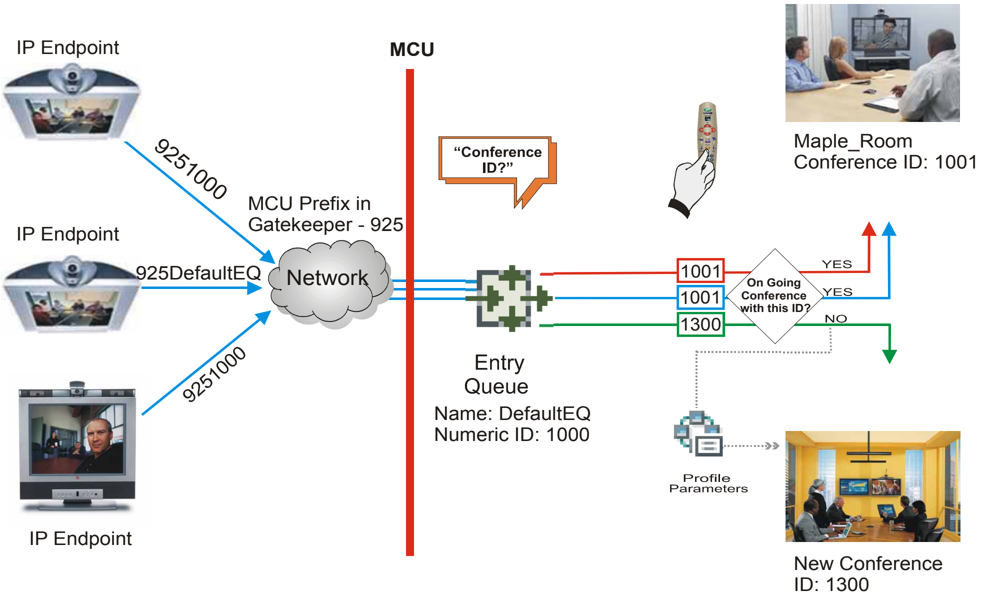


Figure D-1 Ad Hoc Conference Initiation without Authentication

To enable this workflow, the following components must be defined in the system:

- An Entry Queue IVR Service with the appropriate audio file requesting the Conference ID
- An Ad Hoc-enabled Entry Queue with an assigned Profile

Ad Hoc Conferencing with Authentication

The MCU can work with an external database application to validate the participant's right to start a new conference. The external database contains a list of participants, with their assigned parameters. The conference ID entered by the participant is compared against the database. If the system finds a match, the participant is granted the permission to start a new conference.

To work with an external database application to validate the participant's right to start a new conference, the Entry Queue IVR Service must be configured to use the external database application for authentication. In the external database application, you must define all participants (users) with rights to start a new conference using Ad Hoc conferencing. For each user defined in the database, you enter the conference ID, Conference Password (optional) and Chairperson Password (when applicable), billing code, Conference general information (corresponding to the User Defined 1 field in the Profile properties) and user's PIN code. The same user definitions can be used for conference access authentication, that is, to determine who can join the conference as a participant and who as a chairperson.

Entry Queue Level - Conference Initiation Validation with an External Database Application

Starting a new conference with external database application validation entails the following steps:

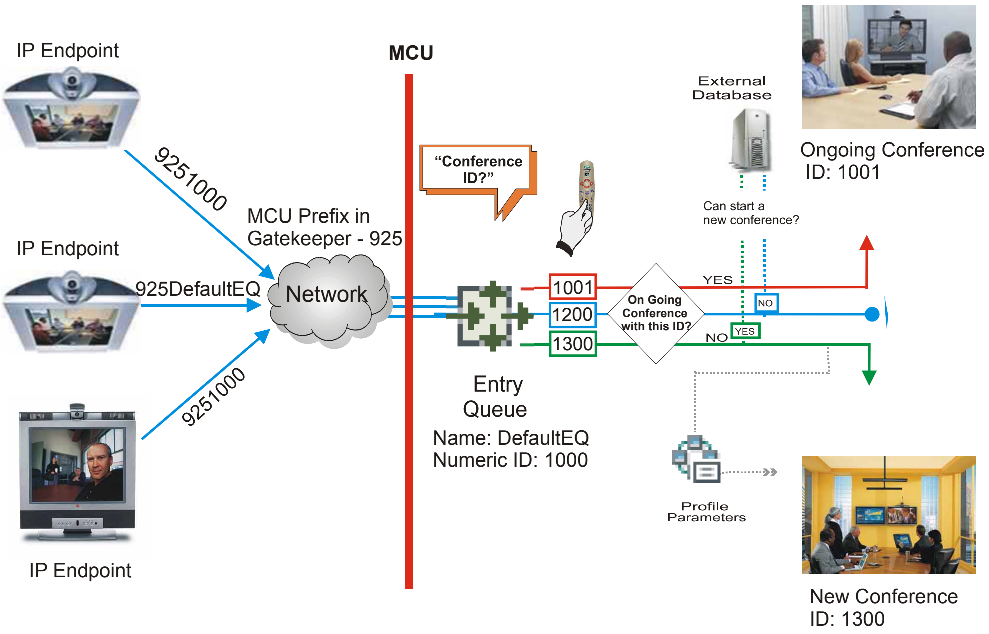


Figure D-2 Conference Initiation Validation with External Database Application

1. The participant dials in to an Ad Hoc-enabled Entry Queue.
2. The participant is requested to enter the Conference ID.
3. The participant enters the conference ID via his/her endpoint remote control using DTMF codes. If there is an ongoing conference with this Conference ID, the participant is moved to that conference where another authentication process can occur, depending on the IVR Service configuration.
4. If there is no ongoing conference with that Conference ID, the MCU verifies the Conference ID with the database application that compares it against its database. If the database application finds a match, the external database application sends a response back to the MCU, granting the participant the right to start a new ongoing conference.

   If this Conference ID is not registered in the database, the conference cannot be started and this participant is disconnected from the Entry Queue.
5. The external database contains a list of participants (users), with their assigned parameters. Once a participant is identified in the database (according to the conference ID), his/her parameters (as defined in the database) can be sent to the MCU in the same response granting the participant the right to start a new ongoing conference. These parameters are:

   — Conference Name

   — Conference Billing code

   — Conference Password

   — Chairperson Password

— Conference Information, such as the contact person name. These fields correspond to Info 1, 2 and 3 fields in the *Conference Properties - Information* dialog box.

— Maximum number of participants allowed for the conference

— Conference Owner

These parameters can also be defined in the conference Profile. In such a case, parameters sent from the database overwrite the parameters defined in the Profile. If these parameters are not sent from the external database to the MCU, they will be taken from the Profile.

6. A new conference is started based on the Profile assigned to the Entry Queue.
7. The participant is moved to the conference.
   If no password request is configured in the Conference IVR Service assigned to the conference, the participant that initiated the conference is directly connected to the conference, as its chairperson.

   If the Conference IVR Service assigned to the conference is configured to prompt for the conference password and chairperson password, without external database authentication, the participant has to enter these passwords in order to join the conference.

To enable this workflow, the following components must be defined in the system:

- A Conference IVR Service with the appropriate prompts. If conference access is also validated with the external database application it must be configured to access the external database for authentication.
- An Entry Queue IVR Service configured with the appropriate audio prompts requesting the Conference ID and configured to access the external database for authentication.
- Create a Profile with the appropriate conference parameters and the appropriate Conference IVR Service assigned to it.
- An Ad Hoc-enabled Entry Queue with the appropriate Entry Queue IVR Service and Conference Profile assigned to it.
- An external database application with a database containing Conference IDs associated with participants and their relevant properties.
- Define the flags required to access the external database in System Configuration.

  For more information, see Figure , "*MCU Configuration to Communicate with an External Database Application*" on page **D-9**.

Conference Access with External Database Authentication

The MCU can work with an external database application to validate the participant's right to join an existing conference. The external database contains a list of participants, with their assigned parameters. The conference password or chairperson password entered by the participant is compared against the database. If the system finds a match, the participant is granted the permission to access the conference.

To work with an external database application to validate the participant's right to join the conference, the Conference IVR Service must be configured to use the external database application for authentication.

Conference access authentication can be performed as:

- Part of the Ad Hoc conferencing flow where the participants must be authorized before they can enter the conference created in the Ad Hoc flow
- Independent of Ad Hoc conferencing where conference access is validated for all conferences running on the MCU regardless of the method in which the conference was started.

Conference access authentication can be implemented for all participants joining the conference or for chairpersons only.

Conference Access Validation - All Participants (Always)

Once the conference is created either via an Ad Hoc Entry Queue, or a standard ongoing conference, the right to join the conference is authenticated with the external database application for all participants connecting to the conference.

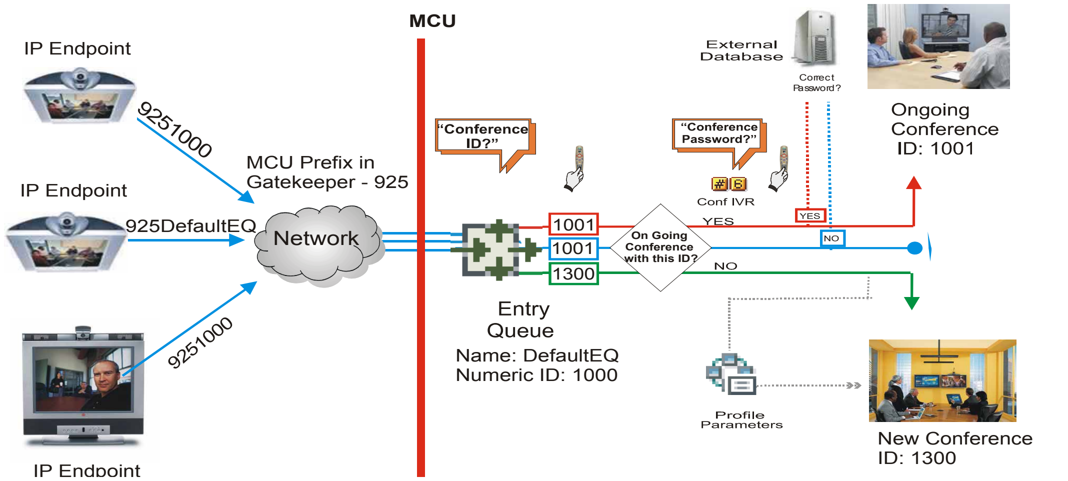


Figure D-3 *Conference Access - Conference Password validation with External Database Application*

Joining the conference entails the following steps:

- When the conference is started (either in the Ad Hoc flow or in the standard method), all participants connecting to the conference are moved to the Conference IVR queue where they are prompted for the conference password.
- When the participant enters the conference password or his/her personal password, it is sent to the external database application for validation.
- If there is a match, the participant is granted the right to join the conference. In addition, the external database application sends to the MCU the following parameters:
  - — Participant name (display name)
  - — Whether or not the participant is the conference chairperson
  - — Participant Information, such as the participant E-mail. These fields correspond to Info 1, 2, 3 and 4 fields in the *Participant Properties - Information* dialog box.

  If there is no match (i.e. the conference or personal password are not defined in the database), the request to access the conference is rejected and the participant is disconnected from the MCU.

- If the Conference IVR Service is configured to prompt for the chairperson identifier and password, the participant is requested to enter the chairperson identifier.
  - — If no identifier is entered, the participant connects as a standard, undefined participant.
- If the chairperson identifier is entered, the participant is requested to enter the chairperson password. In this flow, the chairperson password is **not** validated with the external database application, only with the MCU.
  - — If the correct chairperson password is entered, the participant is connected to the conference as its chairperson.
  - — If the wrong password is entered, he/she is disconnected from the conference.

To enable conference access validation for all participants the following conferencing components are required:

- The external database must hold the conference password or the participant personal password/PIN code or the participant's Alias.
- The Conference IVR Service assigned to the conference (defined in the Profile) must be configured to authenticate the participant's right to access the conference with the external database application for all requests. In addition it must be configured to prompt for the Conference Password.

Conference Access Validation - Chairperson Only (Upon Request)

An alternative validation method at the conference level is checking only the chairperson password with the external database application. All other participants can be checked only with the MCU (if the Conference IVR Service is configured to prompt for the conference password) or not checked at all (if the Conference IVR Service is configured to prompt only for the chairperson password).

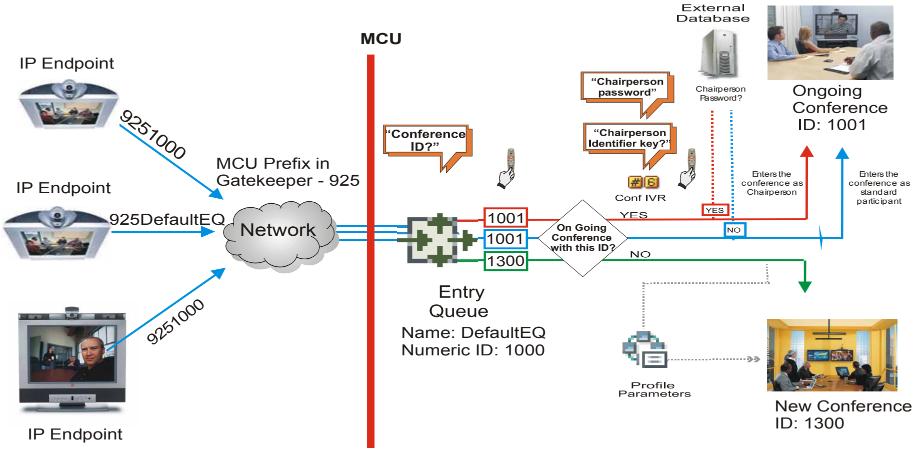


Figure D-4 *Conference Access - Chairperson Password validation with external database application*

Joining the conference entails the following steps:

- When the conference is started (either in the Ad Hoc flow or in the standard method), all participants connecting to the conference are moved to the conference IVR queue where they are prompted for the conference password.

- If the Conference IVR Service is configured to prompt for the Conference password, the participant is requested to enter the conference password. In this flow, the conference password is **not** validated with the external database application, only with the MCU.
  - — If the wrong password is entered, he/she is disconnected from the conference.
- If the correct conference password is entered, the participant is prompted to enter the chairperson identifier key.
  - — If no identifier is entered, the participant is connected to the conference as a standard participant.
- If the chairperson identifier is entered, the participant is prompted to enter the chairperson password.
- When the participant enters the chairperson password or his/her personal password, it is sent to the external database application for validation.
  - — If the password is incorrect the participant is disconnected from the MCU.
- If there is a match, the participant is granted the right to join the conference as chairperson. In addition, the external database application sends to the MCU the following parameters:
  - — Participant name (display name)
  - — Participant Information, such as the participant E-mail. These fields correspond to Info 1, 2, 3 and 4 fields in the *Participant Properties - Information* dialog box.

To enable conference access validation for all participants the following conferencing components are required:

- The external database must hold the Chairperson Password or the participant's Alias.
- The Conference IVR Service assigned to the conference (defined in the Profile) must be configured to check the external database for the Chairperson password only when the participant enters the chairperson identifier key (either pound or star). In addition, it must be configured to prompt for the chairperson identifier key and password.

System Settings for Ad Hoc Conferencing and External Database Authentication

Ad Hoc Settings

Before a participant can initiate an Ad Hoc conference (with or without authentication), the following components must be defined:

- **Profiles**

  Defines the conference parameters for the conferences that will be initiated from the Ad Hoc-enabled Entry Queue. For more details, see "*Conference Profiles"* on page **2-1**.
- **Entry Queue IVR Service with Conference ID Request Enabled**

  The Entry Queue Service is used to route participants to their destination conferences, or create a new conference with this ID. For details, see "*IVR Services"* on page **16-1**.

  In Ad Hoc conferencing, the Conference ID is used to check whether the destination conference is already running on the MCU and if not, to start a new conference using this ID.

- **Ad Hoc - enabled Entry Queue**

  Ad Hoc conferencing must be enabled in the Entry Queue and a Profile must be assigned to the Entry Queue. In addition, an Entry Queue IVR Service supporting conference ID request. For details, see "*Entry Queues"* on page **7-1**..

Authentication Settings

- **MCU Configuration**

  Usage of an external database application for authentication (for starting new conferences or joining ongoing conferences) is configured for the MCU in the System Configuration. For details, see "*MCU Configuration to Communicate with an External Database Application"* on page **D-9**.
- **Entry Queue IVR Service with Conference Initiation Authentication Enabled**

  Set the Entry Queue IVR Service to send authentication requests to the external database application to verify the participant's right to start a new conference according to the Conference ID entered by the participant. For details, see "*Enabling External Database Validation for Starting New Ongoing Conferences"* on page **D-10**.
- **Conference IVR Service with Conference Access Authentication Enabled**

  Set the Conference IVR Service to send authentication requests to the external database application to verify the participant's right to connect to the conference as a standard participant or as a chairperson. For details, see "*Enabling External Database Validation for Conferences Access"* on page **D-10**.
- **External Database Application Settings**

  The external database contains a list of participants (users), with their assigned parameters. These parameters are:

  — Conference Name

  — Conference Billing code

  — Conference Password

  — Chairperson Password

  — Conference Information, such as the contact person name. These fields correspond to Info 1, 2 and 3 fields in the *Conference Properties - Information* dialog box.

