Polycom

Administrator Guide

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Polycom® RealPresence® Collaboration Server 800s, Virtual Edition

6001 America Center Drive
San Jose, CA 95002
USA

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Overview

About the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server, Virtual Edition Administrator's Guide

The *Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server, Virtual 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:

- Customize the Collaboration Server conferencing entities such as conference Profiles, IVR Services, Meeting Rooms, Entry Queues, etc., to your organization's needs (optional). In the CloudAxis solution environment, these entities should be defined in the Polycom® RealPresence® Distributed Media Application™ (DMA®).
- Define Collaboration Server Users.
- Advanced conference Management
- Define video protocols and resolution configuration for CP conferencing
- Optional. Configure Templates, the Address Book and schedule Reservations. 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 to support Polycom third party and partner environments such as Microsoft, IBM, Cisco, Avaya, Broadsoft and Siemens.
- 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 800s / Virtual Edition Getting Started Guide* provides description of basic conferencing operations. It will help you perform the following tasks:

- Unpack the Collaboration Server system and install it on a rack (Collaboration Server 800s only).
- Connect the required cables to the Collaboration Server (Collaboration Server 800s only).
- 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 (RMX) Web Client, and basic operation of your Collaboration Server system, see the *Polycom® RealPresence Collaboration Server 800s / 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.

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 Server 800s and Polycom® RealPresence® Collaboration Server, Virtual Edition Administrator's Guide System

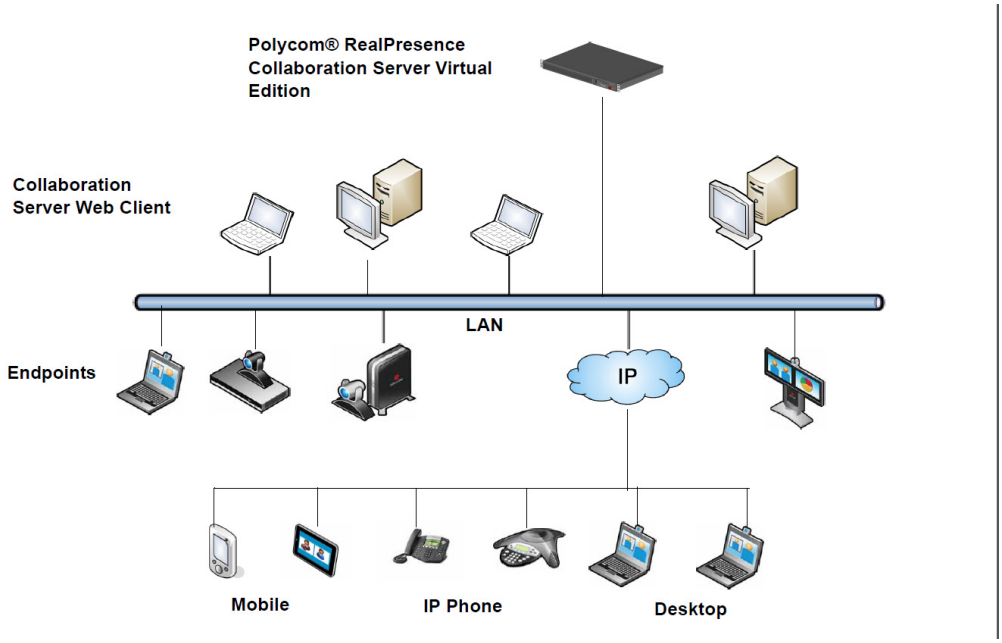
The Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition are high performance, scalable, IP-network (H.323 and SIP) MCUs that provide feature-rich and easy-to-use multipoint voice and video conferencing.

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® RealPresence® Resource Manager - centrally manages, monitors and delivers Cloud based Video as a Service (VaaS) and enterprise video collaboration.
- Polycom® RealPresence® Access Director™ - 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.

Multipoint Video Conferencing using a RealPresence Collaboration Server 800s/Virtual Edition



The RealPresence Collaboration Server 800s / Virtual Edition provide multipoint voice and video conferencing.

The RealPresence Collaboration Server 800s / 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 MCU units..

In the RealPresence Collaboration Server 800s unit, MCU management and IP conferencing are performed via two different LAN ports. The networks can be separated in Maximum Security Environments.

Network Services Guidelines

IP Networks

In the RealPresence Collaboration Server 800s, system management and IP conferencing are performed via a single LAN port.

Management uses LAN1 and IP network Services use LAN2. When enabling multiple services, management and the IP network service (1) share LAN1, the second IP network service (2) uses LAN2.

Workstation Requirements

The RMX 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%

The following table lists the environments (Web Browsers and Operating Systems) with which the Collaboration Server Web Client and RMX Manager applications are supported.

Collaboration Server Wen Client/RMX Manager Environment Interoperability Table

Web Browser	Operating System
Internet Explorer 7	Windows Vista™
	Windows 7
Internet Explorer 8	Windows 7
Internet Explorer 9	Windows 7 and Windows 8
Internet Explorer 10*	Windows 8

.Net Framework 2.0 is required and installed automatically.
If ActiveX installation is blocked, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Administrator's Guide*.

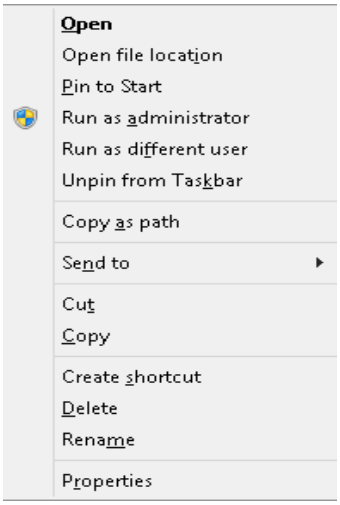
.Net Framework 2.0 SP1 or above is required and installed automatically. Internet Explorer must be enabled to allow running Signed ActiveX.
If ActiveX installation is blocked, see the ActiveX Bypass.

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 RMX 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.**



.Net Framework 2.0 SP1 or above is required and installed automatically. Internet Explorer must be enabled to allow running Signed ActiveX.
If ActiveX installation is blocked, see the ActiveX Bypass.

For Windows 7™ Security Settings, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, *Windows 7™ Security Settings*.

For Internet Explorer 8 configuration, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, *Windows 7™ Security Settings*.

Conferencing Modes Overview

The MCU system offers the following types of conferences (Conferencing Modes), based on the video protocol and the video display during the conference:

- AVC-based Conferencing - *CP Only* (Video Transcoding)
- SVC-based Conferencing (Media Relay) - *SVC Only*
- Mixed AVC and SVC Conferencing - *CP and SVC*

AVC Conferencing

AVC-based Conferences allow endpoints that support AVC video to connect to these conferences. *AVC (Advanced Video Coding)* video refers to the H.264 video protocols used to send and receive video. On the Collaboration Server system it also includes all the standard video protocols such as H.261, H.263, and RTV.

All endpoints (including SVC-enabled 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.

Based on the video processing required during the conference, the Collaboration Server offers the Continuous Presence Conferencing Mode for *AVC-based conferencing.*

The Conferencing Mode determines the video display options (full screen or split screen with all participants viewed simultaneously) and the method in which the video is processed by the MCU (with or without using the MCU's video resources).

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. It 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.

AVC-based 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 Profile and H.264 High Profile) and at any resolution and frame rate, provided they meet the minimum requirements set for the conference:
  - Video Resolutions: from QCIF, CIF and up to 1080p30
  - Frame rates up to 30fps

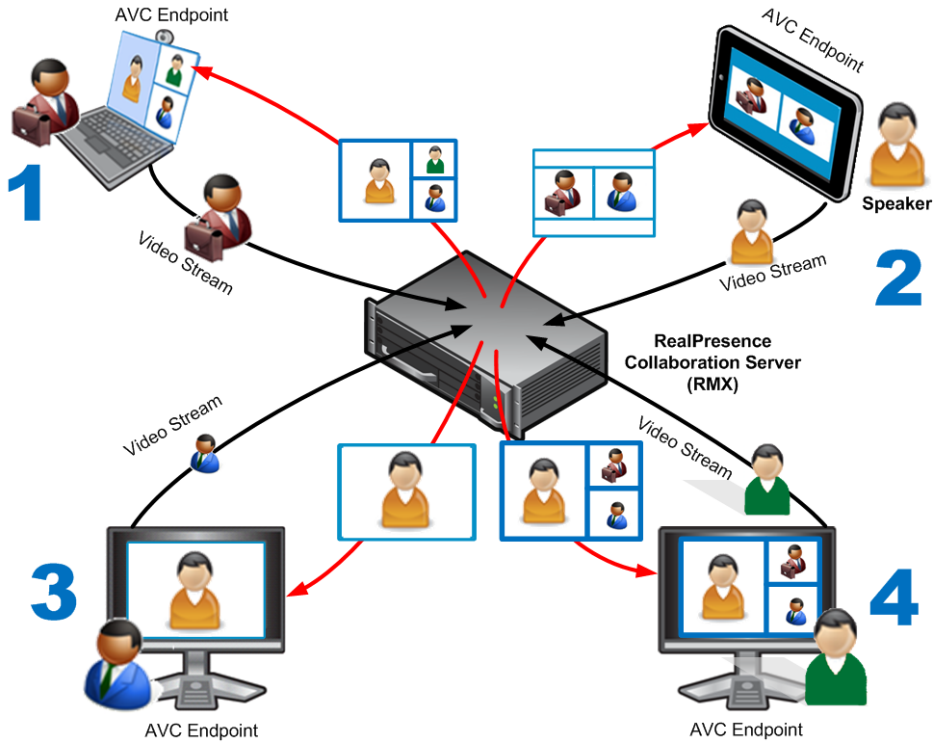
In Continuous Presence conferences, the MCU receives the video stream from each endpoint at the video rate, video resolution and frame rate that it is capable of sending, and it superimposes all the received streams into one video stream that includes the input from the other endpoints arranges in the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

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.

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 **CP 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 Basic Conferencing Parameters

The main parameters that define the quality of an AVC-based video conference and its display are:

- **Line (Bit) Rate -** The transfer rate of video and audio streams. The higher the line (bit) rate, the better the video quality. The MCU supports the following line rates:
  - CP Conferences: 64kbps to 4096kbps
- **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.
- **Video resolution**:

- **H.261 CIF/QCIF** – 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.
- **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 type, compliant endpoints are able to connect to conferences at the following resolutions:
  - **720p** (1280 x 720 pixels) - all Collaboration Server types
  - **1080p** (1920 x 1080 pixels) in Collaboration Server

- **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 Sharing Content During Conferences.
- **Encryption –** Used to enhance media security at conference and participant levels. For more information, see Implementing Media Encryption for Secured Conferencing..
- **Conference Recording -** The Collaboration Server enables audio and video recording of conferences using Polycom RSS recording system.
- **Lecture Mode (CP Conferences only)** – 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).
- **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.

- **Telepresence Mode (CP Conferences only) -** enables the connection of numerous high definition telepresence rooms and of different models (such as TPX and RPX) into one conference maintaining the telepresence experience. This mode is enabled by a special license.
- **TIP Support (CP Conferences only)** - *TIP* is a proprietary protocol created by Cisco for deployment in Cisco TelePresence systems (CTS). Polycom's solution is to allow the Collaboration Server to natively inter-operate with Cisco TelePresence Systems, ensuring optimum quality multi-screen, multipoint calls.

SVC-based Conferencing

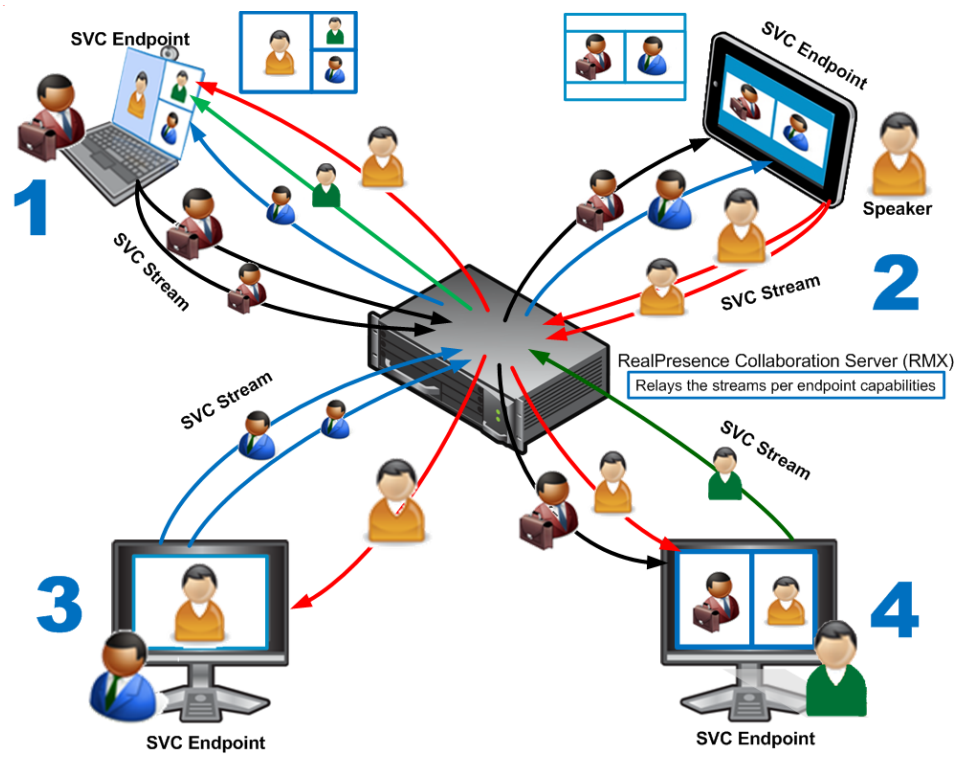
The SVC-Based conferencing mode provides video without transcoding by the MCU, hence requiring less video resources while providing better error resiliency and lower latency.

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 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 the Polycom® RealPresence® Desktop for Windows® 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.

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 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 Collaboration Server functions as the multipoint media relay to the endpoints. For voice activated selection of the video streams, the 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 Collaboration Server functions as a media relay server providing low cost production benefits. The Collaboration Server reduces bandwidth usage by only selecting the necessary video stream to be sent to the endpoints.

SVC Conferencing Guidelines

You can run SVC-based conferences when following the guidelines listed below.

- 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)
  - Ad Hoc conferencing via Meeting Rooms and ongoing conferences
- 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 CP 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 a RealPresence DMA system is part of the solution, the DMA is used as the SIP proxy and the SVC endpoint subscribes to the RealPresence DMA system for call control. If a RealPresence DMA system 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.
  - IVR functionality
  - Conference Gathering phase
  - All DTMF enabled features during the conference
  - Manual selection of video layout
  - Chairperson functionality
  - Media Encryption
  - 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. The table below 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.

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

SVC Conferencing - Maximum Supported Resolutions per Simulcast Stream

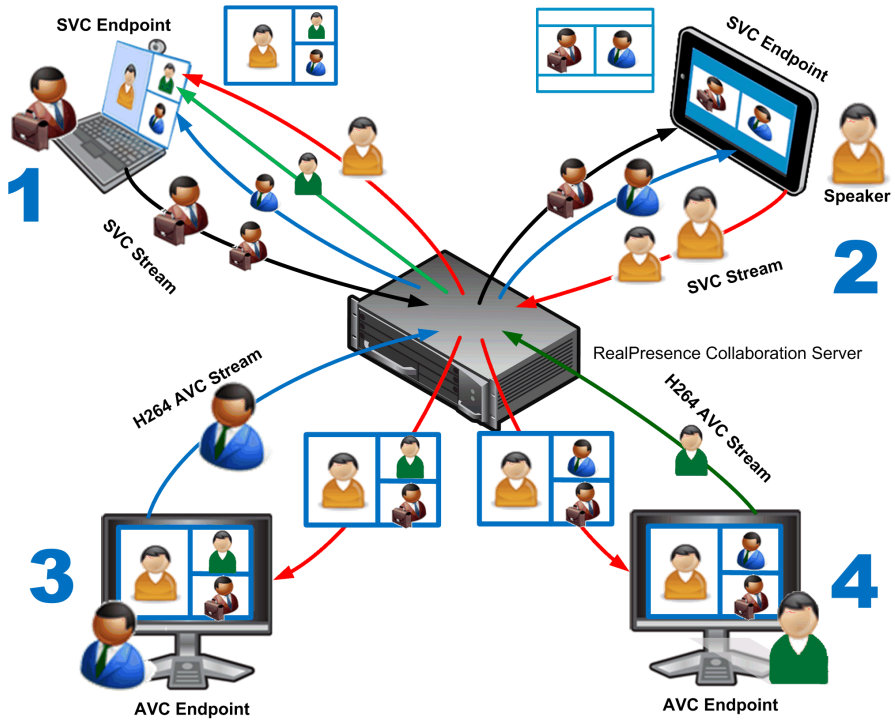
Conference Line Rate (kbps)	Profile	Maximum Resolution	Max. Frame Rate (fps)	Audio Rate (kbps)
192 - 256	Base Profile	180p	30fps	48
128 - 192	Base Profile	180p	15fps	48

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 Collaboration Server like the SVC-based conferences, while AVC endpoints, for example, send only one 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-encoded video bit streams to the SVC-enabled endpoints in the conference according to their request. This enables the video conference layouts to be automatically assembled by the endpoint. AVC endpoints connected to the conference send a single 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 decoded 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 decodes the SVC bit streams and AVC bit streams, then 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.

MCU 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:

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

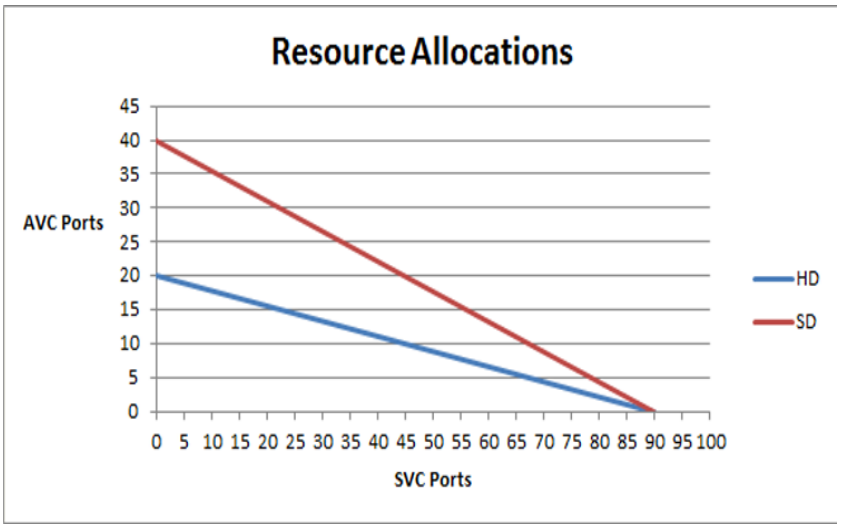
The first four resource types in the resource capacity allocations table are endpoints in a CP only conference or a mixed CP and SVC conference before the actual resource allocations occur.

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:



Using Conference Profiles

In the Polycom® RealPresence® CloudAXIS™ Suite, the Conference Profiles are defined in the RealPresence DMA system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

Conference Profiles include conference parameters such as Conferencing Mode, conference line rate, video and content sharing resolutions and settings, video layout, encryption, Lost Packet Recovery (LPR), etc. Profiles eliminate the need to define these parameters for each new conference created on the MCU. They are stored on the Collaboration Server and they enable you to define all types of conferences.

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

Conference Profiles are assigned to Conferences, Meeting Rooms, Reservations 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*. Profiles can be defined for AVC (Advanced Video Codec) CP conferencing Mode, SVC (Scalable Video Codec) conferencing Mode or Mixed CP and SVC Conferencing Mode.

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.

Conferencing Parameters Defined in a Profile

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, according to the selected Conferencing Mode. When defining a new conference Profile, the system uses default values for the selected conferencing Mode.

Conferencing Capabilities in the Various Conferencing Modes

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

Conferencing Capabilities in the Different Conferencing Modes

Feature	CP Only	Mixed CP & SVC	SVC Only
Conference Type			
Operator Conferences	✓	✗	✗
Entry Queues	✓*	✓*	✓*
Permanent Conference	✓	✓	✓
Cascading	✓**	✓**	✗
Conferencing Feature			
IVR	✓	✓	✓ Reduced IVR set for SVC endpoints
Dial Out	✓	✗	✗
Auto Redial	✓	✓	✗
LPR	✓	✓***	✓***
Content	✓ All Content Settings, All Content Protocols	✓‡ Graphics Only, H.264 Cascade & SVC Optimized (only)	✓ 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	✓	✓	✓

Conferencing Capabilities in the Different Conferencing Modes

Feature	CP Only	Mixed CP & SVC	SVC Only
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.
‡. Content Line Rate is fixed at 128Kbps.

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:

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 for CP Conferences
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

Default CP Only Conference Profile Settings

Setting	Value
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

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:

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
Enable FECC	Disabled
Video Quality	Sharpness
Maximum Resolution	Auto

Default SVC Only Conference Profile Settings

Setting	Value
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

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:

Default Mixed CP and SVC Conference Profile Settings

Setting	Value
Profile Name	Factory_Mix_SVC_CP_Video_Profile
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

Default Mixed CP and SVC Conference Profile Settings

Setting	Value
Enable FECC	Enabled
Video Quality	Sharpness
Maximum Resolution	Auto
Content Settings	Graphics
Content Protocol	H.264 Cascade and SVC Optimized (only)
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

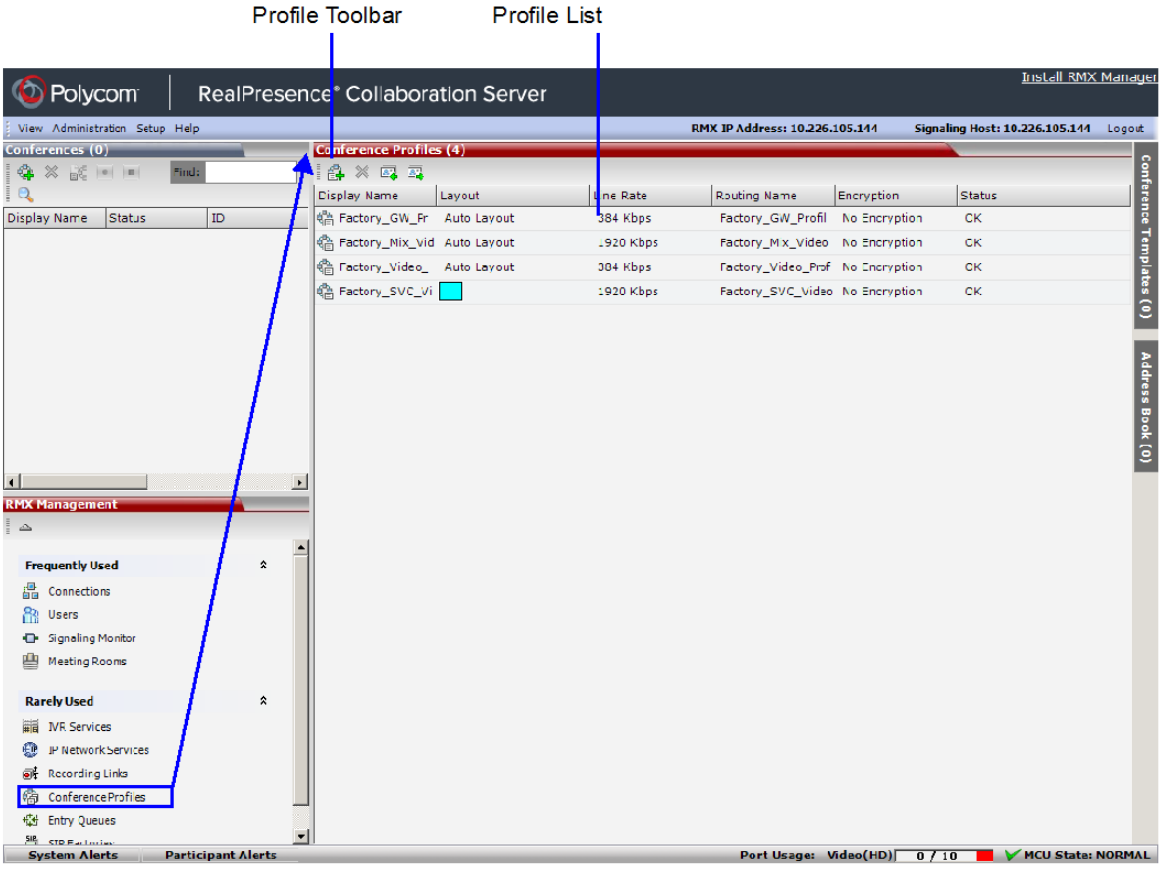
Viewing the List of Conference Profiles

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

To list Conference Profiles:

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

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



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:

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 .

Conference Profiles Pane Columns

Field	Description
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.

Profiles Toolbar

The Profile toolbar provides quick access to the Profile functions:

Profile Toolbar buttons

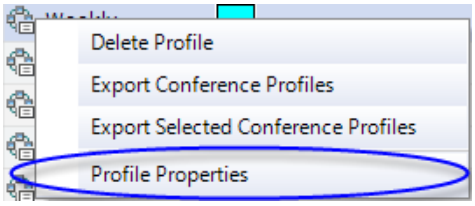
Button	Button Name	Description
	New Profile	To create a new Profile.
	Delete Profile	To delete a Profile, click the Profile name and then click this button.
	Import Profile	To import Conference Profiles from another MCU in your environment.
	Export Profile	To export Conference Profiles to a single XML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



The Profile **Properties - General** dialog box opens.

2 Modify the required Profile parameter(s).

3 Click **OK**.

Deleting a Conference Profile

You can delete Profiles from the Profiles list.

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

To delete a Conference Profile:

1. List the Profile that are currently defined in the system. For details, see Viewing the List of Conference Profiles.
2. In the **Conference Profiles** list, select the Conference Profile you want to delete.
3. Click the Delete Profile ( ) button.
   or
   Right-click the Conference Profile to be deleted and select **Delete Profile** from the menu.
4. In the confirmation dialog box, click **OK**.

   The Conference Profile is deleted.

Defining New Profiles

In the Polycom® RealPresence® CloudAXIS™ Suite, the Conference Profiles are defined in the Polycom® RealPresence® DMA® system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

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

Profiles can be defined for the following **Conferencing Modes**: AVC (Advanced Video Codec) CP , SVC (Scalable Video Codec) or Mixed CP and SVC. The Profile tabs and options change according to the selected Conferencing Mode and only supported options are available for selection. Unsupported options are disabled (grayed out).

CP Conferencing Mode also offers a special functional conference - Operator Conference.

To facilitate the definition process of a new Profile, the system displays default values for each parameter so you need only to modify the required settings.

To define a new Profile:

1. In the **RMX Management** pane, expand the **Rarely Used** list.
2. In the **RMX Management** pane, click **Conference Profiles**.
3. In the **Conference Profiles** pane, click the **New Profile** button.

   The **New Profile – General** dialog box opens.
4. In the **Display Name** field, enter the Profile name.

5 Select the appropriate **Conferencing Mode**: **CP**, **SVC Only** or **CP and SVC.**

The New Profile tabs and options change according to the selected Conferencing Mode and only supported options are available for selection.

6 Define the Profile parameters as described in:

- Defining AVC CP Conferencing Profiles
- Defining SVC Conference Profiles
- Defining Mixed CP and SVC Conferencing Profiles

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 for Exporting and Importing Conference Profiles

- Only Collaboration Server system 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 Exporting and Importing Conference Templates.

Exporting Conference Profiles

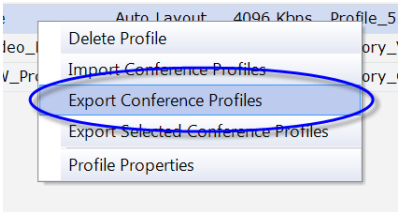
Conference Profiles are exported to a single XML file that can be used to import the Conference Profiles on multiple MCUs. Using the Export Conference Profile feature, you can export all or selected Conference Profiles from an MCU.

Exporting All Conference Profiles from an MCU

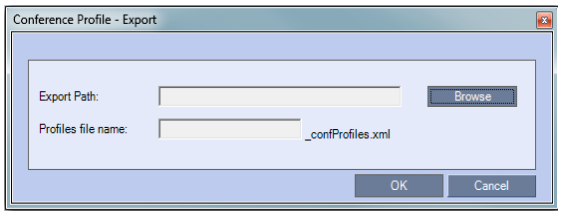
To export all Conference Profiles from an MCU:

1 List the Profile that are currently defined in the system. For details, see Viewing the List of Conference Profiles.

2 In the Conference Profiles List toolbar, click the **Export Conference Profiles**  button or right-click anywhere in the Conference Profiles pane, and then click **Export Conference Profiles**.



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

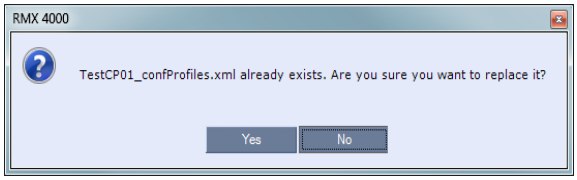


3 In the **Export Path** field, click **Browse** to navigate to the location of the desired path where you want to save the exported file.

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.



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.

Exporting Selected Conference Profiles

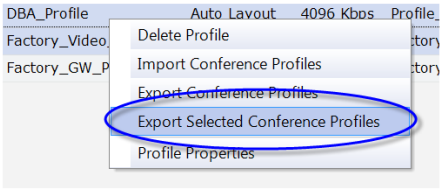
You can select a single Conference Profile or multiple Conference Profiles and export them to a file to be imported to other MCUs in your environment.

To export selected Conference Profiles:

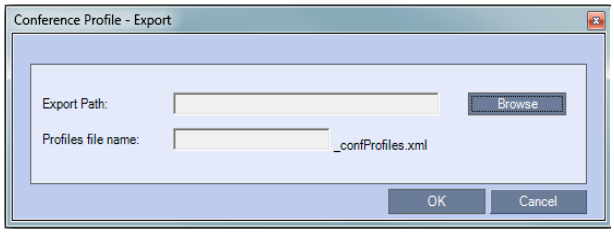
1 List the Profile that are currently defined in the system. For details, see Viewing the List of Conference Profiles.

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

3 In the Conference Profiles List toolbar, click the **Export Conference Profiles** button or right-click the selected Conference Profiles, and then click **Export Selected Conference Profiles**.



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

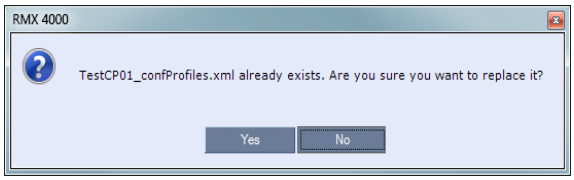


4 In the **Export Path** field, click **Browse** to navigate to the location of the desired path where you want to save the exported file.

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.



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.

Importing Conference Profiles

If your environment includes two or more MCUs, import previously exported Conference Profiles to your MCU to save configuration time and ensure that all MCUs use the same conferencing parameters.

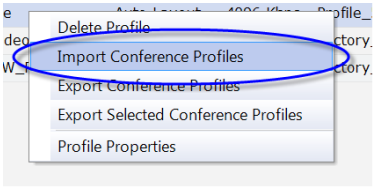
Conference Profiles are not imported when a Conference Profile with that name already exists or if an IVR Service which is assigned to any of the imported Profile does not exist in the MCU.

Conference Profiles are not imported when a Conference Profile with that name already exists or if an IVR Service which is assigned to any of the imported Profile does not exist in the MCU.

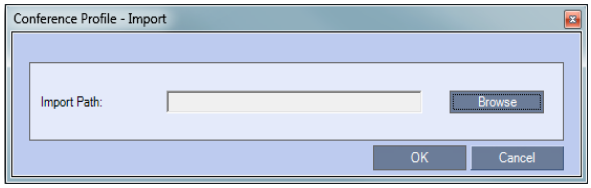
To import Conference Profiles:

1 Display the Conference Profiles List. For details, see Viewing the List of Conference Profiles.

2 In the Conference Profiles List toolbar, click the **Import Conference Profiles** button or right-click the Conference Profiles pane, and then click **Import Conference Profiles**.



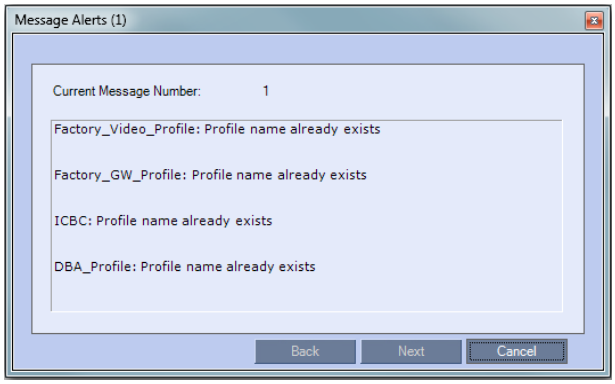
The **Conference Profile - Import** dialog box is displayed.



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

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

When Conference Profiles cannot be imported, a **Message Alert** window is displayed with the profiles that were not imported.



Conference Profiles that are not problematic are imported.

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

The imported Conference Profiles appear in the Conference Profiles list.

Defining AVC-Based Conference Profiles

In the RealPresence CloudAxis Solution, the Conference Profiles are defined in the Polycom® RealPresence® DMA® component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

AVC-based Conference Profile options differ according to the selected Conferencing Mode. To facilitate the definition process of a new Profile, the system displays default values for each parameter so you need only to modify the required settings.

Defining AVC CP Conferencing Profiles

When defining a new Profile, you select the parameters that determine the video display on the participant's endpoint, the quality of the video, content sharing parameters, whether the conference will be recorded, encryption, Telepresence mode and other conferencing parameters.

The following parameters are defined:

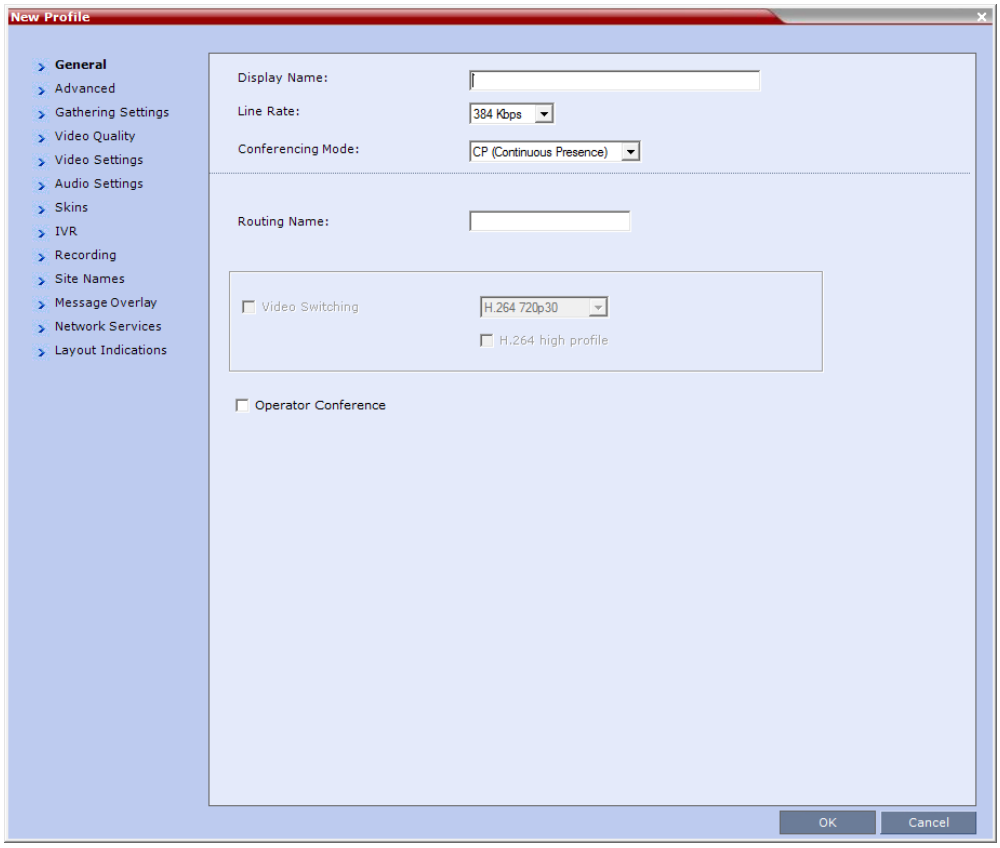
- New AVC CP Profile - General Parameters
- New AVC CP Profile - Advanced Parameters
- New AVC CP Profile - Gathering Settings Parameters
- New AVC CP Profile - Video Quality Parameters
- New AVC CP Profile - Video Settings Parameters
- New AVC CP Profile - Audio Settings Parameters
- New AVC CP Profile - IVR Parameters
- New AVC CP Profile - Recording Parameters
- New AVC CP Profile - Site Names Parameters
- New AVC CP Profile - Message Overlay Parameters
- New AVC CP Profile - Network Services Parameters
- New AVC CP Profile - Layout Indications Parameters

To define a new CP Profile:

1 In the **RMX Management** pane, click **Conference Profiles**.

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

The **New Profile – General** dialog box opens.



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

New AVC CP Profile - General Parameters

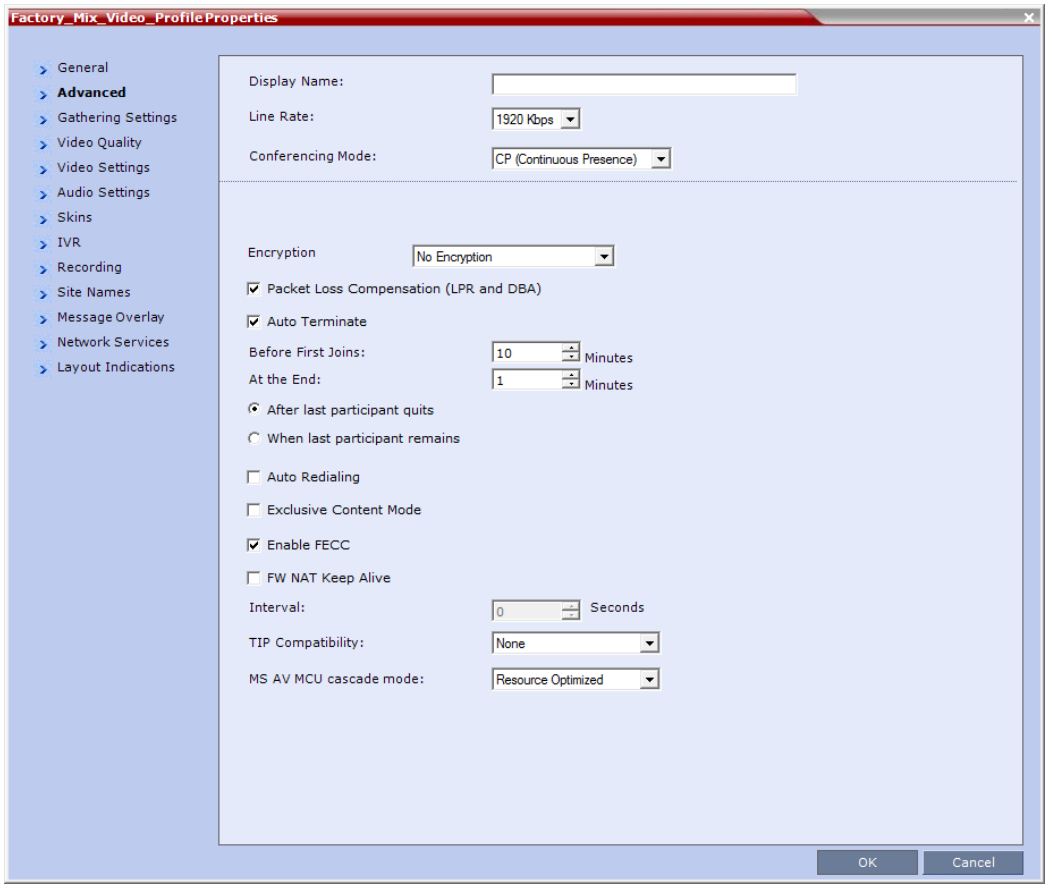
Field/Option	Description
Display Name	Enter a unique Profile name, as follows: • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately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

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. **Note:** This field is displayed in all tabs.
Conferencing Mode	Select the required Conferencing Mode. The selection affects the available tabs and their fields. 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 **(CP Only)**	Select this option to define the profile of an Operator conference. When defining an *Operator Conference*, the **Send Content to Legacy Endpoints** option in the *Video Quality* tab is cleared and disabled.

4 Click the **Advanced** tab.

The **New Profile – Advanced** dialog box opens.



5 Define the following parameters:

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.
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.

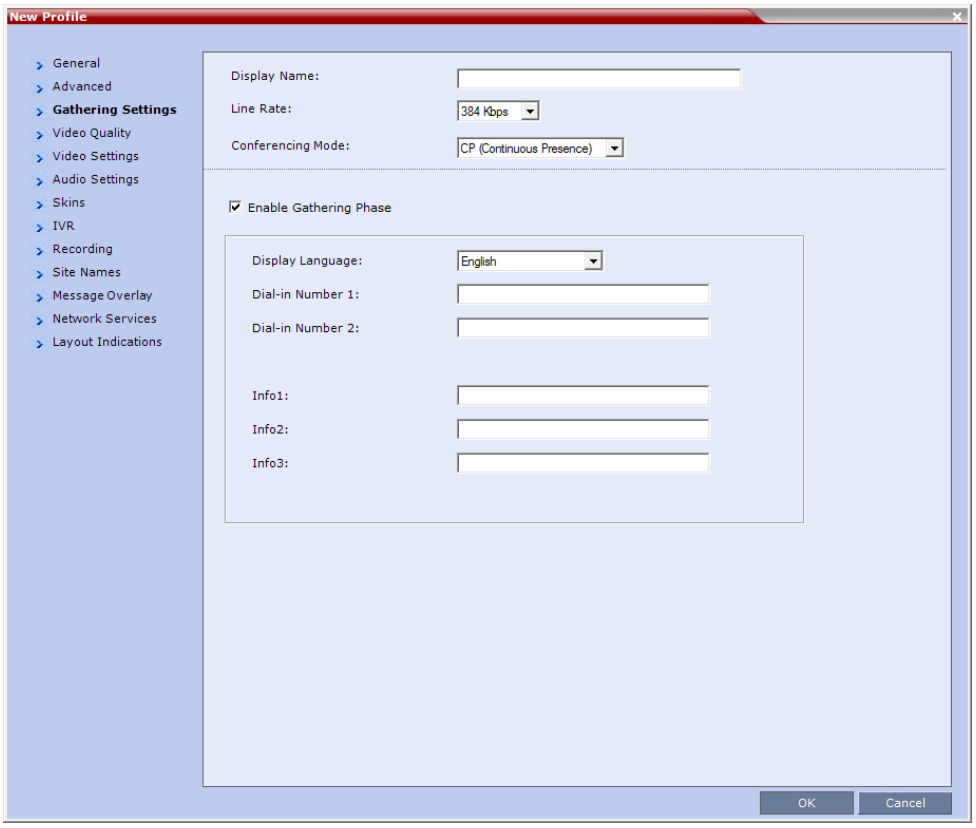
New AVC CP Profile - Advanced Parameters (Continued)

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). Default idle time is 1 minute. **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 this option to limit the Content broadcasting to one participant, preventing other participants from interrupting the Content broadcasting while it is active.
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.
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.

New AVC CP Profile - Advanced Parameters (Continued)

Field/Option	Description
MS AV MCU Cascade Mode	This enables you to set Cascade Mode as either Resource Optimized or Video Optimized. • **Resource Optimized** System resource usage is optimized by allowing high resolution connections only at high line rates and may result in lower video resolutions for some line rates. This option allows youto 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. For more information, see H.264 Base Profile and High Profile Comparison.

6 **For CP Conferences only:** Click the **Gathering Settings** tab.



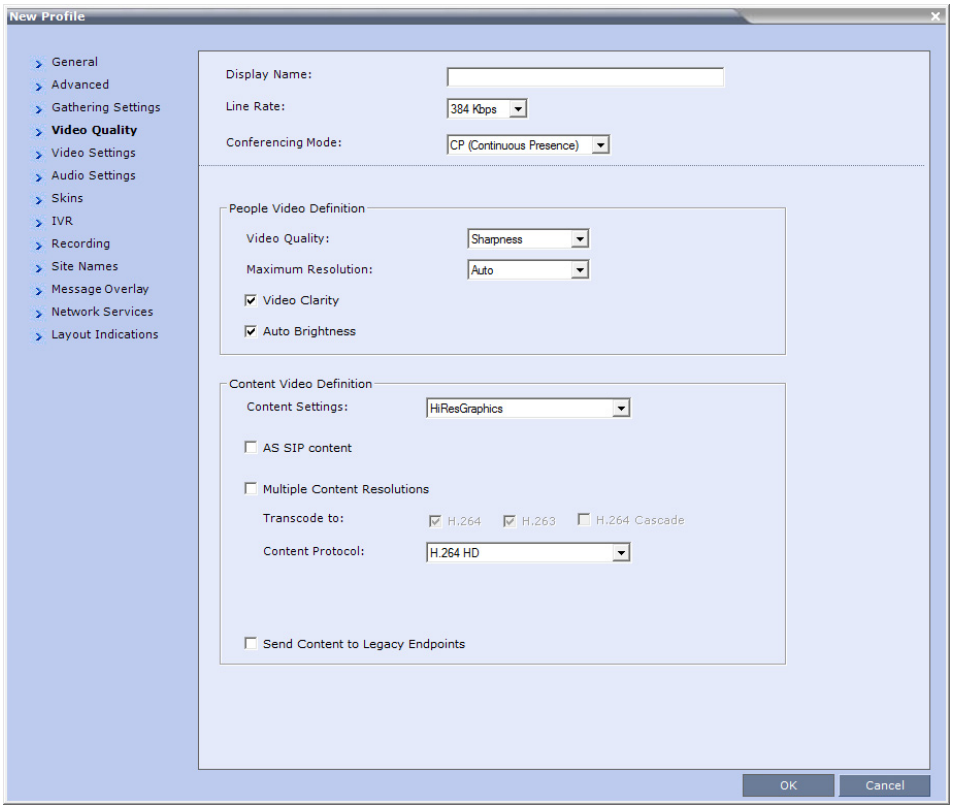
7 **Optional.** Define the following fields if the conference is not launched by the *Polycom Conferencing Add-in for Microsoft Outlook*:

- If the conference is launched by the *Polycom Conferencing Add-in for Microsoft Outlook* the field information is received from the meeting invitation and existing field value are overridden. For more information see Polycom Conferencing for Microsoft Outlook® .
- Gathering is not supported in Cascading Conferences.

For more information see Auto Scan and Customized Polling in Video Layout (CP Conferences Only).

8 Click the **Video Quality** tab.

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



9 Define the following parameters:

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. Depending on the amount of movement contained in the conference video, select either: • **Motion** – For a higher frame rate without increased resolution. When selected, *Video Clarity* is disabled. • **Sharpness** – For higher video resolution and requires more system resources. **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.
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** • **HD1080** 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.
AS SIP Content	AS-SIP is an implementation of SIP that utilizes SIP's built in security features. When selected, content is shared using the Multiple Resolutions mode and is not supported in any other Content sharing mode. For more information, see Enabling AS-SIP Content.

New AVC CP Profile - Video Quality Parameters (Continued)

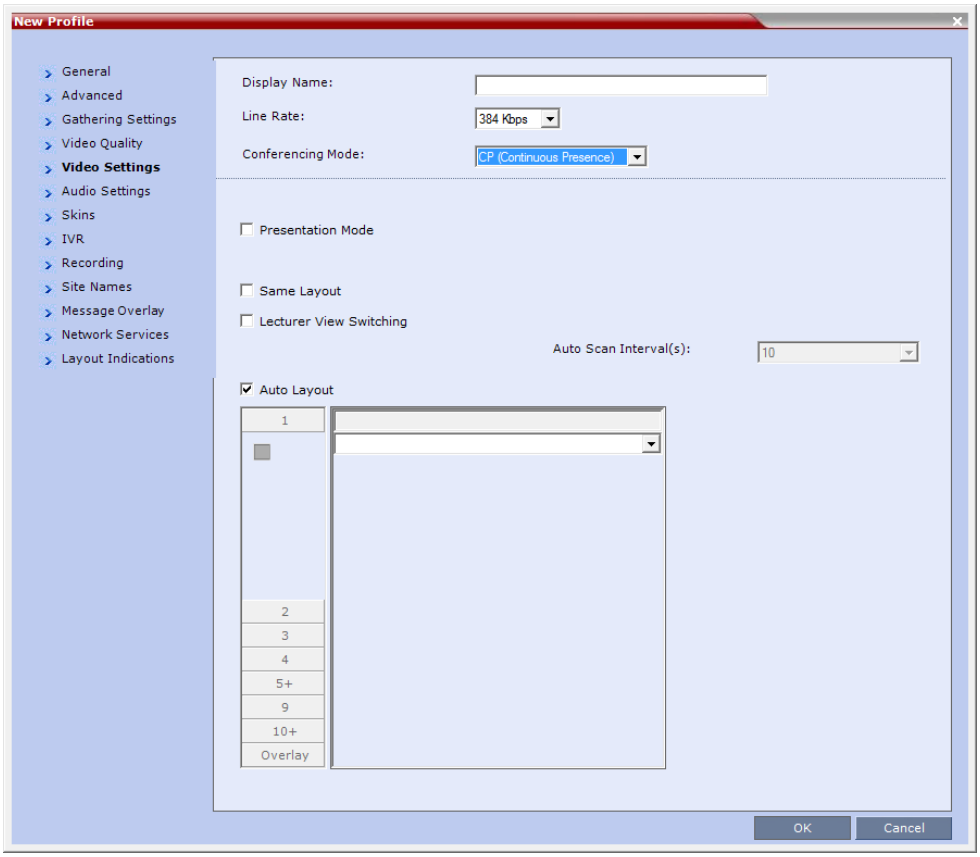
Field/Option	Description
Multiple Content Resolutions	Click this check box to enable the Multiple Content Resolutions mode, in which content is shared in multiple streams, one for each video protocol: H.263 and H.264. This allows endpoints with different protocols to connect and disconnect without having to restart Content sharing in the middle of a conference. For more information, see Sharing Content Using Multiple Content Resolutions Mode. When enabled, the H.264 is always selected and can not be deselected. **Note:** If **Multiple Content Resolutions** is selected in a Cascading environment, the **Content Protocol** must be set to **H.264 Cascade and SVC Optimized** and **H.264 Cascade** must be checked as the **Transcode to:** setting. **Optional.** Select additional protocols: • **H.263** - if the conference will include H.263-capable endpoints that do not support H.264 protocol for content sharing. • **H.264 Cascade** - if the conference will include cascading links that should use a fixed video format for content sharing. **Optional.** If **H.264 Cascade** is selected, select the desired Content Resolution.
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) When selected, content is shared using **H.263** if a mix of H.263-supporting and H.264-supporting endpoints are connected, or **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. **Note:** When **Multiple Content Resolutions** is selected, the Content Protocol field is hidden. For more information, see Content Protocols and Defining Content Sharing Parameters for a Conference.
H.264 High Profile (Check Box)	The H.264 High Profile check box is un-checked by default and is displayed next to the Content Protocol drop-down menu if all the following conditions are met: • The selected Conferencing Mode is AVC-CP. • Multiple Resolutions (Content Transcoding) is not selected. • The selected Content Protocol is Cascade and SVC Optimized. If H.264 HD, H.264 Cascade and SVC Optimized is selected, the Content Resolution is set according to the line rate. • TIP Compatibility (in the Profile - Advanced dialog box) is selected as None or Video Only.

New AVC CP Profile - Video Quality Parameters (Continued)

Field/Option	Description
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.
Content Rate drop-down menu	The Content Rate drop-down menu is displayed next to the Content Resolution drop-down menu when: • H.264 Cascade and SVC Optimized is the selected Content Protocol and • CustomizedContentRate is the selected Content Setting.
Send Content to Legacy Endpoints (CP only)	When enabled (default), Content can be sent to H.323/SIP endpoints that do not support H.239 Content (legacy endpoints) over the video (people) channel. For more information see Sending Content to Legacy Endpoints (AVC Only). **Notes:** • When enabled, an additional HD video resource is allocated to the conference. • This option is valid when sending Content as a separate stream is enabled by the System Flag ENABLE_H239 set to YES. • Select this option when Avaya IP Softphone will be connecting to the conference. • If the **Same Layout** option is selected, the Send Content to Legacy Endpoints selection is cleared and is disabled. • Once an endpoint is categorized as Legacy, it will not be able to restore its content to the Content channel and will receive content only in the video channel.

10 Click the **Video Settings** tab.

The **New Profile - Video Settings** dialog box opens.



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

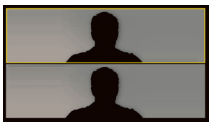
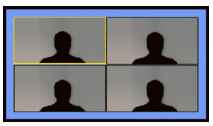
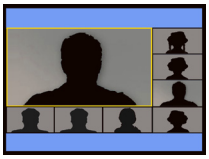
New AVC CP Profile - Video Settings Parameters

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.
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).

New AVC CP Profile - Video Settings Parameters (Continued)

Field/Option	Description
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 Auto Layout – Default Layouts in CP Conferences. 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. **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.

Auto Layout – Default Layouts in CP Conferences

Number of Video Participants	Auto Layout Default Settings
0–2	
3	
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 Modifying System Flags.

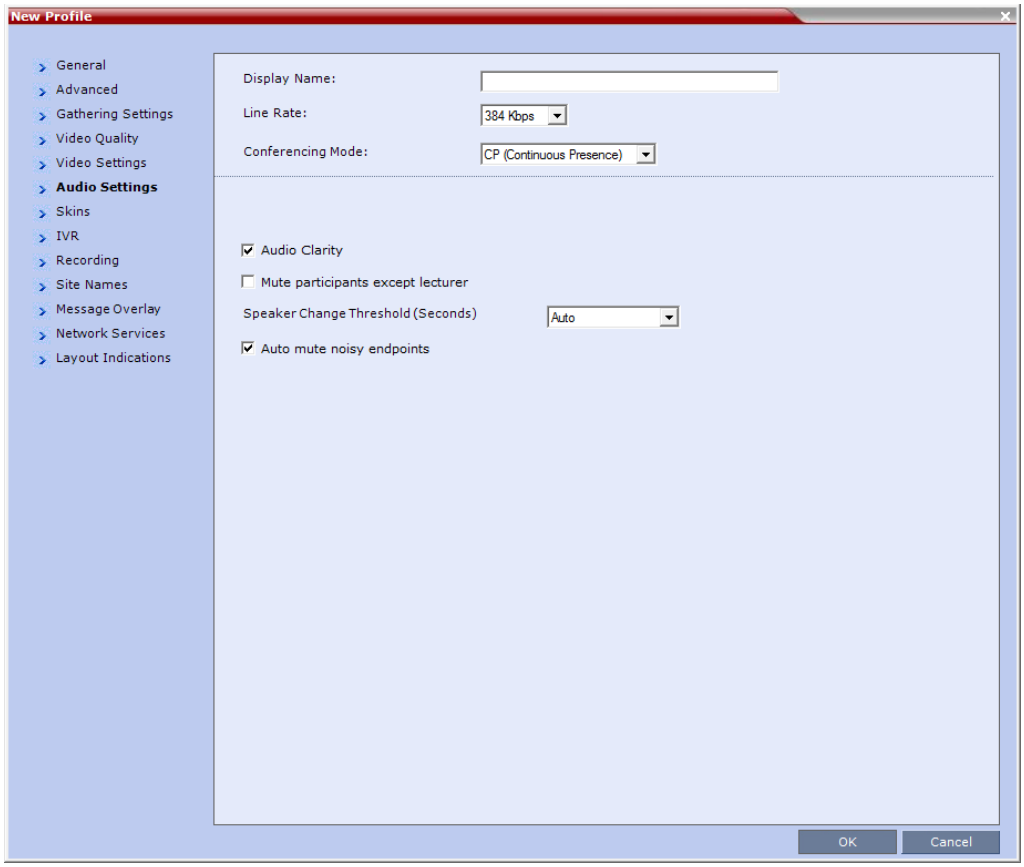
12 To select the **Video Layout** for the conference, click the required number of windows from the layouts bar and then select the windows array. The selected layout is displayed in the Video Layout pane.

Video Layout Options

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

13 Click the **Audio Settings** tab.

The **New Profile - Audio Settings** dialog box opens.



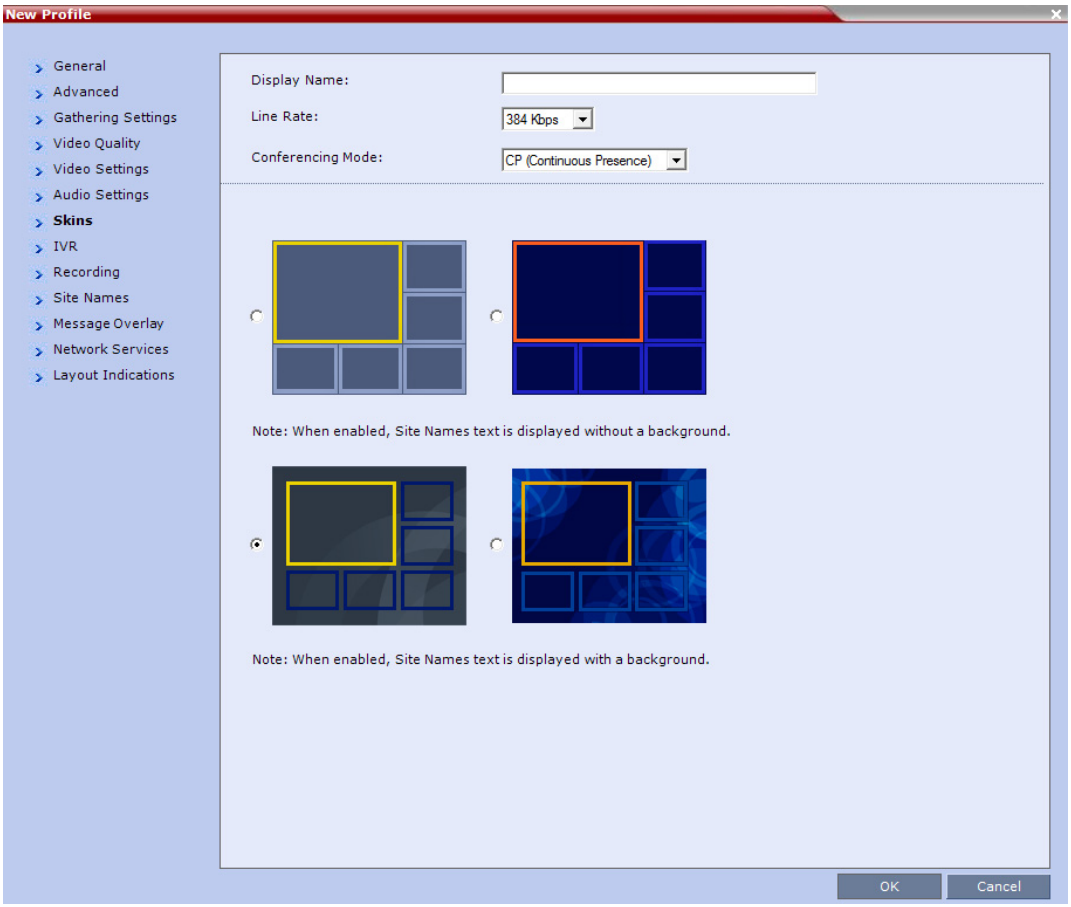
14 Define the following parameters:

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 Message Overlay function. 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. • This 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. • 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 participant) are muted. Only the lecturer is not muted.
Speaker Change Threshold	Indicates the amount of time a participant must speak continuously before becoming the speaker. Select the desired threshold: • **Auto** (Default, 3 seconds) • **1.5** seconds • **3** seconds • **5** seconds

15 Click the **Skins** tab to modify the background and frames.

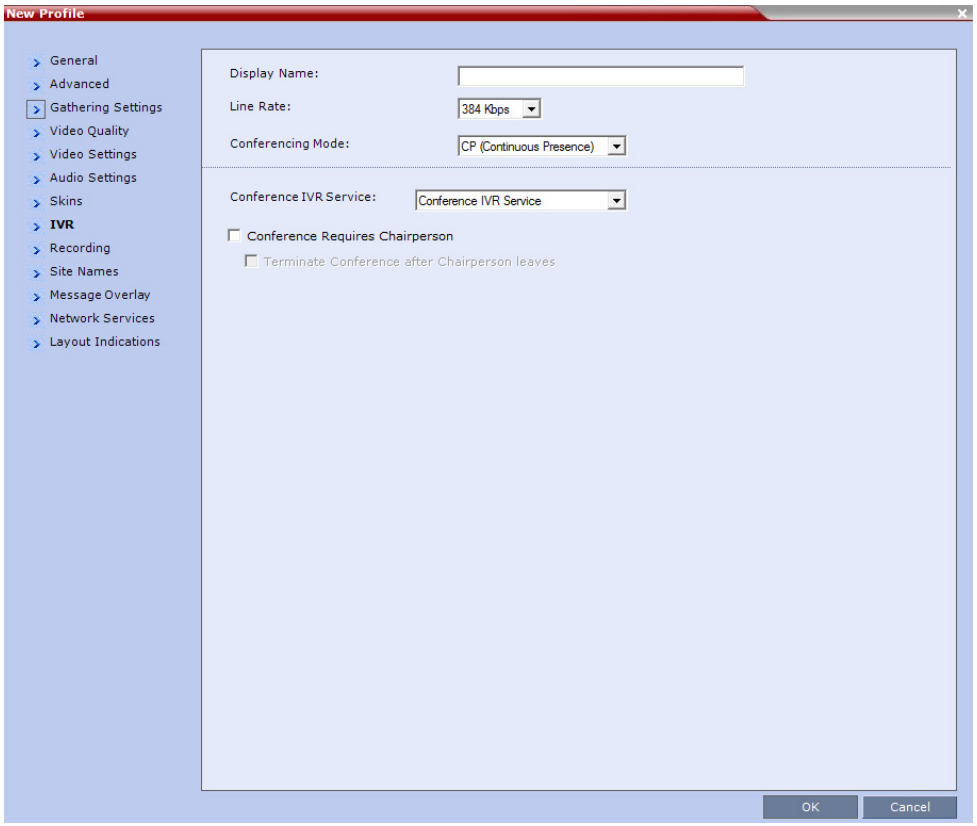
The **New Profile - Skins** dialog box opens.



16 Select one of the Skin options.

17 Click the **IVR** tab.

The **New Profile - IVR** dialog box opens.



18 If required, set the following parameters:

New AVC CP Profile - IVR Parameters

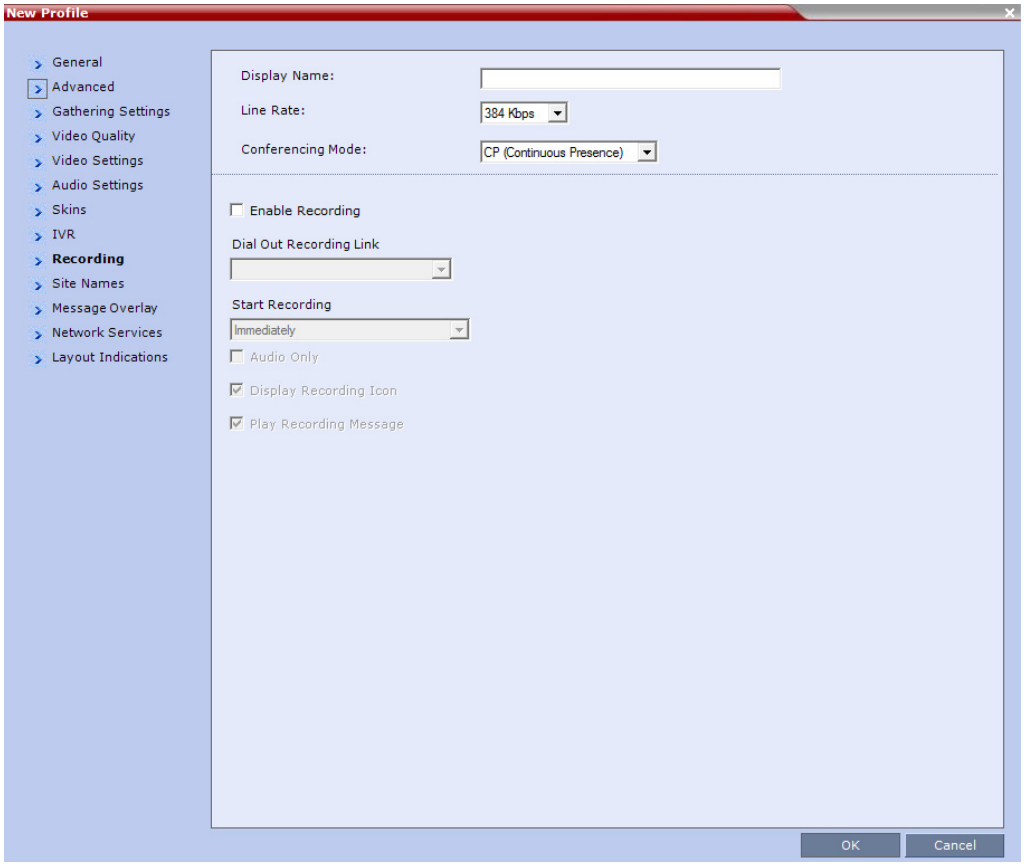
Field/Option	Description
Conference IVR Service	The default conference IVR Service is selected. You can select another conference IVR Service if required.
Conference Requires Chairperson	Select this option to allow the conference to start only when the chairperson connects to the conference and to automatically 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.

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. Terminate Conference After Chairperson Leaves is not supported in cascaded environments.

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

The **New Profile - Recording** dialog box opens.



20 Define the following parameters:

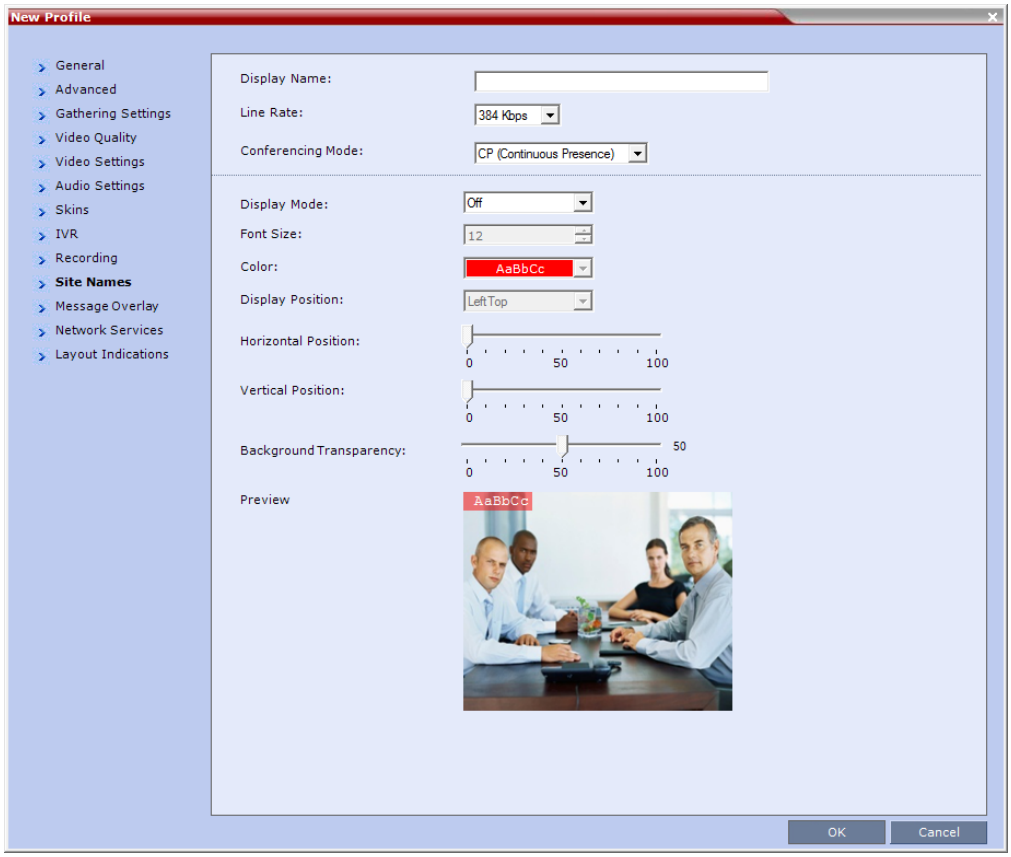
New AVC CP Profile - Recording Parameters

Parameter	Description
Enable Recording	Select this check box to enable the Recording of the conference. 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* (Recording and Streaming Server). For more information see Recording Conferences
Start Recording	Select when to start the recording: • **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. Clear the selection to prevent the display of the recording icon.

The Recording link (which is listed as a participant in the conference) 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 only the conference Audio.

21 Click the **Site Names** tab.

The **New Profile - Site Names** dialog box opens.



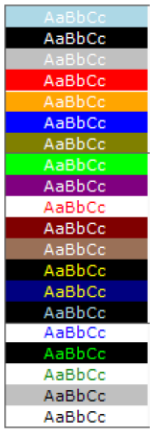
Using the **Site Name** dialog box, you can control the display of the site names by defining the font, size, color, background color and transparency and position within the Video Window. For a detailed description of the site names options see Site Names Definition.

22 Define the following parameters:

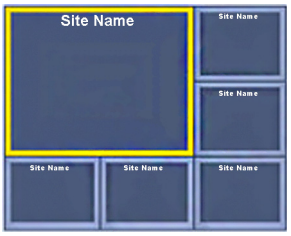
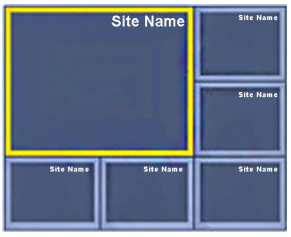
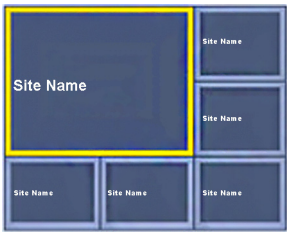
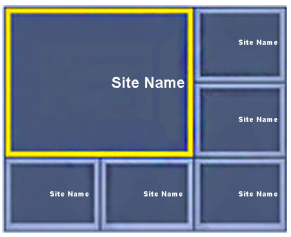
New AVC CP Profile - Site Names Parameters

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 and all other fields in this tab are grayed and disabled

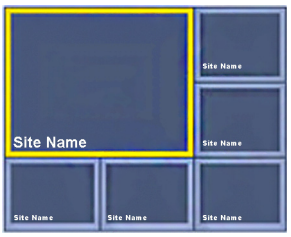
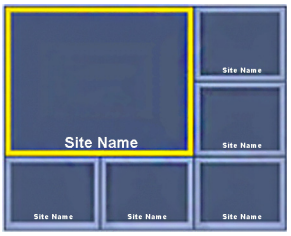
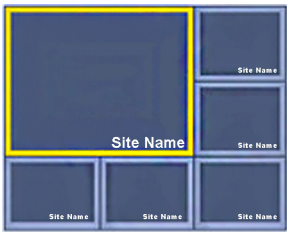
New AVC CP Profile - Site Names Parameters

Field	Description
Font Size	Click the arrows to adjust the font size (in points) for the display of Site Names. 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. Range: 9 - 32 points Default: 12 points
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 (Classic)**  **Default:** White Text No Background (For contrast, no background is shown as black when the text is white.) **Picture Skin**  **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.

New AVC CP Profile - Site Names Parameters

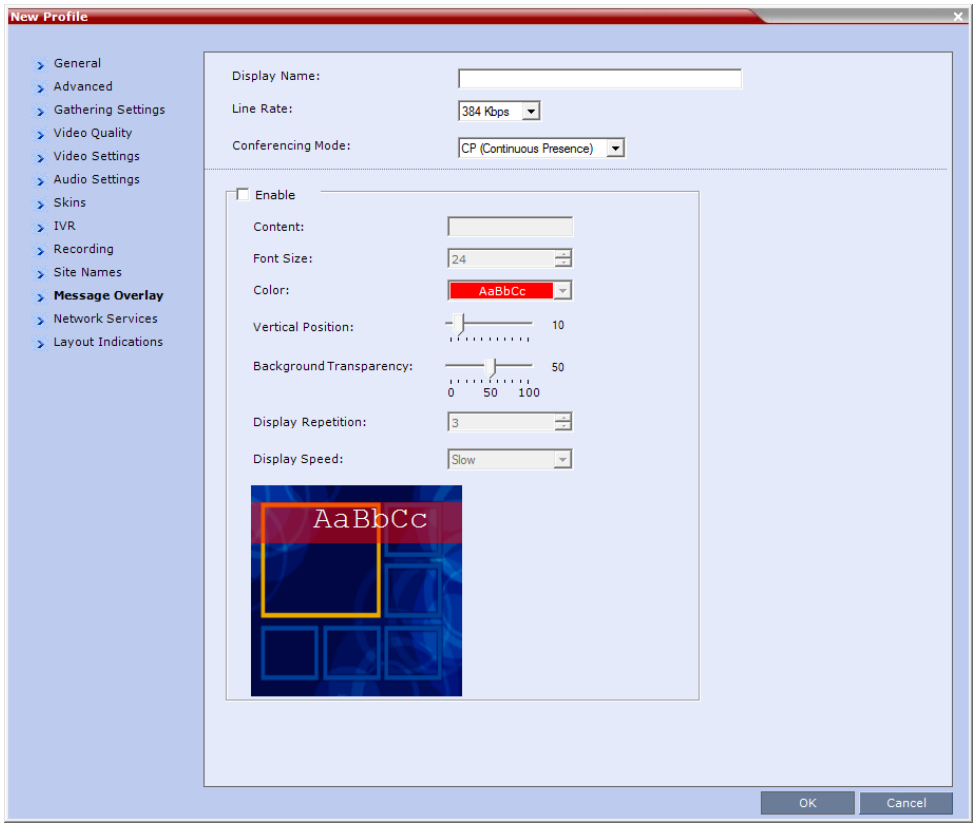
Field	Description
Display Position	Select the pre-set position for the display of the Site (endpoint) Names. **Selection** — **Site Names Position** **LeftTop** (Default)  **Top**  **RightTop**  **LeftMiddle**  **RightMiddle** 

New AVC CP Profile - Site Names Parameters

Field	Description
Display Position (cont.)	**LeftBottom** 
	Bottom 
	RightBottom 

23 Click the **Message Overlay** tab.

The **New Profile - Message Overlay** dialog box opens.



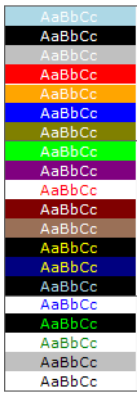
Message Overlay enables you to send text messages to all participants during ongoing Continuous Presence conferences.

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 Sending Text Messages During a Conference Using Message Overlay.

24 Define the following fields:

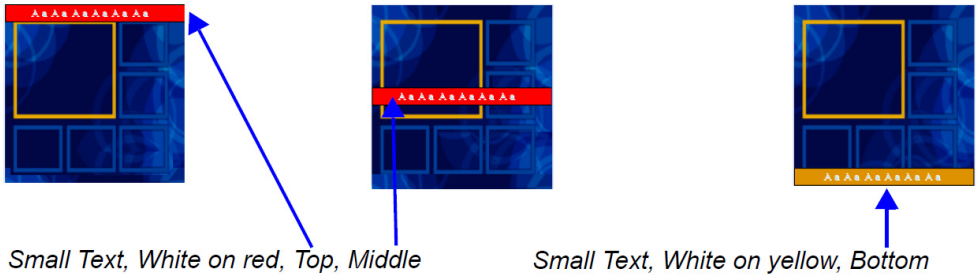
New AVC CP Profile - Message Overlay Parameters

Field	Description
Enable	This option is disabled by default. Select this check box to enable Message Overlay or clear it to disable it.
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 display of the message text. Font size range: 9 - 32 points, default: 24 points **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 displayed text. The choices are:  Default: White text on red background.
Vertical Position	Move the slider to the **right** to move the vertical position of the displayed text downward within the Video Layout. Move the slider to the **left** to move the vertical position of the displayed text 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 text. 0 = No transparency (solid background color). Move the slider to the **right** to increase the transparency of the background of the message 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

New AVC CP Profile - Message Overlay Parameters

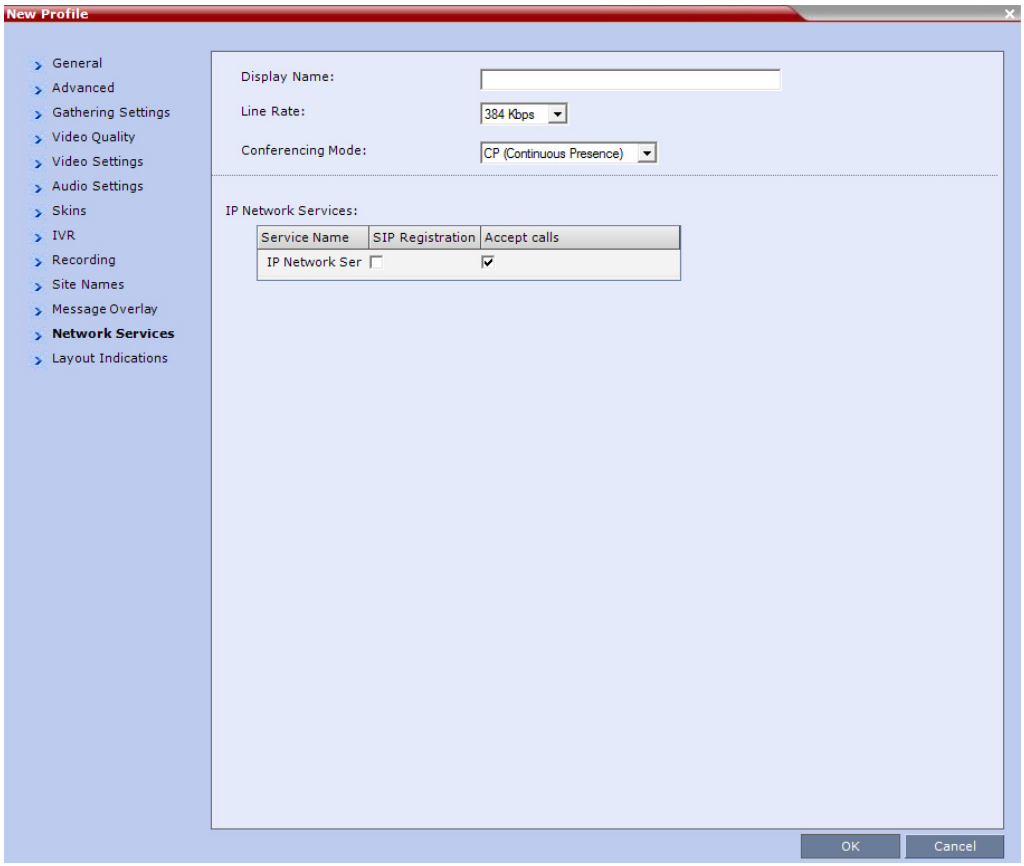
Field	Description
Display Speed	Select whether the text message display is static or moving across the screen, the speed in which the text message moves: **Static, Slow, Fast** Default: Slow

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



25 Click the **Network Services** tab.

The **New Profile - Network Services** dialog box opens.



Registration of conferencing entities such as ongoing conferences, Meeting Rooms, Entry Queuesand 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.

26 Define the following parameters:

New AVC CP Profile - Network Services Parameters

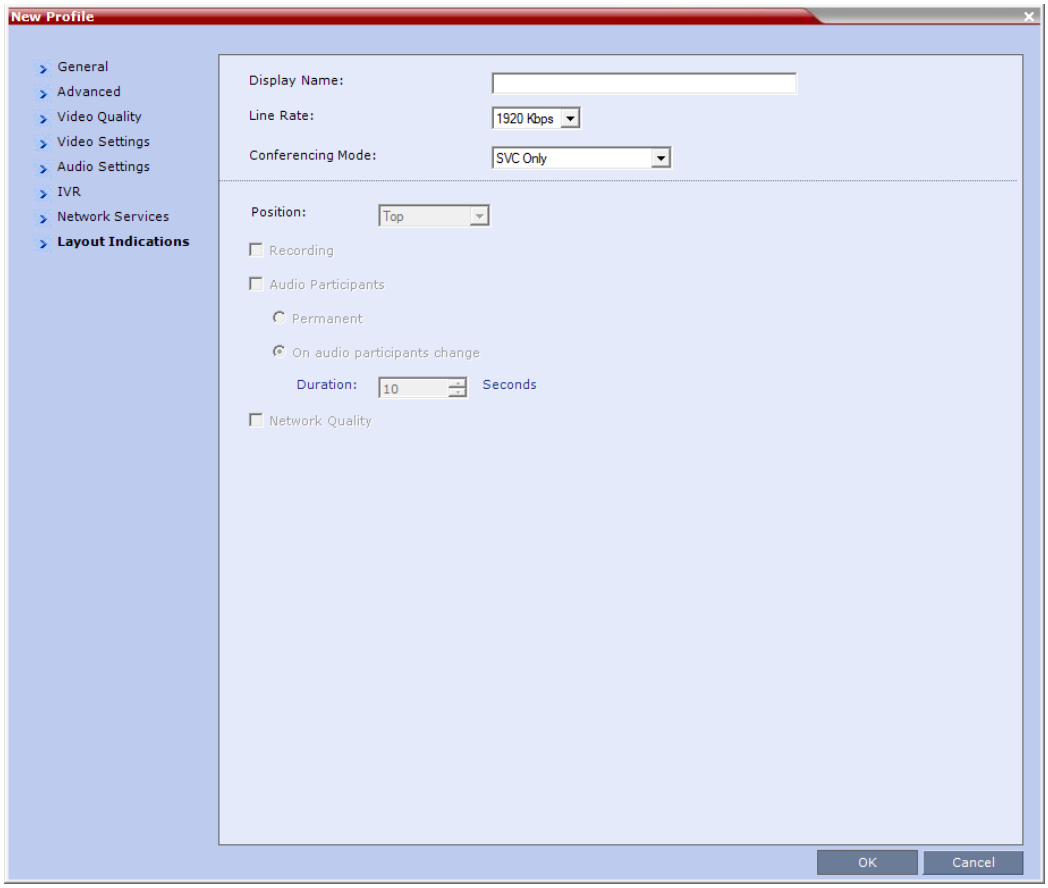
Parameter	Description
IP Network Services	
Service Name	This column lists all the defined Network Services, one or several depending on the system configuration.

New AVC CP Profile - Network Services Parameters

Parameter	Description
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.

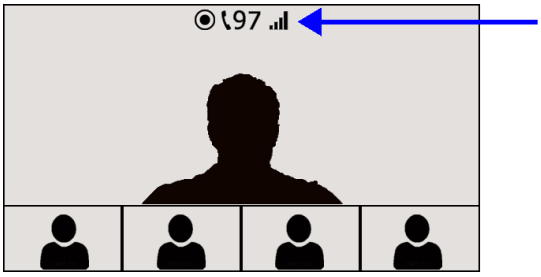
27 Click the **Layout Indications** tab.

The **Layout Indications** dialog is displayed.



The Layout Indications are displayed in the conference video layout as an icon group that includes:

- Recording
- Audio Participants
- Network Quality



The Layout Indications tab is only displayed if either AVC-CP or AVC-CP and SVC (mixed mode) is selected.

Beginning with version 8.4, the configuration options of the Layout Indications tab have replaced the following system flags:

- DISABLE_SELF_NETWORK_IND
- SELF_IND_LOCATION

For further information on layout indications see Layout Indications (AVC Endpoints).

28 Modify the following parameters to configure the display, position, and duration of the indication icons.

New AVC CP Profile - Layout Indications Parameters

Field	Description
Position	Use the Position drop-down menu to configure the display position of the indication icons group. Icons can be displayed in the following positions: • Top-left • Bottom-left • Top center (default) • Bottom • Top-right • Bottom-right
Recording	Select the check box to display the Recording icon. The Recording indication icon is a duplicate of the Display Recording Icon field in the Recording tab of the Profile dialog. For more information, see Recording Conferences.
Audio Participants	Select the check box to display the Audio Participants icon. • **Permanent -** Sets the Audio Participant Indication to display permanently. In this setting it is only displayed when audio participants are connected. • **On audio participants change** - Sets the Audio Participant Indication to display for a short period only when the number of audio participants changes. ◦ **Duration** - Sets the amount of time the Audio Participant icon displays. For more information see Audio Participants Indication.
Network Quality	To display the Network Quality icon, select the check box. For more information see Network Quality Indication (AVC Endpoints).

29 Click **OK** to complete the Profile definition.

A new Profile is created and added to the Conference Profiles list.

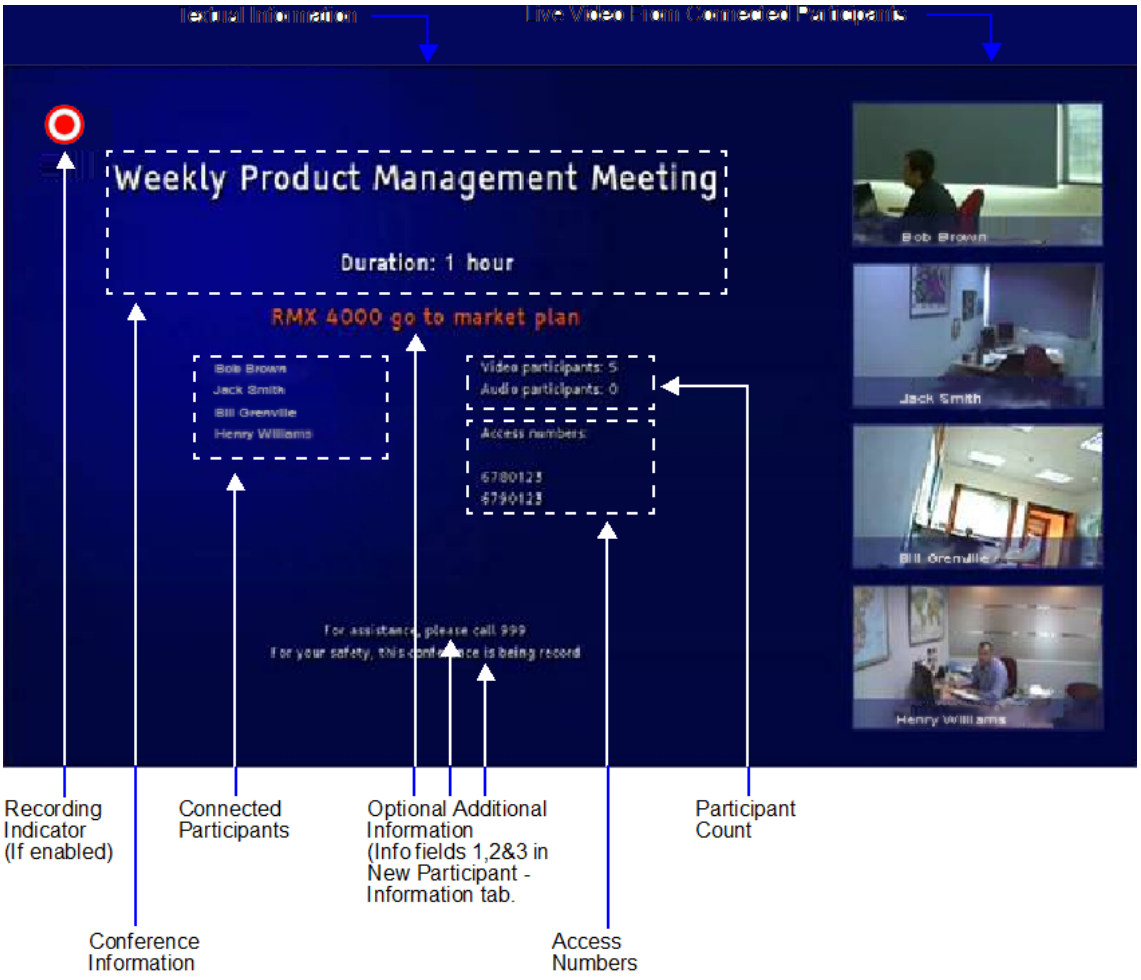
Additional Information for Setting CP Profiles

This section includes detailed explanation of various CP Profile settings:

- Gathering Phase
- Overlay Layouts
- Site Names Definition
- Sending Text Messages During a Conference Using Message Overlay
- Selecting the Chinese Font for Text Display

Gathering Phase

The *Gathering Phase* of an AVC (CP only) conference is the time period during which participants are connecting to a conference. During the *Gathering Phase*, a mix of live video from connected endpoints is combined with both static and variable textual information about the conference into a slide which is displayed on all connected endpoints.



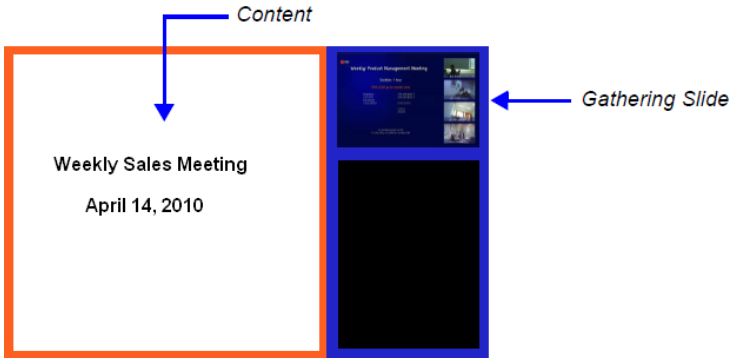
During the Gathering Phase, the audio of all participants can be heard, and the video of active speakers is displayed in the video windows as they begin talking.

All connected participants are kept informed about the current conference status including names of connected participants, participant count, participant type (video/audio) etc.

Gathering Phase Guidelines

- Gathering Phase is only available in AVC only (CP only) conferences. It is not supported in SVC Only conferences.
- The Gathering Phase slide can be displayed at any time during the conference by entering the *Show Participants* DTMF code, ***88**.
  Note: When the display of the Gathering Phase slide is removed, the message overlay text is also removed.
- The names of the first eight participants to connect are displayed. If eight or more participants connect, the 8th row displays "**…**".
- Static text in the Gathering Phase slide such as the field headings: Organizer, Duration, Video/Audio Participants, Access Number, IP are always displayed in the language as configured in the *Polycom Virtual Meeting Rooms Add-in for Microsoft Outlook*. The following languages are supported:
  - English
  - French
  - German
  - International Spanish
  - Korean
  - Japanese
  - Simplified Chinese
- Dynamic text in the Gathering Phase slide such as the meeting name, participants' names, access numbers and the additional information entered in the Info1/2/3 fields of the Gathering Settings tab of the conference Profile are displayed in the language of the meeting invitation.
- The language of a Gathering Phase slide of a conference configured to include a Gathering Phase that is not launched by the Polycom Conferencing Add-in for Microsoft Outlook is configured by the administrator. Using the Collaboration Server Web Client, the administrator selects the language for the Gathering Phase slide. The language selected can be different to that of the Collaboration Server Web Client used by the administrator to perform the configuration.

- Content can be sent during the Gathering Phase. The content is displayed in the large video window of the participant's layout while the Gathering slide is displayed in a smaller video window in the layout.



- Gathering is not supported in Cascading Conferences.

Gathering Phase Duration

The duration of the Gathering Phase can be customized by the administrator so that it is long enough to be viewed by most connected participants yet short enough so as not to over extend into the scheduled conferencing time.

The Gathering Phase duration is configured for the Collaboration Server, by the following System Flags in system.cfg in the **Setup >System Configuration**:

- **CONF_GATHERING_DURATION_SECONDS**

  Range: 0 - 3600 seconds

  Default: 180 seconds

  The Gathering Phase duration of the conference is measured from the scheduled start time of the conference.

  Example: If the value of the flag is set to **180**, the Gathering slide is displayed for three minutes to all participants starting at the conference Start Time, and ending three minutes after the conference Start Time.

  For participants who connect before Start Time, the Gathering slide is displayed from the time of connection until the end of the Gathering duration period.

- **PARTY_GATHERING_DURATION_SECONDS**

  Range: 0 - 3600 seconds

  Default: 15 seconds

  The value of this flag determines the duration of the display of the Gathering slide for participants that connect to the conference after the conference Start Time.

  Participants connecting to the conference very close to of the end of the Gathering Phase (when there are fewer seconds left to the end of the Gathering Phase than specified by the value of the flag) have the Gathering slide displayed for the time specified by the value of the flag.

  Example: If the value of the flag is set to **15**, the Gathering Phase slide is displayed to the participant for 15 seconds.

Enabling the Gathering Phase Display

The Gathering Phase is enabled for per conference in the Conference Profile. The profile also includes the dial-in numbers and the optional additional information to display on the slide.

Conferences that are configured to include a Gathering Phase that are not launched by the *Polycom Conferencing Add-in for Microsoft Outlook* need the following information to be entered via the ***New Profile or Profile Properties — Gathering Settings*** dialog box:

- Display Name (Optional, the Meeting Name is used if left blank.)
- Displayed Language
- Access Number 1 / 2 (Optional.)
- Additional Information (Optional free text)
  - Info 1
  - Info 2
  - Info 3

Conferences launched by the *Polycom Conferencing Add-in for Microsoft Outlook* receive this information from the meeting invitation.

For more information see Defining New Profiles .

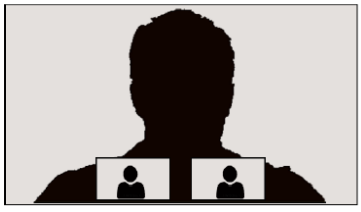
Overlay Layouts

In Overlay Layouts additional participant endpoints can be displayed over the full screen display of the conference speaker.

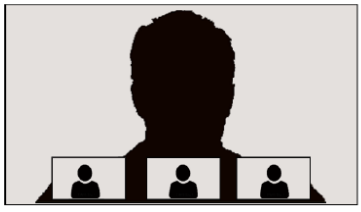
The following Overlay Layouts are available for use in CP Conferences:

1Standalone Endpoint

2 Standalone Endpoints



3 Standalone Endpoints



Although the following Overlay Layout is included in the **Profiles - Video Settings** dialog box, it is not available for use in any *Conferencing Mode* and is only available when included in the *Polycom® Multipoint Layout (MLA)* application:

Single Overlay Cell: 2-4 Screens

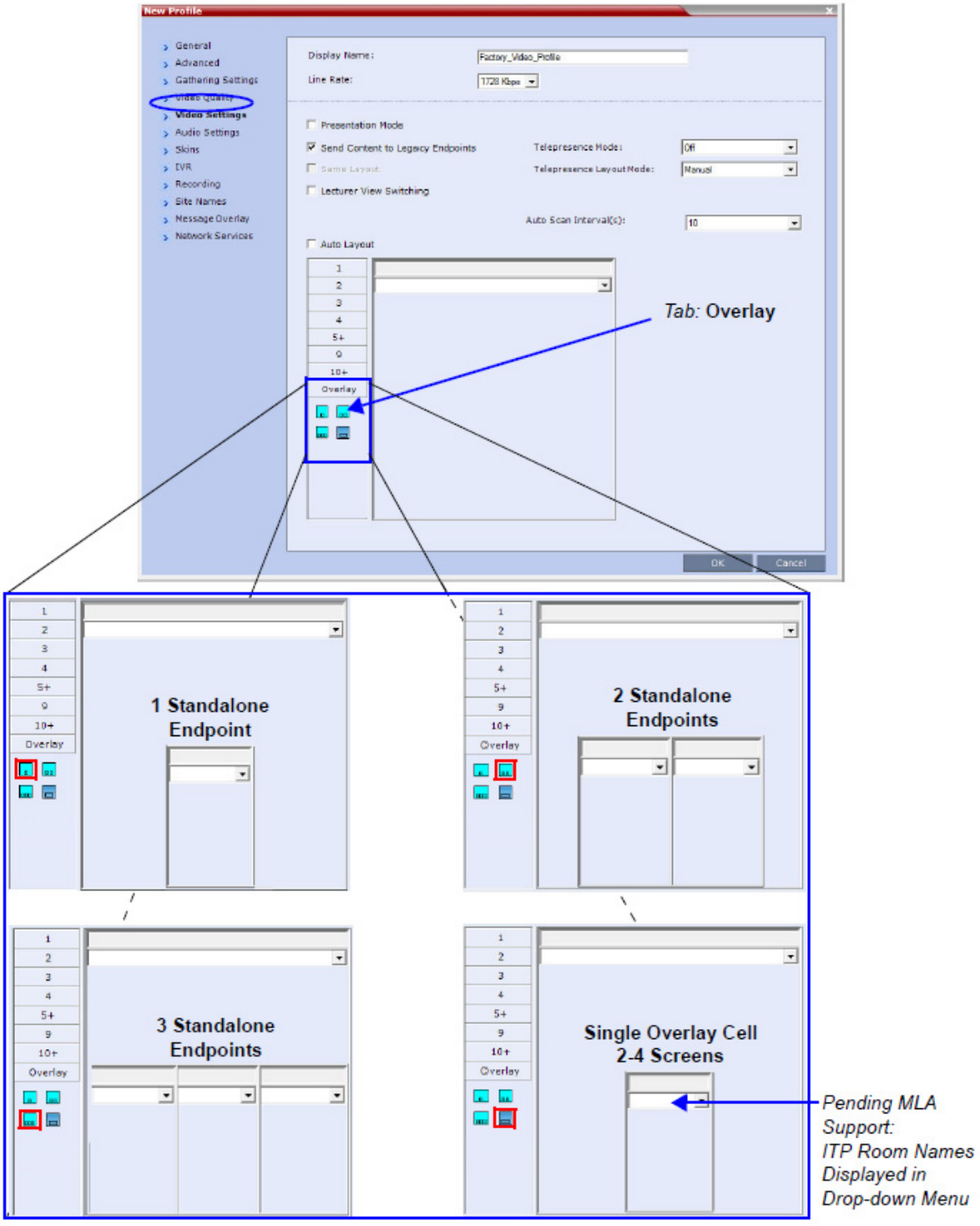
These Overlay Layouts will only be available in *ITP (Telepresence)* conferences when support for Overlay Layouts is included in the *Polycom® Multipoint Layout (MLA)* application.

Guidelines for using the Overlay Layouts

- The Overlay Layouts are supported:
  - In CP Conferencing Mode only.
  - With ITP, non-ITP and CTS endpoints used only as standard endpoints.
  - With both new and classic Skins in Collaboration Server CP mode. For more information see the *Polycom® RealPresence Collaboration Server 800s/Virtual Edition Administrator's Guide*, "*Skins*" on Click the Skins tab to modify the background and frames.
- Overlay Layouts are not supported in ITP conferences as they are not supported by the *MLA* application.
- The Overlay Layouts are 20% of the height of the endpoint display and are supported on endpoints of both 16:9 and 4:3 aspect ratios.
- Overlay Layouts are recommended for use with high resolution endpoints.
- Overlay Layouts are not selected as defaults by the system and are not included in the Auto Layout settings.
- Message Overlay is not affected by the use of Overlay Layouts and is displayed on top of the video layouts.
- Site Names are displayed for all cells. Because the smaller cells are located at the bottom of the large cell, when enabling Site Names it is advisable not to locate the Site Name at the bottom of the cells.
- Standalone Endpoint Cells are displayed each with a border. For all Overlay Layouts, border color is dependent on the selected Skin.
- System behavior for Video Forcing and Personal Layout Control when using the Overlay Layouts during an ongoing conference is the same as for other video layouts.
- Overlay Layouts are only available for selection for the Conference Layout and are not available for selection for Personal Layout.
- During an ongoing conference you cannot select the Overlay Layouts via *PCM* or *Click&View*.
- *PCM* menus can be used when the Overlay Layouts are active, and they are displayed as the top layer.