  — Maximum number of participants allowed for the conference

  — Conference Owner

  — Participant name (display name)

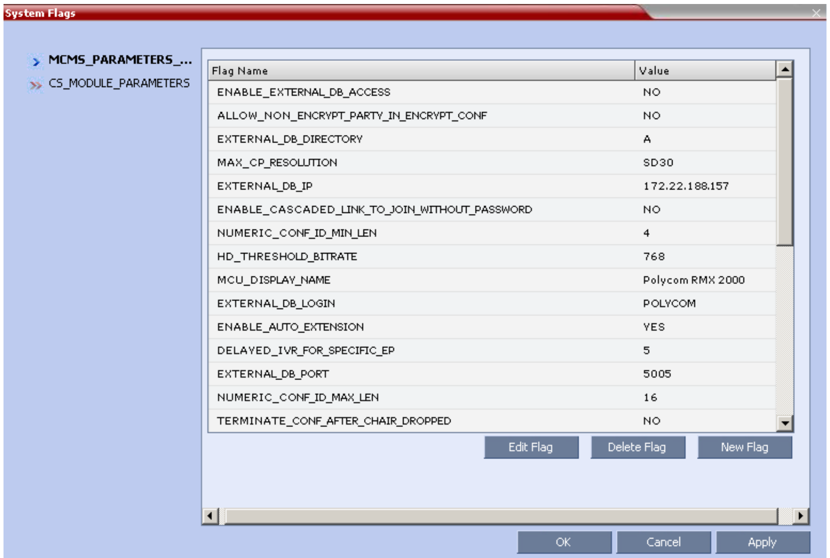
  — Participant Information, such as the participant E-mail. These fields correspond to Info 1, 2, 3 and 4 fields in the *Participant Properties - Information* dialog box.

MCU Configuration to Communicate with an External Database Application

To enable the communication with the external database application, several flags must be set in the System Configuration.

To set the System Configuration flags:

1 On the *Setup* menu, click **System Configuration**.
  The *System Flags* dialog box opens.



Flag Name	Value
ENABLE_EXTERNAL_DB_ACCESS	NO
ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF	NO
EXTERNAL_DB_DIRECTORY	A
MAX_CP_RESOLUTION	SD30
EXTERNAL_DB_IP	172.22.188.157
ENABLE_CASCADED_LINK_TO_JOIN_WITHOUT_PASSWORD	NO
NUMERIC_CONF_ID_MIN_LEN	4
HD_THRESHOLD_BITRATE	768
MCU_DISPLAY_NAME	Polycom RMX 2000
EXTERNAL_DB_LOGIN	POLYCOM
ENABLE_AUTO_EXTENSION	YES
DELAYED_IVR_FOR_SPECIFIC_EP	5
EXTERNAL_DB_PORT	5005
NUMERIC_CONF_ID_MAX_LEN	16
TERMINATE_CONF_AFTER_CHAIR_DROPPED	NO

2 Modify the values of the following flags:

Table D-1 Flag Values for Accessing External Database Application

Flag	Description and Value
ENABLE_EXTERNAL_DB_ACCESS	The flag that enables the use of the external database application.
EXTERNAL_DB_IP	The IP address of the external database application server. default IP: 0.0.0.0.
EXTERNAL_DB_PORT	The port number used by the MCU to access the external application server. Default Port = 80.
EXTERNAL_DB_LOGIN	The user name defined in the external database application for the MCU.
EXTERNAL_DB_PASSWORD	The password associated with the user name defined for the MCU in the external database application.
EXTERNAL_DB_DIRECTORY	The URL of the external database application.

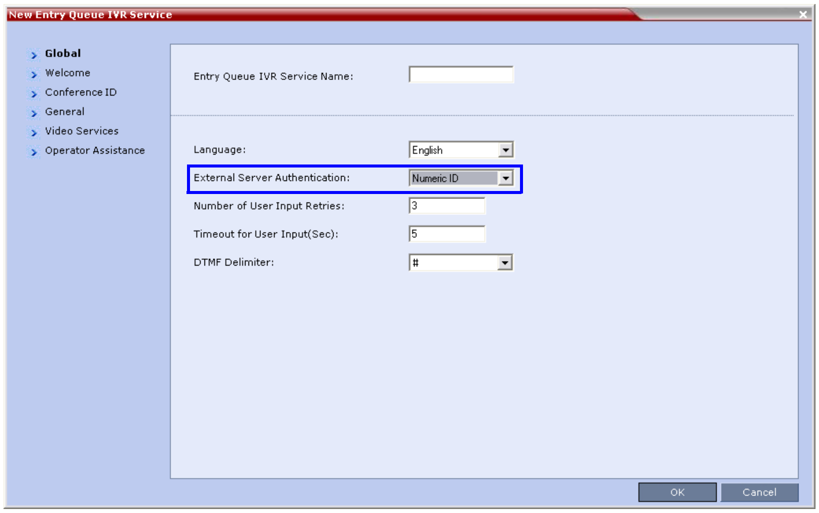
3 Click **OK**.

4 Reset the MCU for flag changes to take effect.

Enabling External Database Validation for Starting New Ongoing Conferences

The validation of the participant's right to start a new conference with an external database application is configured in the *Entry Queue IVR Service - Global* dialog box.

>> Set the *External Server Authentication* field to **Numeric ID**.



Entry Queue IVR Service Name:

Language: English

External Server Authentication: Numeric ID

Number of User Input Retries: 3

Timeout for User Input(Sec): 5

DTMF Delimiter: #

Enabling External Database Validation for Conferences Access

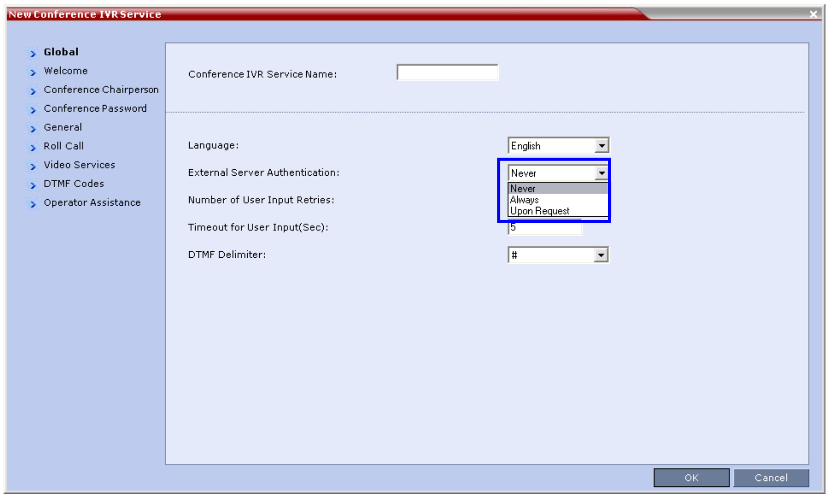
The validation of the participant's right to join an ongoing conference with an external database application is configured in the *Conference IVR Service - Global* dialog box.

You can set the system to validate all the participants joining the conference or just the chairperson.

>> Set the *External Server Authentication* field to:

— **Always -** to validate the participant's right to join an ongoing conference for all participants

— **Upon Request** - to validate the participant's right to join an ongoing conference as chairperson



Conference IVR Service Name:

Language: English

External Server Authentication: Never

Number of User Input Retries:

Timeout for User Input(Sec): 5

DTMF Delimiter: #

Appendix E

Participant Properties Advanced Channel Information

The following appendix details the properties connected with information about audio and video parameters, as well as, problems with the network which can affect the audio and video quality.

Table E-1 *Participant Properties - Channel Status Advanced Parameters*

Field	Description
Media Info	
Algorithm	Indicates the audio or video algorithm and protocol.
Frame per packet (audio only)	The number of audio frames per packet that are transferred between the MCU and the endpoint. If the actual Frame per Packets are higher than Frame per Packets declared during the capabilities exchange, a Faulty flag is displayed.
Resolution (video only)	Indicates the video resolution in use. If the actual resolution is higher than resolution declared in the capabilities exchange, the Faulty flag is displayed. For example, if the declared resolution is CIF and the actual resolution is 4CIF, the Faulty flag is displayed.
Frame Rate (video only)	The number of video frames per second that are transferred between the MCU and the endpoint.
Annexes (video only)	Indicates the H.263 annexes in use at the time of the last RTCP report. If the actual annexes used are other than the declared annexes in the capabilities exchange, the Faulty flag is displayed.
Channel Index	For Polycom Internal use only.

Table E-1 Participant Properties - Channel Status Advanced Parameters (Continued)

Field	Description
RTP Statistics	
Actual loss	The number of missing packets counted by the IP card as reported in the last RTP Statistics report. If a packet that was considered lost arrives later, it is deducted from the packet loss count. Packet loss is displayed with the following details: • ***Accumulated N*** - number of lost packets accumulated since the channel opened. • ***Accumulated %*** - percentage of lost packets out of the total number of packets transmitted since the channel opened. • ***Interval N*** - number of packets lost in the last RTP report interval (default interval is 5 minutes). • ***Interval %*** - percentage of lost packets out of the total number of packets transmitted in the last RTP report interval (default interval is 5 minutes). • ***Peak*** - the highest number of lost packets in a report interval from the beginning of the channel's life span.
Out of Order	The number of packets arriving out of order. The following details are displayed: • ***Accumulated N*** - total number of packets that arrived out of order since the channel opened. • ***Accumulated %*** - percentage of packets that arrived out of order out of the total number of packets transmitted since the channel opened. • ***Interval N*** - number of packets that arrived out of order in the last RTP report interval (default interval is 5 minutes). • ***Interval %*** - percentage of packets that arrived out of order out of the total number of packets transmitted in the last RTP report interval (default interval is 5 minutes). • ***Peak*** - the highest number of packets that arrived out of order in a report interval from the beginning of the channel's life span.

Table E-1 Participant Properties - Channel Status Advanced Parameters (Continued)

Field	Description
Fragmented	Indicates the number of packets that arrived to the IP card fragmented (i.e., a single packet broken by the network into multiple packets). This value can indicate the delay and reordering of fragmented packets that require additional processing, but is not considered a fault. The Fragmented information is displayed with the following details: • ***Accumulated N*** - total number of packets that were fragmented since the channel opened. • ***Accumulated %*** - percentage of fragmented packets out of the total number of packets transmitted since the channel opened. • ***Interval N*** - number of fragmented packets received in the last RTP report interval (default interval is 5 minutes). • ***Interval %*** - percentage of fragmented packets out of the total number of packets transmitted in the last RTP report interval (default interval is 5 minutes). • ***Peak*** - the highest number of fragmented packets in a report interval from the beginning of the channel's life span.

Appendix F

Secure Communication Mode

The *RealPresence Collaboration Server* can be configured to work in *Secure Mode* by configuring the *Collaboration Server* and the *Collaboration Server Web Client* to work with SSL/TLS.

In this mode, a SSL/TLS Certificate is installed on the MCU, setting the MCU Listening Port to secured port 4433.

TLS is a cryptographic protocol used to ensure secure communications on public networks. TLS uses a *Certificate* purchased from a trusted third party *Certificate Authority* to authenticate public keys that are used in conjunction with private keys to ensure secure communications across the network.

The *Collaboration Server* supports:

- TLS 1.0
- SSL 3.0 (Secure Socket Layer)

SSL 3.0 utilizes 1024-bit RSA public key encryption.

TLS certificates can be generated using the following methods: CSR, PFX and PEM; each giving different options for *Encryption Key* length. Table F-1 lists the *SIP TLS Encryption Key* length support for the various system components.

Table F-1 *SIP TLS - Encryption Key Support by System Component*

System Component	Key Generation Method	Key Length (bits)	Key generated by
SIP Signaling	CSR	2048	Collaboration Server
	PFX / PEM	1024 or 2048	User
Management	CSR	2048	Collaboration Server
LDAP			

Certificate Configuration and Management

All *Polycom* devices used in a *Maximum Security Environment* require security certificates.

Certificate Template Requirements

The specific security certificate requirements for *Collaboration Servers* used in *Maximum Security Environments* are:

- Support of 2048-bit encryption keys.

- Support of *Extended Key Usage* (*EKU*) for both:
  - *Client Authentication*
  - *Server Authentication*

The certificate template used by your *CA* server may need modification to meet the Collaboration Server requirements.

Certificate Requirements for Polycom Devices

Each *Polycom* device must have security certificates for the entire *Chain Of Trust.*

The Collaboration Server must have:

- The public certificate of each server in the *CA Chain* or hierarchy that issued its certificate.
  For example: *RootCA* ↔ *IntermediateCA* ↔ *SubCA*

The public certificates of the chain that issued the administrator's identity certificate. For example: *UserRootCA* ↔ *UserIntermediateCA* ↔ *UserSubCA*

Configure Certificate Management

Within a *PKI* environment, certificate revocation policies are used to ensure that certificates are valid. Certificates can expire or be revoked for various reasons (RFC 5280).

The Collaboration Server enforces these certificate revocation policies through *Certificate Revocation Lists* (*CRLs*). *CRLs* are required for each *CA Chain* in use by the Collaboration Server. These *CRL* files must be kept current

Switching to Secure Mode

The following operations are required to switch the Collaboration Server to *Secure Mode*:

- Purchase and Install the *SSL/TLS certificate*
- Modify the *Management Network* settings
- Create/Modify the relevant *System Flags*

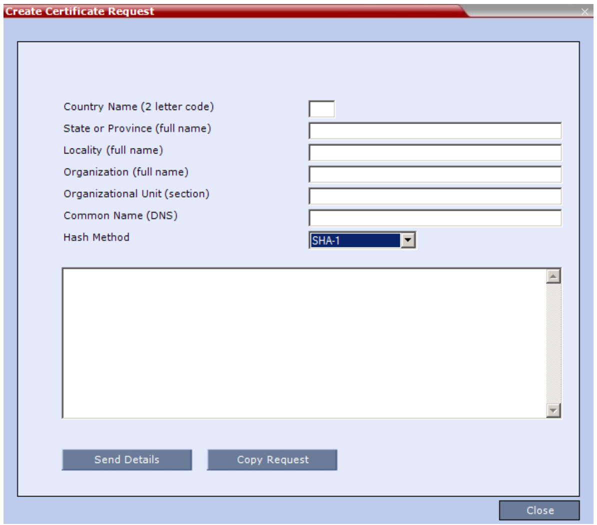
Purchasing a Certificate

Once a certificate is purchased and received it is stored in the Collaboration Server and used for all subsequent secured connections.

To create/purchase a certificate:

1 In the Collaboration Server menu, click **Setup > RMX Secured Communication > Create certificate request**.

The *Create Certificate Request* dialog box is displayed.


Create Certificate Request
Country Name (2 letter code)
State or Province (full name)
Locality (full name)
Organization (full name)
Organizational Unit (section)
Common Name (DNS)
Send Details
Copy Request
Close

Hash Method: SHA-1

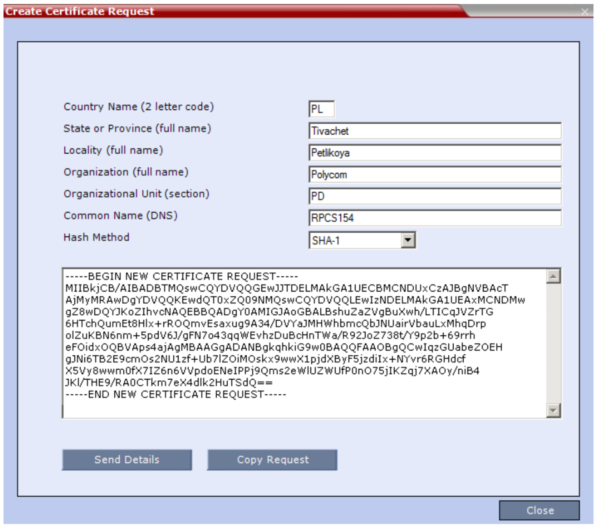
2 Enter information in all the following fields:

Table F-2 Create Certificate Request

Field	Description
Country Name	Enter any 2 letter code for the country name.
State or Province	Enter the full name of the state or province.
Locality	Enter the full name of the town/city/location.
Organization	Enter the full name of your organization for which the certificate will be issued.
Organizational Unit	Enter the full name of the unit (group or division) for which the certificate will be issued.
Common Name (DNS/ IP)	Enter the *DNS MCU Host Name*. This *MCU Host Name* must also be configured in the *Management Network Properties* dialog box.
Hash Method	Select the Hash method for the certificate.

3 Click **Send Details**.

The Collaboration Server creates a *New Certificate Request* and returns it to the *Create Certificate Request* dialog box along with the information the user submitted.



Country Name (2 letter code): PL
State or Province (full name): Tivachet
Locality (full name): Petlikoya
Organization (full name): Polycom
Organizational Unit (section): PD
Common Name (DNS): RPCS154
Hash Method: SHA-1

```
-----BEGIN NEW CERTIFICATE REQUEST-----
MIIBkjCB/AIBADBTMQswCQYDVQQGEwJJTDELMAkGA1UECBMCNDUxCzAJBgNVBAcT
AjMyMRAwDgYDVQQKEwdQT0xZQ09NMQswCQYDVQQLEwIzNDELMAkGA1UEAxMCNDMw
gZ8wDQYJKoZIhvcNAQEBBQADgY0AMIGJAoGBALBshuZaZVgBuXwh/LTICqJVZrTG
6HTchQumEt8Hlx+rROQmvEsaxug9A34/DVYaJMHWhbmcQbJNUairVbauLxMhqDrp
olZuKBN6nm+5pdV6J/gFN7o43qqWEvhzDuBcHnTWa/R92JoZ738t/Y9p2b+69rrh
eFOidxOQBVAps4ajAgMBAAGgADANBgkqhkiG9w0BAQQFAAOBgQCwIqzGUabeZOEH
gJNi6TB2E9cmOs2NU1zf+Ub7lZOiMOskx9wwX1pjdXByF5jzdiIx+NYvr6RGHdcf
X5Vy8wwm0fX7IZ6n6VVpdoENeIPPj9Qms2eWlUZWUfP0nO75jIKZqj7XAOy/niB4
JKl/THE9/RA0CTkm7eX4dlk2HuTSdQ==
-----END NEW CERTIFICATE REQUEST-----
```

4 Click **Copy Request** to copy the *New Certificate Request* to the workstation’s clipboard.

5 Connect to your preferred *Certificate Authority’s* website using the web browser.

6 Follow the purchasing instructions at the *Certificate Authority’s* website.
Paste (**Ctrl + V)** the *New Certificate Request* as required by the *Certificate Authority*.

The *Certificate Authority* issues the TLS/SSL certificate, and sends the certificate to you by e-mail.

Installing the Certificate

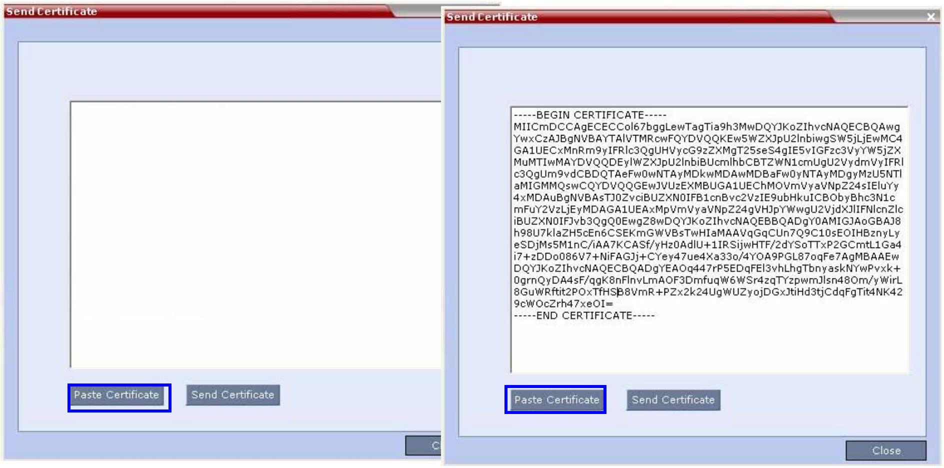
To install the certificate:

After you have received the certificate from the *Certificate Authority:*

1 **Copy** (**Ctrl + C**) the certificate information from the *Certificate Authority’s* e-mail to the clipboard.

2 In the *Collaboration Server* menu, click **Setup > RMX Secured Communication > Send Certificate**.

3 Click **Paste Certificate** to paste the clipboard content into the *Send Certificate* dialog box.



```
-----BEGIN CERTIFICATE-----
MIICmDCCAgECECCol67bggLewTagTia9h3MwDQYJKoZIhvcNAQECBQAwg
YwxCzAJBgNVBAYTAlVTMRcwFQYDVQQKEw5WZXJpU2lnbiwgSW5jLjEwMC4
GA1UECxMnRm9yIFRlc3QgUHVycG9zZXMgT25seS4gIE5vIGFzc3VyYW5jZX
MuMTIwMAYDVQQDEylWZXJpU2lnbiBUcmlhbCBTZWN1cmUgU2VydmVyIFRl
c3QgUm9vdCBDQTAeFw0wNTAyMDkwMDAwMDBaFw0yNTAyMDgyMzU5NTl
aMIGMMQswCQYDVQQGEwJVUzEXMBUGA1UEChMOVmVyaVNpZ24sIEluYy
4xMDAuBgNVBAsTJ0ZvciBUZXN0IFB1cnBvc2VzIE9ubHkuICBObyBhc3N1c
mFuY2VzLjEyMDAGA1UEAxMpVmVyaVNpZ24gVHJpYWwgU2VjdXJlIFNlcnZlc
iBUZXN0IFJvb3QgQ0EwgZ8wDQYJKoZIhvcNAQEBBQADgY0AMIGJAoGBAJ8
h98U7klaZH5cEn6CSEKmGWVBsTwHIaMAAVqGqCUn7Q9C10sEOIHBznyLy
eSDjMs5M1nC/iAA7KCASf/yHz0AdlU+1IRSijwHTF/2dYSoTTxP2GCmtL1Ga4
i7+zDDo086V7+NiFAGJj+CYey47ue4Xa33o/4YOA9PGL87oqFe7AgMBAAEw
DQYJKoZIhvcNAQECBQADgYEAOq447rP5EDqFEl3vhLhgTbnyaskNYwPvxk+
0grnQyDA4sF/qgK8nFlnvLmAOF3DmfuqW6WSr4zqTYzpwmJlsn48Om/yWirL
8GuWRftit2POxTfHSB8VmR+PZx2k24UgWUZyojDGxJtiHd3tjCdqFgTit4NK42
9cWOcZrh47xeOI=
-----END CERTIFICATE-----
```

4 Click the **Send Certificate** button to send the certificate to the Collaboration Server.

The MCU validates the certificate.

— If the certificate is not valid,an error message is displayed.

— If the certificate matches the private key, and the task is completed, a confirmation message indicating that the certificate was created successfully is displayed.

A *System Restart* is **not** required at this point.

The certificate expiry date is checked daily. An active alarm is raised two weeks before the certificate is due to expire, stating the number of days to expiry.

If the certificate expires, the Collaboration Server continues to work in secure mode and an *Active Alarm* is raised with *Security mode failed – Certificate expired* in the description field.

Certificates are deleted when an administrator performs a *Restore Factory Defaults* with the *Comprehensive Restore* option selected.

Creating/Modifying System Flags

The following *System Flags* in *system.cfg* control secure communications.

- *RMX_MANAGEMENT_SECURITY_PROTOCOL*
- *EXTERNAL_DB_PORT*

Table F-3, below, lists both flags and their settings.

If the *System Flag*, *RMX_MANAGEMENT_SECURITY_PROTOCOL* does not exist in the system, it must be created by using the *Setup* menu.

For more information see "*Modifying System Flags"* on page 20-1.

Table F-3 System Flags

Flag	Description
RMX_MANAGEMENT_SECURITY_PROTOCOL	Enter the protocol to be used for secure communications. Default: TLSV1_SSLV3 (both).

Table F-3 System Flags (Continued)

Flag	Description
EXTERNAL_DB_PORT	The external database server port used by the Collaboration Server to send and receive XML requests/responses. For secure communications set the value to 4433. Default: 5005.

The Collaboration Server must be restarted for modified flag settings to take effect.

Enabling Secure Communication Mode

After the SSL/TLS Certificate is installed, secure communications are enabled by modifying the properties of the *Management Network* in the *Management Network* properties dialog box.

When *Secure Communications Mode* is enabled:

- Only **`https://`** commands from the browser to the *Control Unit IP Address* of the Collaboration Server are accepted.
- The Collaboration Server listens only on secured port 4433.
- All connection attempts on port 8080 are rejected.
- A secure communication indicator (🔒) is displayed in the browser's status bar.

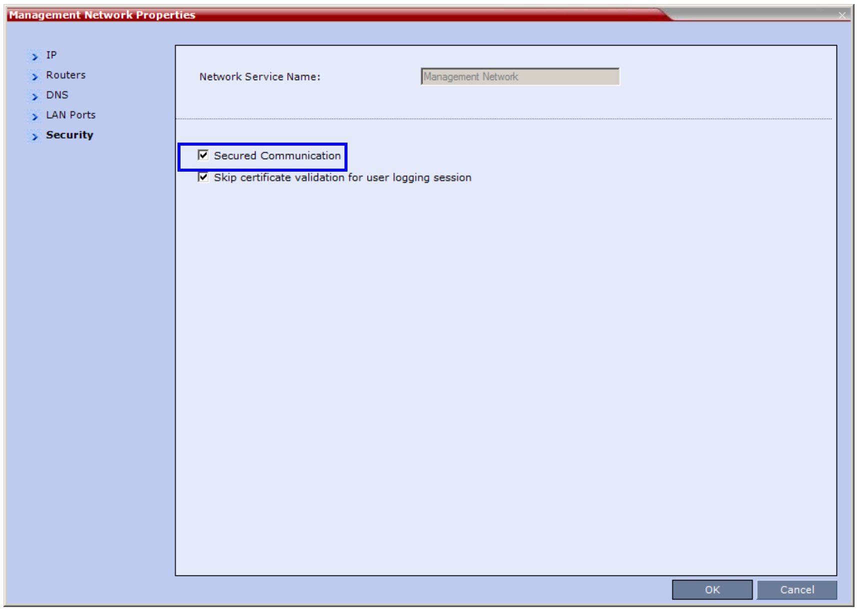
To enable secure communications mode:

1 In the *Collaboration Server Management* pane, click **IP Network Services.**

2 In the *IP Network Services* list pane, double-click the **Management Network** entry.

3 Click the **Security** tab.

The *Management Security Properties* dialog box is displayed.



Management Network Properties

Network Service Name: Management Network

☑ Secured Communication

☑ Skip certificate validation for user logging session

4 Select the **Secured Communication** check box.

5 Click **OK**.

Securing an External Database

TLS 1.0 is used when securing communications between the Collaboration Server and an external database. The certificate is installed on the database server and the Collaboration Server is the client. When the certificate is installed on the database server, all client requests and responses are transferred via secure port 4433.

It is important to verify that the external database application is operating in secure mode before enabling secure external database communications on the Collaboration Server. The Collaboration Server checks the validity of external database's certificate before communicating. If there is a certificate error an *Active Alarm* is raised with *Error in external database certificate* in the description field.

To enable secure Collaboration Server Communications with an External Database:

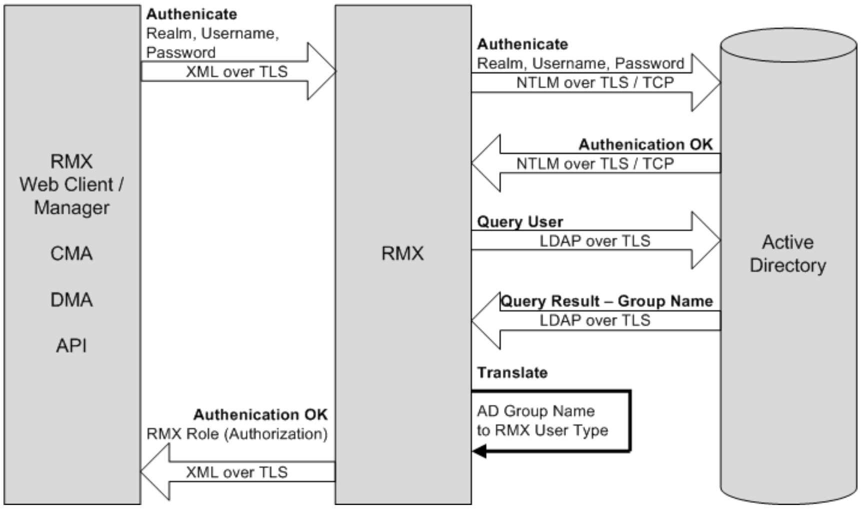
>> Set the Collaboration Server to communicate with the database server via port 4433 by setting the value of the *System Flag* `EXTERNAL_DB_PORT` in *system.cfg* to 4433.

For more information see "*Modifying System Flags"* on page **20-1**.

MS Active Directory Integration

It is possible to configure direct interaction between the Collaboration Server and *Microsoft Active Directory* for *Authentication* and *Authorization* of *Management Network* users.

The following diagram shows a typical user authentication sequence between a *User*, Collaboration Server and *Active Directory*.



Directory and Database Options

Standard Security Mode

Internal Collaboration Server database + External Database

First authentication is via the internal Collaboration Server database. If it is not successful, authentication is via the *External Database*.

Internal Collaboration Server database + External Database + Active Directory

- **Management Logins**

  First authentication is via the internal Collaboration Server database. If it is not successful, authentication is via the *Active Directory*.
- **Conference Queries** (*Chairperson Password, Numerical ID* etc.)

  First authentication is via the internal Collaboration Server database. If it is not successful, authentication is via the *External Database*.

Guidelines

- The Collaboration Server maintains a local record of:
  - *Audit Events* – users that generate these events are marked as being either internal or external.
  - Successful user logins
  - Failed user login attempts
- User passwords and user lockout policy for external users are managed via *Active Directory’s* integration with the user’s host machine.
- Enabling or disabling *Active Directory* integration does not require a reset.
- In *Standard Security Mode* multiple accounts of all user types are supported.
- Multiple *Machine Accounts* with various roles are supported.
- *Microsoft Active Directory* is the only directory service supported.
- *Active Directory* integration is configured as part of the *Management Network*.
- In *Standard Security Mode,* the *Active Directory* can be queried using *NTLM* with or without *TLS* encryption.
- Server and client certificate validation requests use *LDAP* with or without *TLS* encryption.

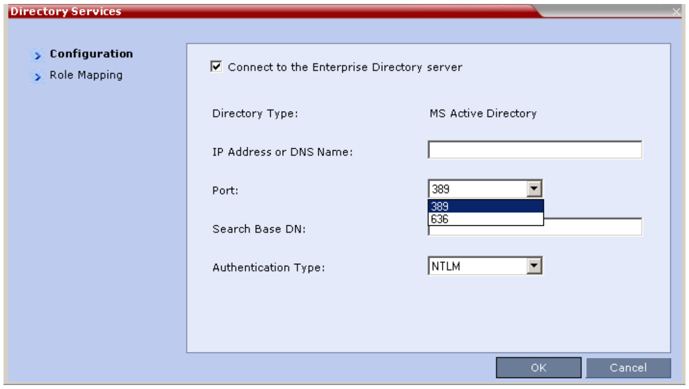
> When using *LDAP* over *TLS*, in addition to using port **389** with *STARTTLS*, the administrator has the option of using port **636**.

Enabling Active Directory Integration

To configure Directory Services:

1. On the *Collaboration Server Menu*, click **Setup > Directory Services**.

   The *Directory Services - Configuration* dialog box is displayed.



☑ Connect to the Enterprise Directory server

Directory Type: MS Active Directory

IP Address or DNS Name:

Port: 389

389
636

Search Base DN:

Authentication Type: NTLM

2. Modify the following fields.

Table F-4 Directory Services - Configuration

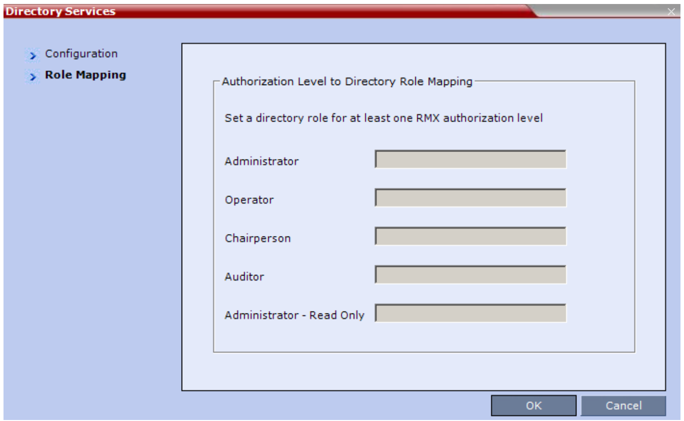
Field	Description
Connect to the Enterprise Directory Server	Select this check box to enable or disable the *Active Directory* feature.
IP Address or DNS Name	Enter the IP address or DNS name of the Enterprise Directory Server (Active Directory).
Port	Select the *Port* according to the *Authentication Protocol* to be used: • **389** - *NTLM* over *TCP* • **636** - *NTLM* over *TLS*

Search Base DN	Enter the starting point when searching for *User* and *Group* information in the *Active Directory*.
	For example if the *Domain Name* is:
	`mainoffice.bigcorp.com.uk`
	The entry in this field should be:
	`CN=Users,DC=mainoffice,DC=bigcorp,DC=come,DC=uk`

Authentication Type	Only NTLM can be used.

3 Click the **Role Mapping** tab.

The *Directory Services - Role Mapping* dialog box is displayed.



Authorization Level to Directory Role Mapping

Set a directory role for at least one RMX authorization level

Administrator

Operator

Chairperson

Auditor

Administrator - Read Only

Each of the Collaboration Server user types: *Administrator*, Administrator Read-Only, *Auditor*, *Operator* and *Chairperson* can be mapped to only one *Active Directory Group* or *Role* according to the customer's specific implementation.

— A Collaboration Server user that belongs to multiple *Active Directory Groups* is assigned to the *Group* with the least privileges.

4 Map the *Collaboration Server User Types,* to their *Active Directory* roles by modifying the following fields.

Table F-5 *Directory Services - Role Mapping*

Field	Description
Administrator	At least one of these *User Types* must be mapped to an *Active Directory Role*.
Administrator Read-Only	
Operator	
Chairperson	
Auditor	

5 Click **OK**.

Appendix G

Setting the Collaboration Server for Integration Into Microsoft Environment

Integration into Microsoft environment (using Lync endpoints) is supported in AVC CP Conferencing Mode only.

Overview

- The Polycom® Visual Communications offers high quality video and audio multipoint conferencing by integrating the Polycom network devices and endpoints into Microsoft® platforms. The Polycom® RealPresence® Collaboration Server (Collaboration Server) system can be integrated into Lync Server 2010 environment (Microsoft Wave 14).

Point-to-point and multipoint audio and video meetings can be initiated from Lync client, Windows Messenger and Polycom video endpoints (HDX and VSX) when the environment components are installed and configured.

- Multipoint calls are enabled when the Collaboration Server is installed in the Microsoft environment and is configured for unified communications. Routing to conferences can be performed by the Lync Server by *Matched URI dialing* - using the SIP URI address.

Only TLS connections to the Collaboration Server will work, TCP connections will not work.
The Collaboration Server does not support working with multiple Edge servers.

TLS certificates can be generated using the following methods: CSR, PFX and PEM; each giving different options for *Encryption Key* length. Table G-1 lists the *SIP TLS Encryption Key* length support for the various system components.

Table G-1 *SIP TLS - Encryption Key Support by System Component*

System Component	Key Generation Method	Key Length (bits)	Key generated by
SIP Signaling	CSR	2048	Collaboration Server
	PFX / PEM	1024 or 2048	User

Table G-1 SIP TLS - Encryption Key Support by System Component (Continued)

System Component	Key Generation Method	Key Length (bits)	Key generated by
Management	CSR	2048	Collaboration Server
LDAP			

Conferencing Entities Presence

Conferencing entities (Meeting Rooms, Entry Queues and SIP Factories) can be registered with the SIP server (Lync server) enabling the addition of these conferencing entities to the buddy list while displaying their presence (availability status: Available, Offline, or Busy). Lync Server client users can connect to conferencing entities directly from the buddy list.

The configuration of the environment to enable Presence, is usually done once the basic configuration is completed.