Selecting the Overlay Layouts

The Overlay Layouts are selected in the **New Profile - Video Settings** dialog box, in the **Overlay** tab of the Video Layout tree.



Site Names Definition

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 in the **Profile - Site Name** dialog box.

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.

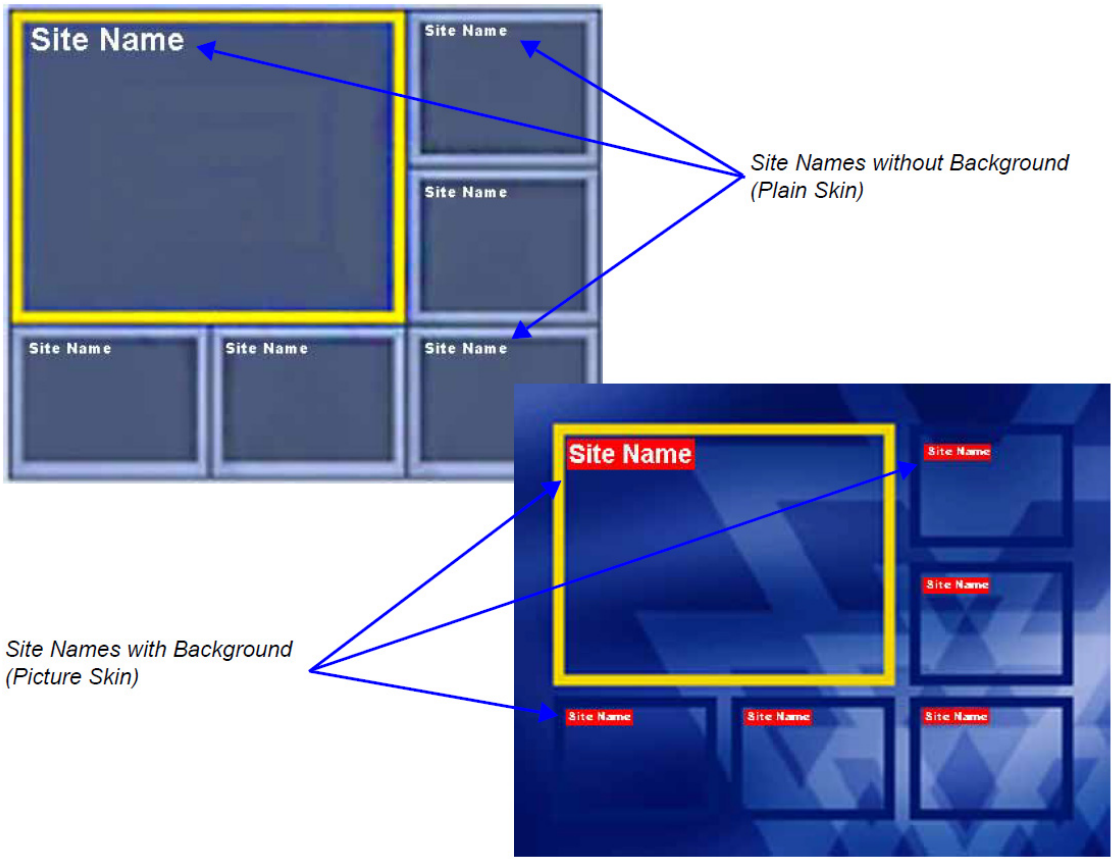
Shorten the Site Name Display

The **SIP_OMIT_DOMAIN_FROM_PARTY_NAME** System Flag can be used to remove Domain Names from SIP dial-in participants' Site Names. This prevents long domain names being appended to SIP participant names, as frequently happens when the Collaboration Server is used with a DMA.

The flag must be manually added to the System Configuration and its value modified as follows:

- **YES** (Default) - The domain name is omitted from SIP dial-in participant names.
- **NO** - The domain name remains as part of SIP dial-in participant names.

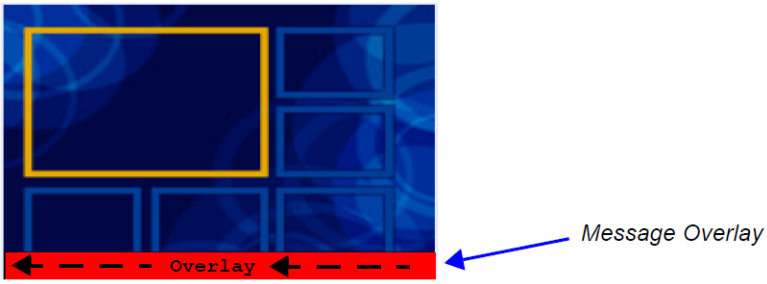
For more information, see Modifying System Flags.


Site Name
Site Name
Site Name
Site Name
Site Name
Site Name
Site Names without Background
(Plain Skin)
Site Name
Site Name
Site Name
Site Name
Site Name
Site Name
Site Name
Site Names with Background
(Picture Skin)

Sending Text Messages During a Conference Using Message Overlay

The Message Overlay option in the Conference Profile allows the operator or administrator to send text messages to all participants during an ongoing conference.

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



Guidelines

- Text messaging using Message Overlay is supported in:
  - Continuous Presence (CP) conferences
  - *Same Layout* mode
  - Encrypted conferences
  - With Unicode or ASCII characters
- Text messages using Message Overlay cannot be displayed:
  - In *Lecture Mode*
  - When the *PCM* menu is active
  - On endpoints that have their video suspended
- Text messaging using Messages Overlay can be enabled, disabled or modified (content and display parameters) during the ongoing conference.
- 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 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 Content or display characteristics (position, size, color and speed) are immediately visible to all participants. When there is a current Message Overlay:
  - The current message is stopped immediately, even it has not completed all of its repetitions.
  - 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 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.
- The text messages cannot be sent via the Content channel.

- Message Overlay text settings are not saved in the *Conference Template* when saving an ongoing conference as a Conference Template.
- Text messages can also be sent to individual or several participants during the ongoing conferences. For more details, see the*Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide* Sending Text Messages During a Conference Using Message Overlay.

For a detailed description of all the Message Overlay parameters, see New AVC CP Profile - Message Overlay Parameters.

Selecting the Chinese Font for Text Display

When using the RMX Web Client or the RMX Manager in Chinese (either Simplified Chinese or Traditional Chinese is selected as an available language in the **Setup > Customize Display Settings > Multilingual Setting**, you can select one of several Chinese fonts for use when sending text over video. The font is used to display text for the following:

- Display of Site Names
- Test messages sent using Message Overlay
- Text displayed on the Gathering slide when Chinese is selected as the display language

Selecting the Chinese Font

The Chinese fonts can be selected in the CP Conference **Profile - Advanced** dialog box only.

The following Chinese fonts are available for selection:

- Heiti (Default)

- Songti
- Kaiti
- Weibei

The Chinese font cannot be changed during an existing conference. It can only be modified in the conference profile.

A participant moved to another conference will be shown the font used by the new conference.

Defining SVC and Mixed CP and SVC Conference Profiles

In the Polycom® RealPresence® CloudAXIS™ Suite, the Conference Profiles are defined in the RealPresence DMA system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

Defining SVC Conference Profiles

The SVC conference Profile definition is started by selecting SVC as the Conferencing Mode. The dialog boxes and their options change as the conference behavior and the MCU video processing change. For example, site name display is performed and controlled by the SVC endpoint and not by the MCU as in CP conferences.

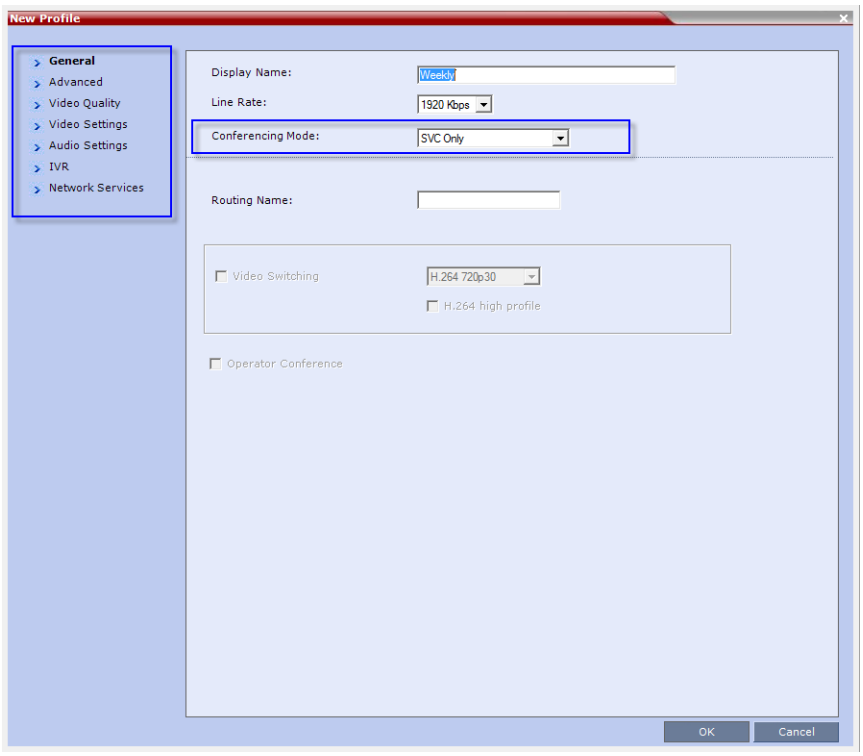
To define SVC Only Profile:

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

   The **New Profile – General** dialog box opens.

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

3 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).

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

New SVC Profile - General Parameters

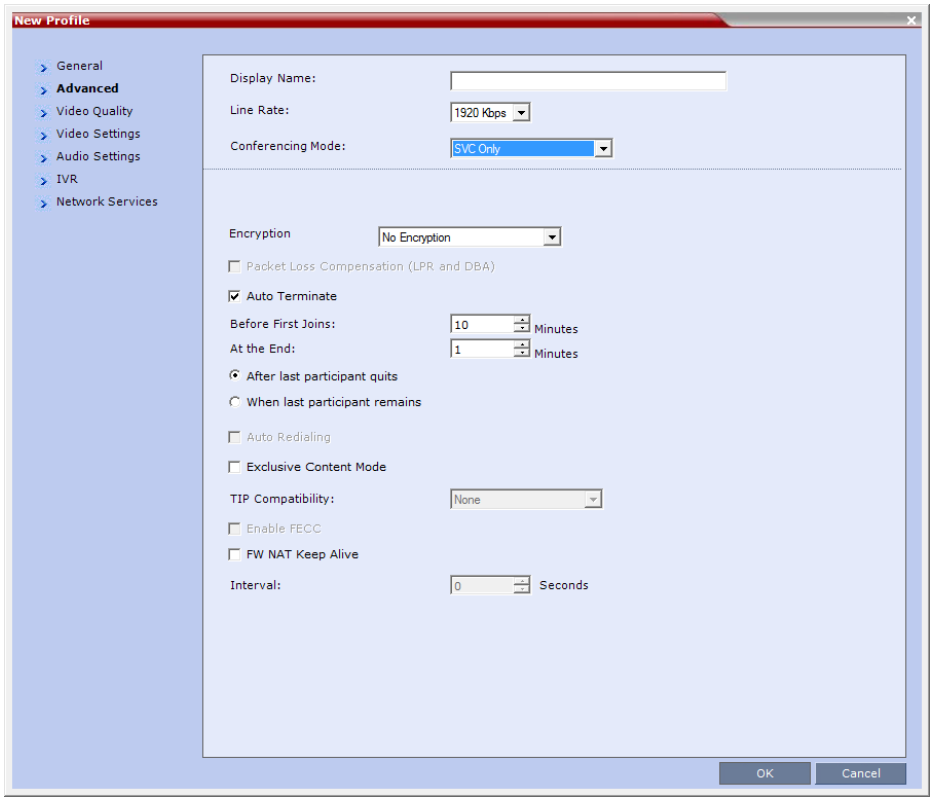
Field/Option	Description
Display Name	Enter a unique Profile name, as follows: • English text uses ASCII encoding and can contain the most characters (length varies according to the field). • European and Latin text length is approximately half the length of the maximum. • Asian text length is approximately one third of the length of the maximum. 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 **1920**kbps. **Note:** This field is displayed in all tabs.

New SVC Profile - General Parameters

Field/Option	Description
Routing Name	Enter the Profile name using ASCII characters set. You can define the Routing Name or it can be 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.



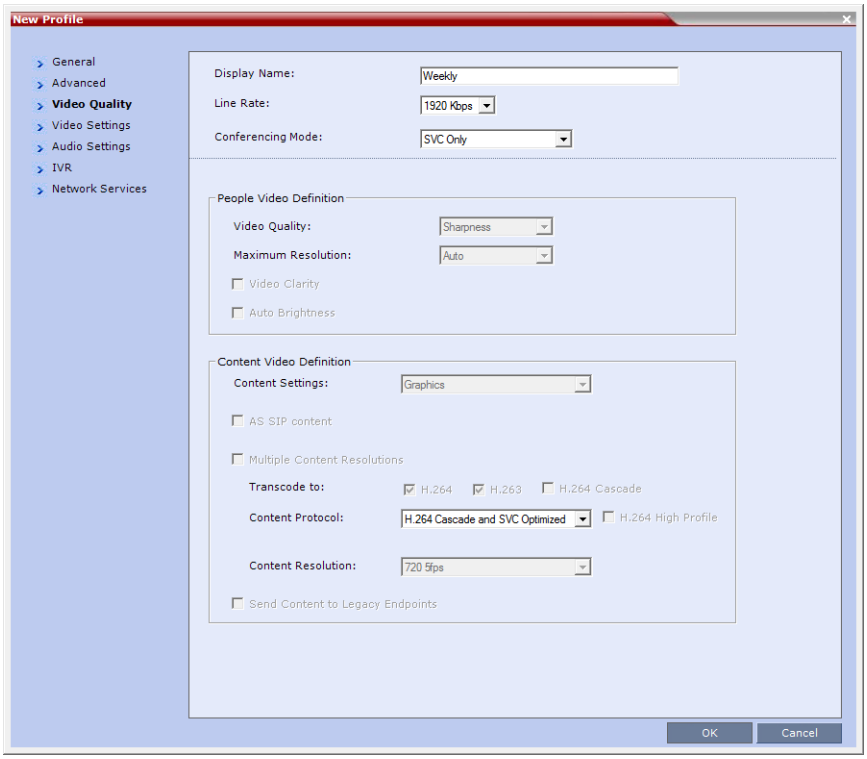
6 Define the following supported parameters:

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. For more information, see Packet Loss Compensation (LPR and DBA) AVC CP Conferences.
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, a *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.



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

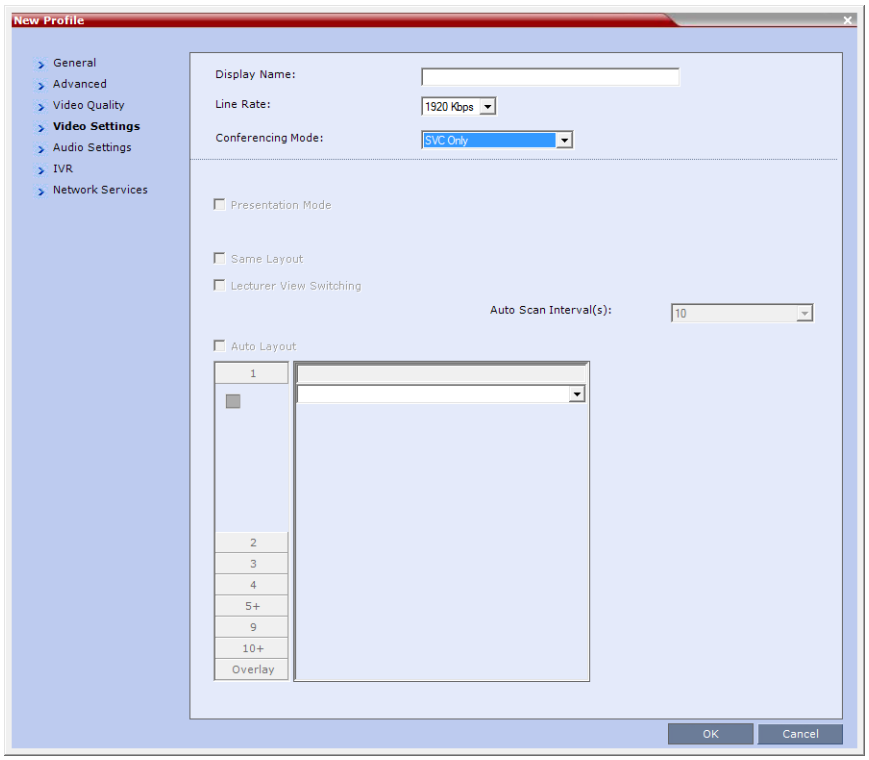
New SVC Profile - Video Quality Parameters

Field/Option	Description
People Video Definition	
Video Quality	Only **Sharpness** is available in SVC Conferencing Mode. The MCU sends the video stream in the resolution required by the endpoint.
Maximum Resolution	Only **Auto** is available in SVC Conferencing Mode. The MCU sends the video stream in the resolution required by the endpoint.
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 Video Preview (AVC Participants Only).

New SVC Profile - Video Quality Parameters

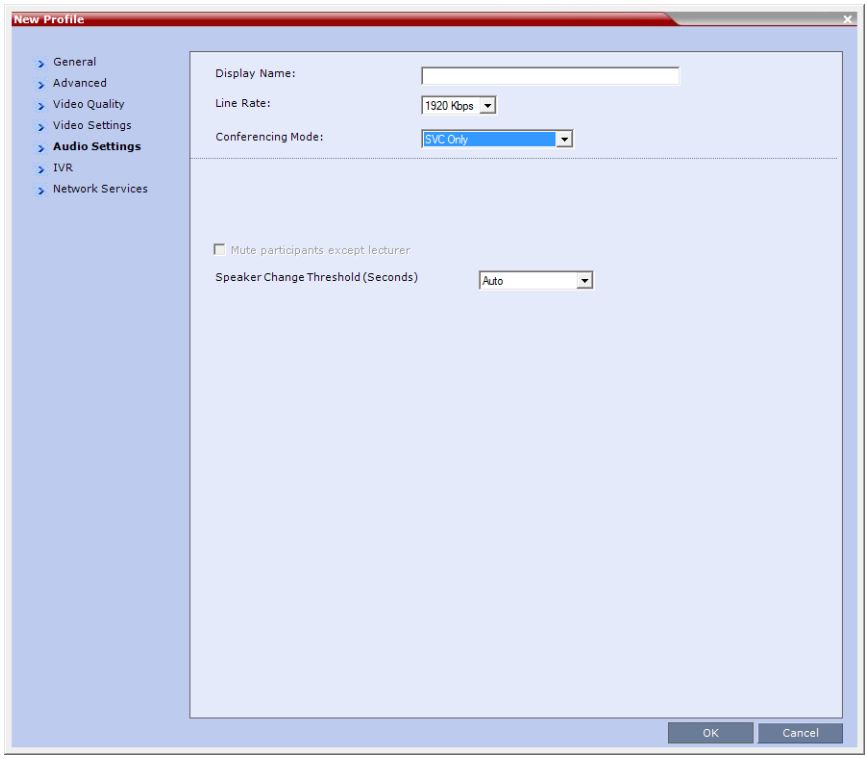
Field/Option	Description
Content Protocol	**H.264 Cascade and SVC Optimized** is the only available Content Protocol for content sharing during SVC-based conferences. In this mode, all *Content* is shared using the *H.264* content protocol and all endpoints must use the set video resolution and frame rate (720p 5fps). Endpoints that do not support these settings cannot share content.

9 Click the **Video Settings** tab.



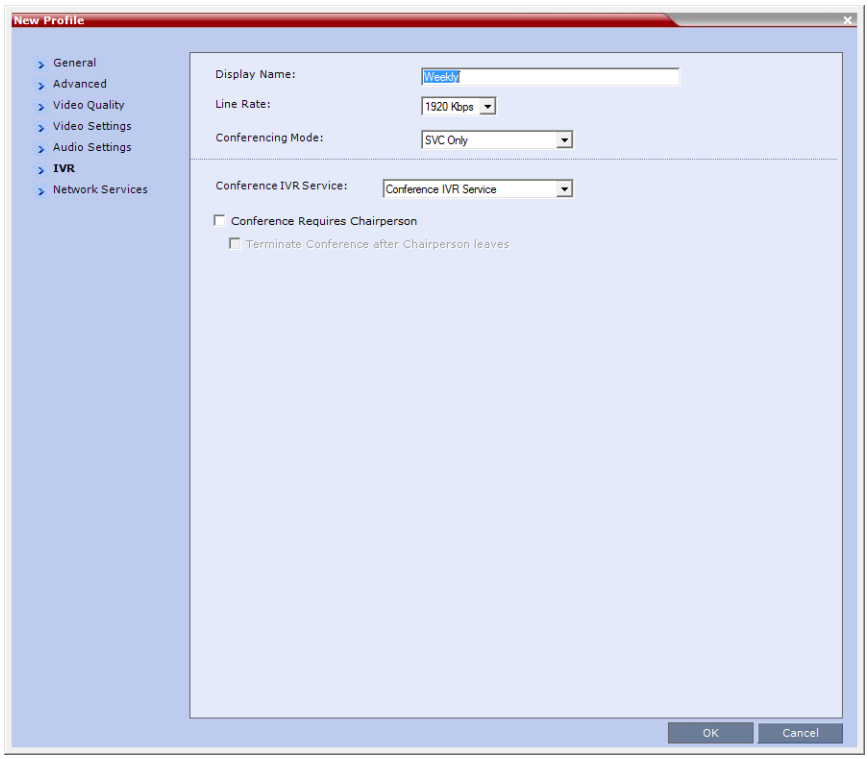
In SVC conferences, each endpoint determines its own video layout and there is no conference level layout selected. Therefore, all the Video Settings parameters are disabled.

10 Click the **Audio Settings** tab.



11 If required, define the **Speaker Change Threshold**: **Auto** (Default, 3 seconds), **1.5.3.5**.
It indicates the amount of time a participant must speak continuously before becoming the speaker.

12 Click the **IVR** tab.



13 If required, set the following parameters:

New SVC Profile - IVR Parameters

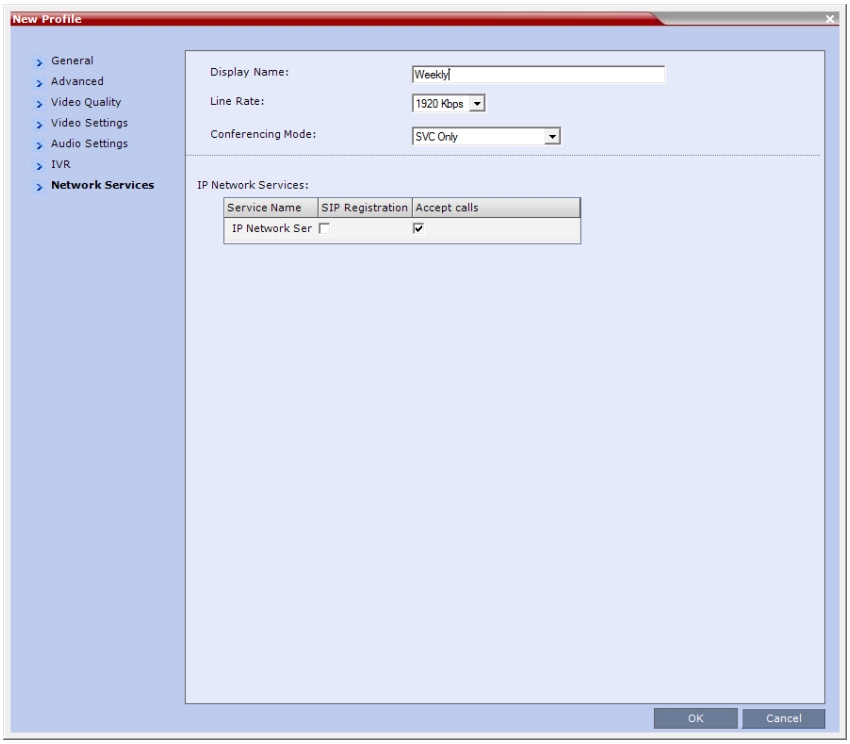
Field/Option	Description
Conference IVR Service	The default conference IVR Service is selected. You can select another conference IVR Service if required.
Conference Requires Chairperson	Select this option to allow the conference to start only when the chairperson connects to the conference and to automatically 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. Terminate Conference After Chairperson Leaves is not supported in cascaded environments.

The following IVR features are not supported during SVC conferences:

- Roll Call
- Invite Participants
- Entry and Exit tones
- Click & View
- PCM

14 Click the **Network Services** tab.

The N**ew Profile - Network Services** tab opens.



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.

15 Define the following parameters:

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.

New SVC Profile - Network Services Parameters

Parameter	Description
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.

16 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 Polycom® RealPresence® CloudAXIS™ Suite, the Conference Profiles are defined in the RealPresence DMA system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

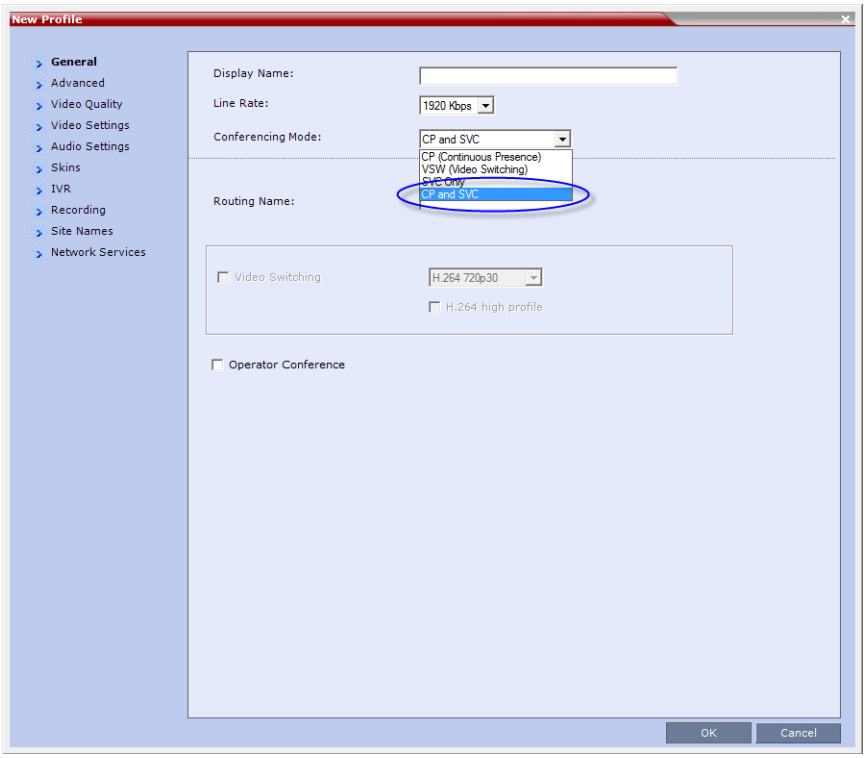
The mixed CP and SVC Profile is based on the CP Profile with a few of the CP options disabled for compatibility between AVC and SVC protocols and to enable the media conversion between these two modes. The **Gathering Settings** and the **Message Overlay** options are unavailable in this Conferencing Mode.

In a mixed CP and SVC conference, the Chairperson can be either an AVC-enabled or SVC-enabled endpoint.

To configure a mixed AVC and SVC conference:

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

   The **New Profile - General** dialog box is displayed.
3. In the **Conferencing Mode** list, select **CP and SVC** to define a mixed AVC and SVC conference.



Using the various Profile tabs, you can define the following profile parameters:

- **CP and SVC Profile - Advanced** parameters - these parameters are the same as for CP conferences. For details, see New AVC CP Profile - Advanced Parameters.
- **CP and SVC Profile - Video Quality** parameters - to enable the sharing of video between SVC and AVC, the common denominator parameters (in this conference, the SVC parameters) are selected for the conference. For more details, see New SVC Profile - Video Quality Parameters.
- **CP and SVC Profile - Video Settings** parameters - the video layout parameters apply only to the AVC-enabled endpoints and do not apply to SVC-enabled endpoints as the SVC endpoints generate their own layout. Options that are not supported in SVC conferencing are disabled in this dialog box, for example, Telepresence Mode. For more details, see New AVC CP Profile - Video Settings Parameters.
- **CP and SVC Profile - Audio Settings** parameters - options that are not supported in SVC conferencing are disabled in this dialog box. For more details, see New AVC CP Profile - Audio Settings Parameters.
- **CP and SVC Profile - Skins** parameters - the display of a video skin applies only to the AVC-enabled endpoints and do not apply to SVC-enabled endpoints as the SVC endpoints generate their own layout.
- **CP and SVC Profile - IVR** parameters - to enable the same IVR behavior and DTMF usage for SVC and AVC, the common denominator parameters (in this conference, the SVC parameters) are selected for the conference. For more details, see New SVC Profile - IVR Parameters.
- **CP and SVC Profile - Recording** parameters - these parameters are the same as for CP conferences as the recording is done in AVC format. For details, see New AVC CP Profile - Recording Parameters.
- **CP and SVC Profile - Site Names** parameters - these parameters are the same as for CP conferences as they apply the AVC-enabled endpoints. SVC-enabled endpoints generate the site name display independent of the MCU. For details, see New AVC CP Profile - Site Names Parameters.
- **CP and SVC Profile - Network Services** parameters - these parameters are the same as for CP and SVC conferences . For details, see New AVC CP Profile - Network Services Parameters.

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 Defining SVC and Mixed CP and SVC Conference Profiles on page 74.

The 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.
- 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 the table below.

Video Quality vs. Line Rate

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 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 571.

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.

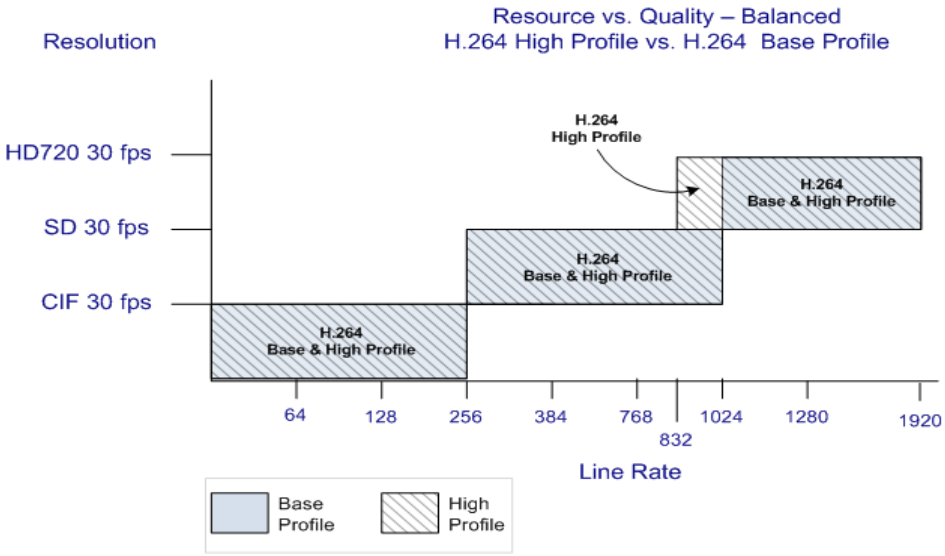
Video Resource Usage

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

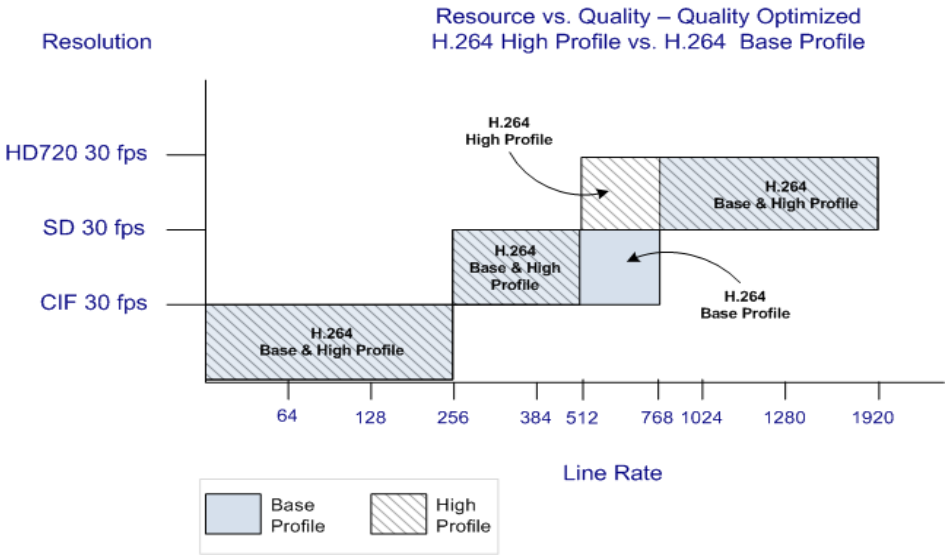
H.264 Base Profile and High Profile Comparison

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.

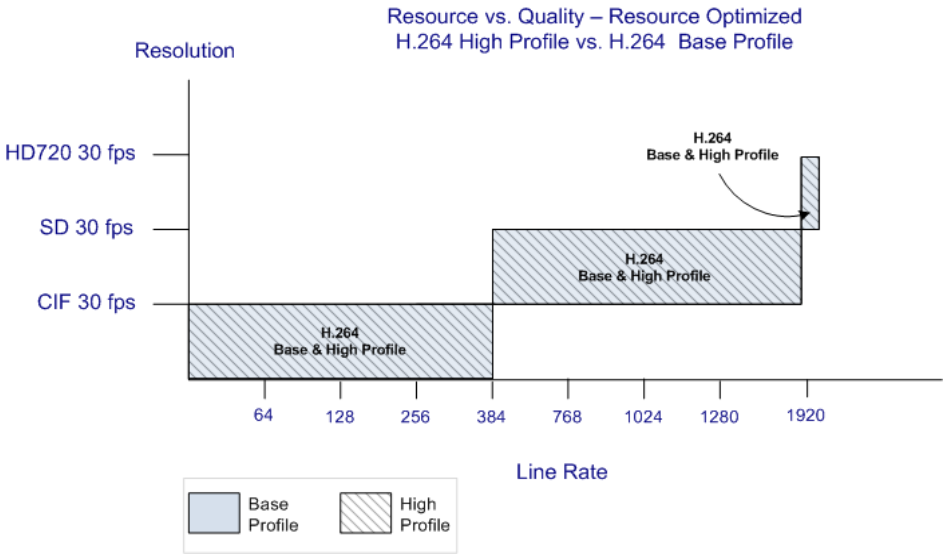
Resolution usage for H.264 High Profile and H.264 Base Profile at various line rates when Resolution Configuration is set to Resource-Quality Balanced



Resolution usage for H.264 High Profile and H.264 Base Profile for Motion at various line rates when Resolution Configuration is set to Video Quality Optimized



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.

Default Minimum Threshold Line Rates and Video Resource Usage

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		Profile	Optimization Mode					
			Balanced		Resource		Video Quality	
			Sharpness	Motion	Sharpness	Motion	Sharpness	Motion
HD1080p30	Default kbps	High	1536		4096		1024	
		Base	4096		4096		1728	
HD720p60	Default kbps	High		1280		1920		832
		Base		1920		1920		1280
HD720p30	Default kbps	High	832		1920		512	
		Base	1024		1920		832	

Resolution		Profile	Optimization Mode					
			Balanced		Resource		Video Quality	
			Sharpness	Motion	Sharpness	Motion	Sharpness	Motion
SD 60	Default kbps	High		768		1024		512
		Base		1024		1024		768
SD 30	Default kbps	High	256		384		256	
		Base	256		384		256	
CIF 60	Default kbps	High		256		384		256
		Base		384		384		256
CIF 30	Default kbps	High	64	64	64	64	64	64
		Base	64	64	64	64	64	64

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 89 and Default Minimum Threshold Line Rates and Resource Usage Summary on page 91.

Guidelines

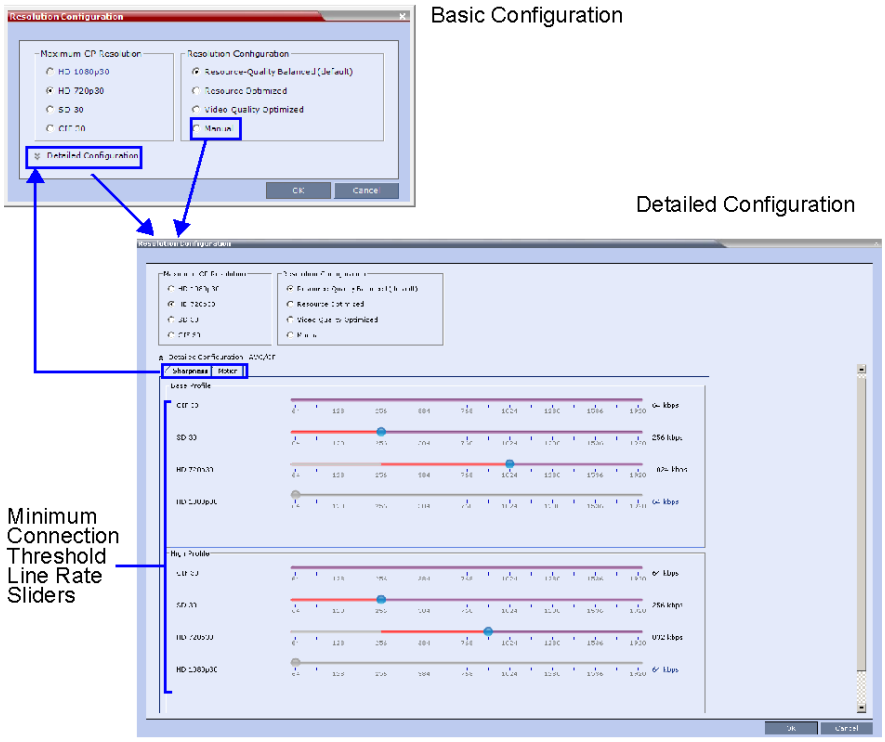
- *Resolution Slider* settings affect all *Continuous Presence* (*CP*) conferences running on the Collaboration Server. *Video Switched* conferences are not affected.
- A system restart is not needed after changing the *Resolution Slider* settings.
- *Resolution Slider* settings cannot be changed if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

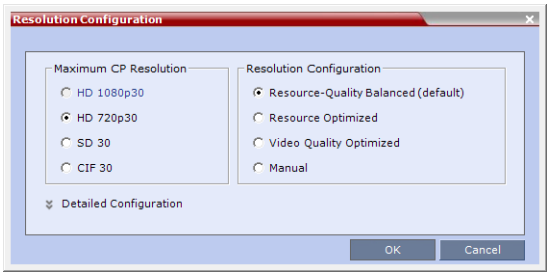


Resolution Configuration - Basic

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

- Max CP Resolution Pane

- Resolution Configuration Pane



Maximum CP Resolution Pane

- The Collaboration Server can be set to one of the following *Maximum CP Resolutions*:HD 1080p30
- 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

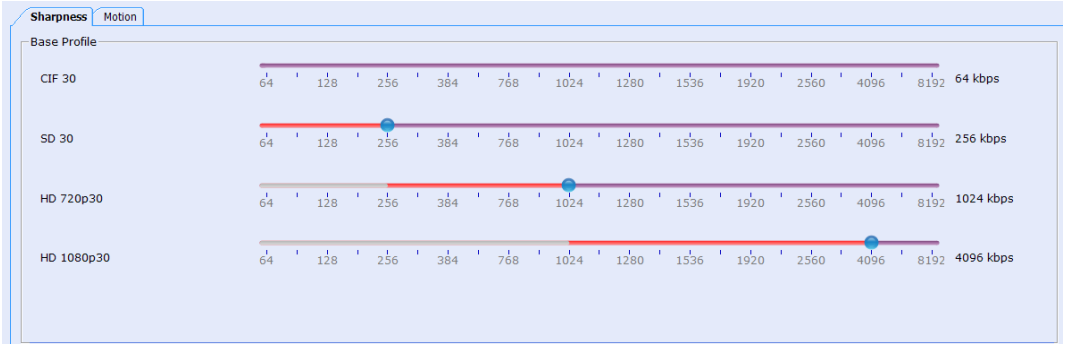
H.264 High Profile allows higher quality video to be transmitted at lower bit rates.

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.



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

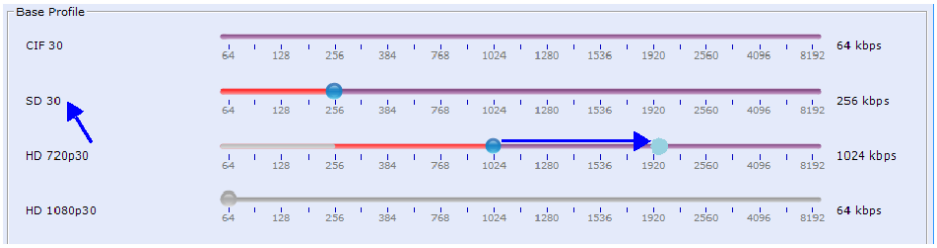
  Although the default minimum threshold bit rates provide acceptable video quality, the use of higher bit rates usually results in better video quality.

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:



- 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:

- HD1060p30
- 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 754.

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.
- HD 1080p 1920 x 1080 pixels at 30 fps.

The video resolution transmitted to any endpoint is determined by the endpoint's capabilities, the conference line rate and the Conference Profile's Motion and Sharpness settings.

w448p Resolution

For improved interoperability with *Tandberg MXP 990/3000* endpoints, the appropriate *System Flag* settings will force the Collaboration Server to send *w448p* (768x448 pixels) at 25fps as a replacement resolution for *WSD15* (848x480) and *SD15* (720x576 pixels).

Guidelines

- The *w448p* resolution is supported:
  - In *CP* mode.

- At conference line rates of 384kbps and 512kbps.
- With *H.323* and *SIP*.

  H.323 endpoints must identify themselves as **Tandberg MXP** during capabilities exchange.
- In all *Video Layouts*.
- In *1x1 Layout*:
  - When *Video Clarity* is **Off**, the Collaboration Server transmits the same resolution as it receives.
  - When *Video Clarity* is **On**, the Collaboration Server changes the transmitted resolution to *w448p*.

  For more information see the *Polycom® RealPresence Collaboration Server 800s/Virtual Edition Administrator's Guide*, Video Clarity on page **2-9** "*Video Clarity"* on page **2-7**.

- Resource consumption for the w*448p* resolution is the same as for *SD* and *WSD* resolutions.

The following table lists the video outputs from the Collaboration Server to the *Tandberg Endpoints* for both *16:9 Aspect Ratio* when the w*448p* resolution is enabled.

Video Output to Tandberg Endpoints- Aspect Ratio 16:9

	Video Quality		Line Rate Kbps	Resolution	Frame Rate fps	Resolution	Frame Rate fps
Network Environment	Tandberg	Collaboration Server		Tandberg to Collaboration Server		Collaboration Server to Tandberg	
H.323 SIP	Motion	Sharpness	384	512x288	30	768x448	25
			512	768x448	30	768x448	25
H.323 SIP	Sharpness*	Sharpness	384	1024x576	15	768x448	25
			512	1024x576	15	768x448	25

***** It is recommend to set the endpoint to **Motion** to ensure the transmission of the higher frame rates of 25fps/30fps to the Collaboration Server.

The following table list the video outputs from the Collaboration Server to the *Tandberg Endpoints* for *4:3 Aspect Ratio* when the w*448p* resolution is enabled.

Video Output to Tandberg Endpoints - Aspect Ratio 4:3

	Video Quality		Line Rate Kbps	Resolution	Frame Rate fps	Resolution	Frame Rate fps
Network Environment	Tandberg	Collaboration Server		Tandberg to Collaboration Server		Collaboration Server to Tandberg	
H.323 SIP ISDN	Motion	Sharpness	384	576x448 ‡	25	768x448	25
			512	576x448 ‡	25	768x448	25

Network Environment	Video Quality		Line Rate Kbps	Resolution	Frame Rate fps	Resolution	Frame Rate fps
	Tandberg	Collaboration Server		Tandberg to Collaboration Server		Collaboration Server to Tandberg	
H.323 SIP ISDN	Sharpness*	Sharpness	384	4CIF	15	768x448	25
			512	4CIF	15	768x448	25

***** It is recommend to set the endpoint to **Motion** to ensure the transmission of the higher frame rates of 25fps/30fps to the Collaboration Server.

‡ *MXP 990/3000* endpoints transmit 576x448 pixels. Other *MXP* endpoints may transmit other resolutions eg. *CIF*.

Content

Sharing and receiving *Content* is supported.

Bandwidth allocated to the *Content* channel during *Content* sharing may cause the video resolution to be decreased as from *w448p* to *w288p*.

When *Content* sharing stops and the full bandwidth becomes available, video resumes at the previous *w448p* resolution.

For more information see the *Polycom® RealPresence Collaboration Server 800s/Virtual Edition Administrator's Guide,* "*H.239 Protocol"* on page **4-2**"*H.239 Protocol"* on page **4-2**.

Packet Loss Compensation

If there is *Packet Loss* in the network and *Dynamic Bandwidth Allocation* (*DBA*) is activated, allocating bandwidth for *Lost Packet Recovery,* video resolution decreases from *w448p* to *w288p*.

When *Packet Loss* ceases and *DBA* no longer needs to allocate bandwidth for *Lost Packet Recovery*, the full bandwidth becomes available and video resumes at the previous *w448p* resolution.

For more information see the *Polycom® RealPresence Collaboration Server 800s/Virtual Edition Administrator's Guide,* Packet Loss Compensation (LPR and DBA) AVC CP Conferences.

Enabling Support of the w448p Resolution

w448p resolution support for *Tandberg* endpoints requires setting of the following entities:

- *Tandberg* endpoint
- Collaboration Server flags
- *Collaboration Server Conference Profile*

Collaboration Server System Flag Settings

- On the Collaboration Server, the *Video Quality* field in the *New Profile - Video Quality* dialog box must be set to **Sharpness**.
  For more information see *Polycom® RealPresence Collaboration Server 800s/Virtual Edition Administrator's Guide, "Defining New Profiles"* on page **2-19***"Defining a CP Conference Profile"* on page **2-11**.

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.

Sharing Content During Conferences

Content such as graphics, presentations, documents, or live video can be shared with conference participants.

Content sharing architecture is comprised of various aspects:

- **Content Control Protocols** - H.239 (for H.323), BFCP (for SIP), or People+Content (Polycom's protocol used for CP conferences prior to H.239 creation)
- **Content Media Protocols** - H.263 (AVC only), H.264 (all conferencing modes), or TIP (Cicso's proprietary protocol for TelePresence endpoints).
- **Content Transmission Methods** - Content Video Switching and Multiple Content Resolutions.
- **Sharing Content in Cascading Environments**

Content Control Protocols

Endpoints wishing to share content, request the Content Token from the MCU (in cascaded environments, the Master MCU). The MCU uses the control protocol to grant the Content Token to the requesting endpoint (unless in Exclusive Content Mode).

Endpoints incapable of sharing content using one of the supported control protocols, can view content via the people video layout (see Sending Content to Legacy Endpoints (CP Only)).

Guidelines for Controlling Content Protocol

- Only the Content Token owner may send content.
- Content Token ownership is valid until:
  - A new endpoint requests token ownership (unless in Exclusive Content Mode).
  - The owner relinquishes it.
  - The Content Token owner endpoint disconnects from the conference.
  - It is cancelled by the MCU user

Supported Content Control Protocols

Polycom supports content sharing using one of the following content control protocols:

- **H.239** - For H.323 participants
- **BFCP** - For SIP participants (over TCP or UDP)
- **People+Content** - Polycom's proprietary content control protocol; for H.323 participants
- **TIP** Auto-Collaboration (CISCO TIP participants). (see Prefer TIP) - TIP Auto-Collaboration for H.264 content sharing

Endpoints supporting the content control protocols above can share content within the same conference.

Video endpoints not supporting one of the content control protocols above, can view content on the people video layout if the conference is set to Send Content to Legacy Endpoints (see Sending Content to Legacy Endpoints (CP Only)).

Note that since TIP content implies using H.264 content media protocol, endpoint supporting only H.263 are considered Legacy content endpoints (see Sending Content to Legacy Endpoints (CP Only)) when TIP content is used.

Content Sharing Using H.239 Protocol

This protocol is used by H.323 endpoints.

The H.239 protocol allows compliant endpoints to share content stream simultaneously with video.

Cascaded links declare H.239 capabilities, and are supported in Star and MIH cascading topologies. For more details, see Cascading Conferences - H.239-enabled MIH Topology.

Endpoints may not send content while connecting to an Entry Queue.

Content Sharing Using BFCP Protocol

This protocol is used by SIP endpoints.

The MCU supports BFCP over either TCP or UDP, which enables the MCU to share content with both SIP client types.

Guidelines for Using SIP BFCP Content

For SIP clients supporting BFCP/TCP or BFCP/UDP:

- BFCP content is not supported over SIP links (in Gateway and cascading scenarios). Therefore, in cascading environment, the cascaded link must be defined as H.323 for content to be shared.
- BFCP/UDP is supported in both IPv4 and IPv6 addressing modes. BFCP/TCP is supported only in IPv4 addressing mode.
- Note that Lync endpoints, though considered as SIP endpoints, do not use BFCP to share content, and use a Microsoft proprietary protocol for that purpose. There are two options to bypass that limitation:
  - Polycom CSS plug-in (see Content Sharing via Polycom CSS Plug-in for Microsoft Lync Clients). This is the preferred option, and is used whenever the CSS plug-in is currently used.
  - Treating Lync endpoints as Legacy, thus viewing content via the people video layout (see Sending Content to Legacy Endpoints (CP Only)).
- 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/BFCP can be disabled by manually adding the ENABLE_SIP_PEOPLE_ PLUS_CONTENT system flag, and setting its value to NO.
- SIP and BFCP capabilities are by default declared to all endpoints.

  Capabilities declaration is controlled by the ENABLE_SIP_PPC_FOR_ALL_USER_AGENT system flag, whose default value is YES, meaning BFCP capability is declared to all vendors' endpoints. When set to NO, the MCU declares SIP over BFCP capabilities only to Polycom and Avaya endpoints. Note that a SIP proxy might remove the agent information, thus preventing the capability declaration to Polycom and Avaya endpoints as well.
- Set ENABLE_FLOW_CONTROL_REINVITE system flag to NO when SIP BFCP is enabled.
- If the system flags mentioned above do not exist in the system, they must be manually added (see Modifying System Flags).

- Due to UC-APL requirement, BFCP over TCP is not supported in Ultra Secure Mode (Collaboration Server 1500/1800/2000/4000).

Content Sharing via Polycom CSS Plug-in for Microsoft Lync Clients

From version 8.1, Polycom CSS (Content Sharing Suite) plug-in for Lync clients allows Lync clients to receive and send content via SIP BFCP, without having to use the people video layout.

The CSS plug-in invokes a separate call for content with or without video. This call may be invoked per a Lync client.

BFCP support in dial-out Connections

For SIP dial-out clients supporting both TCP and UDP, the preferred protocol is BFCP/UDP. However, this preference can be modified, by adding the SIP_BFCP_DIAL_OUT_MODE system flag and modifying its value to TCP (see Manually Adding and Deleting System Flags).

The Collaboration Server's content sharing, as determined by the system flags settings and SIP client capabilities, is summarized in the following table:

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	

BFCP support in dial-in Connections

- The MCU shares content with dial-in SIP clients according to their preferred BFCP protocol.
- SIP clients connected as audio-only cannot share content.

Content Sharing Using People+Content Protocol

People+Content utilizes a different content control protocol, and is Polycom's proprietary protocol used prior to H.239.

This protocol is supported in CP conferences, and is applicable for H.323 endpoints.

Guidelines for Content Sharing Using People+Content Protocol

- If an endpoint supports both H.239 and People+Content protocols, H.239 is selected as the preferred communication protocol.
- **H.263** and **H.263 and H.264 Auto-selection** are the only supported content media protocols usage modes (see MCU Usage Modes of Content Protocols).
- People+Content is enabled by default. It can be disabled for all conferences and endpoints by manually adding the ENABLE_EPC System Flag, and setting its value to NO (default value is YES).

- Endpoints supporting People+Content (for example, FX endpoints) may require a different signaling protocol. For these endpoints, manually add the System Flag CS_ENABLE_EPC, and set its value to YES (default value is NO).

Content Media Protocols

The RealPresence Collaboration Server transmits content using the following content media protocols:

- **H.263 (Annex T)** - Base profile.
- **H.264** - Base profile.

  For single MCU conferences, the MCU determines the media protocol by applying the Highest Common principle (see Highest Common).

H.264 Supported Resolutions for AVC (non-TIP) Conferences

Conference Resolution	Soft MCU	Multiple Content Resolution
720p5	✓	✓
720p30	✓	✓
1080p15	✓	✓

- **TIP** - Cisco's proprietary protocol for TelePresence endpoints. The supported TIP content rate is XGA, 5 fps, 512 Kbps, base profile (see Appendix I - Polycom Open Collaboration Network (POCN)).

LPR has no effect on content rate, though due to the increased number of frames, video rate is decreased.

Content Transmission Methods

There are two methods used for content transmission:

- Content Video Switching
- Multiple Content Resolution

Content Video Switching

When using this method, content sent by endpoints is transmitted as-is to all content-capable participants.

The content rate is determined using one of the following methods:

- Highest Common
- Fixed Rate

Highest Common

This method is applied for single MCU conferences.

The Highest Common method is applied in a few levels:

- Within the same protocol - The highest common content rate
- Between protocols - H.263 and H.264

In this method, the content rate is negotiated to highest common capabilities supported by the endpoints connected to the conference. Therefore, if the conference includes participants supporting both lower and higher content capabilities, the content is sent to all endpoints using the lowest endpoint capabilities.

Note, that when content is currently shared using H.263, the MCU determines the content rate, and the endpoint sharing the content determines the resolution and frame rate. However, when content is shared using H.264, the MCU determines content rate, resolution and frame rate (depending on endpoints capabilities and conference settings).

When a new endpoint with lower content capabilities joins while content is shared, content sharing parameters are downgraded to this endpoint capabilities:

- If downgrade affects the protocol (from H.264 to H.263), content sharing is terminated, protocol automatically downgrades to H.263, or in H.264, and content sharing should be manually resumed.
- If downgrade affects only the content rate, resolution or frame rate (in H.264), the relevant content parameters automatically downgrade during content sharing.

Once this endpoint leaves the conference during content sharing, no change occurs in the protocol or content rate. In H.264 content, resolution or frame rate may automatically upgrade.

Content Setting Highest Common Calculation

For Graphics, Hi-res Graphics and Live Video (see Content Settings), the highest common content is calculated for the conference each time an endpoint connects. Highest Common content bit rate is calculated using line rate (the only factor for H.263), resolution, and frame rate (additional factors for H.264). Therefore, if an endpoint connects to an ongoing conference at lower values of these parameters, content parameters are re-calculated and decreased accordingly.

During content sharing, the MCU does not permit endpoints to increase their content bit rate, only their content resolution.

For more information, see H.263 Content Rate Table and H.264 Highest Common Content Rates Table. For information on minimum content rates as determined by system flags, see H.264 HD System Flags.

Fixed Rate

This method must be used for cascaded and SVC-enabled conferences.

In this method the content rate is fixed, and endpoints not supporting this rate do not receive content over the content channel. Endpoints whose capabilities are too low, can **only view** content over the people video layout (see Sending Content to Legacy Endpoints (CP Only)).

For SVC-enabled conferences, the MCU uses H.264, and 128 Kbps as the fixed content rate.

For H.263 cascaded conferences, the MCU automatically uses a **fixed rate** (once a cascaded link is detected) according to the content rates described in Highest Common and Fixed Content Bit Rate Allocation for H.263.

For H.264 cascaded conferences, the MCU uses a **fixed rate** according to the content rates described in H.264 Cascade and SVC Optimized (Fixed) Content Rates Table.

Multiple Content Resolutions

This option is available only in AVC CP conferencing mode.

In this method, the content is shared in multiple streams, one for each video protocol: H.263 (optional) and H.264HD (mandatory). Separate video resources are used for processing the content for each of the required content streams. The MCU then applies the Highest-Common principle for sharing content with each group of endpoints. In cascading conferences, an additional resource is dedicated for the cascading link(s), with a fixed (H.264 Cascade and SVC Optimized) rate shared over the cascaded link(s).

This allows endpoints with different capabilities to connect/disconnect without having to repeatedly restart content sharing process.

Endpoints not supporting the content capabilities set for the conference, receive the content over the people video layout (see Sending Content to Legacy Endpoints (CP Only)).

Multiple Content Resolutions option is not supported in Ultra Secure Mode.

Guidelines for Sharing Contents using Multiple Content Resolutions

- Multiple Content Resolutions is supported only in CP conferences.
- Content is always provided to H.264 HD endpoints.

- The Send Content to Legacy Endpoints option is set, and cannot be modified.
- Additional resources are allocated to the conference (in addition to resources for the conference participants) for content processing:
  - Resources are allocated only upon content sharing beginning, but once allocated, they remain unchanged until conference ends.
  - If the conference configuration (line rate and content settings) yields HD720p30 as the highest content resolution, 1.5 HD video resources are allocated to the conference, whereas a highest resolution of HD1080p15, increases the allocated video resources to 2 HD.
- An additional content stream can be sent to H.263 endpoints, in which case a separate resource is allocated. H.263 supported resolutions are CIF, 4CIF, XGA.
- Content can be sent to cascaded links, in which case an additional video resource is allocated. The links in both MCUs **must** use the same content parameters (meaning content rate, resolution, and frame rate).
- When resources are insufficient, Multiple Content Resolutions mode is disabled, and content sharing reverts to Content Video Switching mode.
  - If **H.264 Cascade** was selected for the conference, the conference uses **H.264 Cascade and SVC Optimized** as the content protocol.
  - If **H.264 Cascade** was not selected for the conference, **H.264 HD** protocol is used.
- H.264 endpoints with no HD or H.263 capabilities, view content over the video layout (see Sending Content to Legacy Endpoints (CP Only)).
- TIP endpoints cannot share content, only view it over the people video layout.
- When **AS SIP** is enabled for the conference, **Multiple Resolutions** is selected and cannot be modified, **H.264** is enabled, and both **H.263** and **H.264 Cascade** are disabled. Resources are allocated on conference beginning. All this enables AS SIP performance in high-traffic environments.

Content Settings

The Content channel can transmit in one of the following modes:

- **Graphics** – For standard graphics. This is the default mode in AVC conferences and the only supported mode for SVC enabled conferences.
- **Hi-res Graphics** (AVC Only) – Requiring a higher bit rate to increase display quality or highly detailed graphics.
- **Live Video** (AVC Only) – Highest bit rate, for video clips or live video display.
- **Customized Content Rate** (AVC Only) - Allowing manual definition of the Conference Content Rate.

Content sharing consumes a quota of the video rate, depending on the content required quality; the better the content required quality (such as in Live Video) the less bit rate remains for the video.

For each of the settings, the MCU allocates an approximate percentage (determined by the table in Content Sharing Reference Tables) of the conference video capability towards content sharing: for Graphics - 33%, for Hi-res Graphics - 50%, and for Live Video - 66%. However, in actuality this percentage might be lower, since the endpoint capabilities are also taken into account when making this calculation.

Customized Content Rate in AVC CP Conferences

Customized Content Rate functionality may be implemented (see Selecting a Customized Content Rate in AVC CP Conferences) when the content rate automatically calculated by the MCU is unsuitable.

For example, in a cascaded environment, the conference rate must be identical for all links. Yet, capabilities may vary widely between the cascading conferences (such as one conference supporting 2 Mbps, while the other - 512kbps).

The rates listed by the MCU are up to 66% of the conference maximum line rate.

Customized Content rate list comprised of a single value of 0, means the conference bit rate capability is too low to share content.

Selecting a line rate too low for the selected Customized Content Rate, results in an error message display.

MCU Usage Modes of Content Protocols

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** - AVC Only
- **H.263 & H.264 Auto Selection** - AVC Only
- **H.264 Cascade and SVC Optimized** - All conferencing modes
- **H.264 HD** - AVC Only

Endpoints without the content capabilities matching the conference content sharing requirements, can connect to the conference but cannot share or view content in the content channel. Depending on Legacy definitions, they can view content in the people video layout (see Sending Content to Legacy Endpoints (CP Only)).

H.263 (AVC CP Conferences)

In this mode, all endpoints share content using H.263 protocol. Select this option when either most endpoints support H.2634, or to share content over a cascading conference, in which case, the cascading link should be created prior to participants joining the conference.

For single MCU scenario, the Highest Common principle (see Highest Common) is applied to determine content parameters. In cascading environments, a fixed rate is used (see Fixed Rate).

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

Select this option to share content using a mix of H.263 and H.264 capable endpoints. Until version 7.6 (including), this option is named **Up to H.264**.

For single MCU scenario, the Highest Common principle (see Highest Common) is applied to determine content parameters. In cascading environments, an H.263 fixed rate is used (see Fixed Rate) regardless of endpoint capabilities.

H.264 Cascade and SVC Optimized

This content sharing option applies for H.264 content media protocol, and provides fixed content rate (see Fixed Rate) and resolution according to the conference line rate. It must be used for Cascading or SVC-enabled conferences.

The **H.264 Cascade and SVC Optimized** option maintains content quality, and prevents content refreshes upon participants connect/disconnect from the conference.

In cascading environments using this option, the cascade link signaling must be H.323.

- **In AVC conferences** - The H.264 Cascade and SVC Optimized option must be used for AVC cascaded conferences sharing H.264 content. The selected content is defined by the conference parameters:
  - Line rate
  - Content settings (Graphics, Hi-res Graphics, or Live Video)
  - Resolution configuration
- **In SVC enabled conferences** - The H.264 Cascade and SVC Optimized option must be used for SVC enabled conferences sharing content over H.264.The used content rate is 128Kbps (fixed) and 720p5 resolution. Endpoints not supporting the required content parameters (content line rate and resolution) cannot share or view content.

For more information see H.264 Cascade and SVC Optimized (Fixed) Content Rates Table.

H.264 HD (AVC CP default)

H.264 HD may be used for content sharing only for a single MCU, but not in cascading conferences.

H.264 Content Sharing Properties

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

When this protocol option is selected, content minimal bit rate is determined. For more information, see Setting the Minimum Content Rate for Each Content Quality Setting for H.264 HD.

Content parameters are determined by the Collaboration Server applying the Highest Common principle (see Highest Common), for H.264 media content protocol only.

Guidelines for Sharing Content Using H.264 HD

- Only endpoints supporting HD H.264 content (at least HD720p5) can share content.
- This option is not available in SVC-enabled conferencing modes.

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

System flags determine the minimum line rate required for endpoints to share H.264 high quality content for each of the Content Settings: Graphics, Hi Resolution Graphics and Live Video.

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

To change the system flag value, the flag must be first manually added to the System Configuration (see Modifying System Flags.

Content Sharing Related Issues

Sharing Content in Cascaded Environments

In cascaded environments, content must be shared using the same content rate, and in H.264 conferences, the same resolution and frame rate as well.

Sharing content has different constraints and guidelines over each of the signaling protocols:

- **Over H.323** cascaded links -
  - Fixed content rate is used.
  - In cascading environments with non-Polycom MCUs, the Polycom MCU must be defined as Master
  - **For H.263** cascading links:
    - The cascading link must be created before connecting the participants.
    - The cascaded link's Master/Slave role must be determined, in topologies including more than two MCUs.
  - **For H.264** cascading links, the following should be determined in advance:
    - The **H.264 Cascade and SVC Optimized** content protocol must be selected.
    - The cascaded link's Master/Slave role must be configured for the link's participants.
- **Over SIP** - No content sharing is supported over SIP cascaded links.

Gateway calls are implemented as cascaded links, typically SIP, in which case, content cannot be shared via the content channel. It may be viewed over the people video layout, depending on Legacy configuration (see Sending Content to Legacy Endpoints (CP Only)).

Sending Content to Legacy Endpoints (CP Only)

The Collaboration Server can be configured to send content to endpoints not supporting the conference content parameters (legacy endpoints) over the people video layout, thus allowing the participants to view content. However, these endpoints cannot share content.

Guidelines for Sending Content to Legacy Endpoints

- A separate HD video resource is allocated to the conference for content sending to legacy endpoints.

  Allocation is performed only once a legacy content endpoint is connected to the conference, and a content session is initiated and transmitted via the people video layout. Once the resource is allocated, it remains allocated to the conference until its end.

  If the system cannot allocate the resource required for sending the content, the conference status changes to Content Resource Deficiency, and content cannot be sent to the legacy endpoints.
- Endpoints receiving content via the people video layout, use the same video protocol and resolution they use for receiving video.
- Content cannot be sent to Legacy endpoints when Same Layout mode is selected for the conference.
- When content is transmitted, endpoints' Site Names cannot be viewed.

- Content can be sent to Legacy endpoints in gateway calls, depending on content configuration on Master/Slave MCUs (for example, if the gateway conference is using H.263 content media protocol, and the conference MCU is using H.264 content media protocol).
- Content becomes unavailable to Legacy endpoints moved to an Operator conference.
- A Polycom FX endpoint dialing in to a Collaboration Server receives content using People+Content. However, an FX endpoint dialed out from a Collaboration Server with receives content via the people video layout using People+Content only if Send Content to Legacy Endpoints is enabled in the Conference Profile.

Content Display on Legacy Endpoints

When content is sent to content legacy endpoints, their video layout automatically changes to the content layout defined by the system flag LEGACY_EP_CONTENT_DEFAULT_LAYOUT, and the content is shown in the larger/top-left cell. The video layouts of the other conference participants do not change.

The switch to the content layout occurs in Auto Layout, Presentation Mode, Lecture Mode, and when a layout is selected for the conference.

In Lecture Mode, when content is sent to legacy endpoints, switching to the content layout, results in the content shown in the lecturer/speaker window, while the lecturer is shown in the second window. If the layout contains more than two windows, all other windows are empty. The non-lecturer legacy content participants see the lecturer in full screen.

The LEGACY_EP_CONTENT_DEFAULT_LAYOUT flag's default is a layout of 1+4, where the content is shown in the larger window, and the conference participants are shown in the smaller windows. This default can be changed (see Changing the Default Layout for Content Display on Legacy Endpoints).

When content is stopped, the layout of the legacy participants terminates as well.

Legacy participants can change their layout using Click&View. In such a case, the content is forced to the larger/top-left cell.

The Collaboration Server user can also change the layout for the legacy content endpoints participants (selecting personal layout).

When forcing a video participant to the Content window (instead of Content), the Content display can be restored only by selecting any other video layout.

Sending Content to Legacy Endpoints in Telepresence Mode

To ignore personal layouts during Telepresence conferences (while working with MLA), set the value of the flag FORCE_LEGACY_EP_CONTENT_LAYOUT_ON_TELEPRESENCE to YES.

If the layout for displaying content in Legacy endpoints include multiples cells, MCU may populate Telepresence room streams sources in remote cells.

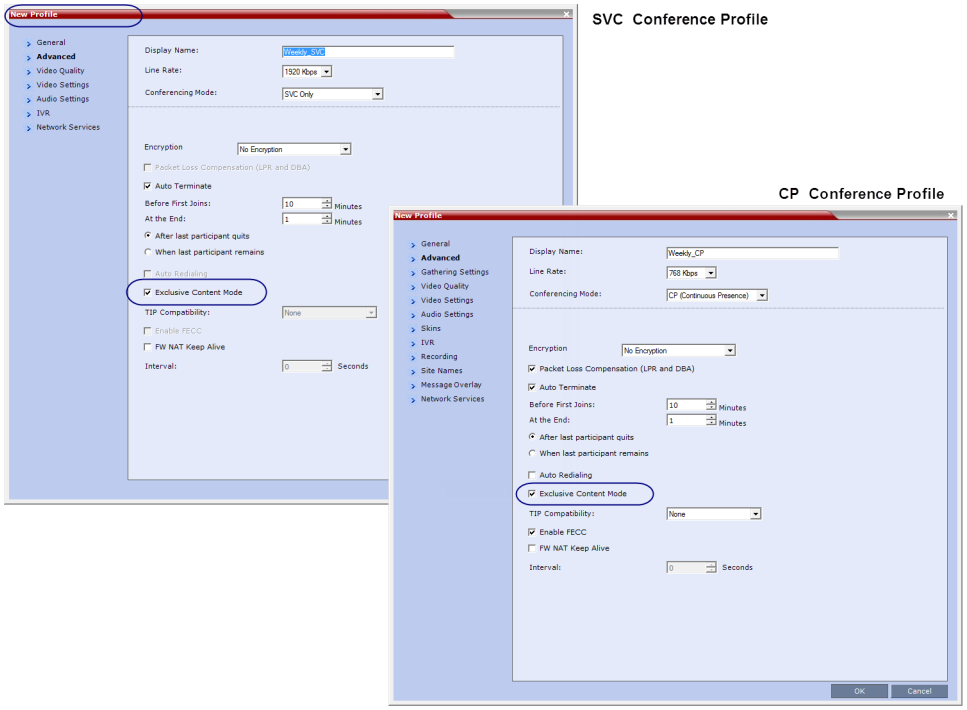
Exclusive Content Mode

In this mode, the MCU prevents participants other than the current content sharer, from sharing content. Exclusive content may be limited to the lecturer, as described below.

To modify exclusive content mode by granting (or cancelling) of token ownership, see Giving and Cancelling Token Ownership (AVC Participants)).

Guidelines for Sharing Content in Exclusive Content Mode

- Exclusive Content Mode is available in all conferencing modes.
- Exclusive Content Mode is enabled or disabled (system default) by a check box in the **Conference Profile** - **Advanced** tab, or during an ongoing conference using the **Conference Properties** - **Advanced** tab.



- In Exclusive Content Mode, when the RESTRICT_CONTENT_BROADCAST_TO_LECTURER system flag is set to:
  - **NO** - The first participant to send content becomes the Content Token holder, and releasing the Content Token allows other participants to acquire the token, and begin transmitting content.
  - **YES** - Only the designated Lecturer can be the Content Token holder.
- The Exclusive Content Mode check box replaces the EXCLUSIVE_CONTENT_ MODE system flag used in previous versions to control exclusive content mode for the system.
- In Exclusive Content Mode, an endpoint attempting to send content, immediately after another endpoint starts sending content, results with a momentary interruption (slide), before resuming normal content stream.

Forcing Other Content Capabilities

The H239_FORCE_CAPABILITIES system flag allows additional control of content sharing:

- **When set to NO (default)** - The MCU merely verifies the endpoint supports the content protocols: H.263 or H.264.
- **When set to YES -** The MCU verifies frame rate, bit rate, resolution, annexes, and all other parameters of content as declared by an endpoint during the capabilities negotiation phase. If the endpoint does not support the content capabilities of the MCU, the participant cannot share content over a dedicated content channel.

Managing Noisy Content Connections

The system can identify participants sending frequent content display refresh requests (usually as a result of a problematic network connection), which cause frequent refreshing of content display, and degrading of viewing quality.

When the system identifies such “noisy” participants, it automatically suspends these requests to avoid affecting the quality of the content viewed by the other conference participants. This process is controlled by the following system flags:

- **MAX_INTRA_REQUESTS_PER_INTERVAL_CONTENT -**

  The maximum number of refresh (intra) requests per 10-second intervals allowed for an endpoint. Beyond that number, content sent by this participant is identified as “noisy”, and its refresh requests are suspended.

  Default setting: 3
- **MAX_INTRA_SUPPRESSION_DURATION_IN_SECONDS_CONTENT -**

  The duration, in seconds, for ignoring the participant’s content display refresh requests.

  Default setting: 10
- **CONTENT_SPEAKER_INTRA_SUPPRESSION_IN_SECONDS** -

  The interval, in seconds, between content refresh (intra) requests sent from the MCU to the content sender due to refresh requests initiated by other conference participants. Additional refresh requests received within that interval are deferred to the next interval.

  Default setting: 5

Useful Procedures in Content Sharing

For all MCUs, content sharing parameters are defined in the conference profile **Video Quality** dialog box.

Defining Content Sharing Parameters for a Conference

For RealPresence Collaboration Server VE, the available content options change according to the selected conferencing mode.

To set the content sharing parameters:

» In the **Content Video Definition** section, set the values for the **Content Settings** and **Protocol** as follows:

Content sharing 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. For a description of each of these options, see Content Settings.
AS-SIP Content	AS-SIP is a SIP implementation utilizing SIP's built-in security features. When selected, content is shared using the Multiple Resolutions mode, and is not supported in any other content sharing mode.
Multiple Resolutions *(CP conferencing mode only)*	Select this check box to enable Multiple Content Resolutions mode, for both H.263 and H.264 content protocols. When enabled, **H.264** is always selected and cannot be modified. **Optional.** Select additional protocols: • **H.263** - If the conference includes endpoints with H.263 capabilities. • **H.264 Cascade** - If the conference includes cascading links and you want to define the video settings for content sharing. For more information, see Sharing Content Using Multiple Content Resolutions Mode.
Content Protocol	• **H.263** (AVC Only) - Content is shared using the H.263 protocol. • **H.263 & H.264 Auto Selection** (AVC Only) - Content is shared according to conference participants capabilities. • **H.264 HD** (AVC Only, default) - Content is shared using the H.264 HD protocol. • **H.264 Cascade and SVC Optimized** - Content is shared using the H.264 content protocol (fixed), and optimized for use in SVC only and cascaded conferences. For a detailed description of each of these settings, see Content Media Protocols.

Content sharing Options

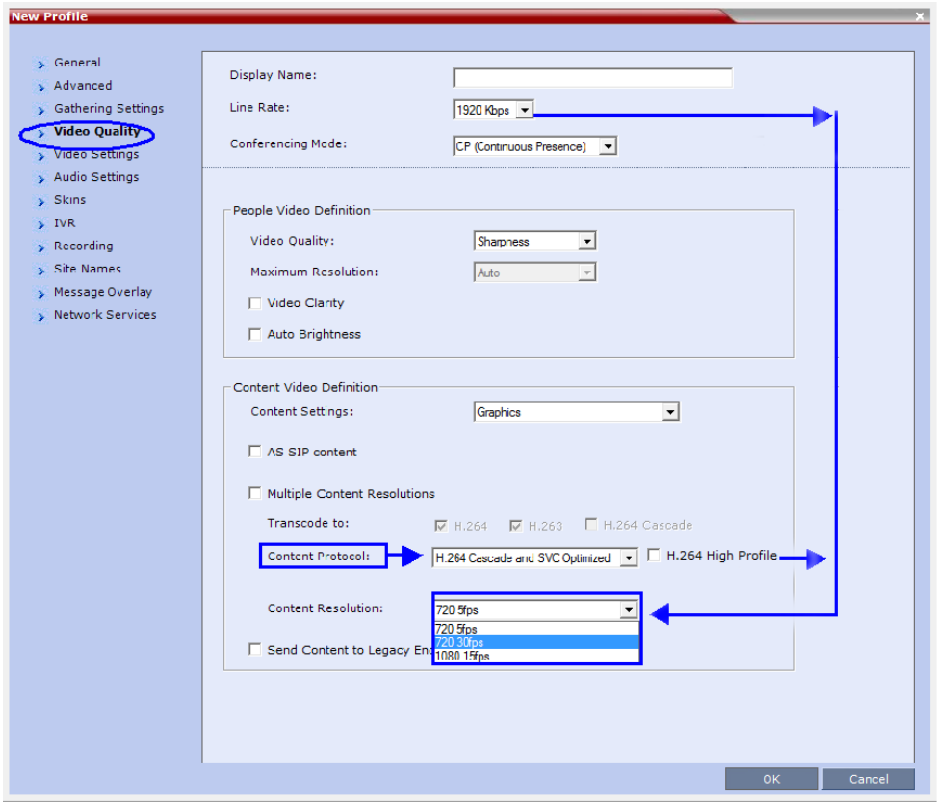
Field	Description
Content Resolution	Select a Content Resolution from the pull-down menu. The Content Resolutions available for selection are dependent on the content sharing mode (Highest Common or Multiple Content Resolutions), Line Rate and Content Settings selected for the conference. For a full list of Content Resolutions see H.264 Supported Resolutions for AVC (non-TIP) Conferences. **Note:** This field is displayed only when **H.264 Cascade and SVC Optimized** is selected, and is enabled for selection in CP conferences (AVC CP) Only. This option is disabled in SVC conferences.
Send Content to Legacy Endpoints	When enabled (default), content is sent to H.323/SIP (Collaboration Server VE) endpoints not supporting MCU content control protocol (legacy endpoints) over the people video layout (see Sending Content to Legacy Endpoints (CP Only)).

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

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

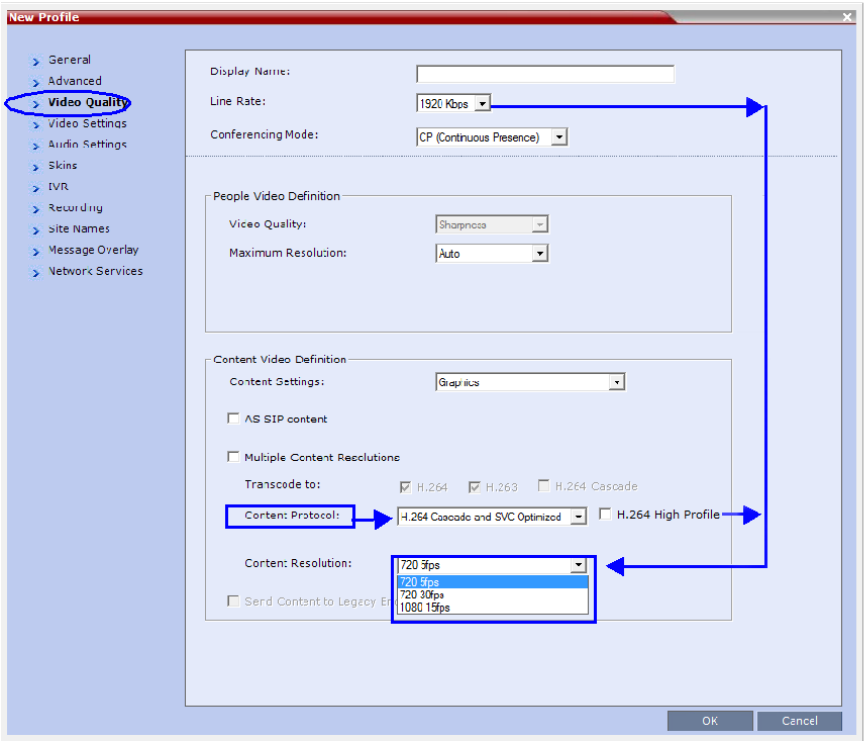
The **Content Resolution** value determines the fixed resolution and frame rate to be used for content sharing in cascaded conferences.

The **Content Resolutions** available for selection are dependent on the **Line Rate** and **Content Settings** selected for the conference.



AVC CP Conferencing Mode

The **Content Rate** drop-down menu list depends on **Customized Content Rate** being selected as the **Content Setting**, and cannot exceed 66% of the conference line rate capability.



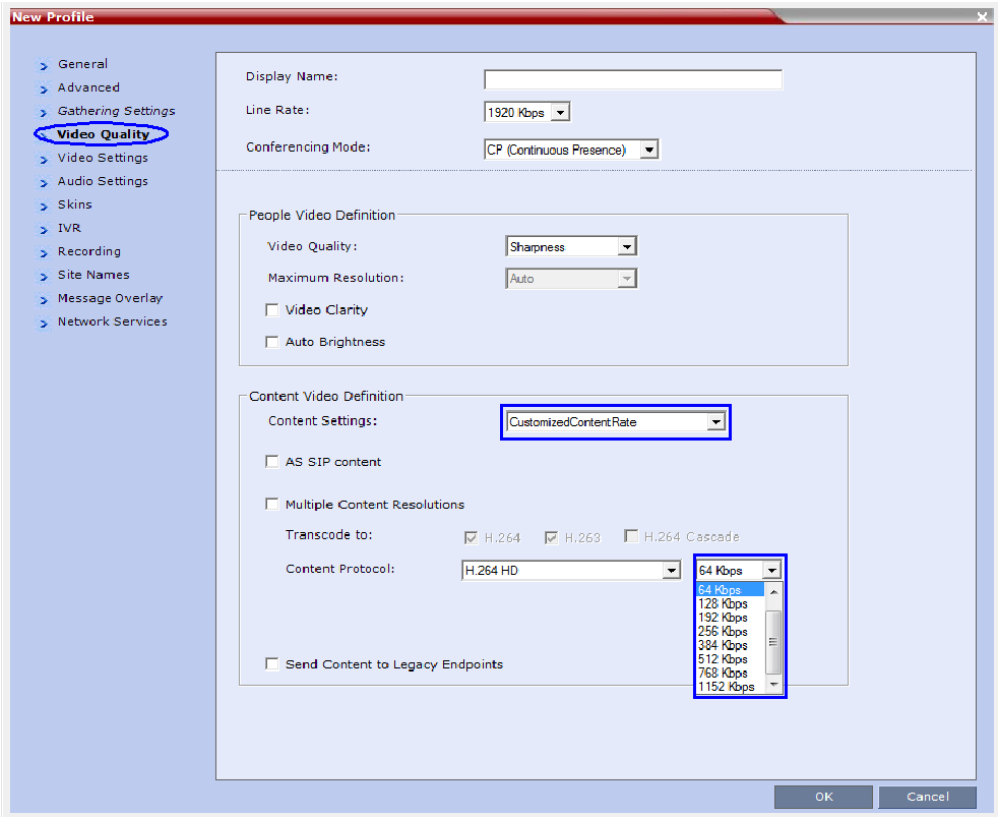
CP Conferencing Mode§

Selecting a Customized Content Rate in AVC CP Conferences

To Select the Customized Content Rate:

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 rates vary according to the selected conference **Line Rate**.



2 Select the required content rate.

If Customized Content Rate is already selected along with the content rate, and you attempt to modify the conference line rate to a value that does not support the selected customized content rate, an error message is displayed.

Modify the Content Line Rate or Conference Line Rate, or modify the Content Setting.

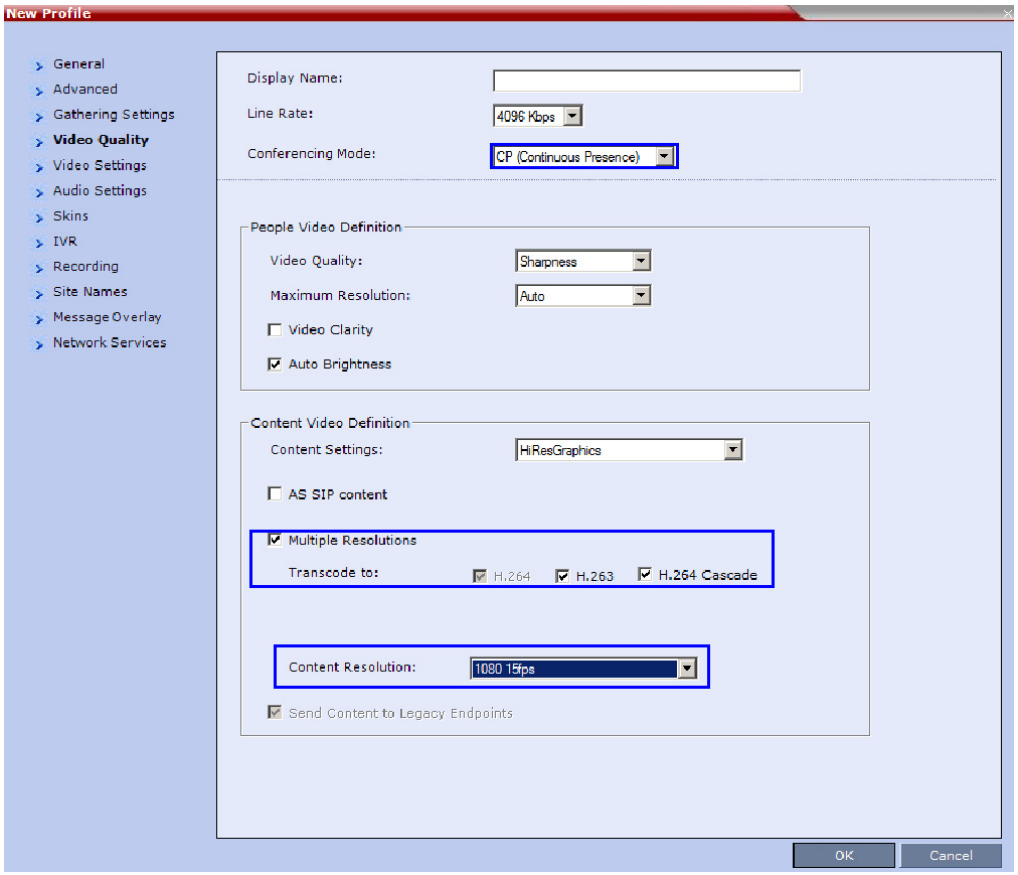
Sharing Content in Multiple Content Resolutions Mode

For information on this content sharing mode see Multiple Content Resolutions.

The Multiple Content Resolutions mode can be modified in the conference Profile, in the **Video Quality** Tab.

To enable Multiple Content Resolutions*:*

1 If the Conference Mode is not CP (Continuous Presence), select **CP**.



2 Select the **Multiple Resolutions** check box.

By default, **H.264** is always selected, and cannot be modified.

3 Select additional protocols:

- **H.263** - If the conference includes H.263 (only) capable endpoints.
- **H.264 Cascade** - If the conference includes cascading links.

If **H.264 Cascade** is selected, select the **Content Resolution**.

4 Click **OK**.

Changing the Default Layout for Content Display on Legacy Endpoints

The default layout used for displaying content for legacy endpoints, is defined by the system flag LEGACY_EP_CONTENT_DEFAULT_LAYOUT.

The configured default layout is 1+4 ( CP_LAYOUT_1P4VER). You can change the default layout configuration by modifying this flag value.

To modify System Flags:

1. On the Collaboration Server menu, click **Setup > System Configuration**.

   The **System Flags** dialog box opens.

2. In the **MCMS_PARAMETERS** tab, double-click the **LEGACY_EP_CONTENT_DEFAULT_LAYOUT** flag.

   The **Edit Flag** dialog box is displayed.

3. In the **Value** field, modify the flag value to the desired layout (see Legacy Content Endpoint Default Layouts Table).

4. Click **OK**.

   The flag is updated in the **MCMS_PARAMETERS** list.

5. Click **OK**.

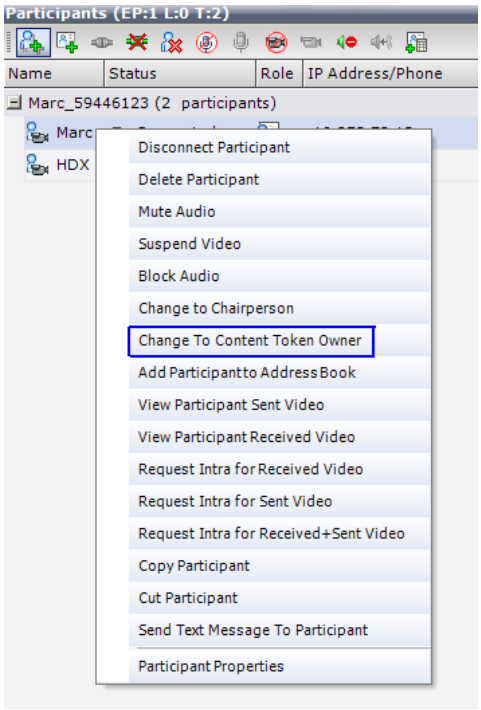
6. For flag changes (including deletion) to take effect, reset the MCU (see Resetting the RMX).

Giving and Cancelling Token Ownership (AVC Participants)

For information on exclusive content ownership, see Exclusive Content Mode.

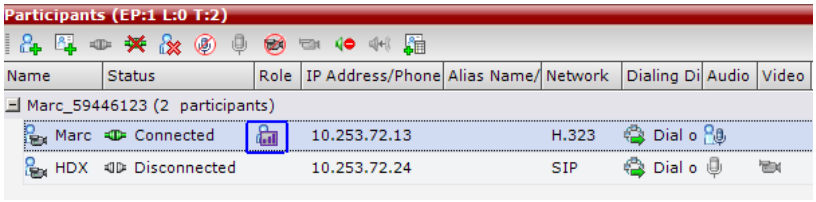
To give token ownership:

1 In the **Participants** list, right-click the AVC-enabled endpoint you wish to define as the exclusive Content Token owner.



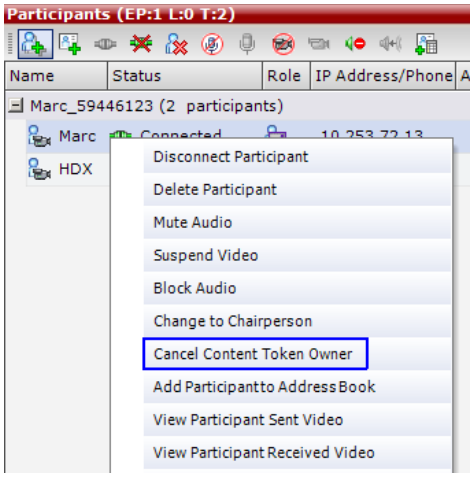
2 Select **Change To Content Token Owner** in the drop-down menu.

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



To cancel token ownership:

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



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

Content Token ownership is cancelled for the endpoint.

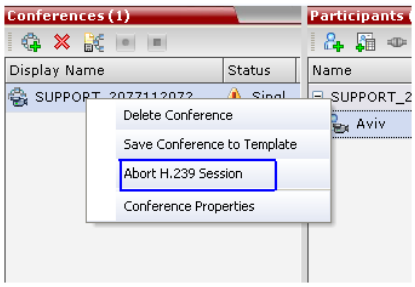
Stopping a Content Session

In some cases, when a participant ends the content sharing session by his/her endpoint, the content token is not released, preventing other participants from sharing content.

The Collaboration Server user can withdraw the content token from the current holder, and 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 select **Abort H.239 Session**.



Content Sharing Reference Tables

Resolutions and Content Rate Reference Tables

The values in the tables below indicate the **maximum** negotiated content rate (for both H.263 and H.264) and resolution/frame rate (only for H.264).

When using Highest Common, endpoints may lower the content parameters, whereas when using fixed rate, endpoints must comply with the content parameters as determined by the MCU.

The **actual** content parameters used by the content sharing endpoint are determined by that endpoint, and may be lower.

H.263 Content Rate Table

The table below describes the content rates for both Highest Common and Fixed content bit rates.

Highest Common and Fixed Content Bit Rate Allocation for H.263

Content Settings / MCU	64	128	256	384	512	768	1024	1152 / 1280	1536	1920
	96	192	320			832		1472	1728	
Graphics 33%										
RPCS VE	0	64	64	128	128	256	256	384	512	512
Hi-res Graphics 50%										
RPCS VE	0	64	128	192	256	384	512	512	768	1024
Live Video 66%										
RPCS VE	0	64	128	256	384	512	512	768	1024	1280

H.264 Resolution per Content Rate Table

The table below describes the resolution as negotiated by the MCU according to the content rate.

Maximum Negotiated Resolution and Frame Rate per Content Rate for H.264 Base Profile

Bit Rate Allocated to Content Channel (Kbps)	Maximum Negotiated Content	
	Resolution	Frames/Second
64-512	H.264 HD720	5
512-768	H.264 HD720	30
768-1280	H.264 HD1080	15

H.264 Highest Common Content Rates Table

The table below summarizes the Highest Common maximum content rates as negotiated by the MCU. This table applies only to single MCU (non-cascading) and non-SVC enabled conferences.

The values in the table below are inapplicable if lower than those of the respective system flags (see Setting the Minimum Content Rate for Each Content Quality Setting for H.264 HD).

Highest Common Content Bit Rate for H.264 Base Profile

Content Settings / MCU	64	96	128	192	256	320	384	512	768	832	1024	1152	1280	1472	1536	1728	1920
Graphics 33%																	
RPCS VE			64		64		128	128	256		256	384			512		512
Hi-res Graphics 50%																	
RPCS VE			64		128		192	256	384		512	512			768		1024
Live Video 66%																	
RPCS VE			64		128		256	384	512		512	768			1024		1280

H.264 Cascade and SVC Optimized (Fixed) Content Rates Table

The table below summarizes the content rates as they are determined according to resolutions and conference line rates.

For information on the supported resolutions for each of the MCU types see H.264 Resolution per Content Rate Table.

Highest Common Content Bit Rate for H.264 Base Profile

Content Settings / MCU	64 / 96	128 / 192	256 / 320	384	512	768 / 832	1024	1152 / 1280 / 1472	1536 / 1728	1920
Graphics 33%										
RPCS VE		64	64	128	128	256	256	384	512	512
Hi-res Graphics 50%										
RPCS VE		64	128	192	256	384	512	512	768	768
Live Video 66%										
RPCS VE		64	128	256	384	512	512	768	1024	1280

Bit Rate of Content Channel per Line Rate for H.264 Cascade and SVC Optimized for Base Profile

Cascade Resolution	64 / 96	128 / 192	256 / 320	384	512	768	832 / 1024	1152	1280	1472 / 1536 / 1728	1920
Graphics 33%											
720p5		64	64	128	128	256	256	256	256	256	256
720p30										512	512
1080p15											768
Hi-res Graphics 50%											
720p5		64	128	192	256	384	384	384	512	512	512
720p30								512	512	512	512
1080p15										768	768
Live Video 66%											
720p5		64	128	256	384	512	512	768	768	768	768
720p30							512	768	768	768	768
1080p15								768	768	768	768

Legacy Content Endpoint Default Layouts Table

The table below describes the supported Legacy content layouts, and their corresponding LEGACY_EP_CONTENT_DEFAULT_LAYOUT flag values.

LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values

Layout	Flag Value
	CP_LAYOUT_1X1
	CP_LAYOUT_1X2
	CP_LAYOUT_1X2HOR
	CP_LAYOUT_1X2VER
	CP_LAYOUT_2X1
	CP_LAYOUT_1P2HOR
	CP_LAYOUT_1P2HOR_UP
	CP_LAYOUT_1P2VER
	CP_LAYOUT_2X2
	CP_LAYOUT_1P3HOR_UP
	CP_LAYOUT_1P3VER
	CP_LAYOUT_1P4HOR_UP
	CP_LAYOUT_1P4HOR
	CP_LAYOUT_1P4VER
	CP_LAYOUT_1P5
	CP_LAYOUT_1P7
	CP_LAYOUT_1P8UP

LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values

Layout	Flag Value
	CP_LAYOUT_1P8CENT
	CP_LAYOUT_1P8HOR_UP
	CP_LAYOUT_3X3
	CP_LAYOUT_2P8
	CP_LAYOUT_1P12
	CP_LAYOUT_4X4

Implementing Media Encryption for Secured Conferencing

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 Dial Out Recording Link Encryption.
- 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.

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.In CISCO TIP environments, dial in endpoints that are registered to CUCM can only connect as non-encrypted when the conference encryption is set to “Encrypt when possible” as the CUCM server sends the Invite command without SDP.

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), *U*ndefined 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

The following table 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.

Connection of Participants to the Conference based on 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.

The following table 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.

Connection of Undefined Participants to the Entry Queue Based on 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 5-53:

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.

The following table 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.

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.

Enabling Media Encryption for a Conference

Media encryption is enabled at three levels:

- MCU level - Setting the Encryption Flags

- Conference level - Enabling Encryption in the Profile
- Participant level - Enabling Encryption at the Participant Level

You must first set the system flags for the MCU before media encryption can be enabled for the conference and participants.

Setting the Encryption Flags

Enabling the media encryption for the MCU is usually performed once an it is applicable to all conferences running on the MCU.

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**.

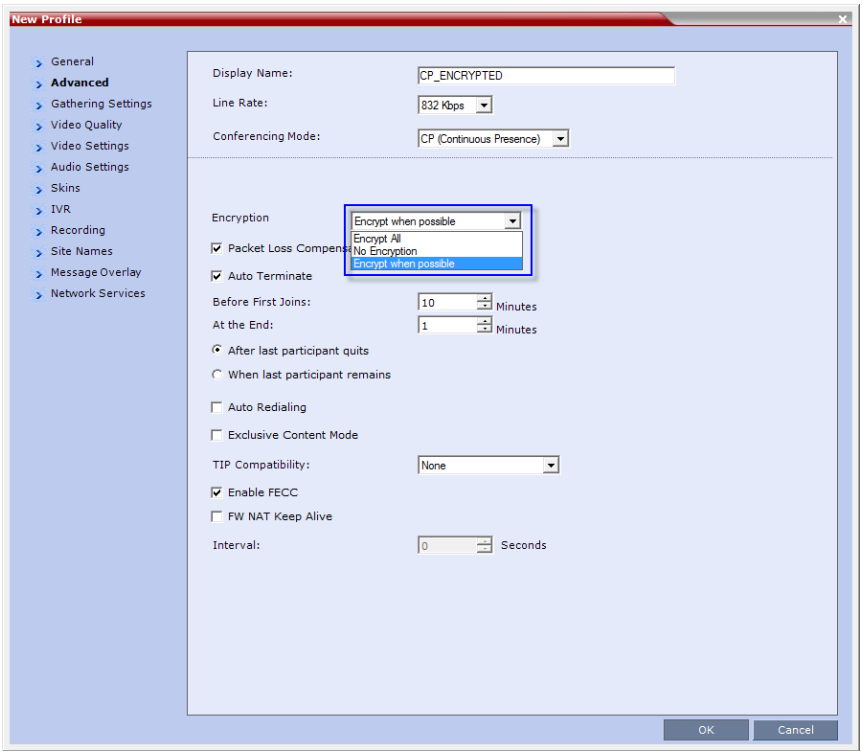
   For more information, see Modifying System Flags.
5. Reset the MCU for flag changes to take effect.

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:



- **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.

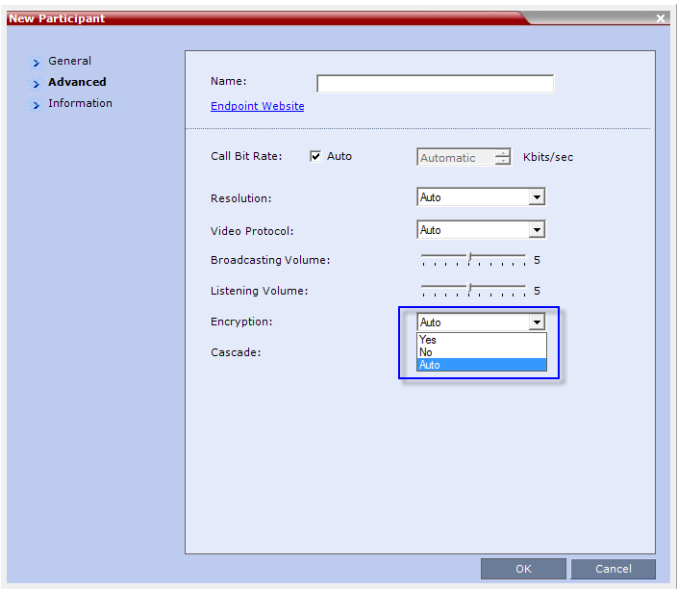
  For more information about recording encrypted conferences, see Dial Out Recording Link Encryption.