For more details, see "*Adding Presence to Conferencing Entities in the Buddy List"* on page **G-22**.

Collaboration Server Integration into the Microsoft Lync Server 2010 and Lync Server 2013 Environments

From Version 7.8, the The RMX interoperability level with Lync 2013 is identical to Lync 2010. Lync 2013 is backward compatible with all RMX Lync 2010 features.

In the Lync Server 2010 environment, only the Matched URI dialing (using the SIP URI address) is available as the call routing method.

> Non-Lync endpoints connected to the same CP AVC-based conference as Lync endpoints running on the Collaboration Server, cannot participate in the desktop sharing session initiated by Lync participants.

Configuring the Polycom-Microsoft Solution

See the *Polycom Unified Communications Deployment Guide for Microsoft Environments*, *"Deployment Process for Polycom Collaboration Server Systems"* for detailed steps on how to deploy a Polycom Collaboration Server system for use with the video conferencing solution in Microsoft Lync Server 2010 environment.

Media Over TCP

Media is automatically transmitted through TCP when UDP, the default transport protocol, is not available.

The media transport protocol type (UDP/TCP) is displayed in the *Participant Properties - Channel Status - Advanced* dialog box.

The media transport protocol type is displayed for the following IP addresses:

- Collaboration Server IP Address
- Participant IP Address

Network Error Recovery

When a short network error occurs, for example 5 seconds, Collaboration Server automatically recovers, enabling calls in Microsoft Lync to continue the video or audio conference without disconnecting. However, when a longer network error occurs, the call is disconnected. The presence status mode is correctly updated from *Busy* to *Available*. There is no configuration required for this procedure.

SIP Dialog Recovery

Collaboration Server has the ability to automatically recover from a SIP dialog failure, which can occur in long duration calls in Meeting Rooms using the Microsoft Lync client. There is no configuration required for this procedure.

Content Sharing via Polycom CSS (Content Sharing Suite) Plug-in for Lync Clients

The Polycom CSS (Content Sharing Suite) Plug-in for Lync clients allows Lync clients to receive and send Content on a separate channel, without having to use the video channel. Content is transmitted using SIP BFCP. For more details, see "*Sharing Content via the Polycom CSS Plug-in for Lync Clients"* on page **G-18**.

Configuring the Collaboration Server for Microsoft Integration

The Collaboration Server is integrated in and Microsoft Lync Server environments by setting its *Transport Type* (in the SIP server configuration) to **TLS** and creating a certificate that is sent to the Collaboration Server. This procedure is also required when encryption of SIP signaling is used.

In addition, if the DNS server was not enabled in the *Network Management Service* on the Collaboration Server, it must be enabled for the integration in the Lync Server (Wave 14) environments.

Modify the Collaboration Server Management Network Service to Include the DNS Server

The *Management Network* that is defined during first entry setup does not include the definition of the DNS which is mandatory in Microsoft environment and has to be modified.

> In *Multiple Networks* configurations, a *DNS* server can be specified for each *IP Network Service* and for the *Collaboration Server Management Network Service*.

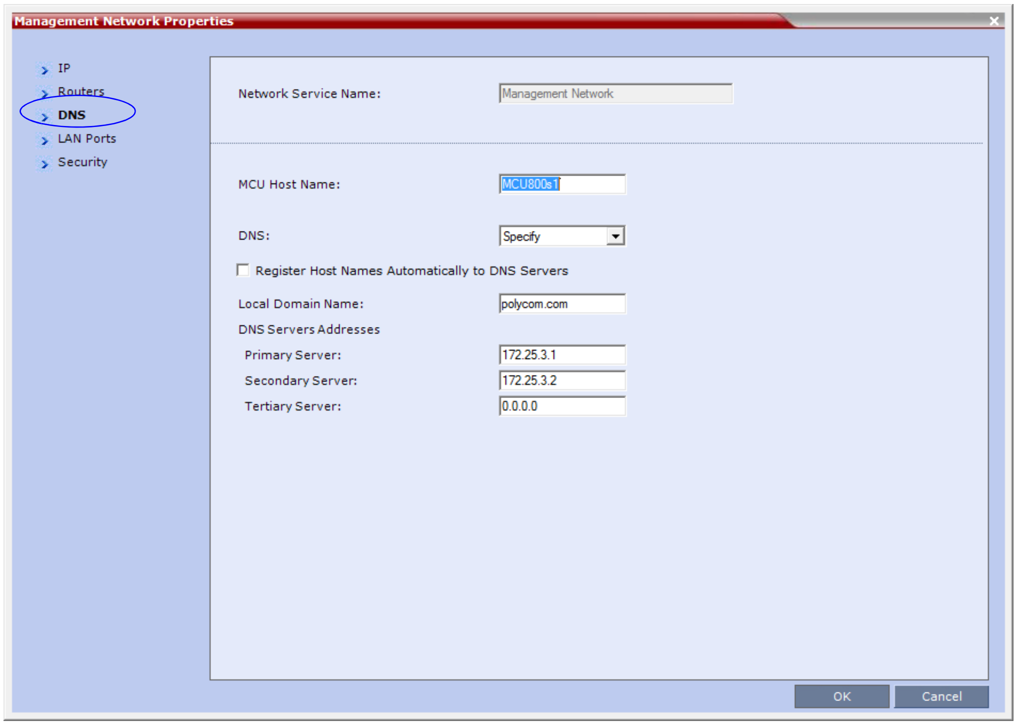
To add the definition of the DNS to the Management Network in the Collaboration Server:

1 Using the Web browser, connect to the Collaboration Server.

2 In the *Collaboration Server Management* pane, expand the **Rarely Used** list and click **IP Network Services** (IP).

3 In the *IP Network Services* pane, double-click the **Management Service** **.**
The *Management Network Properties - IP* dialog box opens.

4 Click the **DNS** tab.



Network Service Name: Management Network

MCU Host Name: MCU800s1

DNS: Specify

□ Register Host Names Automatically to DNS Servers

Local Domain Name: polycom.com

DNS Servers Addresses

Primary Server: 172.25.3.1

Secondary Server: 172.25.3.2

Tertiary Server: 0.0.0.0

5 In the *DNS* field, select **Specify** to define the DNS parameters.

6 View or modify the following fields:

Table 8 Management Network Properties – DNS Parameters

Field	Description
MCU Host Name	Enter the name of the MCU on the network. This name must be identical to the FQDN name defined for the Collaboration Server in the OCS and DNS. Default name is Collaboration Server.
Shelf Management Host Name	Displays the name of the entity that manages the Collaboration Server hardware. The name is derived from the MCU host name. Default is RMX_SHM.
DNS	Select: • **Off** – if DNS servers are not used in the network. • **Specify** – to enter the IP addresses of the DNS servers. **Note:** The IP address fields are enabled only if **Specify** is selected.
Register Host Names Automatically to DNS Servers	Select this option to automatically register the MCU Signaling Host and Shelf Management with the DNS server.
Local Domain Name	Enter the name of the domain where the MCU is installed as defined in the Office Communications Server/Lync Server.

Table 8 Management Network Properties – DNS Parameters (Continued)

Field	Description
DNS Servers Addresses:	
Primary Server	The static IP addresses of the DNS servers (the same servers defined in the Office Communications Server/Lync Server). A maximum of three servers can be defined.
Secondary Server	
Tertiary Server	

7 Click **OK**.

Defining a SIP Network Service in the Collaboration Server and Installing the Security Certificate

Your RealPresence Collaboration Server system should be installed according to standard installation procedures. For details, see the *Polycom® RealPresence Collaboration Server Virtual Edition Getting Started Guide*.

When configuring the *Default IP Network Service* on first entry, or when modifying the properties of the existing *Default IP Network Service*, the SIP environment parameters must be set as described in this section.

The Security Certificate

There are two methods to create and send the security certificate that is required for configuration of the integration of the Collaboration Server in the Microsoft environment:

- The CSR method
- The PFX method (Recommended method for Lync Server, Wave 14)

The CSR Method

In the CSR method, the security certificate is created as part of the *SIP Server* configuration in the IP Network Service configuration.

Using the CSR Method, the following processes are performed:

- Creating the certificate request (in the *Default IP Network Service - SIP Server* dialog box).
- Sending the certificate request to a Certificate Authority.
- Receiving the certificate from the Certificate Authority.
- Installing the certificate in the Collaboration Server (in the *Default IP Network Service - SIP Server* dialog box).

The PFX Method

In the PFX method, the security certificate is created in advance, in the Lync Server environment.

For detailed description of this procedure in the Lync Server environment, see the *Polycom Unified Communications Deployment Guide for Microsoft Environments*..

Certificates are deleted when an administrator performs a *Restore Factory Defaults* with the *Comprehensive Restore* option selected.

Configuring the Collaboration Server IP Network Service

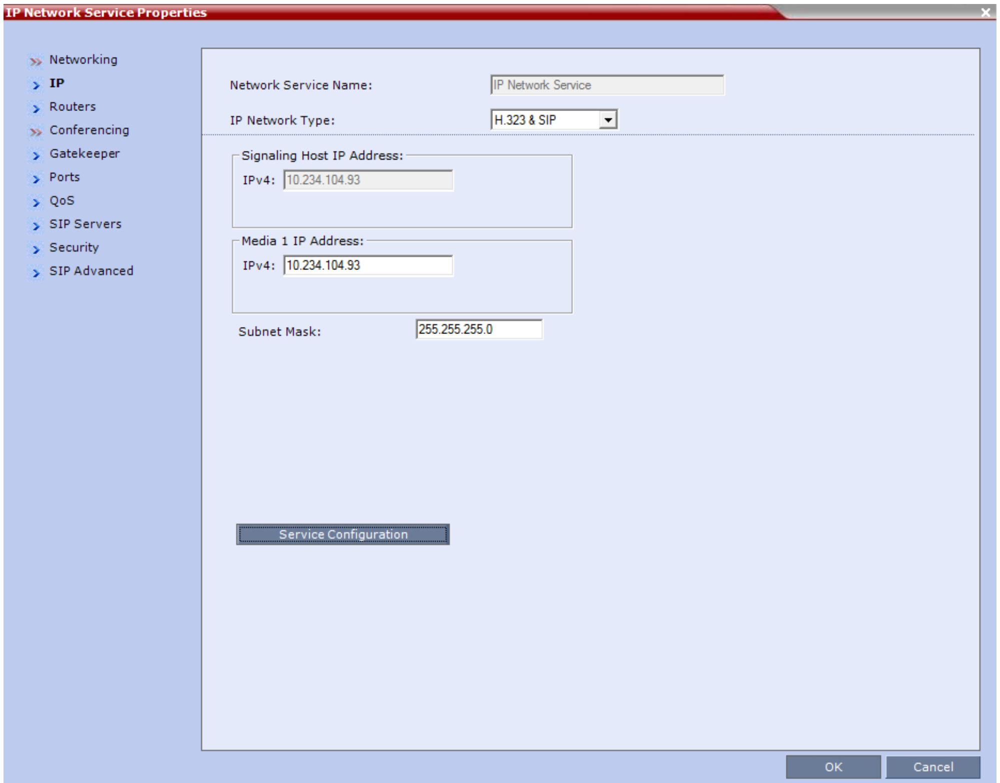
To configure the Collaboration Server IP Network Service:

1 Using the Web browser, connect to the Collaboration Server.

2 In the *RealPresence Collaboration Server Management* pane, expand the **Rarely Used** list and click **IP Network Services** (IP).

3 In the *IP Network Services* pane, double-click the **Default IP Service** (323, SIP, or SIP 323) entry.

The *Default IP Service - Networking IP* dialog box opens.



Network Service Name: IP Network Service

IP Network Type: H.323 & SIP

Signaling Host IP Address:
IPv4: 10.234.104.93

Media 1 IP Address:
IPv4: 10.234.104.93

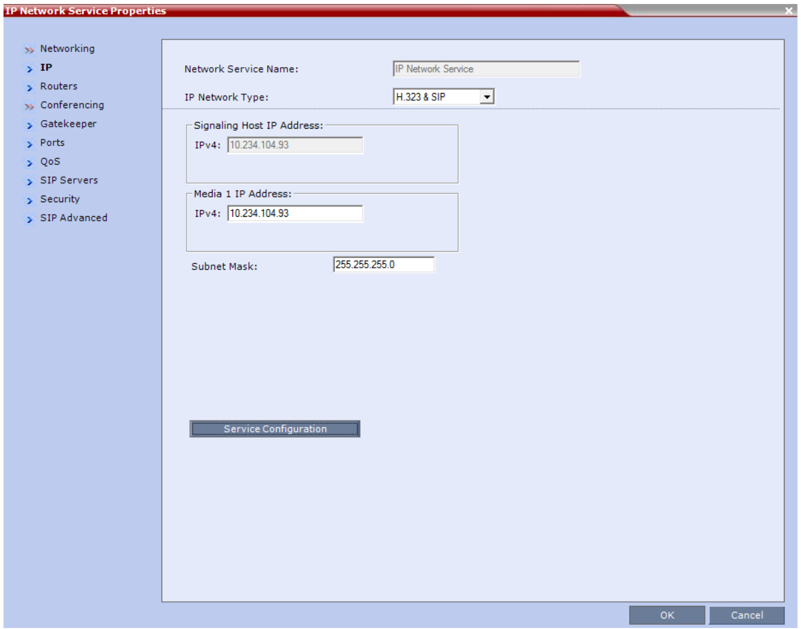
Subnet Mask: 255.255.255.0

4 Make sure the *IP Network Type* is set to **H.323 & SIP** even though SIP will be the only call setup used with Office Communications Server 2007.

5 Make sure that the correct parameters are defined for the *Signaling Host IP Address* and *Subnet Mask*.

> Make sure that the IP address of the Collaboration Server Signaling Host is the same one defined as a trusted host in Lync Server 2010.

6 Click the **SIP Servers** tab.



Network Service Name: IP Network Service

IP Network Type: H.323 & SIP

Signaling Host IP Address:
IPv4: 10.234.104.93

Media 1 IP Address:
IPv4: 10.234.104.93

Subnet Mask: 255.255.255.0

7 In the *SIP Server,* select **Specify.**

8 In the *SIP Server Type,* select **Microsoft**.

9 Enter the IP address of the Office Communications Server 2007 or Lync Server 2010 and the *Server Domain Name* as defined in the OCS/Lync Server and in the *Management Network* for the DNS.

10 If not selected by default, change the *Transport Type* to **TLS**.
The *Create Certificate* and *Send Certificate* buttons are enabled.

11 If you are using the CSR method, and the **CSR** option is not selected by default, change the *Certificate Method* to **CSR**.

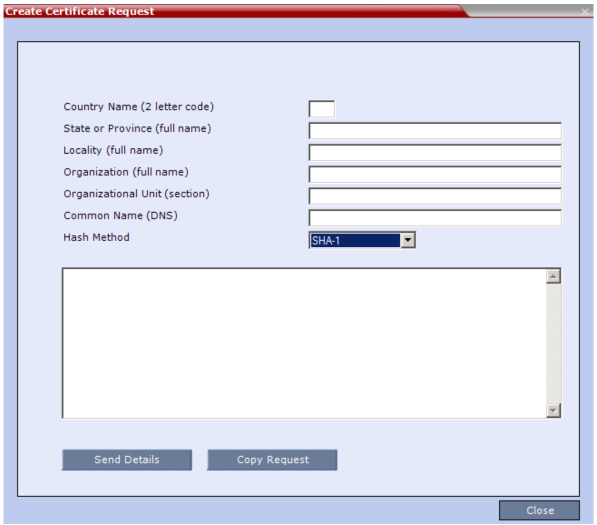
If you are using the PFX method, in the *Certificate Method* field select **PEM/PFX**.
At this point the procedure changes according to the selected certificate method.

If you have selected PEM/PFX, skip to step **27** on **page G-12.**

CSR Method - Creating the Certificate

12 Click the **Create Certificate** button.

The *Create Certificate Request* dialog box is displayed.


Create Certificate Request
Country Name (2 letter code)
State or Province (full name)
Locality (full name)
Organization (full name)
Organizational Unit (section)
Common Name (DNS)
Hash Method
SHA-1
Send Details
Copy Request
Close

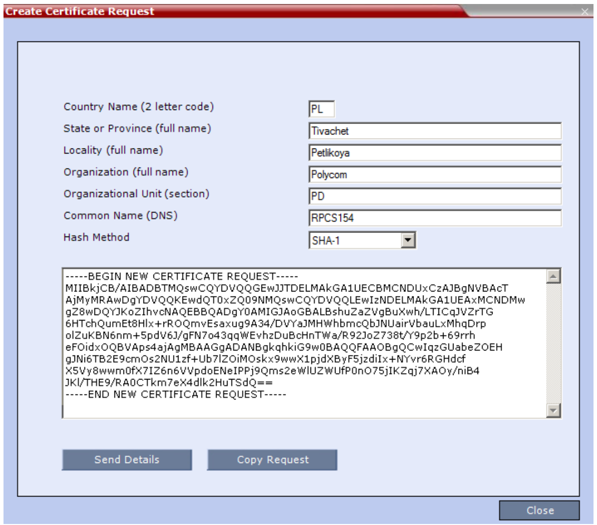
13 Enter information in all the following fields:

Table G-1 Create Certificate Request

Field	Description
Country Name	Enter any 2 letter code for the country name.
State or Province	Enter the full name of the state or province.
Locality	Enter the full name of the town/city/location.
Organization	Enter the full name of your organization for which the certificate will be issued.
Organizational Unit	Enter the full name of the unit (group or division) for which the certificate will be issued.
Common Name (DNS/ IP)	Enter the *DNS MCU Host Name*. This *MCU Host Name* must also be configured in the *Management Network Properties* dialog box.
Subject Alternative Name	
Hash Method	Select the hash method to be used to hash the certificate request.

14 Click **Send Details**.

The Collaboration Server creates a *New Certificate Request* and returns it to the *Create Certificate Request* dialog box along with the information the user submitted.



Country Name (2 letter code): PL
State or Province (full name): Tivachet
Locality (full name): Petlikoya
Organization (full name): Polycom
Organizational Unit (section): PD
Common Name (DNS): RPCS154
Hash Method: SHA-1

```
-----BEGIN NEW CERTIFICATE REQUEST-----
MIIBkjCB/AIBADBTMQswCQYDVQQGEwJJTDELMAkGA1UECBMCNDUxCzAJBgNVBAcT
AjMyMRAwDgYDVQQKEwdQT0xZQ09NMQswCQYDVQQLEwIzNDELMAkGA1UEAxMCNDMw
gZ8wDQYJKoZIhvcNAQEBBQADgY0AMIGJAoGBALBshuZaZVgBuXwh/LTICqJVZrTG
6HTchQumEt8Hlx+rROQmvEsaxug9A34/DVYaJMHWhbmcQbJNUairVbauLxMhqDrp
olZuKBN6nm+5pdV6J/gFN7o43qqWEvhzDuBcHnTWa/R92JoZ738t/Y9p2b+69rrh
eFOidxOQBVAps4ajAgMBAAGgADANBgkqhkiG9w0BAQQFAAOBgQCwIqzGUabeZOEH
gJNi6TB2E9cmOs2NU1zf+Ub7lZOiMOskx9wwX1pjdXByF5jzdiIx+NYvr6RGHdcf
X5Vy8wwm0fX7IZ6n6VVpdoENeIPPj9Qms2eWlUZWUfP0nO75jIKZqj7XAOy/niB4
JKl/THE9/RA0CTkm7eX4dlk2HuTSdQ==
-----END NEW CERTIFICATE REQUEST-----
```

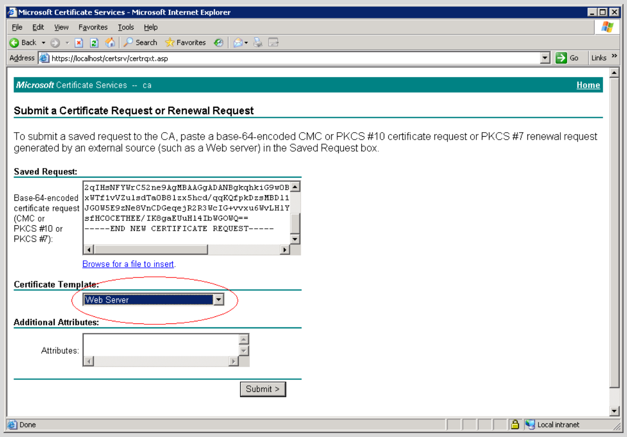
15 Click **Copy Request** to copy the *New Certificate Request* to the workstation's clipboard.

16 Connect to your preferred *Certificate Authority's* website using the web browser.

17 Follow the purchasing instructions at the *Certificate Authority's* website.

18 Paste (**Ctrl + V)** the *New Certificate Request* as required by the *Certificate Authority*.

When creating the certificate request in the Certificate Authority site, make sure that the **Web Server** option is selected as the Certificate Template, as shown in the example below.


Microsoft Certificate Services - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites
Address https://localhost/certsrv/certrqxt.asp
Go Links
Microsoft Certificate Services -- ca
Home
Browse for a file to insert.
Submit >
Done
Local intranet

Submit a Certificate Request or Renewal Request

To submit a saved request to the CA, paste a base-64-encoded CMC or PKCS #10 certificate request or PKCS #7 renewal request generated by an external source (such as a Web server) in the Saved Request box.

Saved Request:

Base-64-encoded certificate request (CMC or PKCS #10 or PKCS #7):

```
2qIHsNFYWrC52ne9AgMBAAGgADANBgkqhkiG9w0B
xWTf1vVZulsdTaOB8lzx5hcd/qqKQfpkDzsMBD11
JGOW5E9zNe8VnCDGeqejR2R3WcIG+vvxu6WvLH1Y
sfHCOCETHEE/IK8gaEUuH14IbWGOWQ==
-----END NEW CERTIFICATE REQUEST-----
```

Certificate Template: Web Server

Additional Attributes:

Attributes:

The *Certificate Authority* issues the TLS/SSL certificate, and sends the certificate to you by e-mail.

If the process of purchasing the certificate is short, you may leave the *IP Network Service - SIP Servers* dialog box open. Otherwise, close it without saving the changes to the Transport Type and Certificate Method.

CSR Method - Sending the certificate

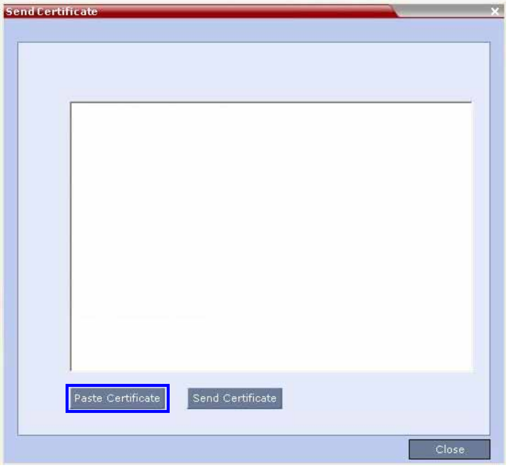
After you have received the certificate from the *Certificate Authority:*

> If you have closed the *IP Network Service - SIP Servers* dialog box, repeat steps 1 to 11 in the procedure "*Defining a SIP Network Service in the Collaboration Server and Installing the Security Certificate"* on page **G-6**.

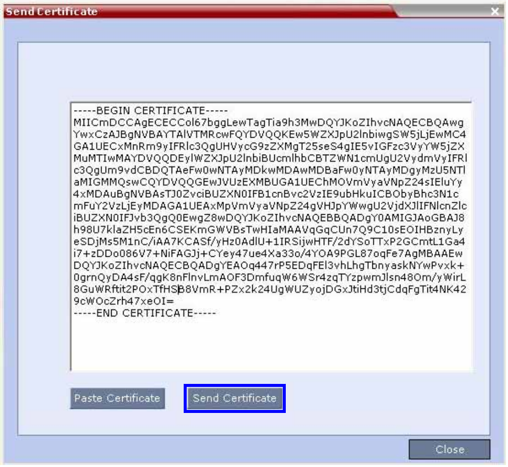
19 Open the *Certificate Authority* e-mail and **Copy** (**Ctrl + C**) the certificate information from the *Certificate Authority's* e-mail to the clipboard.

20 In the *IP Network Service - SIP Servers* dialog box, click the **Send Certificate** button. The *Send Certificate* dialog box opens.

21 Click **Paste Certificate** to paste the clipboard content into the *Send Certificate* dialog box.



22 Click the **Send Certificate** button to send the certificate to the Collaboration Server.



```
-----BEGIN CERTIFICATE-----
MIICmDCCAgECECCol67bggLewTagTia9h3MwDQYJKoZIhvcNAQECBQAwg
YwxCzAJBgNVBAYTAlVTMRcwFQYDVQQKEw5WZXJpU2lnbiwgSW5jLjEwMC4
GA1UECxMnRm9yIFRlc3QgUHVycG9zZXMgT25seS4gIE5vIGFzc3VyYW5jZX
MuMTIwMAYDVQQDEylWZXJpU2lnbiBUcmlhbCBTZWN1cmUgU2VydmVyIFRl
c3QgUm9vdCBDQTAeFw0wNTAyMDkwMDAwMDBaFw0yNTAyMDgyMzU5NTl
aMIGMMQswCQYDVQQGEwJVUzEXMBUGA1UEChMOVmVyaVNpZ24sIEluYy
4xMDAuBgNVBAsTJ0ZvciBUZXN0IFB1cnBvc2VzIE9ubHkuICBObyBhc3N1c
mFuY2VzLjEyMDAGA1UEAxMpVmVyaVNpZ24gVHJpYWwgU2VjdXJlIFNlcnZlc
iBUZXN0IFJvb3QgQ0EwgZ8wDQYJKoZIhvcNAQEBBQADgY0AMIGJAoGBAJ8
h98U7klaZH5cEn6CSEKmGWVBsTwHIaMAAVqGqCUn7Q9C10sEOIHBznyLy
eSDjMs5M1nC/iAA7KCASf/yHz0AdlU+1IRSijwHTF/2dYSoTTxP2GCmtL1Ga4
i7+zDDo086V7+NiFAGJj+CYey47ue4Xa33o/4YOA9PGL87oqFe7AgMBAAEw
DQYJKoZIhvcNAQECBQADgYEAOq447rP5EDqFEl3vhLhgTbnyaskNYwPvxk+
0grnQyDA4sF/qgK8nFlnvLmAOF3DmfuqW6WSr4zqTYzpwmJlsn48Om/yWirL
8GuWRftit2POxTfHSB8VmR+PZx2k24UgWUZyojDGxJtiHd3tjCdqFgTit4NK42
9cWOcZrh47xeOI=
-----END CERTIFICATE-----
```

23 Click the **Close** button.

24 In the *IP Network Service - SIP Servers* dialog box, complete the SIP Servers definitions.

25 Click **OK**.

The MCU validates the certificate.

— If the certificate is not valid, an error message is displayed.

— If the certificate matches the private key, and the task is completed, a confirmation message indicating that the certificate was created successfully is displayed.

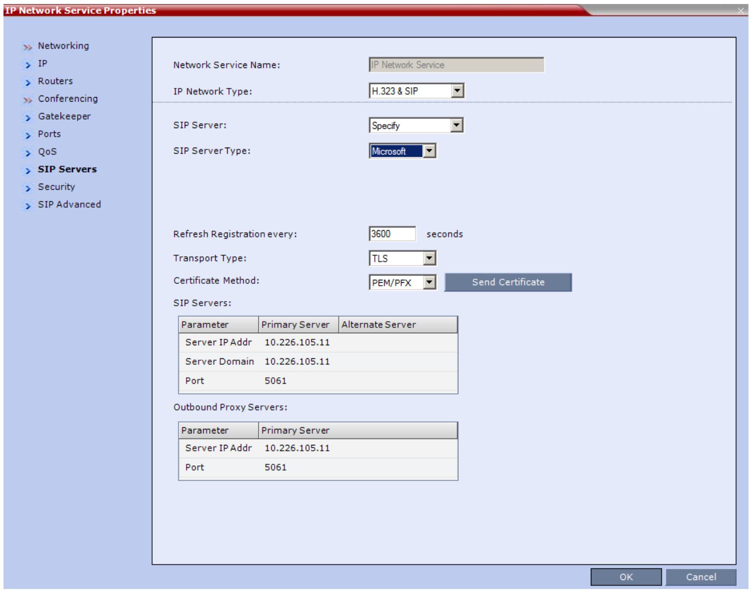
Once the certificate is installed in the Collaboration Server you can complete the definition procedure or continue with the Collaboration Server configuration for ICE dialing. For details, see "" on page **G-32**.

26 If no additional configuration is required, reset the Collaboration Server.

PFX Method - Sending the Certificate

The PFX certificate request is created in the Lync server. This certificate is received from the Certificate Authority it can be sent to the Collaboration Server, as described in the following procedure:

27 Click the **Send Certificate** button.



Network Service Name: IP Network Service

IP Network Type: H.323 & SIP

SIP Server: Specify

SIP Server Type: Microsoft

Refresh Registration every: 3600 seconds

Transport Type: TLS

Certificate Method: PEM/PFX

SIP Servers:

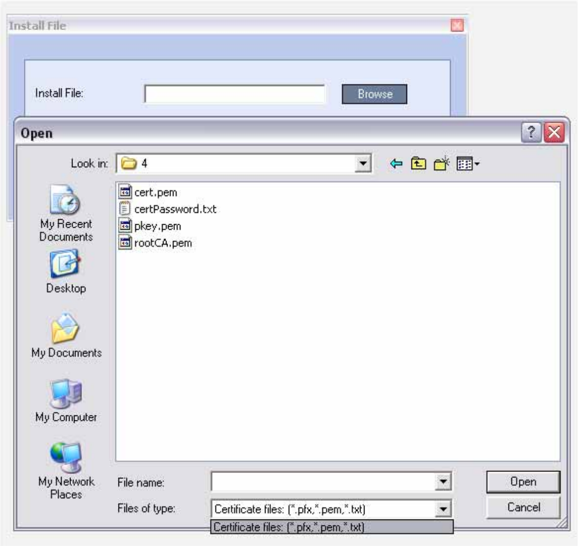
Parameter	Primary Server	Alternate Server
Server IP Addr	10.226.105.11	
Server Domain	10.226.105.11	
Port	5061	

Outbound Proxy Servers:

Parameter	Primary Server
Server IP Addr	10.226.105.11
Port	5061

The *Install File* dialog box opens.

28 Click the **Browse** button.
The *Open* dialog box appears, letting you select the certificate file(s) to send to the MCU.



Depending on the method used when the certificate file(s) were created, send the certificate file(s) to the Collaboration Server according to the contents of the file set that was created:

— The certificate files *pkey.pem*, *cert.pem* and a *certPassword.txt*. The files were created by a Certificate Authority and are sent as is to the Collaboration Server together with the required password contained in the *certPassword.txt* file.
This is the recommended method.

— The files *pkey.pem* and *cert.pem*. The certificate files were created by a Certificate Authority and are sent as is to the Collaboration Server.

— A **.pfx* file and a *certPassword.txt* file. The file *certPassword.txt* is manually created if the **.pfx* file was created by the OCS using a password. The **.pfx* file will be converted internally by the Collaboration Server using the password included in the certPassword.txt into three certificate files named *pkey.pem* and *cert.pem*.

— A **.pfx* file if the certificate file was created in the OCS without using a password. The **.pfx* file will be converted internally by the Collaboration Server into three certificate files named *pkey.pem* and *cert.pem*.

29 In the file browser, select all files to be sent in one operation according to the contents of the set:

— One ***.pfx** file, or

— Two files: one ***.pfx** file and **certPassword.txt**, or

— Three files:**pkey.pem, cert.pem** and **certPassword.txt**

30 Click **Open**.
The selected file(s) appear in the *Install Files* path.

31 Click **Install**.
The files are sent to the Collaboration Server and the *Install File* dialog box closes.

32 In the *Default IP Service - Networking IP* dialog box, click **OK**.

33 In the *Reset Confirmation* dialog box, click **No** to modify the required system flags before resetting the MCU, or click **Yes** if the flag was already set**.**

> Reset can be performed after setting the system flags (for example, setting the MS_ENVIRONMENT flag). Sometimes the system fails to read the *.pfx file and the conversion process fails, which is indicated by the active alarm "SIP TLS: Registration server not responding" and/or "SIP TLS: Registration handshake failure". Sending *.pfx file again, as described in this procedure and then resetting the system may resolve the problem.

Collaboration Server System Flag Configuration

Enabling the Microsoft Environment

The Collaboration Server can be installed in Microsoft R2 environments. To adjust the Collaboration Server behavior to the Microsoft environment in each release, system flags must be set.

To configure the system flags on the Polycom Collaboration Server system:

1 On the *Collaboration Server* menu, click **Setup > System Configuration**.

The *System Flags - MCMS_PARAMETERS_USER* dialog box opens.

2 Scroll to the flag **MS_ENVIRONMENT** and click it.
The *Edit Flag* dialog box is displayed.

3 In the *Value* field, enter **YES** to set the Collaboration Server SIP environment to Microsoft solution.

> Collaboration Server set to MS_ENVIRONMENT=YES supports SIP over TLS only and not over TCP.

4 Click **OK** to complete the flag definition.

5 When prompted, click **Yes** to reset the MCU and implement the changes to the system configuration. After system reset the Collaboration Server can register to the OCS server and make SIP calls.

> Sometimes the system fails to read the *.pfx file and the conversion process fails, which is indicated by the active alarm "SIP TLS: Registration server not responding" and/or "SIP TLS: Registration handshake failure". Sending *.pfx file again, as described in this procedure and then resetting the system may resolve the problem.

In some configurations, the following flags may require modifications when **MS_ENVIRONMENT** flag is set to YES:

Table G-2 Additional Microsoft Environment Flags in the Collaboration Server MCMS_PARAMETERS_USER Tab

Flag Name	Value and Description
SIP_FREE_VIDEO_RESOURCES	Default value in Microsoft environment: **NO.** When set to NO, video resources that were allocated to participants remain allocated to the participants as long as they are connected to the conference even if the call was changed to audio only. The system does not allocate the resources to other participants ensuring that the participants have the appropriate resources in case they want to return to the video call. The system allocates the resources according to the participant's endpoint capabilities, with a minimum of one CIF video resource. When this flag is set to YES, video ports are dynamically allocated or released according to the in the endpoint capabilities. For example, when an audio Only call is escalated to Video and vice versa or when the resolution is changed.
SIP_FAST_UPDATE_INTERVAL_ ENV	Default setting is **0** to prevent the Collaboration Server from automatically sending an Intra request to all SIP endpoints. Enter **n** (where n is any number of seconds other than 0) to let the Collaboration Server automatically send an Intra request to all SIP endpoints every n seconds. It is recommended to set the flag to 0 and modify the frequency in which the request is sent at the endpoint level (as defined in the next flag).

Setting the audio protocol for the Microsoft Client running on a single core PC

By default, Lync Clients are connected to conferences using the G.722.1 audio algorithm. However, when these clients are hosted on single processor workstations, they may experience audio quality problems when this algorithm is used.

The *System Flag* **FORCE_AUDIO_CODEC_FOR_MS_SINGLE_CORE** is used to force the use of a specific Audio algorithm such as G.711 when a *Lync Client* is detected as being hosted on a single core processor.

This flag can be set to:

- **AUTO** – No forcing occurs and the Collaboration Server negotiates a full set of Audio algorithm during capabilities exchange.
- **G711A/U** or **G722** – Set this flag value according to the hosting workstation capabilities. If the Collaboration Server detects single core host during capabilities exchange it will assign a *G.711* or *G.722* Audio algorithm according to the flag value.