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**.



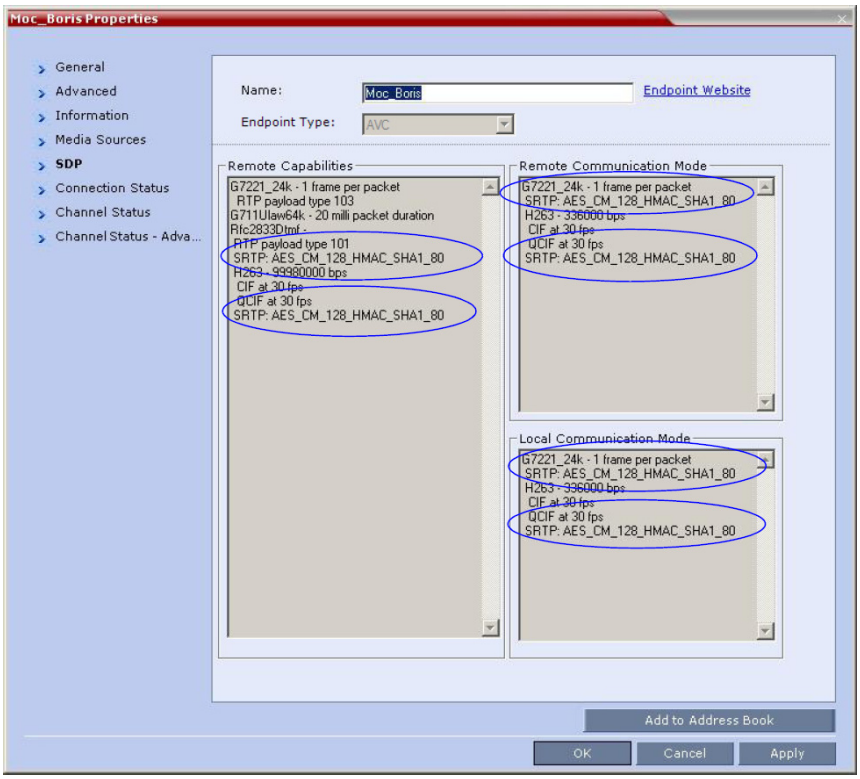
- **Auto** - The participant inherits the conference/Entry Queue encryption setting. The participant connects as encrypted only if the conference is defined as encrypted.
- **Yes** - The participant joins the conference/Entry Queue as encrypted.
- **No** - The participant joins the conference/Entry Queue 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.

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).



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** dialog box, Security Failure indication, and the Cause box identifies the encryption related situation.

For more information about monitoring, see Conference and Participant Monitoring .

Setting Conferences for Telepresence Mode (AVC CP)

Collaboration Server supports the Telepresence Mode in AVC CP conferences allowing multiple participants to join a telepresence conference from RPX and OTX high definition rooms as well as traditional, standard definition video conferencing systems.

OTX (Telepresence) and *RPX (Realpresence)* room systems are configured with high definition cameras and displays that are set up to ensure that all participants share a sense of being in the same room.

Participants using two RealPresence RPX HD 400 Room Systems

The following are examples of situations where an Collaboration Server is needed for *Telepresence* configurations:

- RPX to OTX
- RPX 2-cameras/screens to RPX 4-cameras/screens
- 3 or more RPXs
- 3 or more OTXs

Collaboration Server Telepresence Mode Guidelines

System Level

- The Collaboration Server system must be licensed for **Telepresence Mode**.
- The system must be activated with a **Telepresence** enabled license key.

Conference Level

- The **Telepresence Mode** and **Telepresence Layout Mode** fields are only displayed in the Conference Profile dialog box if the Collaboration Server has a Telepresence license installed.
- A Telepresence conference must have Telepresence Mode enabled in its profile.
- In Telepresence Mode, ITP sites are automatically detected.

- When Telepresence mode is selected in a conference profile, the following options are disabled:
  - Borders
  - Site Names
  - Speaker Indication
  - Skins
  - Same Layout
  - Presentation Mode
  - Auto Layout
  - Lecture Mode
- The master (center) camera is used for video, audio and content.
- *Conference Templates* can be used to simplify the setting up Telepresence conferences where precise participant layout and video forcing settings are crucial. Conference Templates*:*
  - Save the conference Profile.
  - Save all participant parameters including their Personal Layout and Video Forcing settings.
- An ongoing Telepresence conference can be saved to a Conference Template for later re-use.

  For more information see Using Conference Templates.

Automatic Detection of Immersive Telepresence (ITP) Sites

When the conference *Telepresence Mode* is set to Auto (Default) *ITP* endpoints are automatically detected.

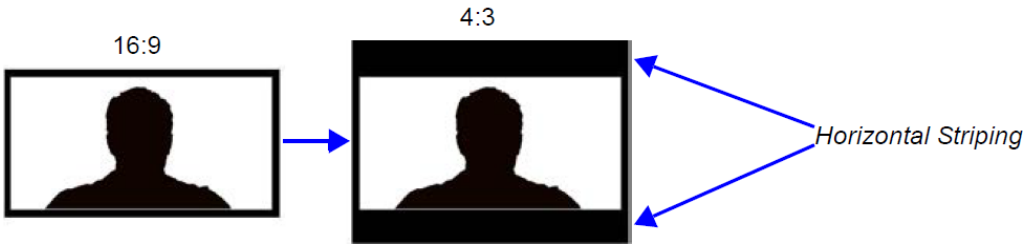
If an *ITP* endpoint is detected in such conference, *ITP* features are applied to **all** endpoints and the Collaboration Server sends conference video with the following options disabled:

- Borders
- Site names
- Speaker indication
- Skins
- Same Layout
- Presentation Mode
- Auto Layout
- Lecture Mode

The ITP features are dynamic, and if all ITP endpoints disconnect from the conference, normal conference video is resumed for the remaining all participants. ITP features are re-applied to all participants should an ITP endpoint re-connects to that conference.

Horizontal Striping

Horizontal Striping is used by the Collaboration Server in order to prevent cropping and preserve the aspect ratio of video for all Telepresence Modes.

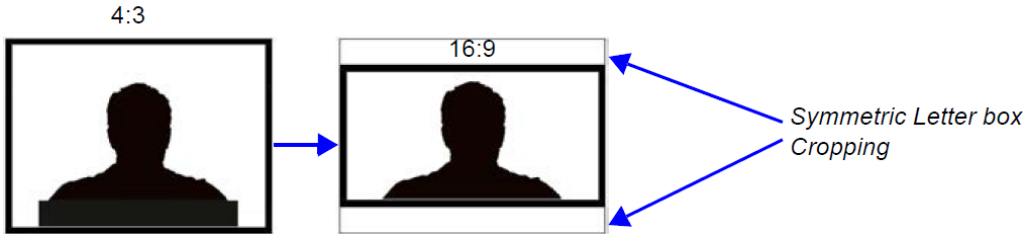


Cropping

Cropping is used by the Collaboration Server in order to preserve the aspect ratio of video for all Telepresence Modes.

Cropping is controlled by the **ITP_CROPPING** system flag in the system configuration, providing different cropping options according to the endpoints participating in the Telepresence conference.

By default, the flag is set to **ITP**. In this mode, the area to be stripped is cropped equally from the top and the bottom (as shown in the example below). For more details, see Modifying System Flags.



Gathering Phase with ITP Room Systems

When a conference is configured to include a Gathering Phase, only one endpoint name is displayed for the ITP room in the connected participant list of the Gathering slide. The ITP room endpoint with the suffix “1” in its name receives the Gathering slide.

Aspect ratio for standard endpoints

Standard endpoints (non-ITP) receive video from the Collaboration Server with the same aspect ratio as that which they transmitted to the Collaboration Server.

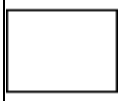
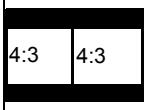
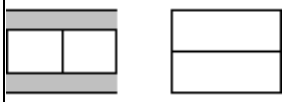
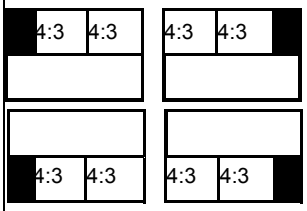
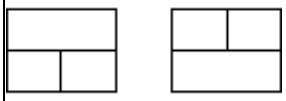
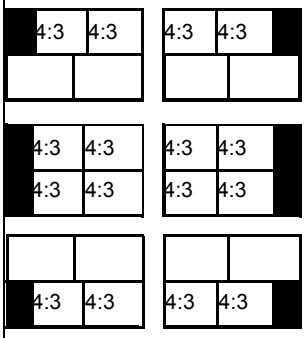
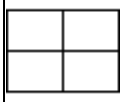
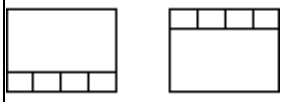
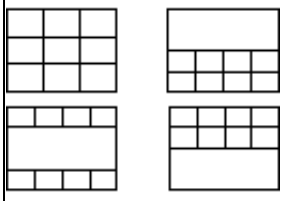
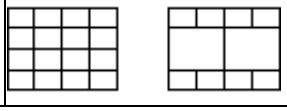
Skins and Frames

When Telepresence Mode is enabled, no Skin is displayed and the system uses a black background. Frames around individual layout windows and the speaker indication are disabled.

RPX and OTX Video Layouts

Additional video layouts have been created to give Telepresence operators more video layout options when configuring OTX and RPX room systems. These additional video layout options are available to all endpoints on both conference layout and Personal Layout levels.

OTX / RPX – Additional Video Layouts

Number of Endpoints	Layouts
1	
2	 
3	 
4	 
5	
9	
10+	

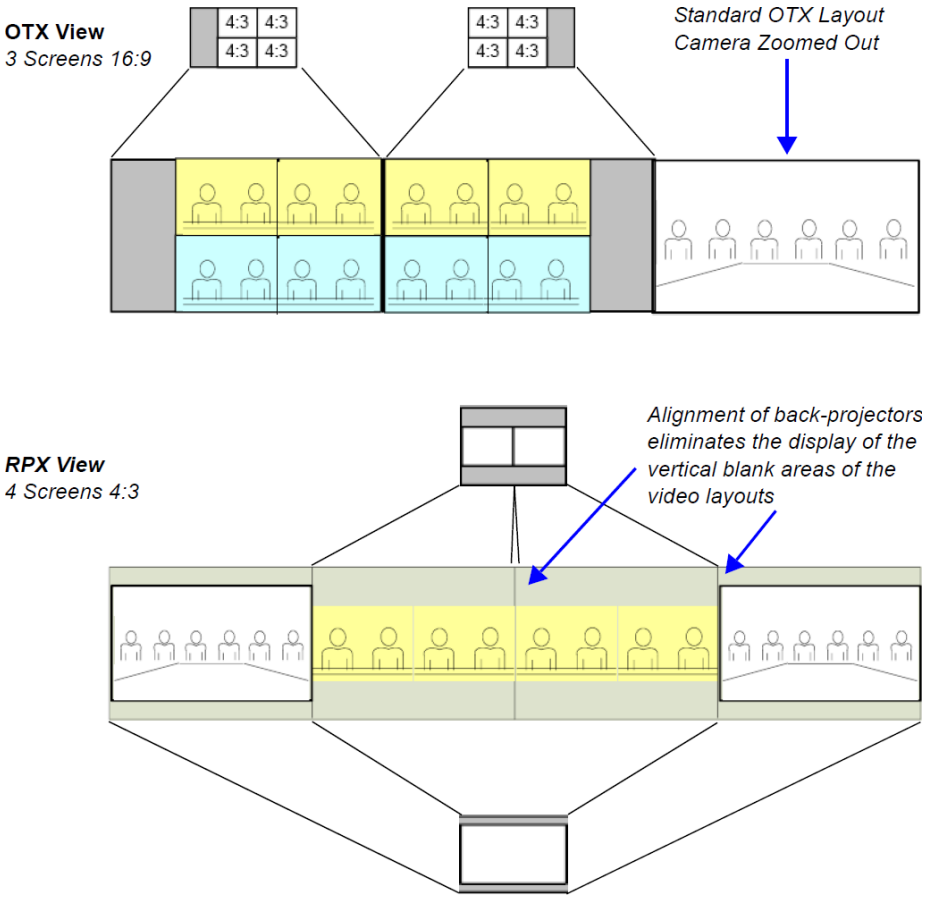
The following example illustrates the use of standard and additional Collaboration Server *Telepresence* layouts when connecting four Room Systems as follows:

- Two OTX Room Systems

- 2 active cameras
- 6 screens

- Two RPX Room Systems
  - 8 cameras
  - 8 screens

RPX and OTX Room System connected using the RealPresence Collaboration Server



Room Switch Telepresence Layouts

The Room Switch Telepresence layouts normally controlled by the MLA can be managed by the MCU to speed updating the conference layouts in large conferences with many endpoints.

Whether the MLA or the MCU controls the Room Switch Telepresence layouts is determined by the **MANAGE_TELEPRESENCE_ROOM_SWITCH_LAYOUTS** flag. This flag must be manually added before changing its value. No system reset is required.

The values are:

- **NO** (Default) - The MCU does not manage Telepresence Room Switch Layouts and they continue to be managed by the MLA.
- **YES** - The MCU manages Telepresence Room Switch Layouts.

When the MCU controls the Telepresence Room Switch layouts (**MANAGE_TELEPRESENCE_ROOM_SWITCH_LAYOUTS = YES**) the display is affected according to the Telepresence Mode Settings in the Conference Profile as follows:

- If the Telepresence Mode = **ON**
  - If no ITP endpoints are connected to the conference, the RMX Room Switch layout applies, in which case only the speaker is seen.
  - When a single participant using an ITP endpoint with either single or multiple screens connects to the conference, the participant will see black screens.
- If the Telepresence Mode = **AUTO**
  - If no ITP endpoints are connected to the conference, the RMX CP layout applies (unless the conference layout is defined).
  - When a single participant using an ITP endpoint with multiple screens connects to the conference, the participant will see black screens.
  - When a single participant using an ITP endpoint with a single screen connects to the conference, the MCU will display a self-view of the participant.
- When a TIP system with 3 screens joins a conference, the layout is updated on all screen simultaneously.
- When a *Polycom* ITP system with 2, 3, or 4 screens joins the conference, the layout is updated on all screens simultaneously.

Telepresence Display Decision Matrix

How the speaker video is displayed on the screens of the conference participants is dependent on the relationship between the number of screens the speaker endpoint contains and the number of screens of the endpoints of the other conference participants.

The following *Telepresence Display Decision Matrix"* table shows how the speaker video will be displayed on the various participant endpoints when the MCU is managing *Telepresence Room Switch* conference layouts.

Number of Screens		Speaker Endpoint			
		1	2	3	4
Participant Endpoint	1	Speaker EP: 1 Displaying EP (mirror): 1 EP1	Speaker EP: 2 1 Displaying EP (mirror): 2 1 EP1	Speaker EP: 2 1 3 Displaying EP (mirror): 2 1 3 EP1	Speaker EP: 4 2 1 3 Displaying EP (mirror): 4 2 1 3 EP1
	2	Speaker EP: 1 Displaying EP (mirror): ■ 1 EP-2 EP1	Speaker EP: 2 1 Displaying EP (mirror): 2 1 EP2 EP1	Speaker EP: 2 1 3 Displaying EP (mirror): ■ 2 1 3 EP2 EP1	Speaker EP: 4 2 1 3 Displaying EP (mirror): 4 2 1 3 EP2 EP1
	3	Speaker EP: 1 Displaying EP (mirror): ■ 1 ■ EP2 EP1 EP3	Speaker EP: 2 1 Displaying EP (mirror): 2 1 ■ EP2 EP1 EP3	Speaker EP: 2 1 3 Displaying EP (mirror): 2 1 3 EP2 EP1 EP3	Speaker EP: 4 2 1 3 Displaying EP (mirror): ■ 4 2 1 3 EP2 EP1 EP3
	4	Speaker EP: 1 Displaying EP (mirror): ■ ■ 1 ■ EP4 EP2 EP1 EP3	Speaker EP: 2 1 Displaying EP (mirror): ■ 2 1 ■ EP4 EP2 EP1 EP3	Speaker EP: 2 1 3 Displaying EP (mirror): 2 1 3 ■ EP4 EP2 EP1 EP3	Speaker EP: 4 2 1 3 Displaying EP (mirror): 4 2 1 3 EP4 EP2 EP1 EP3

For example, if the speaker's endpoints has two screens and the participant's endpoint only one, the participant's display is divided into two video layout cells with each video layout cell showing the input of one of the speaker's screens (endpoint).

If the participant endpoint has two screens, and the speaker endpoint only one, the speaker's video will be displayed on one of the participant's screens, while the second screen remains black.

Guidelines for Managing the Room Switch Telepresence Layouts by the MCU

- Only Room Switch layouts can be managed by the MCU. CP (Continuous Presence) layouts continue to be managed by the MLA.
- Only CP-AVC conferences are supported.
- Lync Clients (with CSS add-in) are supported.
- SVC endpoints are not supported.
- It is recommended that the *Speaker Change Threshold* be set to 3 seconds.
- Telepresence endpoints are named using a text name followed by a number. For example, if an OTX Telepresence room is named Oak, the three endpoint names would be Oak1, Oak2, and Oak3.
- Lecture mode is not supported in Telepresence Room Switch conferences managed by the MCU. (This is because in Lecture mode, unlike Room Switch mode, the lecturer receives the CP layout of conference participants.)
- Personal layouts are disabled. Therefore, any features that use personal layouts like Click&View can not be used to change the layout, and Click&View DTMF digits will be ignored.

- Changing the flag affects only future conferences. Conferences currently running are not affected.
- The *Send Content To Legacy Endpoints* feature is enabled by default when Telepresence mode is enabled.
- Layout attributes (no skins, no site names and no borders) should continue for Telepresence layouts managed by the RMX.

Sending Content to Legacy Endpoints in Telepresence Conferences

The Collaboration Server can be configured to manage the layouts of to *H.323/SIP/ISDN* endpoints that do not support *H.239 Content* (legacy endpoints) over the video (people) channel in Telepresence conferences when *Content* is being sent. This feature is controlled using the FORCE_LEGACY_EP_CONTENT _LAYOUT_ON_TELEPRESENCE flag. This flag must be added to change the value.

The values of the flag are:

- **NO** (Default) - The MCU does not manage the layouts while *Content* is sent. Personal layout changes, for example, by MLA, override the default MCU layout. Legacy endpoints may not display *Content* in Telepresence conferences due to layout changes.
- **YES** - The MCU manages the layouts while *Content* is sent. Personal layout changes, for example, by MLA, are ignored. The layouts for legacy endpoints are managed by the MCU.

Guidelines for Sending Content to Legacy Endpoints in Telepresence Conferences

- MLA layout change requests for legacy endpoints will be ignored until *Content* is stopped. At that point, MLA can be used again.
- *Click&View* can not be used to change the layout while *Content* is being sent.
- The Polycom Touch Control can not be used to change the layout while *Content* is being sent.

Content Display on Legacy Endpoints in Telepresence Conferences

When *Content* is sent to legacy endpoints in Telepresence conferences, their video layout automatically changes to a “Content layout” which is defined by the system flag LEGACY_EP_CONTENT_DEFAULT_LAYOUT. If MLA is managing the Telepresnce layout prior to *Content* being sent, the MCU takes over managing the layout of Legacy endpoints once *Content* is started. The video layouts of the other conference participants continue to be managed by MLA.

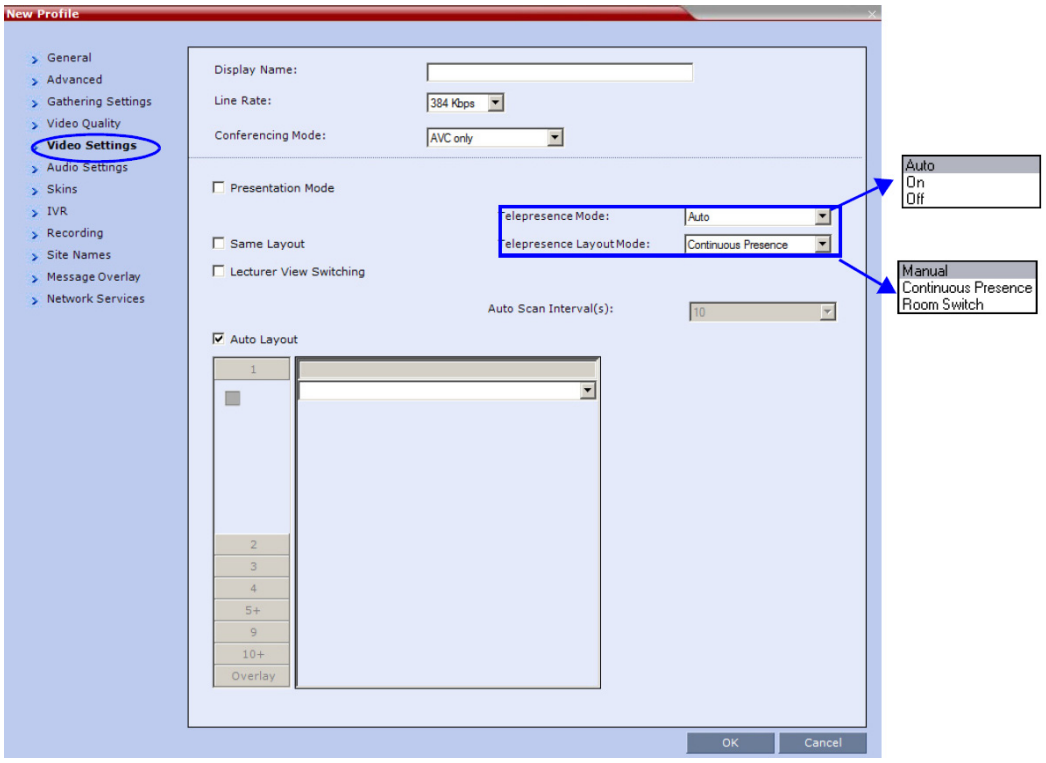
If MLA was managing the Telepresence layouts, when *Content* ends, control of the layouts for legacy endpoints goes back to the MLA after a short time.

Enabling Telepresence Mode

Telepresence Mode must be configured in a new or existing Conference Profile.

To enable Telepresence in a new or existing Conference Profile:

1. In the **RMX Management** pane, click **Conference Profiles**.
2. Click the **New Profiles** button or open an existing Conference Profile.
3. Define the various profile *G*eneral, Advanced, Gathering Settings and Video Quality parameters.
   For more information on defining Profiles, see Defining New Profiles .
4. Click the **Video Settings** tab.



5. In the **Telepresence Mode** field, select one of the following options:
   - **OFF** - When OFF is selected, normal conference video is sent by the Collaboration Server.
   - **AUTO** (Default) **-** The ITP features are dynamic. When AUTO is selected and an ITP endpoint is detected, ITP features are applied to the conference video for all participants. If all ITP endpoints disconnect from the conference, normal conference video is resumed for all remaining participants. ITP features are re-applied for all participants should an ITP endpoint re-connect to the conference.

     When *Telepresence Mode* is set to **Auto** and a one-screen *Telepresence* unit is in use, the *Collaboration Server* controls layouts instead of the *MLA.* For more information see *Polycom Multipoint Layout Application (MLA) User's Guide for Use with Polycom Telepresence Solutions*.
   - **ON** - ITP features are always applied to the conference video for all participants regardless of whether there are ITP endpoints connected or not.

6 In the **Telepresence Layout Mode** field, select the Telepresence Layout Mode to be used in the conference. This field is used by VNOC operators and Polycom Multi Layout Applications to retrieve Telepresence Layout Mode information from the Collaboration Server.

   The following modes can be selected (as required by the VNOC and Polycom Multi Layout Applications):

   - **Manual**
   - **Continuous presence** - Room Continuous Presence **(**Default)
   - **Room Switch** - Voice Activated Room Switching

7 Select the required video layout.

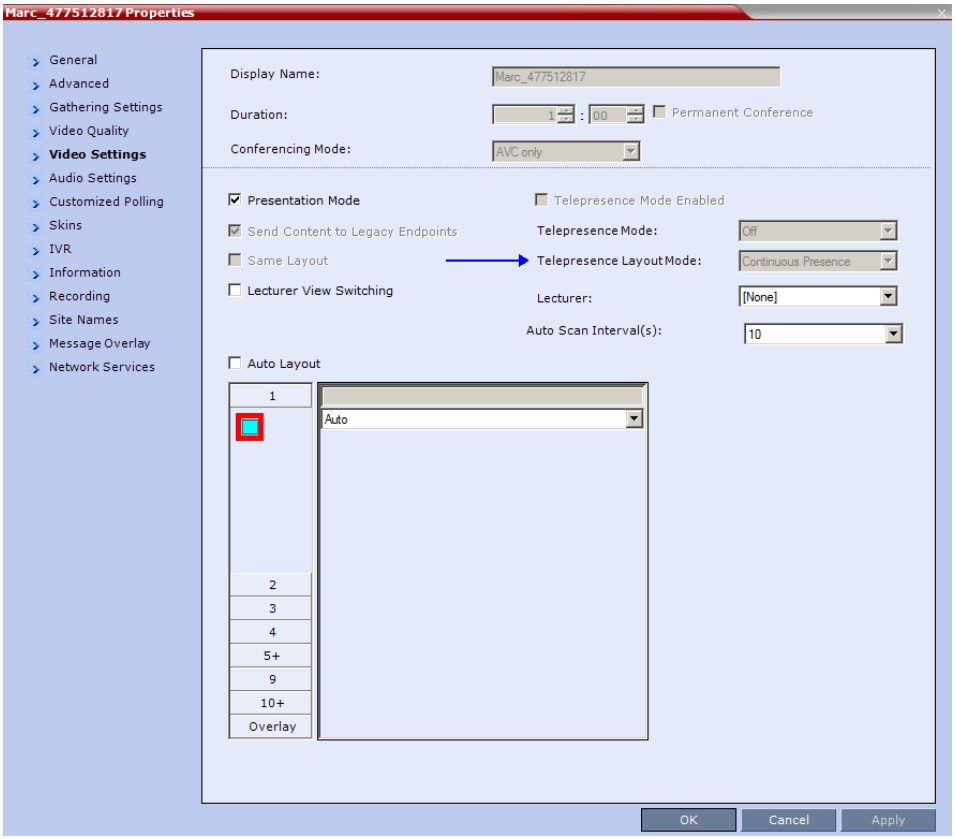
8 Click OK.

When Telepresence Mode is enabled, the **Skins** options are disabled as the system uses a black background and the frames and speaker indication are disabled.

Monitoring Telepresence Mode

Monitoring Ongoing Conferences

An additional status indicator, *Telepresence Mode Enabled*, is displayed in the *Conference Properties - Video Settings* tab when monitoring ongoing conferences.

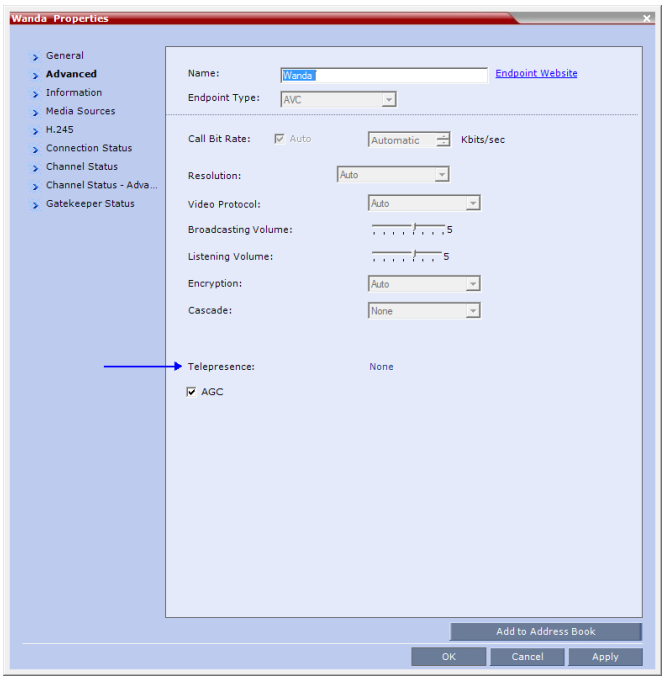


The *Telepresence Mode Enabled, Telepresence Mode* and *Telepresence Layout Mode* fields are only enabled if the Collaboration Server has a *Telepresence* license installed.

If Telepresence Mode is enabled, a check mark is displayed in the check box. This option is grayed as this is a status indicator and cannot be used to enable or disable Telepresence Mode.

Monitoring Participant Properties

An additional status indicator, *Telepresence*, is displayed in the **Participant Properties - Advanced** tab when monitoring conference participants.



The Telepresence mode of the participant is indicated:

- *RPX* - the participant's endpoint is transmitting 4:3 video format.
- *OTX* - the participant's endpoint is transmitting 16:9 video format.
- *None*.

Creating Multiple Cascade Links Between Telepresence Conferences

You can create multiple Cascading links between Collaboration Servers hosting conferences that include *Immersive Telepresence Rooms* (*ITP*) such as Polycom's OTX and RPX Room Systems.

Guidelines for Creating Multiple Cascading Links between Conferences

- *Basic Cascading* topology is used. For more information see the *<Italic>RealPresence Collaboration Server 800s Administrator's Guide, Basic Cascading Using IP Cascaded Link*.
- Multiple Cascade Links between conferences are implemented by creating a Link Participant which consists of a main link and sub-links which are automatically generated and sequentially numbered. For more information see Creating a Link Participant, Creating a Link Participant.
- All cascaded links must use H.323 protocol.
- Multiple Cascade Links are supported in CP conferencing mode.
- The number of cascading links is defined manually according to the maximum number of Room System cameras in the cascaded conference.
- When the active speaker is in an Immersive Telepresence Room, Multiple Cascade Links are used, one link for each of the Room System's cameras.
  - An RPX 4xx Room System requires 4 Cascaded Links to carry the video of its 4 cameras.
  - An RPX 2xx Room System requires 2 Cascaded Links to carry the video of its 2 cameras.
  - An OTX 3xx Room System requires 3 Cascaded Links to carry the video of its 3 cameras. The OTX Room System must be configured as Room Switch in order to send multiple streams. When configured in CP Mode, its cameras zoom out and all 3 screens are sent as one stream.
- The number of links is defined when creating the Link Participant. Each conference in the cascade must have a Link Participant with the same number of Multiple Cascade Links defined. Calls from Link Participants not defined with the same number of links are rejected. Number of cascading links is not identical for all conferences is listed as the Call Disconnection Cause. For more information see Creating a Link ParticipantCreating a Link Participant and Monitoring Multiple Cascade LinksMonitoring Multiple Cascade Links.
- Although it is possible to disconnect and reconnect specific Multiple Cascade Links using the RealPresence Collaboration Server Web Client / RealPresence Collaboration Server Manager it not advisable to do so.
  - If the main link is disconnected all sub-links are disconnected and deleted. Reconnecting the main link reconnects all sub-links.
  - If a sub-link is disconnected it remains disconnected until it is manually reconnected.
  - The number of Multiple Cascade Links cannot be modified while any of the links are in a disconnected state. All previous links must be deleted before modification is possible.

    For more information see Monitoring Multiple Cascade LinksMonitoring Multiple Cascade Links.
- A Link Participant can be dragged from the address book into a conference.
  - If it is the first Link Participant in the conference, the number of Multiple Cascade Links defined for the participant are created and connected.

  - If it is not the first Link Participant in the conference, the number of Multiple Cascade Links defined for the participant is ignored.
- If there are insufficient resources to connect all Multiple Cascade Links in either of the RMXs, none of the links are connected and resources deficiency -0 is listed as the Call Disconnection Cause. For more information see Monitoring Multiple Cascade LinksMonitoring Multiple Cascade Links.
- Multiple Cascade Links that are not used by MLA are inactive but continue to consume resources.
- All RMXs participating in the cascade must have the same Telepresence Mode definitions, either all defined as CP or all defined as Room Switch.
- When Multiple Cascade Links are defined in the Conference Profile, the Layout Type field of the Link Participant's Participant Properties - Media Sources dialog box is set to Conference and cannot be modified.
- TIP Telepresence Rooms (CTS) are supported without *Content.* For more information see the Collaboration With Cisco's Telepresence Interoperability Protocol (TIP).

Enabling and Using Multiple Cascade Links

The settings required to enable Multiple Cascade Links on the RMX are minimal and are described in Creating a Link ParticipantCreating a Link Participant.

Most of the layout configuration is performed using *Polycom's Multipoint Layout Application (MLA)*.

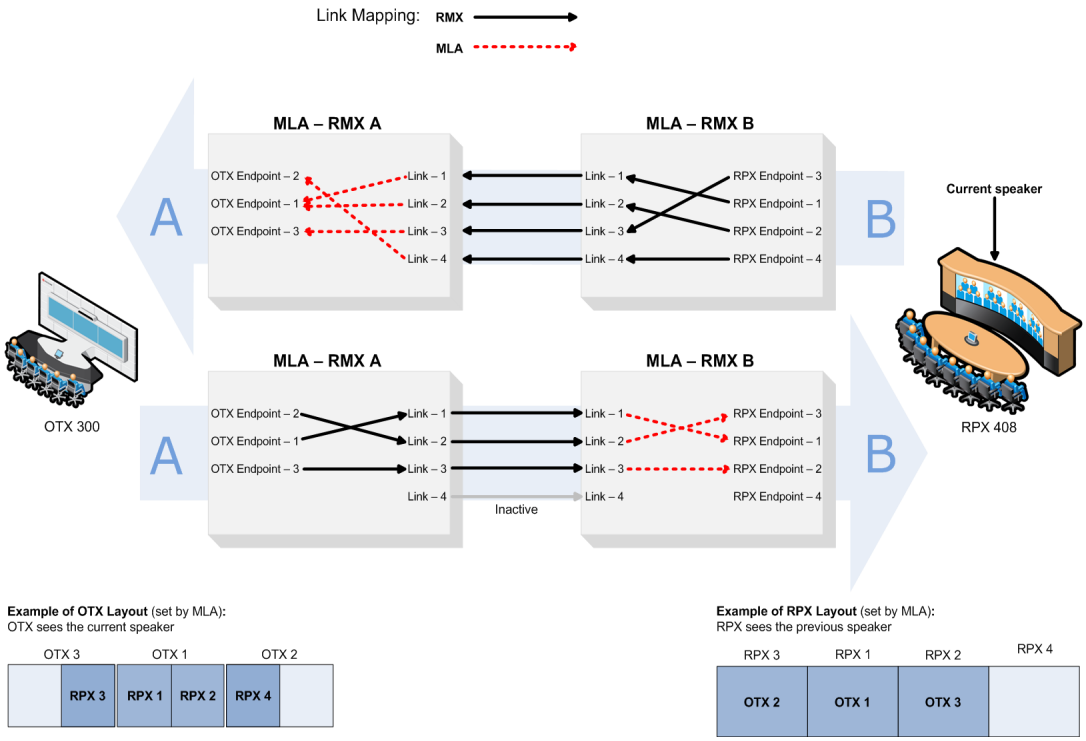
The figures, RMX Telepresence Layout Mode - Room Switch and RMX Telepresence Layout Mode - Continuous Presence show example layouts and media flows when MLA is configured for a cascading conference between two RMXs.

In the figure RMX Telepresence Layout Mode - Room Switch:

- The OTX Room System connects to RMX A.
- The RPX Room System connects to RMX B.
- This layout requires that the *Telepresence Layout Mode* to be set to **Room Switch** in the *Conference Profiles* of the *Cascading Conferences* in each RMX.
- The current speaker is a participant in the RPX ITP Room.

- Directional media flows, A ⇆ B, are shown separately for readability purposes.

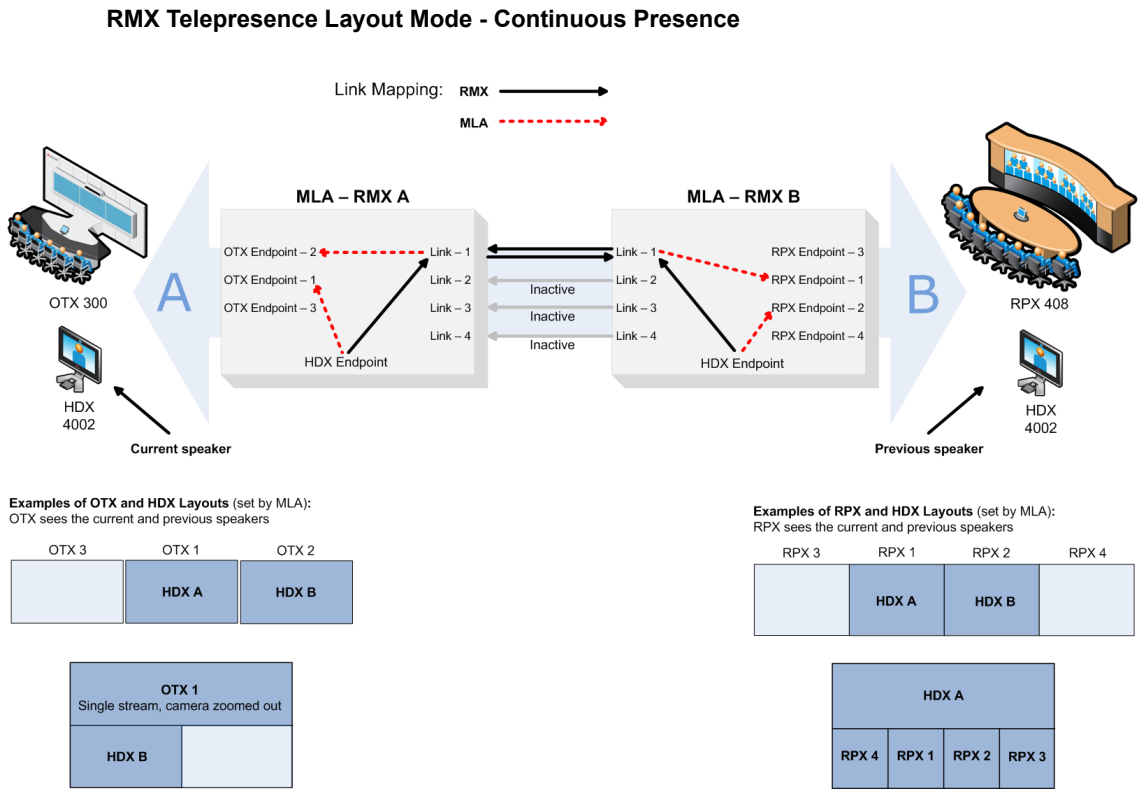
RMX Telepresence Layout Mode - Room Switch



In this figure:

- An HDX endpoint and an OTX Room System connects to RMX A.
- An HDX endpoint and an RPX Room System connects to RMX B.
- This layout requires that the **Telepresence Layout Mode** to be set to **Continuous Presence** in the Conference Profiles of the Cascading Conferences in each RMX.

- The current speaker is the HDX endpoint connected to RMX A.



For more information see:

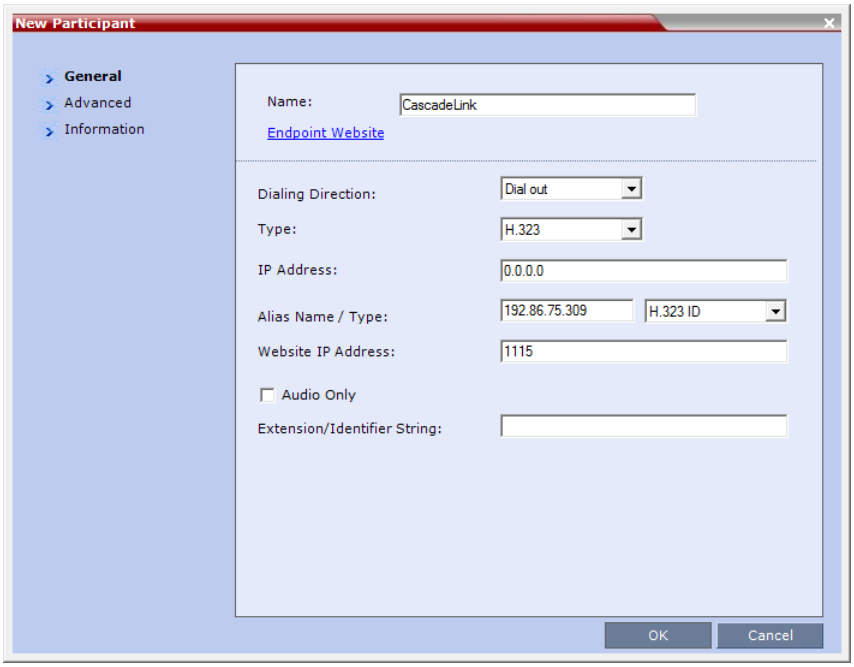
- Telepresence Layout Mode .
- Polycom® Multipoint Layout Application (MLA) User's Guide for Use with Polycom Telepresence Solutions
- Polycom® Immersive Telepresence (ITP) Deployment Guide

Creating a Link Participant

Link Participant in the Dial Out RMX

The Link Participant is defined in the **New Participant** dialog box.

In the **General** tab:



- **Dialing Direction** must be selected as **Dial out.**
- ***Type*** must be selected as **H.323**.

  For more information see the Creating a Cascade Enabled Dial-out/Dial-in Participant Link .

In the **Advanced** tab:

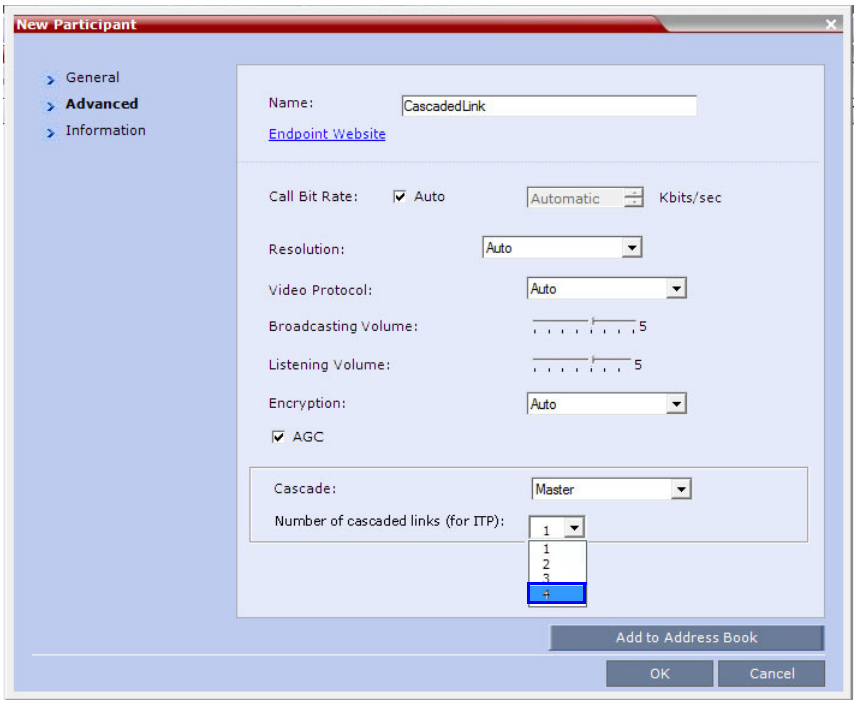
(This field is only enabled if the RMX system is licensed for *Telepresence Mode*.)

- In the **Cascade** drop-down menu, select either **Master** or **Slave**.

- In the **Number of cascaded links (for ITP)** drop-down menu, select the maximum number of *Multiple Cascade Links* required according to the number of Room System endpoints in the cascaded conference.

  This field enables the administrator to select the maximum number of *Multiple Cascade Links* required according to the number of Room System endpoints in the cascaded conference.

  For example if an *RPX 4xx* is included, the number of links required is 4.



The RMX automatically adds a number suffix to the name of the *Link Participant*, for example if the *Participant Link Name* is *CascadeLink* and the *Number of cascaded links (for ITP)* field is set to 4, the following *Multiple Cascade Links* are created:

- CascadeLink-1
- CascadeLink-2
- CascadeLink-3
- CascadeLink-4

Participant Link in the Dial In RMX

The call from Participant Link defined in the Dial-out RMX is identified by the Dial-in RMX as having been initiated by a Participant Link.

Suffixes are appended the Multiple Cascade Links according to the **Number of cascaded links** (for ITP) field depending on whether the Dial -In Participant Link is defined or un-defined:

Participant Link is un-defined

The Multiple Cascade Link names are automatically assigned by the RMX.
For example on a RMX 1500 the names of the links are:

- POLYCOM RMX 1500-1
- POLYCOM RMX 1500-2
- POLYCOM RMX 1500-3, etc.

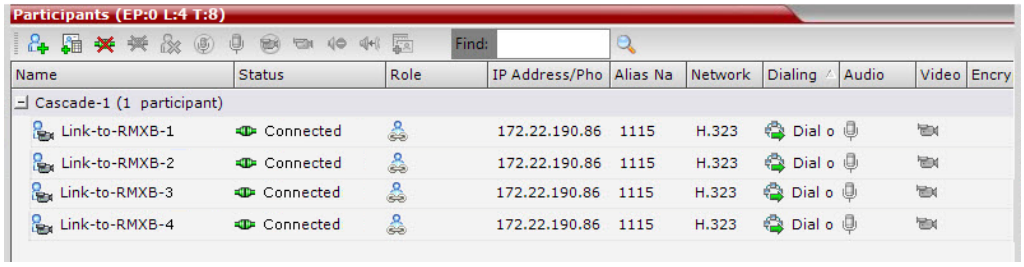
Participant Link is a defined

The Multiple Cascade Link names are assigned according to the name of the defined participant that is to function as the cascade link and the Number of cascaded links (for ITP) information sent by the calling Dial-Out Participant Link.
For example if the defined participant that is to function as the cascade link is named Cascade_Link_From_B the names of the links are:

- Cascade_Link_From_B-1
- Cascade_Link_From_B-2
- Cascade_Link_From_B-3, etc.

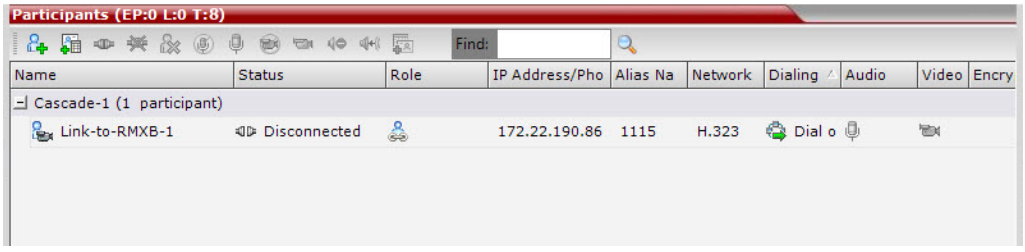
Monitoring Multiple Cascade Links

Multiple Cascade Links connections can be monitored in the *Participants* list of the *RMX Web Client* / *RMX Manager* main screen:

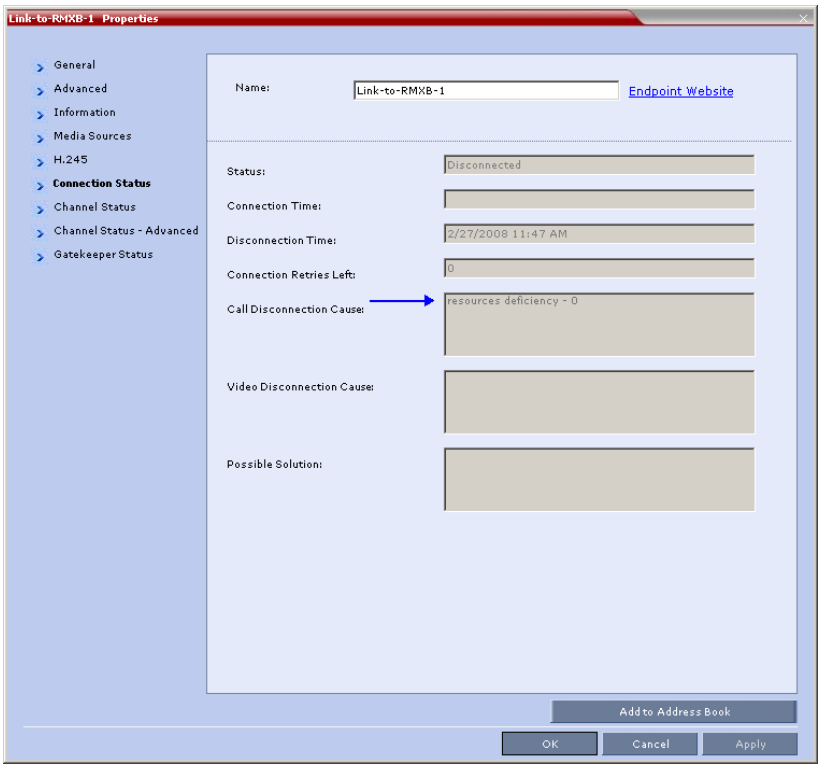


Disconnection Causes

- If there are insufficient resources to connect all the required links:
  - None of the links are connected.
  - The first link is listed as **Disconnected** in the *Participants* list of the RMX Web Client / RMX Manager main screen.



- Resource deficiency is listed as the **Call Disconnection Cause** in the **Participant Properties - Connection Status** dialog box.



- If a calling Link Participant is not defined with same number of links as all the other Link Participants in the cascaded conferences:
  - The call is rejected.
  - The **Call Disconnection Cause** is: Number of cascading links is not identical for all conferences.

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 Polycom® RealPresence® Collaboration Server 800s and Polycom® 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:

- Video Preview (AVC Participants Only)
- Auto Scan and Customized Polling in Video Layout (CP Conferences Only)
- Packet Loss Compensation (LPR and DBA) AVC CP Conferences
- Network Quality Indication (AVC Endpoints)
- Lecture Mode (AVC CP Only)
- Audio Algorithm Support

Video Preview (AVC Participants Only)

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 supported in CP Conferencing Mode only.
- 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.
- 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 Polycom Desktop application (such as CMAD) window cannot be open and running simultaneously on the same PC as both require the same DirectDraw resource.

Workstation Requirements to Display Video Preview

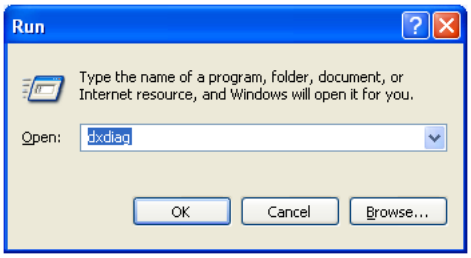
To be able to display the video preview window, the following minimum requirements must be met:

- 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

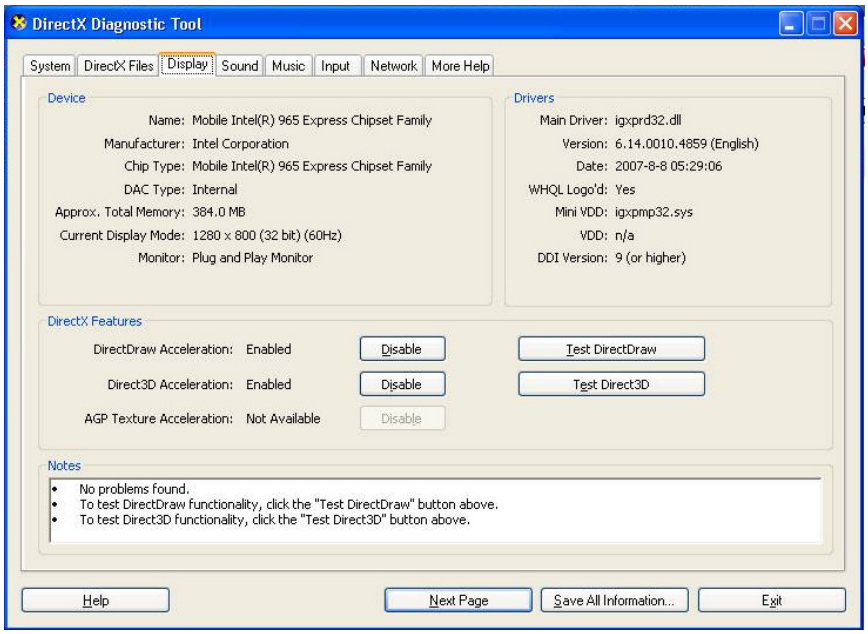
To ensure that your workstation can display the video preview window:

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**.



   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**.



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.

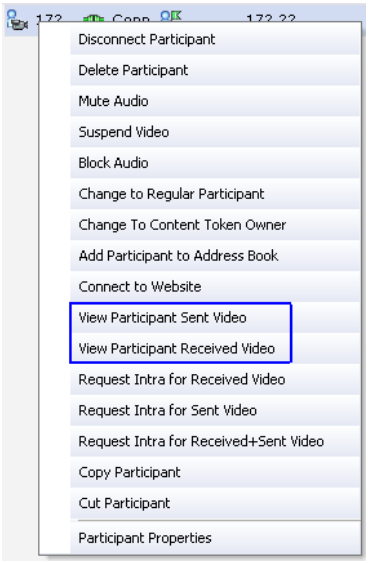
Previewing the Participant Video

You can preview the video sent from the participant to the conference (MCU) and the video sent from the conference to the participant by selecting the appropriate option from the Participant's pop-up menu.

To preview the participant video:

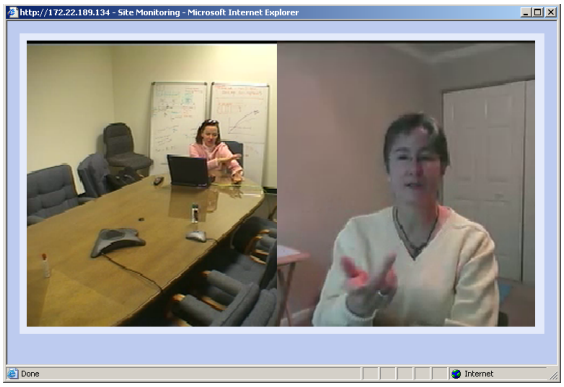
1 List the conference participants in the **Participants** pane.

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



- **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.



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 Conferences Only)

Auto Scan enables you 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 for Using Auto Scan and Customized Polling

- 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 the Auto Scan and Customized Polling (CP Only Conferences)

Auto Scan and Customized Polling are enabled during the ongoing conference, in the **Conference Properties - Video Settings** dialog box.

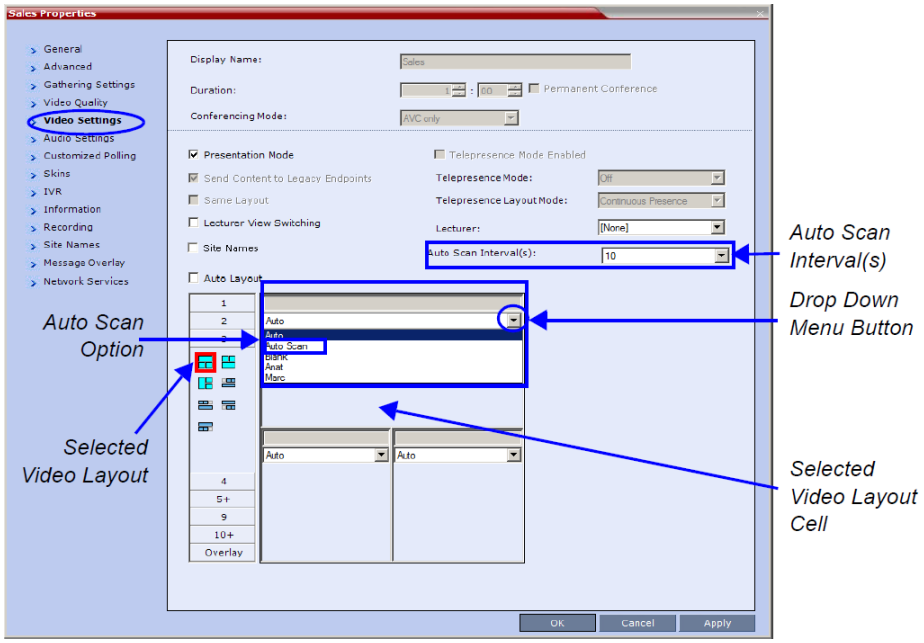
Enabling the Auto Scan

You enable the Auto Scan feature by selecting it in the Video Layout cell.

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** dialog box is displayed.



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)** list the scanning interval in seconds.

6 Click the **Apply** button to confirm and keep the dialog box open, or Click **OK**.

Customized Polling

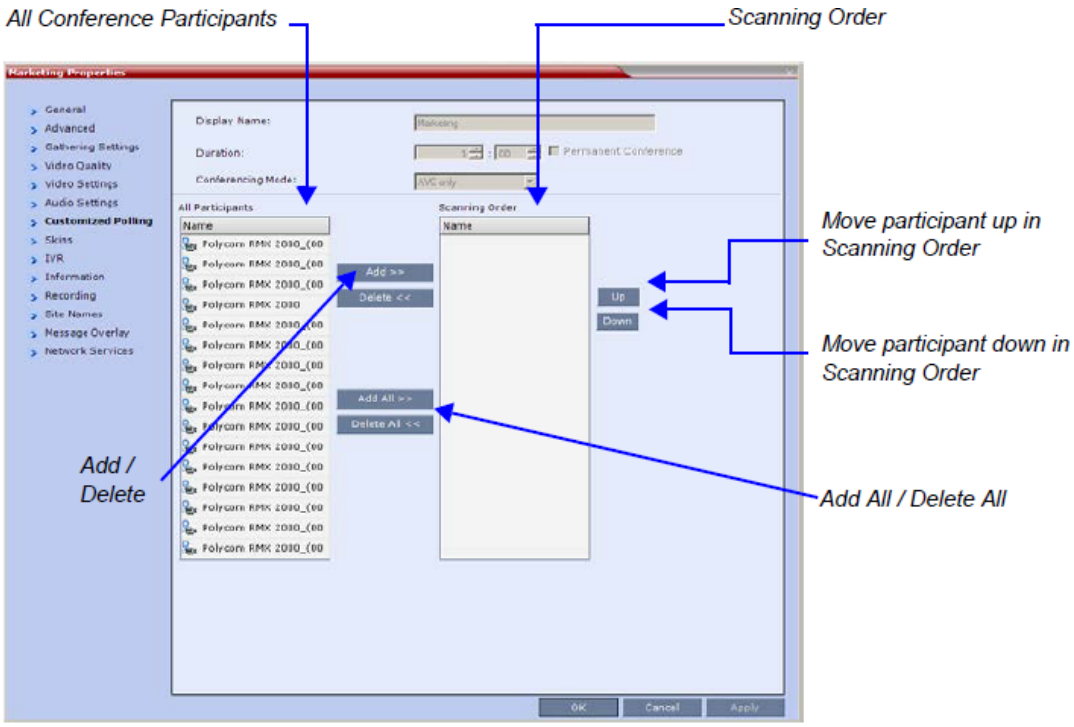
The order in which the Auto Scanned participants are displayed in the Auto Scan enabled cell of the video layout can be customized.

To define the scanning order in the Customized Polling tab:

1 Open the **Conference Properties** dialog box.

2 Click the **Customized Polling** tab.

The Customized Polling dialog box 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:

Customized Polling Dialog Box Command 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*.

3 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.

4 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.

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

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

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

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

9 Click the **Apply** button to confirm and keep the dialog box open, or click **OK**.

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 IP (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

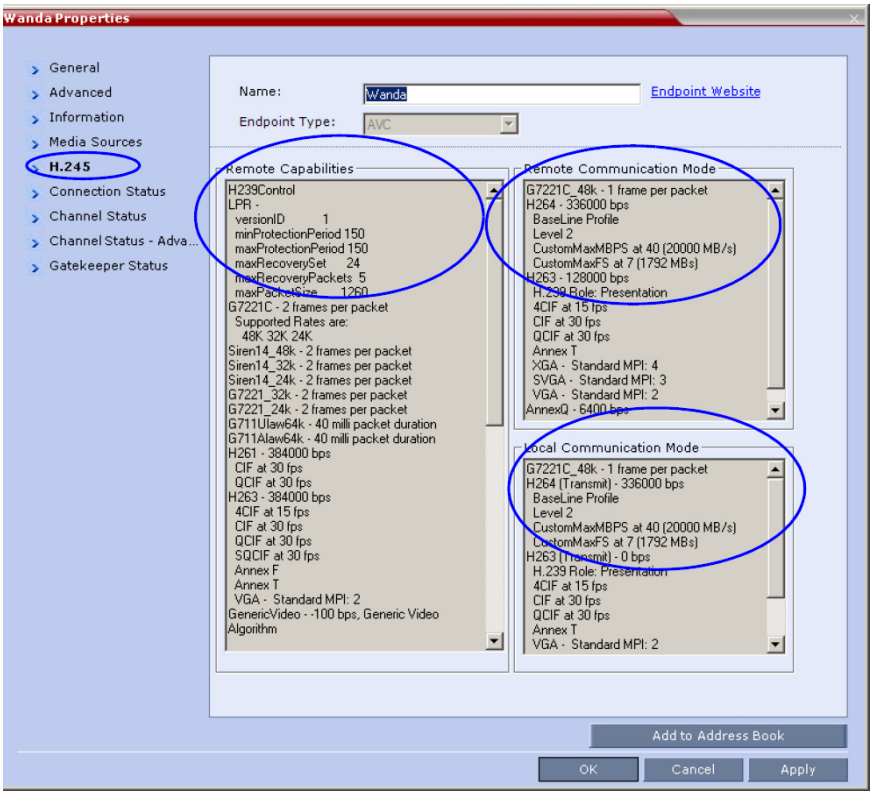
LPR is enabled or disabled in the *Conference Profile* dialog box.

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

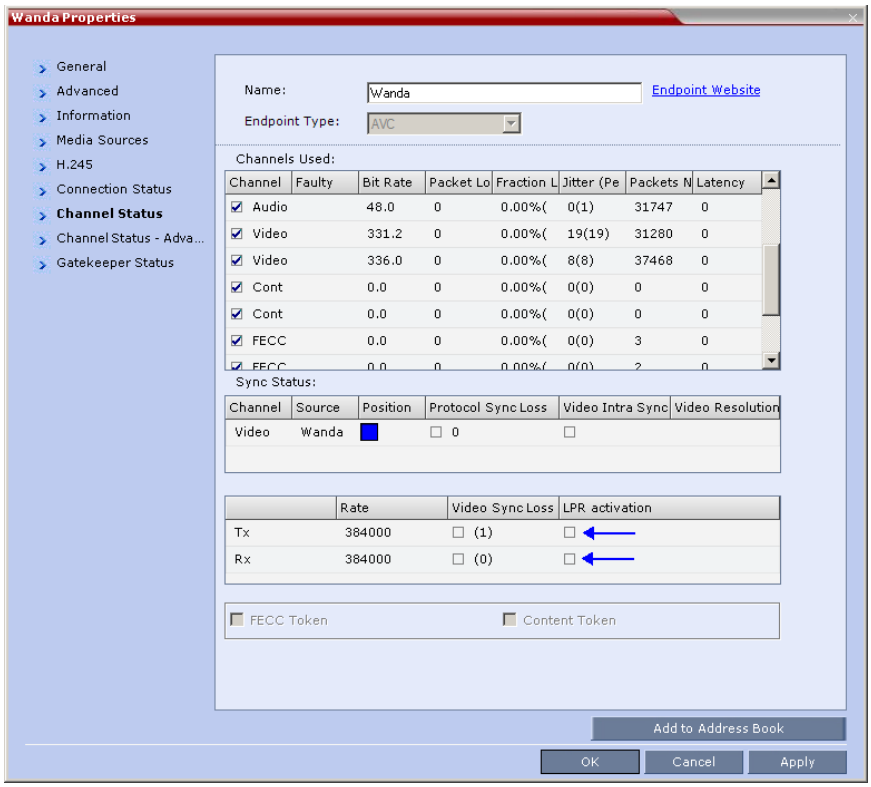
  For more information, see Defining New Profiles .

Monitoring Lost Packet Recovery

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

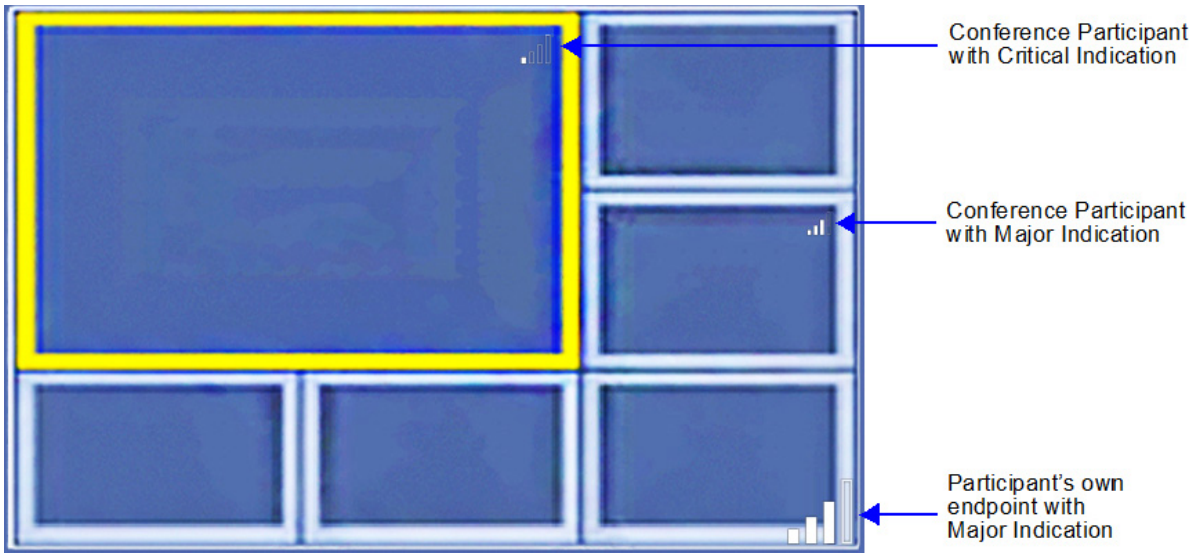


In the **Participant Properties – Channel Status** tab, check box indicators show LPR activation in the local and remote (transmit and receive) channels.



Network Quality Indication (AVC Endpoints)

If network quality issues occur, the *Network Quality* icon provide information to participants about their own network quality and that of other participants displayed in the cells of the conference Video Layout.



Network Quality Levels

Network quality is determined by the percentage of packet loss according to the following default threshold values:

- Packet loss less than **1%** is considered *Normal*
- Packet loss in the range of **1% - 5%** is considered *Major*
- Packet loss above **5%** is considered *Critical*.

Major *Critical*

When network quality improves from Critical to Major remaining stable for 5 seconds, the Network Quality Indicator is changed accordingly and when network quality improves from Major to Normal, remaining stable for 5 seconds, the Network Quality Indicator is no longer displayed.

Indication Threshold Values

The default Major and Critical indication threshold values can be modified by manually adding the following System Flags and modifying their values as required.

Network Quality Icon - Indication Threshold Flags

Flag	Description
NETWORK_IND_MAJOR_PERCENTAGE	The percentage degradation due to packet loss required to change the indicator from *Normal* to *Major*. Default: 1
NETWORK_IND_CRITICAL_PERCENTAGE	The percentage degradation due to packet loss required to change the indicator from *Major* to *Critical*. Default: 5

For more information see the *Polycom® RealPresence Collaboration Server 800s/Virtual Edition Administrator's Guide*, Manually Adding and Deleting System Flags.

Guidelines for Displaying the Network Quality icons

Network Quality icons are not supported in SVC Conferencing Mode.

Network Quality icons are displayed for:

- The video channel only in AVC Conferencing Mode.
  Content, audio and FECC channel quality issues are not indicated.
- The participant's own endpoint:
  - Network Quality icons are displayed by default and can be disabled
  - For media transmitted to and received from the Collaboration Server (Video in / Video out).
- Participants displayed in the cells of the conference video layout:
  - Network Quality icons are not displayed by default and can be enabled
  - The media transmitted from the endpoint to the Collaboration Server (Video in).

Customizing Network Quality Icon Display

Display of the Network Quality icon can be customized for the participant's own endpoint or for the Participants displayed in the cells of the conference Video Layout.

The display of Network Quality icon (showing or hiding the icon) and the position of the icon in the video layout cell can be customized by manually adding the following System Flags and modifying their values as required.

Network Quality Icon - Display Customization Flags

Flag	Description
DISABLE_SELF_NETWORK_IND	Disable the display of the *Network Quality* icon of the participant's own endpoint. Default: NO Range: YES / NO
DISABLE_CELLS_NETWORK_IND	Disable the display of Network Quality icons displayed in the cells of the conference Video Layout. Default: YES Range: YES / NO
SELF_IND_LOCATION	Change the location of the display of the Network Quality icon of the participant's own endpoint. Default: BOTTOM_RIGHT Range: • TOP_ LEFT • TOP • TOP_RIGHT • BOTTOM_ LEFT • BOTTOM • BOTTOM_RIGHT
CELL_IND_LOCATION	Change the location of the display of Network Quality icons displayed in the cells of the conference Video Layout. Default: TOP_RIGHT Range: • BOTTOM_LEFT • BOTTOM_RIGHT • TOP_LEFT • TOP_RIGHT

For more information see the *Polycom® RealPresence Collaboration Server 800s/Virtual Edition Administrator's Guide*, Manually Adding and Deleting System Flags.

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.

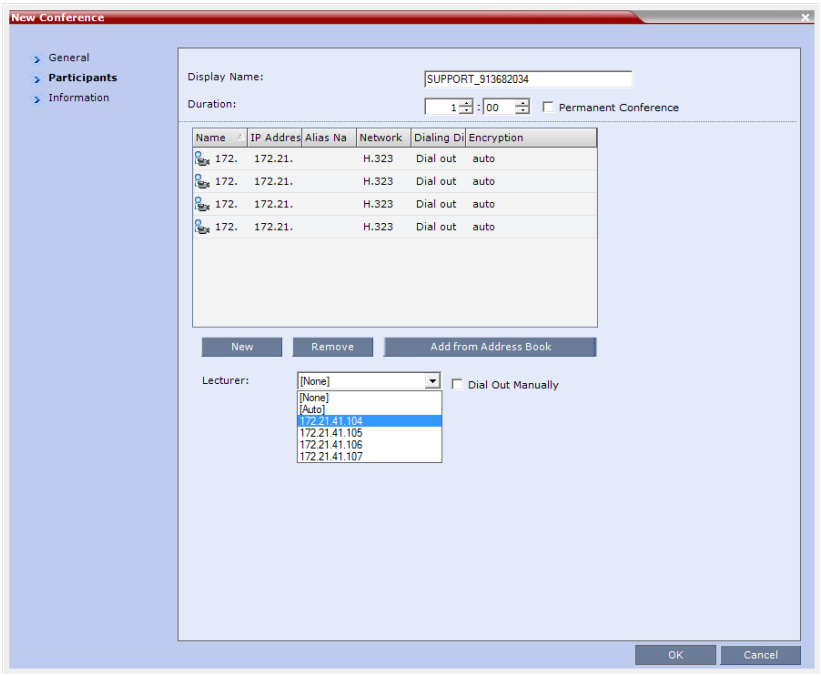
Selecting the Conference Lecturer

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:



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.

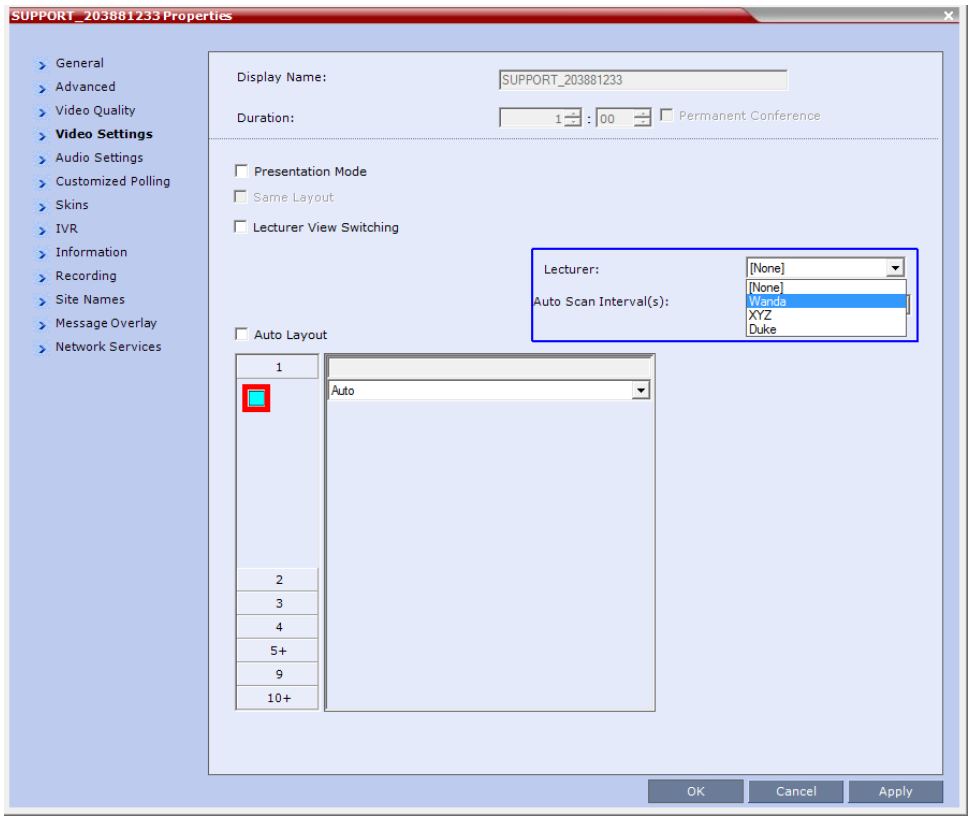
Automatic selection of the lecturer:

➢ 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.

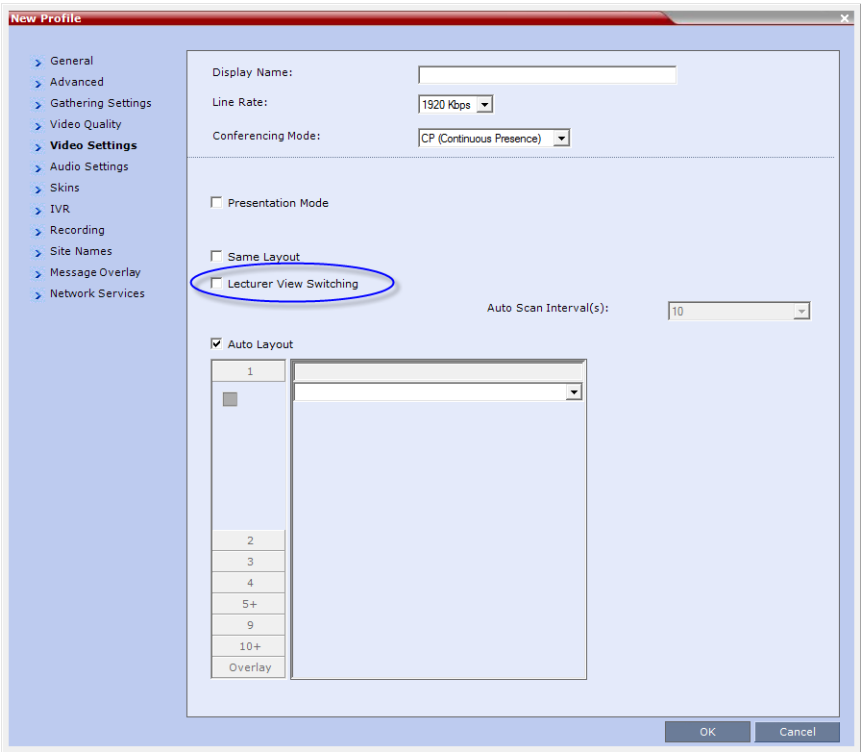


Defined dial out participants and dial in participants are considered to be two separate participants even if they have the same IP address/number. Therefore, if a defined dial-out participant is added to the conference and the same participant then dials in (before the system dialed out to that participant) 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.



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 .

- 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.

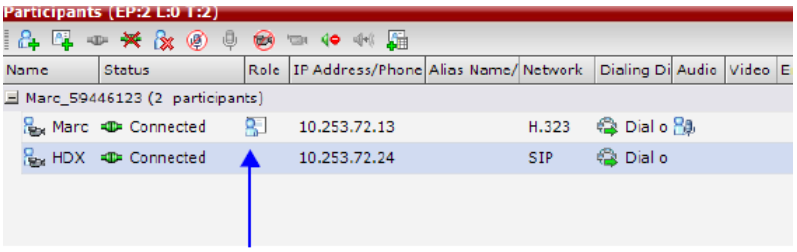
When one of the participants is talking, the automatic switching is suspended, showing the current speaker, and it is resumed when the lecturer resumes talking.

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.



Participant designated as the Lecturer

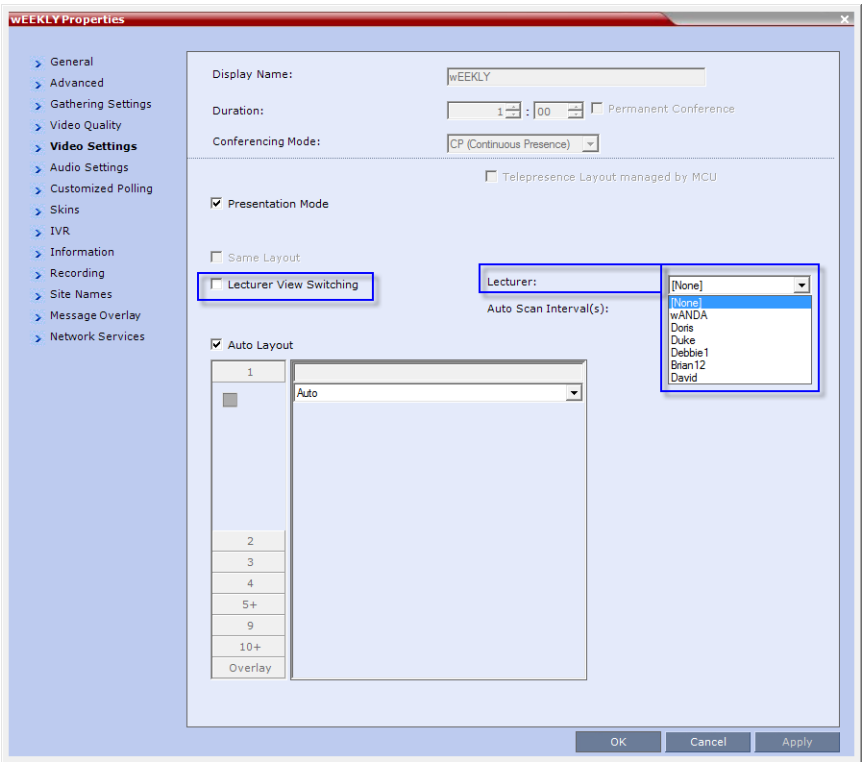
To control the Lecture Mode during an Ongoing Conference:

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.

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



- 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)

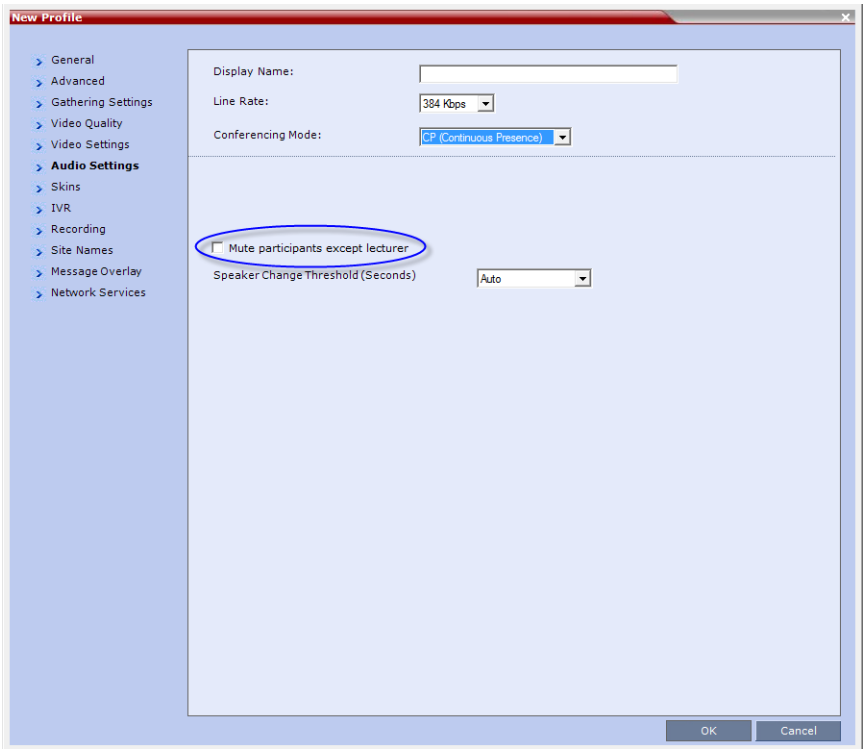
When the *Mute Participants Except Lecturer* option in the Conference Profile is enabled, the audio of all participants in the conference except for the lecturer can be automatically muted upon connection to the conference. This prevents other conference participants from accidentally interrupting the 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 for Muting all the Participants Except the Lecturer

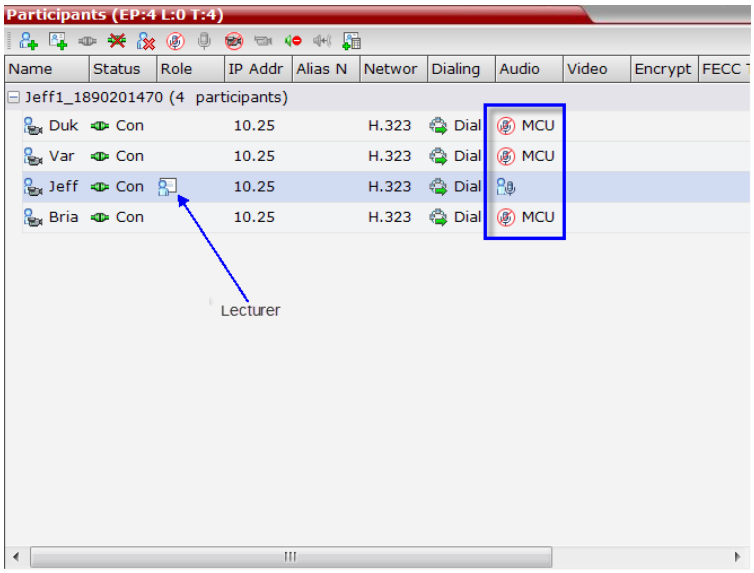
- Both administrators and operators (users) are allowed to set the Mute Participants Except Lecturer option.
- When the Mute Participants Except Lecturer option is enabled, the mute indicator on the participant endpoints are not visible because the mute participants was initiated by the MCU. Therefore, it is recommended to inform the participants that their audio is muted by using the 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.



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.



Audio Algorithm Support

The Collaboration Server supports the following audio algorithms in AVC conferences: G.711, G.722, G.722.1, G.722.1C, G.729A, 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 is 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.

Audio Algorithm Support Guidelines

- Siren 22 is supported in both mono and stereo.
- Stereo is supported in H.323 calls only.
- Siren 22 is supported by Polycom HDX and Group series endpoints, version 2.0 and later.
- 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 conferences.
  - With *Polycom CMAD* and *HDX 3.0.1 and later* and Group series 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, and SirenLPR mono audio algorithms are supported at the following bit rates

Siren22, and SirenLPR Mono vs Bitrate

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

Siren22, and SirenLPR Mono vs Bitrate

Audio Algorithm	Minimum Bitrate (kbps)
Siren22_32k	
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, and SirenLPR audio algorithms are supported at the following bit rates.

Siren22Stereo, and SirenLPR vs Bitrate

Audio Algorithm	Minimum Bitrate (kbps)
Siren22Stereo_128k	1024
SirenLPRStereo_128k	
G.719Stereo_128k	

Siren22Stereo, and SirenLPR vs Bitrate

Audio Algorithm	Minimum Bitrate (kbps)
Siren22Stereo_96k	512
SirenLPRStereo_96k	
G.719Stero_96k	
Siren14Stero_96k	
SirenLPRStereo_64k	384
G.719Stereo_64k	
Siren22Stereo_64k	
Siren14Stereo_64k	

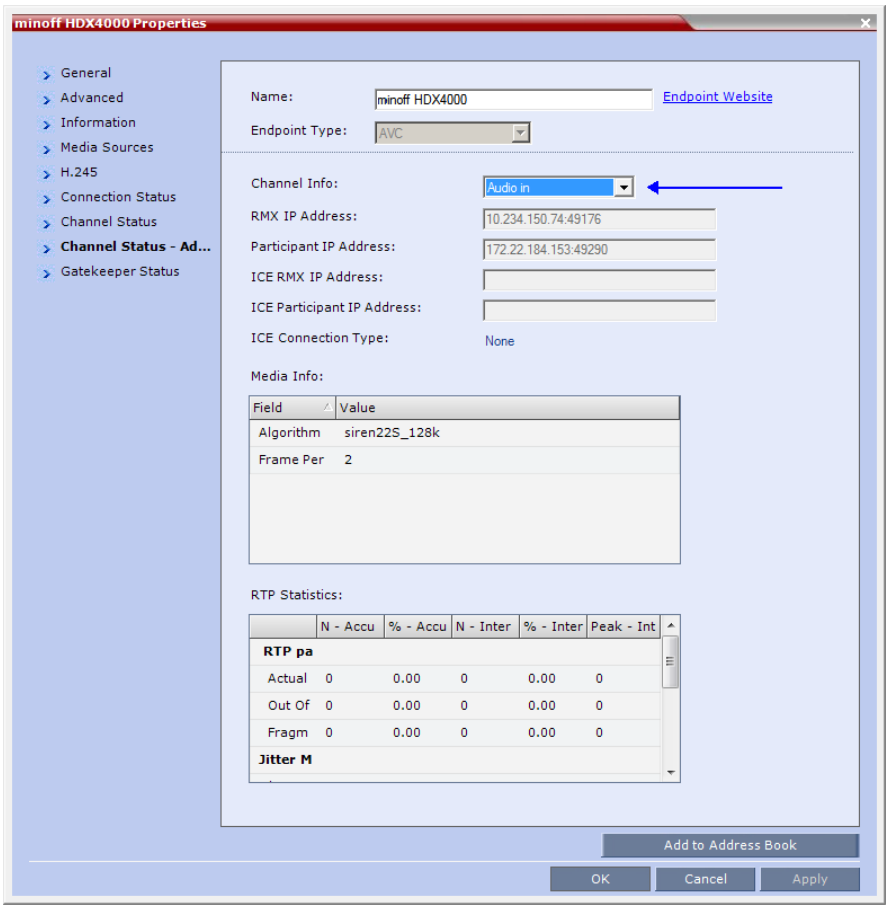
Monitoring Participant Audio Properties

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.



4 Click the **OK** button.

Automatic Muting of Noisy Endpoints

The automatic muting of noisy AVC-enabled endpoints can be used according to the following guidelines:

- The Auto mute noisy endpoints check box in the Profile Properties - Audio Settings dialog box is enabled only when the ENABLE_SELECTIVE_MIXING flag is set to YES (default).
- It affects only AVC-based and audio only endpoints (non-SAC endpoints)
- It does not affect SVC-based endpoints
- It is supported in CP conferences and in Mixed CP and SVC conferences.
- In a mixed CP and SVC conferences, only the AVC-based endpoints can be automatically muted. If the noisy endpoint is SVC-based, its audio channel will not be sent to the AVC-based endpoints, but it will be sent to the other SVC-based endpoints.
- MCU reset is not required when changing the ENABLE_SELECTIVE_MIXING flag setting.
- When upgrading from a version prior to 8.1, the Auto mute noisy endpoints option is not automatically enabled in the existing Profiles and it has to be manually enabled, if required.

- In new Profiles that are created after the upgrade, the Auto mute noisy endpoints option is automatically enabled.

If your conferencing environment includes the Polycom DMA, the conferences that are started from the DMA will not include the Auto mute noisy endpoints parameter as it is not part of the DMA Profiles. In such a case, when the parameter setting is unknown, the system will enable or disable the automatic muting of noisy endpoints according to the flag setting - if the flag is set to YES, it will be enabled in the conference.

The following table summarizes the state (enabled or disabled) of the Automatic muting of noisy endpoints feature depending on the ENABLE_SELECTIVE_MIXING flag setting and the Auto mute noisy endpoints setting in the Profile Properties - Audio Settings:

ENABLE_SELECTIVE_MIXING flag Setting	Auto mute noisy endpoints setting	Automatic muting of noisy endpoints State
YES	Yes (check box selected)	Enabled
YES	No (check box cleared)	Disabled
YES	Unknown (for example, the conference is started from the DMA)	Enabled
NO	Yes (check box selected)	Disabled
NO	No (check box cleared)	Disabled
NO	Unknown (for example, the conference is started from the DMA)	Disabled

Enabling or Disabling the Automatic Muting of Noisy Endpoints

The automatic muting of noisy endpoints can be enabled or disabled at the conference level (in the Conference Profile) or at the system level, by changing the ENABLE_SELECTIVE_MIXING flag setting.

In new MCU installations, the automatic muting of noisy endpoints is automatically enabled on the MCU as the ENABLE_SELECTIVE_MIXING flag is set to YES and the Auto mute noisy endpoints check box in the Profile Properties - Audio Settings tab is selected.

You can disable the automatic muting of noisy endpoints by either setting the system flag to NO or clearing the Auto mute noisy endpoints check box in the Profile Properties - Audio Settings tab.

If required, it is recommended to disabled the automatic muting of noisy endpoints at the conference level, in the conference Profile without changing the flag settings.

In existing MCU sites, following the software upgrade the automatic muting of noisy endpoints is disabled at the conference level in the existing conference Profile and has to be manually enabled in these profiles. This option is automatically enabled when creating a new Profile.

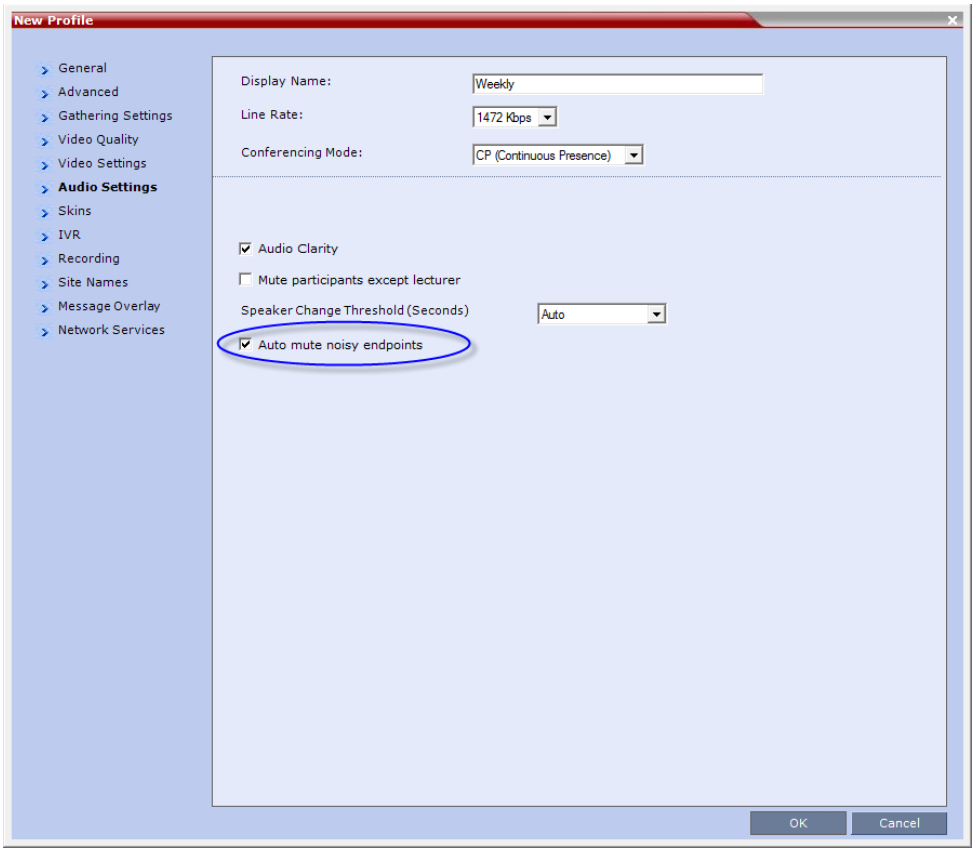
Enabling or Disabling the Automatic Muting of Noisy Endpoints at the Conference Level

If the ENABLE_SELECTIVE_MIXING flag is 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.

If the ENABLE_SELECTIVE_MIXING flag is 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.

To disable/enable the automatic muting of noisy endpoints in the Conference Profile:

1 In a new or existing Conference Profile, click the Audio Settings tab.



- In new Profiles, the Auto mute noisy endpoints check box is selected by default.
- In existing profiles (after software upgrade from a version prior to 8.1), the Auto mute noisy endpoints check box is cleared.

2 To enable the automatic muting of noisy endpoints, click the **Auto mute noisy endpoints** check box.

3 Click **OK**.

Enabling or Disabling the Automatic Muting of Noisy Endpoints at the MCU Level

You can disable the automatic muting of noisy endpoints at the MCU level by changing the ENABLE_SELECTIVE_MIXING flag setting to NO.

In such a case, the automatic muting of noisy endpoints at the conference level (in the Conference Profile - Audio Settings dialog box) is disabled.

To modify the system flag setting:

- To modify NABLE_SELECTIVE_MIXING flag setting to NO, manually add it to system.cfg file and set its value to NO.

  For more details, see Modifying System Flags.

Permanent Conference

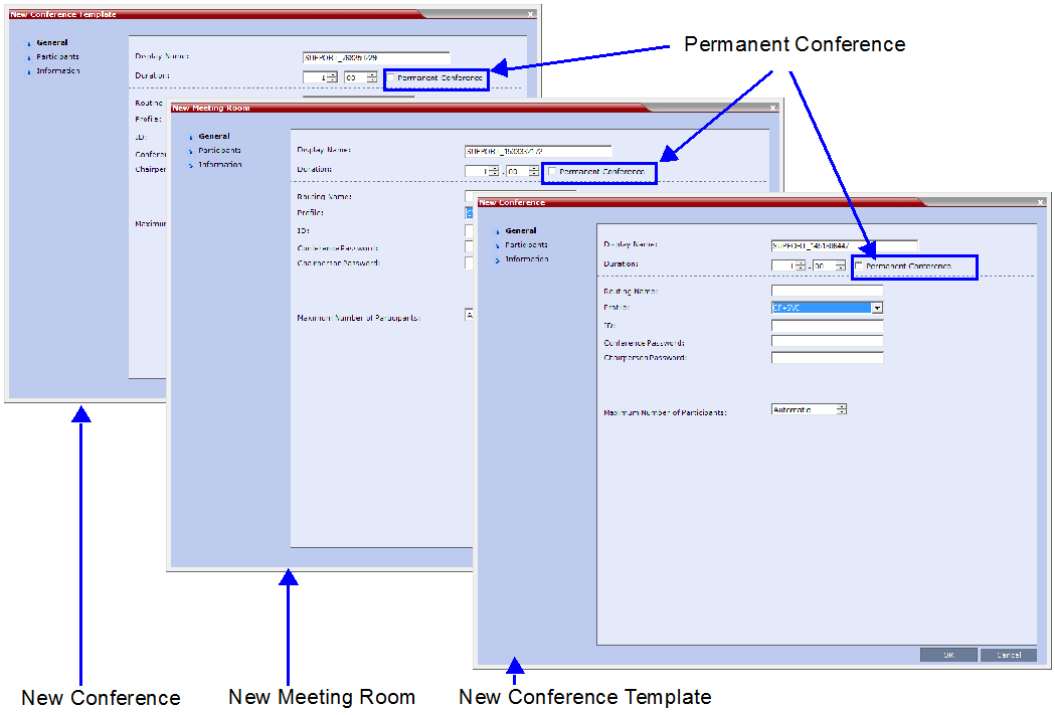
A *Permanent Conference* is any ongoing conference with no pre-determined End Time continuing until it is terminated by an administrator, operator or chairperson.

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.



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.

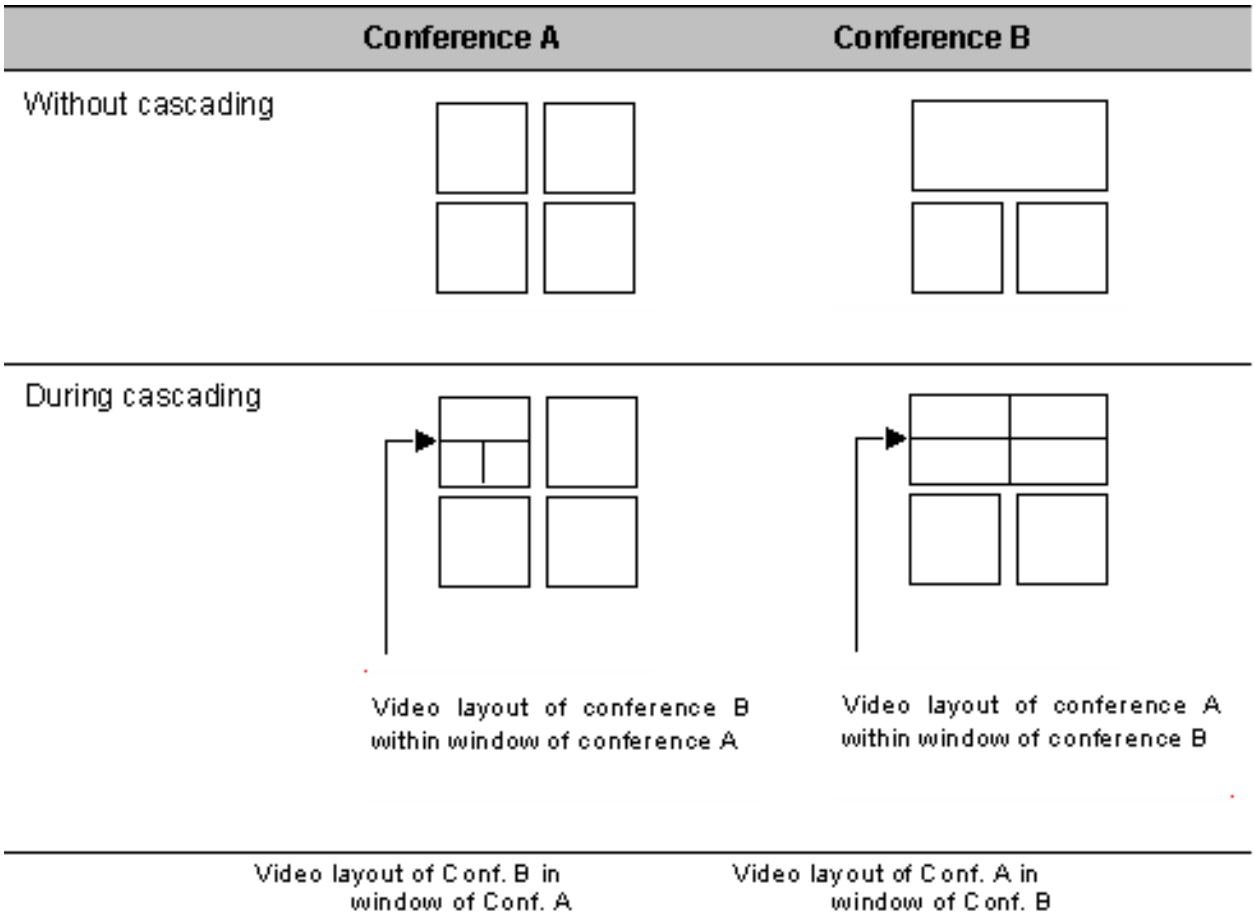
Video Layout in Cascading conferences (CP and mixed CP and SVC)

Cascade links are treated as endpoints in CP conferences and are allocated resources according to Resolution Configuration for CP Conferences on page 6. 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.