Possible values: **AUTO, G711A, G711U, G722**

Default: **G711A**

Microsoft RTV (*Real Time Video*) protocol provides high quality video conferencing capability to *Microsoft OC* (*Office Communicator*) Client endpoints at resolutions up to *HD720p30*. Interoperability between *Polycom HDX* and *OCS* endpoints is improved.

Guidelines

- The *RTV* protocol is supported:
  - In *SIP* networking environments only
  - In CP mode only
- *OCS (Wave 13)* and *Lync Server (Wave 14)* clients are supported.
- *RTV* is supported in *Basic Cascade* mode.
- *RTV* is the default protocol for *OCS* endpoints and *Lync Server* clients connecting to a conference.
- *RTV* participants are supported in recorded conferences.
- *RTV* participant encryption is supported using the *SRTP* protocol.
- *Video Preview* is not supported for *RTV* endpoints.
- *Custom Slides* in *IVR Services* are not supported for *RTV* endpoints.
- *HD720p30* resolution is supported at bit rates greater than 600 kbps. The following table summarizes the resolutions supported at the various bit rates.

Table G-3 RTV - Resolution by Bit Rate

Resolution	Bitrate
QCIF	**Bitrate** <180kbps
CIF30	180kbps < **Bitrate** < 250kbps
VGA (SD30)	250kbps < **Bitrate** < 600kbps *
HD720p30	600kbps < **Bitrate** *

* Dependant on the PC's capability

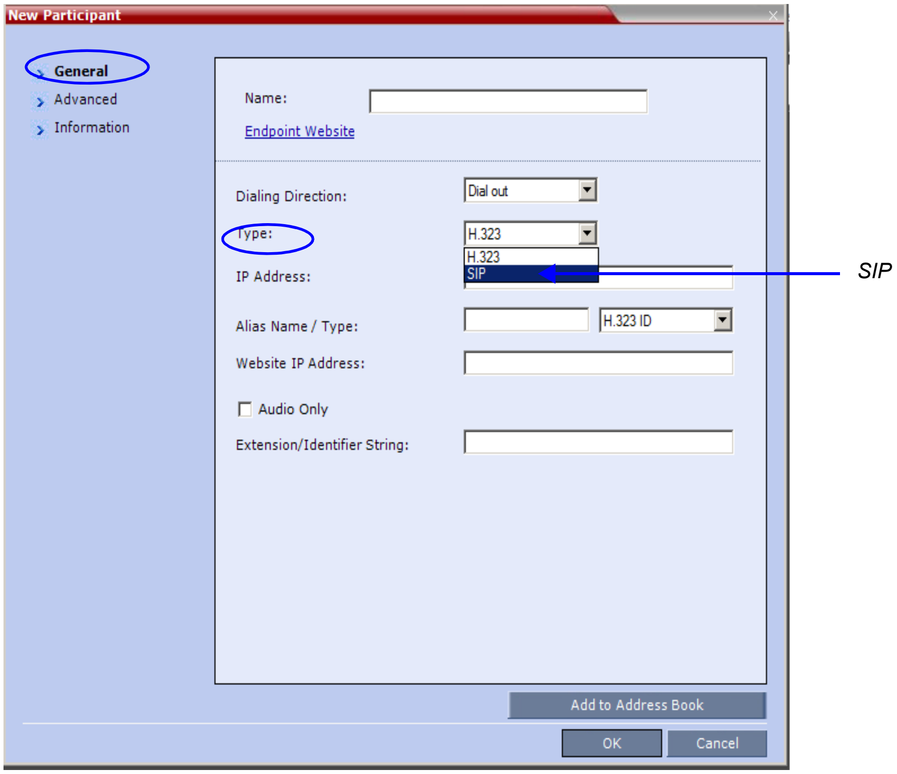
- *System Resource* usage is the same as for the *H.264* protocol. Table G-4 summarizes *System Resource* usage for each of the supported resolutions.

Table G-4 RTV - Resources by Resolution

Resolution	HD Video Resources Used
QCIF / CIF30	0.333
VGA (SD30) / W4CIF	0.5
HD720p30	1

Participant Settings

When defining a new participant or modifying an existing participant, select **SIP** as the participant's networking environment *Type* in the *New Participant* or *Participant Properties - General* tab.



The participants *Video Protocol* in the *New Participant* or *Participant Properties - Advanced* tab should be left at (or set to) its default value: **Auto**.

The **Auto** setting allows the video protocol to be negotiated according to the endpoint's capabilities:

- *OCS* endpoints and *Lync Server* clients connect to the conference using the *RTV* protocol.
- Other endpoints negotiate the video protocol in the following sequence: *H.264*, followed by *RTV*, followed by *H.263* and finally *H.261*.

Protocol Forcing

Selecting *H.264*, *RTV*, *H.263* or *H.261* as the *Video Protocol* results in endpoints that do not support the selected *Video Protocol* connecting as *Secondary* (audio only).

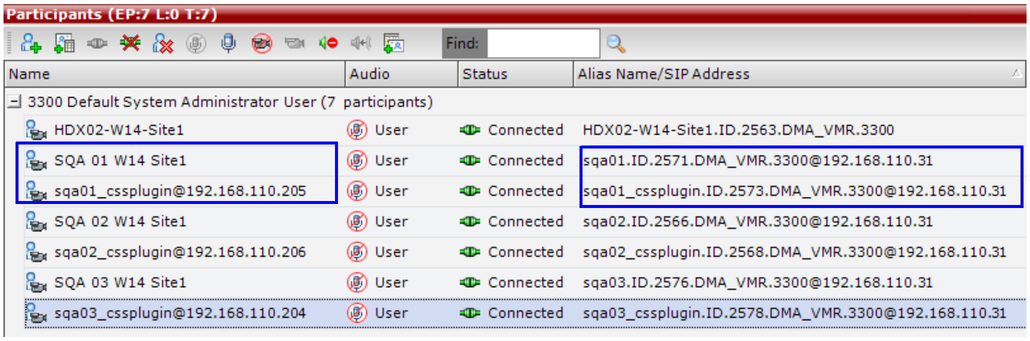
Sharing Content via the Polycom CSS Plug-in for Lync Clients

From version 8.1, Polycom CSS (Content Sharing Suite) Plug-in for Lync clients allows Lync clients to receive and send *Content* on a separate channel, without having to use the video channel. *Content* is transmitted using SIP BFCP.

When Lync clients connect, each endpoint is represented twice in the *RMX Manager* or *Collaboration Server Web Client*. One connection represents the actual Lync client, while the second connection represents the content channel via the Polycom plug-in.

The name of the plug-in "participant" is derived from the name of the Lync client with the suffix "_cssplugin".

When a Lync client connects to a conference, the plug-in connects automatically, regardless of whether the Lync client dials into a conference or is called from the MCU.



Participants (EP:7 L:0 T:7)

Name	Audio	Status	Alias Name/SIP Address
3300 Default System Administrator User (7 participants)			
HDX02-W14-Site1	User	Connected	HDX02-W14-Site1.ID.2563.DMA_VMR.3300
SQA 01 W14 Site1	User	Connected	sqa01.ID.2571.DMA_VMR.3300@192.168.110.31
sqa01_cssplugin@192.168.110.205	User	Connected	sqa01_cssplugin.ID.2573.DMA_VMR.3300@192.168.110.31
SQA 02 W14 Site1	User	Connected	sqa02.ID.2566.DMA_VMR.3300@192.168.110.31
sqa02_cssplugin@192.168.110.206	User	Connected	sqa02_cssplugin.ID.2568.DMA_VMR.3300@192.168.110.31
SQA 03 W14 Site1	User	Connected	sqa03.ID.2576.DMA_VMR.3300@192.168.110.31
sqa03_cssplugin@192.168.110.204	User	Connected	sqa03_cssplugin.ID.2578.DMA_VMR.3300@192.168.110.31

Guidelines

- The maximum resolution for content sharing via the Polycom CSS plug-in is HD720p5.
- The Polycom CSS plug-in supports H.263 and H.264 video protocols for content sharing.
- SVC-enabled endpoints use the AVC (H.264) protocol for sharing content.
- Content can be shared between different types of endpoints, using different network protocols (H.323, SIP and ISDN/PSTN).
- *TIP* content is not supported.
- Lync 2013 is supported.
- *ICE* is not supported.

Configuring the MCU for Content Sharing via the Polycom CSS Plug-in

You can configure the MCU for content sharing via the Polycom CSS plug-in by setting the following parameters:

- Setting the **BLOCK_CONTENT_LEGACY_FOR_LYNC** system flag
- Setting the Content parameters in the conference Profile

Setting the System Flag

By configuring the system flag **BLOCK_CONTENT_LEGACY_FOR_LYNC** you control the system behavior in an environment where some Lync clients use the Polycom CSS plug-in and some do not. This flag must be manually added to the system configuration to change its value.

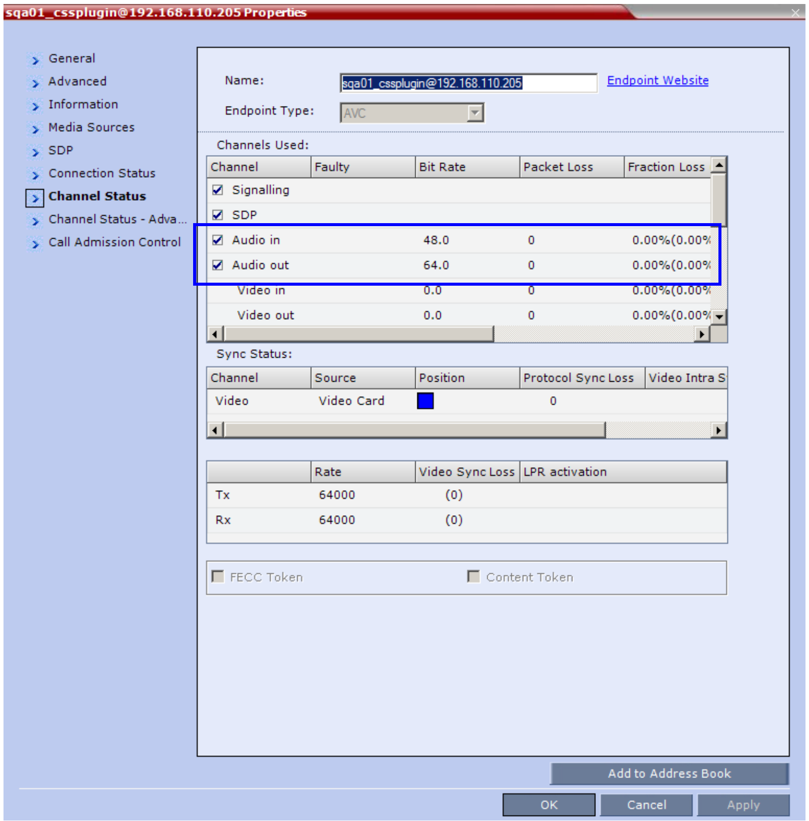
- When set to **NO** (default), *Content* is sent to all Lync clients over the video channel, including those with the Polycom CSS plug-in installed, even when the *Send Content to Legacy Endpoints* is disabled.
  Other, non-Lync legacy endpoints will not be affected by this flag and will receive content according to the *Send Content to Legacy Endpoints* settings in the conference *Profile*.
- When set to **YES**, *Content* is not sent to Lync clients over the video channel including those with the Polycom CSS plug-in installed, even when the *Send Content to Legacy Endpoints* is enabled.
  Other, non-Lync legacy endpoints will not be affected by this flag and will receive content according to the *Send Content to Legacy Endpoints* settings in the conference *Profile*.

Conference Profile Settings

Content is shared in a video switching mode. Therefore, when a Lync client connects to the conference via the Polycom CSS plug-in, the content resolution will be adjusted to the maximum content rate possible by the Lync client, up to a maximum of **720p 5fps** in all line rates, even if you select a higher content rate and resolution in the *Conference Profile*.

Monitoring the Participant connection

- Under properties of the participant representing the CSS plug-in, *Channel Status,* audio channels are shown, but audio is not used in the plug-in. The information can be ignored.



Name: sqa01_cssplugin@192.168.110.205

Endpoint Type: AVC

Channels Used:

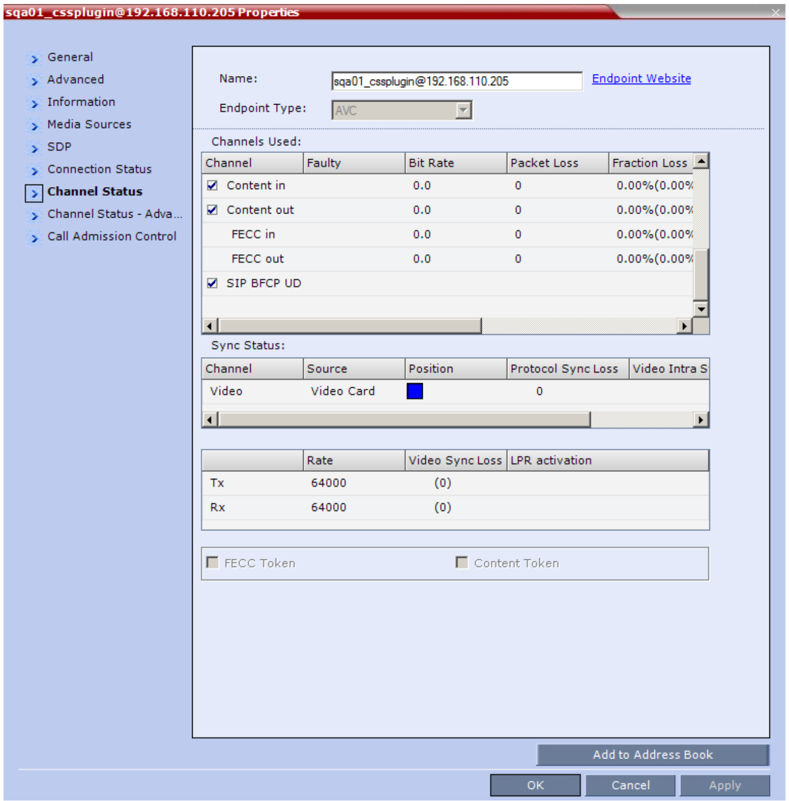
Channel	Faulty	Bit Rate	Packet Loss	Fraction Loss
☑ Signalling				
☑ SDP				
☑ Audio in		48.0	0	0.00%(0.00%
☑ Audio out		64.0	0	0.00%(0.00%
Video in		0.0	0	0.00%(0.00%
Video out		0.0	0	0.00%(0.00%

Sync Status:

Channel	Source	Position	Protocol Sync Loss	Video Intra S
Video	Video Card	🟦	0	

	Rate	Video Sync Loss	LPR activation
Tx	64000	(0)	
Rx	64000	(0)	

□ FECC Token □ Content Token



sqa01_cssplugin@192.168.110.205 Properties

Name: sqa01_cssplugin@192.168.110.205 Endpoint Website

Endpoint Type: AVC

Channels Used:

Channel	Faulty	Bit Rate	Packet Loss	Fraction Loss
☑ Content in		0.0	0	0.00%(0.00%
☑ Content out		0.0	0	0.00%(0.00%
FECC in		0.0	0	0.00%(0.00%
FECC out		0.0	0	0.00%(0.00%
☑ SIP BFCP UD				

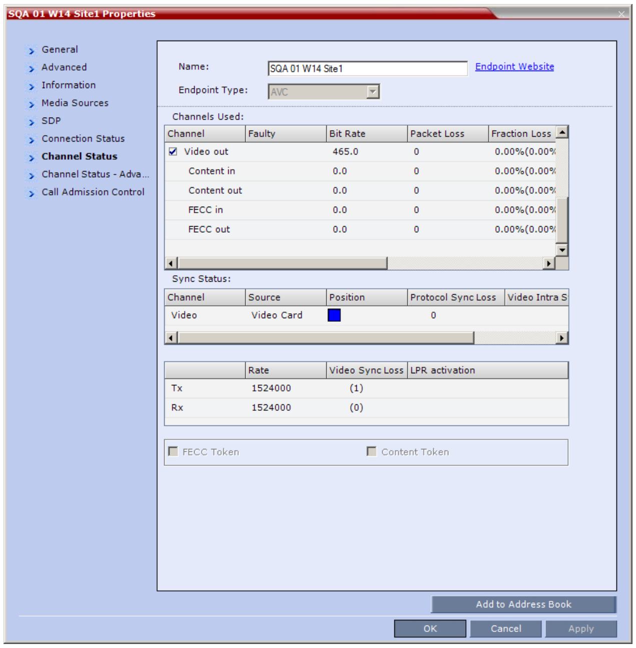
Sync Status:

Channel	Source	Position	Protocol Sync Loss	Video Intra S
Video	Video Card	🟦	0	

	Rate	Video Sync Loss	LPR activation
Tx	64000	(0)	
Rx	64000	(0)	

☐ FECC Token ☐ Content Token

- The properties of the Lync client are those of a video participant. However, the *Content* channel will show 0 as there is no content channel.