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.

Basic Cascading

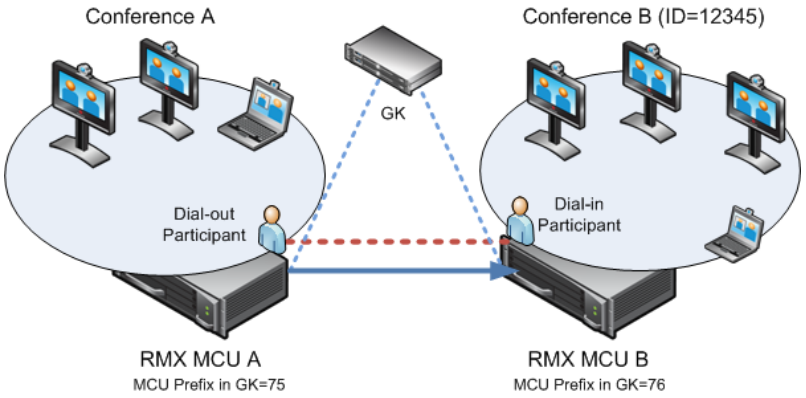
In this topology, a link is created between two conferences, usually running on two different MCUs. The MCUs are usually installed at different locations (states/countries) to save long distance charges by connecting each participant to their local MCU, while only the link between the two conferences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

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**.

Dialing to an Entry Queue

When dialing to an Entry Queue, the dial out participant dials the MCU B prefix or IP address of MCU B and the Entry Queue ID in the format:

[MCU B Prefix/IP address][EQ B ID].

For example, if MCU B prefix is 76 and the Entry Queue ID is 22558, the dial number is **7622558**.

When the participant from conference A connects to the Entry Queue, the system plays to all the participants in Conference A the IVR message requesting the participant to enter the destination conference ID.

At this point, the Conference A organizer or any other participant in the conference can enter the required information for the IVR session using DTMF codes. For example, the meeting organizer enters the destination conference ID - **12345**.

Any DTMF input from conference A is forwarded to the Entry Queue on MCU B to complete the IVR session and enable the move of the participant to the destination conference B.

Once the DTMF codes are entered and forwarded to the Entry Queue on MCU B, the IVR session is completed, the participant moved to the destination conference and the connection between the two conferences is established.

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.

System Flag Settings

The **DTMF_FORWARD_ANY_DIGIT_TIMER_SECONDS** flag determines the time period (in seconds) that MCU A will forward DTMF inputs from conference A participants to MCU B.

Once the timer expires, most of the DTMF codes (excluding five operations as for IP links) entered in conference A will not be forwarded to conference B. This is done to prevent an operation requested by a participant individually (for example, mute my line) to be applied to all the participants in conference B.

Flag range (in seconds): **0 - 360000**

This flag is defined on MCU A (the calling MCU).

If a flag is not listed in the *System Flags* list it must be added before it can be modified. For more details on defining system flags, see Modifying System Flags.

Meeting Rooms

In the RealPresence CloudAxis Solution, the virtual Meeting Rooms are defined in the RealPresence DMA system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

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.

The maximum of number of Meeting Rooms that can be defined is: 1000.

The system is shipped with four default Meeting Rooms:

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

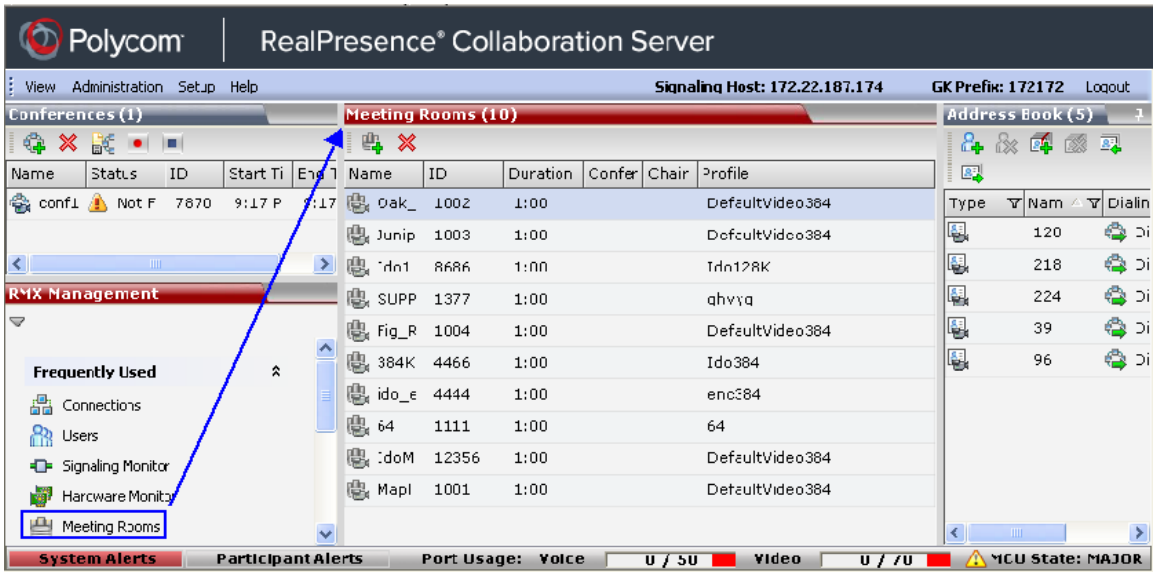
Meeting Rooms List

Meeting Rooms are listed in the **Meeting Room** list pane.

To list Meeting Rooms:

» In the **RMX Management** pane, in the **Frequently Used** list, click the **Meeting Rooms** button .

The **Meeting Rooms** list is displayed.



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:

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.	

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.
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.	
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 a 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.	

Use Time Out as DTMF Delimiter

Users are able to change the behavior of the policy of number collection for VMR Entry Queues, Conference-IDs and Passwords, allowing a time-out to be used as a stop indicator for the input string.

Note: This feature is not supported in the *RealPresence Collaboration Server Virtual Edition.*

In previous versions, a **#** input at the end of the input string indicated completion of the input.

The administrator can configure the system, using the ENABLE_DTMF_NUMBER_WO_DELIMITER system flag to change the previous system behavior, allowing a time-out to be used as a stop indicator for the string input for the local IVR, when the MCU collects the Conference-ID in the local Entry Queue or the Password (chairperson or participant) while routed to the conference.

The flag must be manually added to the System Configuration and its value modified as follows:

Flag Name	Value / Description	
	YES	NO
ENABLE_DTMF_NUMBER_WO_DELIMITER	If the timer expires, the received digits validated even if there is no delimiter. If the received number is not valid, the system will prompt again for the number according to number of retries. that are configured.	**This is the default setting for backward compatibility**. If the timer expires because no delimiter is received, the number input is not valid. The system will prompt again for the number according to number of retries. that are configured.

A System Reset in not required for the flag setting to take effect.

For more information see, `Modifying System Flags` in the *Collaboration Server (RMX) Administrator's Guide*.

Meeting Room Toolbar & Right-Click Menu

The Meeting Room toolbar and right-click menus provide the following functionality:

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.

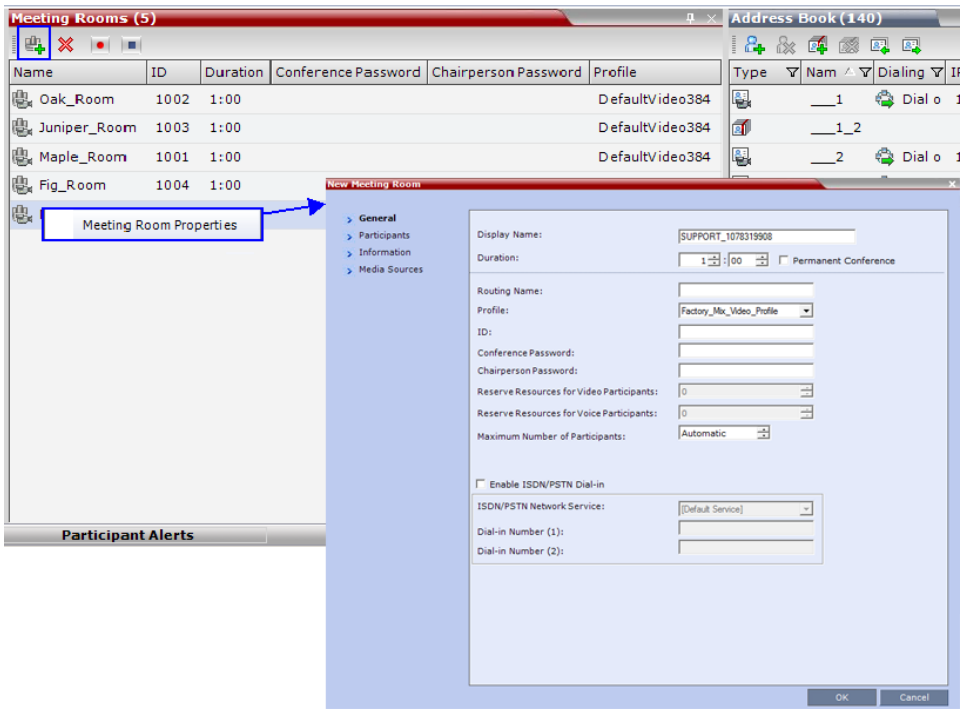
Creating a New Meeting Room

In the RealPresence CloudAxis Solution, virtual Meeting Rooms are defined in the RealPresence DMA system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

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.



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 800s / Virtual Edition Getting Started Guide,* Starting an AVC CP Conference from the Conferences Pane. 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*.

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.

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.

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.

Default Entry Queue properties

The system is shipped with a default Entry Queue whose properties are shown in the following table.

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

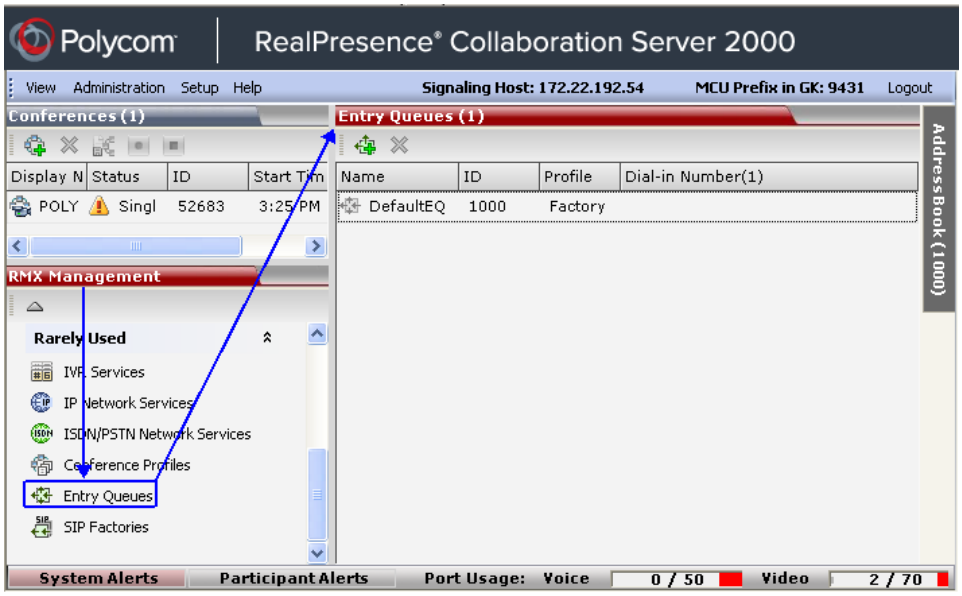
Defining a New Entry Queue

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 Polycom® RealPresence® Collaboration Server 800s and Polycom® 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.

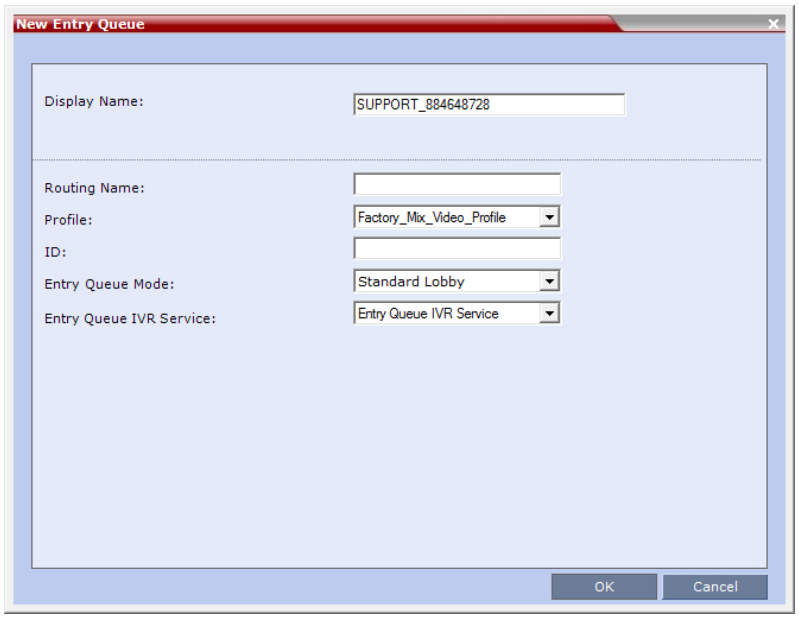
To define a new Entry Queue:

1 In the **RMX Management** pane, In the **Rarely Used** menu, click **Entry Queues.**



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:

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.
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 Definitions Parameters (Continued)

Option	Description
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 Only Service Provider - 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 Only Service Provider Entry Queue does not route the SIP calls to a target conference. Instead the RealPresence DMA system handles the call. For more details, see IVR Provider Entry Queue (Shared Number Dialing).
	External IVR Control - IVR Services can be controlled externally from an application server (such as the DMA) supporting the MCCF-IVR (Media Control Channel Framework-Interactive Voice Response) package. When selected, the connection process of the participant to the conference via the Virtual Entry Queue is controlled and managed by an external IVR service of an application server (for example, DMA).
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**.

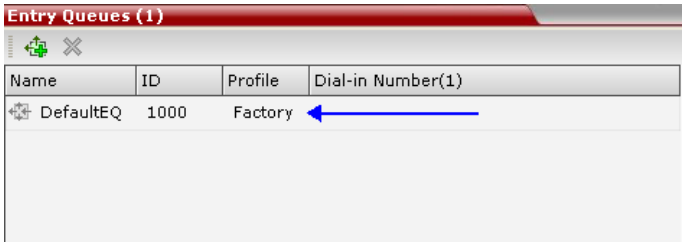
The new Entry Queue is added to the **Entry Queues** list.

Listing Entry Queues

To view the list of Entry Queues:

- In the **RMX Management** pane- **Rarely Used** menu, click **Entry Queues**.

  The Entry Queues are listed in the **Entry Queues** pane.



  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 **RMX Management** pane, **Rarely Used** list, 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 **RMX Management** pane, **Rarely Used** list, click **Entry Queues**.
2. In the **Entry Queues** list, right-click the **Transit Entry Queue** entry, and then click **Cancel Transit Entry Queue**.

IVR Provider Entry Queue (Shared Number Dialing)

In an environment that includes a RealPresence DMA system, the Collaboration Server Entry Queue can be configured to provide the IVR Services on behalf of the RealPresence DMA system to SIP endpoints. It displays the Welcome Slide, plays the welcome message and retrieves the destination conference ID that is entered by the participant using DTMF codes.

To enable this feature, a special Entry Queue that is defined as IVR Only Service Provider is created. This Entry Queue does not forward calls to conferences running on the Collaboration Server and its main functionality is to provide IVR services.

Call Flow

The SIP participant dials the DMA Virtual Entry Queue number, for example 1000@dma.polycom.com.

The DMA forwards the SIP call to the Collaboration Server, to a special Entry Queue that is configured as IVR Only Service Provider. The participant is prompted to enter the conference ID using DTMF codes.

Once the participant enters the conference ID, the conference ID is forwarded to the DMA, enabling the DMA to connect the SIP endpoint to the destination conference or create a new conference and connect the participant to that conference.

Guidelines for Setting the Entry Queue as IVR Provider

- An Entry Queue defined as IVR Only Service Provider does not route the SIP call to a target conference and it cannot be used to route calls on the Collaboration Server. In such a configuration, the DMA handles the calls. Therefore, normal Entry Queues must be defined separately.
- **Operator Assistance** must be disabled in the IVR Service assigned to this Entry Queue.
- Only the conference ID prompts should be configured. Other prompts are not supported in IVR Only Service Provider configuration.
- ISDN and H.323 calls to this Entry Queue are rejected.
- The DMA must be configured to locate the IVR Only Service Provider Entry Queue on the Collaboration Server. To locate the Entry Queue the DMA requires the Entry Queue's ID number and the Collaboration Server Signaling IP address (xxx.xx.xxx.xx).

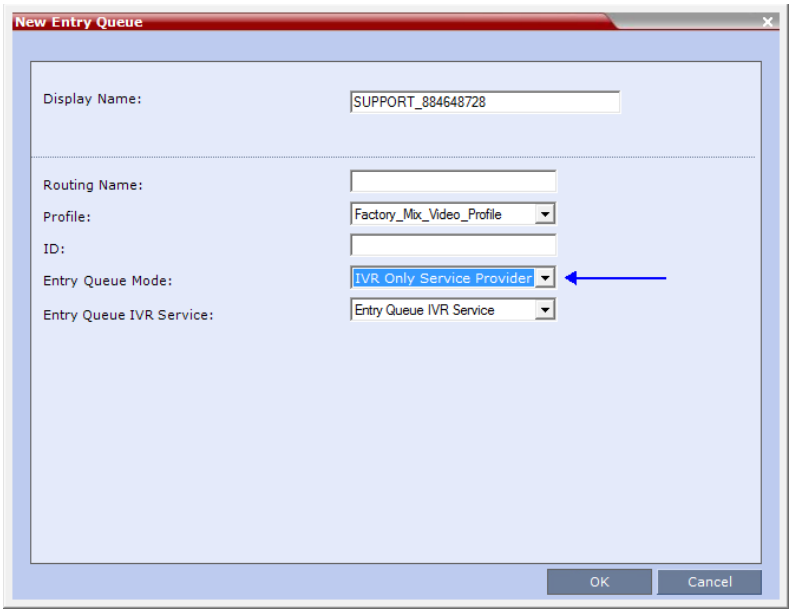
Configuring the Collaboration Server as IVR Provider

Entry Queue IVR Service

If required, create a special Entry Queue IVR Service in which the **Operator Assistance** option is disabled, and only the Conference ID prompts are enabled.

Entry Queue

» In the **New Entry Queue** dialog box, **Entry Queue Mode** list, select **IVR Only Service Provider.**



- Enter the Entry Queue ID that will be used by the DMA to forward the SIP calls to this Entry Queue.
- Select the special **Entry Queue IVR Service** if one was created.

Configuring the MCU to Support External IVR Services via the MCCF-IVR

The support of External IVR Services via the MCCF-IVR package is enabled by default in the Collaboration Server (RMX) systems, by the flag **ENABLE_MCCF** which is set to **YES**.

However, in secured environments where the External IVR Services via the MCCF-IVR package is not required and unused ports should be closed, this flag should be set to **NO**.

To change this flag value from YES to No, you must first add it to the System Configuration. For more details, see Manually Adding and Deleting System Flags.

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, reservations, 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.

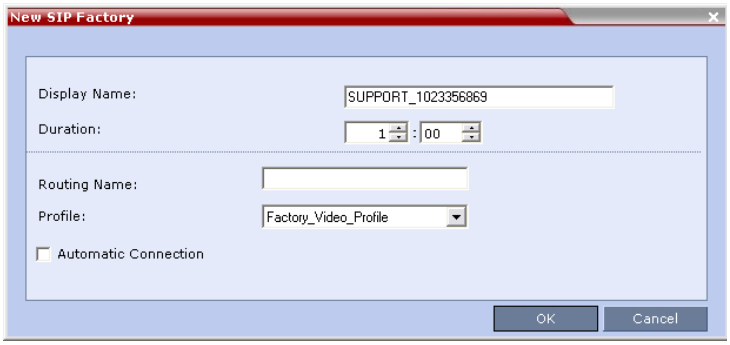
The maximum of number of SIP Factories that can be defined is 40.

Creating SIP Factories

To create a new SIP Factory:

1. In the **RMX Management** pane, **Rarely Used** list, click **SIP Factories**.
2. In the **SIP Factories** list pane, click the **New SIP Factory** button.

   The **New Factory** dialog box opens.



3 Define the following parameters:

New Factory Properties

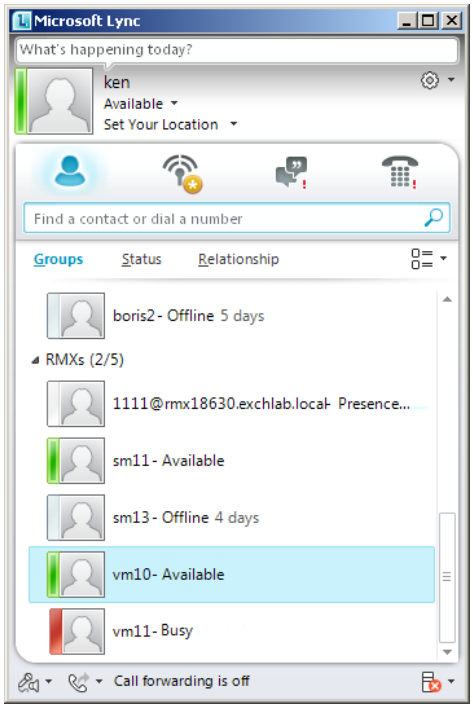
Option	Description
Display Name	Enter the SIP Factory name that will be displayed. The Display Name is the conferencing entity name in native language character sets to be displayed in the Collaboration Server Web Client. In conferences, Meeting Rooms, Entry Queues and SIP factories the system automatically 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 with SIP Servers

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.

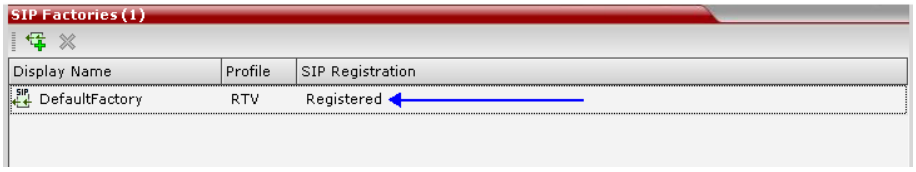


Guidelines for registering Entry Queues and SIP Factories with SIP Servers

- 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 Defining New Profiles.

Monitoring Registration Status

The SIP registration status can be viewed in the **Entry Queue** or **SIP Factory** list panes.



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 is monitored and controlled as any standard ongoing conference.

For more information about Ad Hoc conferencing, see Appendix D, Appendix D - Ad Hoc Conferencing and External Database Authentication.

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® and Polycom® 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**.

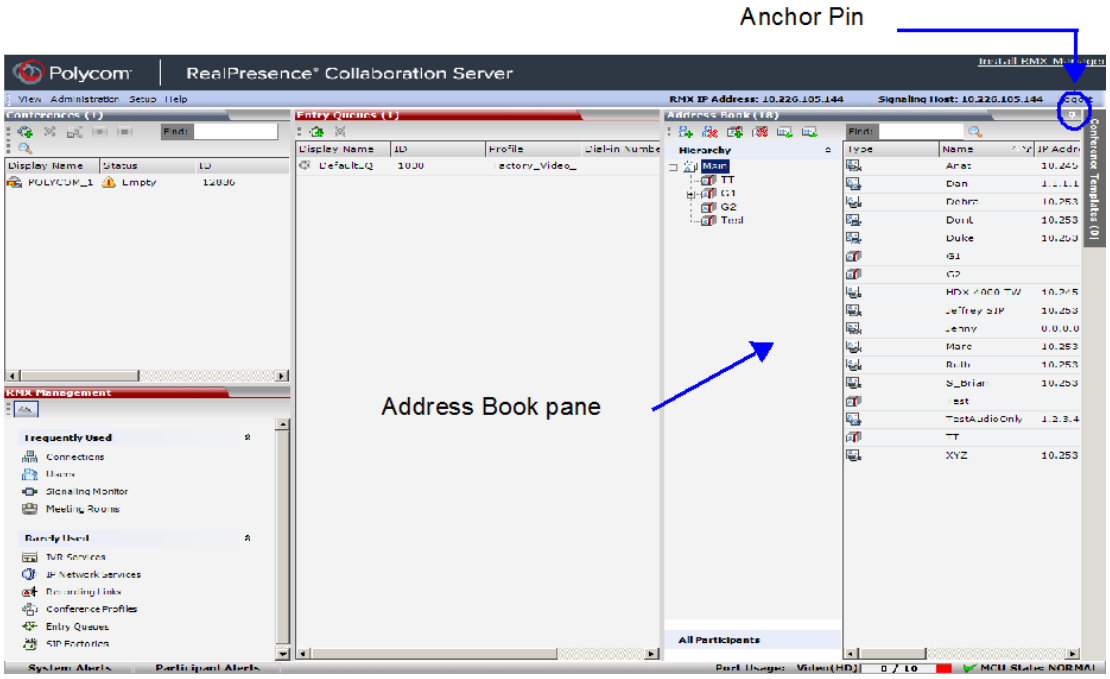
Importing and exporting of Address Books enables organizations to seamlessly distribute up-to-date Address Books to multiple Collaboration Server units. It is not possible to distribute Address Books to external databases running on applications such as the **RealPresence Resource Manager** or **Polycom 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** or **Polycom CMA**, the appropriate system configuration flags must be set.

For more information, see Modifying System Flags.

Integration with the Global Address Book of the Polycom® RealPresence® Resource Manager® or Polycom CMA® is supported. For more information, see Integrating the Global Address Book (GAB) of Polycom RealPresence Resource Manager or Polycom CMA with the Collaboration Server. Integration with the **SE200** GAB (Global Address Book) is not supported.

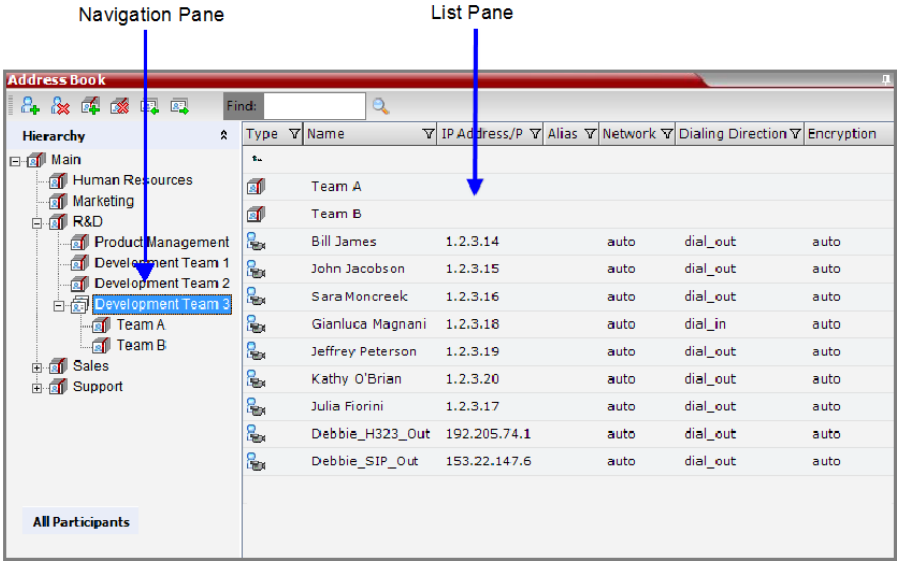
Viewing the Address Book

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.



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.



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:

Participants List Pane

Field/Option	Description
Type	Indicates whether the participant is a video ( ) or voice ( ).
Name	Displays the name of the participant.

Participants List Pane (Continued)

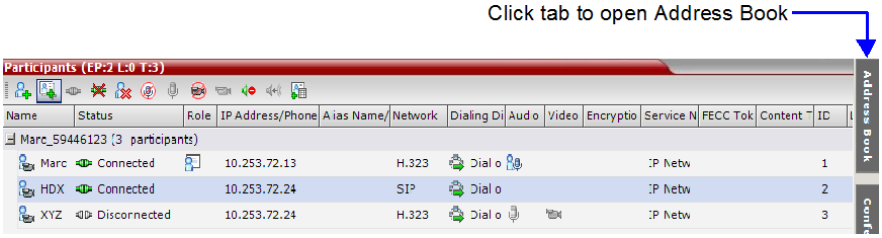
Field/Option	Description
IP Address/Phone	Enter the IP address of the participant's endpoint. • For H.323 participant define either the endpoint IP address or alias. • For SIP participant define either the endpoint IP address or the SIP address.
Network	The network communication protocol used by the endpoint to connect to the conference: **H.323** or **SIP**.
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.

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**.



Adding Participants from the Address Book to Conferences

In the *RealPresence CloudAxis Solution*, Participant Address Book is defined in the Polycom® RealPresence® Resource Manager® component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

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.

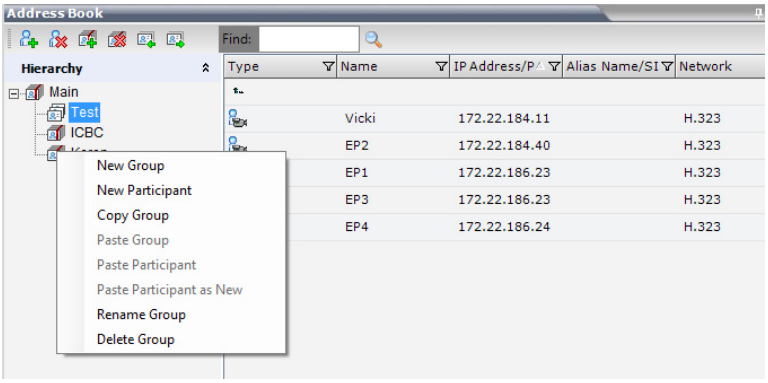
Groups can contain participants and sub-groups. You can define up to ten levels in the “Main” group.

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.



2 Select one of the following actions:

Address Book Navigation

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.
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 (IP) defined for the connection in the participant properties.

Adding a New participant to the Address Book Directly

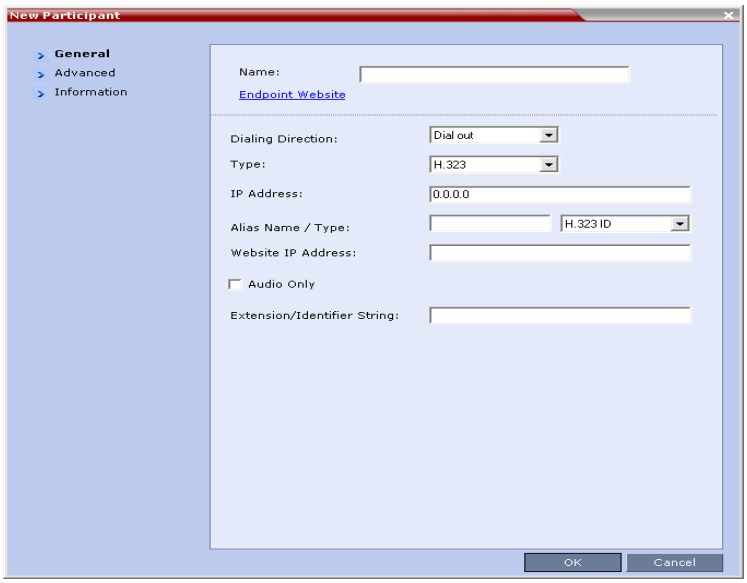
You can add a new participant to the “Main” group or to a group in the **Address Book**. Additionally, you can add a participant from a new conference, ongoing conference, or **Conference Template**.

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.



3 Define the following fields:

New Participant - General

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 800s / Virtual Edition Getting Started Guide*, *Audio and Visual Indications (AVC CP Conferencing)*". **Note:** This field is displayed in all tabs.

New Participant - General (Continued)

Field	Description
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. i
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** or **ISDN/PSTN**. The fields in the dialog box change according to the selected network type.
IP Address (H.323 and SIP)	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.
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 lin. • 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.

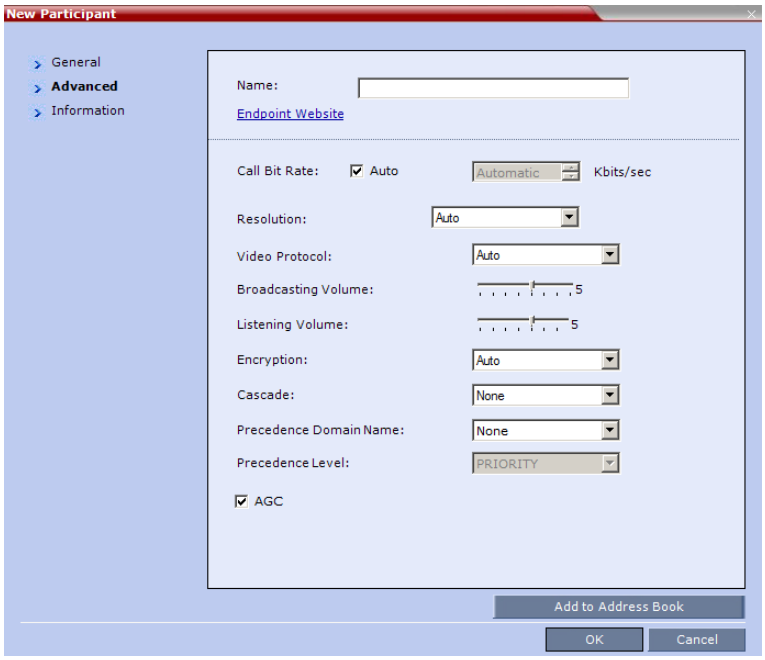
New Participant - General (Continued)

Field	Description
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 **(IP only)**	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.
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	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:



New Participant - Advanced

Field	Description
Video Bit Rate / Auto (IP Only)	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.

New Participant - Advanced (Continued)

Field	Description
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.
Cascaded **(IP Only)**	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.
Precedence Domain Name (**Dial-out SIP Only**)	When Multi Level Precedence and Preemption is used, this is the Precedence Domain Name for the participant. For more information see MLPP (Multi Level Precedence and Preemption).
Precedence Level (**Dial-out SIP Only**)	When Multi Level Precedence and Preemption is used, this is the Precedence Level for the participant For more information see MLPP (Multi Level Precedence and Preemption).
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.

The flag, **REMOVE_IP_IF_NUMBER_EXISTS** controls 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 if an E.164 number (or other) exists.
- **NO** - The IP address of an endpoint will be used.

Guidelines for Substituting E.164 Number in Dial String

- 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:

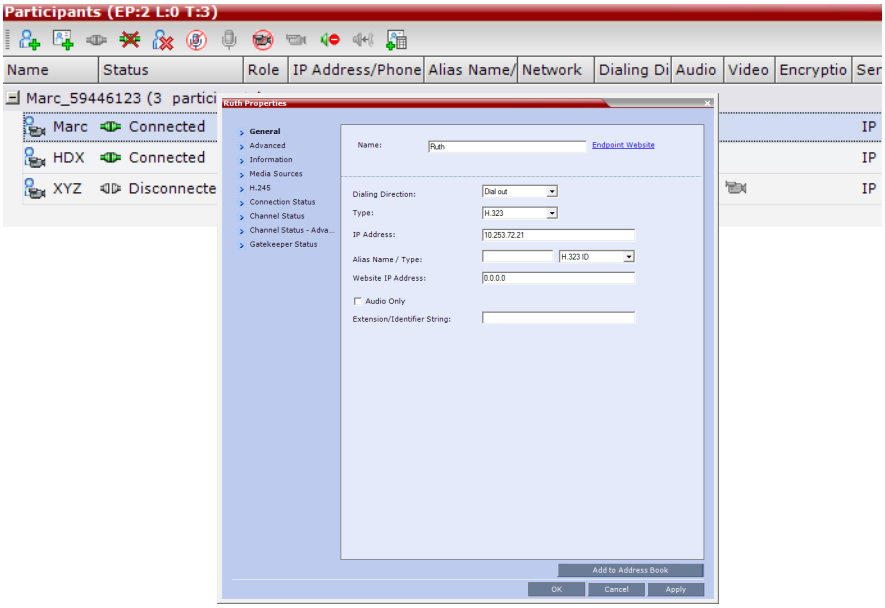
1 During an ongoing conference, select the participant in the *Participant* pane, and either click the **Add Participant to Address Book** button (), or right-click and select **Add Participant to Address Book**.

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.



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

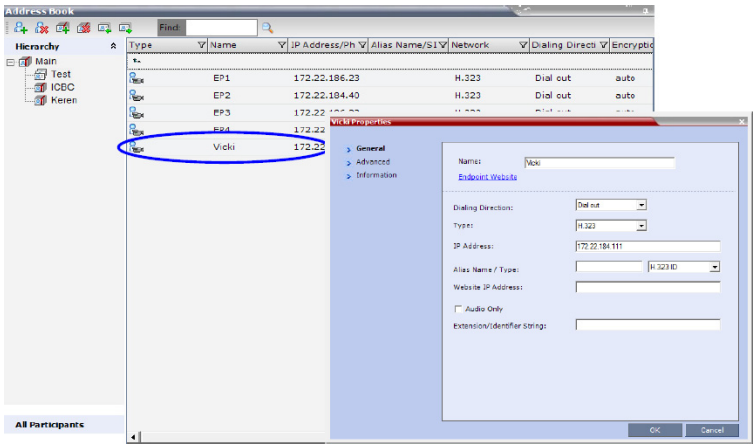
When required, you can modify the participant's properties.

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.



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**.

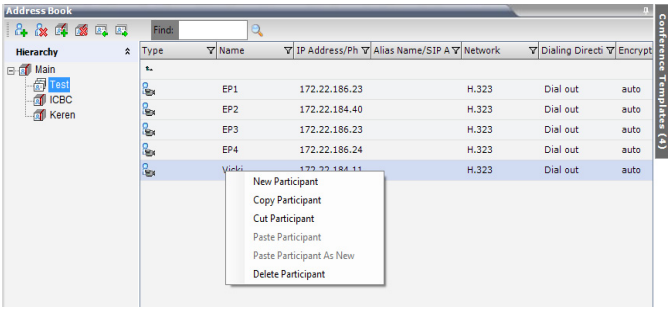
The changes to the participant's properties are updated.

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.



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.

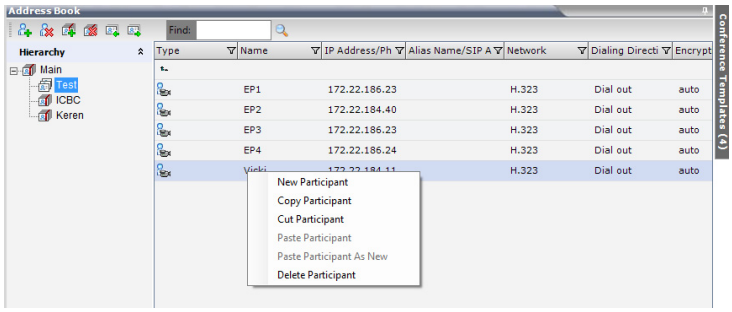
Copying or Moving a Participant

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:



Copy / Move Participant

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:

Paste Participant

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

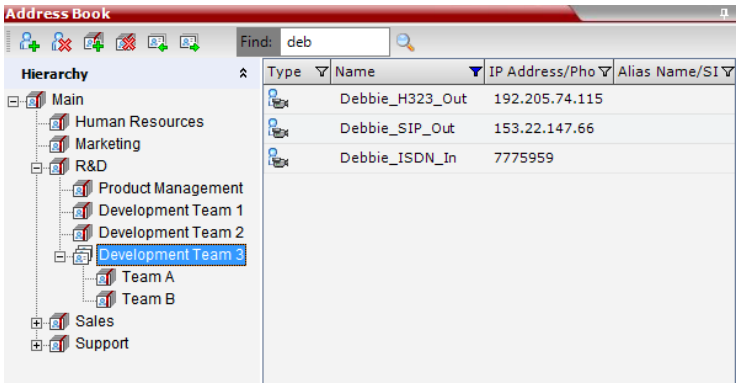
You can search the Address Book for a participant's name or a group name only on the currently selected group/level.

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.



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

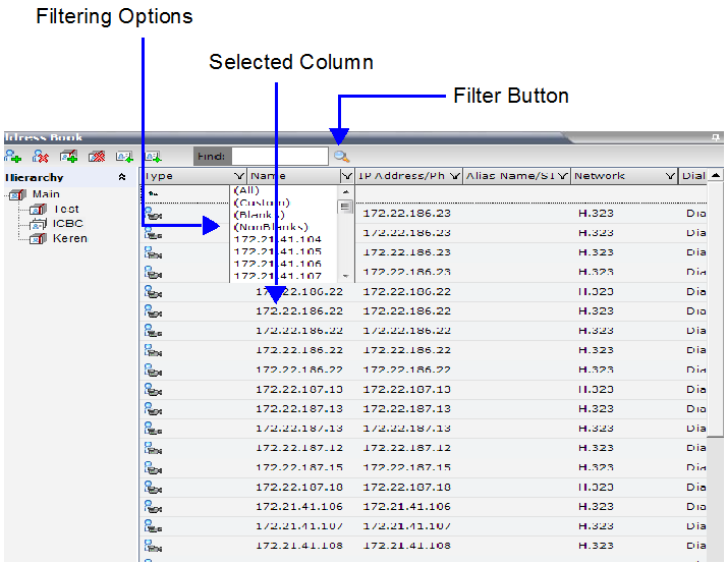
When you use the Find dialog box to search filtered data, only the data that is displayed is searched; data that is not displayed is not searched. To search all the data, clear all filters.

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.

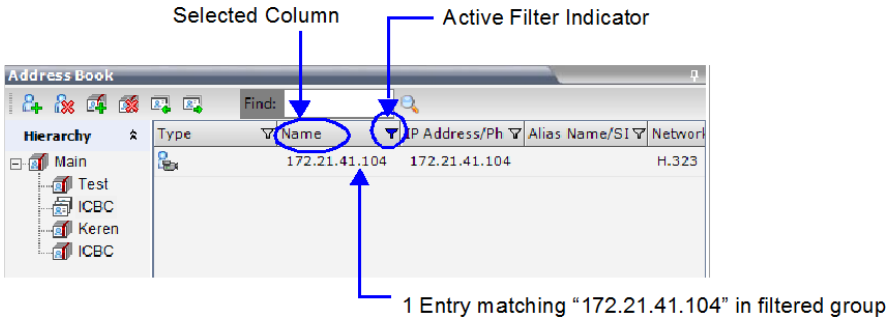
   A drop-down menu is displayed containing all the matching patterns that can be applied to the selected field.



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:



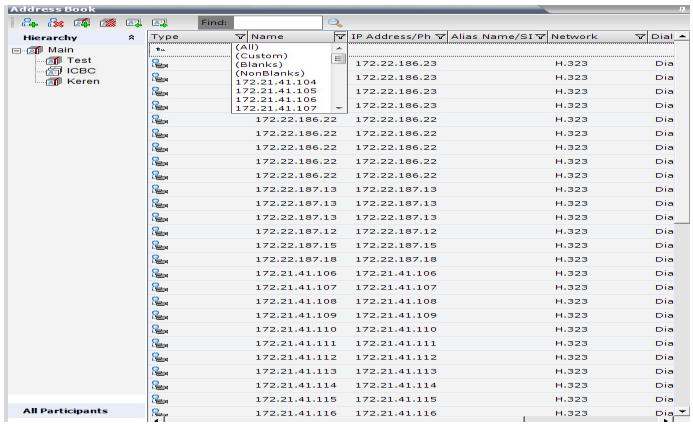
Filtering Address Book Data Using a Custom Pattern

To filter the data in an address book group:

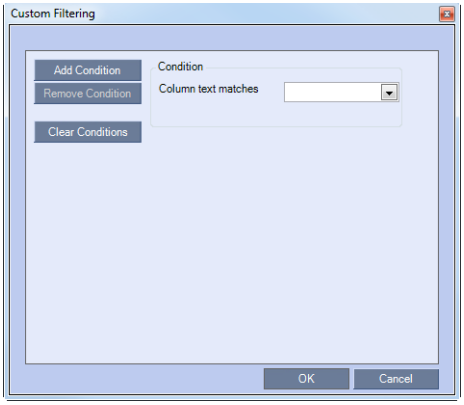
1 In the **Address Book - Navigation** pane, select the group to filter.

2 In the **Address Book - List** pane, in the column that you want to use for filtering, click the **filter** (▼) button.

3 Select the **(Custom)** option from the drop-down list.



The **Custom Filtering** dialog box opens.



4 In the **Condition - Column text matches** field, enter the filtering pattern.

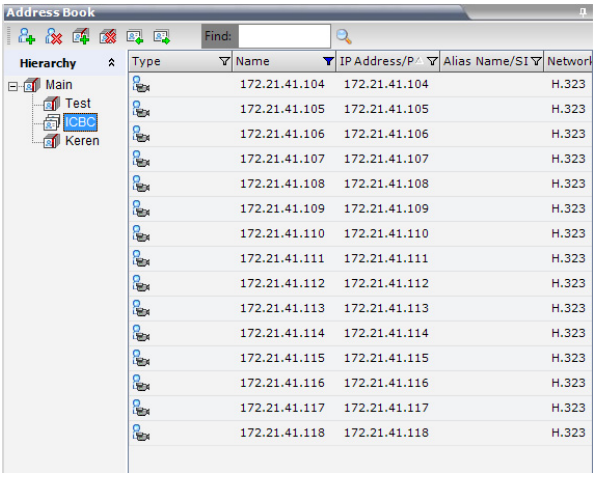
For example, to list only endpoints that include the numerals 41 in their name, enter 41.

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.



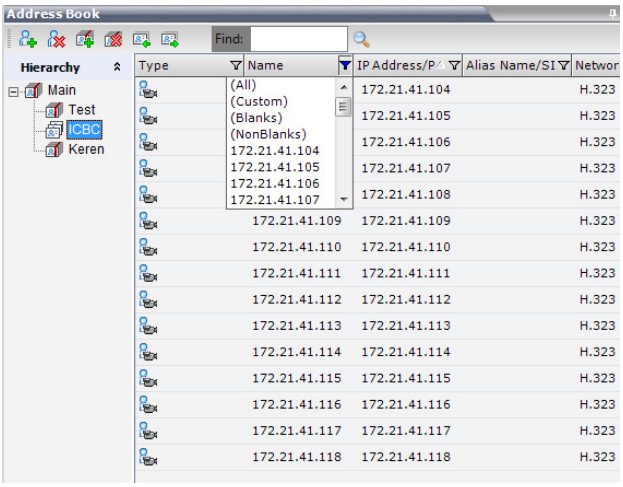
Clearing the Filter

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)**.



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 for Obtaining the Display Name from the Address Book

- 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 for IPv4 participants only .

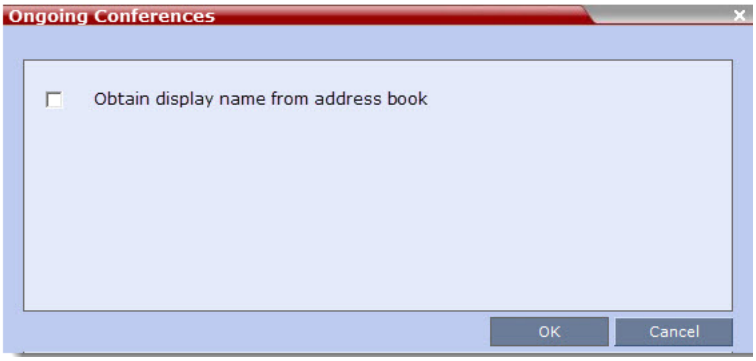
Enabling and Disabling the Obtain Display Name from Address Book Feature

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.



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

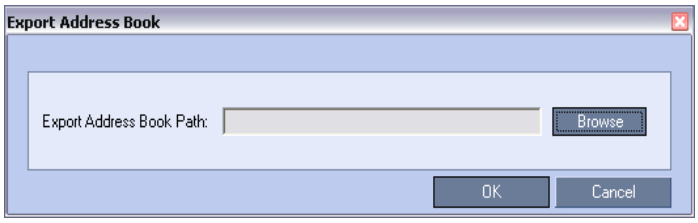
Address Books are proprietary Polycom data files that can only be distributed among Collaboration Server units. The Address Books are exported in XML format, which are editable offline. If no name is assigned to the exported Address Book, the default file name is:
`EMA.DataObjects.OfflineTemplates.AddressbookContent_.xml`

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.

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 or Polycom CMA with the Collaboration Server

The RealPresence Resource Manager 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 or Polycom CMA

- The Collaboration Server can use only one address book at a time. After you integrate the Polycom RealPresence Resource Manager or Polycom CMA with the Polycom Collaboration Server, the CMA address book replaces the Collaboration Server internal address book.
- The Collaboration Server uses the RealPresence Resource Manager or Polycom CMA address book in read-only mode. You can only add or modify CMA address book entries from the RealPresence Resource Manager or Polycom CMA.

The Collaboration Server acts as a proxy to all address book requests between the Collaboration Server Web Client and the CMA. **Ensure that firewall and other network settings allow the Collaboration Server access to the CMA server.**

To Integrate the RealPresence Resource Manager or Polycom CMA Global Address Book (GAB) with the Collaboration Server:

RealPresence Resource Manager or Polycom CMA Side

1 In the RealPresence or Polycom application, manually add the Polycom Collaboration Server system to the RealPresence Resource Manager or Polycom CMA system as directed in the *RealPresence Resource Manager or Polycom CMA Operations Guide*.

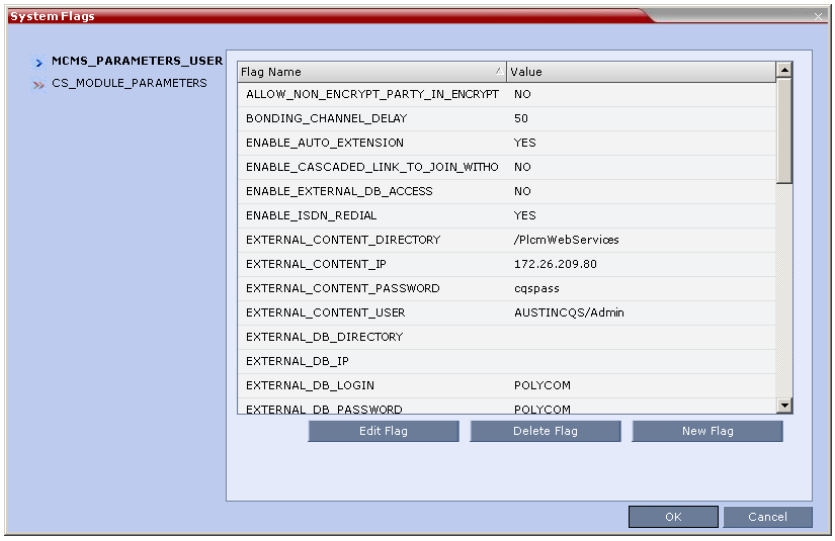
2 In the RealPresence Resource Manager or Polycom CMA application, add a user or use an existing user for Collaboration Server login as directed in the **R***ealPresence Resource Manager or Polycom CMA Operations Guide*.

Write down the User Name and Password as they will be used later to define the Collaboration Server connection to the RealPresence Resource Manager 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.



2 Modify the values of the flags in the table below.

For more information, see Modifying System Flags.

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 or Polycom CMA application. Default: /PlcmWebServices
EXTERNAL_CONTENT_IP	Enter the IP address of the RealPresence Resource Manager 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 or Polycom CMA Global Address Book (GAB). Leave this flag blank to disable address book integration with the RealPresence Resource Manager or Polycom CMA server.
EXTERNAL_CONTENT_PASSWORD	The password associated with the user name defined for the Collaboration Server in the RealPresence Resource Manager or Polycom CMA server.

System Flags for CMA Address Book Integration

Flag	Description
EXTERNAL_CONTENT_USER	The login name defined for the Collaboration Server in the RealPresence Resource Manager or Polycom CMA server defined in the format: domain name/user name.

3 Click **OK** to complete the definitions.

4 When prompted, click **Yes** to reset the MCU and implement the changes to the system configuration.

Scheduling Reservations

In the *RealPresence CloudAxis Solution*, Reservations are scheduled in the RealPresence Resource Manager component and should not be scheduled directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

The *Reservations* option enables users to schedule conferences. These conferences can be launched immediately or become ongoing, at a specified time on a specified date.

Scheduling a conference reservation requires definition of conference parameters such as the date and time at which the conference is to start, the participants and the duration of the conference.

Scheduled conferences (Reservations) can occur once or repeatedly, and the recurrence pattern can vary.

The maximum number of reservations per Collaboration Server is 2000.

Guidelines for Scheduling Reservations

System

- By default, the *Scheduler* is enabled by a *System Flag*. The flag prevents potential scheduling conflicts from occurring as a result of system calls from external scheduling applications such as the Polycom® RealPresence® Resource Manager®, *ReadiManager®*, *SE200, Polycom CMA™* and others via the API.

  If an external scheduling application is used, the flag **INTERNAL_SCHEDULER** must be manually added to the *System Configuration* and its value must be set to NO.
  For more information see Modifying System Flags.

Resources

- System resources are calculated according to the Collaboration Server's license. For more information, see Forcing Video Resource Allocation to CIF Resolution.
- System resource availability is partially checked when reservations are created:
  - If a conference duration extension request is received from an ongoing conference, the request is rejected if it would cause a resource conflict.
  - If several reservations are scheduled to be activated at the same time and there are not enough resources for all participants to be connected:
    - The conferences are activated.
    - Participants are connected to all the ongoing conferences until all system resources are used up.
- If sufficient resources are not available in the system and a scheduled *Reservation* cannot be activated, the *Reservation* is deleted from the schedule.
- Resources are reserved for participants at the highest video resolution supported by the *Line Rate* specified in the conference *Profile* and up to the maximum system video resolution specified by the **Resolution Configuration** dialog box.

- When a new *Reservation* is created in the *Reservation Calendar*, the effect of the new *Reservation* (including its recurrences) on available resources is checked. If resource deficiencies are found an error message is displayed.

  Defined dial-in or dial-out participants, Meeting Rooms, Entry Queues and new connections to Ongoing conferences are not included in the resources calculation.

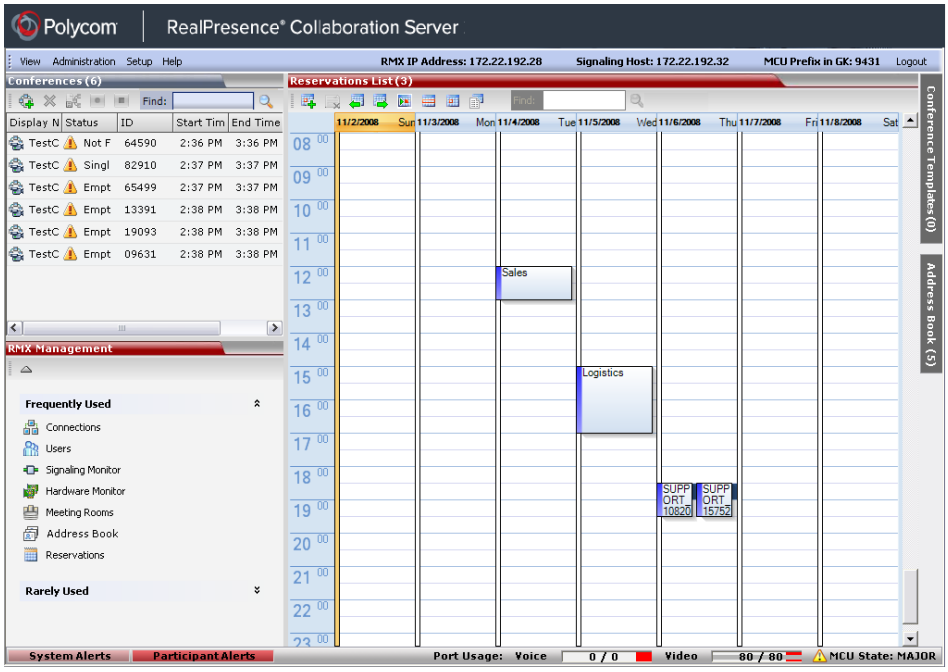
Reservations

- A *Reservation* that has been activated and becomes an ongoing conference is deleted from the *Reservation Calendar* list.
- The maximum number of concurrent reservations is 80. Reservations with durations that overlap (for any amount of time) are considered to be concurrent.
- System resource availability is partially checked when reservations are created:
  - If a conference duration extension request is received from an ongoing conference, the request is rejected if it would cause a resource conflict.
  - If several reservations are scheduled to be activated at the same time and there are not enough resources for all participants to be connected:
    - The conferences are activated.
    - Participants are connected to all the ongoing conferences until all system resources are used up.
- A scheduled *Reservation* cannot be activated and is deleted from the schedule if:
  - An Ongoing conference has the same *Numeric ID*.
  - Sufficient resources are not available in the system.
- If a problem prevents a *Reservation* from being activated at its schedule time, the *Reservation* will not be activated at all. This applies even if the problem is resolved during the *Reservation's* scheduled time slot.
- A Profile that is assigned to a Reservation cannot be deleted.
- Reservations are backed up and restored during **Setup > Software Management >Backup /Restore Configuration** operations. For more information see Software Management.
- All existing reservations are erased by the *Standard Restore* option of the **Administration > Tools > Restore Factory Defaults** procedure.
- *Reservations* can also be scheduled from *Conference Templates*. For more information see Scheduling a Reservation From a Conference Template.

Using the Reservation Calendar

To open the Reservation Calendar:

- In the **RMX Management** pane, click the **Reservation Calendar** button ( ).



Toolbar Buttons

The toolbar buttons functions are described in the table below.

Reservations – Toolbar Buttons

Button	Description
New Reservation	Create a new reservation. The date and time of the new reservation is set according to the highlighted blocks on the *Reservation Calendar*.
Delete Reservation	Click to delete the selected reservation.
Back	Click to show the previous day or week, depending on whether *Show Day* or *Show Week* is the selected.
Next	Click to show the next day or week, depending on whether *Show Day* or *Show Week* is the selected.
Today	Click to show the current date in the Reservation Calendar in either *Show Day* or *Show Week* view.

Button	Description
 Show Week	Change the calendar view to weekly display, showing a calendar week: Sunday through Saturday
 Show Day	Click this button to show the day containing the selected time slot.
 Reservations List	Click to change to List View and display a list of all reservations.
	Used to search for reservations by *Display Name.* (Available in *Reservations List view* only).

Reservations Views

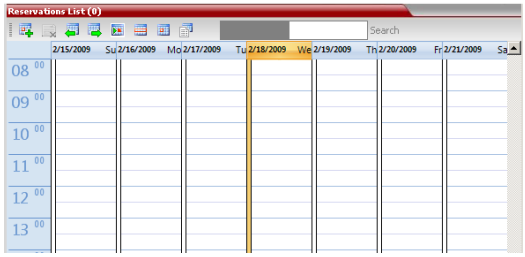
The *Reservation Calendar* list has the following views available:

- Week
- Day
- Today
- List

In all views the *Main Window List Pane* header displays the total number of reservations in the system.

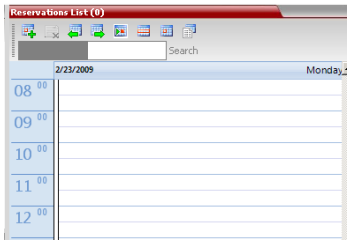
Week View

By default the *Reservation Calendar* is displayed in *Week* view with the current date highlighted in orange.



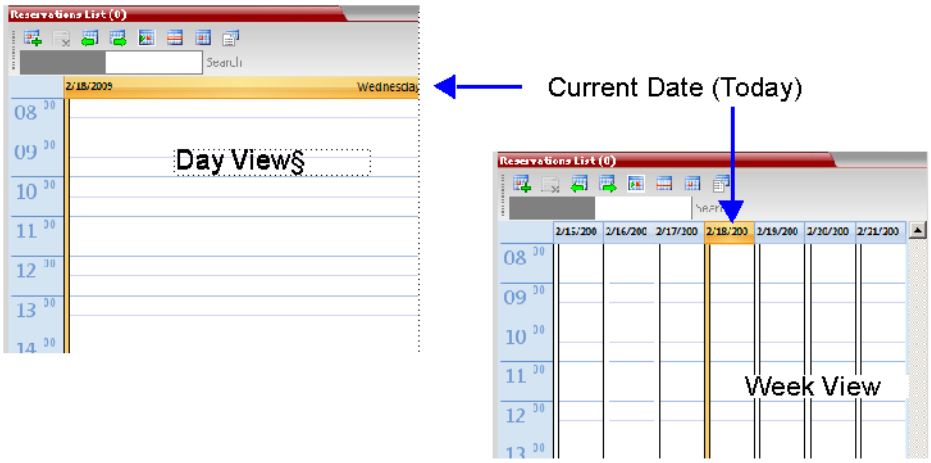
Day View

A single day is displayed.



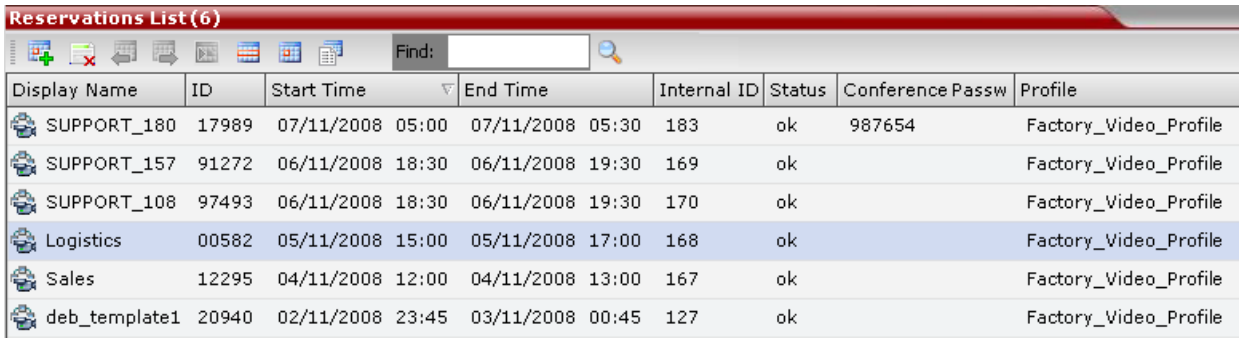
Today View

The current date (*Today*), highlighted in orange, can be viewed in both *Week View* and *Day View*.



List View

List View does not have a calendar based format.



Display Name	ID	Start Time	End Time	Internal ID	Status	Conference Passw	Profile
SUPPORT_180	17989	07/11/2008 05:00	07/11/2008 05:30	183	ok	987654	Factory_Video_Profile
SUPPORT_157	91272	06/11/2008 18:30	06/11/2008 19:30	169	ok		Factory_Video_Profile
SUPPORT_108	97493	06/11/2008 18:30	06/11/2008 19:30	170	ok		Factory_Video_Profile
Logistics	00582	05/11/2008 15:00	05/11/2008 17:00	168	ok		Factory_Video_Profile
Sales	12295	04/11/2008 12:00	04/11/2008 13:00	167	ok		Factory_Video_Profile
deb_template1	20940	02/11/2008 23:45	03/11/2008 00:45	127	ok		Factory_Video_Profile

All *Reservations* are listed by:

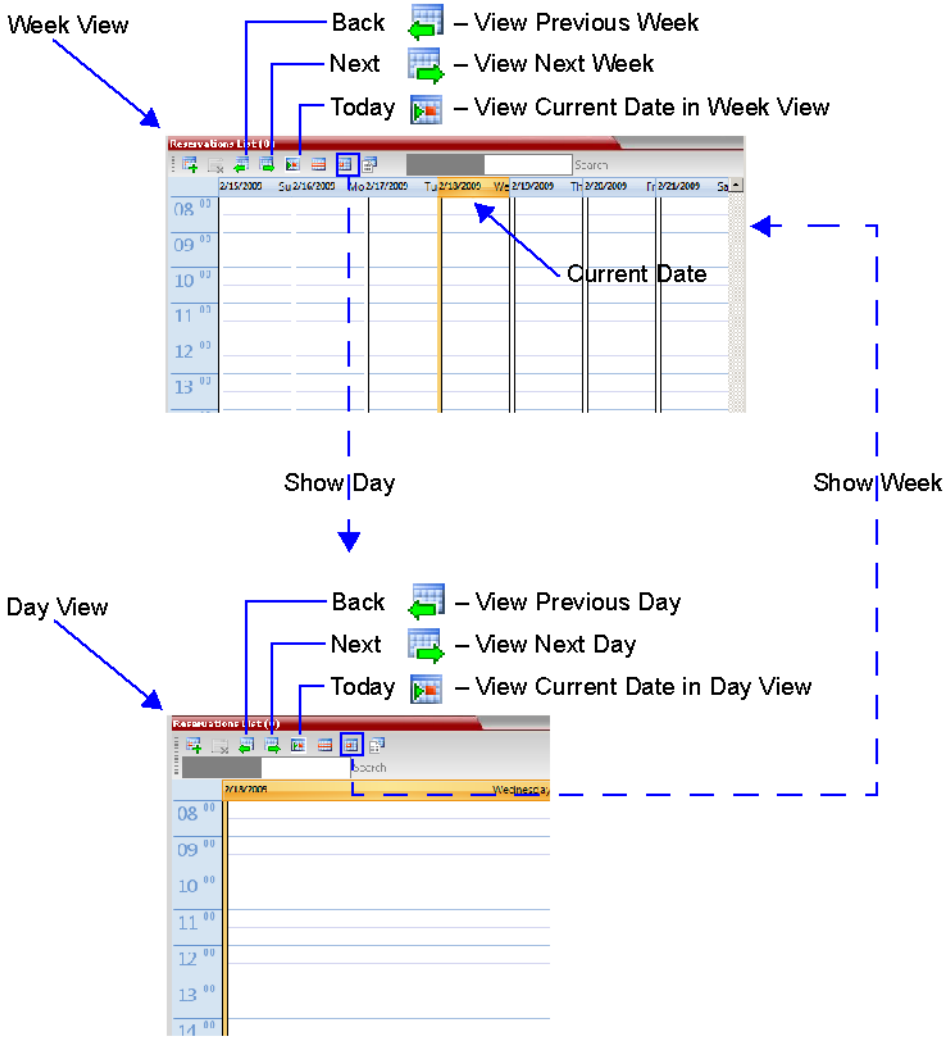
• Display Name	• End Time
• ID	• Status
• Internal ID	• Conference Password
• Start Time	• Profile

The *Reservations* can be sorted, searched and browsed by any of the listed fields.

Changing the Calendar View

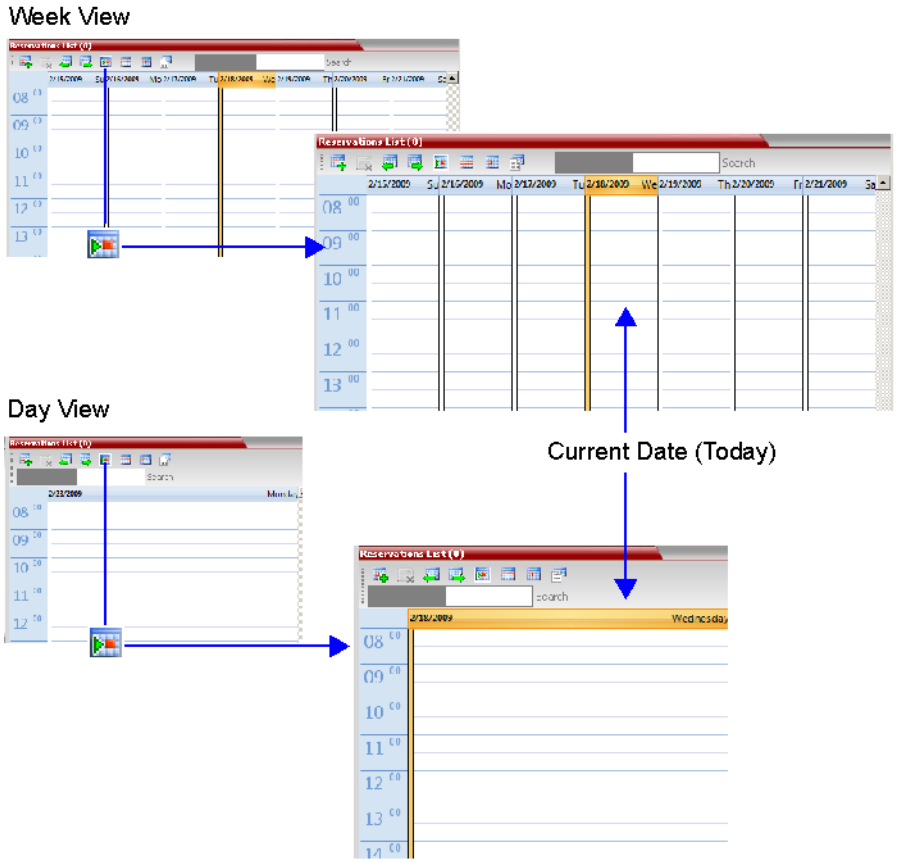
To change between Week and Day views:

- In Week View: In the *Reservation Calendar* toolbar, click **Show Day** ( ) to change to *Day View*.
  or
  In Day View: In the *Reservation Calendar* toolbar, click **Show Week** ( ) to change to *Week View*.



To view Today (the current date):

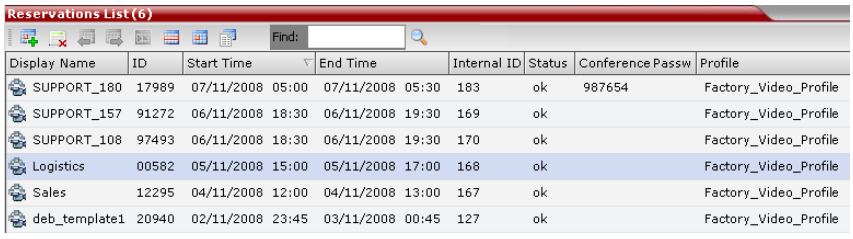
- In *Week View* or *Day View*, in the *Reservation Calendar* toolbar, click the **Today** ( ) button to have the current date displayed within the selected view.



To change to List View:

1 In the *Reservation Calendar* toolbar, click, the **Reservations List** ( ) button.

The *Reservations List* is displayed.



Display Name	ID	Start Time	End Time	Internal ID	Status	Conference Passw	Profile
SUPPORT_180	17989	07/11/2008 05:00	07/11/2008 05:30	183	ok	987654	Factory_Video_Profile
SUPPORT_157	91272	06/11/2008 18:30	06/11/2008 19:30	169	ok		Factory_Video_Profile
SUPPORT_108	97493	06/11/2008 18:30	06/11/2008 19:30	170	ok		Factory_Video_Profile
Logistics	00582	05/11/2008 15:00	05/11/2008 17:00	168	ok		Factory_Video_Profile
Sales	12295	04/11/2008 12:00	04/11/2008 13:00	167	ok		Factory_Video_Profile
deb_template1	20940	02/11/2008 23:45	03/11/2008 00:45	127	ok		Factory_Video_Profile

2 **Optional.** Sort the data by any field (column heading) by clicking on the column heading.

A ▽ or △ symbol is displayed in the column heading indicating that the list is sorted by this field, as well as the sort order.

3 **Optional.** Click on the column heading to toggle the column's sort order.

To return to Calendar View:

- In the *Reservation Calendar* toolbar, click any of the buttons (**Show Week**/**Show Day**/**Today**) to return to the required *Reservation Calendar* view.

Scheduling Conferences Using the Reservation Calendar

In the *RealPresence CloudAxis Solution*, Reservations are scheduled in the RealPresence Resource Manager component and should not be scheduled directly in the RealPresence Collabporatio Server Virtual Edition component.

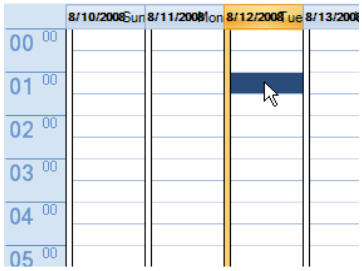
Creating a New Reservation

There are three methods of creating a new reservation:

- Method I – Creating a reservation with default duration of 1 hour
- Method II – Creating a reservation with default duration of ½ hour
- Method III – Interactively define the reservation duration

Each method requires the selection of a starting time slot in the *Reservation Calendar*. The default time slot is the current half-hour period of local time.

In all views, if the **New Reservation** ( ) button is clicked without selecting a starting time slot or if a time slot is selected that is in the past, the *Reservation* becomes an Ongoing conference immediately and is not added to the *Reservations* calendar.



After selecting a starting time slot in the *Reservation Calendar* you can create a reservation with a default duration derived from the creation method used or by interactively defining the duration of the reservation.

Method I – To create a reservation with default duration of 1 hour:

- In the *Reservation Calendar* toolbar, click the **New Reservation** ( ) button to create a reservation of 1 hour duration.

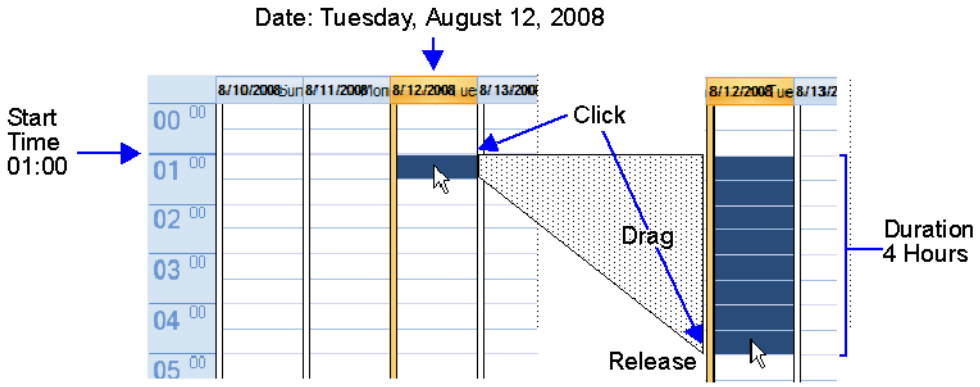
Method II – To create a reservation with default duration of ½ hour:

- Right-click and select **New Reservation** to create a reservation of ½ hour default duration.

Method III – To interactively define the duration:

1. In the calendar, click & drag to expand the time slot to select the required *Date*, *Start Time and Duration* for the reservation.
2. In the *Reservation Calendar* toolbar, click the **New Reservation** ( ) button or right-click and select **New Reservation.**

   Example: The following click & drag sequence would select a reservation for *Tuesday, August 12, 2008*, starting at *01:00* with a duration of *4* hours.



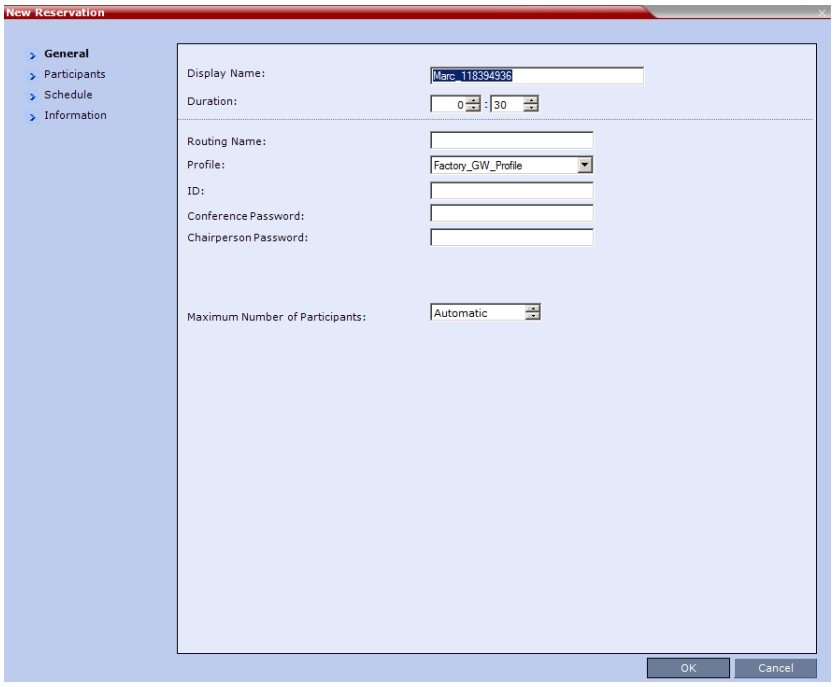
   The duration of reservations created by any of the above methods can be modified in the *Scheduler* tab of the *New Reservation* dialog box.

To create a new reservation:

1. Open the *Reservation Calendar*.
2. Select a starting time slot.

3 Create the reservation using one of the three methods described above.

The **New Reservation – General** tab dialog box opens.

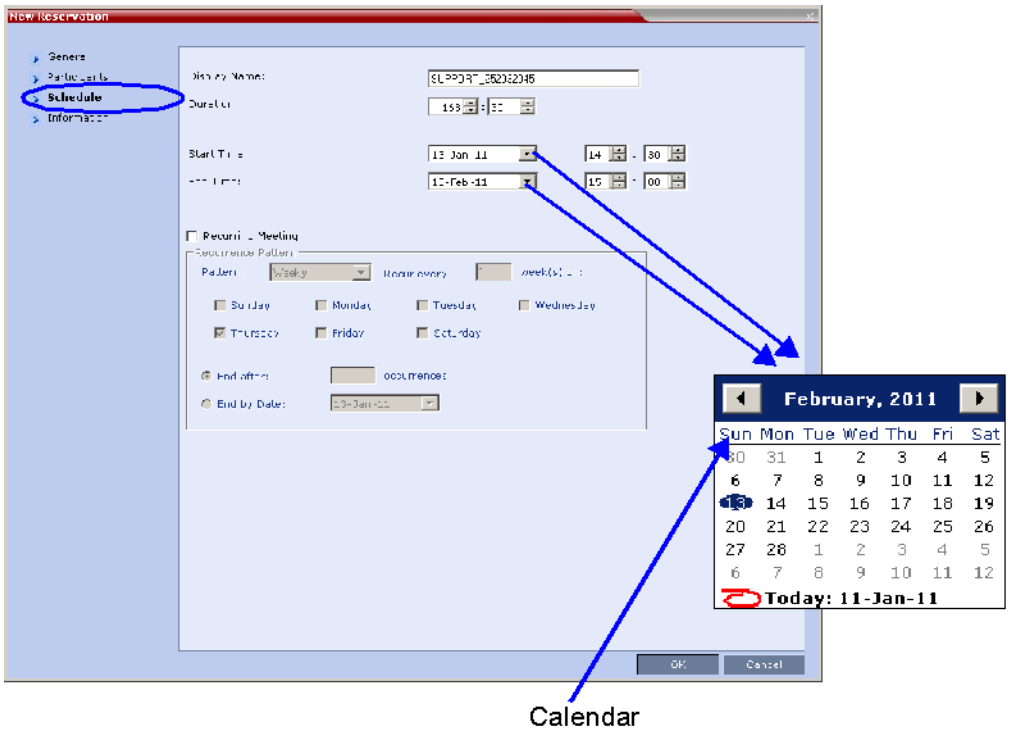


All the fields are the same as for the **New Conference – General** dialog box, described in the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, General Tab.

New Reservation – Reserved Resources

When a *Conference Profile* is assigned to a Meeting Room or a Reservation, the Profile's parameters are not embedded in the Reservation, and are taken from the Profile when the reservation becomes an ongoing conference. Therefore, any changes to the Profile parameters between the time the Reservation or Meeting Room was created and the time that it is activated (and becomes an ongoing conference) will be applied to the conference.
If the user wants to save the current parameters, a different Profile with these parameters must be assigned, or a different Profile with the new parameters must be created.

4 Click the **Schedule** tab.



5 Adjust the new reservation's schedule by modifying the fields as described in the table below.

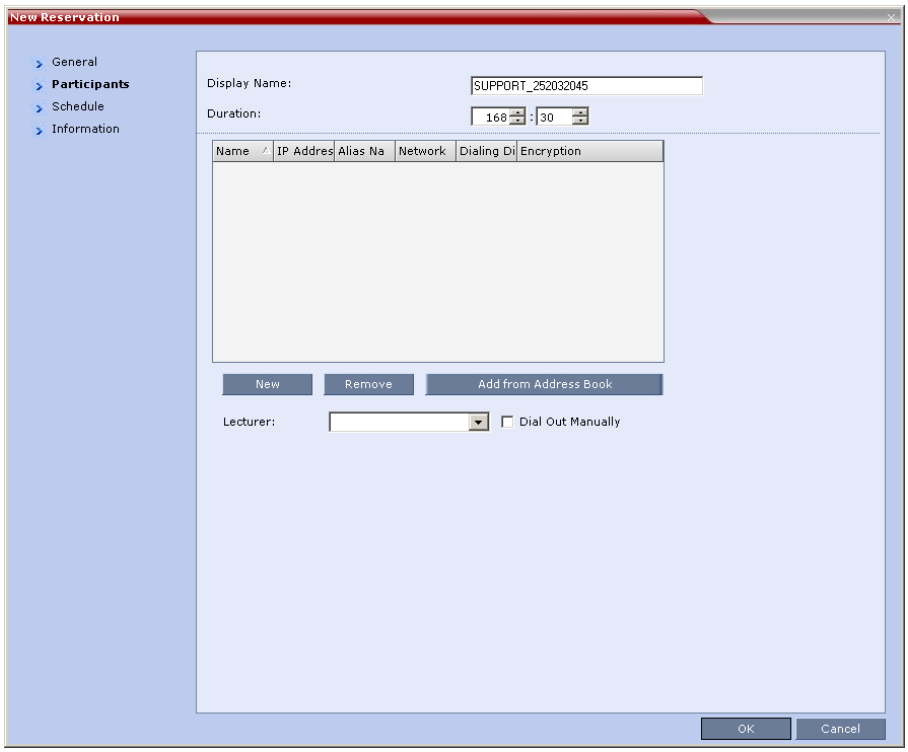
New Reservation – Schedule Tab

Field	Description
Start Time	Select the Start Time of the Reservation.
End Time	Select the End Time of the Reservation.

- The Start/End Times of the Reservation are initially taken from the time slot selected in the Reservation Calendar.
- The Start/End Times can be adjusted by typing in the hours and minutes fields or by clicking the arrow buttons.
- The Start/End dates can be adjusted by typing in the date field or by clicking the arrow buttons or using the calendar.
- The start time of all the reservations can be manually adjusted in one operation. For more information see Adjusting the Start Times of all Reservations.
- End Time settings are initially calculated as Start Time + Duration. End Time settings are recalculated if Start Time settings are changed.
- Changes to End Time settings do not affect Start Time settings. However, the Duration of the Reservation is recalculated.

Field	Description	
Recurring Meeting	Select this option to set up a Recurring Reservation - a series of Reservations to be repeated on a regular basis. To create a recurring reservation, you must define a time period and a recurrence pattern of how often the Reservation should occur: *Daily*, *Weekly* or *Monthly*.	
Recurrence Pattern	Daily	If *Daily* is selected, the system automatically selects all the days of the week. To de-select days (for example, weekends) clear their check boxes.
	Weekly	If *Weekly* is selected, the system automatically selects the day of the week for the Reservation from the day selected in the Reservation Calendar. You can also define the recurrence interval in weeks. For example, if you want the reservation to occur every second week, enter 2 in the *Recur every _ week(s)* field. To define a twice-weekly recurring Reservation, select the check box of the additional day of the week on which the Reservation is to be scheduled and set the recurrence interval to 1.
	Monthly	If *Monthly* is selected, the system automatically selects the day of the month as selected in the Reservation Calendar. You are required to choose a recurrence pattern: • **Day (1-31) of every (1-12) month(s**) - Repeats a conference on a specified day of the month at a specified monthly interval. For example, if the first Reservation is scheduled for the 6th day of the current month and the monthly interval is set to 1, the monthly Reservation will occur on the 6th day of each of the following months. • **The (first, second,...,last) (Sun-Sat) of x month(s)** - Repeats a Reservation in a particular week, on a specified day of the week at the specified monthly interval. For example, a recurrent meeting on the third Monday every second month.
A series of Reservations can be set to end after a specified number of occurrences or by a specific date. Select one of the following methods of terminating the series of Reservations:		
End After	**End After: x Occurrences** - Ends a recurring series of Reservations after a specific number (x) of occurrences. Default: 1 (Leaving the field blank defaults to 1 occurrence.)	
End by Date	**End By Date: mm/dd/yyyy** - Specifies a date for the last occurrence of the recurring series of Reservations. The End By Date value can be adjusted by typing in the date field or by clicking the arrow button and using the calendar utility. Default: Current date.	

6 Click the **Participants** tab.



The fields are the same as for the **New Conference – Participants** dialog box, described in the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Participants Tab.

Participant properties are embedded in the conferencing entity and therefore, if the participant properties are modified in the *Address Book* (or *Meeting Rooms*) after the Reservation has been created they are not applied to the participant when the Reservation is activated.

7 **Optional.** Add participants from the *Participants Address Book*.

For more information see Meeting Rooms and the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guidee*, To add participants from the Address Book: .

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 instead of the IP address. For more information see Substituting E.164 Number in Dial String .

8 **Optional.** Add information to the reservation.

Information entered in the *Information* tab is written to the *Call Detail Record (CDR)* when the reservation is activated. Changes made to this information before it becomes an ongoing conference will be saved to the CDR.

For more information see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Information Tab.

9 Click **OK**.
The *New Reservation* is created and is displayed in the *Reservation Calendar*.

If you create a recurring reservation all occurrences have the same ID.The series number (`_0000n`) of each reservation is appended to its *Display Name*.

Example:

Conference Template name: `Sales`

Display Name for single scheduled occurrence:`Sales`

If 3 recurrences of the reservation are created:

Display Name for occurrence 1: **`Sales_00001`**

Display Name for occurrence 2: **`Sales_00002`**

Display Name for occurrence 3: **`Sales_00003`**

Managing Reservations

Reservations can be accessed and managed via all the views of the *Reservations List*.

Guidelines

- The *Recurrence Pattern* fields in the *Schedule* tab that are used to create multiple occurrences of a *Reservation* are only displayed when the *Reservation* and its multiple occurrences are initially created.
- As with single occurrence *Reservations*, only the *Duration*, *Start Time* and *End Time* parameters of multiple occurrence reservations can be modified after the *Reservation* has been created.
- A single occurrence *Reservation* cannot be modified to become a multiple occurrence reservation.
- *Reservations* can only be modified one at a time and not as a group.
- If *Reservations* were created as a recurring series, the system gives the option to delete them individually, or all as series.

Viewing and Modifying Reservations

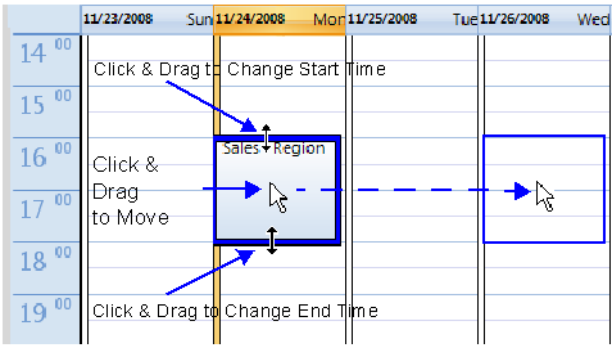
Reservations can be viewed and modified by using the *Week* and *Day* views of the *Reservations Calendar* or by using the *Reservation Properties* dialog box.

Using the Week and Day views of the Reservations Calendar

In the *Week* and *Day* views each *Reservation* is represented by a shaded square on the *Reservation Calendar*. Clicking on a *Reservation* selects the *Reservation*. A dark blue border is displayed around the edges of the *Reservation* indicating that it has been selected.

The *Start Time* of the *Reservation* is represented by the top edge of the square while the *End Time* is represented by the bottom edge.

The cursor changes to a vertical double arrow (↕) when it is moved over the top and bottom sides of the square.



To move the Reservation to another time slot:

1. Select the *Reservation*.
2. Hold the mouse button down and drag the *Reservation* to the desired time slot.
3. Release the mouse button.

To change the Reservation's Start time:

1. Select the *Reservation*.
2. Move the mouse over the top edge of the *Reservation's* square.
3. When the cursor changes to a vertical double arrow (↕) hold the mouse button down and drag the edge to the desired *Start Time*.
4. Release the mouse button.

To change the Reservation's End time:

1. Select the *Reservation*.
2. Move the mouse over the bottom edge of the *Reservation's* square.
3. When the cursor changes to a vertical double arrow (↕) hold the mouse button down and drag the edge to the desired *End Time*.
4. Release the mouse button.

To View or Modify Reservations using the Reservation Properties dialog box:

1. In the *Reservations List*, navigate to the reservation (or its recurrences) you want to view, using the **Show Day**, **Show Week**, **Today**, **Back**, **Next** or **List** buttons.
2. Double-click, or right-click and select **Reservation Properties**, to select the reservation to be viewed or modified.

   The *Reservation Properties – General* dialog box opens.
3. Select the tab(s) of the properties you want to view or modify.
4. **Optional.** Modify the *Reservation Properties*.

5 Click **OK**.
The dialog box closes and modifications (if any) are saved.

Adjusting the Start Times of all Reservations

When utilizing GMT offset (for example, *Daylight Saving Time* change), the start time of the reoccurring reservations scheduled before the Collaboration Server time change are not updated accordingly (although their start times appear correctly in the *Reservations* list, when checking the reservation properties the start time is incorrect).

Following the Collaboration Server time change, the start time of all reoccurring reservations must be manually adjusted in one operation.

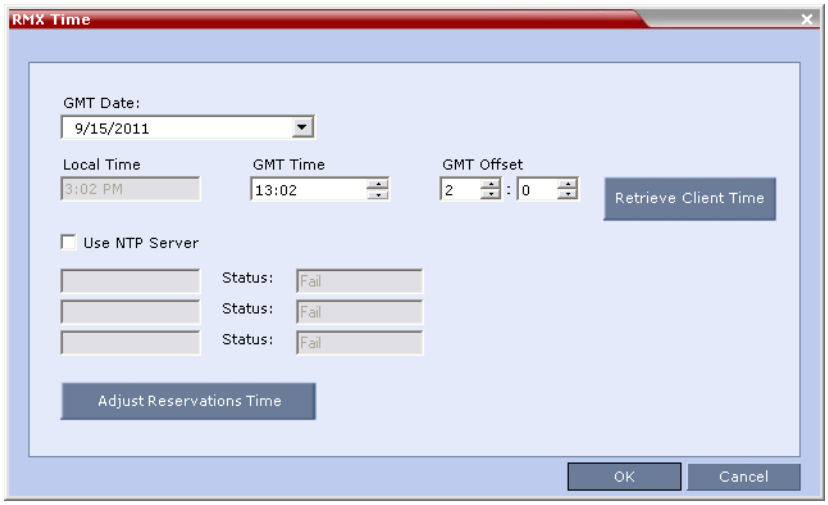
Using this option, the start times of **all** reservations currently scheduled on the Collaboration Server are adjusted with the same offset.

To adjust the reoccurring reservations start time after the GMT Offset has been changed for Daylight Saving Time (DST) or a physical move:.

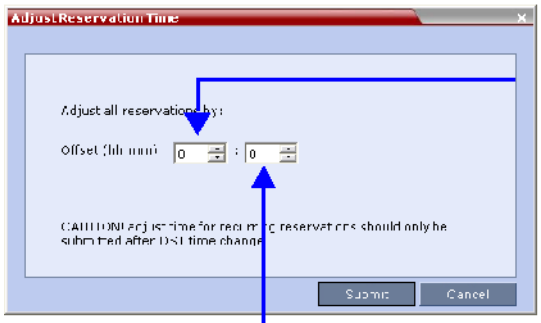
Adjustment of *Reservation Time* should only be performed after adjustment of *Collaboration Server Time* is completed as a separate procedure.

1 On the Collaboration Server menu, click **Setup > RMX Time**.
The **RMX Time** dialog box opens.

2 Click the **Adjust Reservations Time** button.



The **Adjust Reservations Time** dialog box opens.



Click the arrows to adjust the start time by hours. Range is between 12 hours and -12 hours

A positive value indicates adding to the start time

Click the arrows to adjust the start time by minutes. Range is between 45 minutes and -45 minutes.

3 Click the arrows of the *Offset - Hours* box to indicate the number of hours to add or subtract from the current start time; a positive value indicates adding time, while minus (-) indicates subtracting time.

4 Click the arrows of the *Offset - minutes* box to indicate the number of minutes to add or subtract from the current start time of the reservations. Increments or decrements are by 15 minutes.

For example, to subtract 30 minutes from the start time of all the reservation, enter 0 in the *hours* box, and -30 in the *minutes* box.

To add one hour and 30 minutes to the start time, enter 1 in the hours box and 30 in the minutes box.

5 Click the **Adjust** button to apply the change to all the reoccurring reservations currently scheduled on the Collaboration Server.

When adjusting the start time of 1000 - 2000 reservations, an “Internal communication error” message may appear. Ignore this message as the process completes successfully.

Deleting Reservations

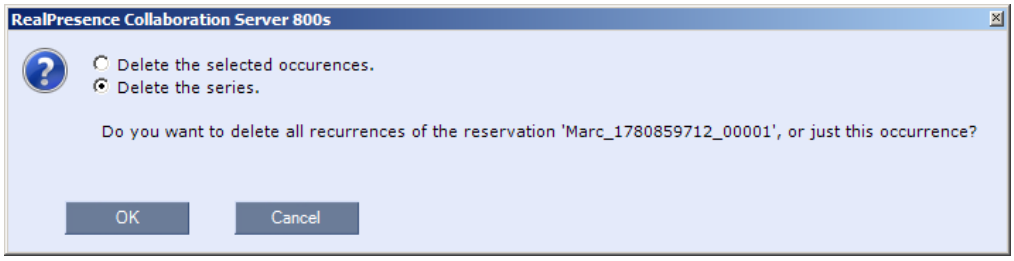
To delete a single reservation:

1. In the *Reservations List*, navigate to the reservation you want to delete, using the **Show Day**, **Show Week**, **Today**, **Back**, **Next** or **List** buttons.
2. Click to select the reservation to be deleted.
3. Click the **Delete Reservation** *(*  *)* button.
   or
   Place the mouse pointer within the *Reservation* block, right-click and select **Delete Reservation**.
4. Click **OK** in the confirmation dialog box.
   The *Reservation* is deleted.

To delete all recurrences of a reservation:

1. In the *Reservations List*, navigate to the *Reservation* or any of its recurrences, using the **Show Day**, **Show Week**, **Today**, **Back**, **Next** or **List** buttons.
2. Click the **Delete Reservation** *(*  *)* button.
   or
   Place the mouse pointer within the *Reservation* or any of its recurrences, right-click and select **Delete Reservation**.

   A confirmation dialog box is displayed.



3. Select **Delete the series.**
4. Click **OK.**

   All occurrences of the *Reservation* are deleted.

Searching for Reservations using Quick Search

Quick Search is available only in *List View.* It enables you to search for *Reservations* by *Display Name*.

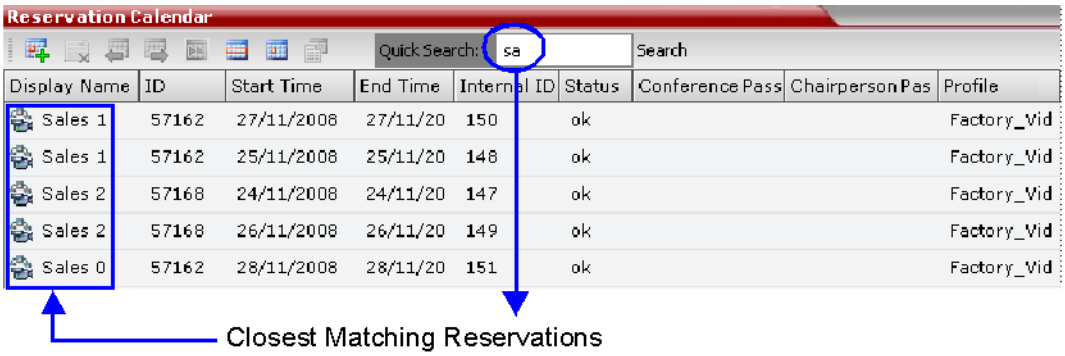
To search for reservations:

1. In the *Reservation Calendar* toolbar, click in the *Quick Search* field.

   The field clears and a cursor is displayed indicating that the field is active.

2 Type all or part of the reservation's *Display Name* into the field and click **Search**.

The closest matching *Reservation* entries are displayed.



3 **Optional.** Double- click the *Reservation's* entry in the list to open the *Reservations Properties* dialog box to view or modify the *Reservation*.

or

Right -click the *Reservation's* entry in the list and select a menu option to view, modify or delete the *Reservation.*

To clear the search and display all reservations:

1 Clear the *Quick Search* field.

2 Click **Search.**

All *Reservations* are displayed.

Operator Assistance & Participant Move

User 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.

In the *RealPresence CloudAxis Solution*, the Conference Profiles are defined in the RealPresence DMA system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

Operator conferences and participant move are supported in AVC CP Conferencing Mode only.

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.
- 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* cannot be defined as a Reservation.
- 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*.
- Participants in Telepresence conferences cannot be moved from their conference, but an operator can join their conference and help them if assistance is required.
- Moving participants from/to an *Operator conference* follows the same guidelines as moving participants between conferences. For move guidelines, see Move Guidelines.
- 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:

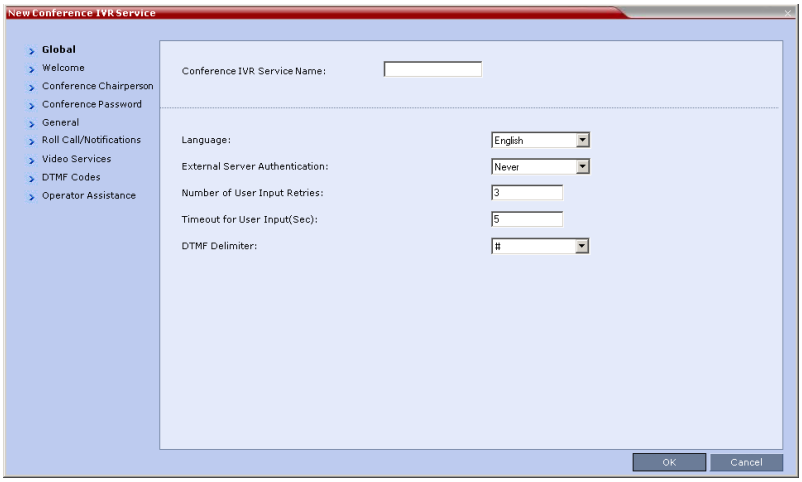
- IVR Service (Entry Queue and Conference) in which Operator Assistance options are enabled.
- A Conference Profile with the *Operator Conference* option enabled.
- An active Operator conference with a connected Operator participant.