SQA 01 W14 Site1 Properties

Name: SQA 01 W14 Site1 Endpoint Website

Endpoint Type: AVC

Channels Used:

Channel	Faulty	Bit Rate	Packet Loss	Fraction Loss
☑ Video out		465.0	0	0.00%(0.00%
Content in		0.0	0	0.00%(0.00%
Content out		0.0	0	0.00%(0.00%
FECC in		0.0	0	0.00%(0.00%
FECC out		0.0	0	0.00%(0.00%

Sync Status:

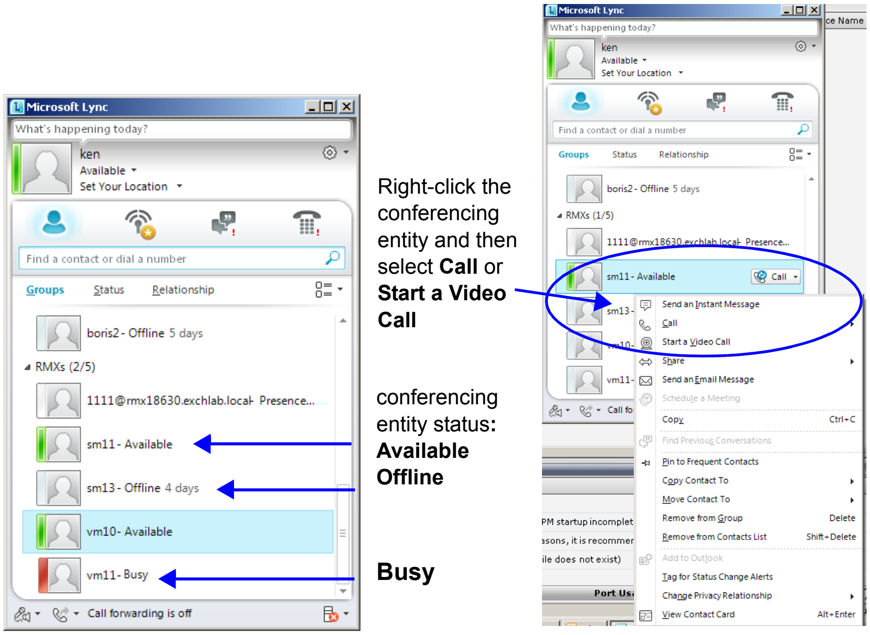
Channel	Source	Position	Protocol Sync Loss	Video Intra S
Video	Video Card	🟦	0	

	Rate	Video Sync Loss	LPR activation
Tx	1524000	(1)	
Rx	1524000	(0)	

☐ FECC Token ☐ Content Token

Adding Presence to Conferencing Entities in the Buddy List

Registration of conferencing entities (Meeting Rooms, Entry Queues and SIP Factories) with the SIP server adds these conferencing entities to the buddy list with their presence. It enables the LYNC Server client users to see the availability status (Available, Offline, or Busy) of these conferencing entities and connect to them directly from the buddy list.



Right-click the conferencing entity and then select **Call** or **Start a Video Call**

conferencing entity status:
Available
Offline

Busy

Guidelines

- Registration with Presence of up to 100 conferencing entities to a single SIP Server is supported. When this number is exceeded, the additional conferencing entity may appear to be successfully registered but the presence status will be shown as 'Offline' in Lync for any entities beyond the limit.
- Lync endpoints cannot connect to conferencing entities that their presence is "offline".
- The Conferencing Entity: Meeting Room, Entry Queue, SIP Factory (*Routing Name*) has to be added to the Active Directory as a User.

  Make sure that a unique name is assigned to the conferencing entity and it is not already used for another user account in the Active Directory.
- The conferencing entity name must not include any upper case letters or special characters: @ # $ % ^ & * ( ) _ - = + | } { : " \ ] [; / ? > < , . (space) ~.
- When the MCU system is shutting down while a Meeting Room is still active, as indicated by its presence, the status remains active for 10 minutes during which Lync endpoints cannot connect to the Meeting Room. After 10 minutes, the Meeting Room Status changes to "offline".
- Registration of the conferencing entity is defined in the Conference Profile (and not in the IP Network Service), enabling you to choose the conferencing entity to register.

- In *Multiple Networks* configuration, an IP Network Service that is enabled for registration in a Conference Profile cannot be deleted.

Enabling the Registration of the Conferencing Entities

The creation of the various conferencing entities is described in the following chapters:

- "*Meeting Rooms*" on page **6-1**
- "*Entry Queues, Ad Hoc Conferences and SIP Factories*" on page **7-1**

Registration with presence of conferencing entities with the SIP Server is enabled by performing the following processes:

- Creating an Active Directory Account for the Conferencing Entity.
- Enabling the Conferencing Entity User Account for Lync Server
- Defining the Microsoft SIP Server in the IP Network Service
- Enabling Registration in the Conference Profile

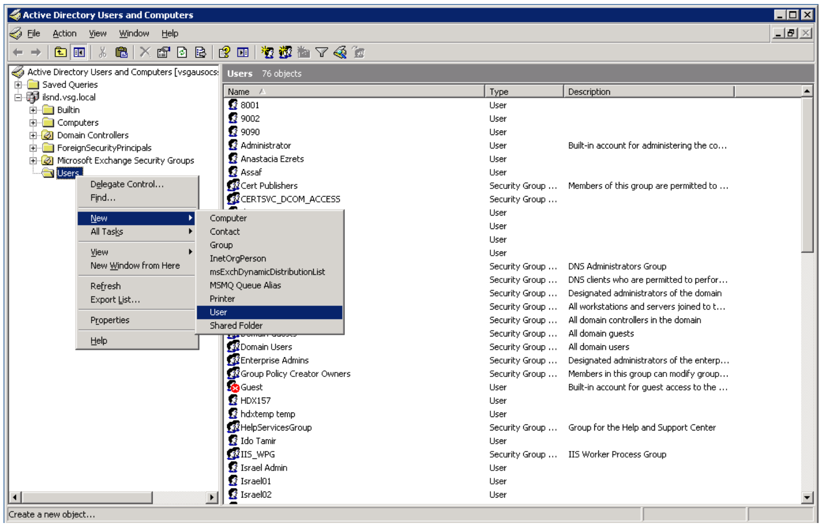
Creating an Active Directory Account for the Conferencing Entity

The User account created for the Conferencing entity is used for registration with the Lync server and to automatically synchronize with the STUN and relay (Edge) servers.

To add the conferencing entity user to the Active Directory:

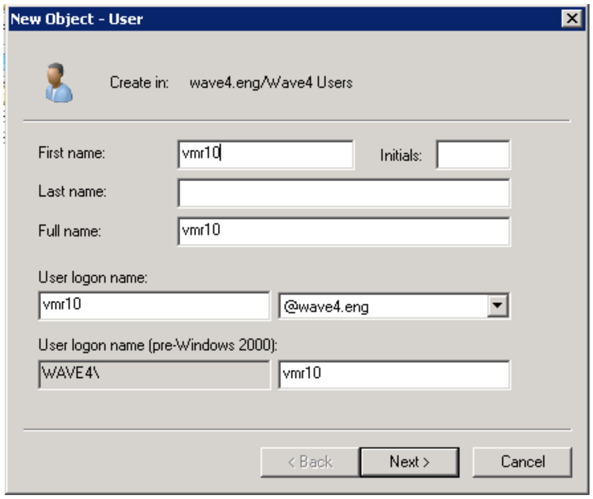
1 Go to **Start > Run** and enter **dsa.msc** to open the *Active Directory Users and Computers* console.

2 In the console tree, select **Users > New > User**.



Name	Type	Description
8001	User	
9002	User	
9090	User	
Administrator	User	Built-in account for administering the co...
Anastacia Ezrets	User	
Assaf	User	
Cert Publishers	Security Group ...	Members of this group are permitted to ...
CERTSVC_DCOM_ACCESS	Security Group ...	
	User	
	User	
	User	
	User	
	Security Group ...	DNS Administrators Group
	Security Group ...	DNS clients who are permitted to perfor...
	Security Group ...	Designated administrators of the domain
	Security Group ...	All workstations and servers joined to t...
	Security Group ...	All domain controllers in the domain
	Security Group ...	All domain guests
Domain Users	Security Group ...	All domain users
Enterprise Admins	Security Group ...	Designated administrators of the enterp...
Group Policy Creator Owners	Security Group ...	Members in this group can modify group...
Guest	User	Built-in account for guest access to the ...
HDX157	User	
hdxtemp temp	User	
HelpServicesGroup	Security Group ...	Group for the Help and Support Center
Ido Tamir	User	
IIS_WPG	Security Group ...	IIS Worker Process Group
Israel Admin	User	
Israel01	User	
Israel02	User	

3 In the *New User* wizard, define the following parameters:



New Object - User

Create in: wave4.eng/Wave4 Users

First name: vmr10

Initials:

Last name:

Full name: vmr10

User logon name:

vmr10 @wave4.eng

User logon name (pre-Windows 2000):

WAVE4\ vmr10

Table G-5 *Active Directory - New User Parameters for the Collaboration Server*

Field	Description
First Name	Enter the name of the conferencing entity user. This name will appear in the buddy list of the Office Communication Server or Lync server. For example, vmr10. **Notes:** • This name must be the identical to the **Routing Name** assigned to the conferencing entity in the Collaboration Server system. It must also be the *User Login Name* in the Active Directory. • The name can include only lower case characters and/or numbers.
Full Name	Enter the same name as entered in the *First Name* field.
User Login Name	Enter the same name as entered in the *First Name* field and select from the drop down list the domain name for this user. It is the domain name defined for the Office Communication Server or Lync server.

4 Click **Next**.

5 Enter the password that complies with the Active Directory conventions and confirm the password.

6 Select the options: **User cannot change password** and **Password never expires**.
Clear the other options.

7 Click **Next**.
The system displays summary information.

8 Click **Finish**.

The new User is added to the Active Directory *Users* list.

9 Repeat for each Collaboration Server conferencing entity.

Enabling the Conferencing Entity User Account for Lync Server

The new Conferencing Entity user must be enabled for registration with the Lync Server.

To enable the Conferencing Entity User Account for Lync Server:

1 On the computer running the Lync Server 2010, go to **Start->All Programs->Microsoft Lync Server 2010>Lync Server Control Panel**.

Windows Security window opens.

2 Enter your User name and Password as configured in the Lync Server and click OK. The *Microsoft Lync Server 2010 Control Panel* window opens.

3 Click the **Users** tab.

4 In the *User Search* pane, click the **Enable Users** heading.

The *New Lync Server User* pane opens.

5 Click the **Add** button.

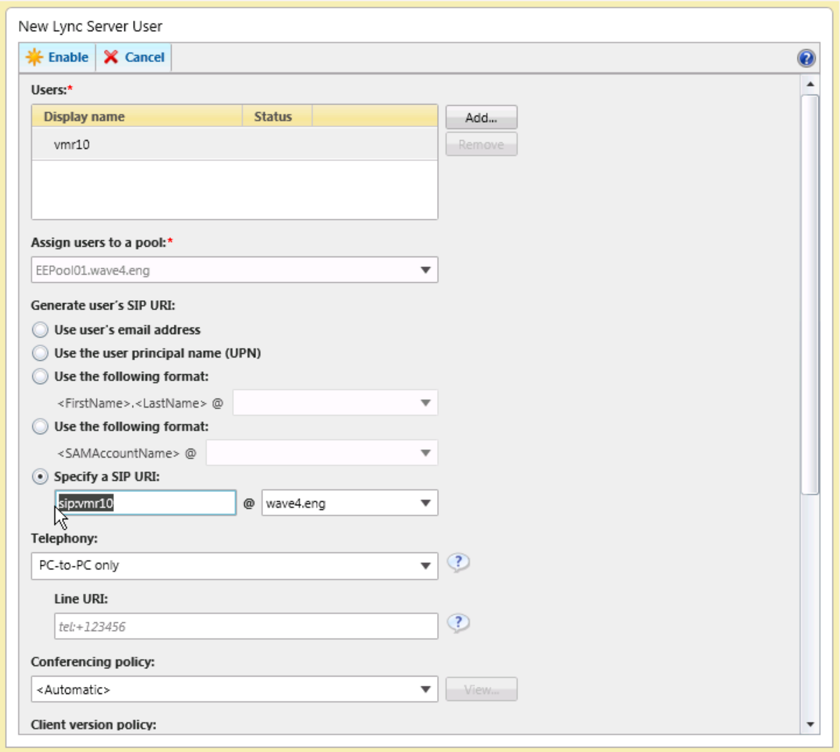
The *Select from Active Directory* dialog box opens.

6 Enter the conferencing entity user name as defined in the Active Directory, and then click the **Find** button.

The requested user is listed in the *Select From Active Directory* dialog box.

7 Select the listed user (conferencing entity user) and click **OK**.
The selected user appears in the *New Lync Server User* pane.

8 Select the following parameters:



Users:*

Display name	Status
vmr10	

Assign users to a pool:* EEPool01.wave4.eng

Generate user's SIP URI:

○ Use user's email address

○ Use the user principal name (UPN)

○ Use the following format: <FirstName>.<LastName> @

○ Use the following format: <SAMAccountName> @

● Specify a SIP URI: sip:vmr10 @ wave4.eng

Telephony: PC-to-PC only

Line URI: tel:+123456

Conferencing policy: <Automatic>

Client version policy:

— In *Assign users to a pool* field, select the required pool.

— In the *Generate user SIP URI*, define the SIP URI of the conferencing entity using one of the following methods:

- Select the **Specify a SIP URI** option and enter the conferencing entity user portion of SIP URI defined in the active directory. This SIP URI must match the conferencing entity Routing Name configured in Collaboration Server. For example, for the meeting room account **sip:vmr10@wave4.eng**, use only the **vmr10** portion of the address.

or

- Select the **Use the user principal name (UPN)** option.

9 Click the **Enable** button.
The selected user appears as enabled in the *User Search* pane.

Defining the Microsoft SIP Server in the IP Network Service

To enable the registration of the conferencing entities the *SIP Server Type* must be set to **Microsoft** and the Lync Server properties in the *IP Network Service - SIP Servers* dialog box.

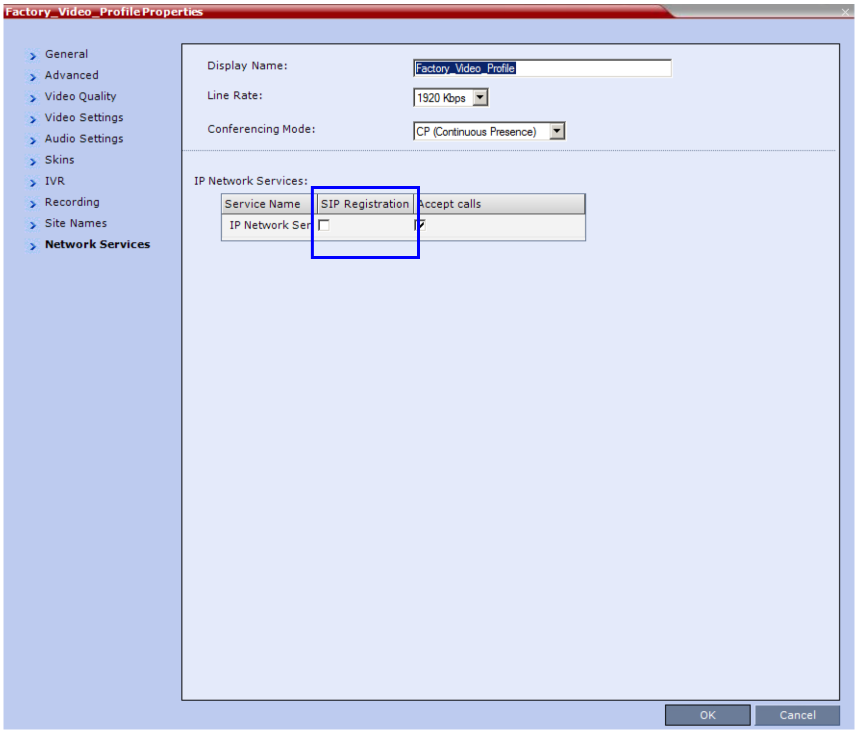
For more details, see "*Configuring the Collaboration Server IP Network Service"* on page **G-7**.

Enabling Registration in the Conference Profile

Registration of conferencing entities such as ongoing conferences, *Meeting Rooms, Entry Queues, SIP Factories* and *Gateway Sessions* with *SIP* servers is done per conferencing entity. This allows better control on the number of entities that register with each *SIP* server.

Selective registration is enabled by assigning a conference Profile in which registration is enabled to the conferencing entities that require registration. Assigning a conference Profile in which registration is disabled (registration check box is cleared) to conferencing entities will prevent them from registering. By default, Registration is disabled in the Conference Profile, and must be enabled in Profiles assigned to conferencing entities that require registration.

Registration can be enabled in the *New Profile - Network Services* dialog box:



Display Name: Factory_Video_Profile

Line Rate: 1920 Kbps

Conferencing Mode: CP (Continuous Presence)

IP Network Services:

Service Name	SIP Registration	Accept calls
IP Network Ser	□	☑

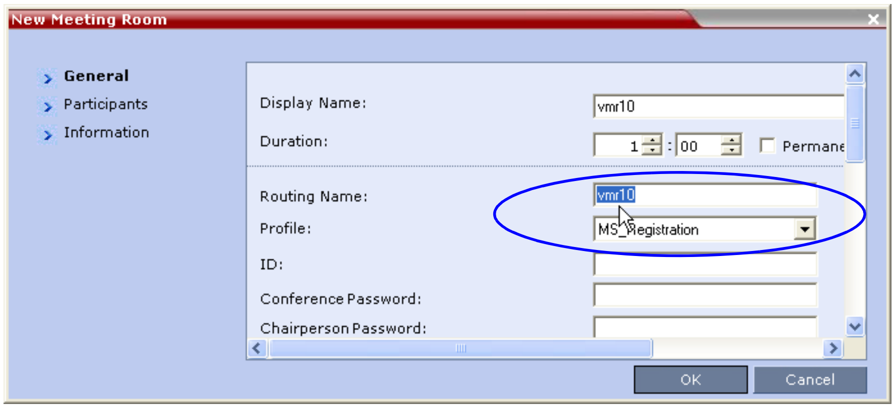
Table G-6 *Profile Properties - Network Services*

Parameter	Description
IP Network Services:	
Service Name	This column lists all the defined *Network Services,* one or several depending on the system configuration (single Network or Multiple Networks).
SIP Registration	To register the conferencing entity to which this profile is assigned, with the SIP Server defined for that *Network Service*, click the *SIP Registration* check box of that *Network Service*.
Accept Calls	To prevent dial in participants from connecting to a conferencing entity when connecting via a certain *Network Service*, clear the *Accept Calls* check box of that *Network Service*.

Verifying the Collaboration Server Conferencing Entity Routing Name and Profile

Collaboration Server conferencing entity can be dialed directly from the buddy list of the Lync client if its routing name matches the user name of Active Directory account you created and Registration is enabled in the Conference Profile assigned to it.