Defining a Conference IVR Service with Operator Assistance Options

In the **RMX 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.



2 Enter the Conference IVR Service **Name**.

3 Define the *Conference IVR Service - Global* parameters. For more information, see Conference IVR Service Properties - Global Parameters.

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.

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 New Conference IVR Service Properties - Conference Chairperson Options and Messages.

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 New Conference IVR Service Properties - Conference Password Parameters.

10 Select the various audio messages that will be played in each case. For more information, see For more information, see New Conference IVR Service Properties - Conference Password Parameters.

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 Conference IVR Service Properties - General Voice Messages.

13 Click the **Roll Call/Notifications** tab.
The **New Conference IVR Service - Roll Call** dialog box opens.

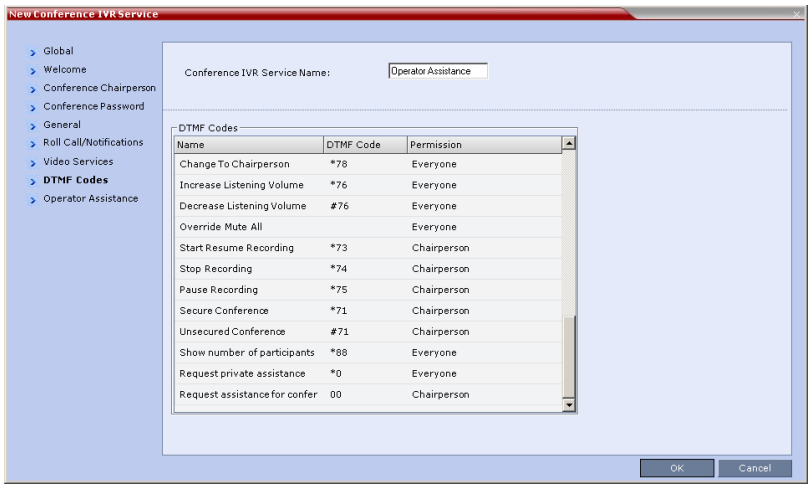
14 Enable the Roll Call feature and assign the appropriate audio file to each message type. For more information, see Conference IVR Service Properties - Roll Call Messages.

15 Click the **Video Services tab**.
The *New Conference IVR Service* - ***Video Services*** **dialog box opens.**

16 Define the **Video Services** parameters. For more information, see New Conference IVR Service Properties - Video Services Parameters.

17 Click the **DTMF Codes** *tab*.

The **New Conference IVR Service - DTMF Codes** dialog box opens.



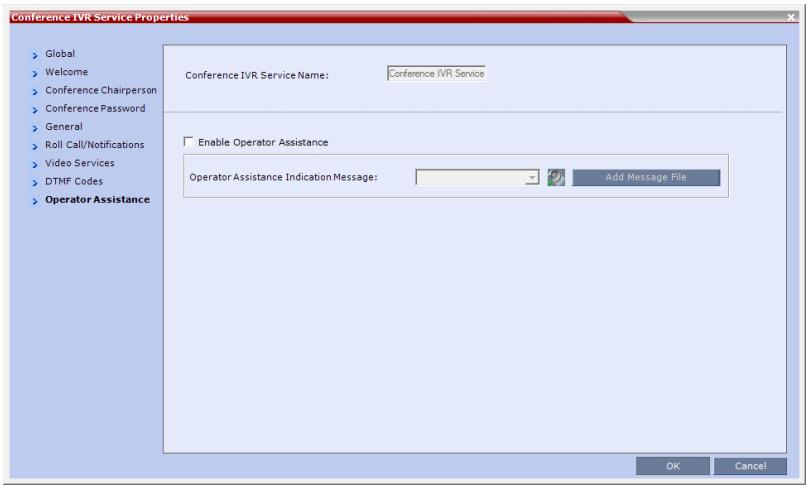
The default DTMF codes for the various functions that can be performed during the conference by all participants or by the chairperson are listed. For the full list of the available DTMF codes, see New Conference IVR Service Properties - DTMF Codes.

18 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.

19 Click the **Operator Assistance** tab.

The **Operator Assistance** dialog box opens.



20 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.

21 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.

22 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 *RMX 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 Entry Queue IVR Service Properties - Global Parameters.

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. *T*his 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.

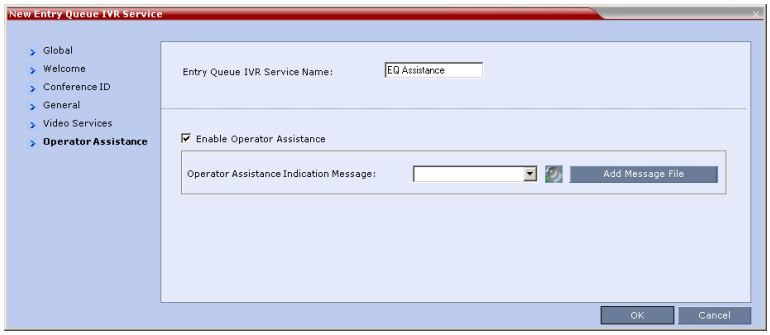
8 Select the required voice messages. For more information, see Entry Queue IVR Service Properties - Conference ID.

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.



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

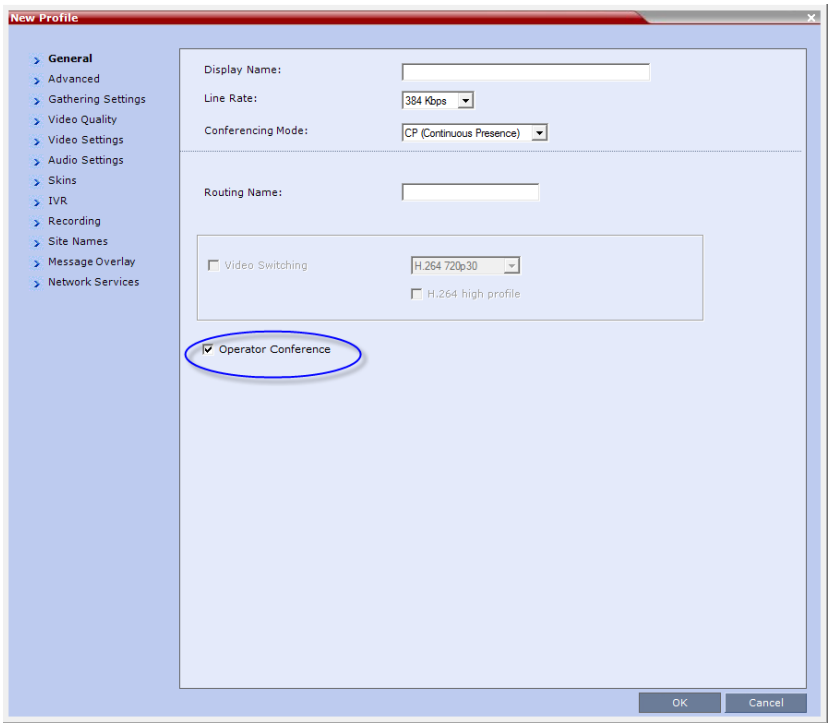
In the *RealPresence CloudAxis Solution*, the Conference Profiles are defined in the RealPresence DMA system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

1 In the **RMX Management** pane, click **Conference Profiles**.

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

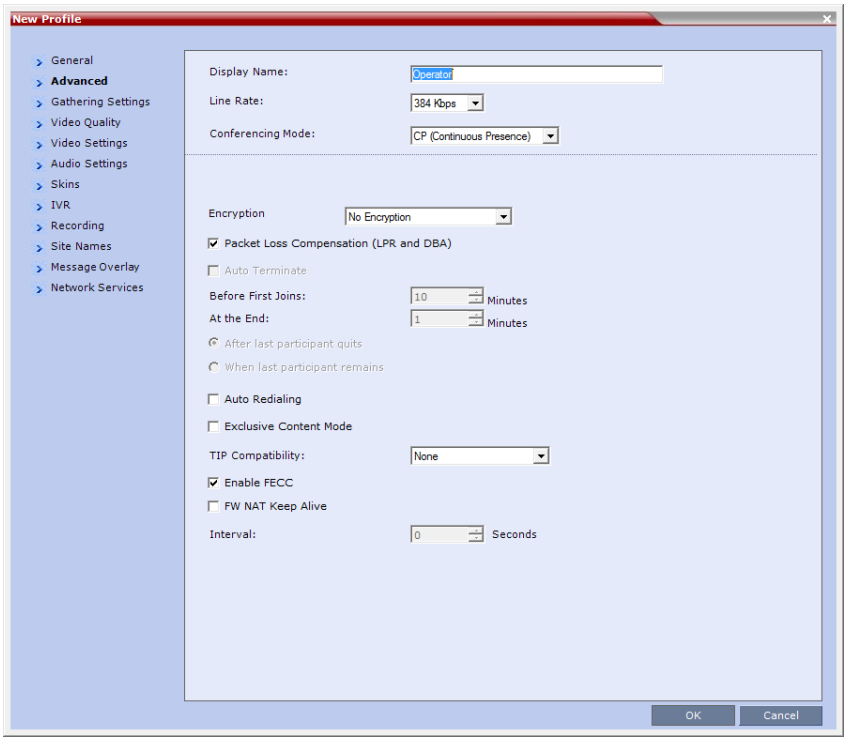
3 Define the Profile name and, if required, the Profile general parameters.
For more details, see New AVC CP Profile - General Parameters.

4 Click the **Operator Conference** check box.



5 Click the **Advanced** tab.

The **New Profile – Advanced** dialog box opens.



6 Define the Profile *Advanced* parameters. For more details, see New AVC CP 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. For more details, see New AVC CP Profile - 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 New AVC CP Profile - Video Settings Parameters.

11 Define the remaining Profile parameters. For more details, see Defining AVC CP Conferencing Profiles.

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

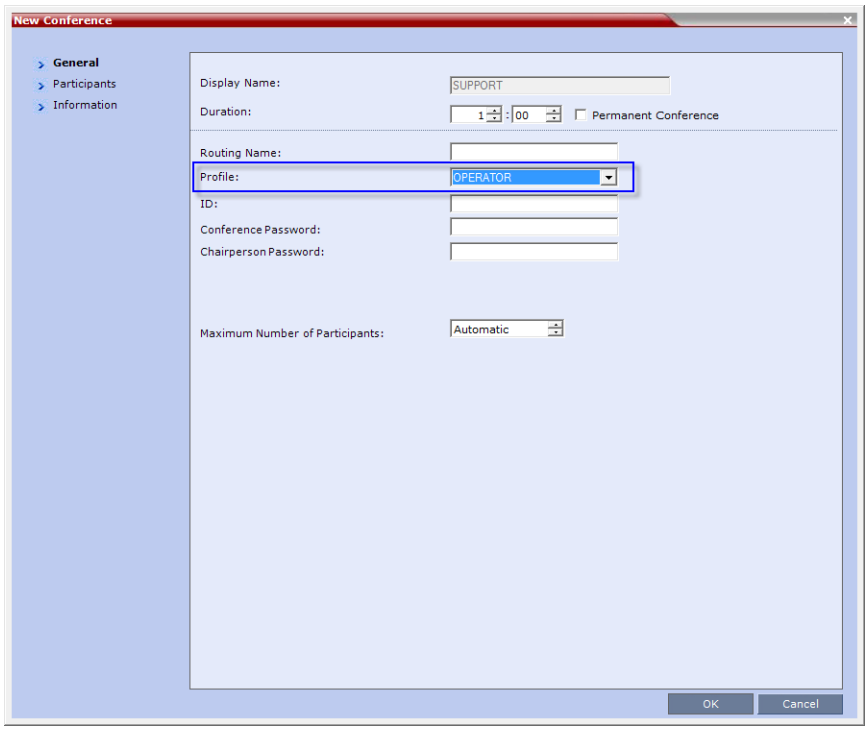
Starting an Ongoing Operator Conference

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.



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:

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.

Field	Description
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 800s / Virtual Edition Getting Started Guide*, Participants Tab.

5 **Optional**. Click the **Information** tab**.**
The **Information** dialog box opens.

6 Enter the required information. For more details, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Information Tab.

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.

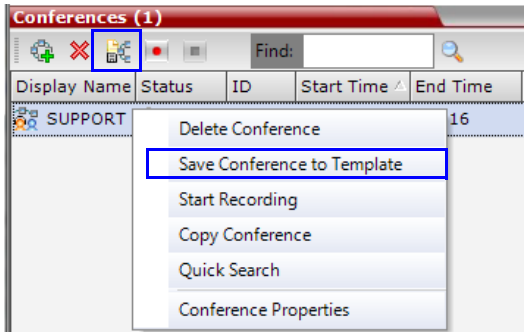
Saving an Operator Conference to a Template

The Operator conference that is ongoing can be saved as a template.

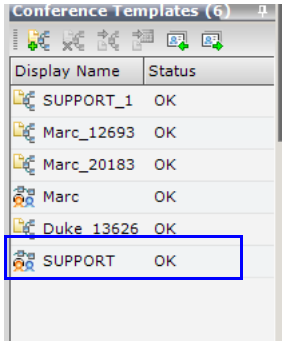
To save an ongoing Operator conference as a template:

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**.



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.



Starting an Operator Conference from a Template

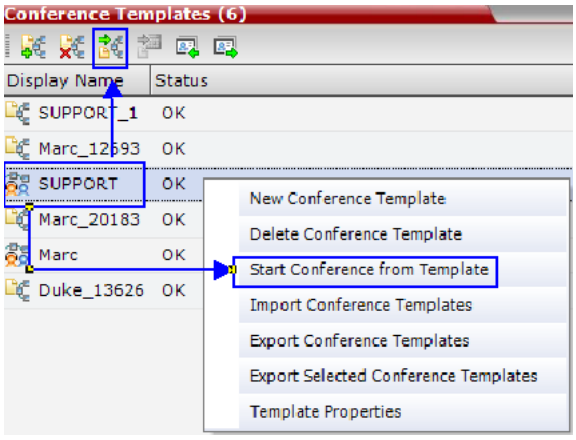
An ongoing Operator conference can be started from an Operator Template saved in the *Conference Templates* list.

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**.



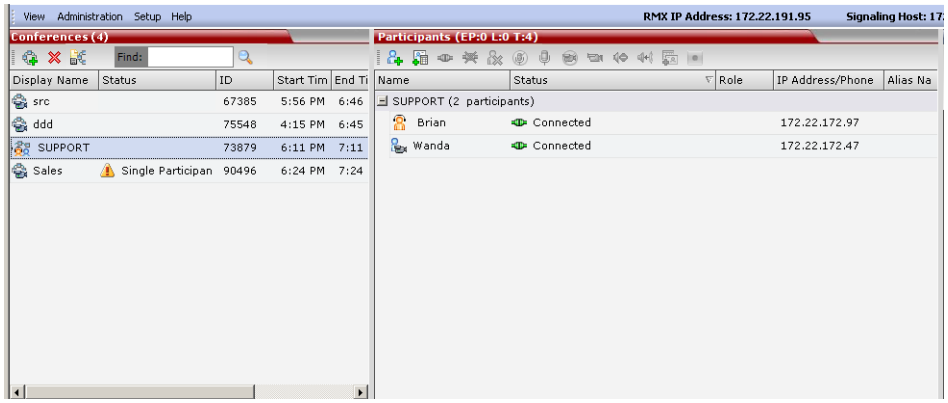
The conference is started.

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.



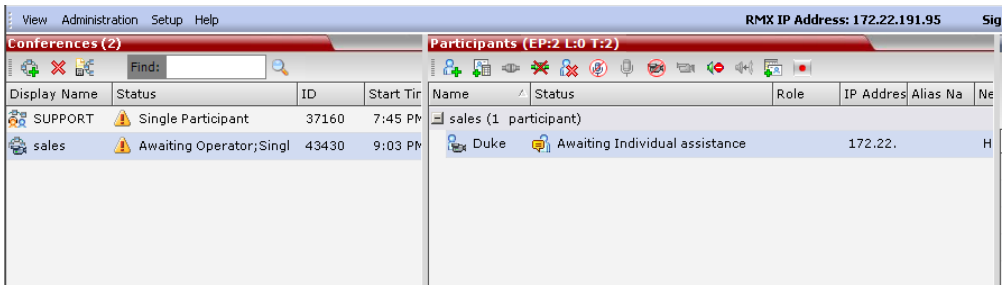
You can view the properties of the *Operator conference* by double-clicking the conference entry in the *Conferences* list or by right-clicking the conference entry and selecting **Conference Properties**. For more information, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Participant Level Monitoring.

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:



- The participant's connection *Status* changes, reflecting the help request. For more information, see Table 5-2.
- 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:

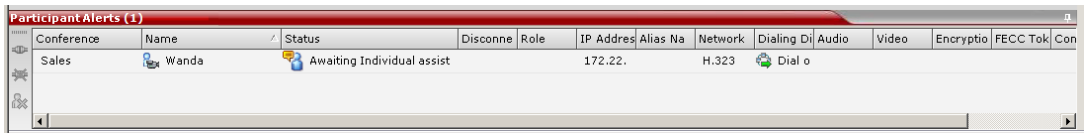
Participants List Status Column Icons and Indications

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.



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.

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 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 a Telepresence conference.
- 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 the following table:

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
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

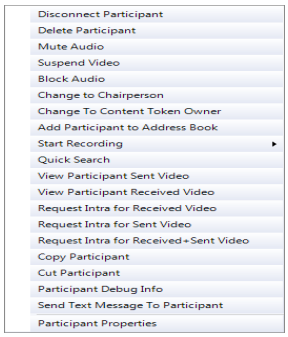
A move can be performed using the following methods:

- Using the participant right-click menu
- Using drag and drop

To move a participant from the ongoing conference using the right-click menu options:

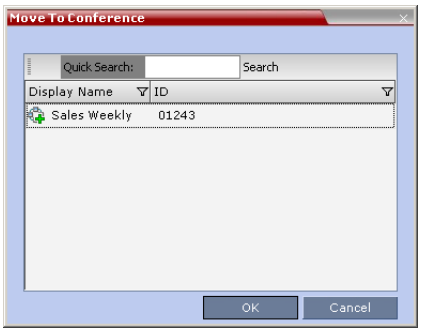
1. In the *Conferences* list, click the conference where there are participants waiting for Operator's Assistance to display the list of participants.

2 In the *Participants* list, right-click the icon of the participant to move and select one of the following options:



- **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.



- **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.

Conference Templates

In the *RealPresence CloudAxis Solution*, the conference templates are defined in the RealPresence DMA system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

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.
- Simplifies the setting up Telepresence conferences where precise participant layout and video forcing settings are crucial.

Guidelines

- The maximum number of templates is 100.A maximum of 200 participants can be saved in a *Conference Template*.

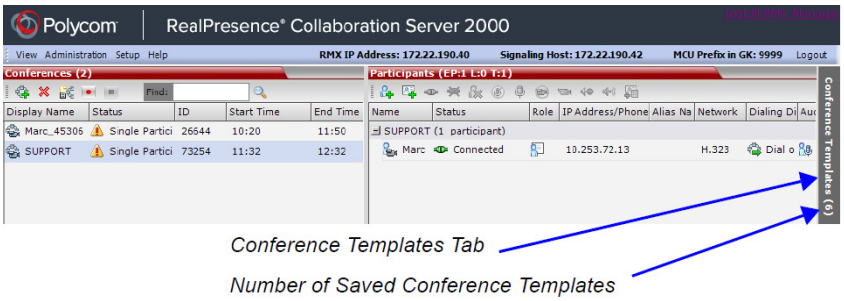
  Trying to start a Conference Template that exceeds the allowed maximum number of participants will result in participants being disconnected due to resource deficiency.
- If the Profile assigned to a conference is deleted while the conference is ongoing the conference cannot be saved as a template.
- 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.
- A Reservation that has become an ongoing conference can be saved as Conference Template.
- 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 Permanent Conference.

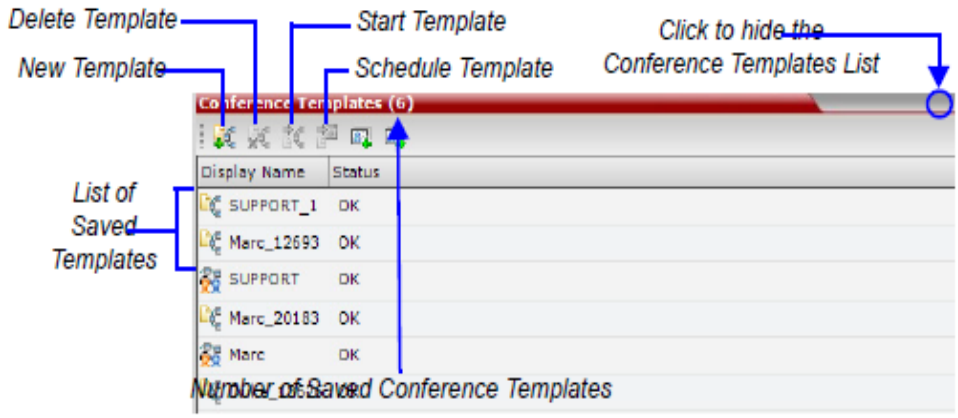
Using Conference Templates

In the *RealPresence CloudAxis Solution*, templates are used in the RealPresence DMA system component and should not be defined directly in the RealPresence Collaboration Server component.

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.



Clicking the tab opens the *Conference Templates* list.



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 800s / Virtual Edition Getting Started Guide,* Customizing the Main Screen.

Clicking the anchor pin ( ) button hides the *Conference Templates* list as a closed tab.

Toolbar Buttons

The Conference Template toolbar includes the following buttons:

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.
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 Scheduling Reservations.

The Conferences List toolbar includes the following button:

Conferences List – Toolbar Button

Button	Description
Save Conference to Template	Saves the selected ongoing conference as a Conference Template.

Creating a New Conference Template

In the *RealPresence CloudAxis Solution*, the Conference Templates are defined in the RealPresence DMA system component and should not be defined directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

There are two methods to create a Conference Template:

- From scratch - defining the conference parameters and participants
- Saving an ongoing conference as Template

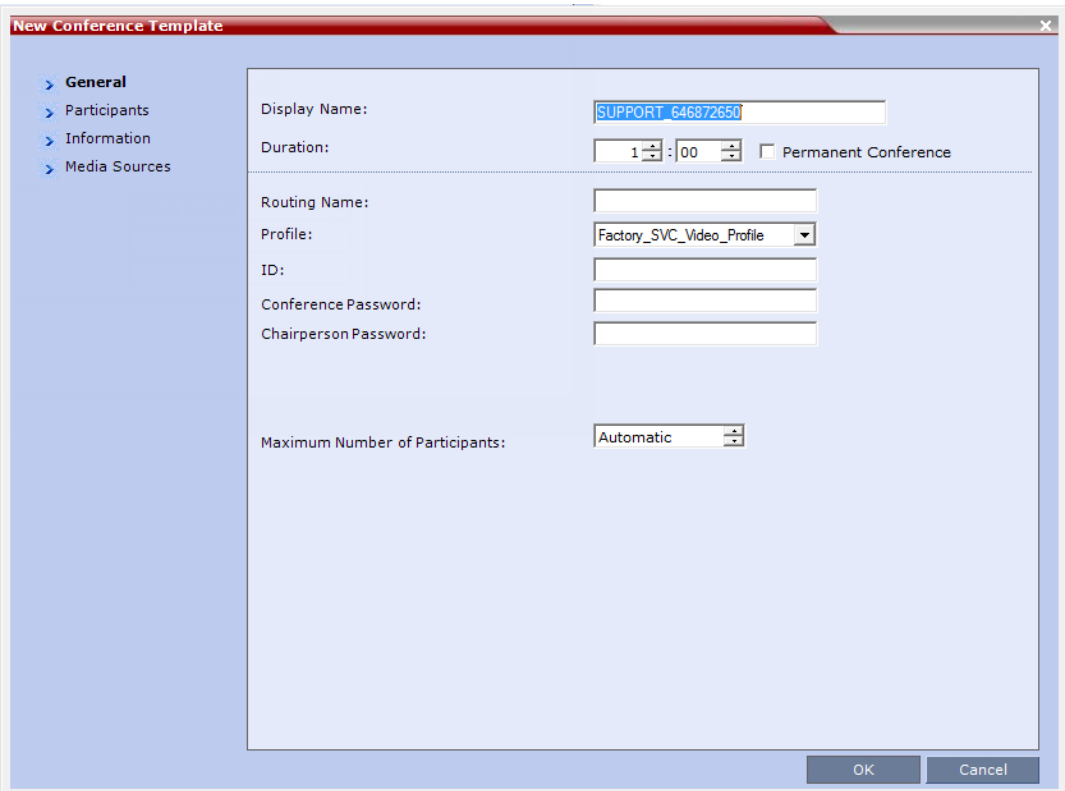
Creating a new Conference Template from Scratch

To create a new Conference Template:

1 In the Collaboration Server main screen, click the **Conference Templates** tab.

2 Click the **New Conference Template** ( ) button.

The **New Conference Template - General** dialog box opens.

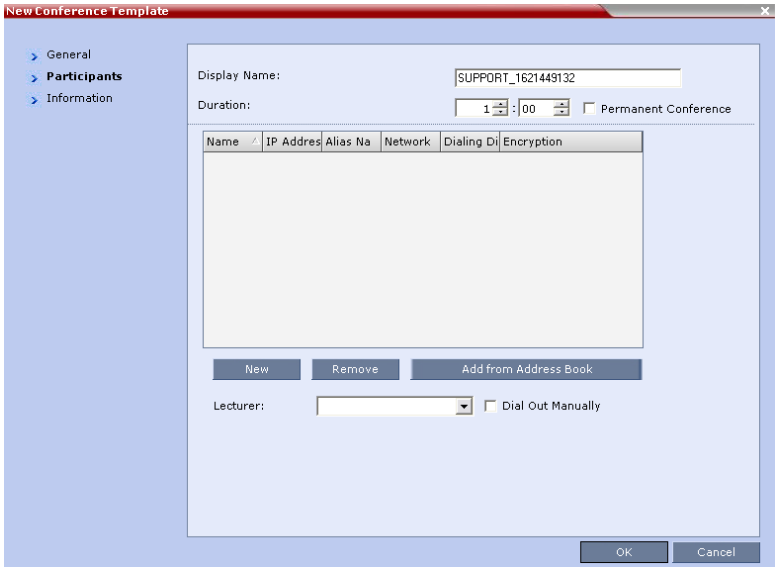


The fields of the **New Template – General** dialog box are identical to those of the **New Conference – General** dialog box. For a full description of the fields see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* General Tab.

3 Modify the fields of the **General** dialog box.

4 Click the **Participants** tab.

The **New Template – Participants** dialog box opens.



The fields of the **New Template – Participants** dialog box are the same as those of the **New Conference – Participant** dialog box.

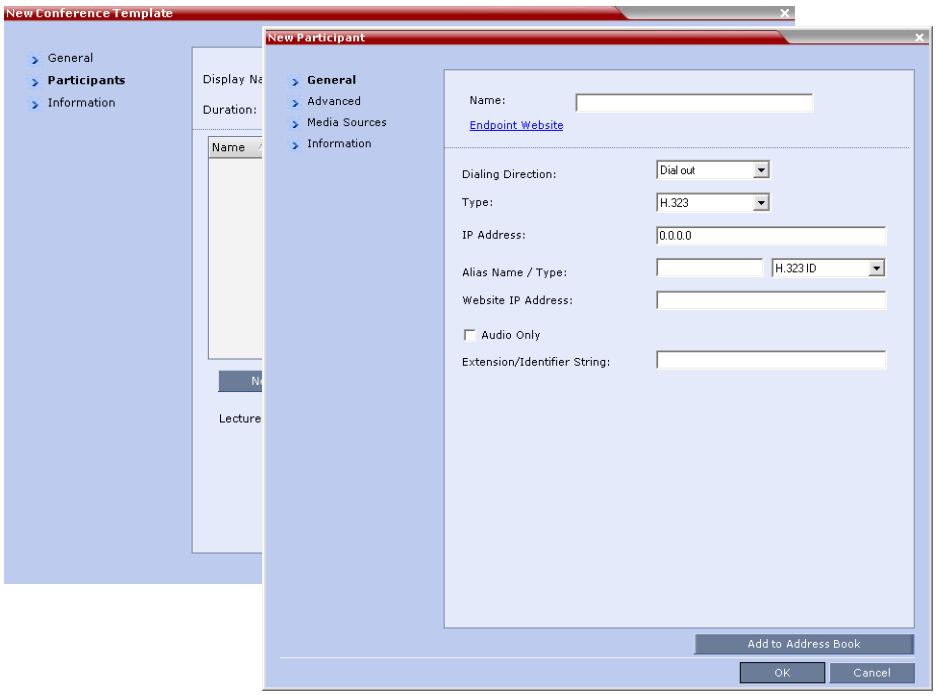
For a full description of these fields see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Participants Tab.

5 **Optional.** Add participants to the template from the Address Book.

6 Click the **New** button.

The **New Participant – General** dialog box opens.

The **New Template – Participant** dialog box remains open in the background.

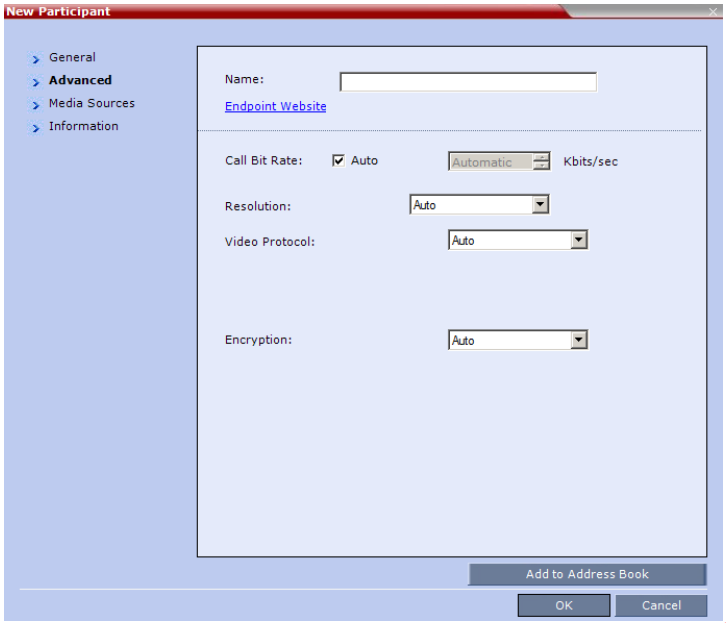


For a full description of the *General* tab fields see Adding a New participant to the Address Book Directly.

7 Modify the fields of the **General** dialog box.

8 Click the **Advanced** tab.

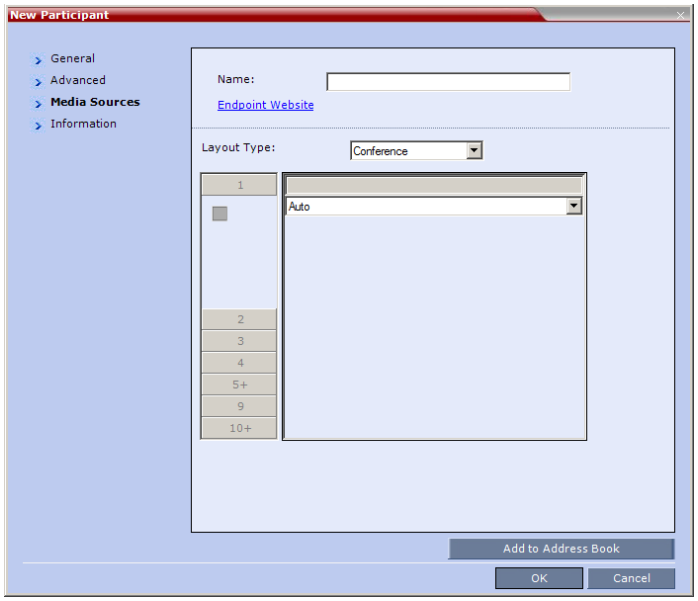
The **New Participant – Advanced** dialog box opens.



9 Modify the fields of the **Advanced** dialog box.

10 Click the **Media Sources** tab.

The **Media Sources** dialog box opens.



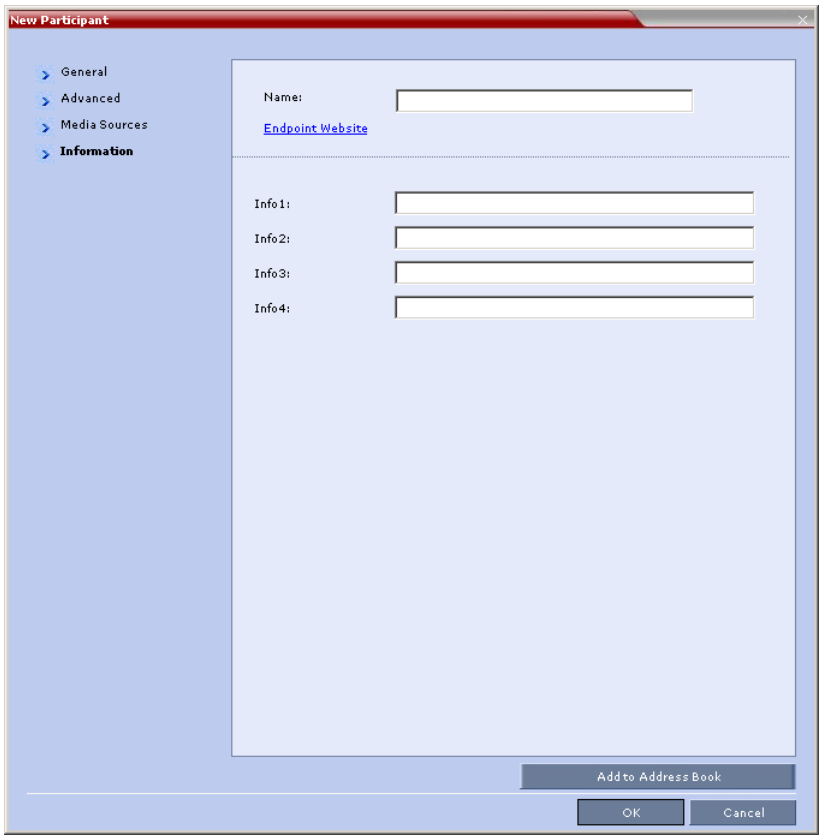
The **Media Sources** dialog box enables you to set up and save Personal Layout and Video Forcing settings for each participant. This is especially important when setting up Telepresence conferences.

For a full description of Personal Layout and Video Forcing settings see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Changing the Video Layout of a Conference (AVC-Based CP and Mixed CP and SVC Conferences) and Video Forcing (AVC-Based CP and Mixed CP and SVC Conferences).

11 Modify the Personal Layout and Video Forcing settings for the participant.

12 **Optional.** Click the **Information** tab.

The **New Participant – Information** dialog box opens.



For a full description of the Information fields see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* Information Tab.

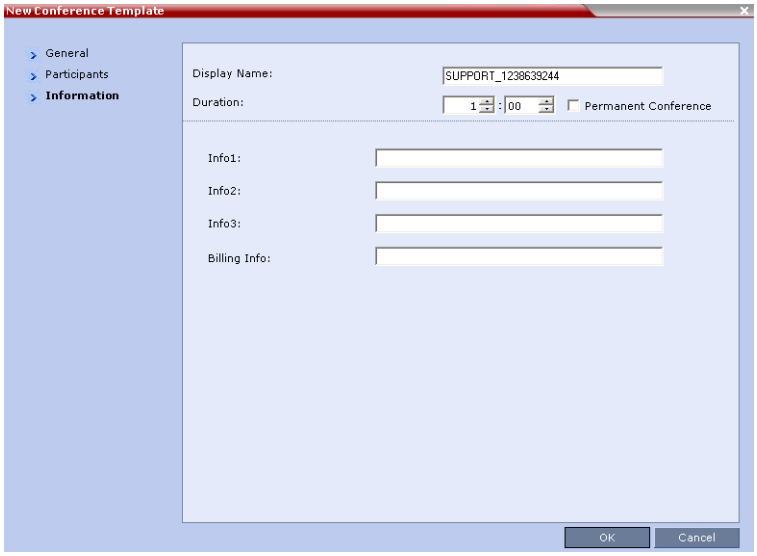
13 Click the **OK** button.

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 6).

14 **Optional.** In the **New Conference Template** dialog box, click the **Information** tab.

The **New Conference Template – Information** dialog box opens.



For a full description of the **Information** fields see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* Information Tab.

15 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.

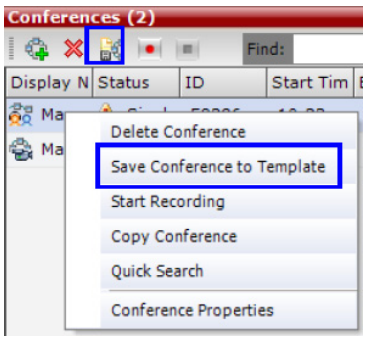
To save an ongoing or AVC-based CP Operator Conference as a template:

1 In the *Conferences List*, select the conference or *Operator Conference* to be saved as a Template.

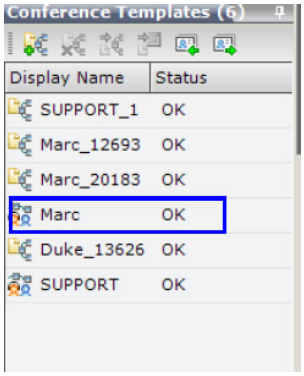
2 Click the **Save Conference to Template** *(*  *)* button.

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.



Starting an Ongoing Conference From a Template

Conference Templates saved from an ongoing conference does not include *Message Overlay* text messages.

In the *RealPresence CloudAxis Solution*, Conferences should be started in the RealPresence DMA system component and should not be started directly in the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition component.

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.

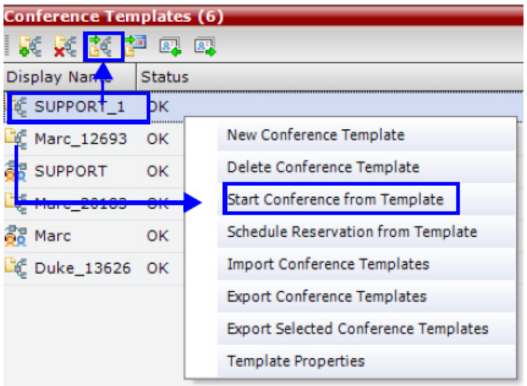
To start an ongoing conference from a Template:

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**.



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:

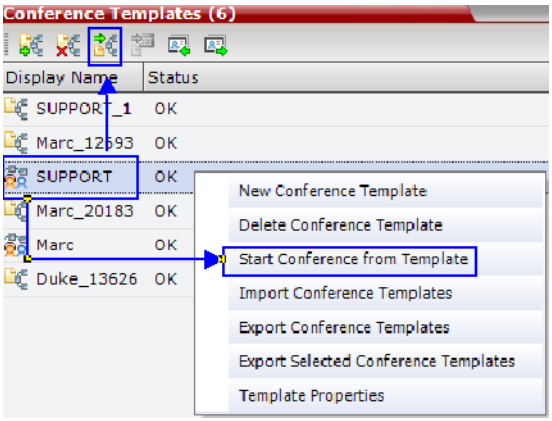
1 In the **Conference Templates** list, select the Operator Template to start as an ongoing Operator conference.

- 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**.



The conference is started.

The name of the ongoing conference in the *Conferences* list is taken from the Conference Template *Display Name*.

Scheduling a Reservation From a Conference Template

A Conference Template can be used to schedule a single or recurring Reservation.

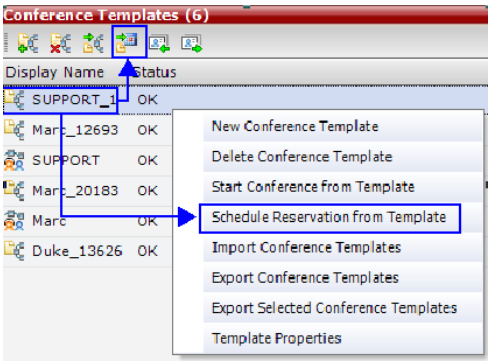
To schedule a Reservation from a Conference Template:

1 In the **Conference Templates** list, select the Conference Template you want to schedule as a Reservation.

2 Click the **Schedule Reservation from Template** ( ) button.

or

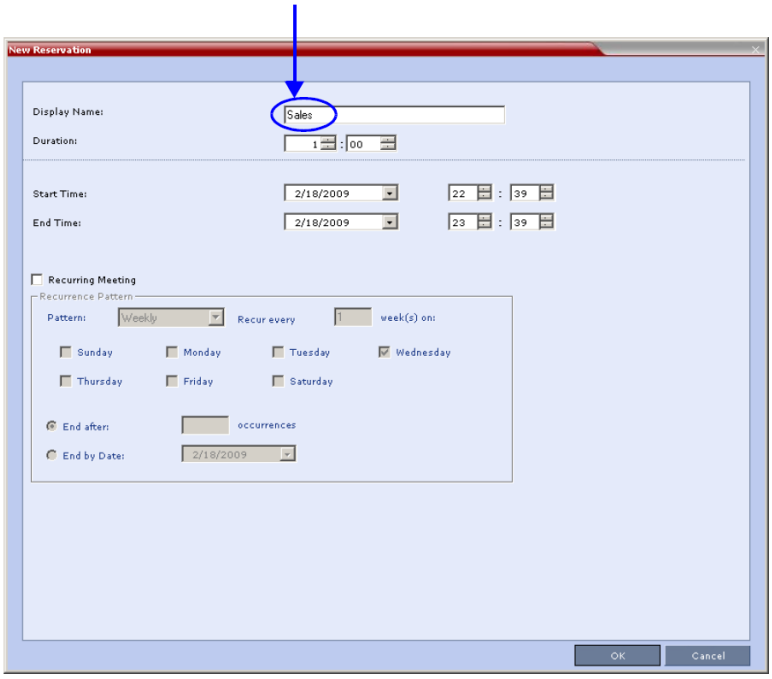
Right-click and select **Schedule Reservation from Template**.



The **Reservation Properties** dialog box is displayed.

The **Display Name** of the Reservation is taken from the Conference Template Display Name.

Conference Template and Reservation Name



For a full description of the **Reservation Properties** fields see Creating a New Reservation.

3 Modify the fields of the Reservation Properties.

4 Click the **OK** button.

A Reservation is created based on the Conference Template. The Reservation can be viewed and modified along with all other Reservations using the Reservations - Calendar View and Reservations List.

If you create a recurring reservation all occurrences have the same ID.The series number (`_0000n`) of each reservation is appended to its Display Name.

Example:

Conference Template name: `Sales`

Display Name for single scheduled occurrence:`Sales`

If 3 recurrences of the reservation are created:

Display Name for occurrence 1: **`Sales_00001`**

Display Name for occurrence 2: **`Sales_00002`**

Display Name for occurrence 3: **`Sales_00003`**

Deleting a Conference Template

One or several Conference Templates can be deleted at a time.

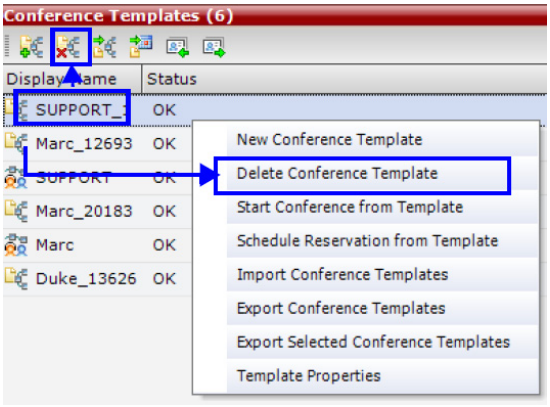
To delete Conference Templates:

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**.



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:

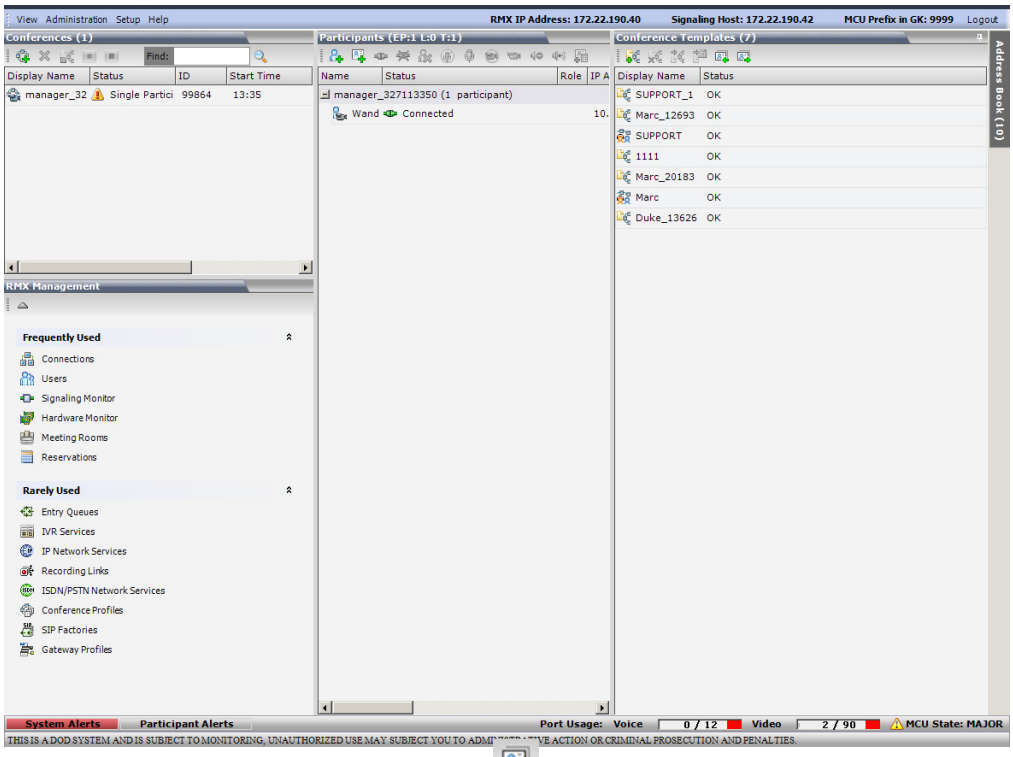
- Export all Conference Templates from an MCU
- Export selected Conference Templates

Exporting All Conference Templates from an MCU

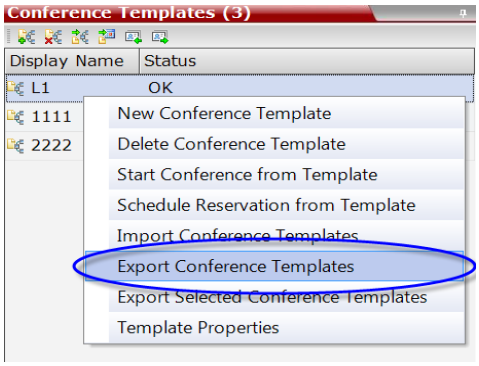
To export all Conference Templates from an MCU:

1 In the Collaboration Server Web Client main window, click the Conference Templates tab.

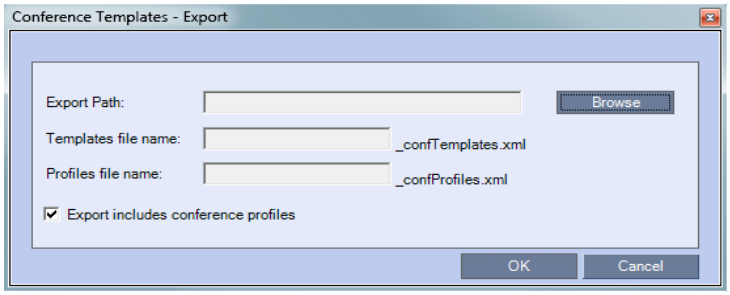
The **Conference Templates** list pane is displayed.



2 Click the **Export Conference Templates** button, or right-click the *Conference Templates* list, and then click **Export Conference Templates**.



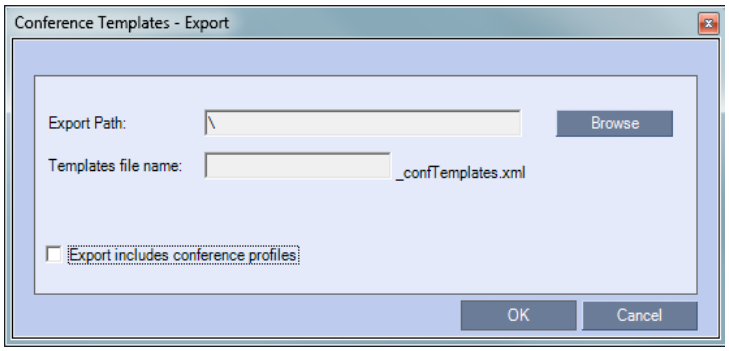
The **Conference Templates - Export** dialog box is displayed.



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.



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.

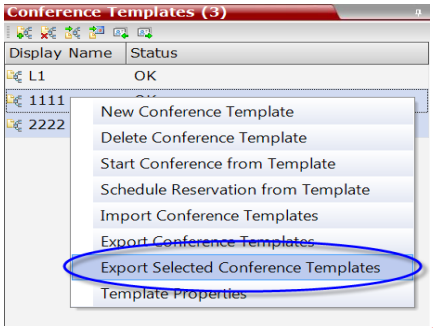
Exporting Selected Conference Templates

You can export a single Conference Template or multiple Conference Templates to other MCUs in your environment.

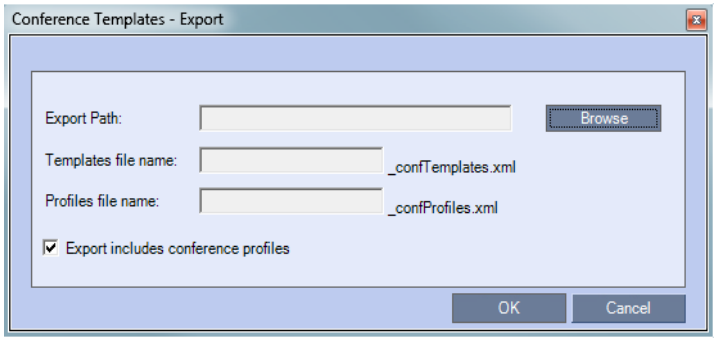
To export selected Conference Templates:

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**.



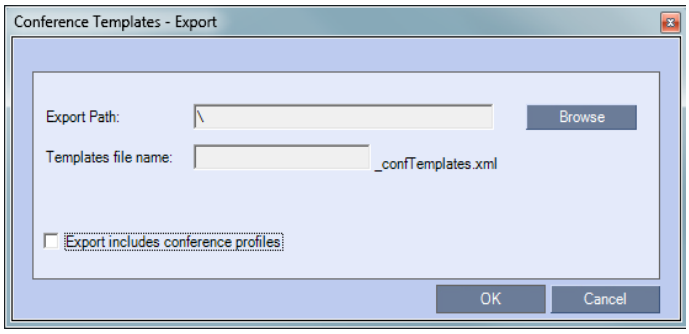
The *Conference Templates - Export* dialog box is displayed.



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.



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

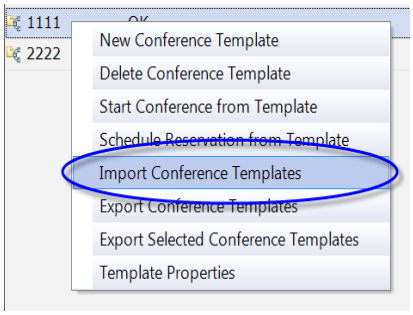
You can import Conference Templates and Conference Profiles from one MCU to multiple MCUs in your environment.

To import Conference Templates:

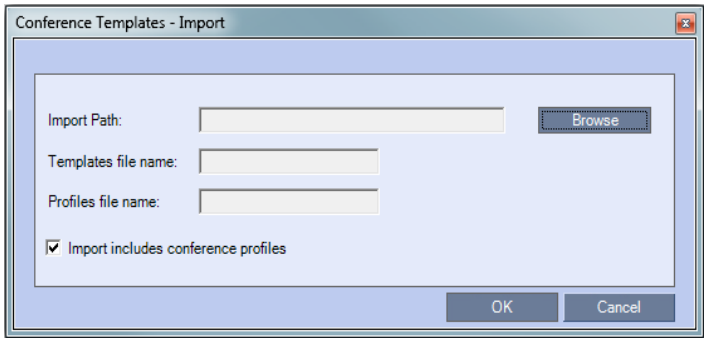
1 In the Collaboration Server Web Client main window, click the Conference Templates tab.

The Conference Templates are displayed.

2 Click the **Import Conference Templates** button or right-click the Conference Templates pane, and then click I**mport Conference Templates.**

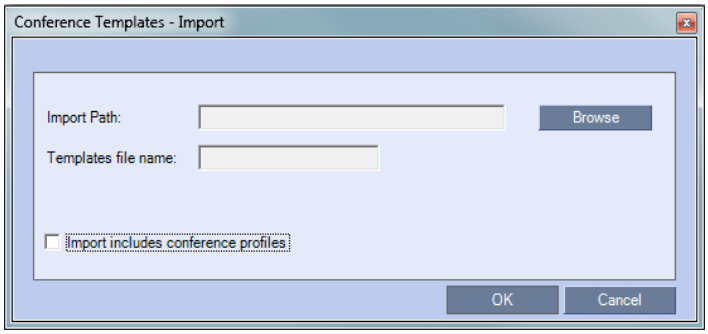


The **Conference Templates - Import** dialog box is displayed.



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.



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

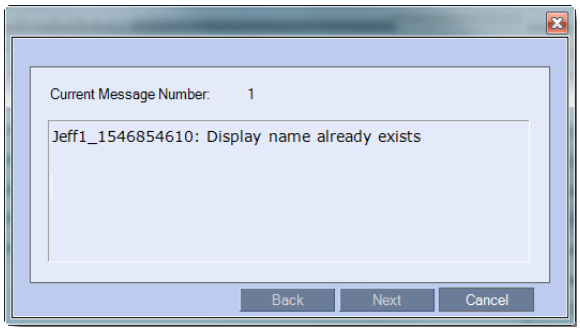
When clicking the exported templates file you want to import, the system automatically displays the appropriate files in the Templates file name field and the **Profiles file name** field (when the **Import includes conference profiles** check box is selected).

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.



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.

Polycom Conferencing for Microsoft Outlook®

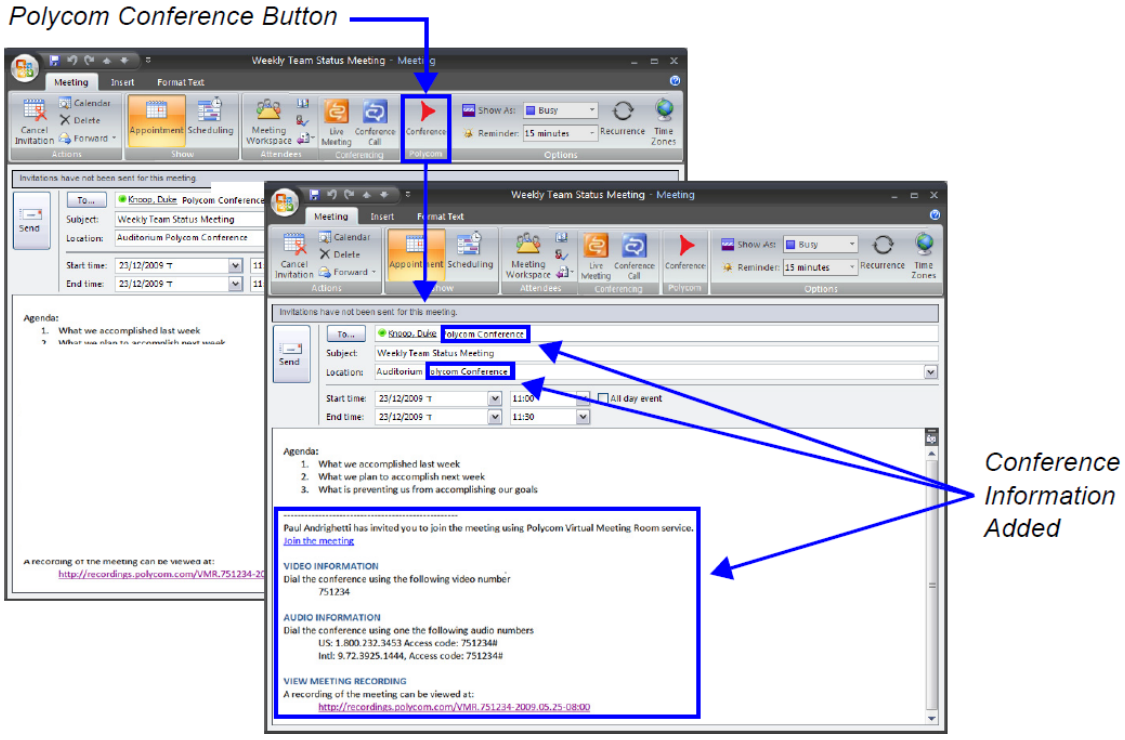
Polycom Conferencing for Microsoft Outlook is supported in AVC CP Conferencing Mode only

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.



A *Gathering Slide* is displayed to connected participants until the 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 Participants Only).

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.

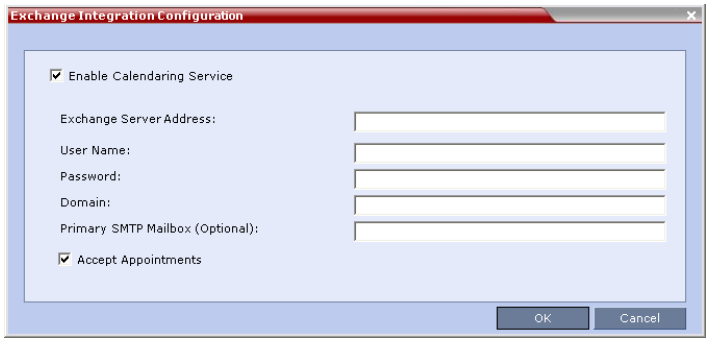
If included in the solution, *Polycom RealPresence DMA system* and calendaring-enabled endpoints share this e-mail account. For more information, see the *Polycom Unified Communications Deployment Guide for Microsoft Environments*.

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**.

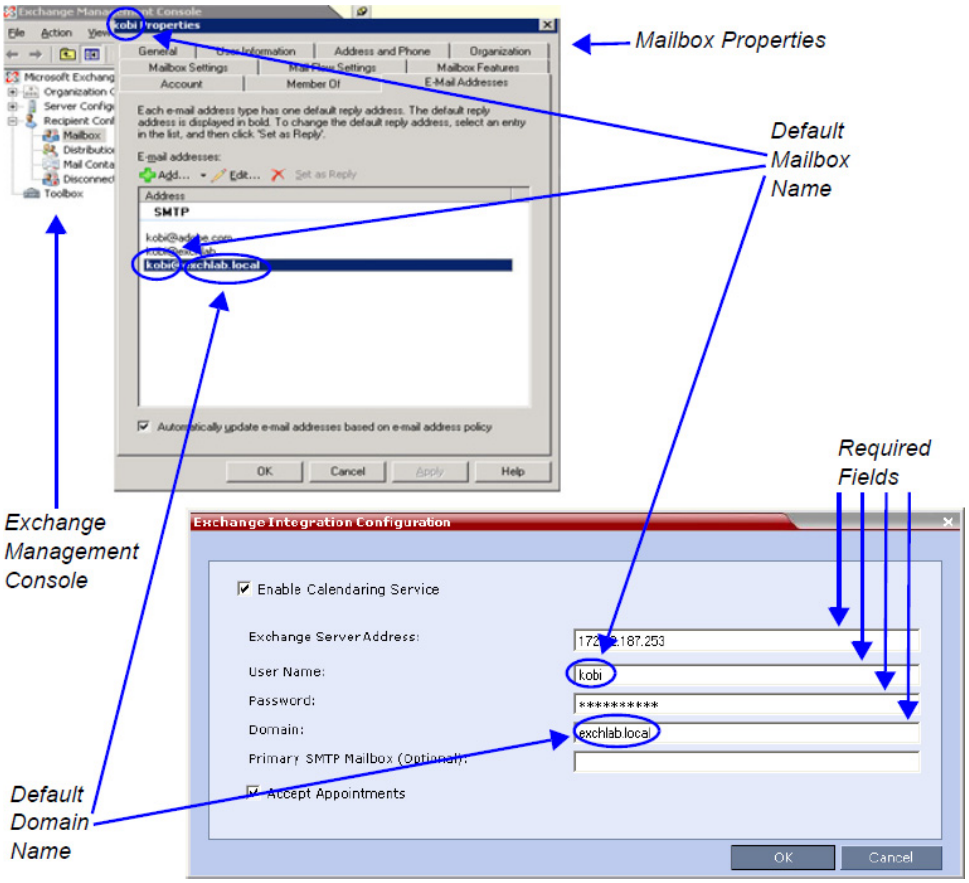
The **Exchange Integration Configuration** dialog box is displayed.



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



a Define the following fields:

Exchange Integration Configuration - Option 1

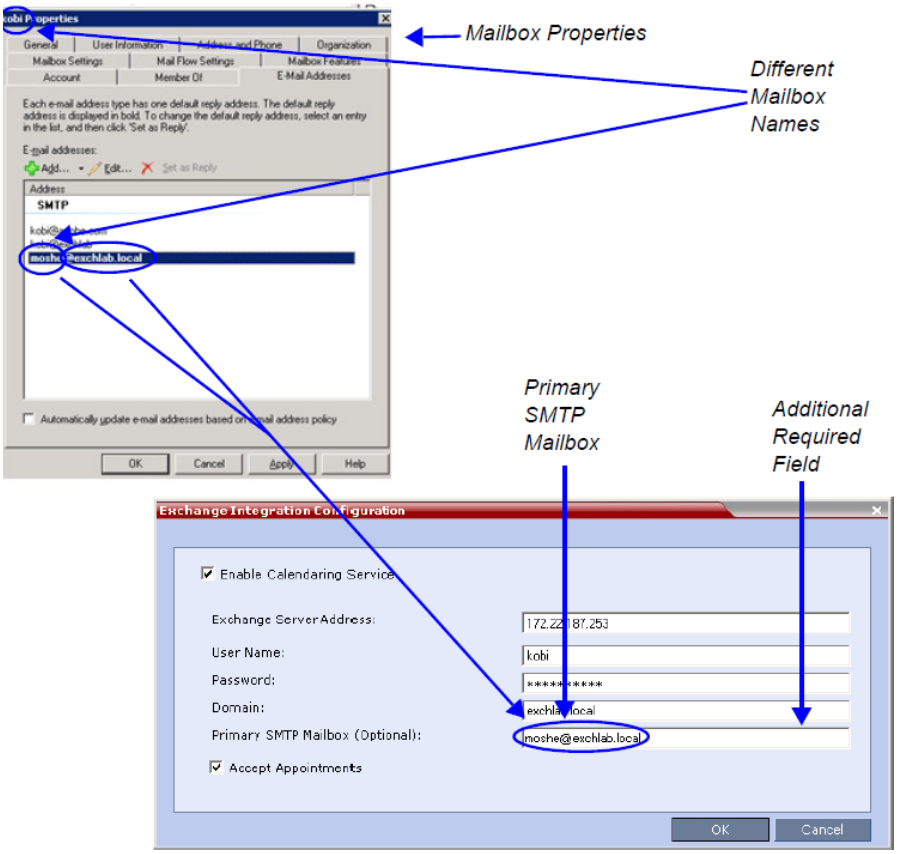
Field	Description
Enable Calendaring Service	Select or clear this check box to enable or disable the Calendaring Service using the Polycom Add-in for Microsoft Outlook. When this check box is cleared all fields in the dialog box are disabled.
Exchange Server Address	Enter the IP address of the Exchange Server.
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.

Exchange Integration Configuration - Option 1

Field	Description
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 **OK**.

Option 2 - Using an alternate Primary SMTP Mailbox



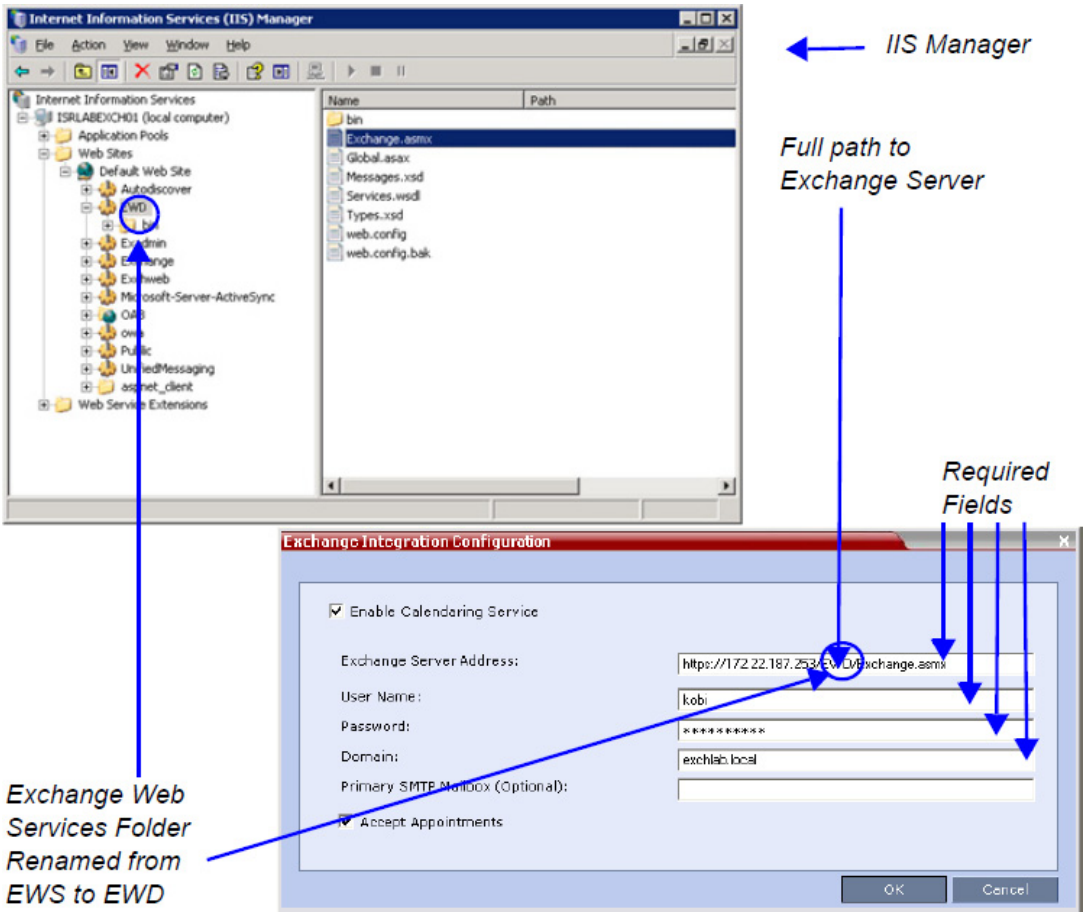
a Define the following fields:

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 **OK**.

Option 3 - Using modified Exchange Server settings



a Define the following fields:

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 in the RealPresence Collaboration Server 800s on page 8. Field length: Up to 80 characters.
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*.

1 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.

Calendaring Guidelines

- The Collaboration Server must have its *MCU* prefix registered in the gatekeeper.

  For more information see Modifying the Default IP Network Service.
- The Collaboration Server must be configured as a *Static Route*.

  For more information see Modifying the Default IP Network Service.

- 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.
- 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*.

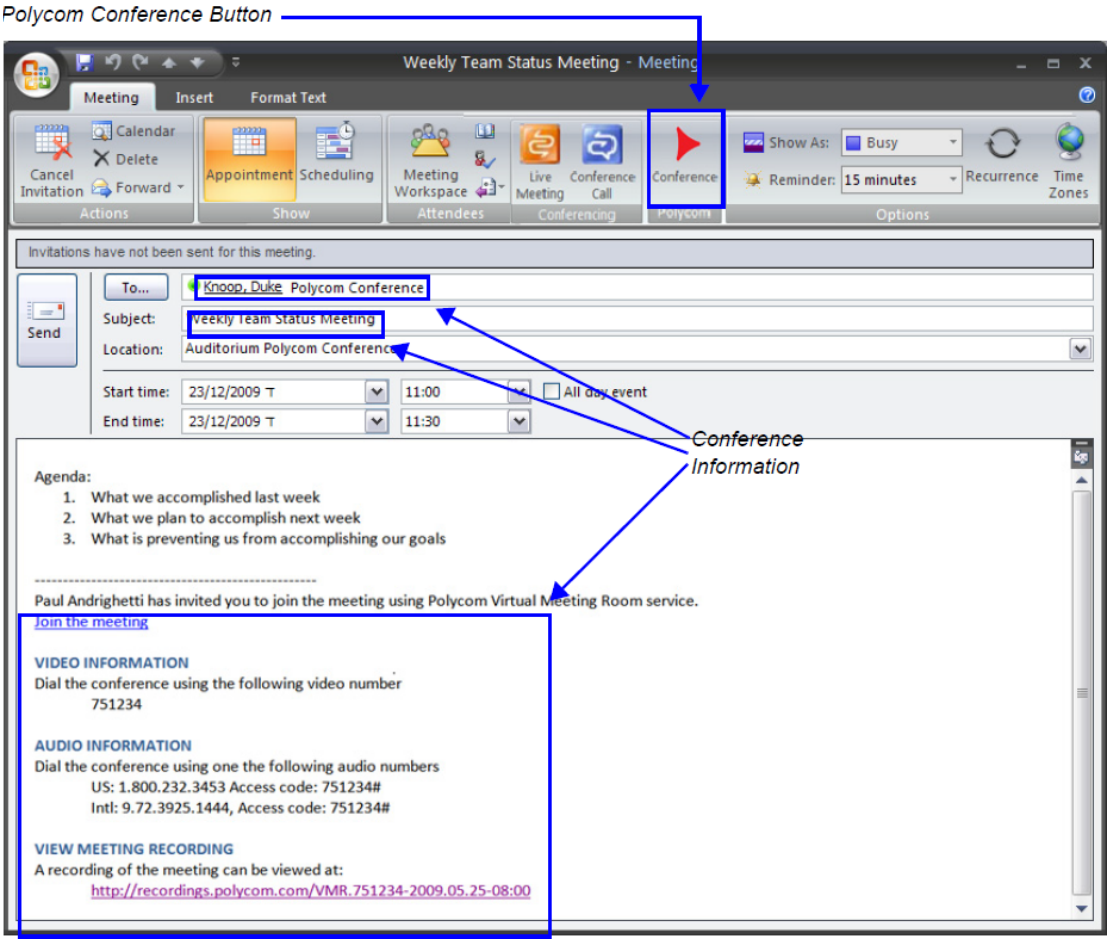
  For more information see Defining AVC CP Conferencing Profiles.
- Meetings can be single instance or have multiple occurrences.
- Attendees that do not have video devices may be invited to the meeting.
- Attendees using e-mail applications that use the *iCalendar* format may be invited to meetings via 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.



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.

The table below summarizes the Collaboration Server's usage of *Microsoft Outlook* data fields included in the meeting invitation.

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/RealPresence 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.

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.

Conference and Participant Monitoring

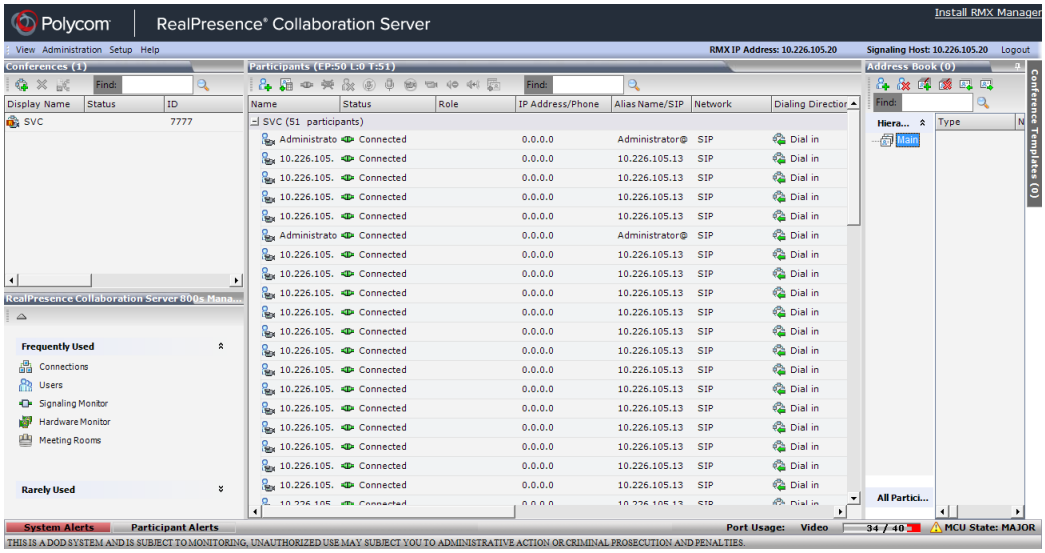
You can monitor ongoing conferences and perform various operations while conferences are running.

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.

General Monitoring

Users can monitor a conference or keep track of its participants and progress. For more information, see *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Monitoring Ongoing Conferences on page 166.



You can click the blinking **Participant Alerts** indication bar to view participants that require attention. For more information, see System and Participant Alerts.

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.

Conference monitoring - Tab list per conferencing mode and user

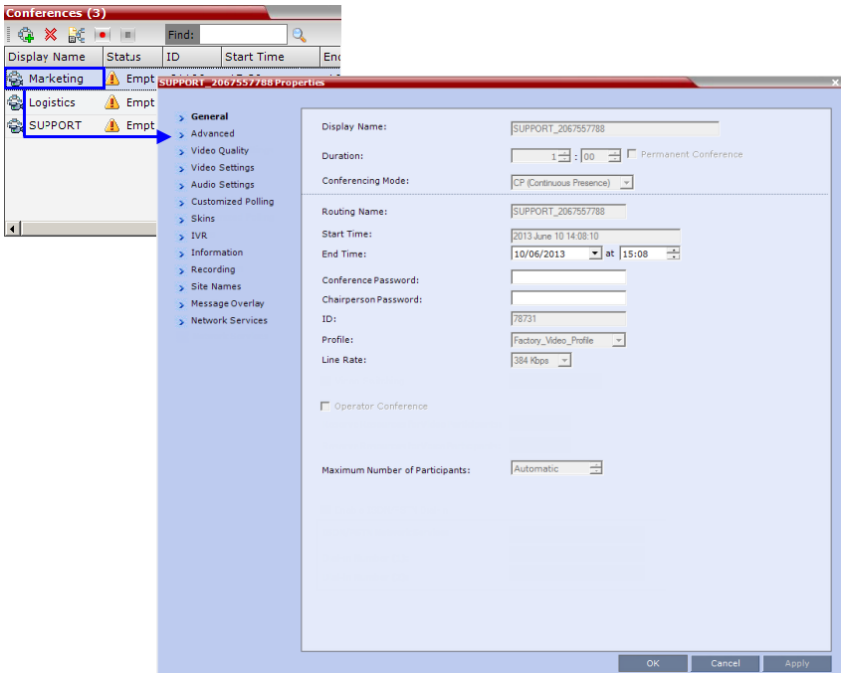
	Admin				Chairperson				Operator			
Tab Name	**CP**	**SVC**	**Mixed**	**VSW**	**CP**	**SVC**	**Mixed**	**VSW**	**CP**	**SVC**	**Mixed**	**VSW**
General	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Advanced	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Gathering Settings	✓	×	×	✓	✓	×	×	✓	✓	×	×	✓
Video Quality	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Video Settings	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Audio Settings	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Customized Polling	✓	×	×	✓	×	×	×	×	✓	×	×	✓
Skins	✓	×	✓	×	✓	×	✓	×	✓	×	✓	×
IVR	✓	✓	✓	✓	×	×	×	×	✓	✓	✓	✓
Information	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Recording	✓	×	✓	✓	✓	×	✓	✓	✓	×	✓	✓
Site Names	✓	×	✓	×	✓	×	✓	×	✓	×	✓	×
Message Overlay	✓	×	×	×	✓	×	×	×	✓	×	×	×
Network Services	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓

Viewing the Properties of Ongoing CP and Mixed CP and SVC Conferences

To view the parameters of an ongoing CP conference:

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

The *Conference Properties - General* dialog box with the **General** tab opens.



The following information is displayed in the *General* tab:

Conference Properties - General

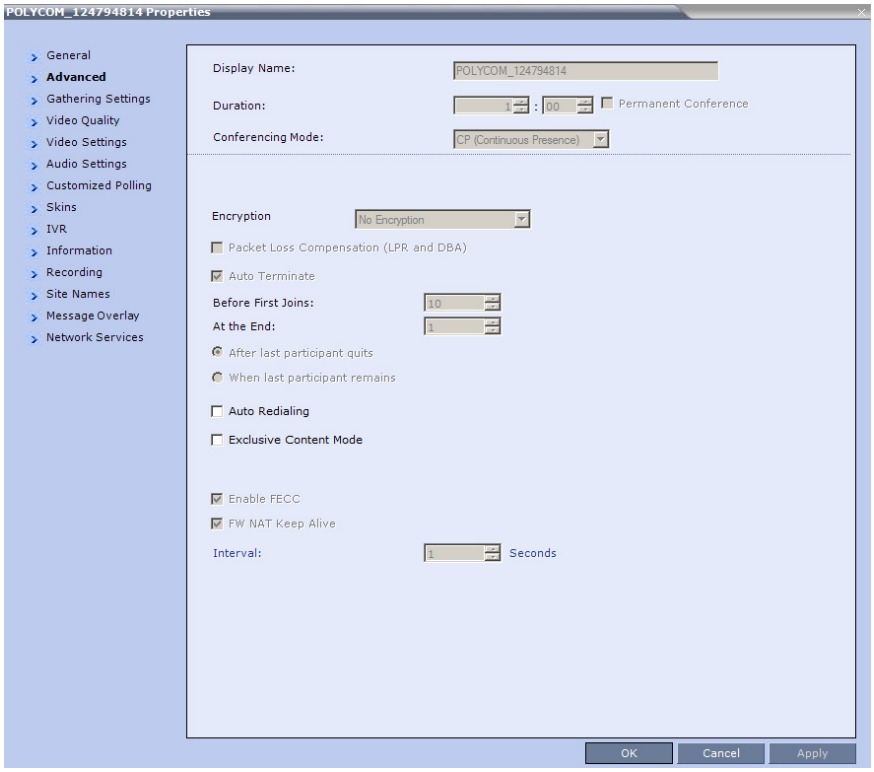
Field	Description
Display Name	The Display Name is the conference name in native language and Unicode character sets to be displayed in the *Collaboration Server Web Client*. **Note:** This field is displayed in all tabs.
Duration	The expected duration of the conference using the format HH:MM. **Note:** 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*.

Conference Properties - General

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.



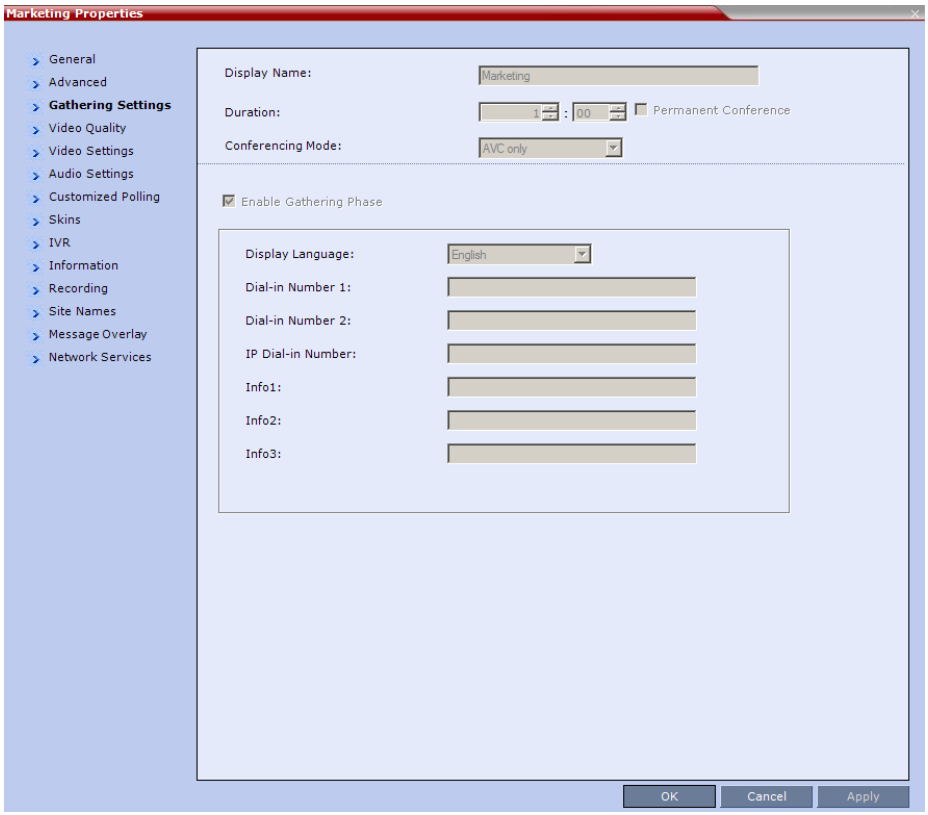
3 The following information is displayed in the *Advanced* tab:

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.
TIP Compatibility	Indicates the *TIP Compatibility* mode implemented for the conference, when the environment implements the *Collaboration Server and Cisco Telepresence Systems (CTS) Integration* solution. • None • Video Only • Video & Content • Prefer TIP For more information, see Collaboration With Cisco's Telepresence Interoperability Protocol (TIP).
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 RealPresence Collaboration Server 800s/Virtual Edition Network Port Usage.

4 Click the **Gathering Settings** tab.

The *Conference Properties - Gathering Settings* dialog box opens.



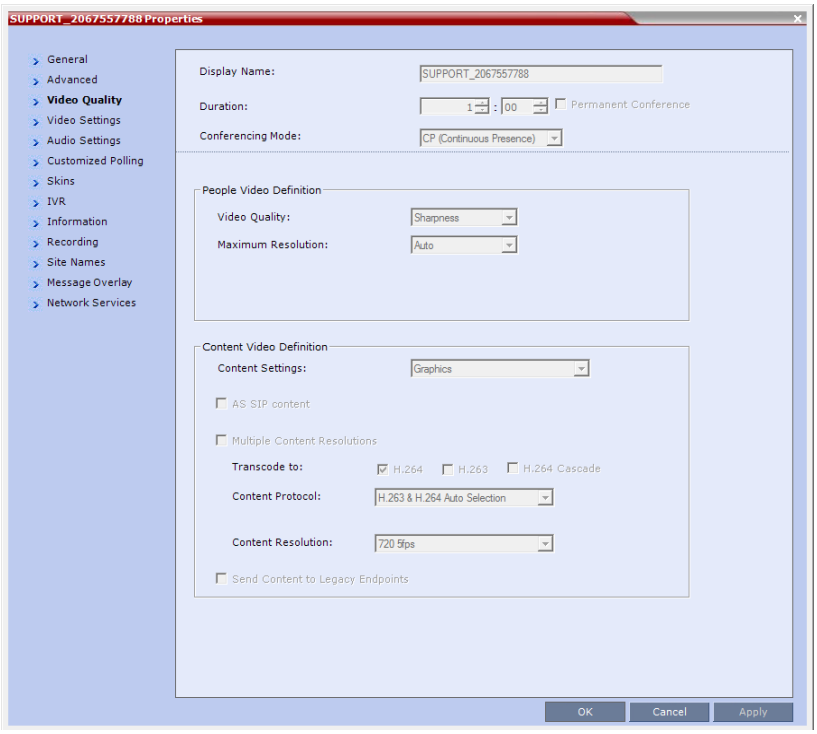
The following information is displayed:

Profile - Gathering Settings

Field/Options	Description
Enable Gathering	Indicates whether the *Gathering Phase* has been enabled.
Display Language	Indicates the language of the *Gathering Slide* field headings. **Note:** When working with the *Polycom Conferencing Add-in for Microsoft Outlook,* the language selected should match the language selected for the conference in the *Polycom Conferencing Add-in for Microsoft Outlook* to ensure that the *Gathering Phase* slide displays correctly.
Info 1	Additional information to be displayed during the *Gathering Phase*.
Info 2	
Info 3	

5 Click the **Video Quality** tab.

The *Conference Properties - Video Quality* dialog box opens.



The following information is displayed:

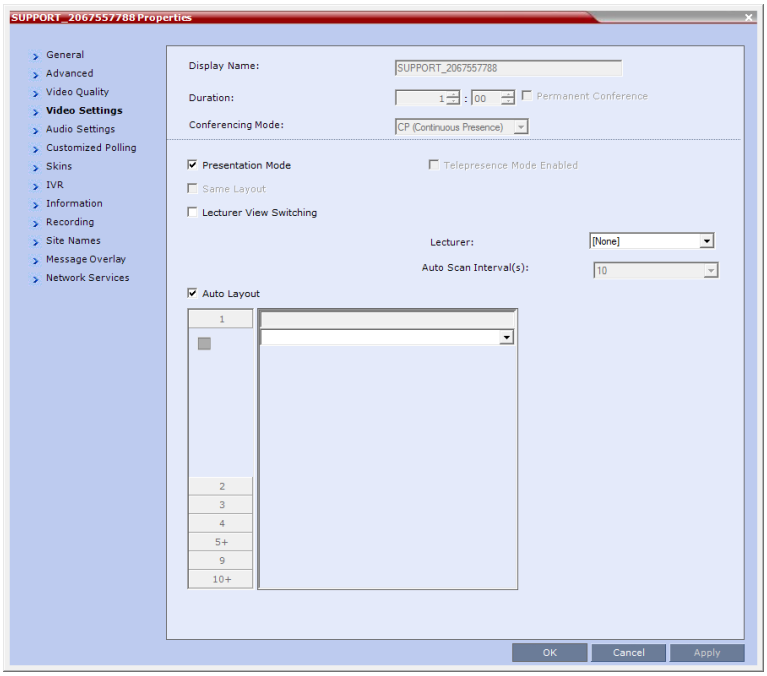
Conference Properties - Video Quality Parameters

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.
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 Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition.

Conference Properties - Video Quality Parameters

Field/Option	Description
Multiple Content Resolutions	Indicates if *Multiple Content Resolutions* mode for content sharing is enabled. In this mode, content is shared in multiple streams, one for each video protocol: H.263 and H.264. This allows endpoints with different protocols to connect and disconnect without having to restart Content sharing in the middle of a conference. For more information, see Sharing Content Using Multiple Content Resolutions Mode.
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.
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.
Send Content to Legacy Endpoints (CP only)	Indicates if the *Send Content to Legacy Endpoints* is enabled. If enabled, Content can be sent to H.323/SIP endpoints that do not support H.239 Content (legacy endpoints) over the video (people) channel. For more information see Sending Content to Legacy Endpoints (AVC Only).

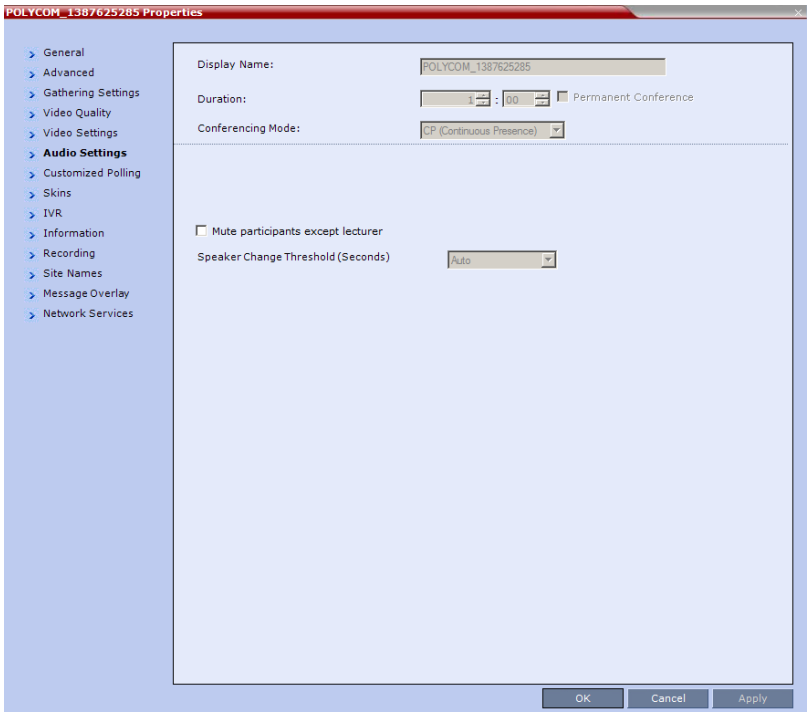
6 Click the **Video Settings** tab to list the video parameters.



Conference Properties - Video Settings Parameters

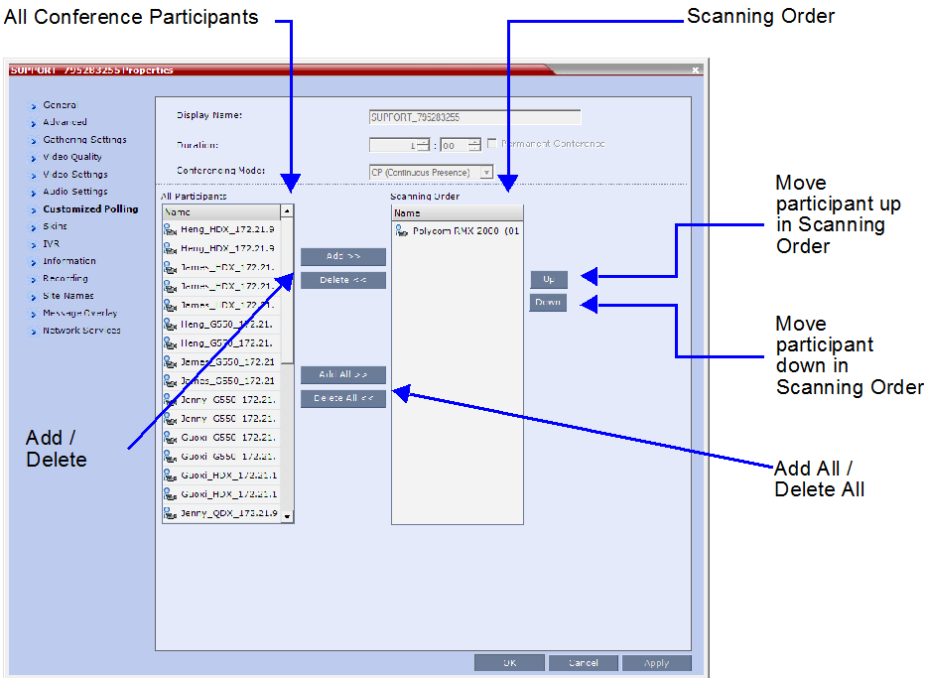
Field	Description
Presentation Mode	When checked, indicates that the Presentations Mode is active. This option is disabled in a mixed CP and SVC conference. For more information, see Supplemental Conferencing Features.
Lecturer View Switching	When checked, the *Lecturer View Switching* enables automatic 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.
Video Layouts (graphic)	Indicates the currently selected video layout.

7 Click the A**udio Settings** tab to view the audio setting for the conference.



8 If needed, you can enable or disable the *Mute participants except lecturer* setting.

9 **CP Only Conferences:** Click the C**ustomized Polling** tab to view and modify the customized polling for the conference.



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 table below.

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*.

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

or

Click **OK** to confirm and return to the *Collaboration Server Web Client Main Screen*.

11 Click the **Skins** tab to view the skin selected for the conference.

You cannot select another skin during an ongoing conference.

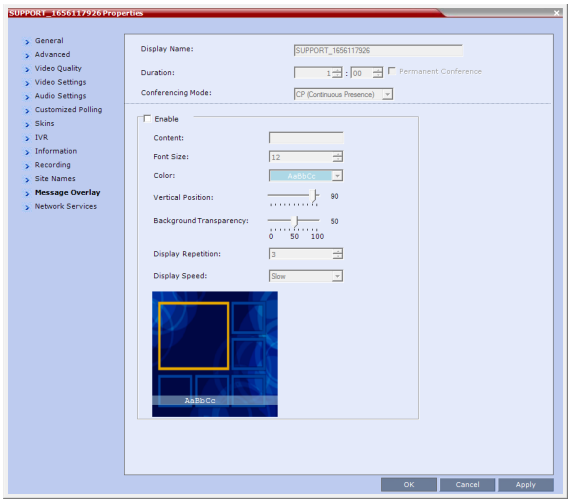
12 Click the **IVR** tab to view the IVR settings.

13 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.

14 Click the **Recording** tab to review the recording settings for the conference.

15 Click the **Site Names** tab to enable or disable the display of site names during the conference, and adjust the display properties.

16 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.



For more information, see Sending Text Messages During a Conference Using Message Overlay on page 71.

17 Click the **Network Services** tab to verify the SIP registration for the conference.

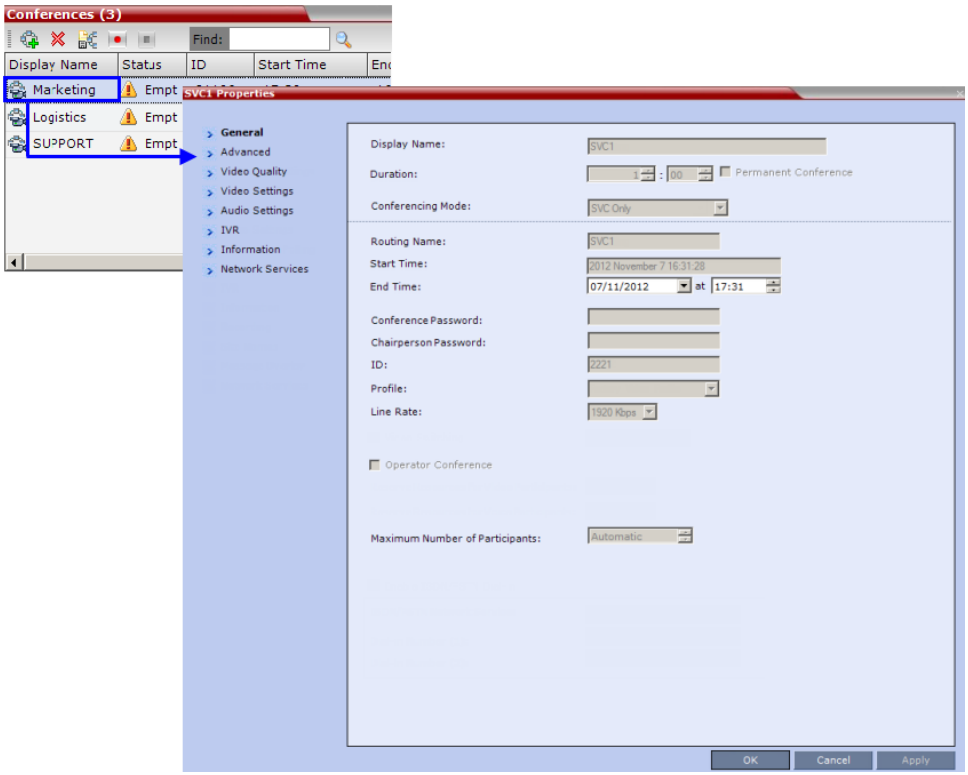
18 Click **OK** to close the *Conference Properties* dialog box.

Viewing the Properties of Ongoing SVC-based Conferences

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.



2 The following information is displayed in the *General* tab:

Conference Properties - General Parameters

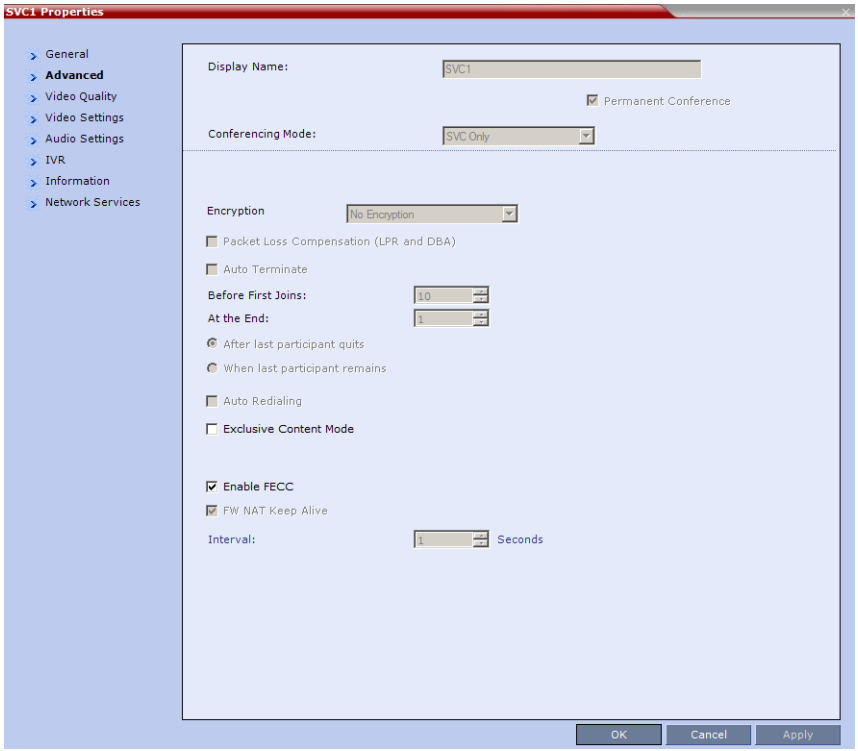
Field	Description
Display Name	The Display Name is the conference name in native language and Unicode character sets to be displayed in the *Collaboration Server Web Client*. **Note:** This field is displayed in all tabs.
Duration	The expected duration of the conference using the format HH:MM. **Note:** 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.
Conference Password	A numeric password for participants to access the conference.

Conference Properties - General Parameters

Field	Description
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.

3 Click the **Advanced** tab.

The *Conference Properties - Advanced* dialog box opens.



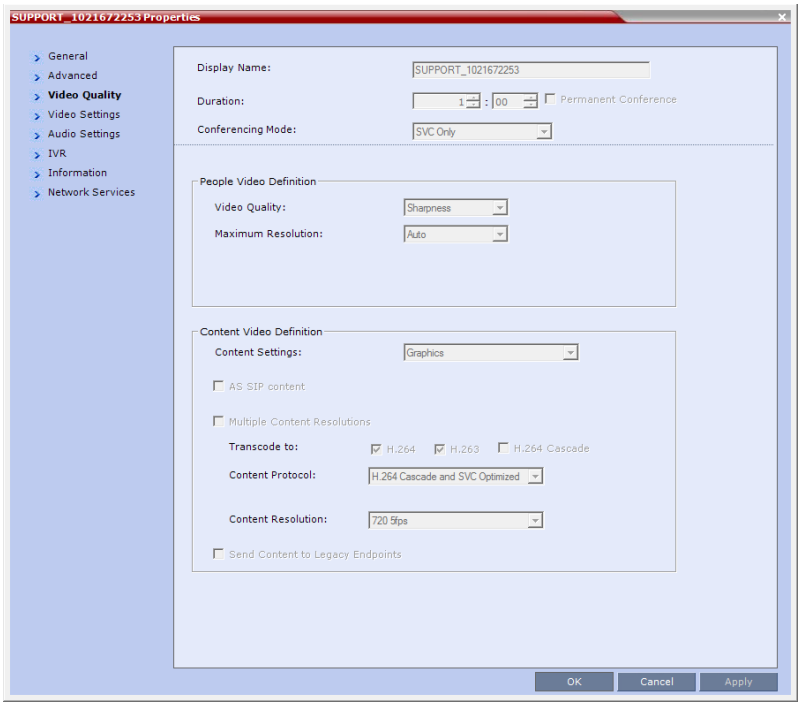
4 The following information is displayed in the *Advanced* tab:

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 800s/Virtual Edition Network Port Usage on page 37.