- To ensure that the Collaboration Server meeting room or conferencing entity is properly configured for registration the following parameters must be defined:
  - The user name on the conferencing entity in Active Directory account must be identical to its **Routing Name** on the Collaboration Server.
    For example, if the SIP URI in the Active Directory is ***`sip:vmr10@wave4.eng`***, it must be defined as ***`vmr10`*** in the *Routing Name* field of that Collaboration Server conferencing entity.



Display Name: vmr10
Duration: 1 : 00 □ Permane
Routing Name: vmr10
Profile: MS_Registration
ID:
Conference Password:
Chairperson Password:

  - In the *Profile* field, make sure that a conference Profile that has been enabled for SIP registration is selected.

Monitoring the Registration Status of a Conferencing Entity in the Collaboration Server Web Client or RMX Manager Application

The Status of the SIP registration can be viewed in the appropriate conferencing Entity list or when displaying its properties.

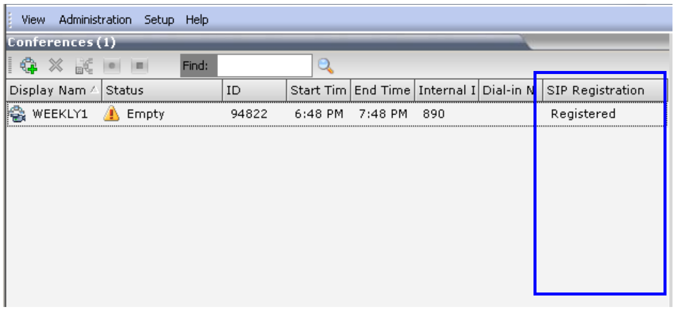
Conferencing Entity List

The list of conferencing entity includes an additional column - *SIP Registration*, which indicates the status of its registration with the SIP server. The following statuses are displayed:

- **Not configured** - Registration with the SIP Server was not enabled in the Conference Profile assigned to this conferencing Entity. In Multiple Networks configuration, If one service is not configured while others are configured and registered, the status reflects the registration with the configured Network Services. The registration status with each SIP Server can be viewed in the *Properties - Network Services* dialog box of each conferencing entity.

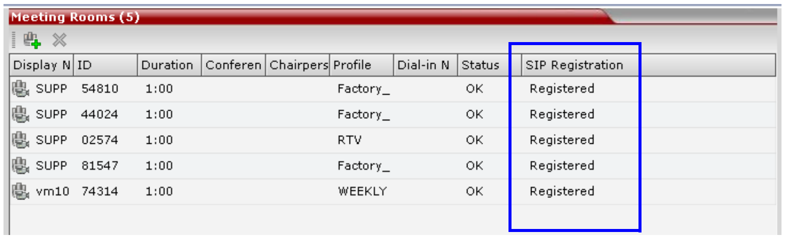
  When SIP registration is not enabled in the conference profile, the Collaboration Server's registering to SIP Servers will each register with an URL derived from its own signaling address. This unique URL replaces the non-unique URL, dummy_tester, used in previous versions.
- **Failed** - Registration with the SIP Server failed.
  This may be due to incorrect definition of the SIP server in the IP Network Service, or the SIP server may be down, or any other reason the affects the connection between the Collaboration Server or the SIP Server to the network.
- **Registered** - the conferencing entity is registered with the SIP Server.

- **Partially Registered** - This status is available only in Multiple Networks configuration, when the conferencing entity failed to register to all the required Network Services (if more than one Network Service was selected for Registration). The registration status with each SIP Server can be viewed in the *Properties - Network Services* dialog box of each conferencing entity.



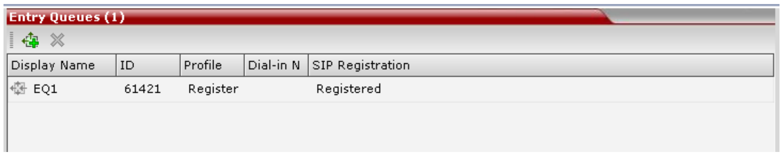
Display Nam	Status	ID	Start Tim	End Time	Internal I	Dial-in N	SIP Registration
WEEKLY1	Empty	94822	6:48 PM	7:48 PM	890		Registered

Figure G-1 *Ongoing Conferences list - SIP Registration*



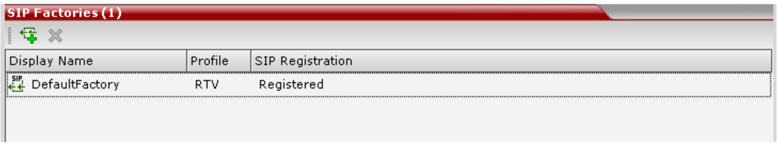
Display N	ID	Duration	Conferen	Chairpers	Profile	Dial-in N	Status	SIP Registration
SUPP	54810	1:00			Factory_		OK	Registered
SUPP	44024	1:00			Factory_		OK	Registered
SUPP	02574	1:00			RTV		OK	Registered
SUPP	81547	1:00			Factory_		OK	Registered
vm10	74314	1:00			WEEKLY		OK	Registered

Figure G-2 *Meeting Rooms list - SIP Registration*



Display Name	ID	Profile	Dial-in N	SIP Registration
EQ1	61421	Register		Registered

Figure G-3 *Entry Queues list - SIP Registration*

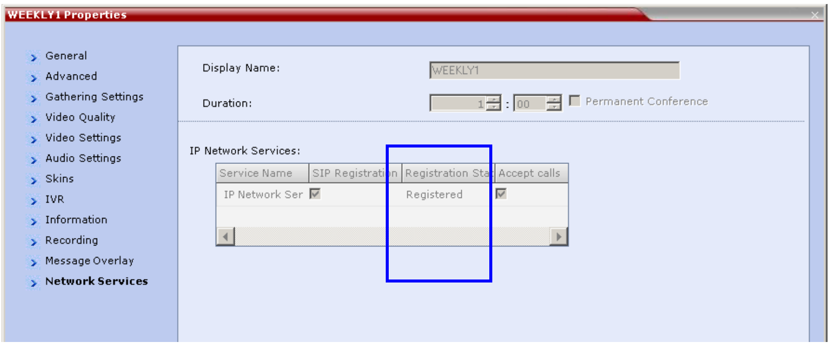


Display Name	Profile	SIP Registration
DefaultFactory	RTV	Registered

Figure G-4 *SIP Factories list - SIP Registration*

Conferencing Entity Properties

Registration status is reflected in the *Properties - Network Services* dialog box:



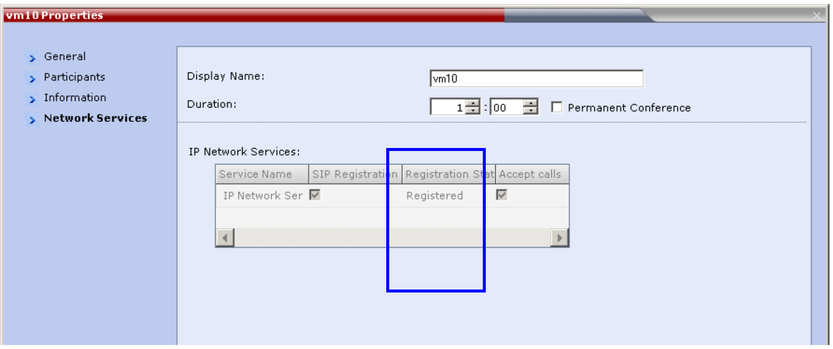
Display Name: WEEKLY1

Duration: 1 : 00 □ Permanent Conference

IP Network Services:

Service Name	SIP Registration	Registration Sta	Accept calls
IP Network Ser	☑	Registered	☑

Figure G-5 *Ongoing conference Properties - Network Services - SIP Registration*



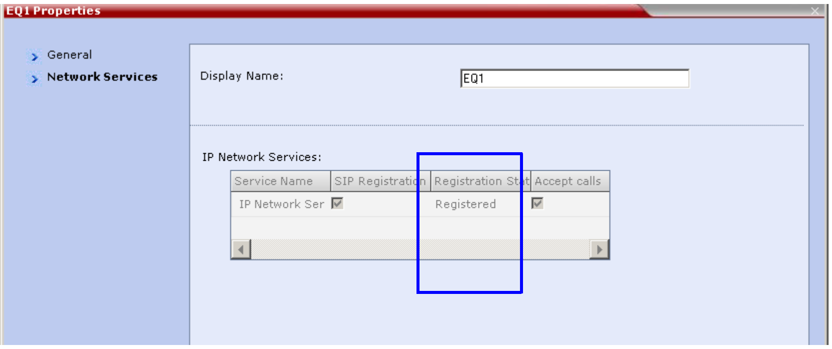
Display Name: vm10

Duration: 1 : 00 □ Permanent Conference

IP Network Services:

Service Name	SIP Registration	Registration Stat	Accept calls
IP Network Ser	☑	Registered	☑

Figure G-6 *Meeting Room Properties - Network Services - SIP Registration*



Display Name: EQ1

IP Network Services:

Service Name	SIP Registration	Registration Stat	Accept calls
IP Network Ser	☑	Registered	☑

Figure G-7 *Entry Queue Properties - Network Services - SIP Registration*

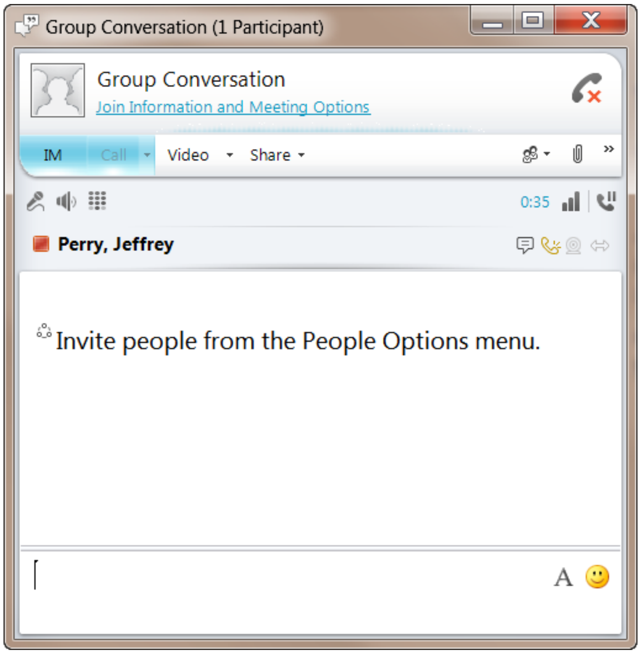
Connecting a Collaboration Server Meeting Room to a Microsoft AV-MCU Conference

Microsoft Lync users can connect an Collaboration Server Meeting Room to a conference running on the Microsoft A/V MCU. This allows Collaboration Server Lync users to connect with a conference in progress on the A/V MCU and be an active participant in the conference. The connection to the A/V MCU is the same configuration as a cascading conference between multiple Collaboration Server MCUs.

To connect to an A/V MCU conference:

1 From the Menu bar, click **Meet Now** to create an ad-hoc conference.

The Group Conversation dialog box is displayed.


Group Conversation (1 Participant)
Group Conversation
Join Information and Meeting Options
IM
Call
Video
Share
0:35
Perry, Jeffrey
Invite people from the People Options menu.
A

2 From the Contacts List on Lync, drag a Virtual Meeting Room (VMR) into the Group Conversation list.

After the Virtual Meeting Room is connected on Lync, an invitation is sent from the A/V MCU to the Collaboration Server using the Centralized Conference Control Protocol (CCCP). The Collaboration Server responds and triggers a standard SIP invite sent from the A/V MCU to the Collaboration Server.

Multiple participants can now connect to both the Collaboration Server Meeting Room and the A/V MCU, and participate in a cascaded conference.

When a conference begins with Audio Only, a Lync user cannot add video to the conference after the VMR is connected to the conference. The conference will remain as Audio Only.

Active Alarms and Troubleshooting

Active Alarms

The following active alarms may be displayed in the Collaboration Server *System Alerts* pane when the Collaboration Server is configured for integration in the OCS environment:

Table G-7 New Active Alarms

Alarm Code	Alarm Description
SIP TLS: Failed to load or verify certificate files	This alarm indicates that the certificate files required for SIP TLS could not be loaded to the Collaboration Server. Possible causes are: • Incorrect certificate file name. Only files with the following names can be loaded to the system: rootCA.pem, pkey.pem, cert.pem and certPassword.txt • Wrong certificate file type. Only files of the following types can be loaded to the system: rootCA.pem, pkey.pem and cert.pem and certPassword.txt • The contents of the certificate file does not match the system parameters
SIP TLS: Registration transport error	This alarm indicates that the communication with the SIP server cannot be established. Possible causes are: • Incorrect IP address of the SIP server • The SIP server listening port is other than the one defined in the system • The OCS services are stopped **Note:** Sometimes this alarm may be activated without real cause. Resetting the MCU may clear the alarm.
SIP TLS: Registration handshake failure	This alarm indicates a mismatch between the security protocols of the OCS and the Collaboration Server, preventing the Registration of the Collaboration Server to the OCS.

Table G-7 New Active Alarms (Continued)

Alarm Code	Alarm Description
SIP TLS: Registration server not responding	This alarm is displayed when the Collaboration Server does not receive a response from the OCS to the registration request in the expected time frame. Possible causes are: • The Collaboration Server FQDN name is not defined in the OCS pool, or is defined incorrectly. • The time frame for the expected response was too short and it will be updated with the next data refresh. The alarm may be cleared automatically the next time the data is refreshed. Alternatively, the OCS Pool Service can be stopped and restarted to refresh the data. • The Collaboration Server FQDN name is not defined in the DNS server. Ping the DNS using the Collaboration Server FQDN name to ensure that the Collaboration Server is correctly registered to the DNS.
SIP TLS: Certificate has expired	The current TLS certificate files have expired and must be replaced with new files.
SIP TLS: Certificate is about to expire	The current TLS certificate files will expire shortly and will have to be replaced to ensure the communication with the OCS.
SIP TLS: Certificate subject name is not valid or DNS failed to resolve this name	This alarm is displayed if the name of the Collaboration Server in the certificate file is different from the FQDN name defined in the OCS. **Note:** Occasionally this alarm may be activated without real cause. Resetting the MCU may clear the alarm.

Troubleshooting

- At the end of the installation and configuration process, to test the solution and the integration with the OCS, create an ongoing conference with two participants, one dial-in and one dial-out and connect them to the conference.
- If the *active* Alarm *"SIP TLS: Registration server not responding"* is displayed, stop and restart the OCS Pool Service.
- If the communication between the OCS and the Collaboration Server cannot be established, one of the possible causes can be that the Collaboration Server FQDN name is defined differently in the DNS, OCS and Collaboration Server. The name must be defined identically in all three devices, and defined as type A in the DNS. The definition of the Collaboration Server FQDN name in the DNS can be tested by pinging it and receiving the Collaboration Server signaling IP from the DNS in return.
- The communication between the OCS and the Collaboration Server can be checked in the Logger files:
  - — SIP 401/407 reject messages indicate that the Collaboration Server is not configured as Trusted in the OCS and must be configured accordingly.
  - — SIP 404 reject indication indicates that the connection to the OCS was established successfully.

Known Issues

- Selecting **Pause my Video** in OC client causes the call to downgrade to audio only call if the call was not in Audio Only mode at all (the call was started as a video call).

  If the call is started as an audio only call and video is added to it, or if the call was started as video call and during the call it was changed to Audio Only and back to video, selecting *Pause my Video* will suspend it as required.
- Rarely, the OC client disconnects after 15 minutes. The OC client can be reconnected using the same dialing method in which they were previously connected (dial-in or dial-out).
- Rarely, all SIP endpoints disconnect at the same time. The SIP endpoint can be reconnected using the same dialing method in which they were previously connected (dial-in or dial-out).

Polycom Solution Support

Polycom Implementation and Maintenance services provide support for Polycom solution components only. Additional services for supported third-party Unified Communications (UC) environments integrated with Polycom solutions are available from Polycom Global Services and its certified Partners. These additional services will help customers successfully design, deploy, optimize and manage Polycom visual communications within their UC environments.

Professional Services for Microsoft Integration is mandatory for Polycom Conferencing for Microsoft Outlook and Microsoft Office Communications Server integrations. For additional information and details please see http://www.polycom.com/services/professional_services/index.html or contact your local Polycom representative.

Appendix H

SIP RFC Support

Table H-1 SIP RFC Support in RealPresence Collaboration Server Systems

SIP RFC	Description	Note
1321	MD5	
2032	RTP Payload for H.261	
2205	RSVP	
2327	Session Description Protocol (SDP)	
2429	RTP Payload for H.263+	
2833	RTP Payload for DTMF	
2617	HTTP Authentication	
2976	SIP Info Method	
3261	SIP	
3264	Offer/Answer Model	
3265	SIP Specific Event Notification	Limited support
3311	SIP Update Method	
3515	SIP Refer Method	Limited support
3550	RTP	
3551	RTP Profile for Audio/Video	
3711	SRTP	
3890	Transport Independent Bandwidth Modifier for SDP	
3891	SIP Replaces header	Limited support
3892	SIP Referred-by Mechanism	Limited support
3984	RTP Payload format for H.264	
4028	Session Timers in SIP	
4145	TCP Media Transport in SDP	
4566	Session Description Protocol (SDP)	

Table H-1 SIP RFC Support in RealPresence Collaboration Server Systems (Continued)

SIP RFC	Description	Note
4568	SDP Security Descriptions	
4573	H.224 RTP Payload (FECC)	
4574	SDP Label Attribute	
4582	Binary Floor Control Protocol (BFCP)	
4583	SDP for BFCP	
4796	SDP Content Attribute	
5168	XML Schema for Media Control (Fast Update)	
cc-transfer	Call Transfer Capabilities in SIP	Limited support
draft-turn-07	TURN spec for firewall traversal in SIP	
draft-rfc3489bis-15	STUN spec for firewall traversal in SIP	