5 Click the **Video Quality** tab.

The *Conference Properties - Video Quality* dialog box opens.

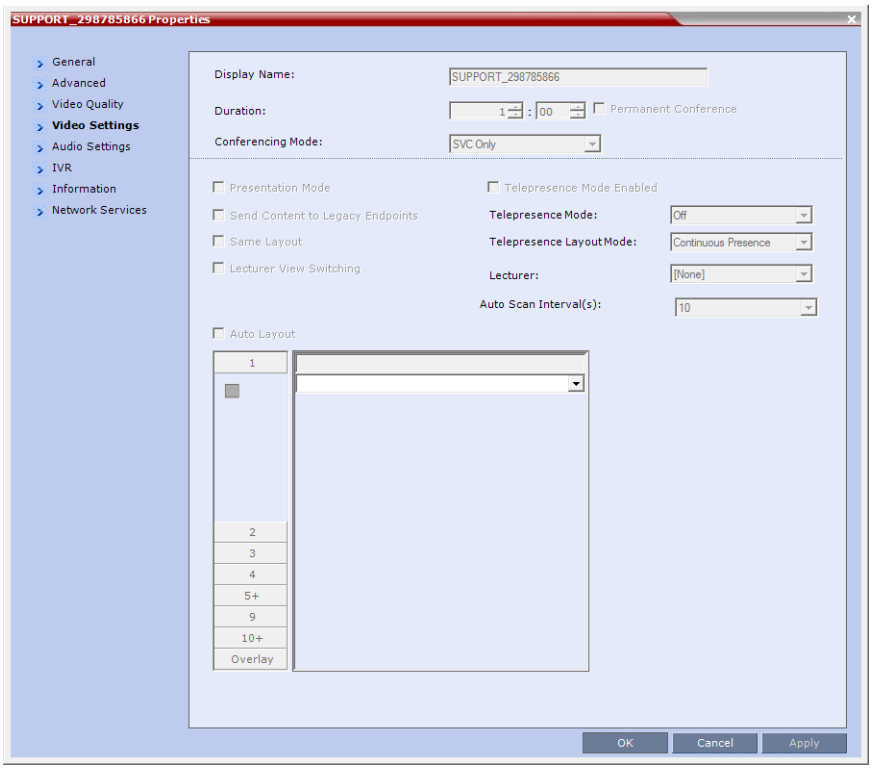


The following information is displayed:

Conference Properties - Video Quality Parameters

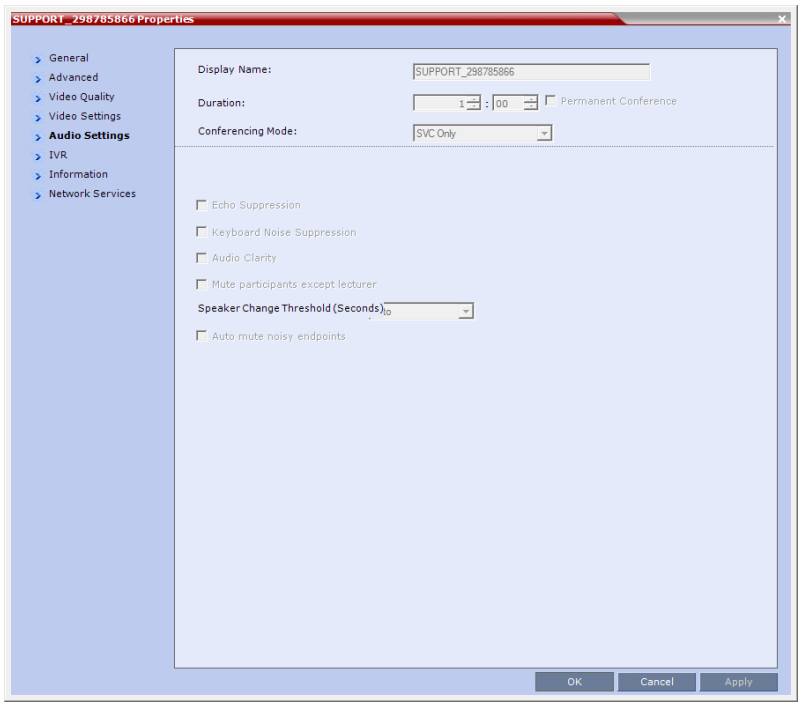
Field/Option	Description
People Video Definition	
Video Quality	Indicates the resolution and frame rate that determine the video quality set for the conference. Only Sharpness is supported.
Maximum Resolution	In *SVC conferencing*, this is always *Auto* (default) - The *Maximum Resolution* remains as selected in the *Resolution Configuration* dialog box.
Content Video Definition	
AS-SIP	This option is not supported with Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition.
Multiple Content Resolutions	*Multiple Content Resolutions* 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.



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.



In SVC conferences, all Audio Settings options are disabled.

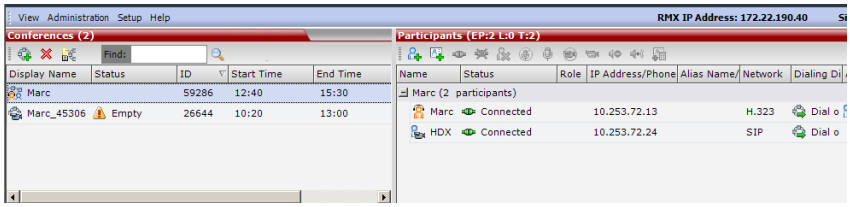
8 Click the **Information** tab to view general information defined for the conference. Changes made to this information once the conference is running are not saved to the CDR.

9 Click **OK** to close the *Conference Properties* dialog box.

Monitoring of 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.



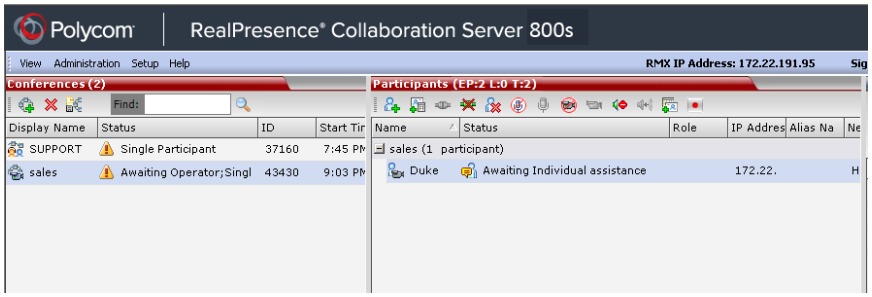
You can view the properties of the *Operator conference* by double-clicking the conference entry in the *Conferences* list or by right-clicking the conference entry and selecting **Conference Properties**. For more information, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Conference Level Monitoring.

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:



- The participant's connection *Status* changes, reflecting the help request. For details, see Table 5-9.
- 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:

Participants List Status Column Icons and Indications

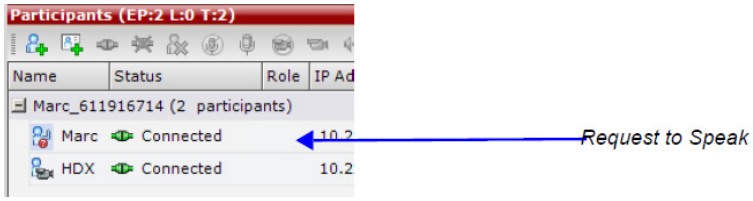
Icon	Status indication	Description
	Awaiting Individual Assistance	The participant has requested the operator's assistance for himself/herself.
	Awaiting Conference Assistance	The participant has requested the operator's assistance for the conference. Usually this means that the operator is requested 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.



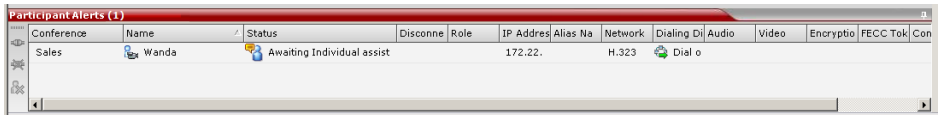
Request to Speak is:

- 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.
- 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.



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.

The table below lists the tabs in the *Participant Properties* dialog box, as viewed by each user type, for each participant connection types.

Participant monitoring - Tab list per participant connection type and user

	Admin		Chairperson		Operator	
Tab Name	**AVC H.323**	**AVC/SVC SIP**	**AVC H.323**	**AVC/SVC SIP**	**AVC H.323**	**AVC/SVC SIP**
General	✓	✓	✓	✓	✓	✓
Advanced	✓	✓	×	×	✓	✓
Information	✓	✓	✓	✓	✓	×
Media Sources	✓	✓	✓	✓	✓	✓
H.245	✓	×	×	×	✓	×
SDP	×	✓	×	×	×	✓
Connection Status	✓	✓	×	×	✓	✓
Channel Status	✓	✓	×	×	✓	✓
Channel Status - Advanced	✓	✓	×	×	×	×
Gatekeeper Status	✓	✓	×	×	×	×
Call Admission Control	×	×	×	×	×	×

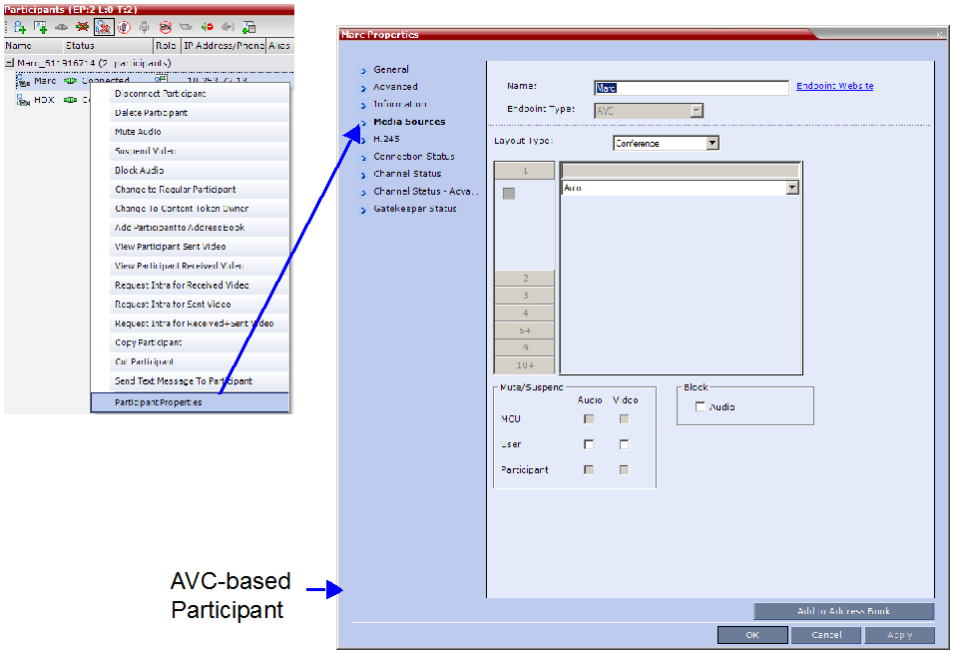
Viewing the Properties of Participants

To view the participant Properties:

» In the *Participant List* pane double-click the participant entry. Alternatively, right-click a participant and then click **Participant Properties**.

The *Participant Properties* dialog box opens, displaying the last opened tab.

Media Sources properties are not available for SVC participants.



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.

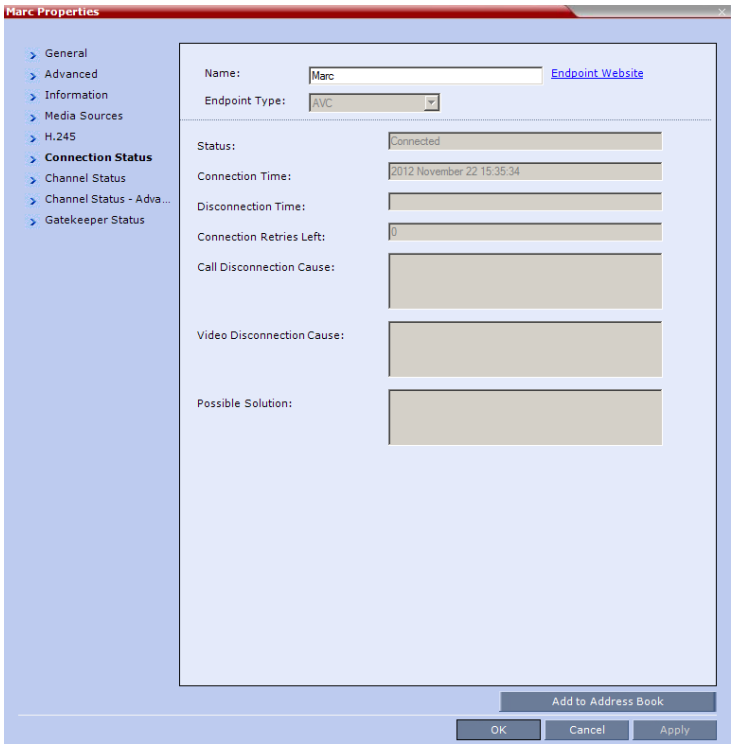
Monitoring IP Participants

The following parameters are displayed for an IP participant.

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 800s / Virtual Edition Getting Started Guide,* Changing the Video Layout of a Conference (AVC-Based CP and Mixed CP and SVC Conferences).
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.
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.

1 Click the **Connection Status** tab to view the connection status, and if disconnected the cause of the disconnection.



This dialog box is the same for AVC-based and SVC-based participants.

The following parameters are displayed:

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:* Appendix A - Disconnection Causes on page 923.

Participant Properties - Connection Status Parameters

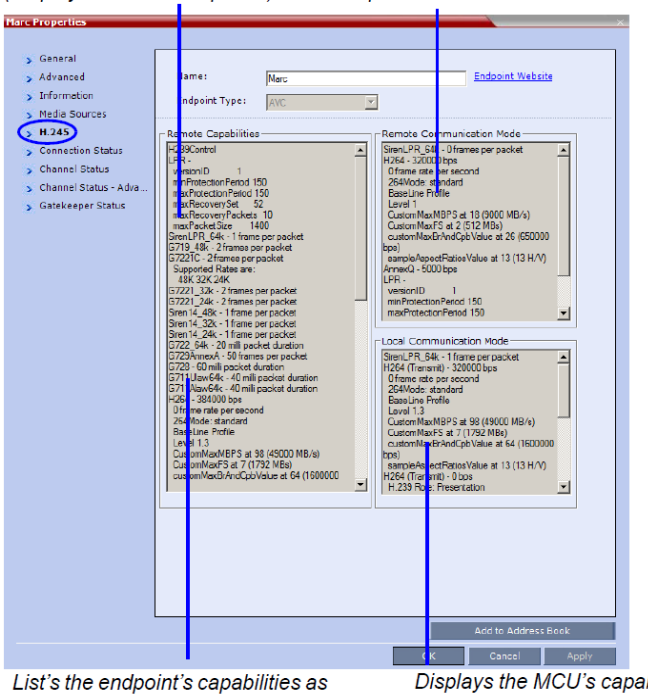
Field	Description
Video Disconnection Cause	Displays the cause the video channel could not be connected. For more information, see *Appendix A:* Appendix A - Disconnection Causes on page 923.
Possible Solution	In some cases, a possible solution is indicated to the cause of the video disconnection.

2 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.

LPR activity (Displayed in all three panes)

Displays the endpoint's actual capabilities used for the connection

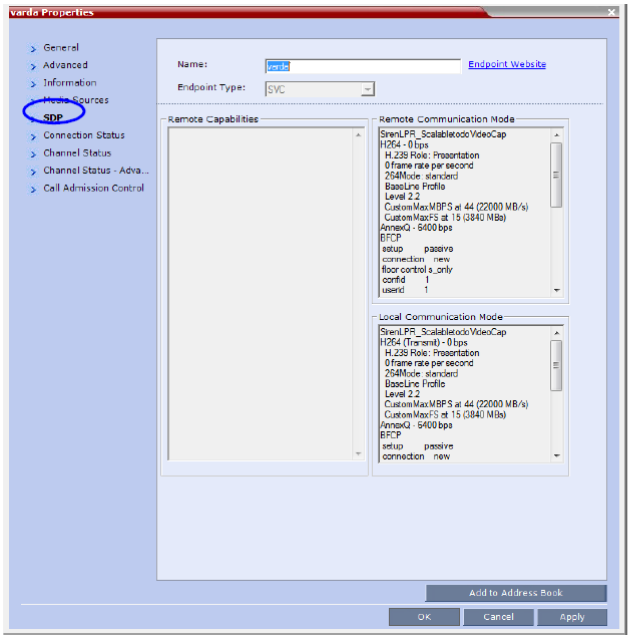
H.323 Participant (AVC-based)



List's the endpoint's capabilities as retrieved from the remote site

Displays the MCU's capabilities used for connection with the participant

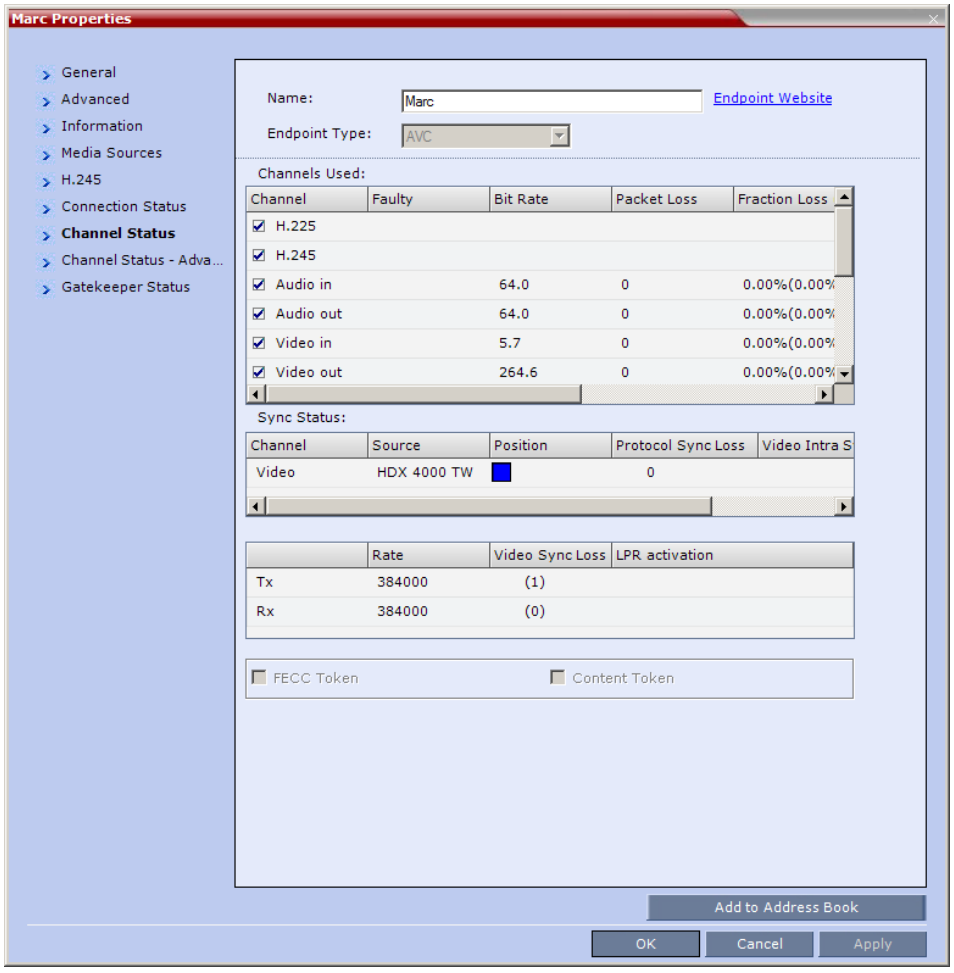
SIP Participant (AVC-based and SVC-based)



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).

3 Click on the **Channel Status** tab to view the status of the various channels.



The following parameters are displayed:

Participant Properties - Channel Status Parameters

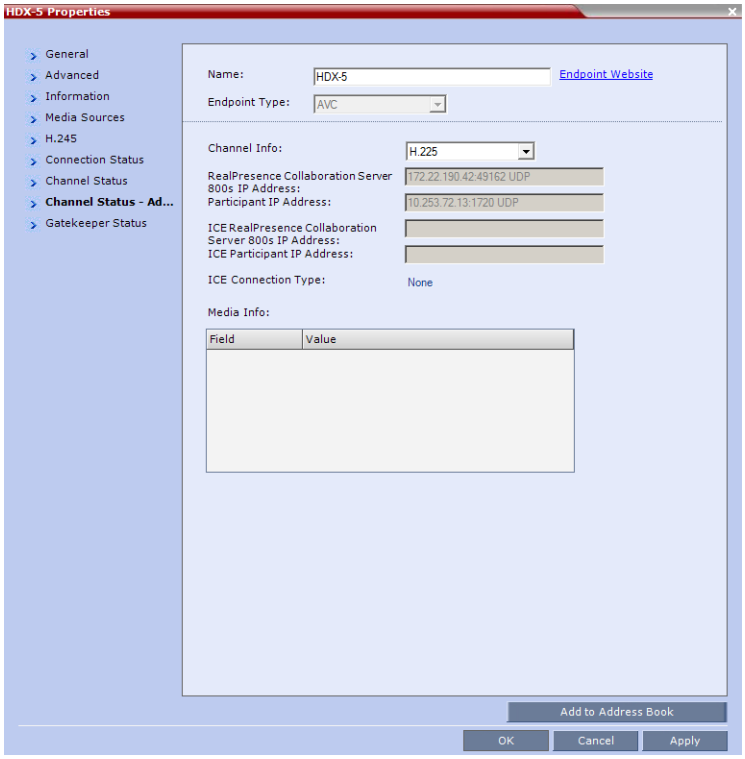
Field	Description
Channels Used	When checked, indicates the channel type used by the participant to connect to the conference: Incoming channels are endpoint to MCU, Outgoing channels are from MCU to endpoint. Channels: • *H.225/Signaling* - The call-signaling channel. • *H.245/SDP* - The Control channel. • *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.

Participant Properties - Channel Status Parameters

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.

4 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:



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.

Participant Properties - Channel Status Advanced Parameters

Field	Description
ICE Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition IP Address	The IP address, port number, and transport protocol of the MCU used to pass through the media when ICE is functional. See Participant Properties - ICE Connection Parameters.
ICE Participant IP Address	The IP address, port number, and transport protocol of the endpoint used to pass through the media when ICE is functional. See Monitoring the Participant Connection in ICE Environment.
ICE Connection Type	Indicates the type of connection between the Collaboration Server and the participant in the ICE environment: • **Local** (or Host) - The endpoint (Remote) is on the same network as the Collaboration Server and the media connection is direct, using local addresses. • **Relay** - Media between the Collaboration Server and the participant passes through a media relay server. • **Firewall** - Media connection between the Collaboration Server and the participant is done using their external IP addresses (the IP addresses as seen outside of the local network).
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.
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.

5 **Optional for H.323 AVC-based participants.** Click the **Gatekeeper Status** tab to view its parameters.

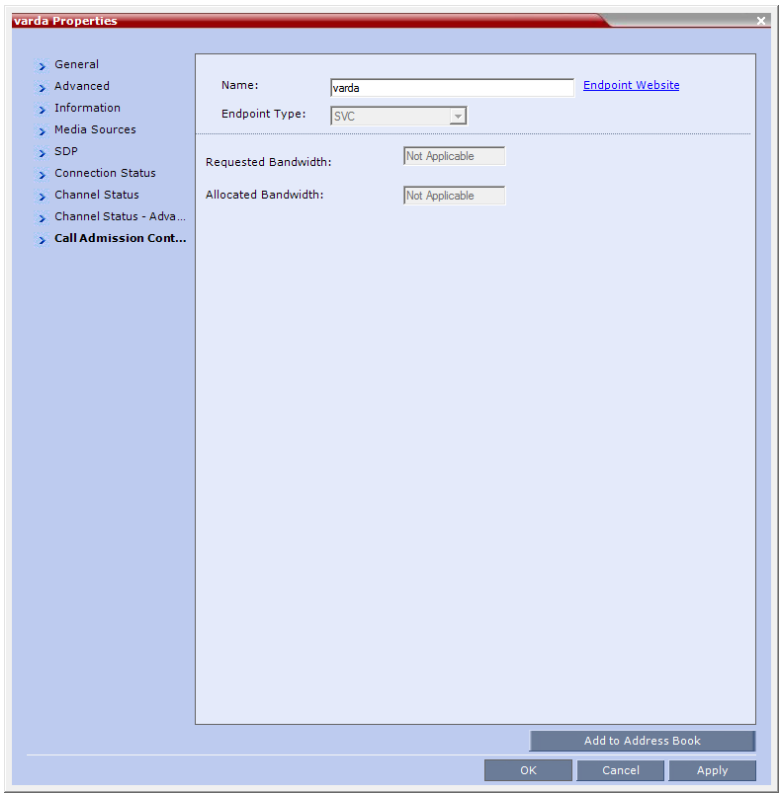
Participant Properties - Gatekeeper Status Parameters

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.

Participant Properties - Gatekeeper Status Parameters

Field	Description
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.

6 **Optional for SIP AVC-based and SVC-based participants.** Click the **Call Admission Control** tab to view its parameters.



Participant Properties - Gatekeeper Status Parameters

Field	Description
Requested Bandwidth	The bandwidth requested by the MCU from the SIP server.

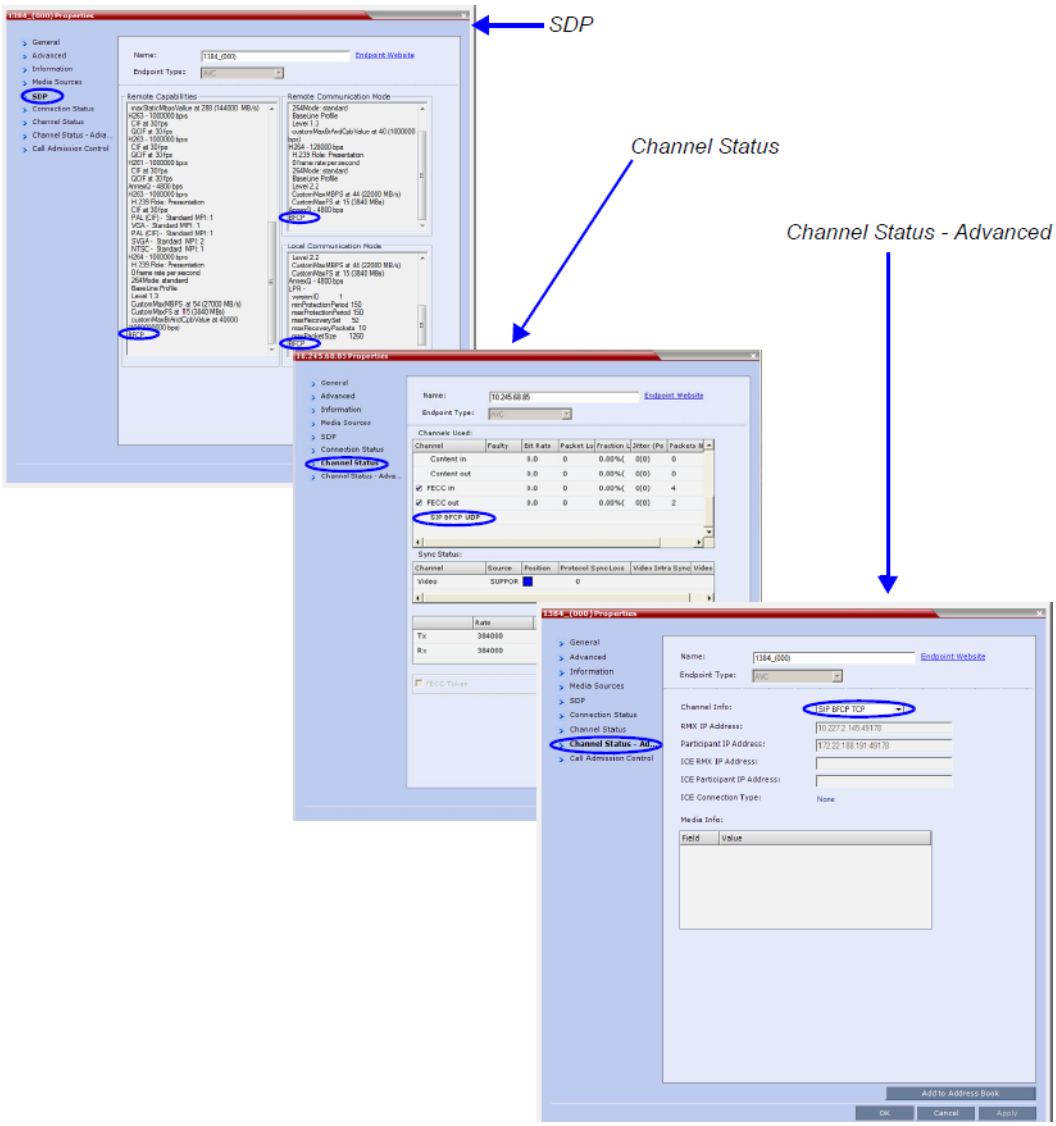
Participant Properties - Gatekeeper Status Parameters

Field	Description
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.



For more information see Participant Level Monitoring.

Detecting Endpoint Disconnection

Endpoint detection can be detected for SIP and H323 protocols.

Note: H323 endpoint disconnection detection is not supported in the RealPresence Collaboration Server 800s**.**

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.

H323 Endpoint Disconnection Detection

In versions previous to version 8.4, when an H.323 endpoint disconnected, round trip messages were still received, even when a gatekeeper was deployed. Because of this, the MCU failed to detect the endpoint disconnection.

From version 8.4 onward, the MCU detects H.323 endpoint connection or disconnection by monitoring RTCP/RTP messages reception using either the audio or video channels. When these messages are not received within a predefined timeout interval, the endpoint is considered disconnected. Therefore as long as either RTCP or RTP messages (interchangeably) are received on either of the video or audio channels, the endpoint is considered connected.

If no messages are received through either channel within the predefined time out interval, the endpoint is disconnected, and a disconnection message, `H.323 remote stopped responding`, is sent to the endpoint.

No channel disconnection detection occurs while an endpoint's video or audio are muted, or while the endpoint is put on hold.

In audio calls, only the audio channel is polled for RTCP/RTP messages.

Configuring the System Flag

The time out used for SIP endpoint disconnection detection is controlled by the **DETECT_SIP_EP_DISCONNECT_TIMER** System Flag which must be added to the System Configuration to view or modify its value.

Range: 0 - 300

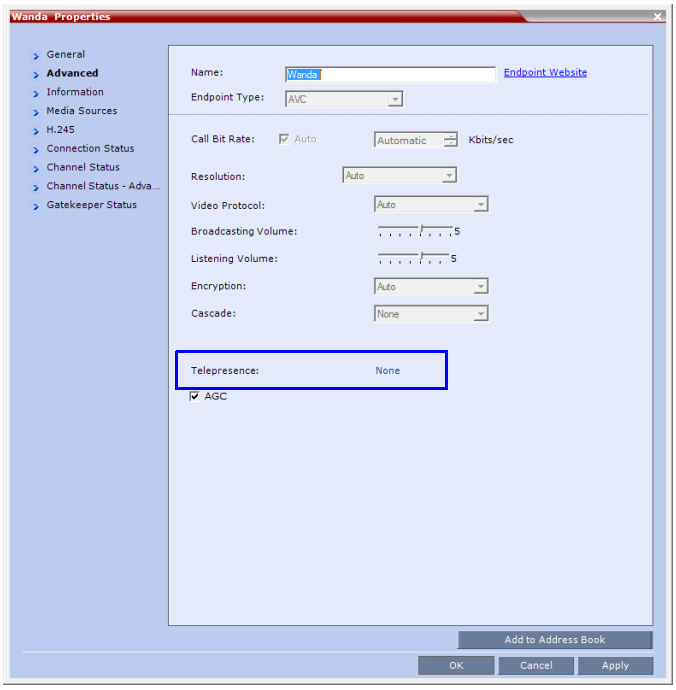
When the value is set between 0 and 14, the feature is disabled and SIP endpoints are not detected for disconnection. When the value is set between 15 and 300, the feature is enabled.

Default: 20

For more information see Manually Adding and Deleting System Flags, and Modifying System Flags in the *RealPresence Collaboration Server (RMX) Administrator's Guide*.

Monitoring Telepresence Participant Properties

A *Telepresence* status indicator is displayed in the *Participant Properties - Advanced* tab when monitoring conference participants.



The *Telepresence* mode of the participant is indicated:

- *RPX* - the participant's endpoint is transmitting 4:3 video format.
- *TPX* - the participant's endpoint is transmitting 16:9 video format.

  None - the participant's endpoint is neither *RPX* nor *TPX*.

Recording Conferences

Conference recording is not available in SVC Conferencing Mode.

Conferences running on the Collaboration Server can be recorded using Polycom RealPresence® Capture Server in the following ways:

- Capture Server dials in MCU for conference recording via an SIP interface from Capture Server. From the Capture Server Admin UI, you can start a conference recording by dialing out to a Collaboration Server. After the Collaboration Server gets the request from the Capture Server, the recording will start if the recording is enabled on the Collaboration Server. During the conference recording, you can pause or stop the recording from the Capture Server Admin UI. Refer to the Polycom RealPresence Capture Server Userâ€™s Guide on how to start a recording by dialing out to an interoperable endpoint.
- Collaboration Server dials out to Capture Server for a conference recording. Recording conferences is enabled via a dial-out Recording Link, which is a dial-out connection from the conference to the recording system.

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 *Dial Out 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 *Dial Out Recording Links* may be defined.

Conference Dial Out 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 Dial Out 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

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 VRRs.

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.

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*.

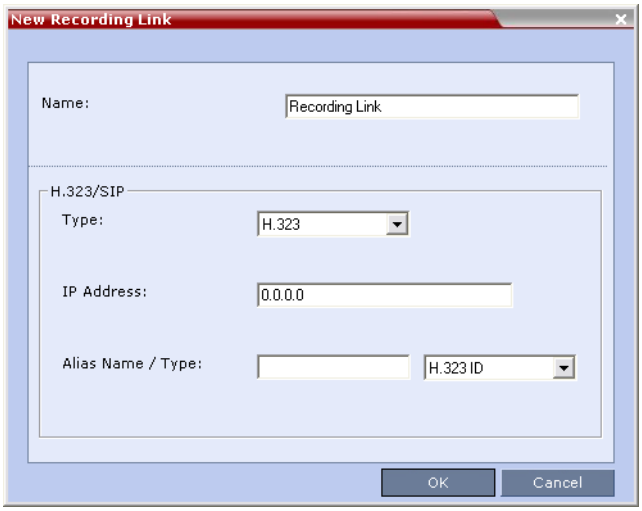
In *Multiple Networks* Configuration, Recording Links use the default Network Service to connect to conferences, therefore the recording system must be defined on the default IP Network Service to enable the recording.

To define a Recording Link:

1 In the *RMX 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.



3 Define the following parameters:

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:

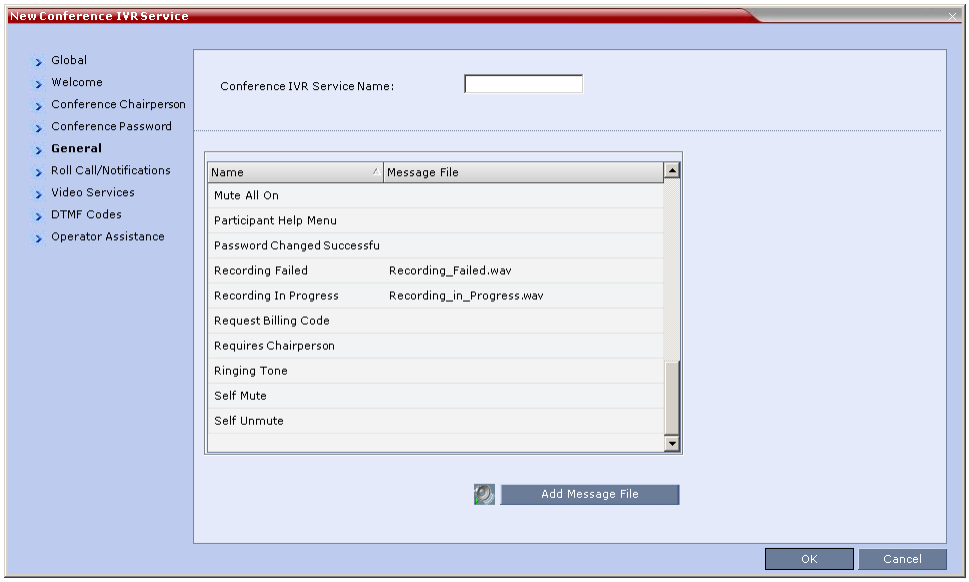
1 In the **RMX Management** pane, click the **IVR Services** ( ) button.

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.



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 *R***ecording 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.

Default DTMF Codes assigned to the recording process

Recording Operation	DTMF Code	Permission
Start or Resume Recording	*2	Chairperson
Stop Recording	*3	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:

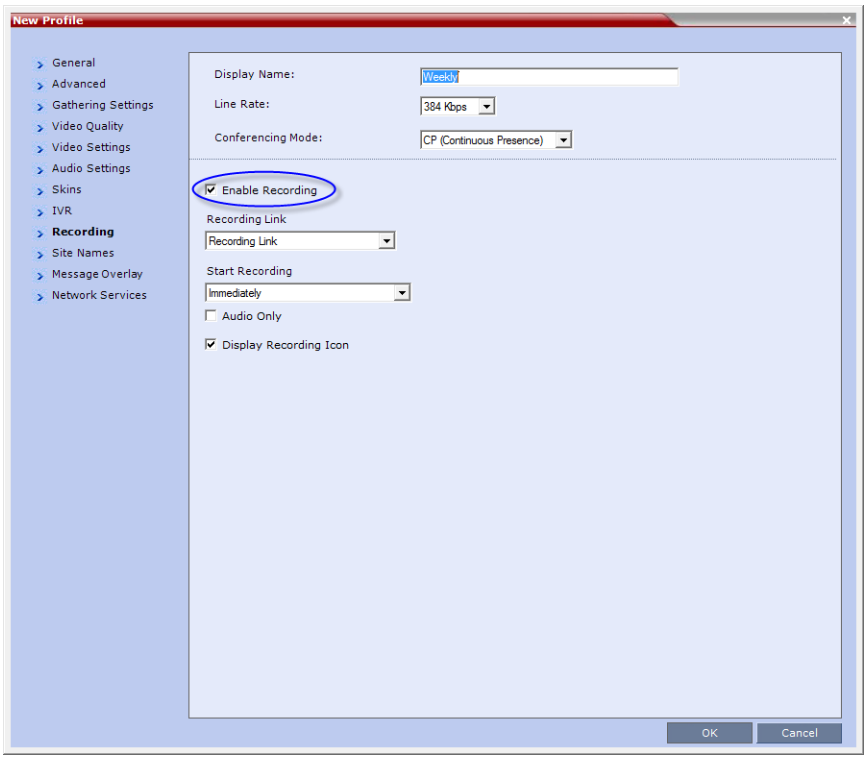
1 In the *Collaboration Server Management* pane, click the **Conference Profiles** () button.

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 Defining AVC CP Conferencing Profiles.

3 In the *Profile Properties* dialog box, click the **Recording** tab.



4 Select the **Enable Recording** check box.

5 Define the following parameters

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*.

Parameter	Description
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.

The following table 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.

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 Packet Loss Compensation (LPR and DBA) AVC CP Conferences.
- Setting the Recording options in the *Recording* tab. For more details, see Enabling the Recording in the Conference Profile.

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 userâ€™s request. Once the recording is initiated for a conference, the MCU connects to the recording device (Capture Server). 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 Capture Server by its Collaboration Server conference name.

A conference participant and the Recording Link cannot have identical names, otherwise the recording process will fail.

Recording 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.

The default settings for Auto Layout for the conference and the Recording Link are summarized in the following table:

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		

Participants	Conference Auto Layout Default Settings	Recording Link Auto Layout Settings
2		
3		
4		
5		
6		
7		
8		
9		
10 or more		

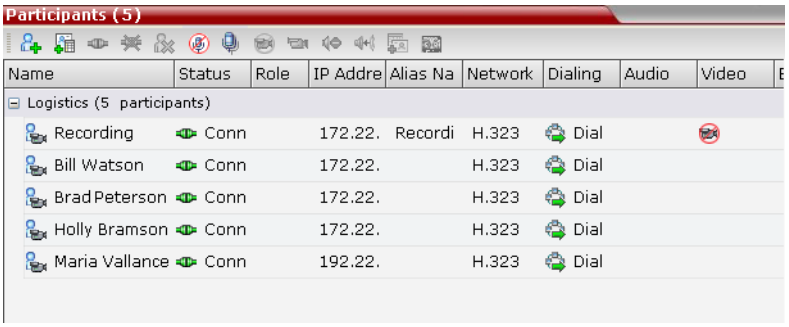
When Capture Server dials in MCU for conference recording, you can choose the video layout from the Capture Server Admin UI. The following layout is supported by the Collaboration Server.

- Auto: automatic layout setting at the Collaboration Server side.
- Single View: record 1*1 layout
- Dual View: record 1*2 layout.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:

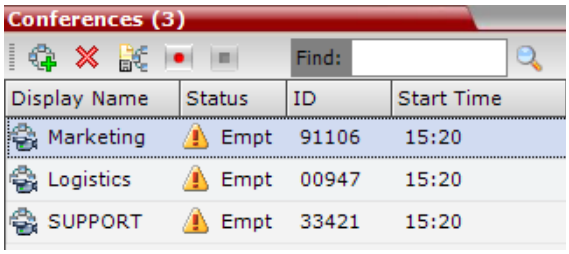


Recording Participant Right-click Options

Name	Description
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.



The recording buttons will only be displayed in the conference tool bar for a conference that is recording-enabled.

Conferences List - Recording Tool bar buttons

Button	Description
	Start/Resume recording. This button toggles with the *Pause* button.
	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*.

Users, Connections, and Notes

Collaboration Server 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.

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.

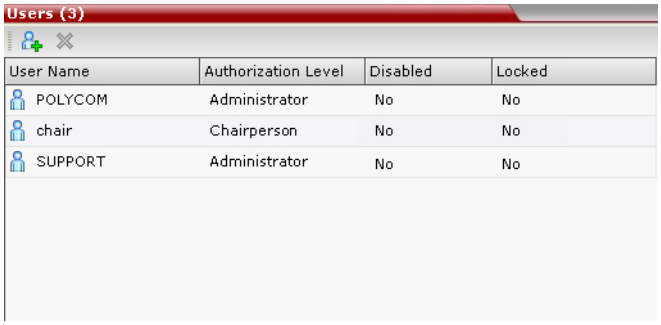
The system is shipped with a default Administrator user called POLYCOM, whose password is POLYCOM. However, once you have defined other authorized Administrator users, it is recommended to remove the default user.

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 *RMX Management* pane, click the **Users** ( ) button.

The **Users** pane is displayed.



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.
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.

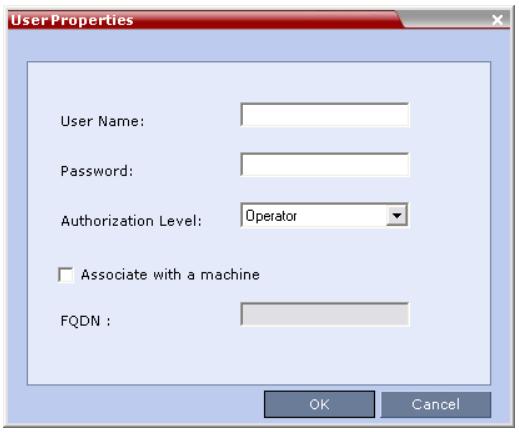
To add a new user to the system:

1 In the *RMX 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.



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 *RMX 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 *RMX 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

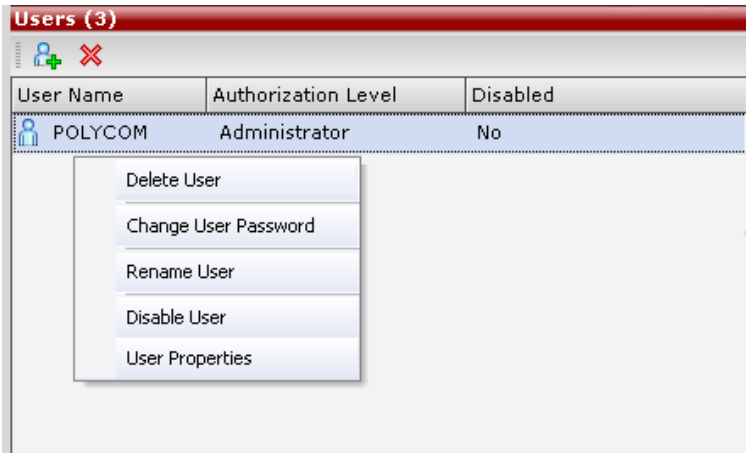
An administrator can disable an enabled user. An indication is displayed in the Users List when the User is disabled. An administrator can enable a disabled User.

To disable a user:

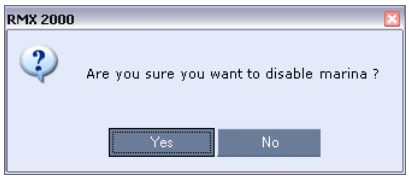
1 In the RMX *Management* pane, click the **Users** ( ) button.

The Users pane is displayed.

2 In the *Users* pane, right-click the user to be disabled and select **Disable User** in the menu.



A confirmation box is displayed.



3 Click **YES**.
The User status in the *Users* list - *Disabled* column changes to **Yes**.

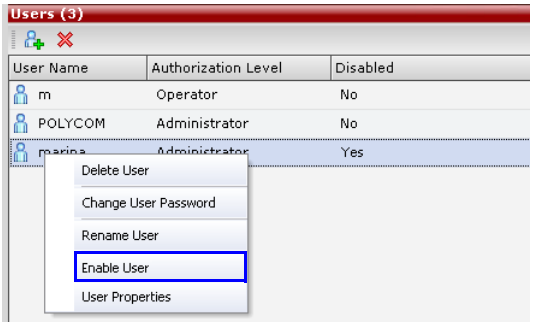
Enabling a User

An administrator can enable a User who was disabled manually by the administrator.

To enable a user:

1 In the *RMX Management* pane, click the **Users** ( ) button.
The *Users* pane is displayed.

2 Right-click the user to be enabled and select **Enable User**.



A confirmation box is displayed.

3 Click **YES**.
The User status in the *Users* list - *Disabled* column changes to **NO**.

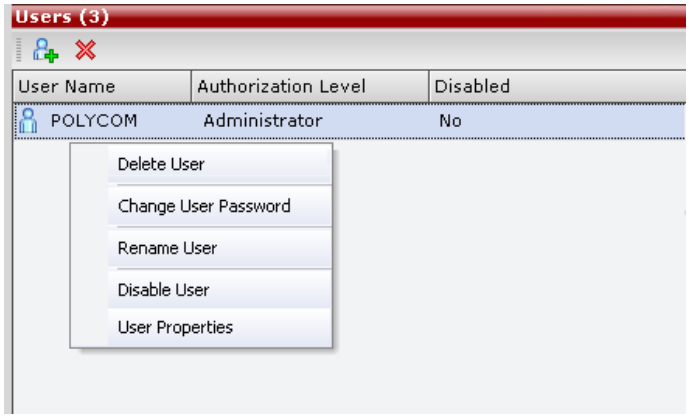
Renaming a User

To rename a user:

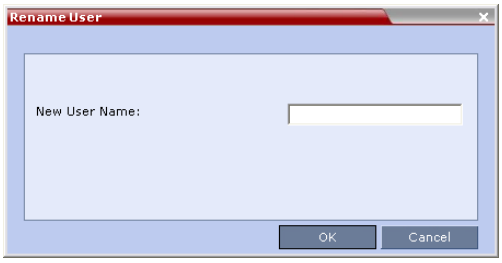
1 In the *RMX Management* pane, click the **Users** ( ) button.

The Users pane is displayed.

2 Right-click the user to be renamed and select **Rename User**.



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 for defining a machine account

- 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* system with which it interacts. This machine account must include a fully-qualified domain name (*FQDN*) for the *RealPresence CMA/DMA* 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 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

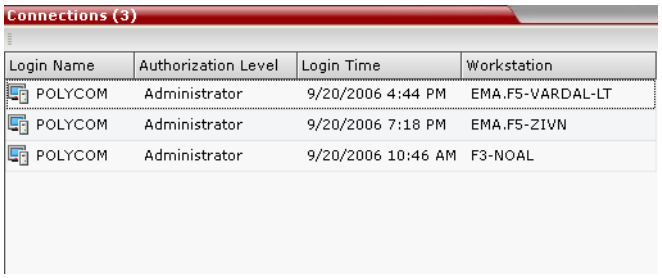
The *Collaboration Server* enables you to list all connections that are currently logged into the MCU, e.g. users, servers or API users. The MCU issues an ID number for each login. The ID numbers are reset whenever the MCU is reset.

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.



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.

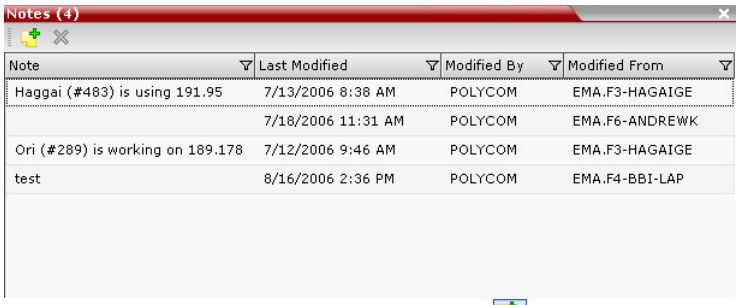
Notes can be read by all system Users concurrently connected to the MCU. Notes that are added to the *Notes* list are updated on all workstations by closing and re-opening the *Notes* window. Notes can be written in any Unicode language.

Using Notes

To create a note:

1 On the *Collaboration Server* menu, click **Administration** > **Notes**.

The **Notes** window opens.

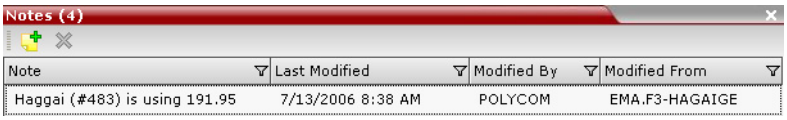


2 In the *Notes* toolbar, click the **New Note** ( ) button, or right-click anywhere inside the *Notes* window and select **New Note**.

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.



To open or edit a note:

- 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.

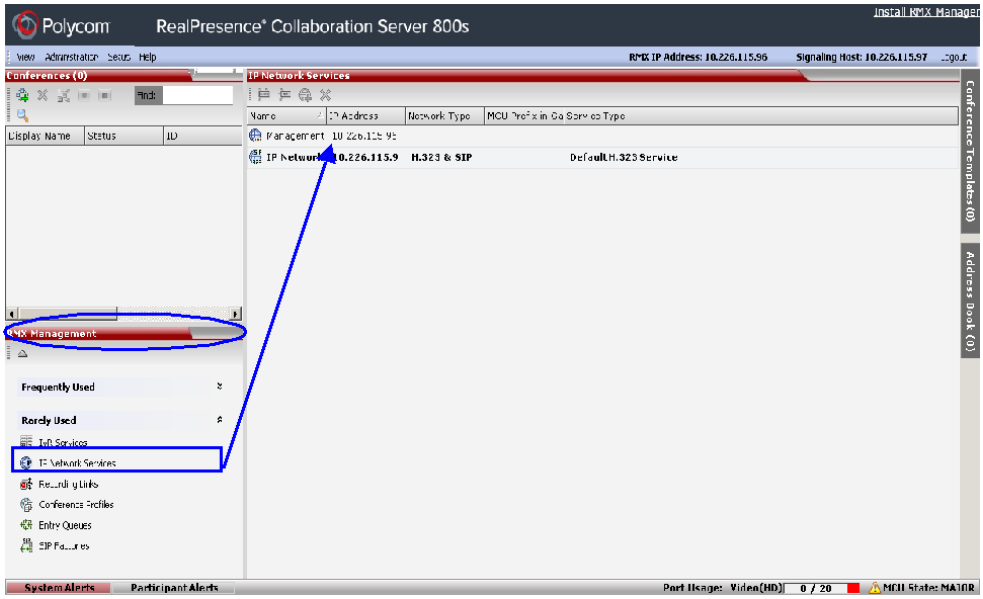
2 Click **OK** to delete the note, or click **Cancel** to keep the note.

IP Network Services

To enable the *Collaboration Server* to function within IP network environments, network parameters must be defined for the *IP Network Services.*

The Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition allows you only to view the parameters of the IP Network Services in the RealPresence Collaboration Server Web Client or the RMX Manager, 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 Manual IP Configuration.

The configuration dialog boxes for the IP network services are accessed via the *Collaboration Server Management* pane of the *Collaboration Server Web Client*.



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 the following IP addressing environments:

- IPv4
- IPv6

- IPv6 & IPv4

When *IPv4* is selected, IPv6 fields are not displayed and conversely when *IPv6* is selected, *IPv4* fields are not displayed. When *IPv6 & IPv4* is selected both *IPv6* and *IPv4* fields are displayed.

For the purposes of comprehensive documentation, all screen captures in this chapter show the dialog boxes as displayed with *IPv6 & IPv4* selected.

Management Network (Primary)

The *Management Network* is used to control the *Collaboration Server*, mainly via the *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 *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 the RealPresence Collaboration Server 800s, the *Management Network* parameters can be set either via a *USB memory stick* or by using a cable to create a private network, during *First Time Power-up*. For more information, see the*Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Installing the RealPresence Collaboration Server 800s and Appendix G - Configuring Direct Connections to the Collaboration Server.

In the RealPresence Collaboration Server Virtual Edition with DHCP available, the *Management Network* parameters are automatically set during *First Time Power-up* and whenever the Collaboration Server is restarted. In the RealPresence Collaboration Server Virtual Edition without DHCP available, the *Management Network* properties must be set manually. For more information, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Manual IP Configuration.

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 *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.

On the RealPresence Collaboration Server 800s, most of the *Default IP Service* is configured by the *Fast Configuration Wizard,* which runs automatically should the following occur:

- First time power-up.
- Deletion of the D*efault IP Service*, followed by a system reset.

On the RealPresence Collaboration Server Virtual Edition with DHCP available, the D*efault IP Service* parameters are automatically set during *First Time Power-up* and whenever the Collaboration Server is restarted. On the RealPresence Collaboration Server Virtual Edition without DHCP available, the D*efault IP Service* properties must be set manually. For more information, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Manual IP Configuration.

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 800s

The *Management Network* parameters need to be modified if you want to:

- Connect directly to the *Collaboration Server* from a workstation
- Modify routes
- Modify DNS information

To view or modify the Management Network Service:

1 In the **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.

3 Modify the following fields:

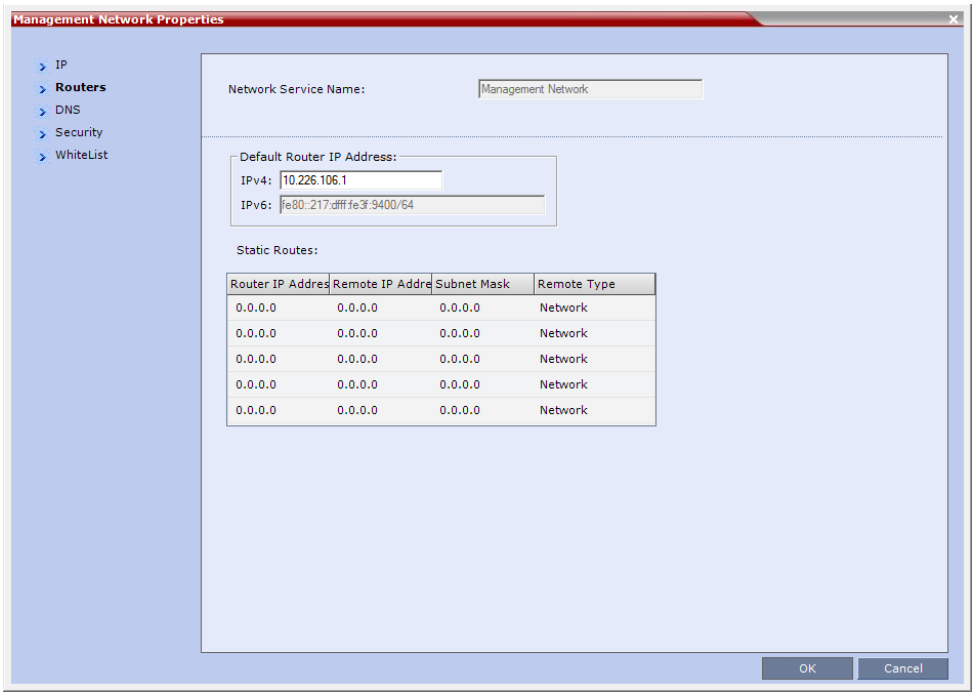
RealPresence Collaboration Server 800s - Default Management Network Service – IP

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.	
IP Version	IPv4	Select this option for IPv4 addressing only.
	IPv6	Select this option for IPv6 addressing only.
	IPv4 & IPv6	Select this option for both IPv4 and IPv6 addressing. Note: If the gatekeeper cannot operate in **IPv6** addressing mode, the H323_RAS_IPV6 **System Flag** should be set to NO. For more information see Manually Added Flags - CS_MODULE_PARAMETERS Tab.

Field	Description		
IPv6 Configuration Method Manual Configuration Method is recommended with IPv6.	Auto (Stateless)	Select this option to allow automatic generation of the following addresses: • Link-Local (For internal use only) • Site-Local • Global	
	Manual	Select his option to enable manual entry of the following addresses: • Site-Local • Global Manual configuration of the following address types is not permitted: • Link-Local • Multicast • Anycast	
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.	
	IPv6	The IPv6 address of the MCU. This IP address is used by the *Collaboration Server* **Web Client** to connect to the Collaboration Server. Note: **Internet Explorer 7™** is required for the *Collaboration Server* **Web Client** to connect to the MCU using IPv6.	
		All	Click the **All** button to display the IPv6 addresses as follows: • **Auto** - If selected, **Site-Local** and **Global** site addresses are displayed. • **Manual** - If selected, only the **Manual** site address is displayed.

Field	Description		
Shelf Management IP Address	IPv4	The IPv4 address of the **RMX Shelf Management Server**. This IP address is used by the *Collaboration Server* **Web Client** for **Hardware Monitoring** purposes.	
	IPv6	The IPv6 address of the **RMX Shelf Management Server.** This IP address is used by the *Collaboration Server* **Web Client** for **Hardware Monitoring** purposes. **Note: Internet Explorer 7™** is required for the *Collaboration Server* **Web Client** to connect to the MCU using IPv6.	
		All	Click the **All** button to display the **IPv6** addresses as follows: • **Auto** - If selected, **Site-Local** and **Global** site addresses are displayed. • **Manual** - If selected, only the **Manual** site address is displayed.
Subnet Mask	Enter the subnet mask of the Control Unit. **Note:** This field is specific to IPv4 and is not displayed in IPv6 only mode.		

4 Click the **Routers** tab.

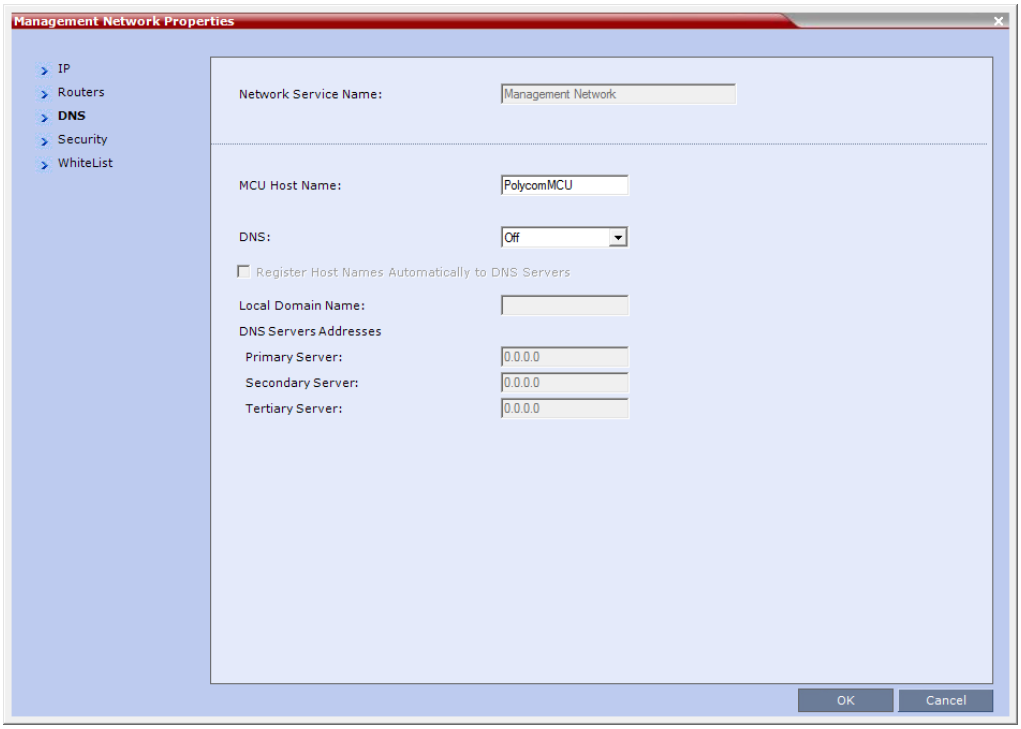


5 Modify the following fields:

RealPresence Collaboration Server 800s - Default Management Network Service – Routers

Field	Description	
Default Router IP Address	IPv4	Enter 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	
Static Routes IPv4 Only Table		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	Enter the IP address of the router.
	Remote IP Address	Enter 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	Enter the subnet mask of the remote network.
	Remote Type	Select 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.

6 Click the **DNS** tab.



7 Modify the following fields:

RealPresence Collaboration Server 800s - Default Management Network Service – DNS

Field	Description
MCU Host Name	Enter the name of the MCU on the network. Default name is RMX
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.
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	

The Secuity and Whitelist tabs are not supported in the RealPresence Collaboration Server 800s.

8 Click **OK.**

9 If you have modified the *Management Network Properties*, reset the MCU.

Modifying the Default IP Network Service in the RealPresence Collaboration Server 800s

The *Default IP Service* parameters need to be modified if you want to change the:

- Network type that the *Collaboration Server* connects to
- IP address of the *Collaboration Server* Signaling Host
- Gatekeeper parameters or add gatekeepers to the Alternate Gatekeepers list
- SIP server parameters

Fast Configuration Wizard

The *Fast Configuration Wizard* enables you to configure the *Default IP Service*. It starts automatically if no *Default IP Network Service* is defined. This happens during *First Time Power-up*, before the service has been defined or if the *Default IP Service* has been deleted, followed by an *Collaboration Server* restart.

The *IP Management Service* tab in the *Fast Configuration Wizard* is enabled only if the factory default *Management IP addresses* were not modified.

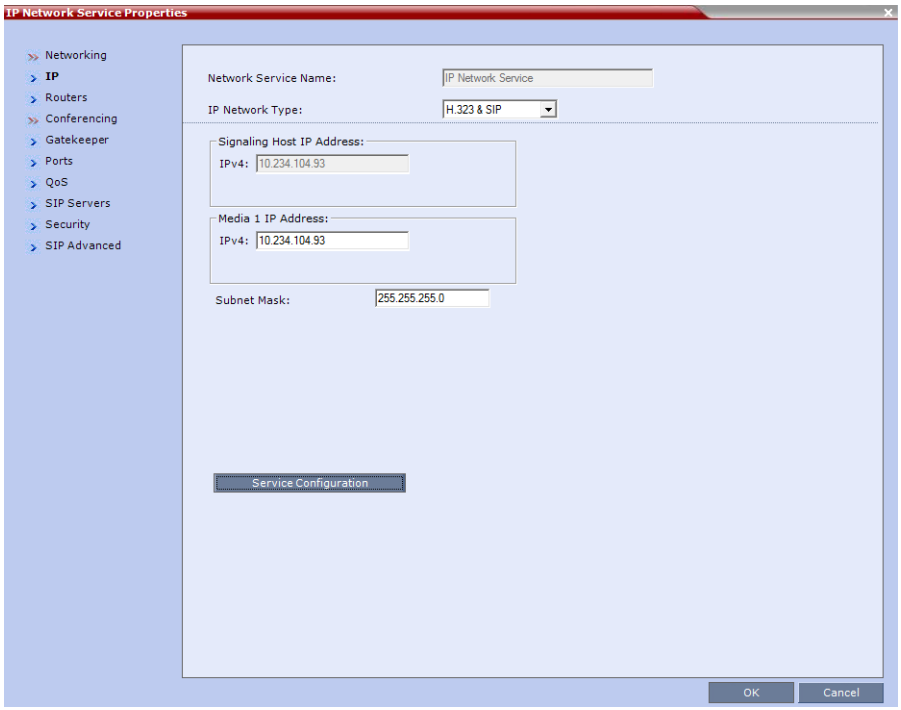
If the *Fast Configuration Wizard* does not start automatically, the *Default IP Service* must be modified through the *IP Network Properties* dialog boxes.

To view or modify the Default IP Service:

1 In the *Collaboration Server Management* pane, click **IP Network Services** ( ).

2 In the *Network* list pane, double-click the **Default IP Service** (, , or) entry.

The *Default IP Service - Networking IP* dialog box opens.



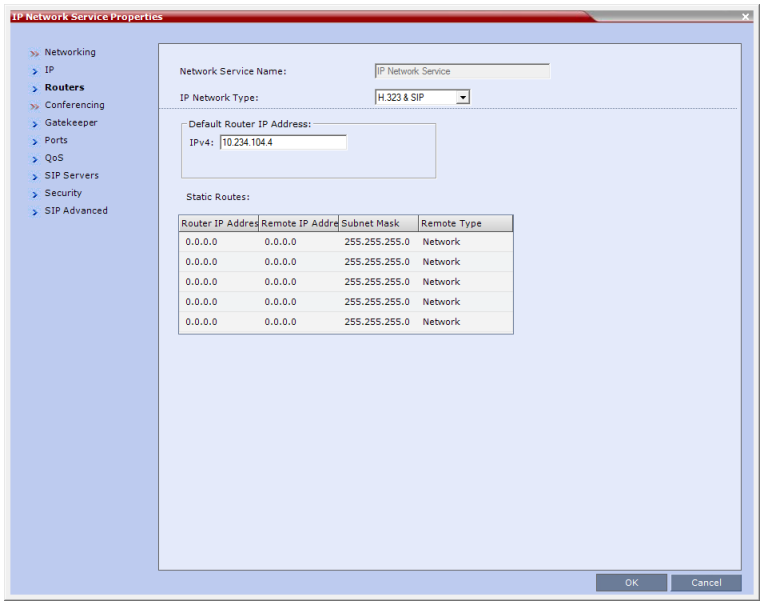
3 Modify the following fields:

RealPresence Collaboration Server 800s - 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 name can 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	Enter the address to be used by IP endpoints when dialing in to the MCU. 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.
Media Card 1 IP Address	Enter the address to be used by IP endpoints when dialing in to the MCU.

Field	Description
Subnet Mask	Enter the subnet mask of the MCU. Default value: 255.255.255.0.

4 Click the **Routers** tab.



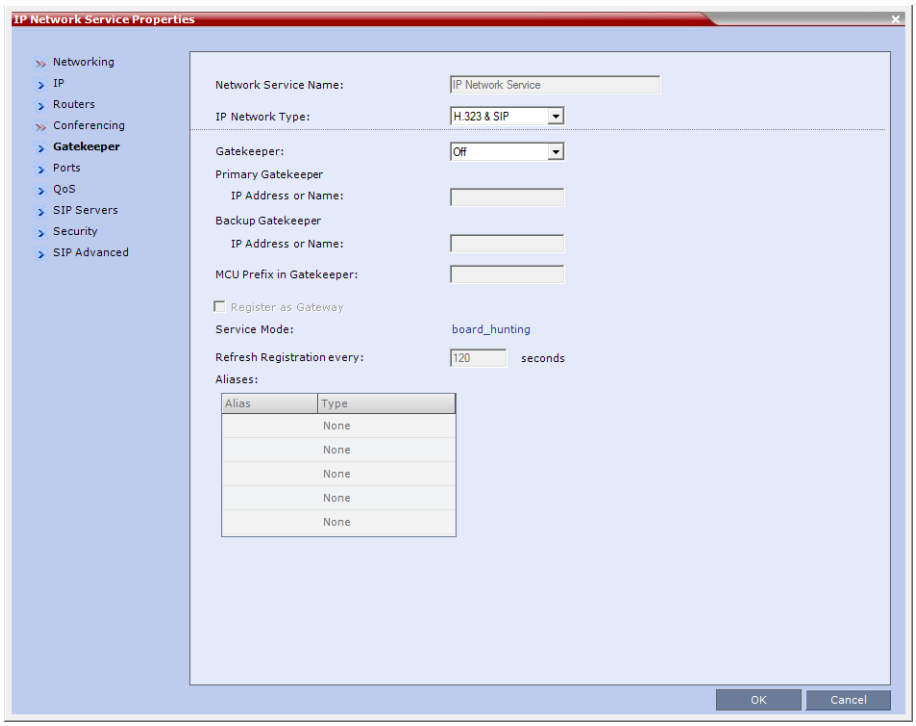
With the exception of *IP Network Type*, the field definitions of the *Routers* tab are the same as for the *Default Management Network.* For more information see step 3.

5 **Optional.** Click the **DNS** tab.

Settings in this dialog box are relevant to *Multiple Network Services* only.

For more information see NAT (Network Address Translation) Traversal.

6 Click the **Gatekeeper** tab.



7 Modify the following fields:

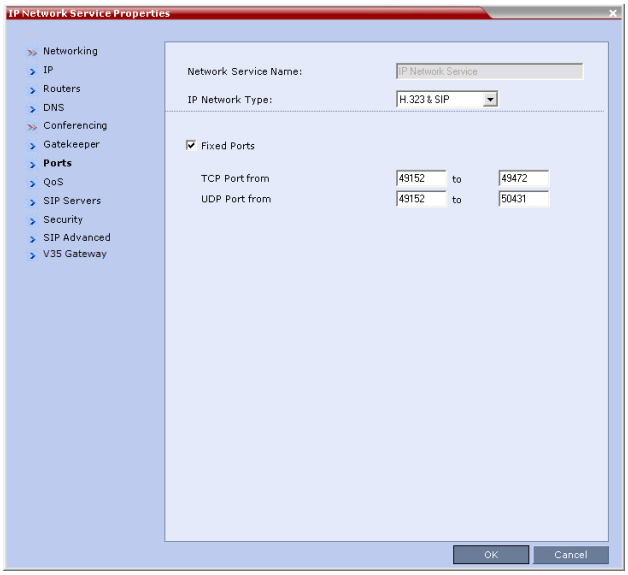
RealPresence Collaboration Server 800s - Default IP Service – Conferencing – Gatekeeper

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.	**Note:** When in *IPv4&IPv6* or in *IPv6* mode, it is easier to use *Names* instead of *IP Addresses*.
Alternate 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.	
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 he 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) • 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.	

Parameters

8 Click the **Ports** tab.

Settings in the *Ports* tab allow specific ports in the firewall to be allocated to multimedia conference calls.



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:

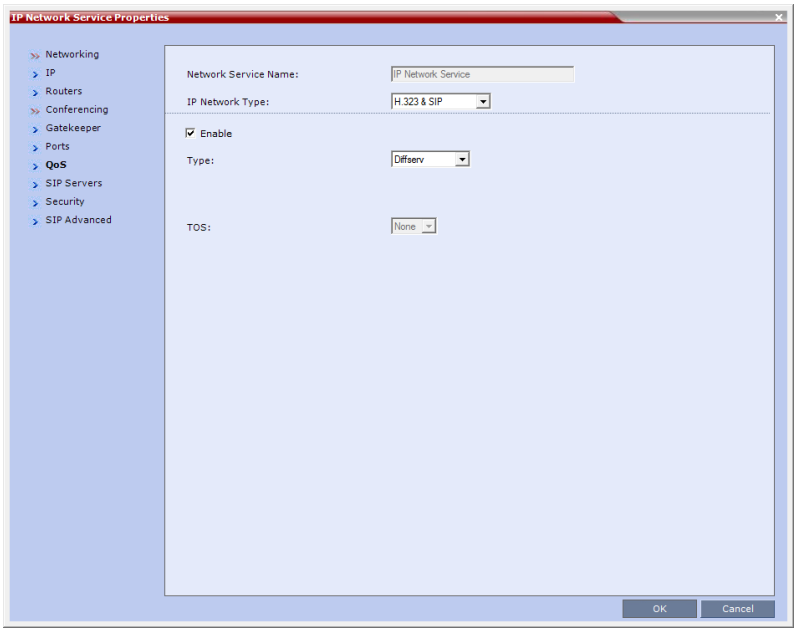
RealPresence Collaboration Server 800s - Default IP Service – Conferencing – Ports

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). **Note:** When ICE Environment is enabled, 8 additional ports are allocated to each call. 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. When ICE environment is enabled, the range must be **2048** ports per media card.

Parameters

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.



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.

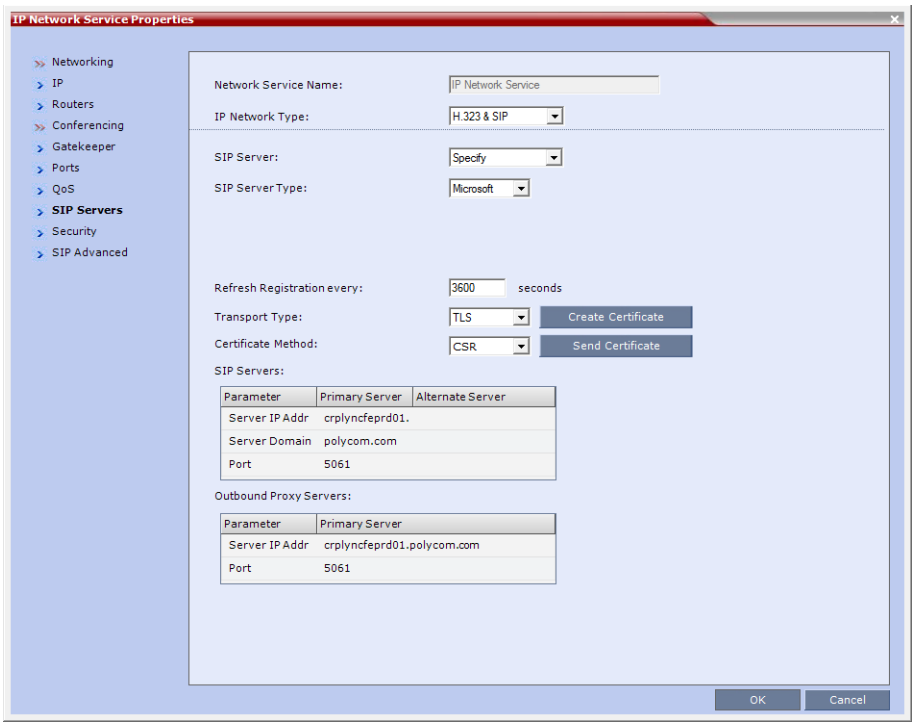
11 View or modify the following fields:

RealPresence Collaboration Server 800s - Default IP Service – Conferencing – QoS

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.
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 0x31. These values are determined by the QOS_IP_VIDEO and QOS_IP_AUDIO flags in the **system.cfg** file. The default priority for Signaling IP traffic is 0x00 and is determined by the QOS_IP_SIGNALING flag in the **system.cfg** file. For more information Modifying System Flags • **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.

Parameters

12 Click the **SIP Servers** tab.



13 Modify the following fields:

RealPresence Collaboration Server 800s - Default IP Network Service – SIP Servers

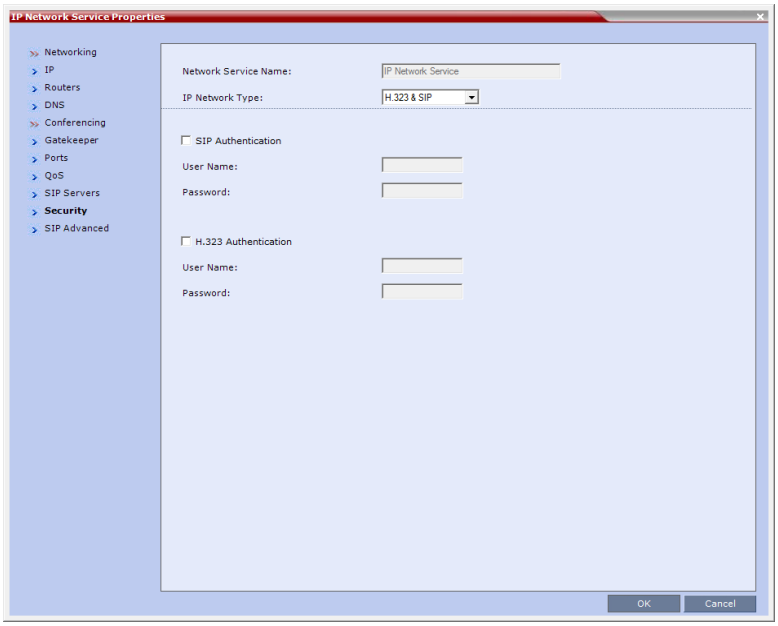
Field	Description
SIP Server	Select: • **Specify** – to manually configure SIP servers. • **Off** – if SIP servers are not present in the network.
SIP Server Type	Select: • Generic - for non Microsoft environments. • Microsoft - for Microsoft environments.
Refresh Registration	This defines the time in 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.

Field	Description
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.
Skip Certificate Validation	When checked, no Certificate Validation is performed.
Revocation Method	For a detailed description of these fields see the **Ultra Secure Mode** chapter, Certificate Management and Certificate Revocation.
Global Responder URL	
Use Responder Specified in Certificate	
Allow Incomplete Revocation Checks	
Skip Certificate Validation for OSCP Responder	
SIP Servers: Primary / Alternate Server Parameter	
Server IP Address	Enter the IP address of the preferred SIP server. If a DNS is used, you can enter the SIP server name. **Note:** When in IPv4&IPv6 or in IPv6 mode, it is easier to use Names instead of **IP Addresses**.
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.

Field	Description
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). **Note:** When in IPv4&IPv6 or in IPv6 mode, it is easier to use **Names** instead of **IP Addresses**.
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.



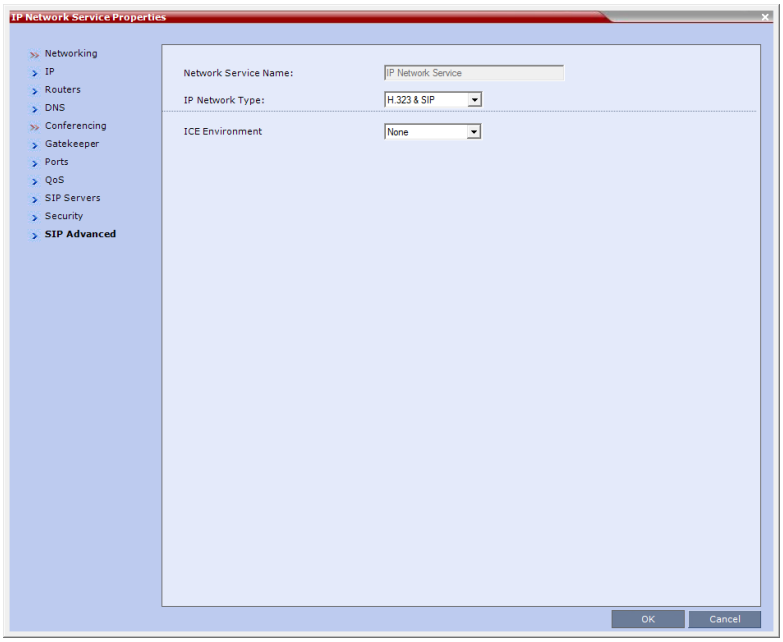
15 Modify the following fields:

RealPresence Collaboration Server 800s - 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.	
	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 SIP proxy. 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.

16 Optional. To configure the ICE environment, click the **SIP Advanced** tab.



17 Modify the following fields:

RealPresence Collaboration Server 800s - Default IP Network Service – SIP Advanced

Field	Description
ICE Environment	Select **MS** (for **Microsoft ICE** implementation) to enable the **ICE** integration.
Server User Name	Enter the *Collaboration Server* **User** name as defined in the **Active Directory**. For example, enter **rmx1234**. This field is disabled if the **ICE Environment** field is set to **None**.

18 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.

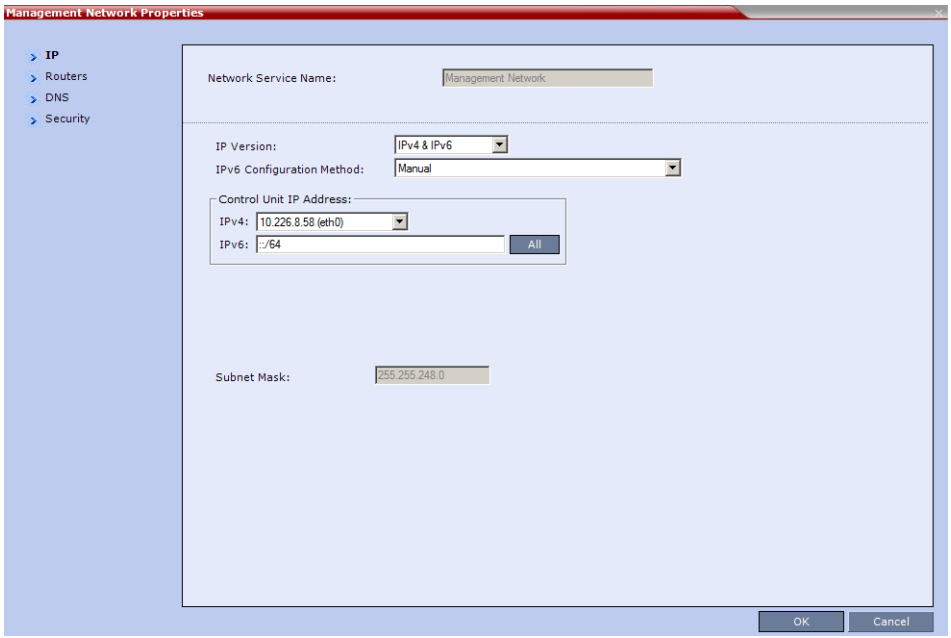
Viewing the Management Network in the RealPresence Collaboration Server Virtual Edition

In the Polycom® RealPresence® Collaboration Server 800s and Polycom® 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 800s / Virtual Edition Getting Started Guide*, Manual IP Configuration on page 21.

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.

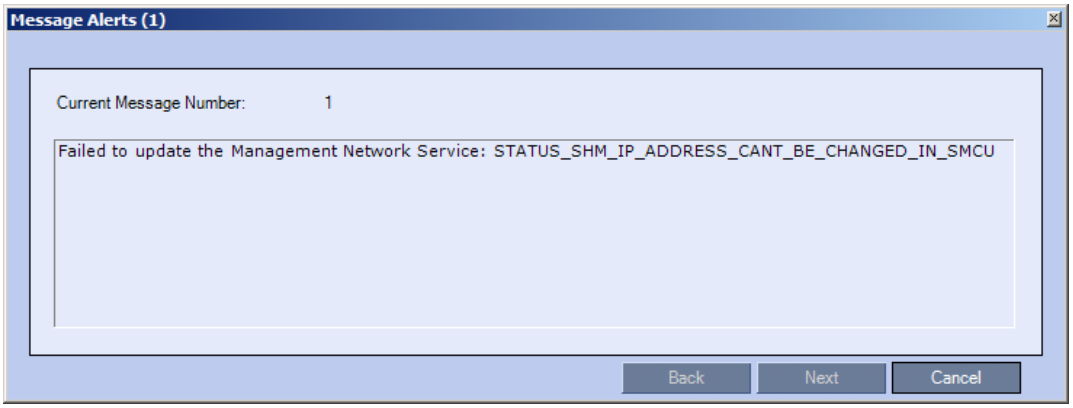


The following fields can be viewed, but can not be modified:

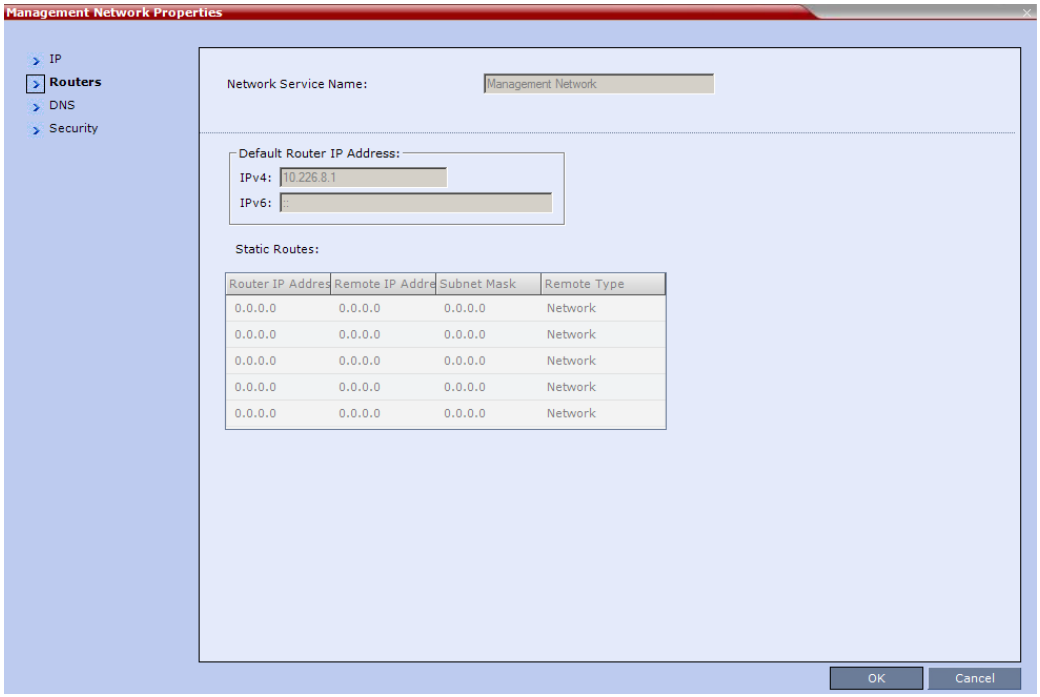
RealPresence Collaboration Server Virtual Edition - Default Management Network Service – IP

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:



3 Click the **Routers** tab.



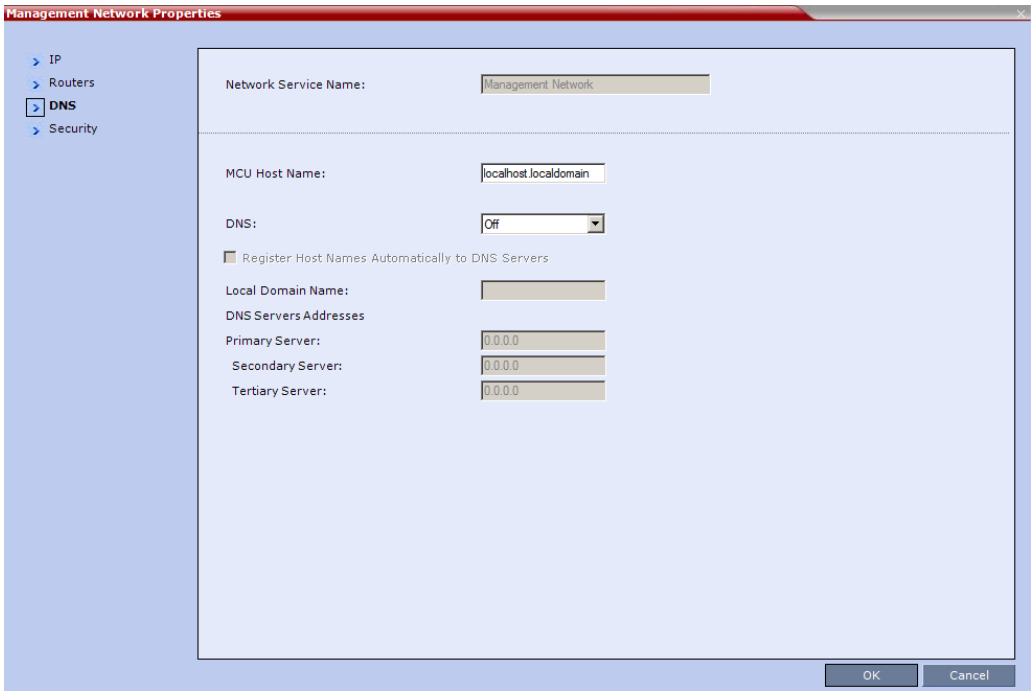
The following fields can be viewed but not modified.

RealPresence Collaboration Server Virtual Edition - Default Management Network Service –

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	
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.

Routers

4 Click the **DNS** tab.

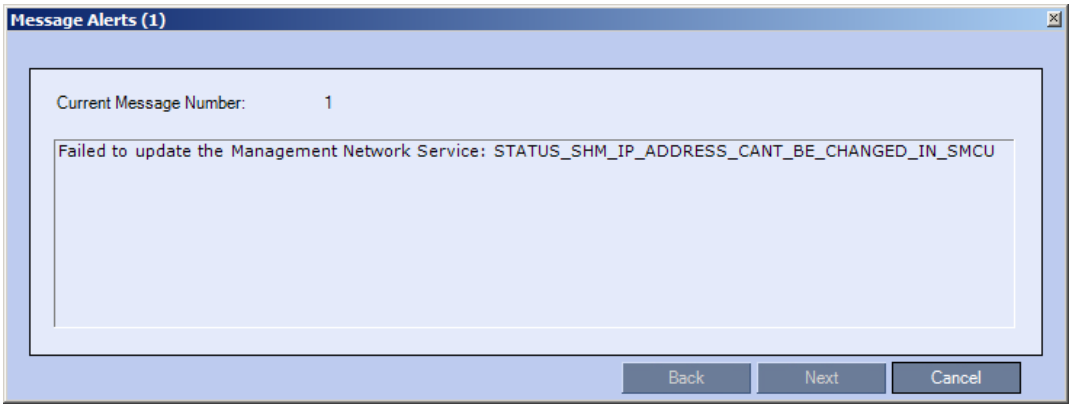


The following fields can be modified, but their values will not be applied:

RealPresence Collaboration Server Virtual Edition - 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:



5 Click **OK.**

IP Network Monitoring

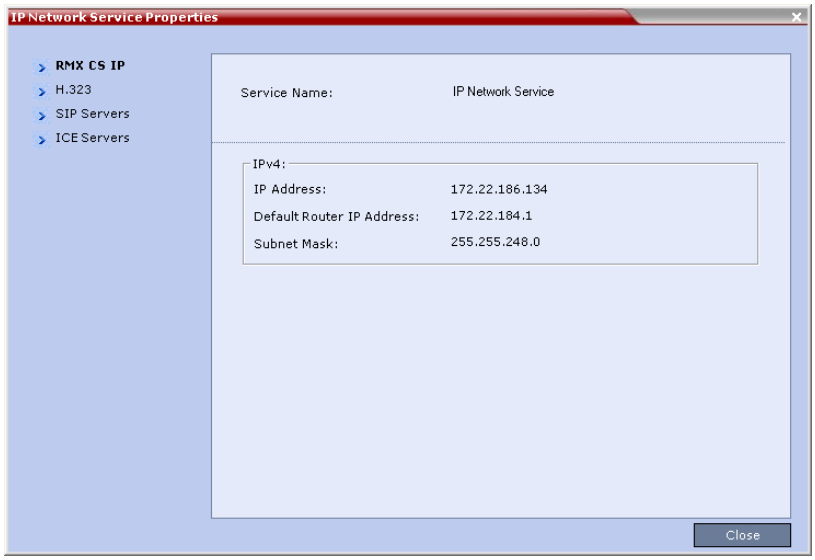
The **Signaling Monitor** is the Collaboration Server entity used for monitoring the status of external network entities such as the gatekeeper, DNS, SIP proxy and Outbound proxy and their interaction with the MCU.

To monitor signaling status:

1 In the **RMX 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:

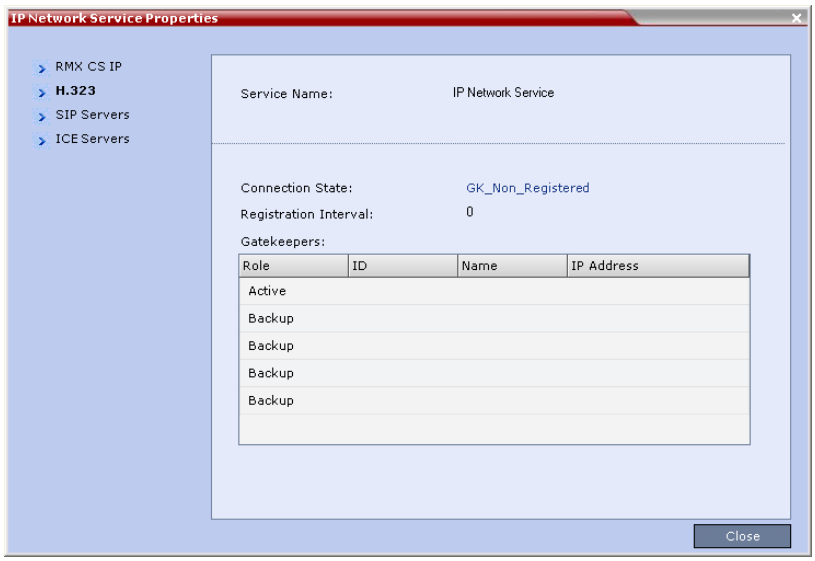


The **RMX CS IP** tab displays the following fields:

IP Network Services Properties – RMX CS IP

Field	Description		
Service Name	In the RealPresence Collaboration Server 800s, the name assigned to the **IP Network Service** by the **Fast Configuration Wizard**. In the RealPresence Collaboration Server Virtual Edition, this is always, “IP Network Service.” **Note:** This field is displayed in all tabs.		
IPv4	IP Address		
	Default Router IP Address	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.	
	Subnet Mask	The subnet mask of the MCU. Default value: 255.255.255.0.	
IPv6	Scope	IP Address **Note:**	
		Global	The Global Unicast IP address of the Collaboration Server.
		Site-Local	The IP address of the Collaboration Server within the local site or organization.
	Default Router IP Address	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.	

3 Click the **H.323** tab.

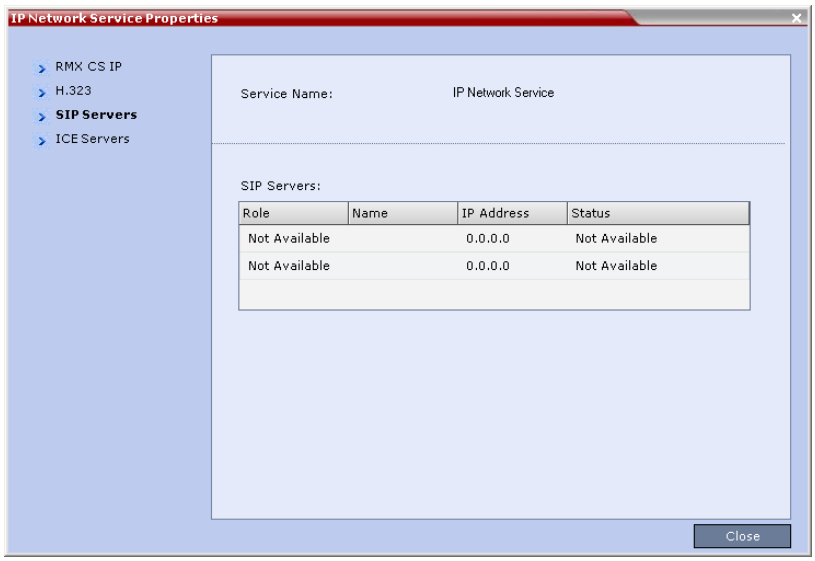


The **H.323** tab displays the following fields:

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.

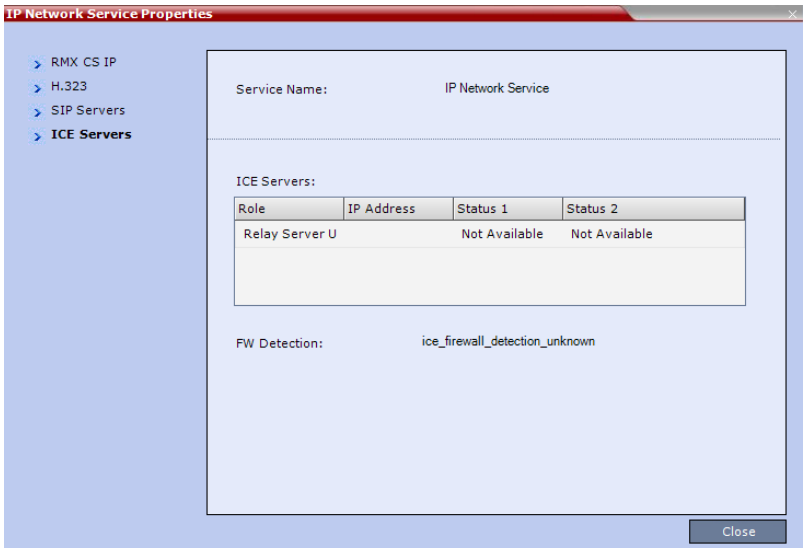


The **SIP Servers** tab displays the following fields:

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.



The **ICE Servers** tab displays the following fields:

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 In systems with multiple media cards, Status 1 refers to the uppermost media card.

IP Network Services Properties – ICE Servers

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

Using IPv6 Networking Addresses for Collaboration Server Internal and External Entities

IPv6 addresses can be assigned to both *Collaboration Server* (*Internal*) and *External Entity* addresses.

Collaboration Server Internal Addresses

Default Management Network Service

- Control Unit
- Signaling Host
- Shelf Management

External Entities

- Gatekeepers (Primary & Secondary)
- SIP Proxies
- DNS Servers
- Default Router
- Defined participants

IPv6 Guidelines

- *Internet Explorer 7*™ is required for the *Collaboration Server Web Client* and *RMX Manager* to connect to the *Collaboration Server* using *IPv6*.
- The default IP address version is *IPv4*.
- The IP address field in the *Address Book* entry for a defined participant can be either *IPv4* or *IPv6*. A participant with an IPv4 address cannot be added to an ongoing conference while the *Collaboration Server* is in *IPv6* mode nor can a participant with an *IPv6* address be added while the *Collaboration Server* is in *IPv4* mode.

  An error message, *Bad IP address version,* is displayed and the *New Participant* dialog box remains open so that the participant's address can be entered in the correct format.

- Participants that do not use the same IP address version as the *Collaboration Server* in ongoing conferences launched from *Meeting Rooms*, *Reservations* and *Conference Templates,* and are disconnected. An error message, *Bad IP address version,* is displayed.

  IP Security (IPSec) Protocols are not supported.

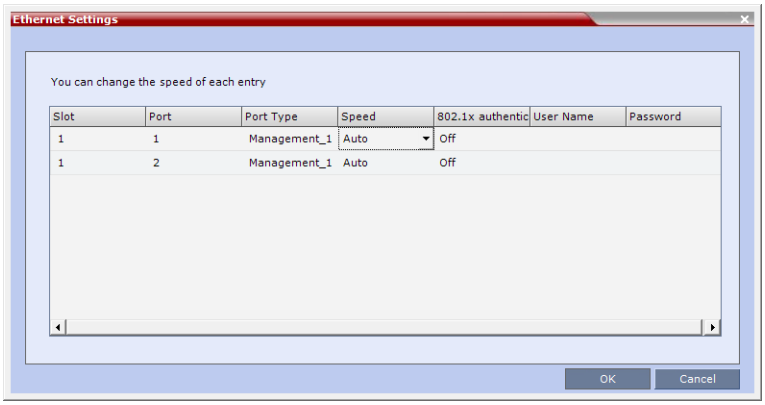
Ethernet Settings

In the *RealPresence Collaboration Server 800s*, the automatically identified speed and transmit/receive mode of each LAN port used by the system can be manually modified if the specific switch requires it.

To modify the automatic LAN port configuration:

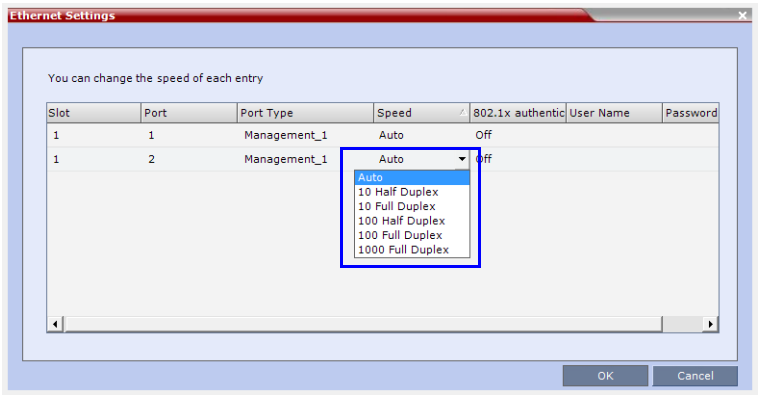
1 On the *Collaboration Server* menu, click **Setup > Ethernet Settings**.

The *Ethernet Settings* dialog box opens.



The Collaboration Server has 2 LAN ports. You can select the speed and transmit/receive mode manually for these ports.

2 In the *Speed* column, click the drop-down arrow of the table entry to modify and select the speed and the transmit/receive mode for each port:



When **Auto** (default) is selected, the negotiation of speed and transmit/receive mode starts at 1000 Mbits/second Full Duplex, proceeding downward to 100 Mbits/second Half Duplex.

Note: To maximize conferencing performance, especially in high bit rate call environments, a 1Gb connection is recommended.

3 Click the **OK** button.

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)

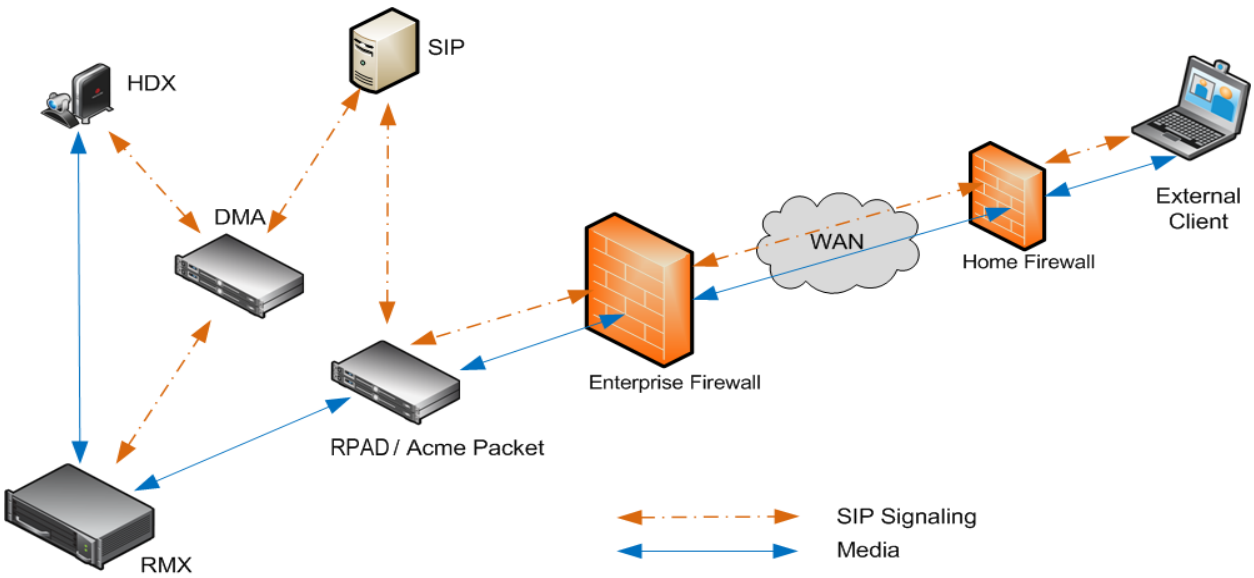
All signaling and media for both SIP and H.323 will be routed through an **SBC**.
The following **SBC** environments are supported:

- **SAM** - 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

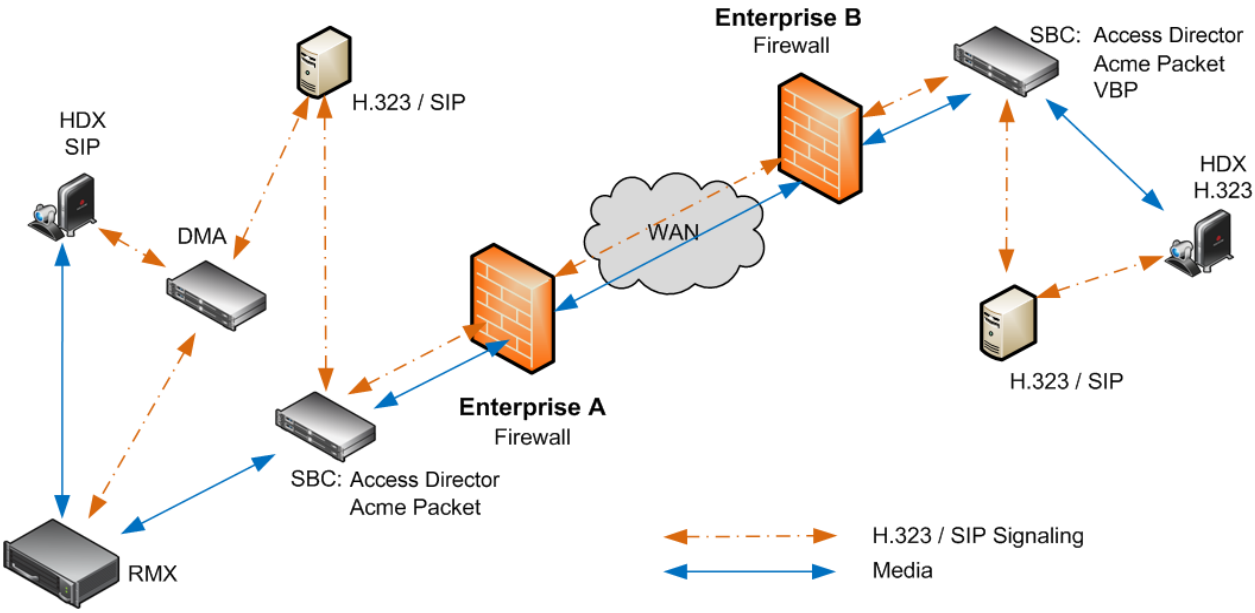


The following **Remote Connection** call flow options are supported:

Remote Connections

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
SIP / H.323	No	SAM Only	⇔	No	H.323

Business to Business Connections



The following **Business to Business** connection call flow options are supported:

Business to Business Connections

Enterprise A Client				Enterprise B Client		
Environment	**Registered**	**SBC**		**SBC**	**Registered**	**Environment**
H.323	Yes	Access Director	⇔	Access Director	Yes	H.323
H.323	Yes	Access Director	⇔	VBP	Yes	H.323
SIP	Yes	Access Director	⇔	Access Director	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.

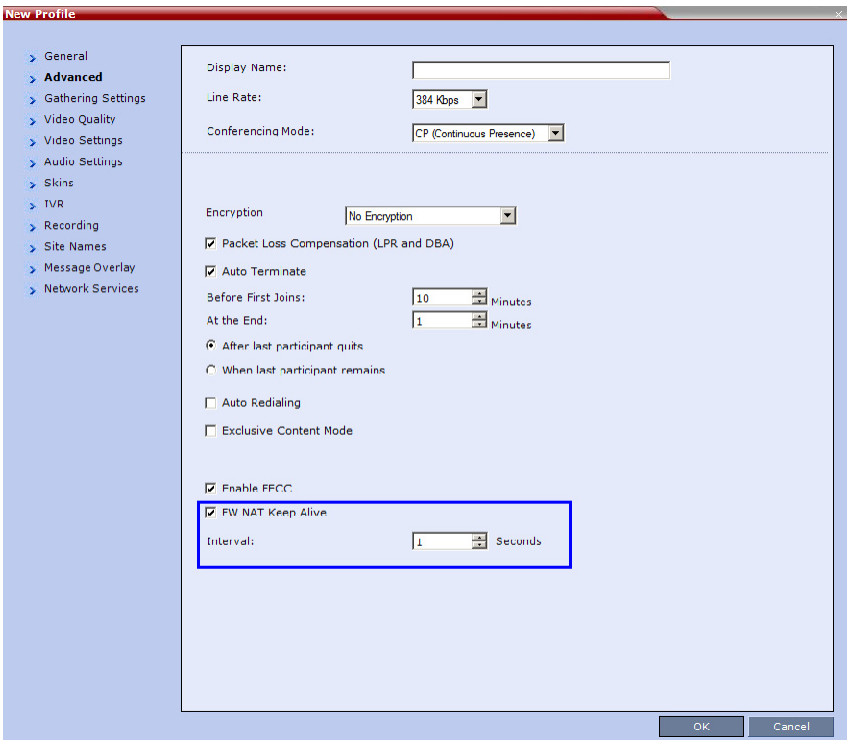
This is necessary because port mappings in the firewall are kept open only if there is network traffic in both directions. The firewall will only allow **UDP** packets into the network through ports that have been used to send packets out.

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 **5 - 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.

SIP Proxy Failover With Polycom® Distributed Media Application™ (DMA™) 7000

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.

Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition Network Port Usage

The following table summarizes the port numbers and their usage in the *Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition*:

Collaboration Server Network Port Usage Summary

Connection Type	Port Number	Protocol	Description	Configurable
HTTP	80	TCP	Management between the *Collaboration Server* and Collaboration Server *Web Client*.	No
HTTPS	443	TCP	Secured Management between the *Collaboration Server* and *Collaboration Server Web Client*.	No
DNS	53	TCP	Domain name server.	Can be disabled in the IP Network Service.
DHCP	68	TCP	Dynamic Host Configuration Protocol.	Can be disabled in the IP Network Service.
SSH	22	TCP	Secured shell. It is the *Collaboration Server* terminal.	No

Connection Type	Port Number	Protocol	Description	Configurable
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.
H.323 H.245	49152 - 59999	TCP	H.245 control. Each outgoingx 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.

Connection Type	Port Number	Protocol	Description	Configurable
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.
SIP -TLS	5061	TCP	SIP -TLS for SIP server, alternate SIP server, outbound proxy and alternate outbound proxy.	No

LAN Redundancy

LAN Redundancy is applicable to the RealPresence Collaboration Server 800s only.

LAN Redundancy enables the redundant LAN port connection to automatically replace the failed port by using another physical connection and NIC (Network Interface Card). When a LAN port fails, IP network traffic failure is averted and network or endpoints disconnections do not occur. When LAN cables are connected to both LAN 2 and LAN 3 ports, the *Collaboration Server* automatically selects which port is active and which is redundant.

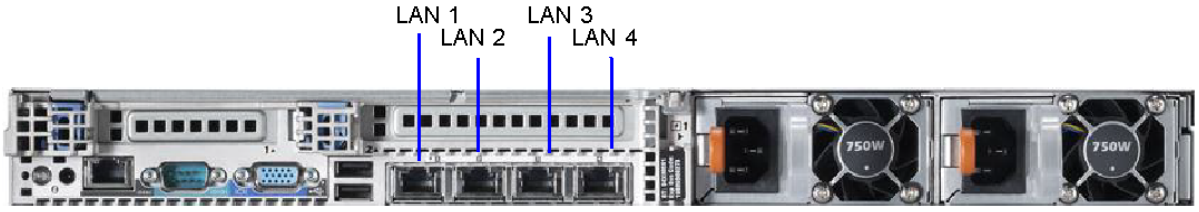
Configuration Requirements

LAN Redundancy is available by default and is enabled by connecting the additional LAN cable to LAN 3, or LAN 4.

Signaling and Media Redundancy

On the RealPresence Collaboration Server 800s, LAN2 is used for media and signaling and LAN 3, and LAN 4 are the redundant media ports:

Collaboration Server - Rear View



Media Redundancy on the *Collaboration Server* is dependent on the settings of the **LAN_REDUNDANCY** and **MULTIPLE_SERVICES** *System Flags* as summarized in the table below.

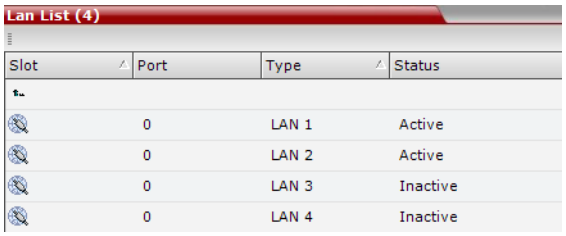
Collaboration Server - Media Redundancy - System Flags

System Flag / Value	Collaboration Server
LAN_REDUNDANCY = NO MULTIPLE_SERVICES = NO	No redundancy
LAN_REDUNDANCY = NO MULTIPLE_SERVICES = YES	
LAN_REDUNDANCY = YES MULTIPLE_SERVICES = NO	Full signaling and media redundancy

Hardware Monitor Indications

With LAN redundancy, when LAN LEDs are lit they indicate that a physical connection of the cables is present but does not indicate their activity status.

In the *Hardware Monitor* pane the *Lan List* displays the *Collaboration Server* LAN ports together with their *Status* indication.



LAN Indications

Status	Description
Active	The LAN port cable is connected.
Inactive	The LAN port cable is not connected.
Standby	The LAN Redundancy option is enabled and this LAN port is the redundant and in standby mode. In case of failure, this port becomes active.

Multiple Network Services

Multiple Network Services are applicable to the RealPresence Collaboration Server 800s only.

Media, signaling and management networks can be physically separated on the *Collaboration Server* system to provide enhanced security. This addresses the requirement in an organization that different groups of participants be supported on different networks. For example, some participants may be internal to the organization while others are external.

Up to three media and signaling networks can be defined for the *RealPresence Collaboration Server 800s* for each media and signaling network connected to the *Collaboration Server*.

The *Management Network* is logically and physically separated from the media and signaling networks. There can be one *Management Network* defined per *Collaboration Server* system.

Each conference on the *Collaboration Server* can host participants from the different IP Network networks simultaneously.

Guidelines

- Multiple Services system mode is a purchasable option and it is enabled in the MCU license.
- Multiple Services system mode is enabled when the system configuration flag **MULTIPLE_SERVICES** is added and set to **YES**.
- On the *RealPresence Collaboration Server 800s*, LAN redundancy cannot be enabled in parallel to Multiple Networks and the **LAN_REDUNDANCY** flag must be set to **NO** when the Multiple Networks option is enabled.
- Participants on different networks can connect to the same conference with full audio, video and content capabilities.
- Traffic on one network does not influence or affect the traffic on other networks connected to the same MCU. If one network fails, it will not affect the traffic in the other connected networks.
- The Signaling Host IP Address and the Media IP Address cannot be different.
- Maximum number of services that can be defined per *Collaboration Server* platform:

  Maximum Number of Network Services

- A *DNS* server can be specified for each *IP Network Service* and for the *Collaboration Server Management Network Service*.
  - In the Network Services that do not include the DNS server, use the IP addresses of the various devices to define them in the Network Services.
- Participants are associated with a Network Service and use it resources as follows:
  - Dial-in participants - according to the network used to place the call and connect to the *Collaboration Server*.
  - Dial-out participant - according to the Network Service selected during the participant properties definition or during conference definition, according to the Network Service selected as default.

Resource Allocation and Capacity

The *Resolution Configuration* settings are configured per MCU and affect the resource capacity of the MCU. They are reflected in the port gauges displayed on the *Collaboration Server* management application's main screen.

In *Multiple Networks* mode, the port gauges do not reflect the resource availability per Network Service.

In *Multiple Networks* mode, the resources of the Network Services are not split between the network services but are used per their availability by all Network Services equally.

First Time Installation and Configuration

First Time Installation and Configuration of the *Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition* consists of the following procedures:

1 Preparations

- Gather Network Equipment and Address Information - get the information needed for integrating the *Collaboration Server* into the local network for each of the networks that will be connected to the *Collaboration Server* unit. For a list of required address, see the *RealPresence Collaboration Server Getting Started Guide,* Gather Network Equipment and Address Information.
- Unpack the *Collaboration Server*. For more details see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Unpacking the RealPresence Collaboration Server 800s.
- Modify the *Management Network parameters* on the USB Key. For more details see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Modifying the Factory Default Management Network Settings on the USB Memory Stick.

2 Hardware Installation and Setup

- Mount the *Collaboration Server* in a rack. For more details see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* Hardware Installation and Rack Mounting.
- Connect the necessary cables. For details, see *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* Connecting the Cables to the MCU.

3 First Entry Power-up and Configuration

- Power up the *Collaboration Server*. For more details see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* Procedure 1: First-time Power-up.
- Register the *Collaboration Server*. For more details see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* Procedure 2: Product Registration.
- Connect to the *Collaboration Server*. For more details see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* Procedure 3: Connection to MCU.
- Configure the *Default IP Network Service* using the information for one of the networks connected to the system. For more details see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* Procedure 4: Modifying the Default IP Service Settings.

4 Modify the required System Flag to enable Multiple Services and reset the MCU.

5 Add the required IP Network Services to accommodate the networks connected to the *Collaboration Server*.

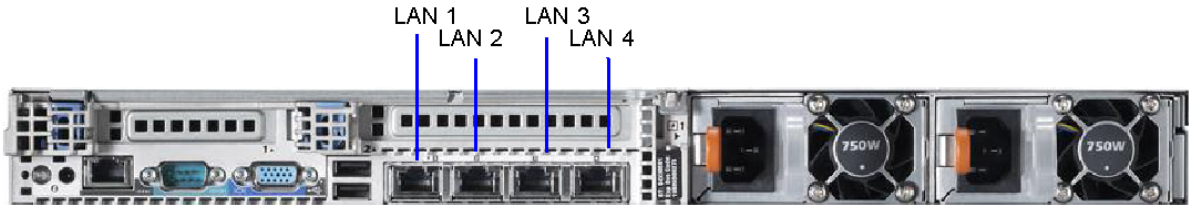
6 Select a Network Service to act as default for dial out and gateway calls for which the Network Service was not selected.

7 Place several calls and run conferences to ensure that the system is configured correctly.

Connecting the Cables to the RealPresence Collaboration Server 800s

On the *Collaboration Server* LAN2 is used for media and signaling and LAN 3 and LAN 4 can be used for multiple Networks configuration.

Collaboration Server - Rear View



Multiple Network LAN Port Usage

LAN Port	Description
1	Management Network LAN connection (mandatory)
2	Signaling and media - first IP Network Service (mandatory)
3	Signaling and media - second IP Network Service (optional)
4	Signaling and media - third IP Network Service (optional)

If LAN Redundancy is configured, Multiple Networks on LAN 3 and 4 cannot be defined.

Collaboration Server Configuration

Once the network cables are connected to the *Collaboration Server*, you can modify the default IP Network Service and add additional Network Services.

System Flags and License Settings

The **MULTIPLE_SERVICES** System Flag determines whether the Multiple Services option will be activated once the appropriate license is installed. Possible Values: **YES / NO** Default: **NO**

This flag must be manually added to the system configuration and set to YES to enable this option. For more information see the *RealPresence Collaboration Server 800s Administrator's Guide*, Manually Adding and Deleting System Flags.

IP Network Service Definition

Use this procedure to define Network Services in addition to the Network Service already defined during first entry installation and configuration. Each of the defined Network Service can be associated with one or more media cards installed in the system (depending on the system type).

To add new/additional Network Services:

1 In the *Device Management* pane, click **IP Network Services** ( ).

2 In the *Network Services* list toolbar, click the **Add Network Service** button.

The *New IP Service - Networking IP* dialog box opens.

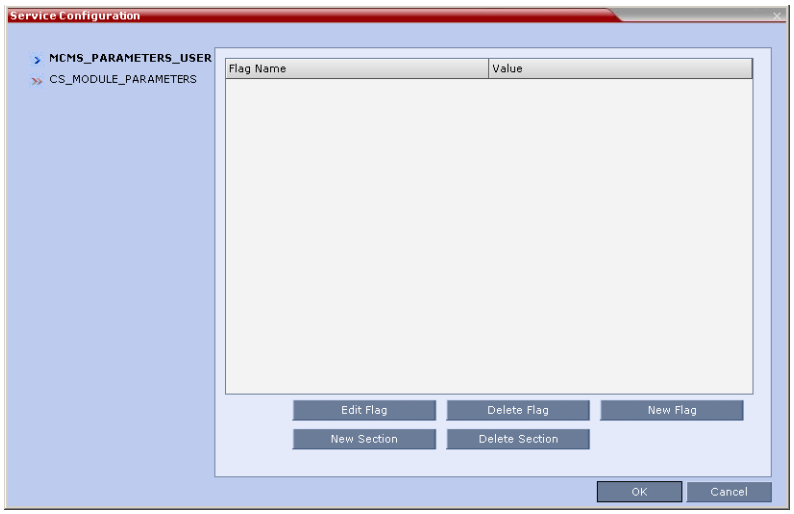
3 Define the following fields:

IP Network Service - IP Parameters

Field	Description
Network Service Name	Enter the IP Network Service name. **Note:** This field is displayed in all IP Signaling dialog boxes and can contain character sets that use Unicode encoding.
IP Network Type	Select the IP Network 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 *Collaboration Server* using this service. **Note:** This field is displayed in all Default IP Service tabs.
Signaling Host IP Address	This field is disabled as only one IP address is used for the signaling.
Media Card IP address	If each of LAN ports designated for signaling and media (LAN2, LAN3 and LAN4) on the system can be used with a different network, each port is assigned to its own Network Service. In such a case, enter the IP address of the port to be assigned to this Network Service. A LAN port that is already assigned to a different Network Service, displays the IP Address of the assigned port and it cannot be assigned to this Network Service (it is disabled).
Subnet Mask	Enter the subnet mask of the *Collaboration Server* in that network service. Default value: 255.255.255.0.

4 **Optional.** Some system flags can be defined per Network Service, depending on the network environment.
To modify these flags, click the **Service Configuration** button.

The *Service Configuration* dialog box opens.



All the flags must be manually added to this dialog box. For a detailed description of the flags and how to add them, see the Manually Adding and Deleting System Flags.

Flags defined per Network Service override their general definition in the System Configuration.

The following flags can be defined per service:

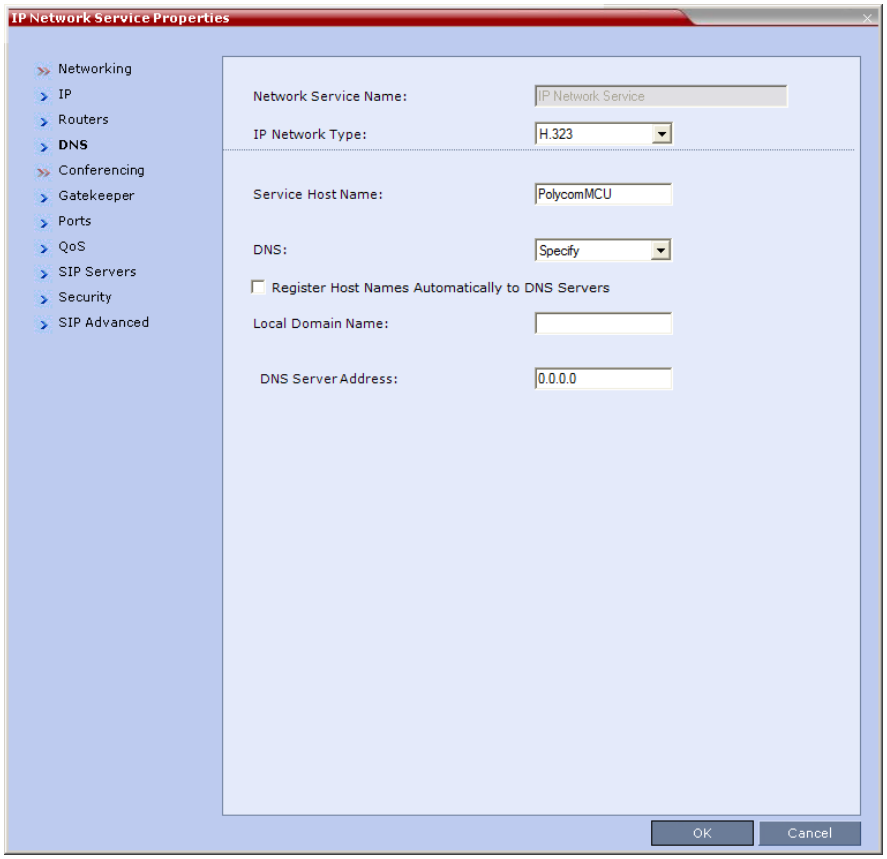
- ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF
- ENABLE_H239
- SIP_ENABLE_FECC
- ENABLE_CLOSED_CAPTION
- ALLOW_NON_ENCRYPT_RECORDING_LINK_IN_ENCRYPT_CONF
- NUMERIC_CONF_ID_LEN
- NUMERIC_CONF_ID_MIN_LEN
- NUMERIC_CONF_ID_MAX_LEN
- ENABLE_CASCADED_LINK_TO_JOIN_WITHOUT_PASSWORD
- MAX_CP_RESOLUTION
- QOS_IP_AUDIO
- QOS_IP_VIDEO
- QOS_IP_SIGNALING
- ENABLE_CISCO_GK
- SIP_FREE_VIDEO_RESOURCES
- FORCE_CIF_PORT_ALLOCATION

- MS_ENVIRONMENT
- SIP_FAST_UPDATE_INTERVAL_ENV
- SIP_FAST_UPDATE_INTERVAL_EP
- H263_ANNEX_T
- H239_FORCE_CAPABILITIES
- MIX_LINK_ENVIRONMENT
- IP_LINK_ENVIRONMENT
- FORCE_STATIC_MB_ENCODING
- FORCE_RESOLUTION
- SEND_WIDE_RES_TO_IP
- DISABLE_WIDE_RES_TO_SIP_DIAL_OUT
- SEND_SIP_BUSY_UPONRESOURCE_THRESHOLD

5 Click the **Routers** tab.

6 Define the routers used in this network and that are other than the routers defined in the Management Network. The field definitions of the *Routers* tab are the same as for the *Default Management Network*. For more information see the RealPresence Collaboration Server 800s - Default Management Network Service – Routers.

7 Click the **DNS** tab.



8 Modify the following fields:

Default Management Network Service – DNS

Field	Description
Service Host Name	Enter the host name of this network Service. Each Network Service must have a unique Host Name otherwise an error message is displayed.
DNS	Select: • **Off** – if no DNS server is used in this network. • **Specify** – to enter the IP address of the DNS server used by this network service. **Notes:** • The IP address field is enabled only if **Specify** is selected. • Only one DNS can be define for the entire topology (that is, only one Network Service can include the DNS definition).
Register Host Names Automatically to DNS Servers	Select this option to automatically register this Network Service Signaling Host with the DNS server.
Local Domain Name	Enter the name of the domain for this network service.
DNS Server Address	Enter the static IP address of the DNS server that is part of this network.

9 Click the **Gatekeeper** tab.

10 Define the *Primary* and *Alternate Gatekeepers* and at least one **Alias** for this network Service. The field definitions of the *Gatekeeper* tab are the same as for the *Default IP Network Service*. For more information see the RealPresence Collaboration Server 800s - Default IP Service – Conferencing – Gatekeeper Parameters.

In *Multiple Services* mode, an Alias must be defined for the specified gatekeeper.

11 **Optional.** Click the **Ports** tab.

Settings in the *Ports* tab allow specific ports in the firewall to be allocated to multimedia conference calls. If required, defined the ports to be used multimedia conference calls handled by this Network Service. The field definitions of the *Ports* tab are the same as for the *Default IP Network Service*.

For more information see the RealPresence Collaboration Server 800s - Default IP Service – Conferencing – Ports Parameters.

12 If required, click the **QoS** tab.

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.

The field definitions of the *QoS* tab are the same as for the *Default IP Network Service*. For more information see the RealPresence Collaboration Server 800s - Default IP Service – Conferencing – QoS Parameters.

13 Click the **SIP Servers** tab.

14 Define the *Primary* and *Alternate SIP Server* for this network Service.

- If Microsoft Office Communications or Lync server are part of this network service, a certificate must be created for this network service. If each network connected to the *Collaboration Server* includes Microsoft Office Communications or Lync server, separate certificates must be created and sent to the *Collaboration Server* for each of these networks.
- If the Network Service does not include a DNS, you must use the IP address of the SIP Server instead of its name.

The field definitions of the *SIP Servers* tab are the same as for the *Default IP Network Service*. For more information see the RealPresence Collaboration Server 800s - Default IP Network Service – SIP Servers.

15 Click the **Security** tab.

The field definitions of the *Security* tab are the same as for the *Default IP Network Service*. For more information see the RealPresence Collaboration Server 800s - Default IP Network Service – Security (SIP Digest).

16 **Optional.** To configure the ICE environment, click the **SIP Advanced** tab.

17 Modify the following fields:

Default IP Network Service – SIP Advanced

Field	Description
Server User Name	Enter the *User* name for this service as defined in the *Active Directory*. For example, enter *rmxNet2*. This field is disabled if the *ICE Environment* field is set to *None*.
ICE Environment	Select **MS** (for *Microsoft ICE* implementation) to enable the *ICE* integration.

18 Click the **OK** button.
The new Network Service is added to the *IP Network Services* list pane.

Setting a Network Service as Default

The default Network Service is used when no Network Service is selected for dial out participants.In addition, the Signaling Host IP address and the MCU Prefix in GK displayed on the *Collaboration Server Web Client* main screen are taken from the default H.323 Network Service.

One IP Network Service can be defined as default for H.323 connections and another Network Service as default for SIP connections. If the IP Network Service supports both H.323 and SIP connections, you can set the same Network Service as default for both H.323 and SIP, or for H.323-only or for SIP-only.

To designate an IP Network Service as the default IP Network Service:

1 In the *Device Management* pane, click **IP Network Services** ( ).

2 In the *Network Services* list pane right-click the IP Network Service to be set as the default, and then click **Set As H.323 Default,** or **Set As SIP Default**.

The next time you access this menu, a check mark is added next to the network service type to indicate its selection as default.

To set this IP Network Service for both H.323 and SIP connections, repeat step 2 and select the option you need.

The following icons are used to indicate the default IP Network Service type:

Default IP Network Service Icons

Icon	Description
	This Network Service supports both SIP and H.323 connections and is designated as default for both SIP and H.323 connections.
	This Network Service supports both SIP and H.323 connections and is designated as default for H.323 connections.
	This Network Service supports both SIP and H.323 connections and is designated as default for SIP connections.
	This Network Service supports only H.323 connections and is set as default for H.323 connections.
	This Network Service supports only SIP connections and is set as default for SIP connections.

Signaling Host IP Address and MCU Prefix in GK Indications

The *Collaboration Server Web Client* displays the *Signaling Host IP Address* and *MCU Prefix in GK* parameters as defined in the **Default H.323 Network Servic**e.

Resolution Configuration

These configurations are set for the system and are applied to all the Network Services.

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 by selecting for each of the conferencing entities whether it will register with the SIP server.

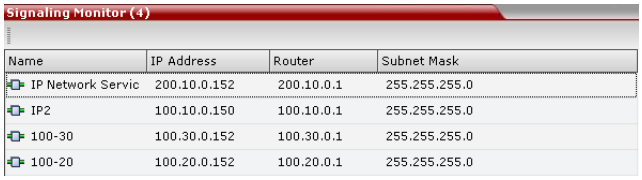
The registration is defined in the *Conference Profile - Network Services* tab.

In the *IP Network Services* table, the system lists all the defined Network Services (one or several depending on the system configuration).

- To register the conferencing entity to which this profile is assigned to a Network Service, in the *Registration* column click the check box of that Network Service.
- You can also prevent dial in participants from connecting to that conferencing entities when connecting via a Network Service.
  In the *Accept Calls* column, clear the check box of the Network Service from which calls cannot connect to the conference.

Signaling Monitor

The Signaling Monitor pane includes the list of the signaling and media IP Network Services defined in the system (up to three in the *Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition*). Double-clicking a Network Service, displays it properties and status.



Conferencing

Each conference on the *Collaboration Server* can host participants from the different IP Network networks simultaneously.

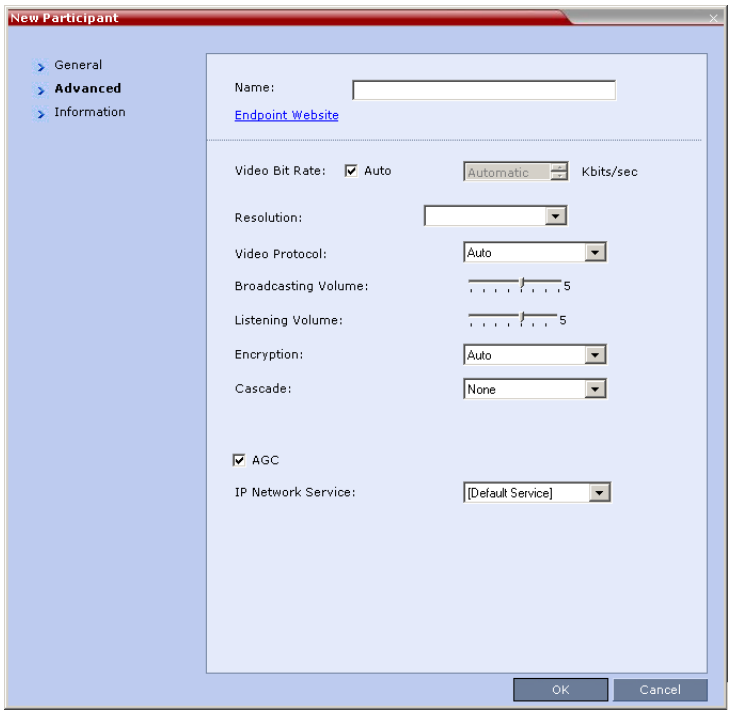
Defining AVC Dial Out Participants

When defining AVC dial out participants, you can select the Network Service to place the call according to the network to which the endpoint pertains. If the endpoint is located on a network other than the selected network, the participant will not be able to connect.

If no Network is selected, the system uses the default IP Network Service.

If no Network is selected, the system uses the IP Network Service selected for reserving the conference resources, and if none is set for the conference it uses the Network Service set as default.

The IP Network Service is selected in the *New Participant - Advanced* dialog box.



Monitoring Conferences

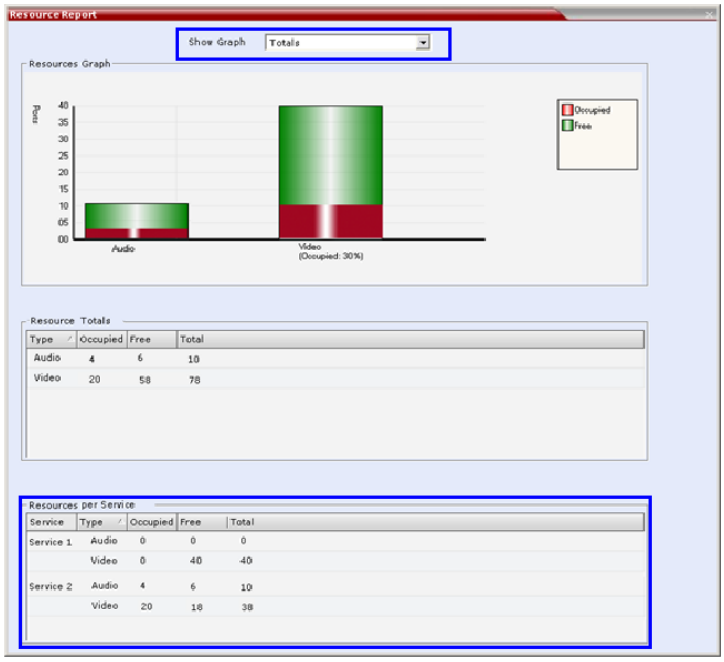
The *Conference Properties - Network Services* dialog box shows for each Network Service with which Network Service's SIP proxy the conference should be registered and if the dial in call will be connected to the conference.

In the *Participant* pane, a new column - *Service Name* was added, indicating the name of Network Service used for the participant's connection.

Resource Report

The *Resource Report* displays the resource usage in total and per Network Service in a table format. The Resources per Service table provides the actual information on resource usage and availability per network Service and provides an accurate snapshot of resources usage in the system.

You can select the graph to display: select either **Totals** (default) or the Network Service.



Port Gauge Indications

The port Gauges displays the total resource usage for the *Collaboration Server* and not per Network Service. Therefore, it may not be an accurate representation of the availability of resources for conferencing, as one Network Service may run out of available resources while another Network Service may have all of it resources available. In such a case, the port gauges may show that half of the system resources are available for conferencing, while calls via the Network Service with no available resources will fail to connect.

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.
- 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.

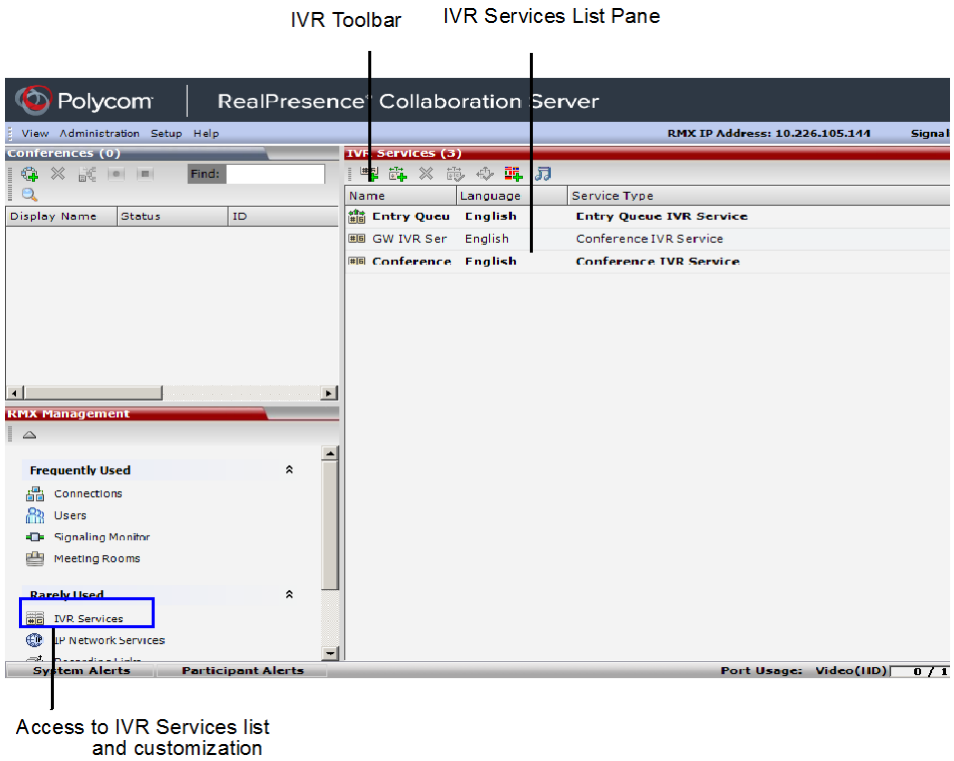
IVR Services List

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 Services Toolbar

The IVR Services toolbar provides quick access to the IVR Service definitions as follows:

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.

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

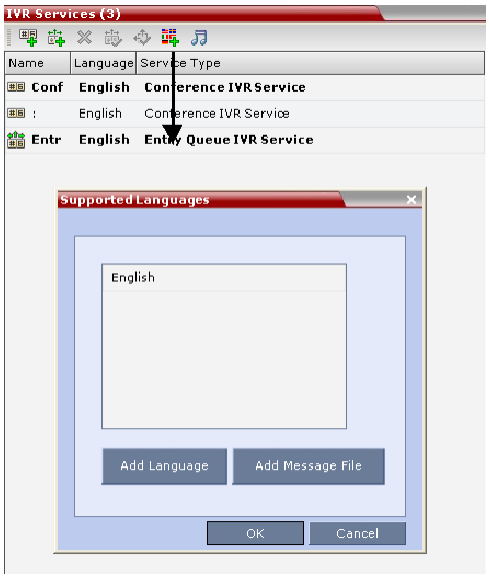
You can define different sets of audio prompts in different languages, allowing the participants to hear the messages in their preferred language.

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.

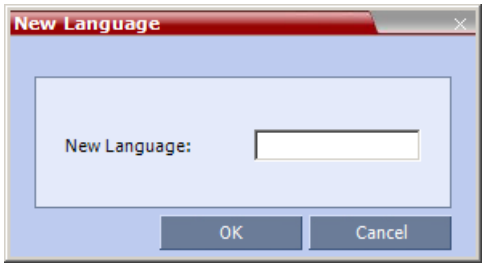
To add a language:

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.



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

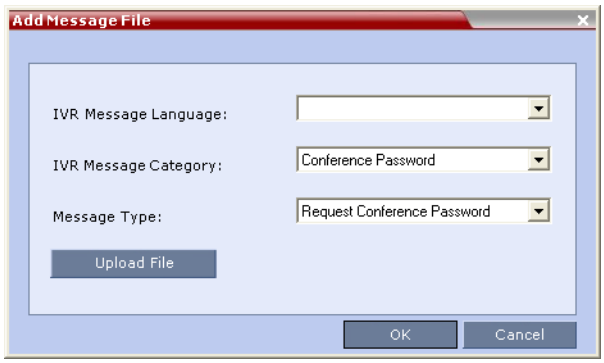
You can upload audio files for the new language or additional files for an existing language now, or you can do it during the definition of the IVR Service. In the latter case, you can skip the next steps.

- 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.

The **Add Message File** dialog box opens.



Audio files are uploaded to the MCU one-by-one.

2 In the **IVR Message Language** list, select the language for which the audio file will be uploaded to the MCU.

3 In the **IVR Message Category** list, select the category for which the audio file is uploaded.

4 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.

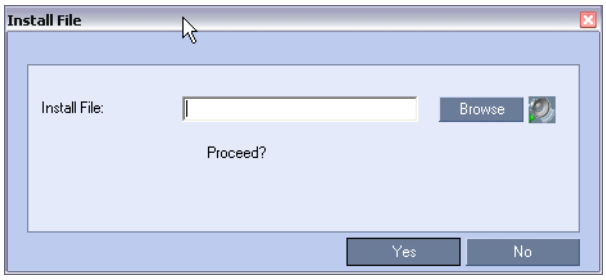
Table 5-2 lists the Message Types for each category:

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.	
Billing Code	Requests the chairperson to enter the conference Billing Code.	
Roll Call	Roll call related messages, such as the message played when a participant joins the conference. Messages are listed in the Conference IVR Service - Roll Call dialog box.	
Conference ID	Requests the participant to enter the required Conference ID to be routed to the destination conference.	

5 Click **Upload File** to upload the appropriate audio file to the MCU.

The **Install File** dialog box opens.



6 Enter the file name or click the **Browse** button to select the audio file to upload.
The **Select Source File** dialog box opens.

7 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.

8 Optional. You can play a .wav file by selecting the **Play** button ( ).

9 Click Yes to upload the file to the MCU.
The system returns to the **Add Message File** dialog box.

10 Repeat step 6 to Click Yes to upload the file to the MCU. The system returns to the Add Message File dialog box. for each additional audio file to be uploaded to the MCU.

11 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.

12 Click OK.

Defining a New Conference IVR Service

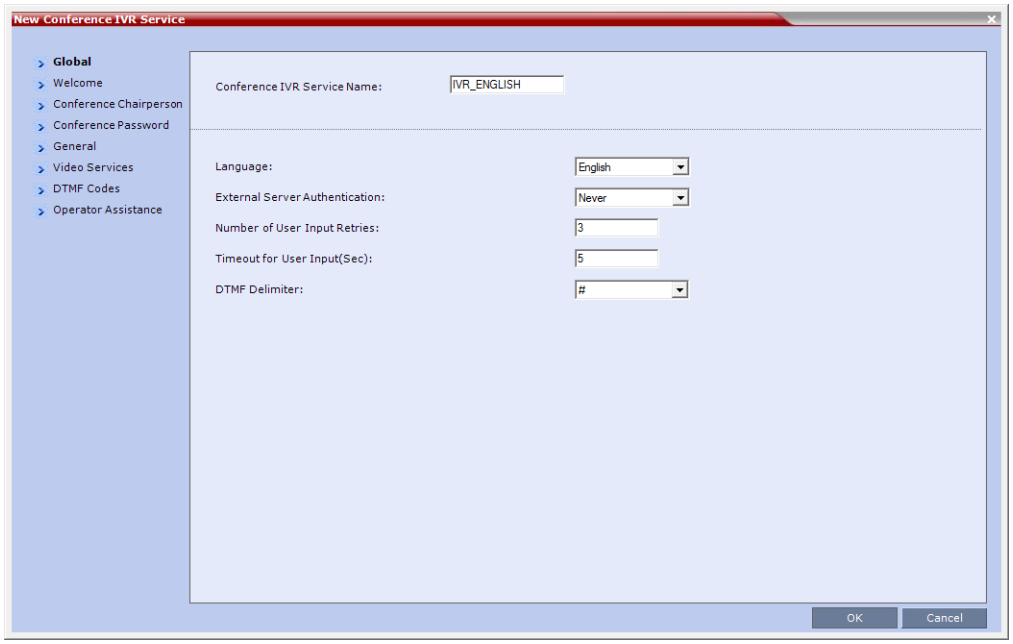
The Collaboration Server is shipped with two default Conference IVR Services and all its audio messages and video slide. You can define new Conference IVR Services or modify the default Conference IVR Service.

Up to 80 IVR Services (Conference IVR Services and Entry Queue IVR Services) can be defined per Collaboration Server.

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.



2 Define the following parameters:

Conference IVR Service Properties - Global Parameters

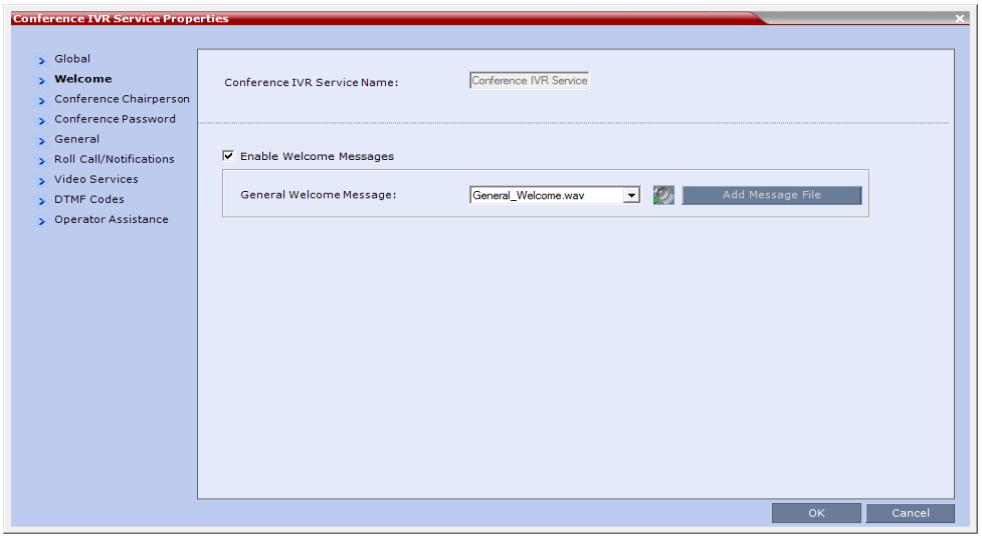
Field/Option	Description
Conference IVR Service Name	Enter the name of the Conference IVR Service. The maximum field length is 20 characters and may be typed in Unicode.
Language For IVR	Select the language of the audio messages and prompts from the list of languages defined in the Supported languages. The default language is English. For more information, see Adding Languages.
External Server Authentication	This option is not supported with Collaboration Server 800s/Virtual Edition. You can configure the IVR Service to use an external database application to verify a participant's right to join the conference. For more information, see Conference Access with External Database Authentication. Select one of the following options: • **Never** – The participant's right to join the conference will not be verified with an external database application (default). • **Always** – Any participant request to join the conference is validated with the external database application using a password. • **Upon Request** – Only the participant request to join the conference as chairperson is validated with the external database application using a password. The validation process occurs only when the participant enters the chairperson identifier key.
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.

Conference IVR Service Properties - Global Parameters

Field/Option	Description
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.

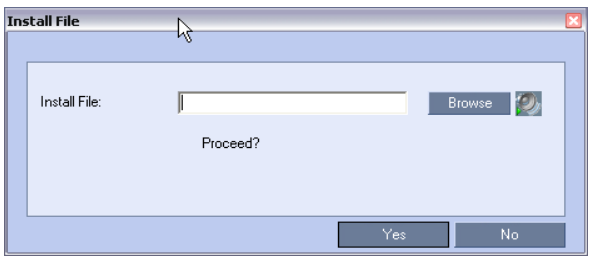


4 Select the **Enable Welcome Messages** check box to define the system behavior when the participant enters the Conference IVR queue. When participants access a conference through an Entry Queue, they hear messages included in both the Entry Queue Service and Conference IVR Service. To avoid playing 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.



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.

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.

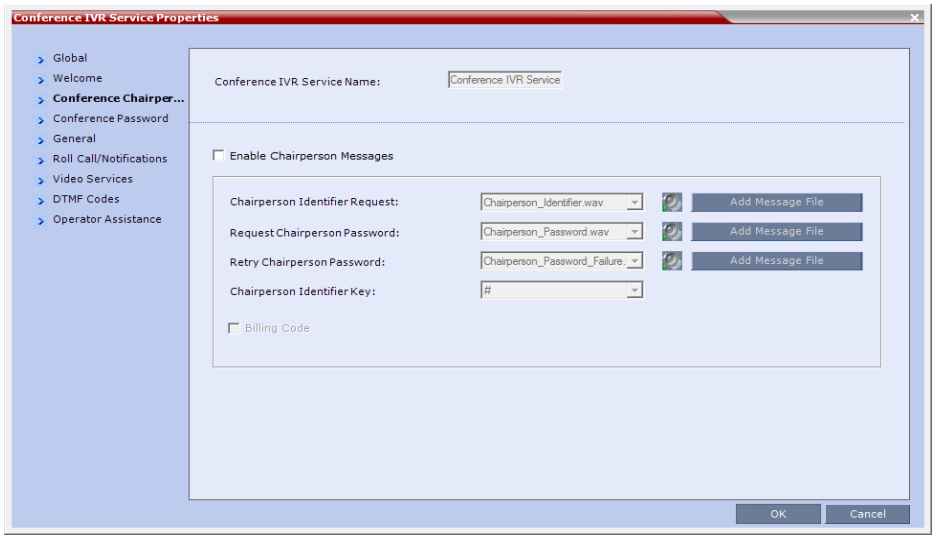
c Optional. You can play a .wav file by selecting the **Play** button ( ).

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.



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.

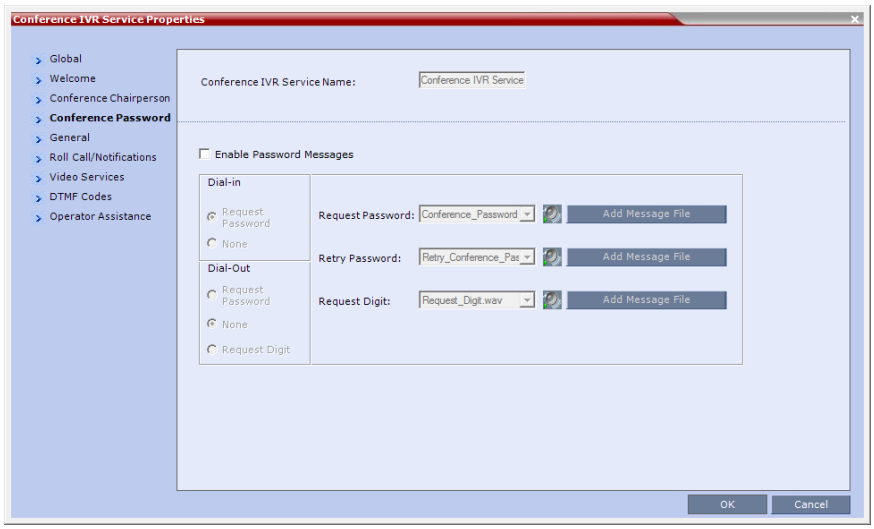
If the files were not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the Collaboration Server.

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.



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.

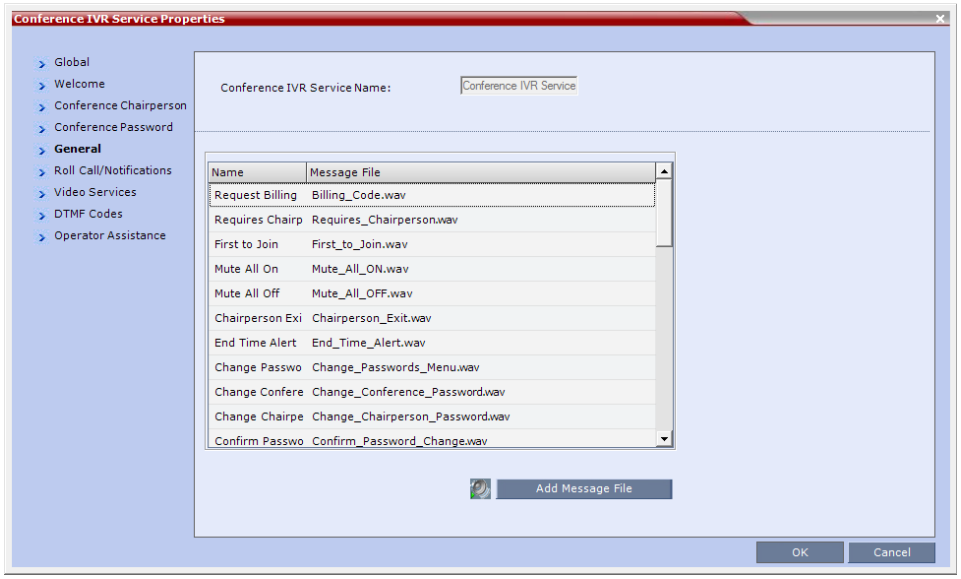
13 Select the various audio messages that will be played in each case.

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.
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.



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:

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.
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.

Conference IVR Service Properties - General Voice Messages

Message Type	Description
Maximum Number of Participants Exceeded	Indicates the participant cannot join the destination conference as the maximum allowed number of participants will be exceeded.
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 **Roll Call/Notifications** tab.

The **New Conference IVR Service** - **Roll Call** dialog box opens.

The Roll Call and Tone Notification options are disabled in SVC and mixed CP and SVC conferences.

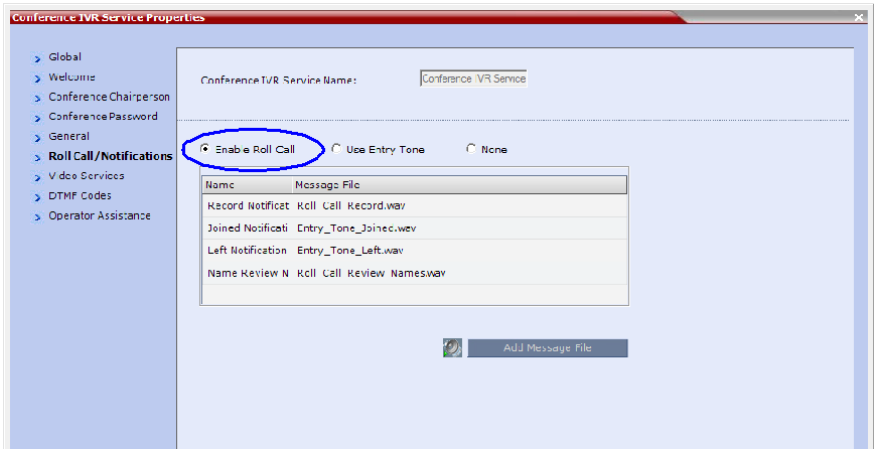
The **Roll Call** feature of the Conference IVR Service is used to record the participants' names for playback when the participants join and leave a conference.

Roll Call announcements played upon a participant's connection or disconnection from a conference (Entry and Exit announcements) can be replaced by tones. These tones can be used as notification when participants join or leave the conference but the identification of the participant is not required.

The system is shipped with two default tones: Entry Tone and Exit tone. When the Tone Notifications option is enabled, no recording of the participant names will occur and the conference chairperson will not be able to ask for a name review during the conference.

19 Select one of the following options to determine the announcement mode:

a To enable the Roll Call feature, select the **Enable Roll Call** option.



b Select **Enable Tones** to enable the **Tone Notifications** option.
The dialog box changes to display the tone notification options and all Roll Call options are disabled. In such a case, skip to step Select the Entry Tone or Exit tone:.

c Select **None** to disable the Roll Call and Tone Notifications features.

If Enable Roll Call option is selected:

20 To assign the audio file to the message type, in the Message File column, click the appropriate table entry. An arrow appears in the **Message File** column.

If the Roll Call option is enabled, you must assign the appropriate audio files to all message types.

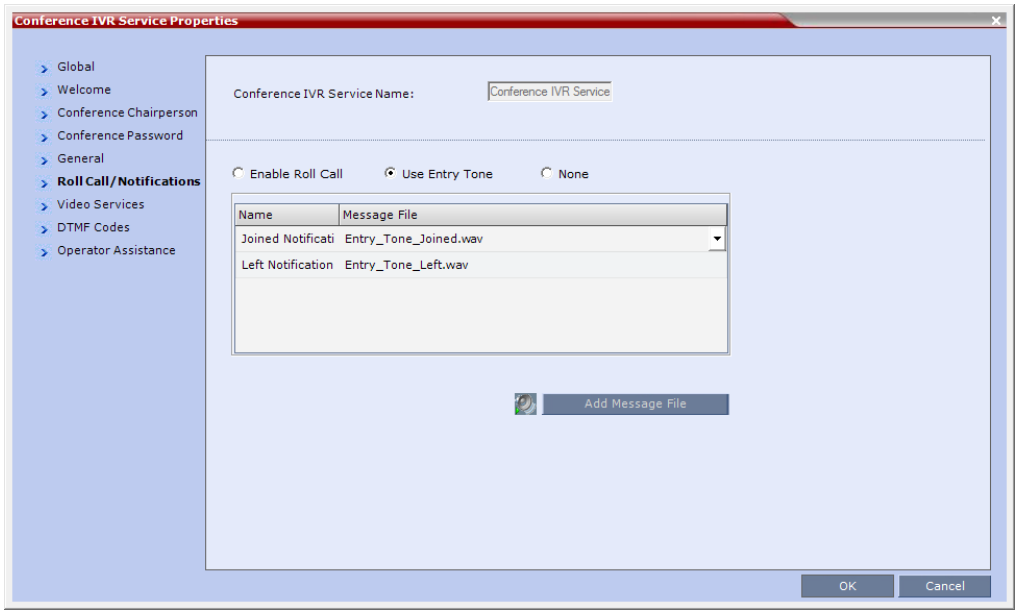
21 Click the arrow to open the **Message File** list and select the appropriate audio file.

Conference IVR Service Properties - Roll Call Messages

Roll Call Message	Description
Roll Call Record	Requests participants to state their name for recording, when they connect to the conference. **Note:** The recording is automatically terminated after two seconds.
Roll Call Joined	A voice message stating that the participant has joined the conference.
Roll Call Left	A voice message stating that the participant has left the conference.
Roll Call Review	Played when Roll Call is requested by the chairperson, introducing the names of the conference participants in the order they joined the conference.

If Enable Tone Notifications option is selected:

22 Select the Entry Tone or Exit tone:



a Click the appropriate table entry in the **Message File** column.
A drop-down list is enabled.

b From the list, select the audio file to be assigned to the event/indication.

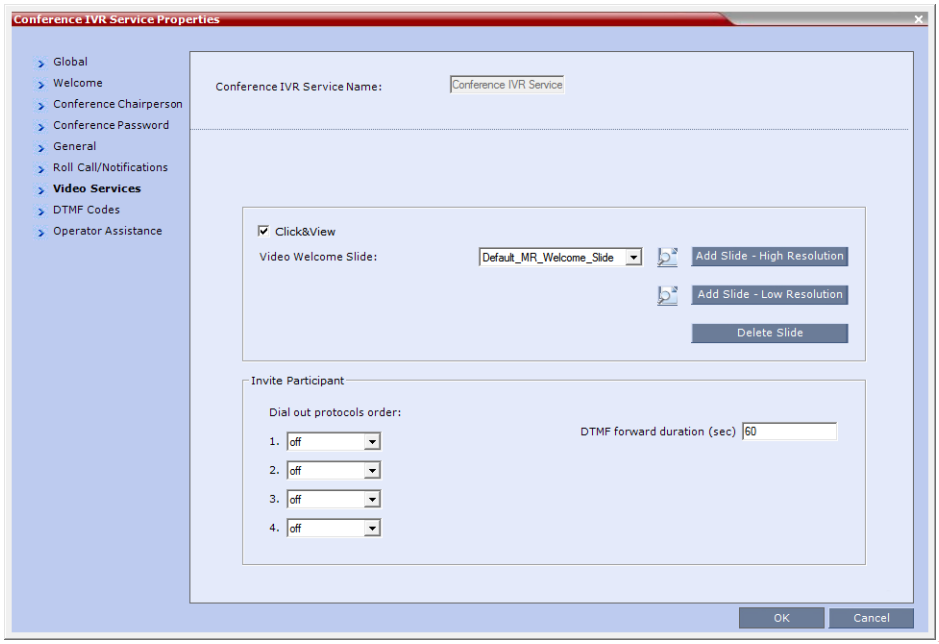
If the **Tones** option is enabled, you must assign the appropriate audio files to all notification types. The Collaboration Server system is shipped with two default tones: Entry_tone.wav and Exit_tone.wav.

If required, you can upload customized audio files that will be played when participants join or leave the conference.

If the option to play a tone when a cascading link connection is established, make sure that the tone selected for Entry or Exit notification differ from the cascading link tone as the latter one cannot be customized.

23 Click the **Video Services tab**.

The **New Conference IVR Service** - **Video Services dialog box opens.**



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.

  The following table summarizes the recommended input slide formats and the resulting slides that are generated:

IVR Slide - Input / Output Formats

	Format	
Slide Resolution	**Input Slides**	**Generated Slides**
High	HD1080p (16:9) or HD720p (16:9)	HD1080p 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.

24 Define the following parameters

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. When *Collaboration Server* is configured to IPv6, the IVR slide is displayed without taking into account the MTU Size.
Invite Participant	See Inviting Participants using DTMF . **Note:** The Invite Participant feature is not available in SVC conferences and for SVC participants in mixed CP and SVC conferences.

25 If the video slide file was not uploaded to the MCU prior to the IVR Service definition, click the:

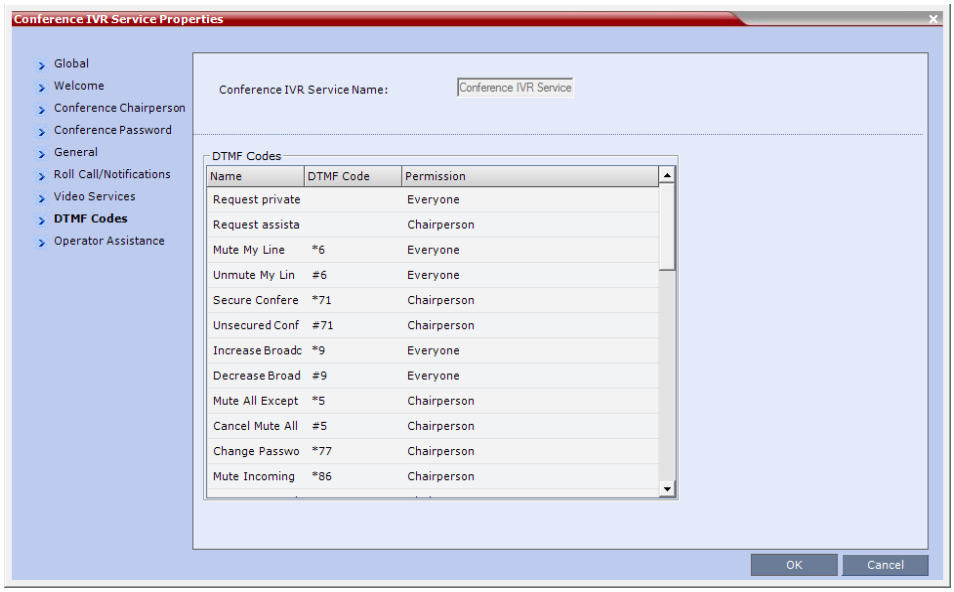
- **Add Slide - Low Resolution** button to upload a **Low Resolution Slide**.
- **Add Slide - High Resolution** button to upload a **High Resolution Slide**.

The **Install File** dialog box opens. The uploading process is similar to the uploading of audio files. For more information, see **To upload an audio file for an IVR message, click Add Message File.**.

- The video slide must be in a .jpg or .bmp file format. For more information, see Creating a Welcome Video Slide.
- Customized H.261 slides are not supported.

26 Click the **DTMF Codes** tab.

The **New Conference IVR Service** - **DTMF Codes** dialog box opens.



- 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

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
Change To Chairperson	*78	Everyone
Override Mute All	Configurable	Everyone
Start Recording	*2	Chairperson
Stop Recording	*3	Chairperson
Pause Recording	*1	Chairperson

New Conference IVR Service Properties - DTMF Codes

Operation	DTMF String	Permission
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

27 To modify the DTMF code or permission:

a In the **DTMF Code** column, in the appropriate entry enter the new code.

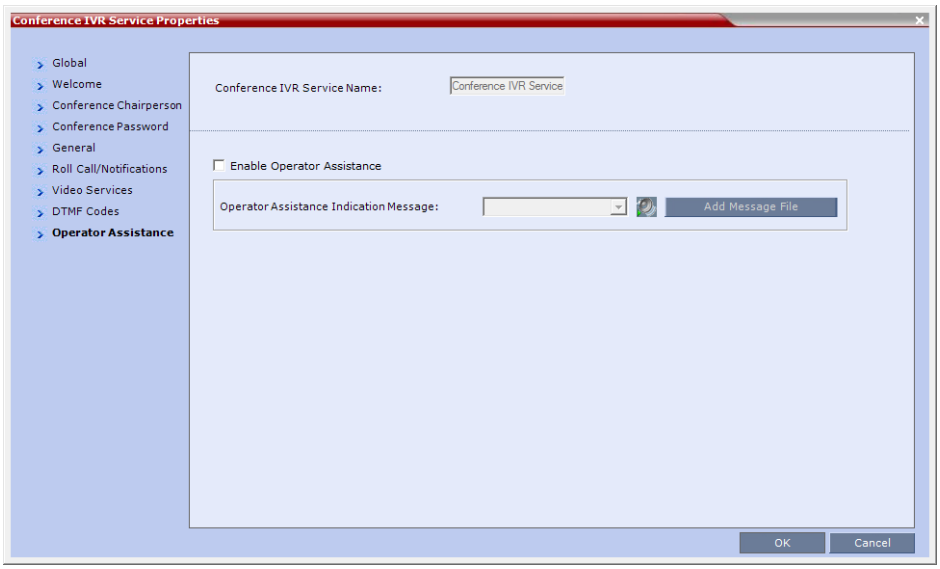
b In the **Permission** column, select from the list who can use this feature (Everyone or just the Chairperson).

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.

28 Click the **Operator Assistance** tab.

The **Operator Assistance** dialog box opens.



29 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.

30 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*.

31 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 the *Polycom® RealPresence Collaboration Server 800s/Virtual Edition Administrator's Guide*, New Conference IVR Service Properties - Conference Chairperson Options and Messages.

- 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.

Controlling the receipt of in-band and out-of-band DTMF Codes

The **RFC2833_DTMF System Flag** controls the receipt of in-band or out-of-band DTMF Codes.

When set to **YES** (default), the RMX will receive DTMF Codes sent in-band. When set to **NO** the RMX receives DTMF Codes sent out-of-band. The RMX always sends DTMF Codes in-band (as part of the Audio Media stream). If you wish to modify the flag value, the flag must be added to the **System Configuration** file. For more information see Modifying System Flags.

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.

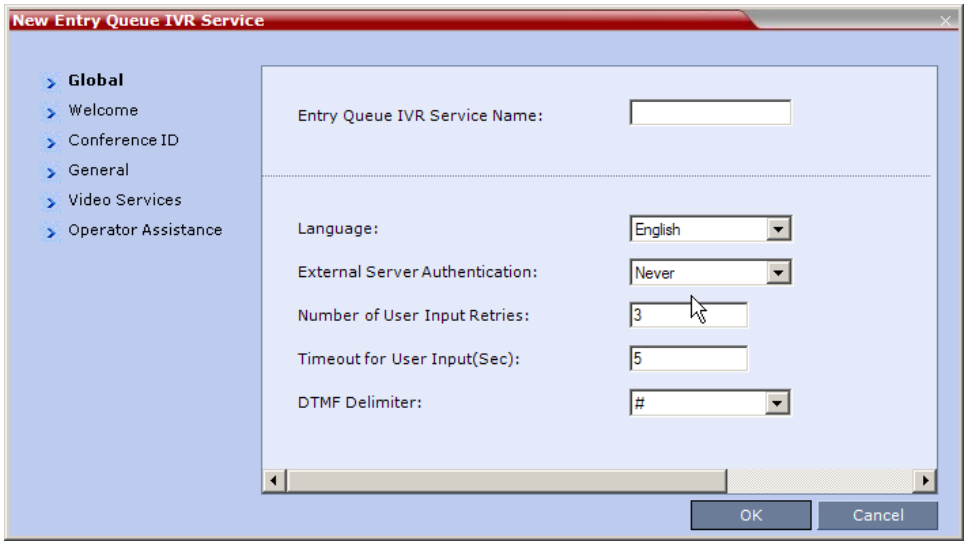
Defining a New Entry Queue IVR Service

To set up a new Entry Queue IVR Service:

1 In the **RMX 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 Fill in the following parameters:

Entry Queue IVR Service Properties - Global Parameters

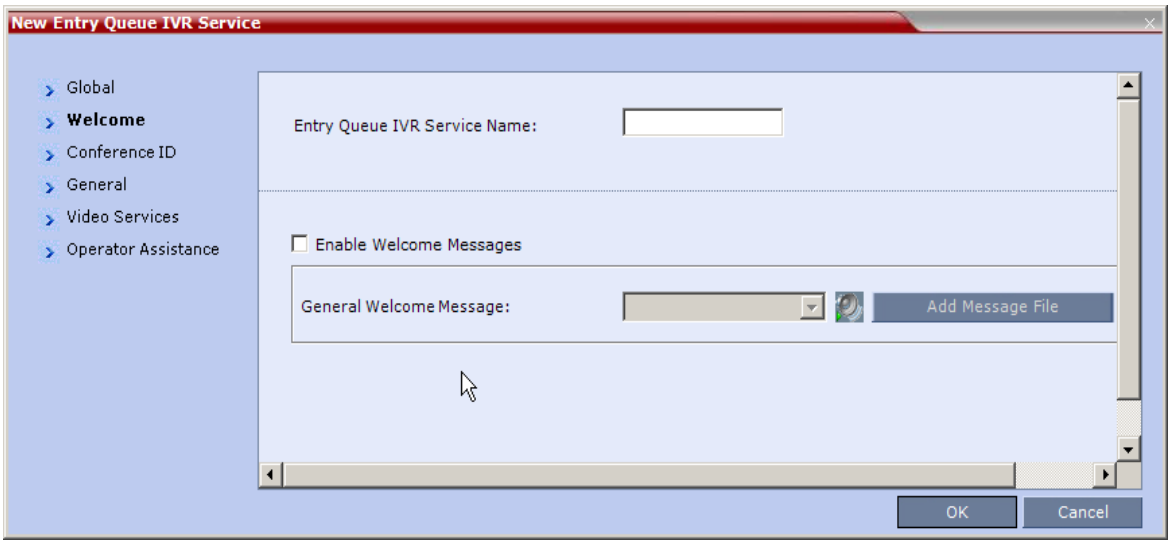
Option	Description
Entry Queue Service Name	(Mandatory) Enter the name of the Entry Queue Service. The name can be typed in Unicode. Maximum field length is 80 ASCII characters.
Language	Select the language in which the Audio Messages and prompts will be heard. The languages are defined in the **Supported Languages** function.
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: Appendix D - Ad Hoc Conferencing and External Database Authentication. 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.

Entry Queue IVR Service Properties - Global Parameters

Option	Description
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.

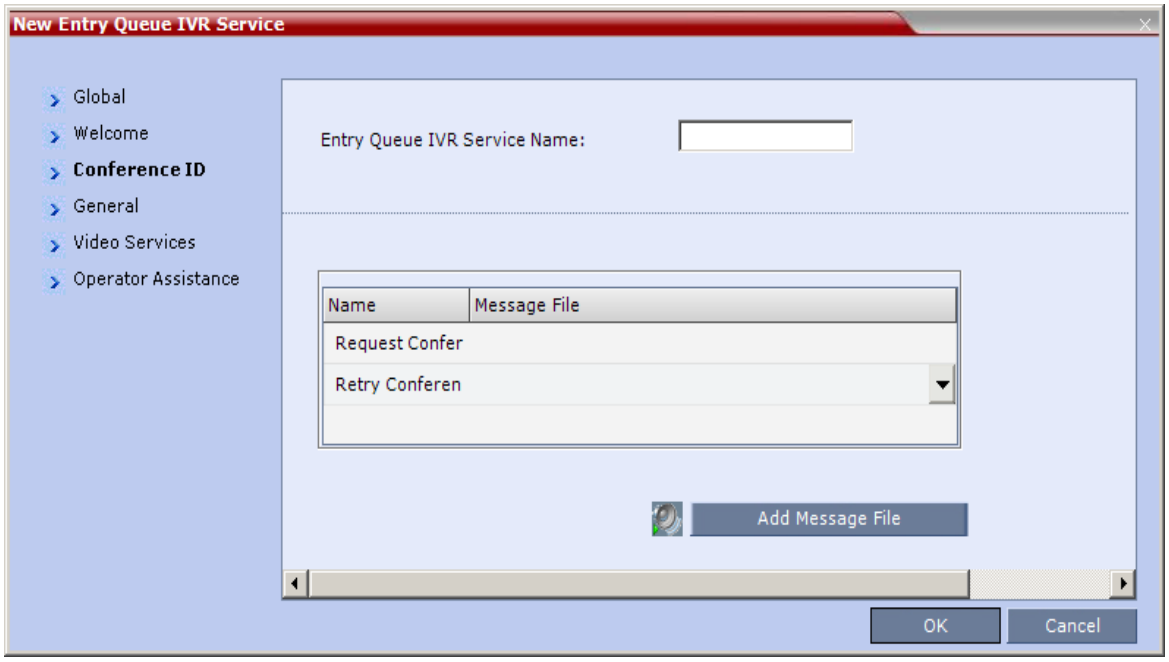


If the files were not uploaded prior to the definition of the IVR Service or if you want to add new audio files, click **Add Message File** to upload the appropriate audio file to the *Collaboration Server*.

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 New Conference IVR Service Properties - Conference Chairperson Options and Messages.

6 Click the **Conference ID tab**.

The **New Entry Queue IVR Service - Conference ID** dialog box opens.



7 Select the voice messages:

Entry Queue IVR Service Properties - Conference ID

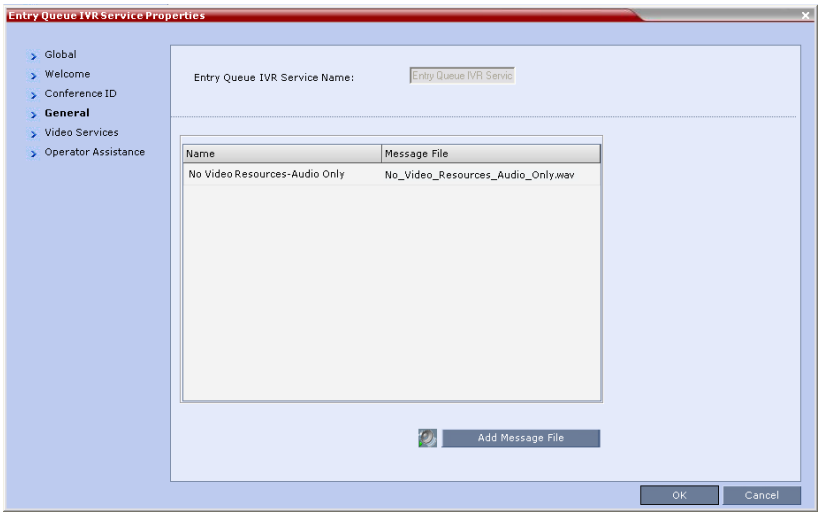
Field/Option	Description
Request Conference ID	Prompts the participant for the conference ID.
Retry Conference ID	When the participant entered an incorrect conference ID, requests the participant to enter the ID again.

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.



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**

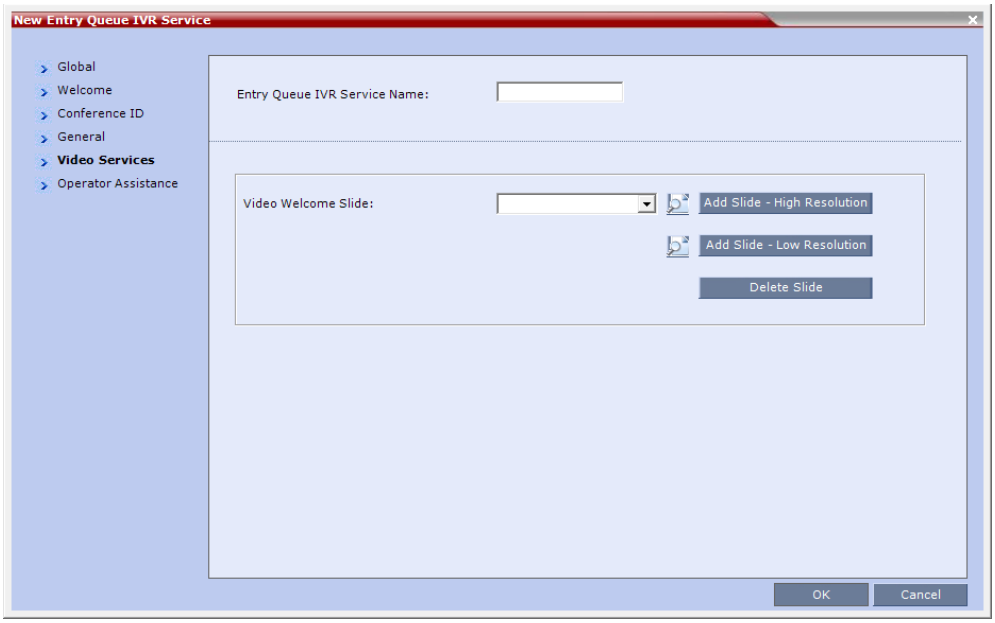
If you wish to modify the flag value, the flag must be added to the **System Configuration** file. For more information see the Modifying System Flags.

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.



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:

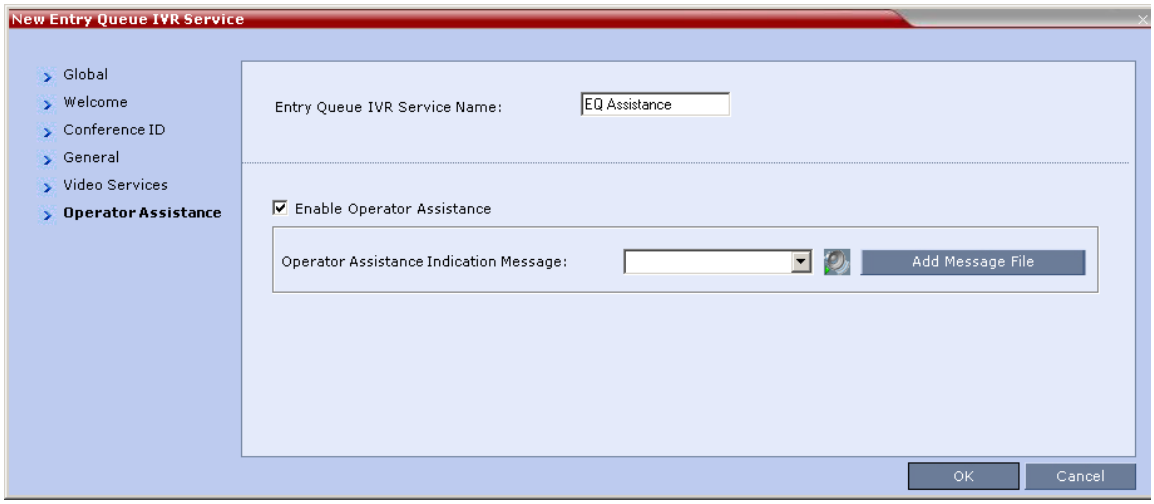
- **Add Slide - Low Resolution** button to upload a **Low Resolution Slide**.
- **Add Slide - High Resolution** button to upload a **High Resolution Slide**.

The **Install File** dialog box opens. The uploading process is similar to the uploading of audio files. For more information, see **step** 6 .

The video slide must be in a .jpg or .bmp file format. For more information, see Creating a Welcome Video Slide.

15 Click the **Operator Assistance** tab.

The **Operator Assistance** dialog box opens.



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.

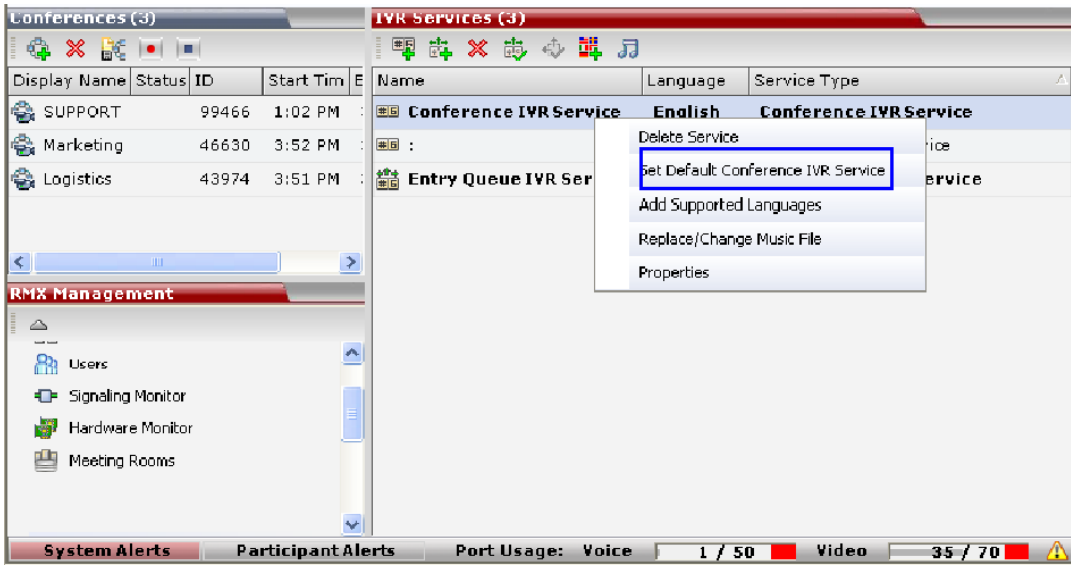
Setting a Conference IVR Service or Entry Queue IVR Service as the Default Service

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**.

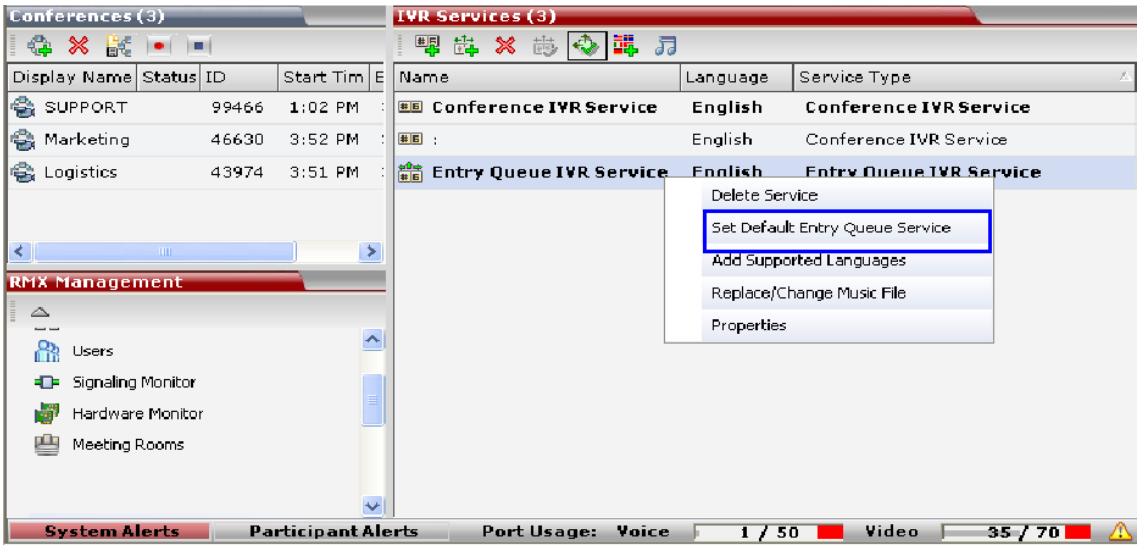


The IVR Service is displayed in bold, indicating that it is the current default service.

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**.



  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 RMX 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.
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:

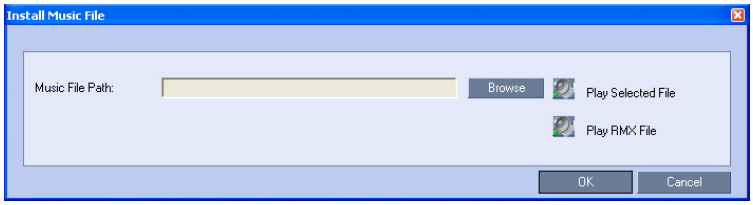
- The file must be in *.wav format.
- Music length cannot exceed one hour.
- The music recording must be in the range of (-12dB) to (-9dB).

Adding a Music File

To replace the Music file:

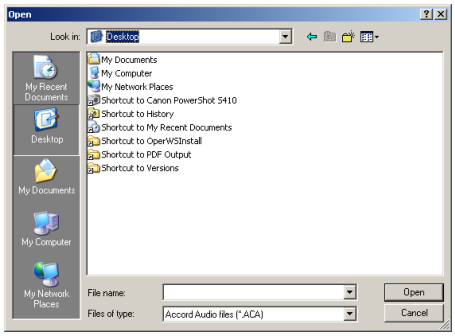
1 In the **RMX Management** pane, click **IVR Services**.

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 RMX File** to play a file already uploaded on the **RMX**.

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 the following procedure. 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.

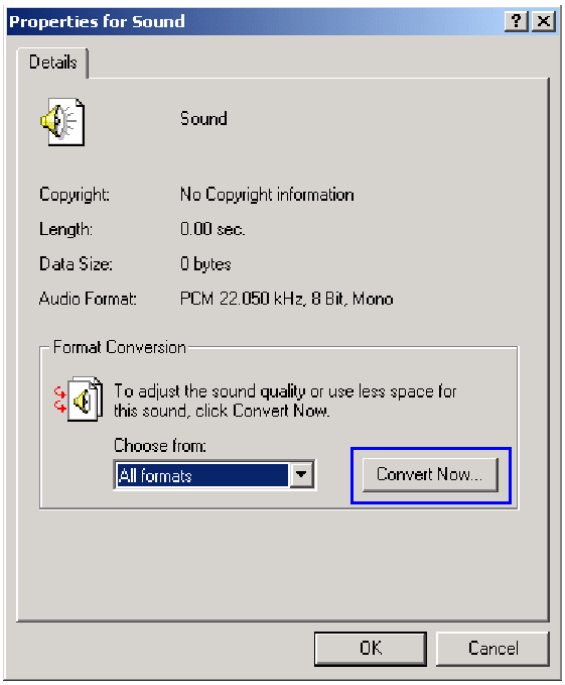
To define the format settings for audio messages:

- **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.

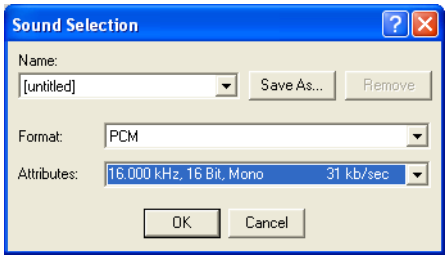
3 Click the **Convert Now** button.



The ***Sound Selection*** dialog box opens.

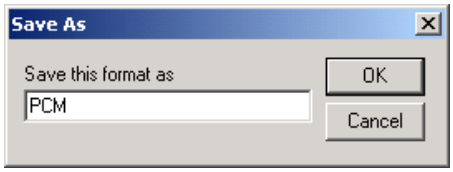
4 In the **Format** field, select **PCM**.

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**.

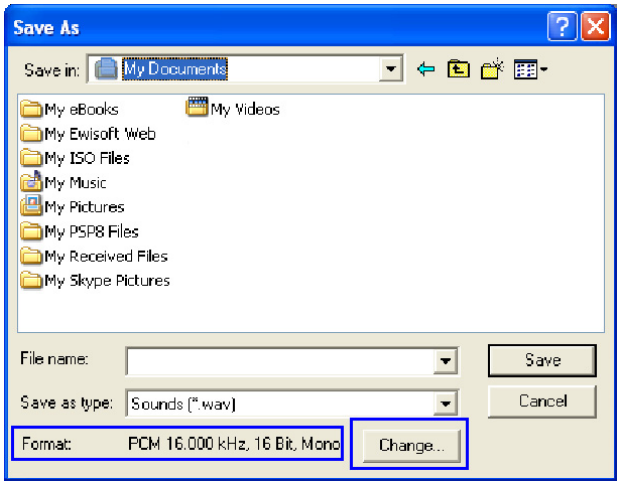
3 Click the **Record** button.
The system starts recording.

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.

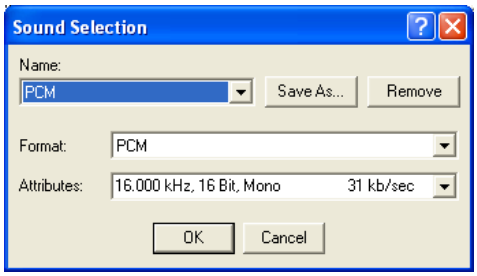
5 Click the **Stop Recording** button.

6 Save the recorded message as a wave file, click **File** > **Save As**.
The **Save As** dialog box opens.



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.



8 In the **Name** field, select the name of the format created in step 7 on Select the location where the format will reside, enter a name and then click OK..

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.

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.

Inviting Participants using DTMF

This feature is disabled in SVC conferences and for SVC participants in mixed CP and SVC conferences.

A participant in a video or audio conference can invite another participant to the conference using the touch-tone DTMF numeric keypad on the participant's endpoint. You can invite a participant using various communication devices, such as a mobile phone, an IP phone, PSTN phones, laptops, or connect to another conference running on another PBX or MCU.

Invite Call Flow

The following flow describes how a participant is invited to the conference using the DTMF codes:

1. During the conference, the participant enters the DTMF code (default is ***72**) on the numeric keypad to invite another participant.
2. The participant is prompted to enter the invited participant's destination number (a number or IP address) including the prefix (if required) and the DTMF delimiter digit ('*' or '#') at the end. The asterisk ('*') is used to denote the dot in the IP address.

   For example: To enter an IP address such as 10.245.22.19, on the DTMF keypad press 10*245*22*19 and then the DTMF delimiter.

   Digits that are entered after the DTMF delimiter and before the participant is connected are ignored.

3. The system automatically dials to the destination according to the protocol order as defined in the **IVR Services Properties - Video Services** tab.

   When the call cannot be completed by the current protocol, the system attempts to connect to the destination using the next protocol according to the protocol order.

   The *Collaboration Server* connects the participant when the call is answered.
4. The last invited participant can be disconnected when the inviting participant enters the DTMF code (default is **#72**) on the numeric keypad.

Entering Additional DTMF Codes

In some environments, the call is answered by an IVR system (for example when connecting to another conference or PBX), requesting a password or a destination number to complete the connection process. In such a case, additional DTMF digits must be entered before the **DTMF forward duration** time has expired and are forwarded to the invited destination. When the additional DTMF codes are entered, they are heard by all the conference participants.

If the DTMF code is not entered on time or if the wrong DTMF code is entered, the participant is prompted for a new input. After the defined number of retries have elapsed, the call is ended.

Error Handling

- If the destination endpoint is busy or the participant did not answer, the system ends the call.

- When an incorrect number is entered, the call fails and an error message is displayed.
- If the destination number is not entered in a specific amount of time (defined in **Timeout for user input** in the **IVR Services - Global** tab), the participant is prompted to enter a destination number again. Depending on the **Number of user input retries** as defined in the **IVR Services - Global** tab, the system will attempt to receive the required input. When all the retries have failed, the call to the invited participant is cancelled.

Guidelines

- Inviting other participants is available to AVC-enabled participants only.
- Participants can be invited to Event Mode, and CP and VSW conferences.
- All network protocols are supported (H.323, SIP, ISDN, and PSTN). It is recommended to select PSTN and not ISDN if PSTN is the only destination protocol. If both PSTN and ISDN are enabled, it is recommended to select the PSTN before ISDN as the connection process for PSTN endpoints will be quicker.
- In an Multiple IP Networks environment, the system will try to connect the participant using each of the IP Network Services listed in the **Conference Profile - Network Services** dialog box. Network services that are excluded from this list are skipped during the dialing sequence.
- In Event Mode conferences, the invited participant connection parameters must match one of the conference levels.
- In CP conferences, the participant initiating the invitation to another participant is able to view the dialing information and connection status. During the dialing process, the dialing string is displayed as the participant name which is replaced by the site name when connected to the conference.
- By default, all participants (Everyone) are granted permission to invite a participant to join a conference. To change the permission to the Chairperson, modify the **Permission** column in the **IVR Service - DTMF Codes** tab.

Enabling the Invite Participants using DTMF Option

The option to invite participants to a conference using the DTMF keypad is enabled in the following **Conference IVR Services** dialog boxes:

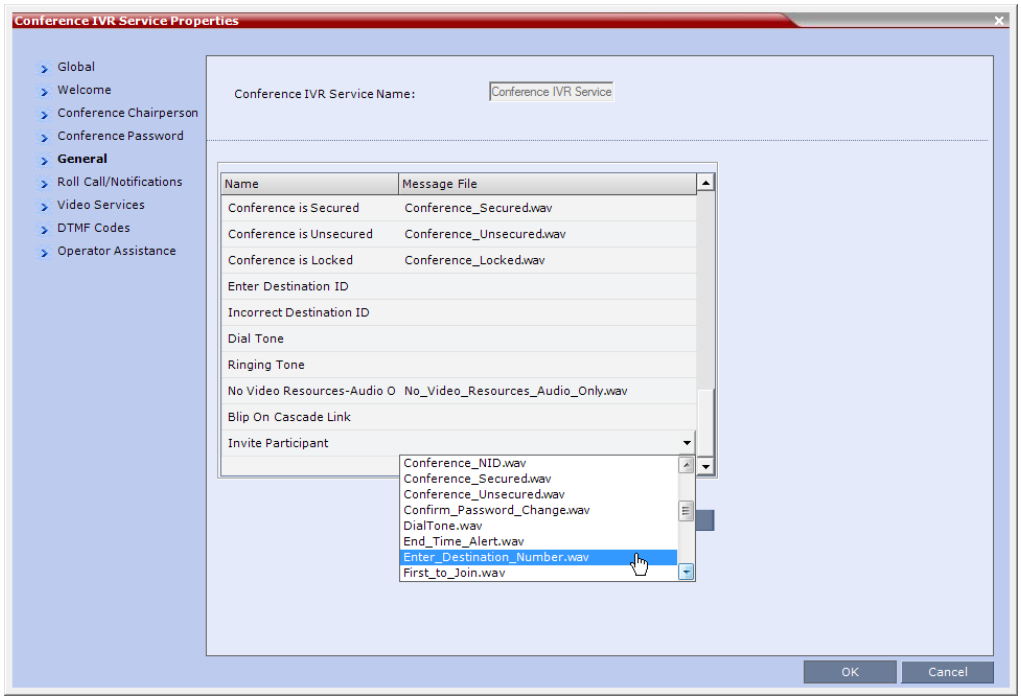
- General
- Video Services
- DTMF Codes

To enable the Invite Participant using DTMF on the *Collaboration Server*:

1 Open an existing or define a new **Conference IVR Service**.
Conference IVR Service - Global dialog box opens**.**

2 Click the **General** tab.

The **Conference IVR Services - General** tab is displayed.

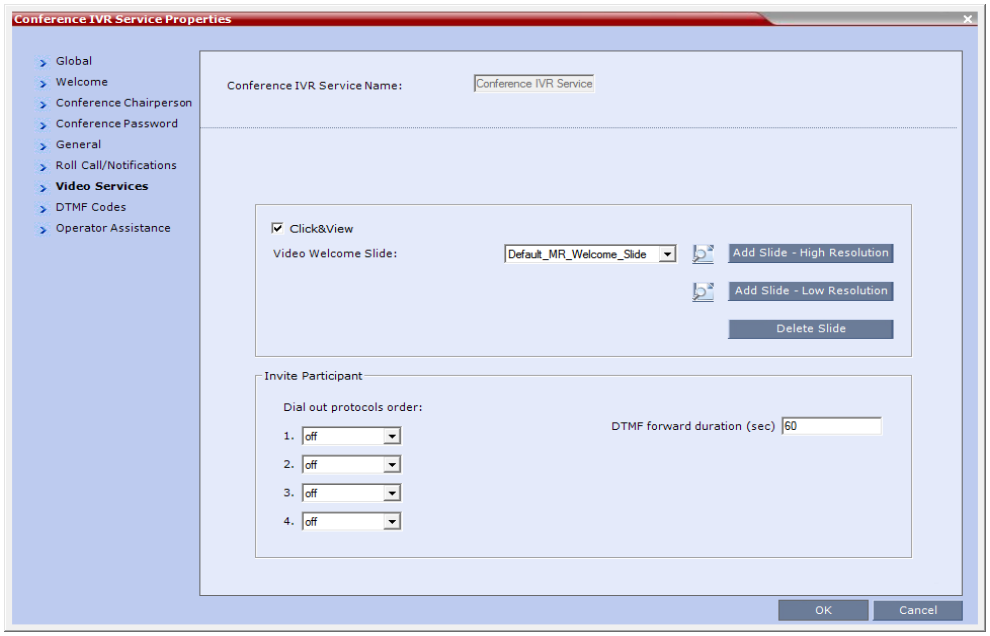


3 In the Message File column of the **Invite Participant** entry, click the drop-down arrow and select the required voice message. The file **Enter_Destination_Number.wav** that is shipped with the system can be used for this message.

To upload a new file, click the **Add Message File**. For more details, see *Creating Audio Prompts and Video Slides*.

4 Click the **Video Services** tab.

The **IVR Services - Video Services** tab is displayed.



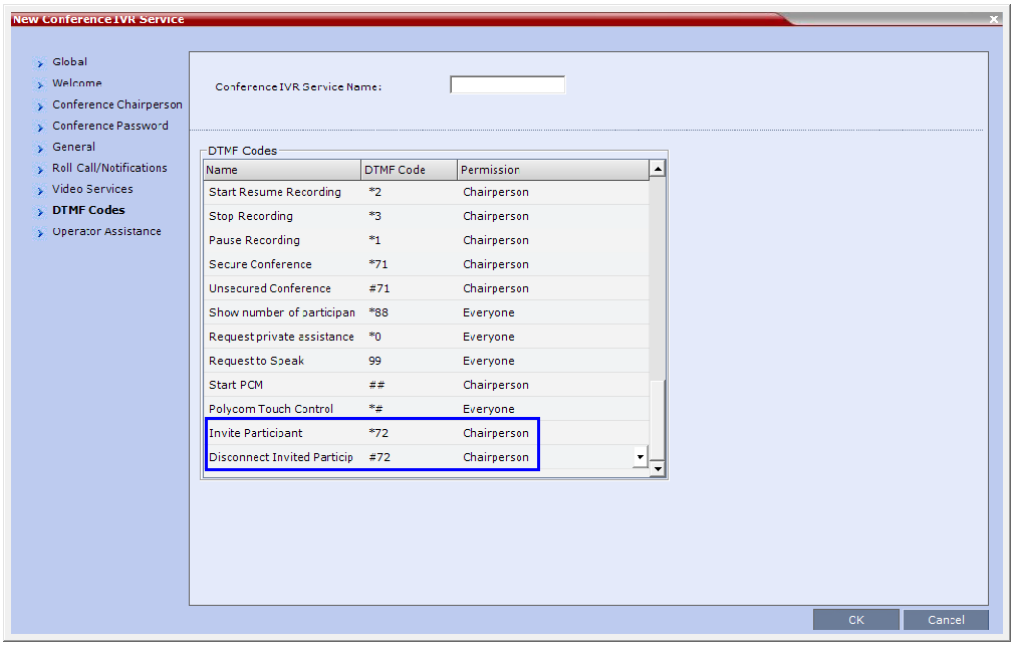
5 Define the following parameters

IVR Services Properties - Video Services Parameters - Invite Participants

Video Services	Description
Dial out protocols order	Select the order of the network protocols that will be used by the system to dial the destination number. The system will start dialing using the first protocol, and if the call is not answered it will continue with the second, third and fourth protocols (if they are enabled) until the call is answered. By default, H.323 is set as the first protocol and SIP as the second while the remaining protocols are disabled (set to Off).
DTMF forward duration	Use this field when connecting to another conferencing entity with an IVR, requiring the input of a password, destination number or ID. Enter the number of seconds that the system will wait for the input of additional DTMF digits such as a password or conference number. The range can be from 10 seconds to 600 seconds. Default is 60 seconds.

6 Click the **DTMF Codes** tab.

The I**VR Services - DTMF Codes** tab is displayed.



7 Make sure that **Invite Participant** and **Disconnect Invited Participant** have DTMF Codes assigned to them. Default system values are *72 (**Invite Participant)** and **#72** (**Disconnect Invited Participant**), however you can enter your own values.
When upgrading from a previous version, default system values may not be assigned if these IVR entries were not defined in your existing IVR Service and have to be manually added to the **DTMF Codes** table.

8 If required, determine who can invite other participants to the conference using DTMF codes by changing the permissions to either **Chairperson** or **Everyone**.

9 Click **OK**.

Disabling the Invite Participant Option

To disable the Invite Participant option:

1 From the **IVR Services - DTMF Codes** tab, delete the DTMF digits from the **DTMF Code** column.

2 Click **OK**.

External IVR Service Control

IVR Services can be controlled externally from an application server supporting the MCCF-IVR (Media Control Channel Framework-Interactive Voice Response) package. The external IVR service is currently being implemented with the integration of the *Polycom RealPresence Virtualization Manager (DMA)* as the application server. When the application server is deployed in the enterprise environment and the Polycom RealPresence Collaboration Server (MCU) is deployed as a media server, the external IVR service can be used to play audio messages, display slides, and collect DTMF input from the participant.

For more information, see Using External IVR Services via the MCCF-IVR Package.

IVR Services Support with TIP Protocol

From Version 8.1, Conference IVR and Entry Queue IVR Services are supported with AVC TIP protocol in conferences that include both TIP-enabled and non-TIP-enabled endpoints. TIP-enabled endpoints can be moved from the Entry Queue to the destination conference if the **TIP Compatibility Modes** settings in the Profile are identical for both conferencing entities (it is recommended to use the same **Profile** for both entities).

The IVR services can be enabled for all **TIP Compatibility Modes:**

- Video only
- Video and Content
- Prefer TIP

IVR media files, WAV for voice messages and JPG for video slides, are all stored on the RealPresence Collaboration Server (RMX).

Guidelines for TIP Support with IVR Services

- IVR default audio files are enabled for all **TIP Compatibility Modes.**
- Only Polycom default Welcome slides are available. Custom **Welcome** slides are not supported.
- TIP-enabled endpoints can send DTMF digits to MCU.
- In an mixed TIP environment, there is no support for content in cascaded conferences. Additionally, Legacy and Lync endpoints cannot view content.

Default IVR Prompts and Messages

The system is shipped with the following audio prompts and messages:

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

Default IVR Messages

Message Type	Message Text	When Played	File Name
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
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

Default IVR Messages

Message Type	Message Text	When Played	File Name
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
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

Default IVR Messages

Message Type	Message Text	When Played	File Name
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
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

Volume Control of IVR Messages, Roll Call and Music

The volume of IVR music, and IVR messages and Roll Call is controlled by the following system flags:

- IVR_MUSIC_VOLUME
- IVR_MESSAGE_VOLUME
- IVR_ROLL_CALL_VOLUME

To control the volume of IVR music, Roll Call and messages:

- Modify the values of the **System Flags** listed in the following table by clicking the menu **Setup > System Configuration**.

If these flags do not appear in the *System Flags* list, they must be manually added. For more information see System Configuration Flags.

Default IVR Messages

Flag	Description
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: 2). 0 – disables playing the music 1 – lowest volume 10 – highest volume
IVR_MESSAGE_VOLUME	The volume of IVR messages varies according to the value of this flag. Possible value range: 0-10 (Default: 2). 0 – disables playing the IVR messages 1 – lowest volume 10 – highest volume **Note:** It is not recommended to disable IVR messages by setting the flag value to 0.

The following **System Flags** do not require an MCU reset:

- IVR_MESSAGE_VOLUME
- IVR_MUSIC_VOLUME
- IVR_ROLL_CALL_VOLUME

For all other flag changes, the MCU must be reset for the modified flag settings (including deletion) to take effect.

IVR Services in TIP-Enabled Conferences

Conference IVR and Entry Queue/Virtual Entry Queues are supported with AVC TIP protocol in conferences that include both TIP-enabled and non-TIP-enabled endpoints.

A Virtual Entry Queue can be configured to either *IVR Only Service Provider* or *External IVR Control* mode.

TIP-enabled endpoints can be moved from the Entry Queue to the destination conference if the **TIP Compatibility Modes** settings in the Profile are identical for both conferencing entities (it is recommended to use the same **Profile** for both entities).

TIP IVR users can access the conference directly or enter the Entry Queue/Virtual Entry Queue and provide a password to access the conference.

The IVR services can be enabled for all **TIP Compatibility Modes:**

- Video only
- Video and Content
- Prefer TIP

IVR media files, WAV for voice messages and JPG for video slides, are all stored on the RealPresence Collaboration Server (RMX).

IVR Services in TIP-Enabled Conferences Guidelines

- IVR default audio files are enabled for all **TIP Compatibility Modes.**
- Only Polycom default **Welcome** slides are available. Custom **Welcome** slides are not supported.
- TIP-enabled endpoints can send DTMF digits to MCU.
- In a mixed TIP environment there is no support for content in cascaded conferences. Additionally, Legacy and Lync endpoints cannot view content.

Entry Queue and Virtual Entry Queue Access

TIP endpoints can dial-in to conferences directly using the IVR, Entry Queue/Virtual Entry Queue and IVR Only Service Provider. For more information see Defining a New Entry Queue.

For more information on Multipoint see the Collaboration With Cisco's Telepresence Interoperability Protocol (TIP).

Configuring the Conference and Entry Queue IVR Services

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 configuration process is the same for TIP and non-TIP enabled Conferences and Entry Queues.

For more information about IVR Services see, Defining a New Conference IVR Service.

For more information about Entry Queues see, Entry Queues.

For more information see Appendix I - Polycom Open Collaboration Network (POCN).

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 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 Properties

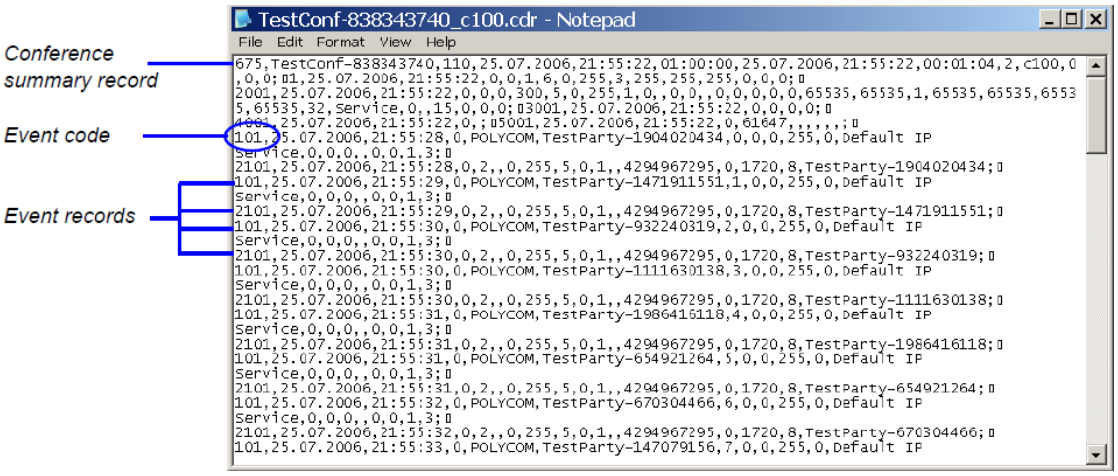
The output of a CDR file depends on the format in which the file was archived and the size of the file.

CDR File Formats

The conference CDR records can be retrieved and archived in the following two formats:

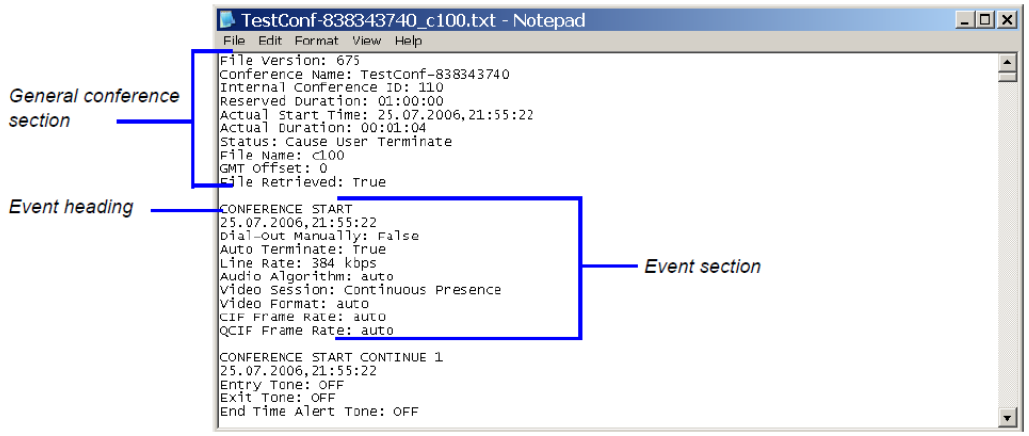
- **Unformatted data** – Unformatted CDR files contain multiple records in "raw data" format. The first record in each file contains general conference data. The remaining records contain event data, one record for each event. Each record contains field values separated by commas. This data can be transferred to an external program such as Microsoft Excel© for billing purposes. The following is a sample of an unformatted CDR file.

  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 following is an example of a formatted CDR file.

Formatted CDR File



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.

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.

Enabling the Multi-Part CDR Option

- 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.
- 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 Appendix 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

You can view the list of CDR files and retrieve them to your local workstation. These files can then be used to generqate billing information, resource usage reports and more by any third party application.

Viewing the Conference Records

You can list all the CDR files that are currently saved on the MCU.

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.

The following fields are displayed:

Conference Record Fields

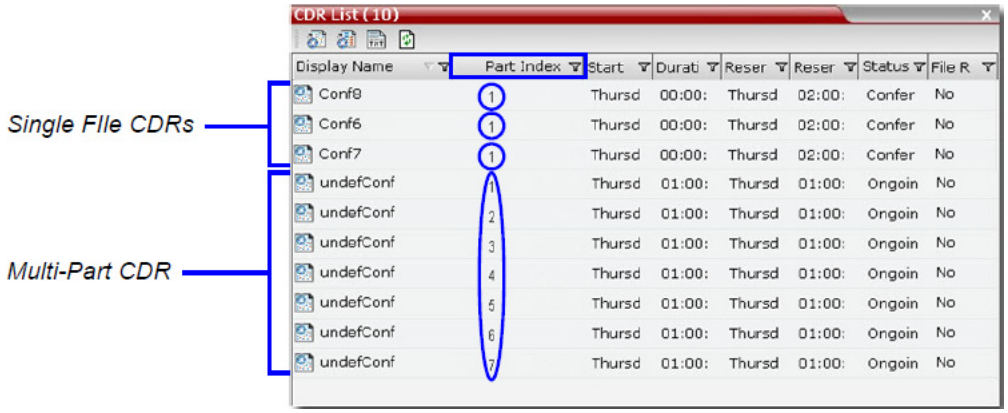
Field	Description
Display Name	The Display Name of the conference and an icon indicating whether or not the CDR record has been retrieved and saved to a formatted text file. The following icons are used:  The CDR record has not been saved.  The CDR record has been saved.
Start Time	The actual time the conference started.

Conference Record Fields (Continued)

Field	Description
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.
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.



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

If the CDR file list is displayed for sometime and you want the latest CDR files to be displayed, you can refresh the 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

You can retrieve the CDR files and store them on your workstation for later use.

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 or select the record to retrieve, and then click the appropriate button on the toolbar as detailed in the following table.

To retrieve multiple CDR records simultaneously, use standard Windows multi-selection methods.

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		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.

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 Polycom® RealPresence® Collaboration Server 800s and Polycom® 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 Client 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.

Installing the RMX Manager Application

The installation of the RMX Manager Application includes two main stages:

- Accessing or downloading the RMX Manager Installer
- Installing the RMX Manager application

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.
- When upgrading the *RMX Manager* from a major version (for example, version 8.7.0) to a maintenance version of that version (for example, 8.7.0.x), the installation must be performed from the same MCU (IP address) from which the major version (for example, version 8.7.0) 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 800s / Virtual Edition Getting Started Guide*, Software Installation.

Once the connection to the Collaboration Server unit is established and the Login window is displayed, the RMX Manager application can be installed.

Upgrade Notes

When upgrading the RMX Manager application, it is recommended to backup the MCU list using the Export RMX Manager Configuration option.

When upgrading the RMX Manager from a major version (for example, version 8.7.0) to a maintenance version of that version (for example, 8.7.0.x), the installation must be performed from the same MCU (IP address) from which the major version (for example, version 8.7.0) 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.

Accessing or downloading the RMX Manager Installer

The RMX Manager installer can be downloaded or accessed and installed on your workstations using one of the following methods:

- Accessing the RMX Manager Application Installer Directly from the MCU
- Downloading the RMX Manager application from the Polycom web site at http://www.polycom.com/support and installing it. The Installation procedure is the same as if you have downloaded the application from the Login screen.
- Accessing the RMX Manager Installer from the Login screen

Accessing the RMX Manager Application Installer Directly from the MCU

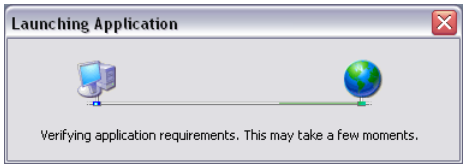
1 Start Internet Explorer and in your browser enter:
http://<Collaboration Server IP Address>/RMXManager.html.

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/RMXManager.html.*

The **RMX Manager Version nnnn** page is displayed.

2 Click the **Install** button.

The installer verifies the application's requirements on the workstation.



3 Continue the Installation as described in Installing the RMX Manager on Your Workstation.

Downloading the Installation files from Polycom Support Site

1 Access the Polycom web site at http://www.polycom.com/support.

2 Click on **Documents and Downloads** and then select **UC Infrastructure** from the drop-down list

3 Select the appropriate RMX/Collaboration Server product.

4 Click the **Web Client (RMX Manager)** link.

The file download dialog box opens.

5 Follow the standard download procedure to either run the installer directly or save the files to your local computer.

6 Continue the Installation as described in Installing the RMX Manager on Your Workstation.

Accessing the RMX Manager Installer from the Login screen

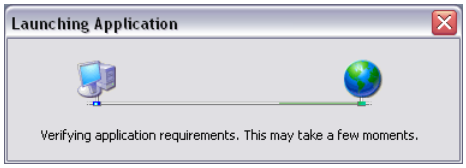
1 Start Internet Explorer and connect to one of the Collaboration Server units in your site. It is recommended to connect to the Collaboration Server installed with the latest software version.

The *Login* screen is displayed.

There is a link to the *RMX Manager Installer* at the top of the right edge of the screen.

2 Click the **Install RMX Manager** link.

The installer verifies the application's requirements on the workstation.

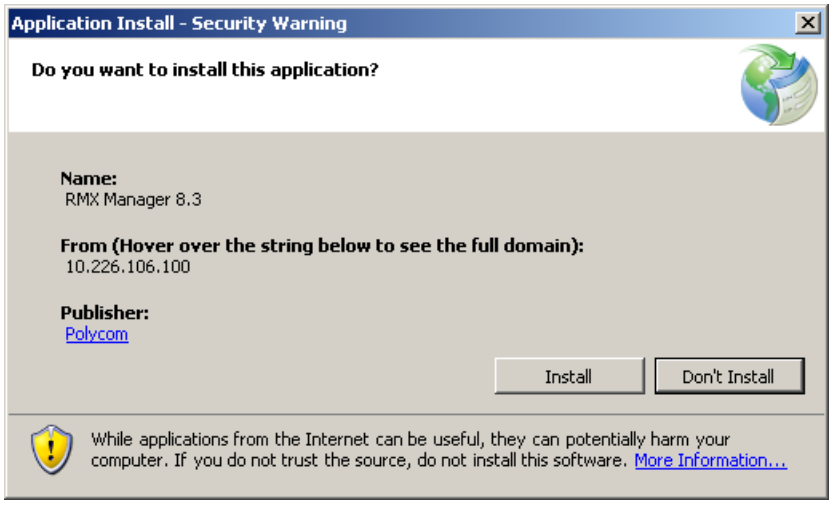


The *Install* dialog box is displayed.

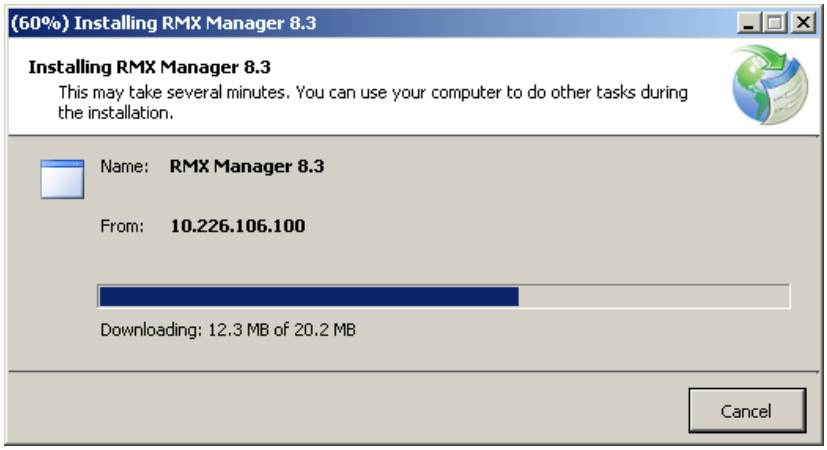
Installing the RMX Manager on Your Workstation

Once the installer has verified that the application's requirements on the workstation are met, the The *Install* dialog box is displayed.

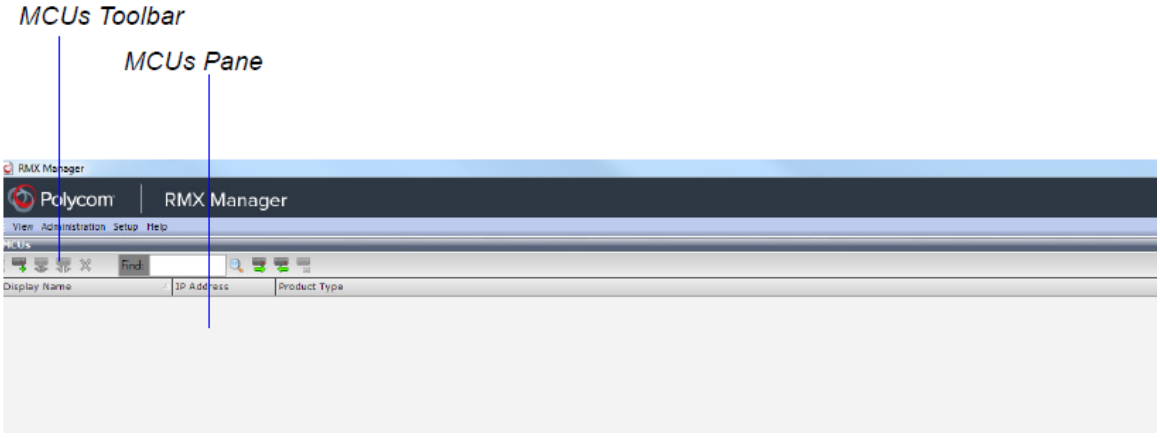
1 Click the **Install** button.



The installation proceeds.



The installation completes, the application loads and the **RMX Manager - MCUs** screen is displayed.



The first time you start the *RMX Manager* application, the *MCUs* pane is empty.

Installing the RMX Manager for Multi-User Capability

The RMX Manager can be installed to be available to all users of a shared computer during the initial installation.

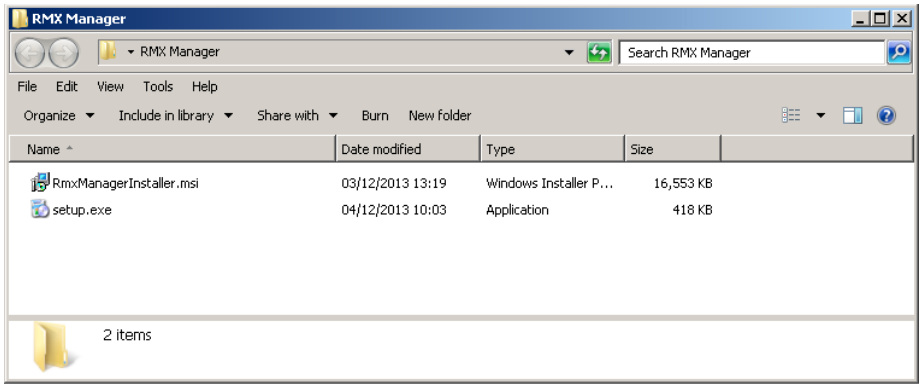
The following procedure is performed after downloading the RMX Manager from the Polycom Support website.

The RMX Manager can still be installed from the Collaboration Server Web Client, but the installation will only be available to the current user.

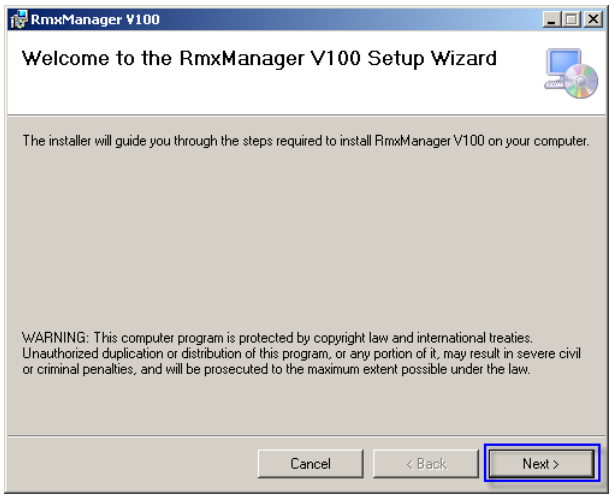
To install the RMX Manager for Multiple Users:

1 Download the RMX Manager installation package from the Polycom Support website.

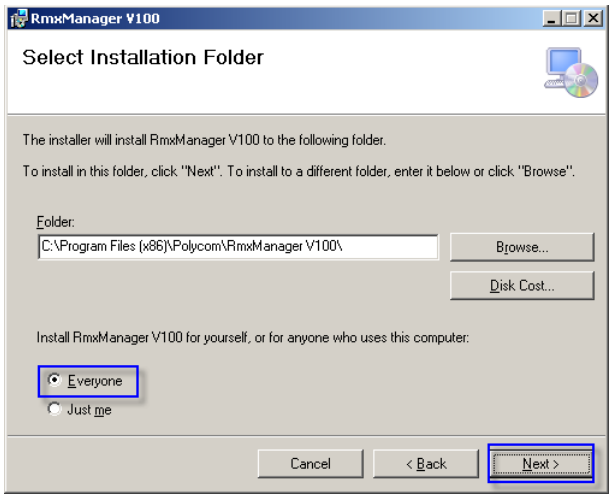
2 Unzip the installation package.



3 Double-click **setup.exe** to open the RMX Manager Setup Wizard.

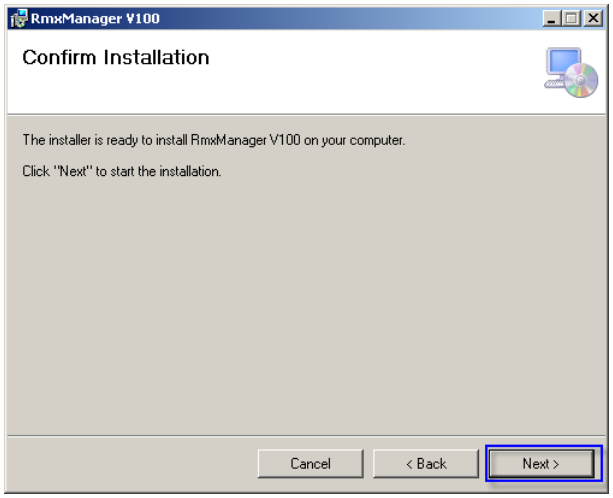


4 Click **Next**.

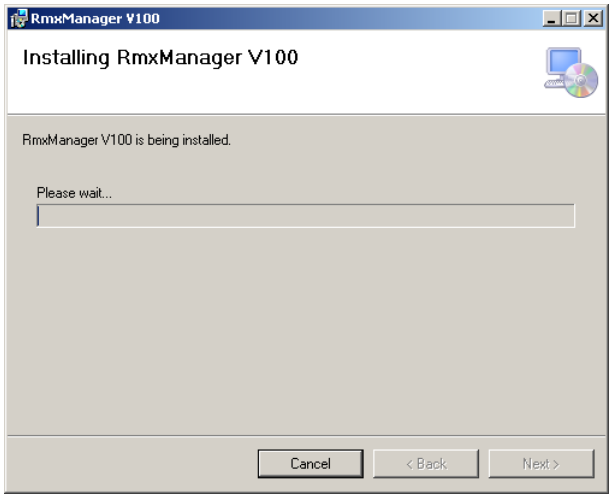


5 Select **Everyone** to install the RMX Manager for all users sharing the computer.
(Select **Just me** to install the RMX Manager just for the current user.)

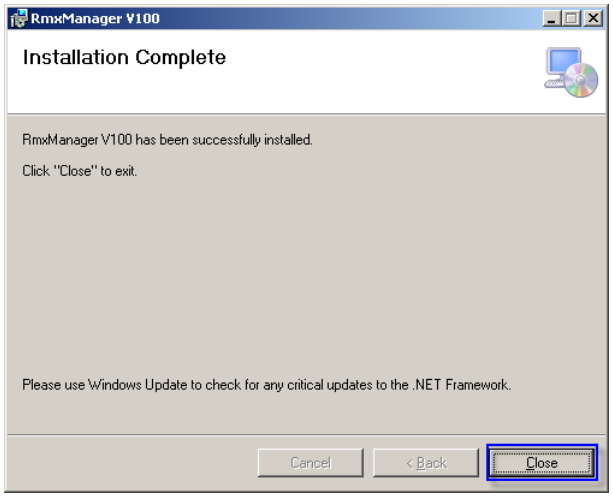
6 Click **Next**.



The installation begins.



When the installation is complete, a confirmation window is displayed.



7 Click **Close** to close the RMX Manager installer.

The installation is complete.

Starting the RMX Manager Application

Once installed, the *RMX Manager* can be run using the `http://` (non-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>/RMXManager.html

or

https://<MCU Control Unit IP Address>/RMXManager.html

2 Press **Enter**.

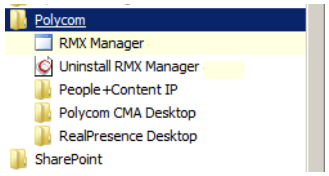
To use the Windows Start menu:

1 Click **Start > Programs.**

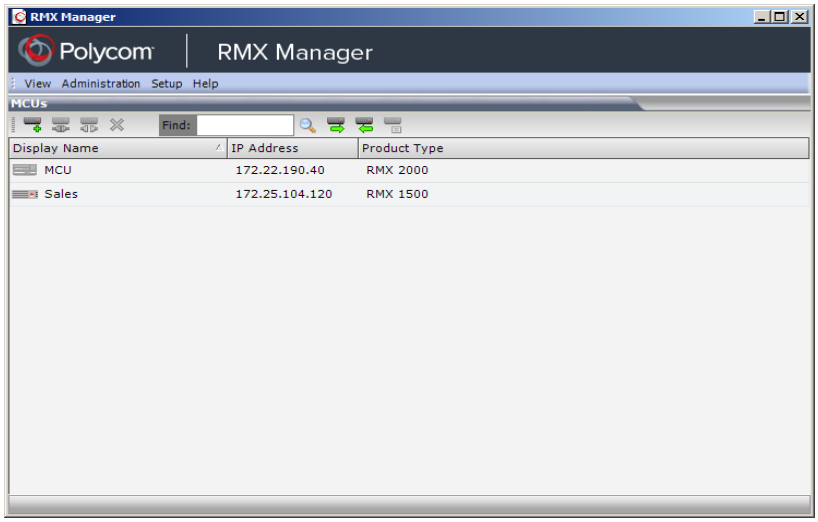
a If the *RMX Manager* is displayed in the recently used programs list, click **RMX Manager** in the list to start the application.

or

b Click **All Programs** > **Polycom** > **RMX Manager**.



The *MCUs* screen is displayed, listing the MCUs currently defined in the *RMX Manager*.



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.

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 VE, 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 Servers. For more details, see Connecting to the MCU.

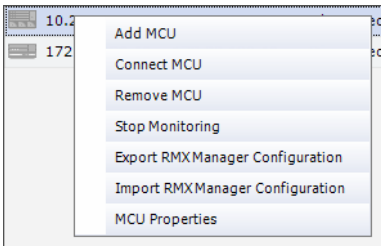
Connecting to the MCU

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.

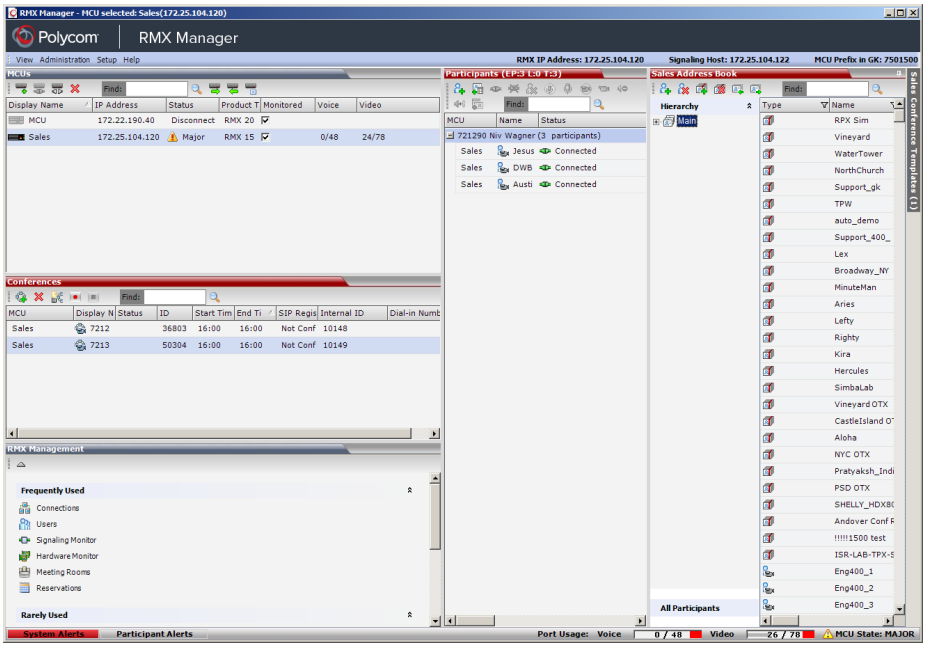
The first *Collaboration Server* unit that is connected to the *RMX Manager* dictates the Authorization Level of Users that can connect to the other MCUs on the list. For example, if the Authorization level of the User POLYCOM is Administrator, all Users connecting to the other MCUs on the list must 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**.

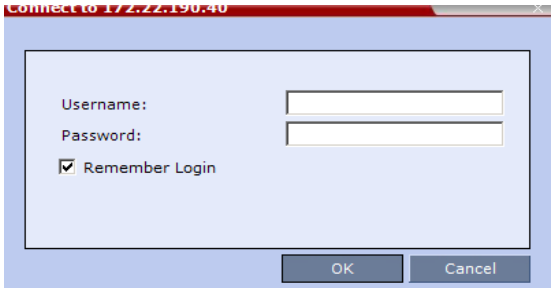


If you are connecting to the MCU from the MCUs opening screen and have defined the Username and Password for the connecting MCU, the system connects to the Collaboration Server, and the RMX Manager Main Screen is displayed.



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.



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**.

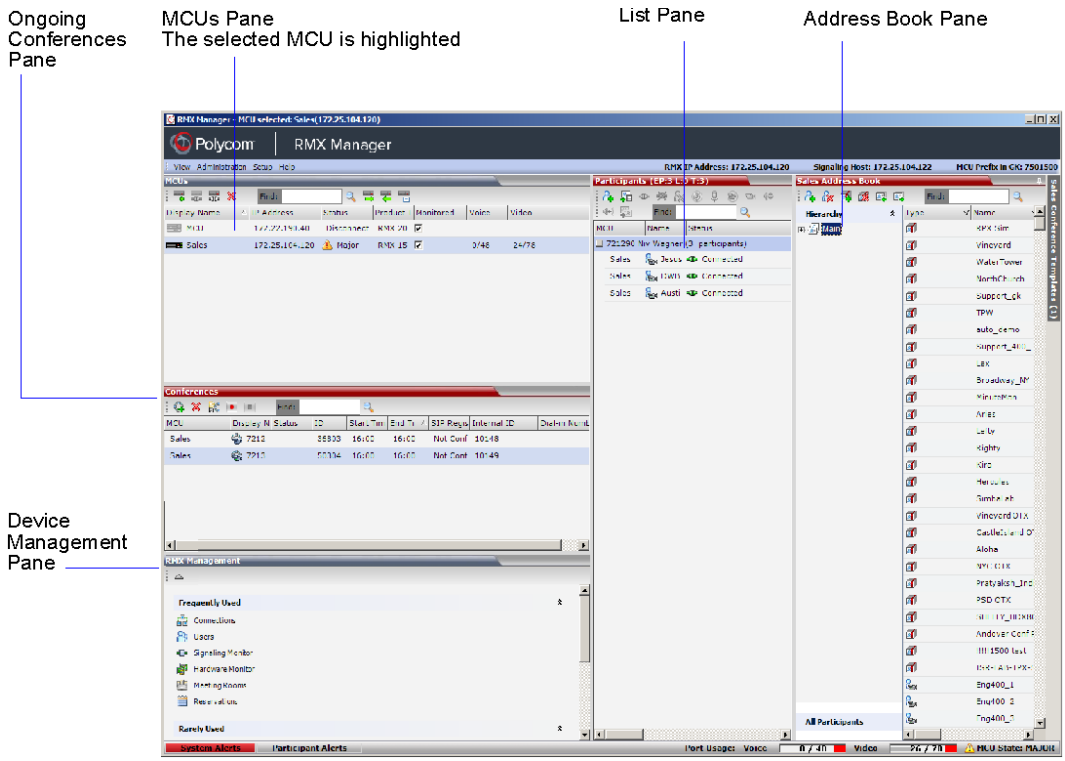
The system connects to the Collaboration Server, and the RMX Manager Main Screen is displayed.

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.

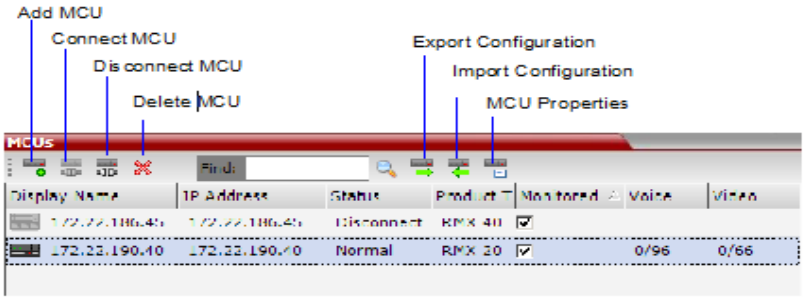


Only one MCU can be selected in the *MCUs* pane. If only one MCU is connected, it is automatically selected. The selected MCU is highlighted.

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.



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:

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 1800, disconnected
	RealPresence Collaboration Server 1800, connected
	RealPresence Collaboration Server 800s, disconnected
	RealPresence Collaboration Server 800s, connected
	RealPresence Collaboration Server Virtual Edition, disconnected
	RealPresence Collaboration Server Virtual Edition, connected

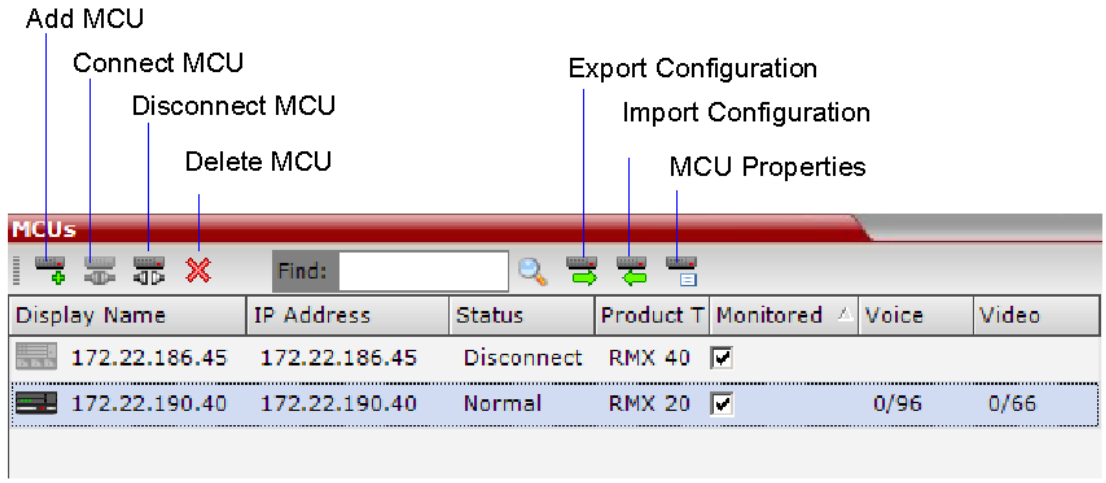
- IP Address - of the MCU's control unit.
- **Status** - The status of the MCU:
  - **Connected** - the MCU is connected to the RMX Manager and can be managed by the RMX Manager user.

- **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 1500/2000/4000, RealPresence Collaboration Server 1800, RealPresence Collaboration Server 800s, RealPresence Collaboration Server Virtual Edition. 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 Resources -** The number of audio resources that are available for conferencing.

MCUs Toolbar

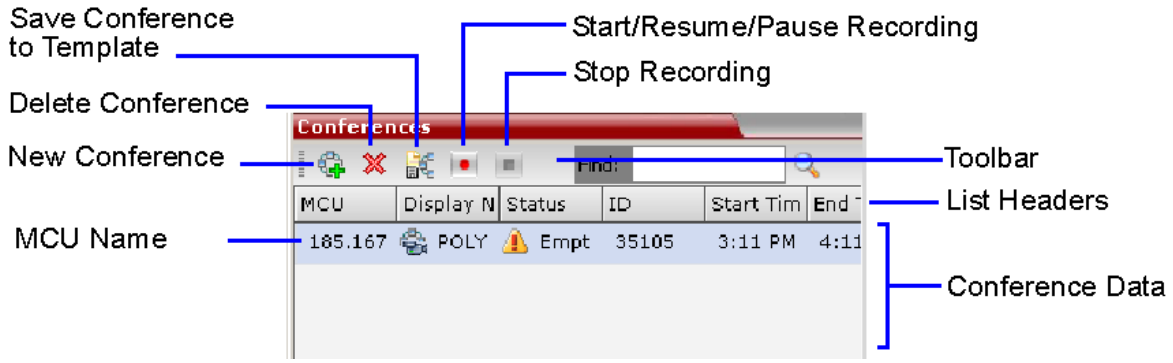
The *MCUs* toolbar contains the following buttons:



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:



If Conference Recording is enabled the following buttons are enabled:

- *Start/Resume Recording* – start/resume recording.
- *Stop Recording* – stop recording.

Pause – toggles with the *Start/Resume* button.**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.

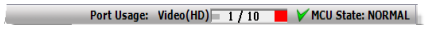
List Pane

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.

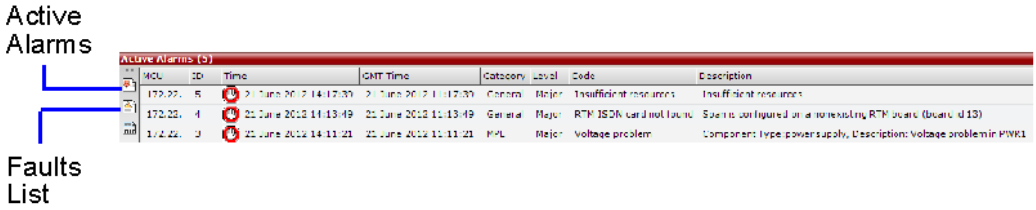


System Alerts

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*.



For more information about **Active Alarms** and **Faults List,** see System and Participant Alerts.

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.

Port Usage Gauge

IThe Port Usage Gauge displays for the selected MCU:

- The total number of Video ports in the system.
- The number of Video ports in use.
- The High Port Usage threshold.

For more details, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Port Usage Gauges.

MCU State

The *MCU State* indicator displays the status of the selected MCU.

For more details, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, The basic unit used for reporting resource usage in the Port Gauges is HD720p30. Results are rounded to the nearest integer..

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 800s / Virtual Edition Getting Started Guide*, Address Book.

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 800s / Virtual Edition Getting Started Guide*, Conference Templates.

Adding MCUs to the MCUs List

The RMX Manager can connect to one or several Collaboration Servers 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.

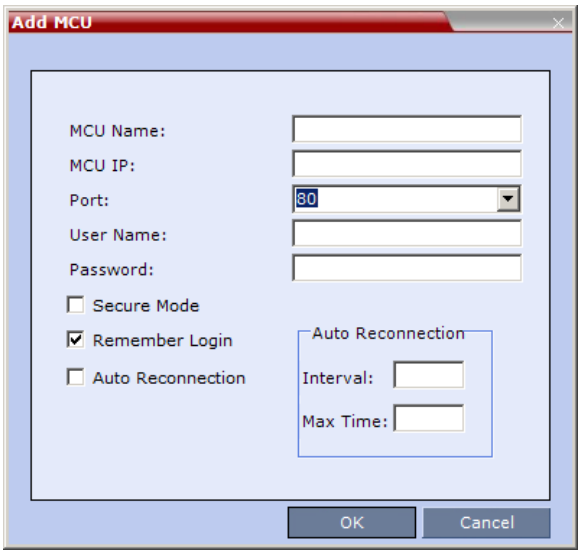
The Collaboration Server must be installed and its IP addresses properly configured in the Management Network Service before defining its connection parameters in the RMX Manager application.

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:



MCU Properties

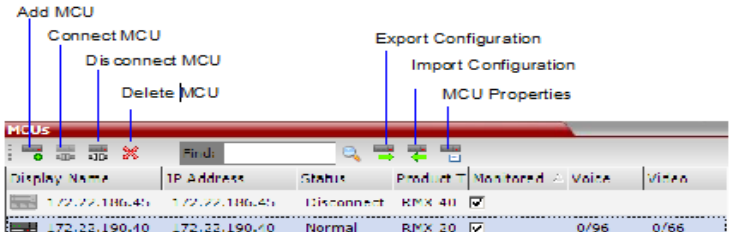
Field	Description
MCU Name	Enter the name of the MCU on the network.
MCU IP	Enter the IP address of the MCU's Control Unit. The IP address must be identical to the one configured in the MCU during first entry Configuration. For more details, see the *Polycom® RealPresence Collaboration Server 800s / 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 **80**. For a Secured connection (using TLS or SSL), enter **443**.
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.

Field	Description
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.



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.
- 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.
- 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.
- Start a *Reservation:*
  - If the *Start Time* of the *Reservation* is past due the conference becomes ongoing immediately.
  - If the *Start Time* of the *Reservation* is in the future the conference becomes ongoing, at the specified time on the specified date.

  For more information, see Starting a Reservation.
- Start any *Conference Template* saved in the *Conference Templates* list.
  For more information, see Starting an Ongoing Conference or Reservation From a Template.

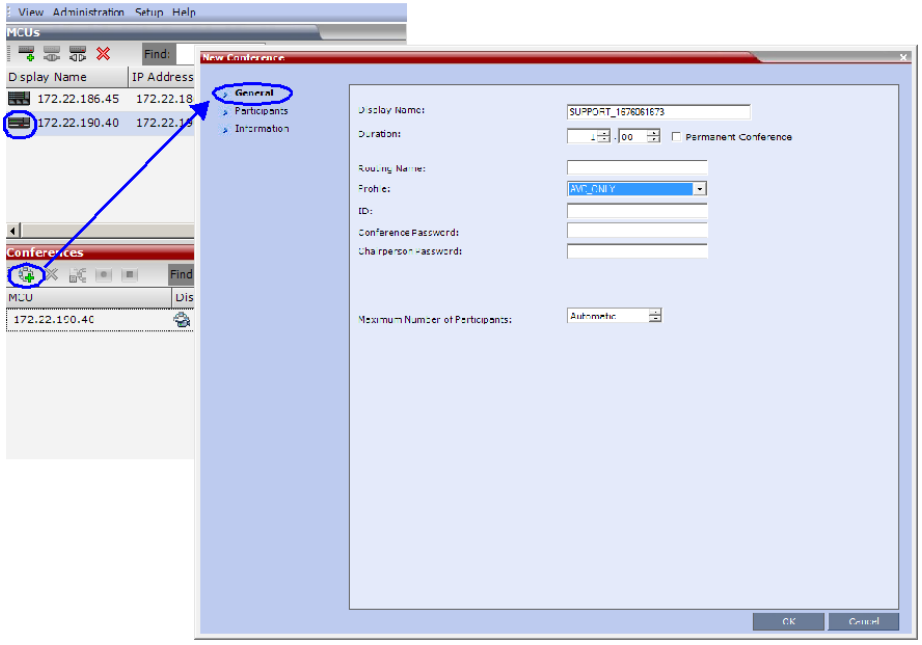
Starting a Conference from the Conferences Pane

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.



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**.

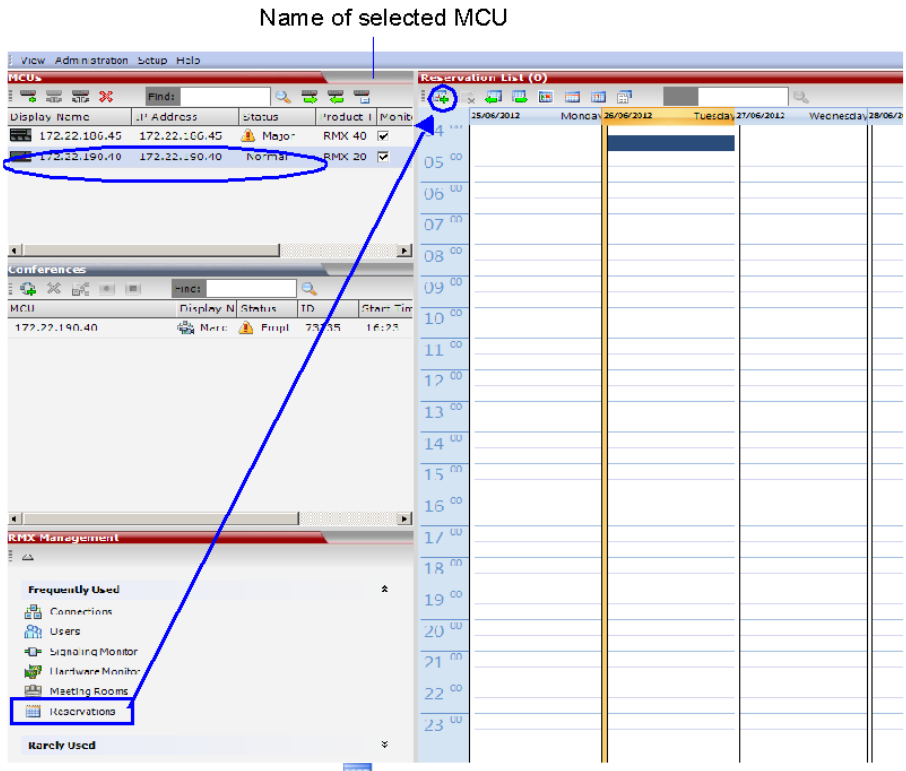
For more details, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Starting an AVC CP Conference from the Conferences Pane.

Starting a Reservation

To start a conference from the Reservation Calendar:

1 In the **MCUs** pane, select the MCU to run the conference.

2 In the **RMX Management** pane, click the **Reservation Calendar** button ( ).
The Reservation Calendar is displayed.



3 Click the **New Reservation** ( ) button.

For more information, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Scheduling an AVC-based Reservation.

Starting an Ongoing Conference or Reservation From a Template

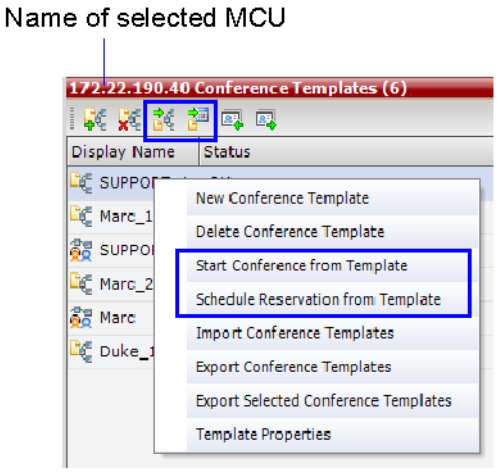
An ongoing conference or a Reservation can be started from any Conference Template saved in the *Conference Templates* list of the selected MCU.

To start an ongoing conference or a reservation from a Template:

1 In the **MCUs** pane, select the MCU to run the conference.
2 In the **Conference Templates** list, select the Template you want to start as an ongoing conference.

3 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.



The conference is started.

For detailed description of *Conference Templates*, see Conference Templates.

Monitoring Conferences

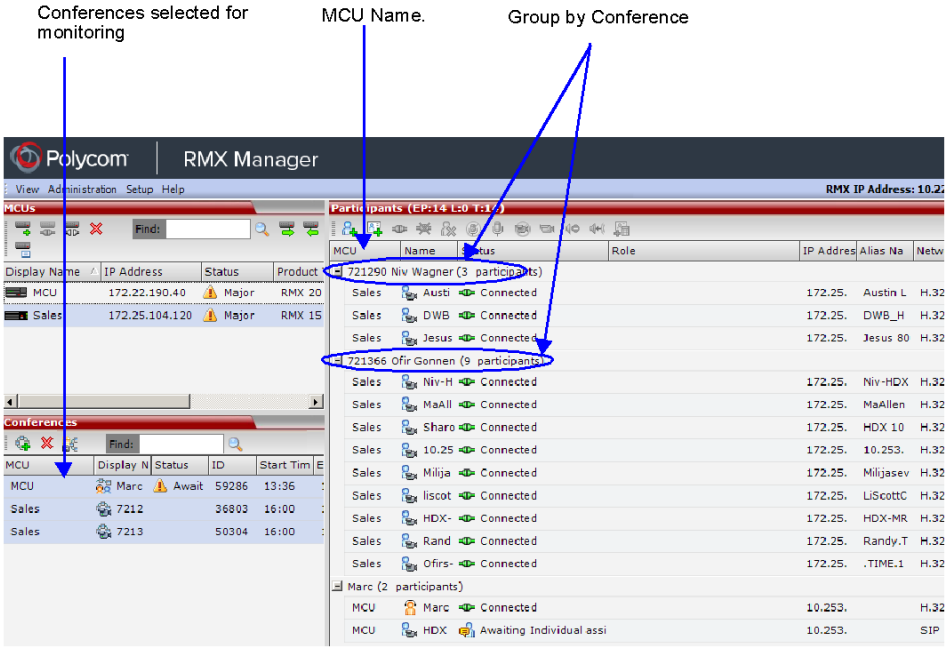
When MCUs are connected to the *RMX Manager* they are automatically monitored, that is, any ongoing conference that is started on that MCU is automatically added to the Conferences pane and its participants are monitored.

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.

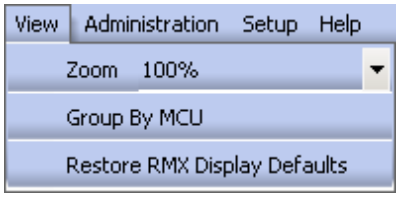


Grouping the Participants by MCU

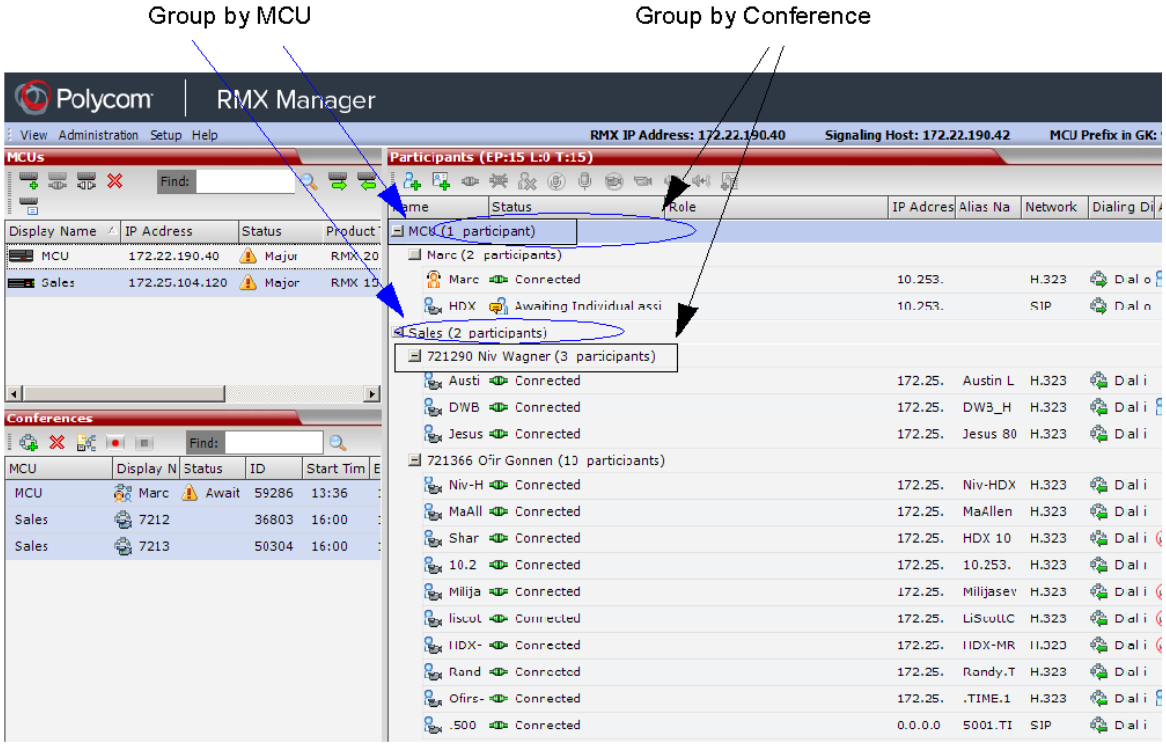
The Participants can be grouped by MCU and then by conferences.

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.



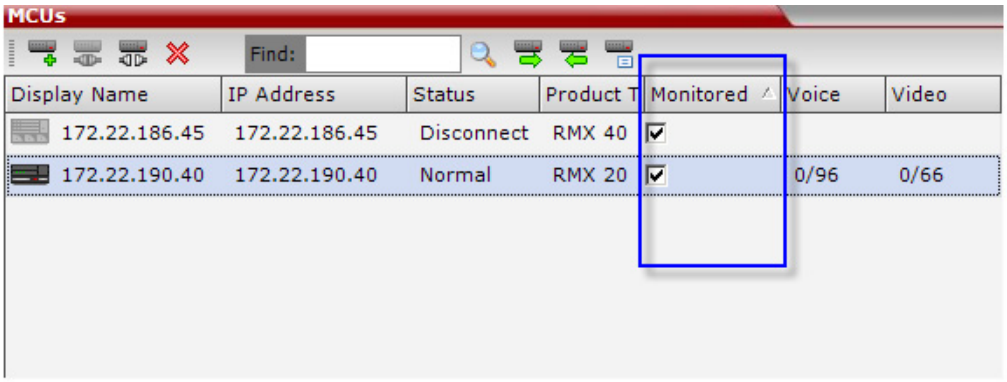
To toggle between the two display modes, click **View > Group by MCU.**

Start Monitoring/Stop Monitoring

By default, all conferences running on connected *Collaboration Server*s are monitored.

You can stop the automatic monitoring of conferences on a specific MCU in one of the following methods:

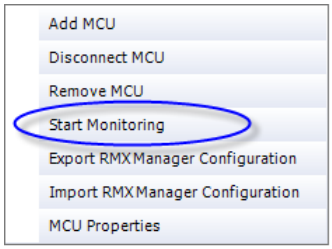
- By clearing the check box in the *Monitored* column in the *MCUs* pane.


MCUs
Find:
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**.

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**.



Modifying the MCU Properties

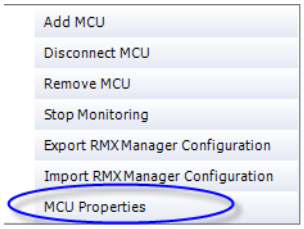
You can view the currently defined MCU settings, and modify them when required, for example, change the MCU name, IP address or Secured mode.

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**.



   The **MCU Properties** dialog box opens.
2. Define/modify the required parameters. For details, see MCU Properties.
3. Click **OK**.

Disconnecting an MCU

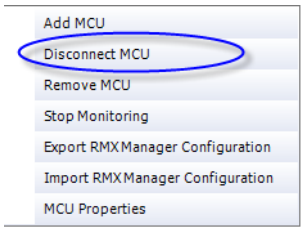
An MCU can be disconnected from the *RMX Manager*, without removing it from the *MCUs* list.

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

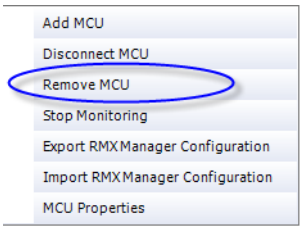
An MCU can be removed from the *RMX Manager*. This function should be used if the MCU hardware was disconnected and removed from the network.

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.

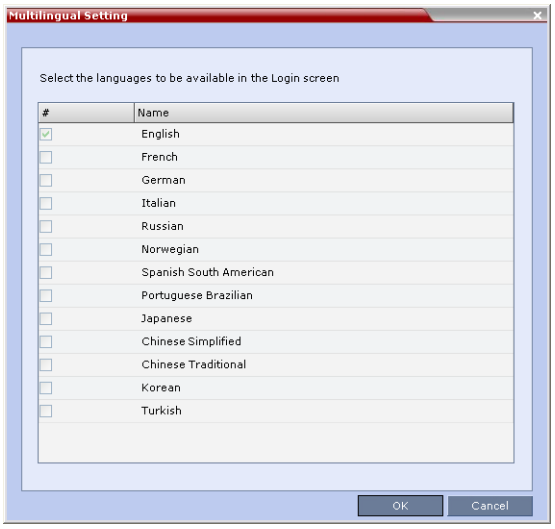
Changing the RMX Manager Language

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.



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.

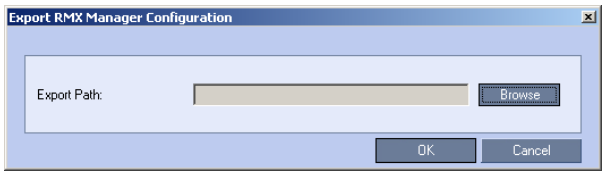
The exported file is save in XML format and can be edited in any text editor that can open XML files.

To Export the RMX Manager Configuration:

1 In the 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**.

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.



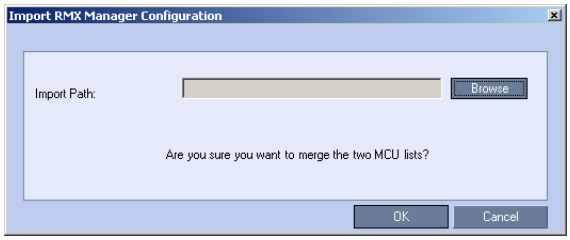
The selected file path is displayed in the **Export Path** box.

3 Click **OK** to export the RMX Manager configuration.

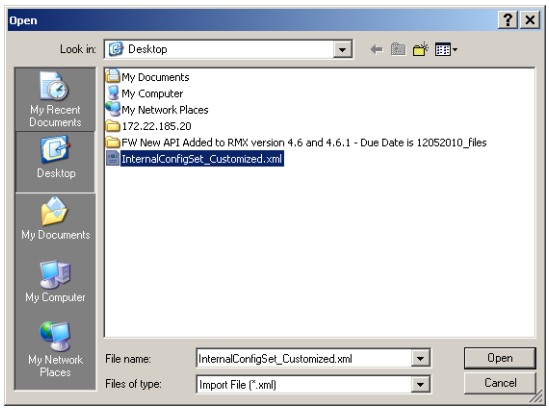
To Import the RMX Manager Configuration:

1 In the 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**.

The **Import RMX Manager Configuration** dialog box opens.



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.



3 Select the XML file previously saved, and click the **Open** button.
The selected file path is displayed in the *Import Path* box.

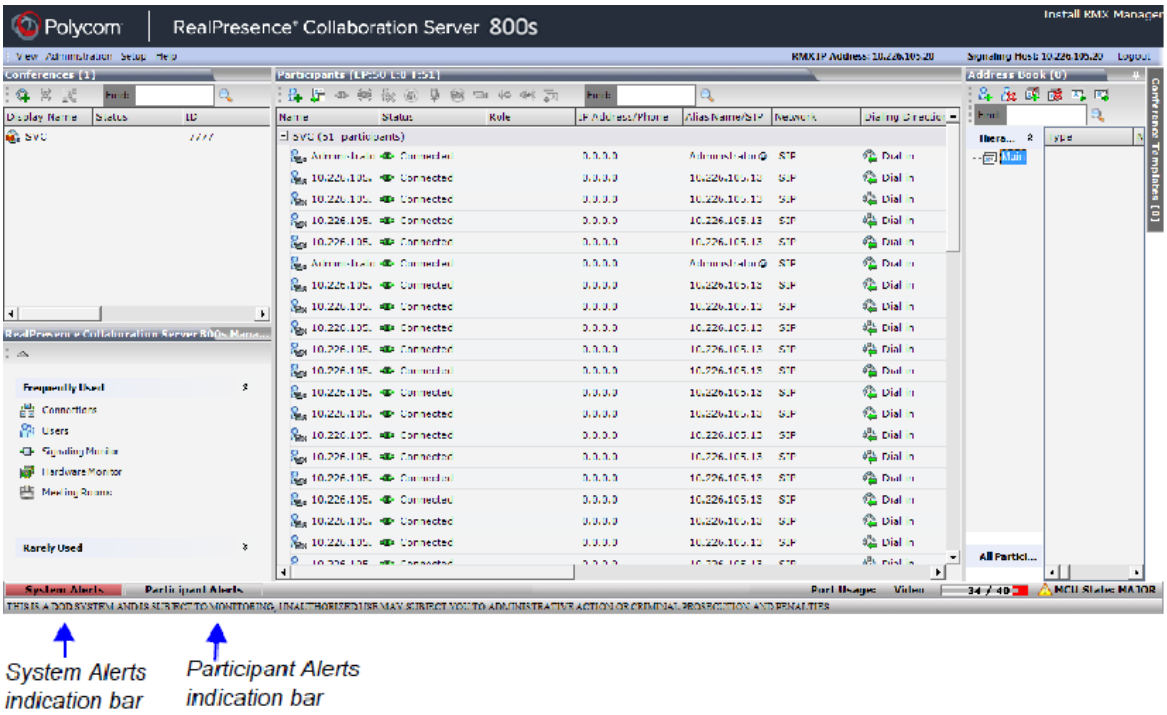
4 Click **OK** to import the file.

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.

Collaboration Server 800s Status 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.

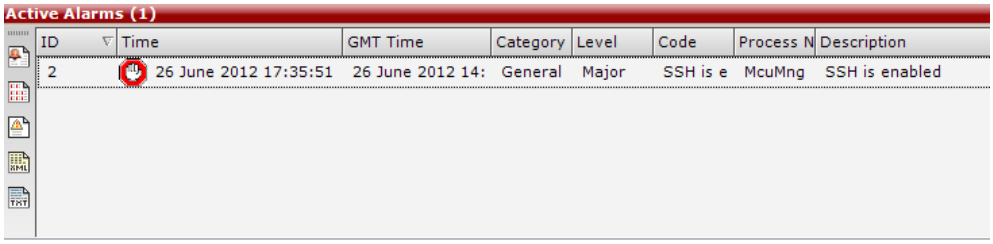
System Alerts

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.



The following columns appear in the **Active Alarms** pane:

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.

2 Click one of the following two buttons to view its report in the **System Alerts** pane:

System Alerts Buttons	
	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.

The Participant Alerts pane displays similar properties to that of the Participant List pane. For more information, see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide,* Participant Level Monitoring.

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.

NTP Servers can be used if:

- NTP servers use Version 4 as it 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.

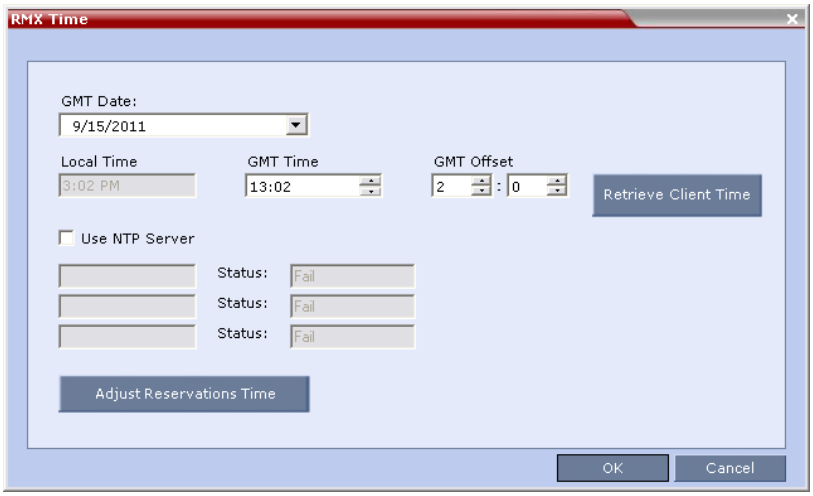
Altering the clock

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.**

The **RMX Time** dialog box opens.



2 View or modify the following fields:

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.

RMX Time – Fields Properties

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**: TheCollaboration 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*)	Use this button to adjust the start time of all the reservations in one operation. For more information see Adjusting the Start Times of all Reservations. Not supported in the RealPresence Collaboration Server.

After resetting the MCU a delay may occur when synchronizing with the external NTP server.

Resource Management

This section describes how the MCU resources are managed by the MCU and how they are used by the MCU to connect participant to conferences.

This section describes:

Resource Capacity

The MCU resources are determined by the MCU type, allocated hardware (Virtual Edition), and the system license you have purchased. The total number of licensed resources is shown in the System Information.

MCU Capacities in CP Only Conferencing and SVC Only Conferencing

The following table describes the resource capacity allocations for the Collaboration Server per Resource type (voice or video protocol) and per resolution.

Resource Capacity Allocation per Resource Type

Resource Type	Number of Resources
VoIP Ports	120
CIF Ports	40
SD Ports (4CIF)	40
HD 720p30	20
HD 1080p30	10
VGA RTV	20
SVC Only	60

One HD720p30 port equals 3 SVC ports or 2 CIF ports.

Resource Usage in SVC Conferencing

During a SVC conference, each SVC-endpoint uses a video port that is equivalent to a third of HD720p30 port. When sharing content an additional video resource is used.

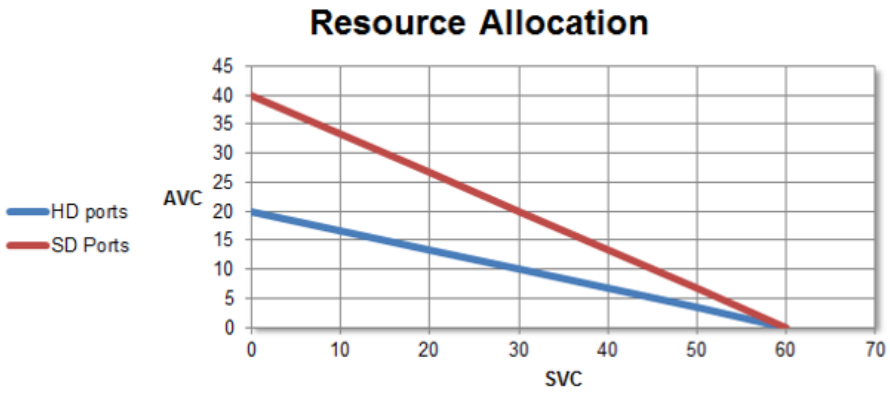
MCU Capacities in Mixed CP and SVC Conferencing

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:



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.

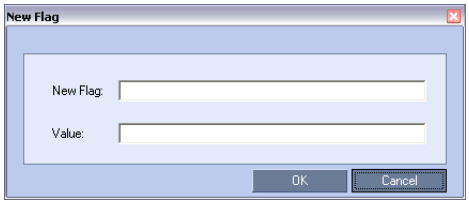
To force CIF resource:

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 Resetting the RealPresence Collaboration Server Virtual Edition.

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 Resetting the RealPresence Collaboration Server Virtual Edition.

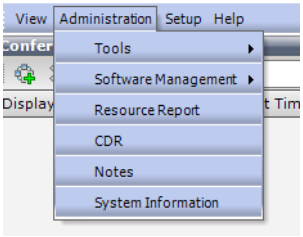
Resource Reports

When viewing the Collaboration Server resource report, the resource allocations are described in AVC HD720p30 units.

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

» In the main toolbar, click **Administration > Resource Report**.

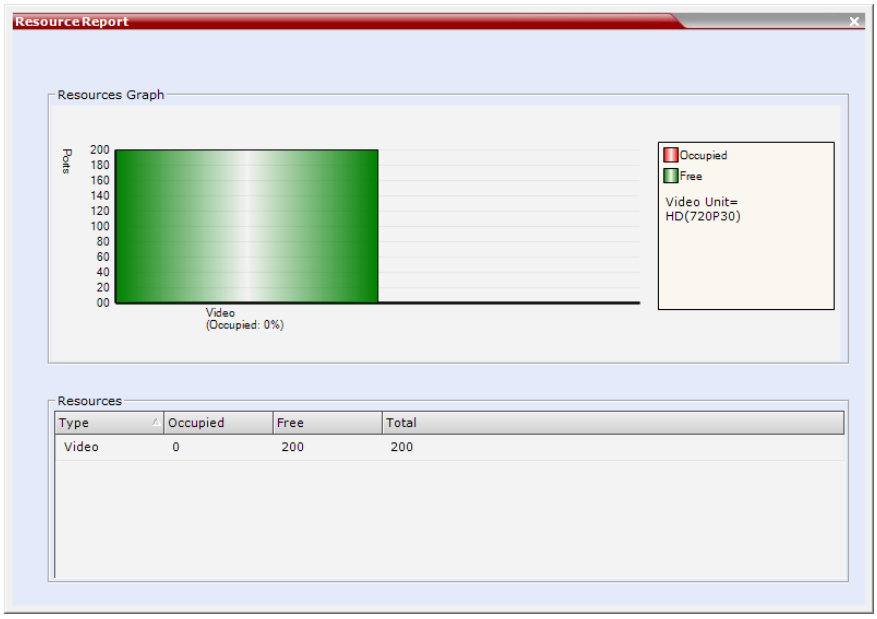


For each resource type, the Resource Report includes the following columns:

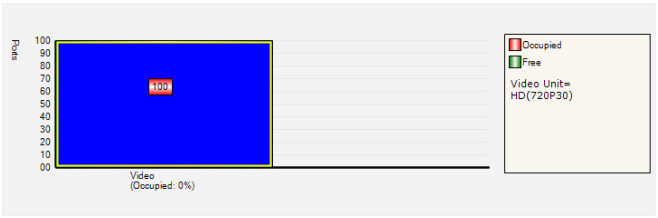
Resource Report Fields Parameters

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.

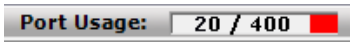


The actual number of occupied or free resources can also be displayed by moving the cursor over the columns of the bar graph. Moving the cursor over the *Video* bar displays the following view:



Port Gauge

The *Port Gauge* in the *Status Bar* show the numbers as they appear in the resource report. In the following example, 20 of the 400 system resources are shown as occupied.



Resource Capacities in AVC CP, SVC and Mixed Mode Conferences

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 tables show the actual resource capacity utilization for both CP only and mixed CP and SVC conferences in AVC HD720p30 units a single

MCU Resource Management by RealPresence Resource Manager, Polycom CMA and Polycom RealPresence DMA System

When the RealPresence Resource Manager, Polycom CMA and Polycom RealPresence DMA system are part of the solution, following a request by the RealPresence Resource Manager 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, 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.In *Flexible Resource Capacity Mode*, *CMA/DMA* receive information about how many *Video* (*CIF*) and *Audio* resources are occupied per conference or MCU according the request type sent by the *CMA* and *DMA*.
- Occupied resources are resources that are connected to ongoing conferences. Disconnected endpoints in an ongoing conference are not counted as occupied resources.
- An ongoing conference that does not include participants and the *Send Content to Legacy Endpoints* option is disabled does not occupy resources. If the *Send Content to Legacy Endpoints* option is enabled, the conference occupies one SD resource.
- 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.

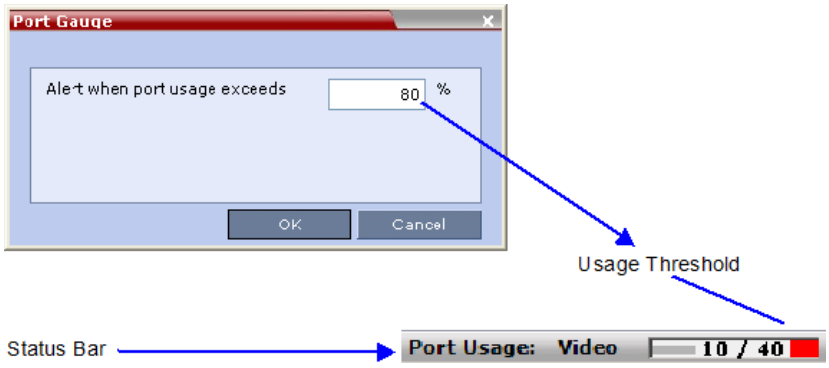
The default port capacity usage threshold is 80%.

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.



2 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%.

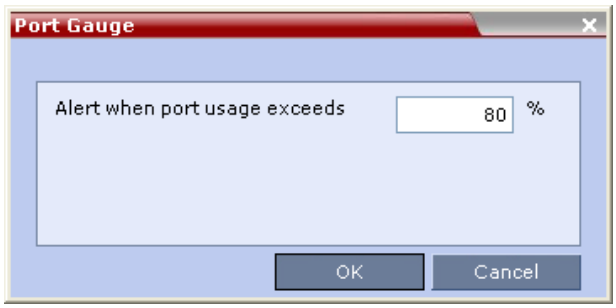
3 Click **OK.**

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.



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

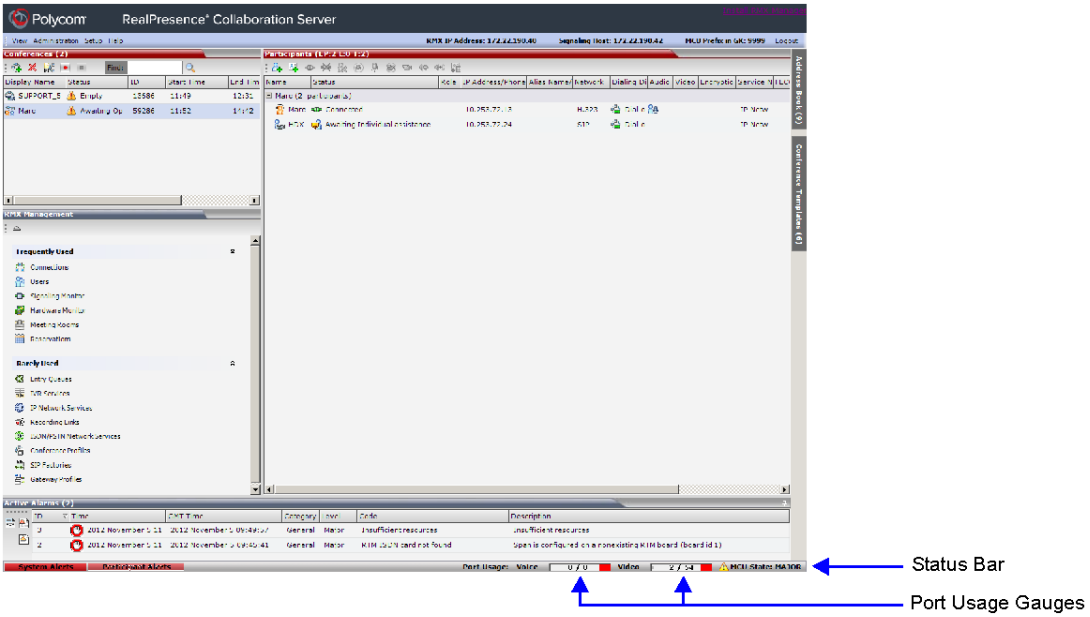
video, and H.323 cascaded links. When the call is rejected by the MCU because of lack of resources, the appropriate indication will be sent by the MCU to the SIP audio endpoint.

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

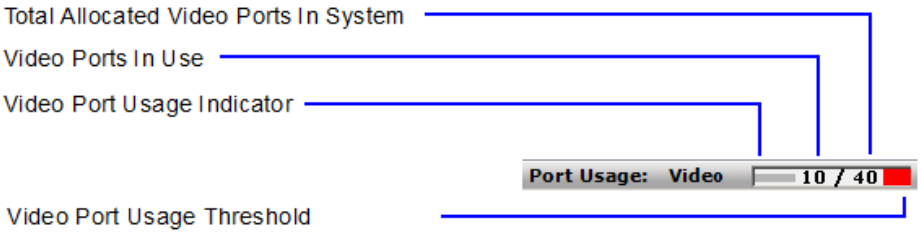
The Port Usage Gauge is displayed in the *Status Bar* at the bottom of the Collaboration Server Web Client screen.



Port Usage Gauge

The Port Usage Gauge displays for the selected MCU:

- The total number of *Video* ports in the system.
- The number of *Video* ports in use.
- The *High Port Usage* threshold.



The basic unit used for reporting resource usage in the Port Gauges is HD720p30. Results are rounded to the next integer.

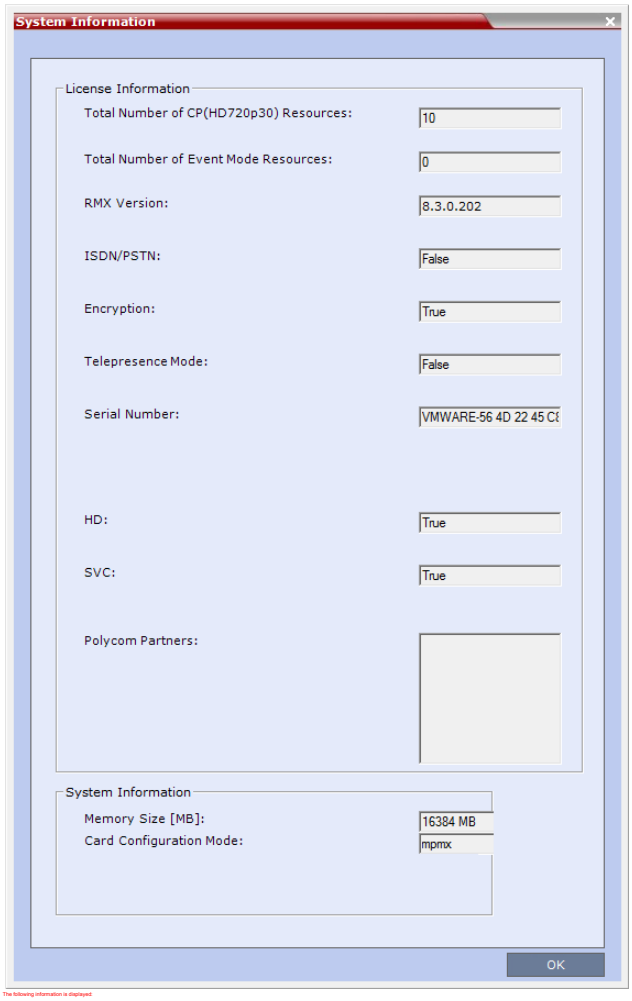
System Information

System Information includes License Information and general system information.

To view the System Information properties box:

- On the Collaboration Server menu, click **Administration > System Information**.

  The S**ystem Information** dialog box is displayed.



System Information

Field	Description
Total Number of CP (HD720p30) Resources	Displays the number of HD720p30 video resources licensed for the system. Each HD720p30 resource represents 3 CIF video resources. Each SVC resource is equivalent to one CIF video resource.

System Information (Continued)

Field	Description
Total Number of Event Mode Resources	Displays the number of video/voice resources 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 MCU.
Encryption	Indicates whether Encryption is included in the MCU license. Encryption is not available in all countries. Range: True / False
Telepresence Mode	The field value indicates whether the system is licensed to work with RPX and TPX Telepresence room systems. 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	Indicates if the MCU is licensed to run SVC-based conferences.
Polycom Partners	Indicates that the System Software contains features for the support of specific Polycom Partner environments.
Memory Size [MB]	Indicates the MCU system memory size in Megabytes. **Note:** If Memory size is 512MB (Collaboration Server 1500/2000/4000 only), *Version 7.1 and later* are not supported. DO NOT upgrade the system to *Version 7.1 and later*.
Card Configuration Mode	Not applicable to Collaboration Server 800s/Virtual Edition. • • •

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:

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.

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.
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.

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.
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.

Unified MIB SNMP Fields (Continued)

Name	Type	Description
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:

- **ColdStart trap**. This is a standard trap which is sent when the MCU is reset.

An Example of a ColdStart Trap

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)

- **Authentication failure trap**. This is a standard trap which is sent when an unauthorized community tries to enter.

An Example of an Authentication Failure Trap

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)

- **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).

An Example of an Alarm Fault 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

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.

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	
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	

SNMP MIB entities that do not generate traps (Continued)

Key	Description	Comment
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
5001	License not found	Not in use (Product activation failure is trapped)

Defining the SNMP Parameters in the Collaboration Server

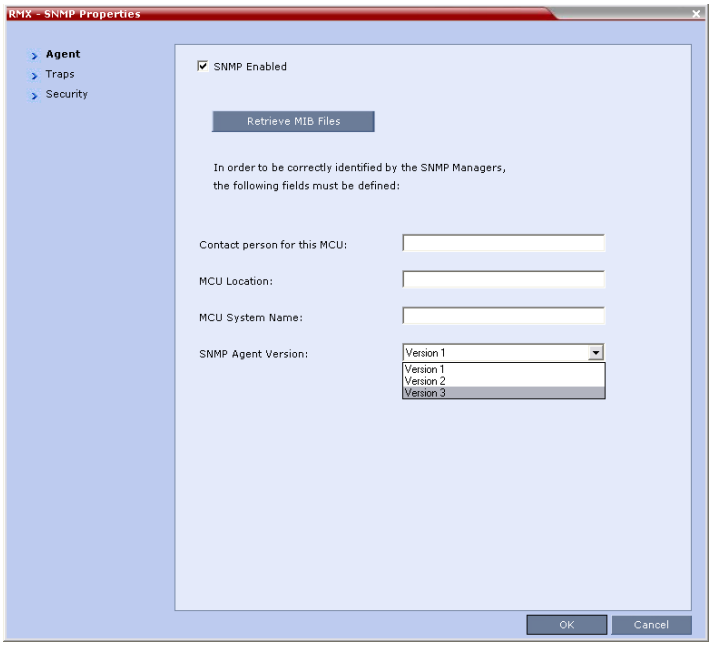
The SNMP option is enabled via the Collaboration Server Web Client application.

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 **RMX SNMP Properties - Agent** dialog box is displayed.



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.

Collaboration Server-SNMP Properties - Agent Options

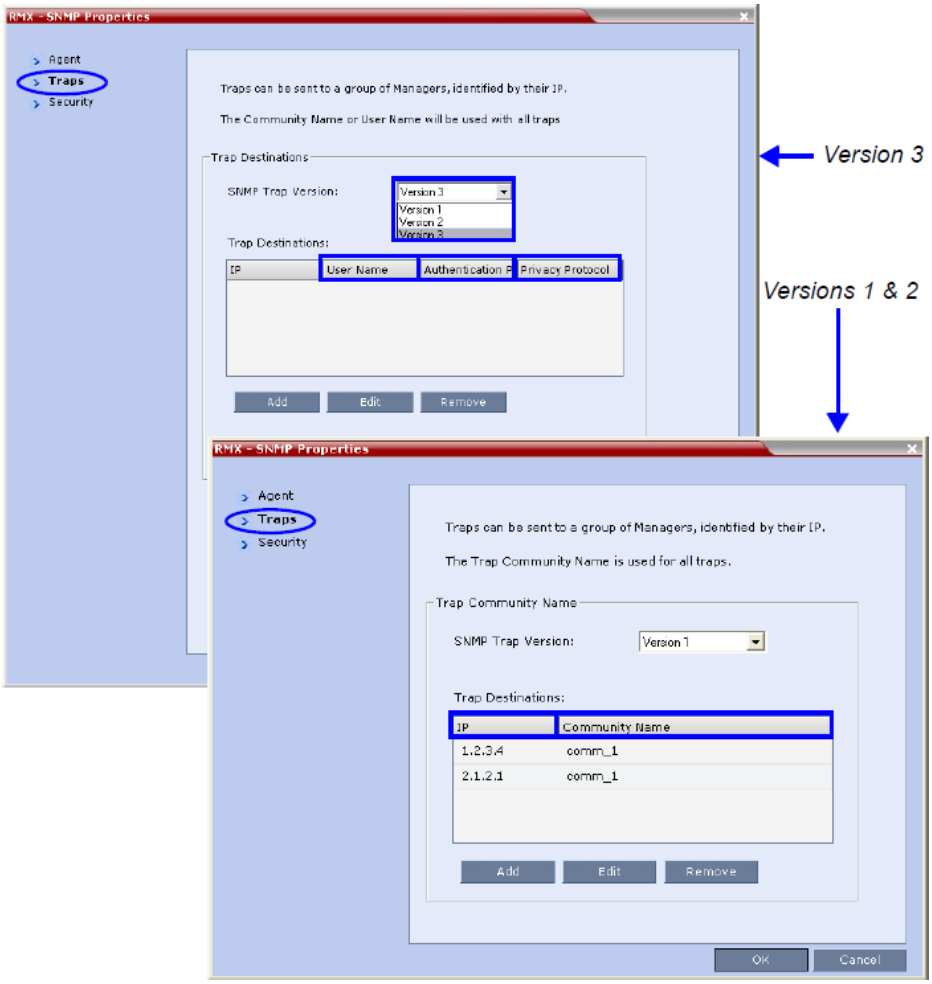
Field	Description
Contact person for this MCU	Type the name of the person to be contacted in the event of problems with the MCU.
MCU Location	Type the location of the MCU (address or any description).

Collaboration Server-SNMP Properties - Agent Options

Field	Description
MCU System Name	Type the MCU's system name.

9 Click the **Traps** tab.

The **SNMP Properties – Traps** dialog box opens.



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.

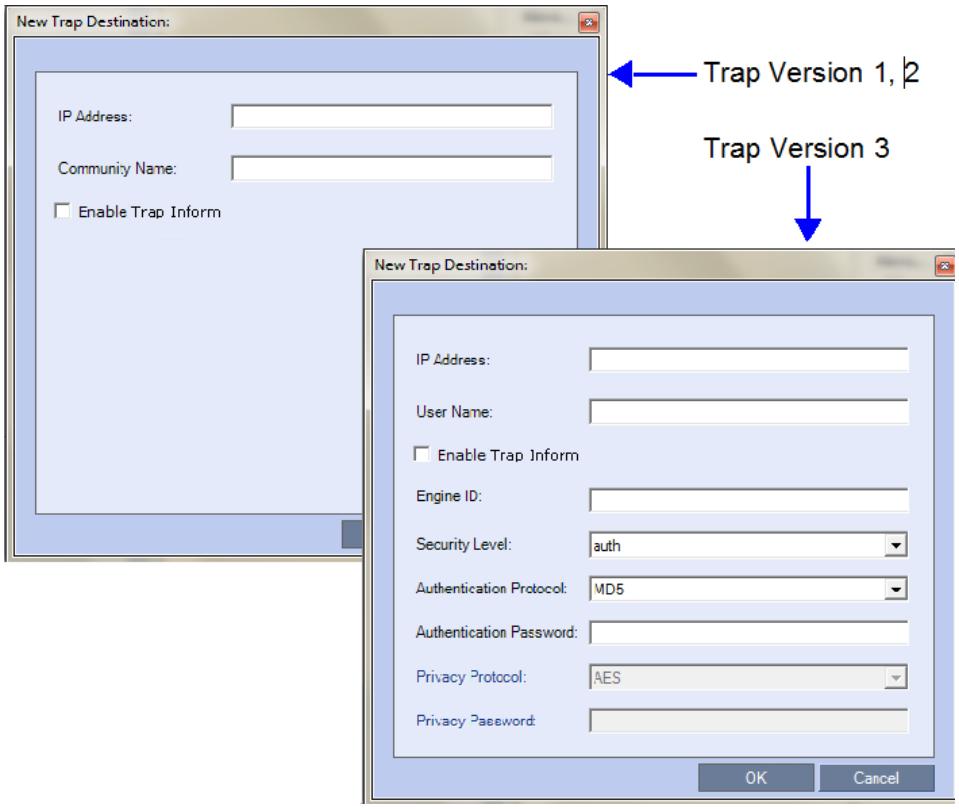
10 Define the following parameters:

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.



12 Define the following parameters:

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

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.

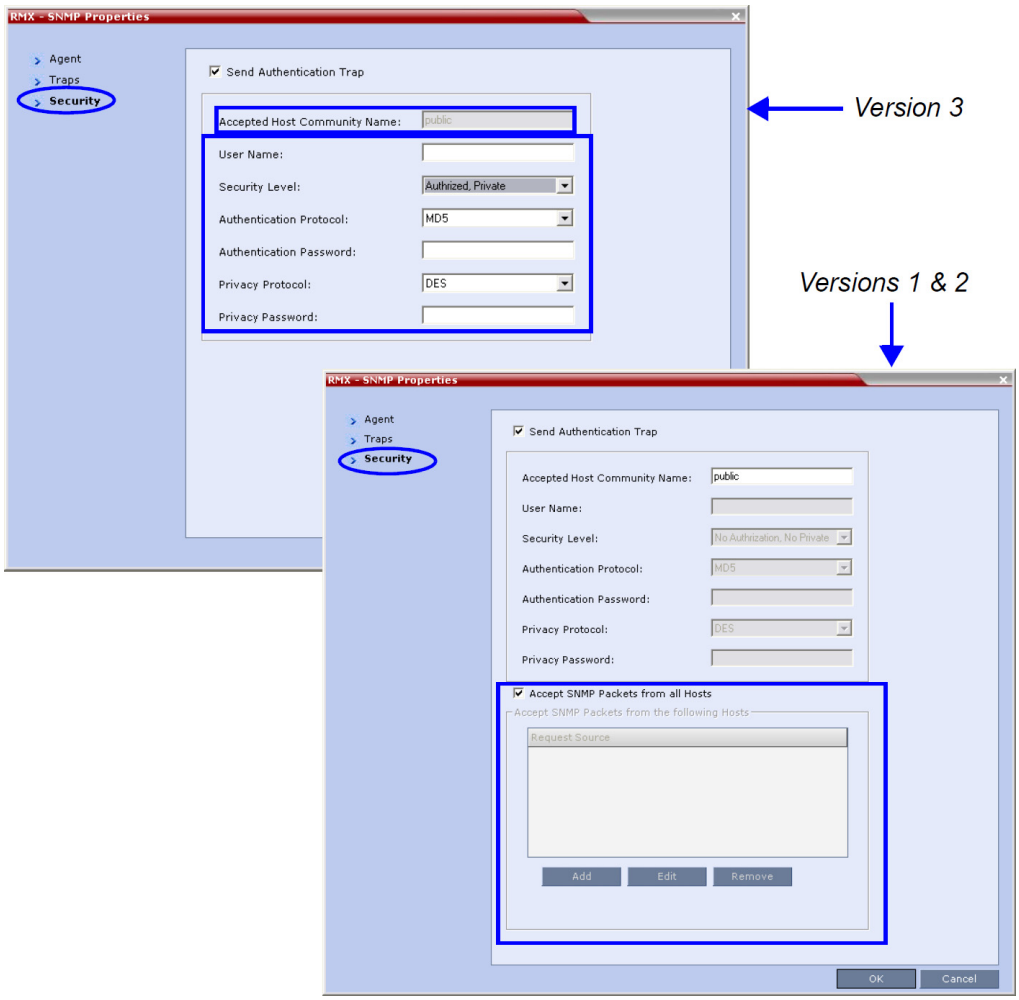
The new IP Address and Community name is added to the Trap Destinations box.

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 **SNMP Properties – Security** dialog box opens.



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.

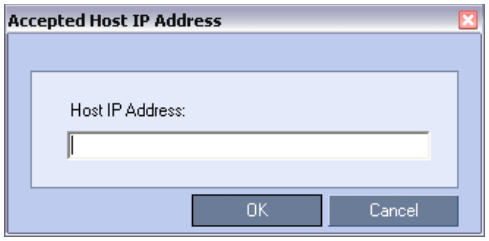
15 Define the following parameters:

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		Version3
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 **SNMP Properties - Security** dialog box, click **OK**.

19

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 the following table.

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

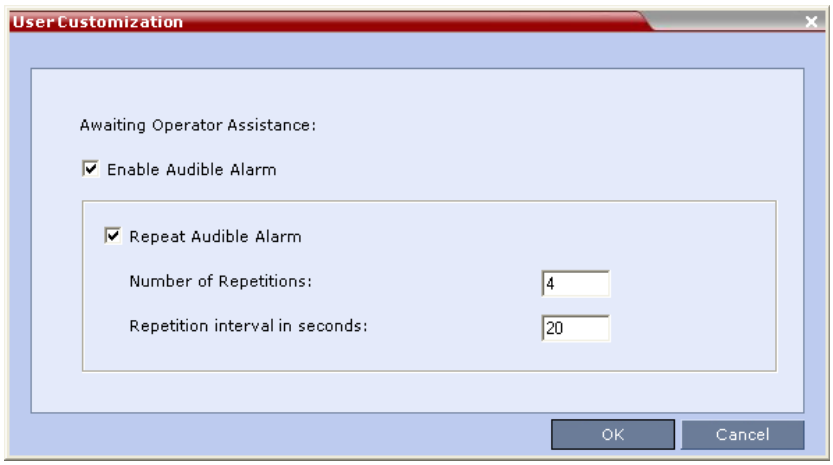
User Customization

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.



2 Define the following parameters:

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.

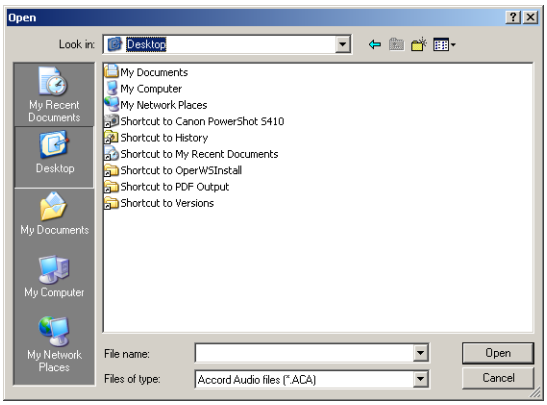
To replace the Audio file on the Collaboration Server Client or RMX Manager:

1 On the Collaboration Server menu, click **Setup** > **Audible Alarms** > **Download Audible Alarm File**.

The **Download Audible Alarm File** window opens.

2 Click the **Browse** button to select the audio file (*.wav) to download.

The *Open* dialog box opens.



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.

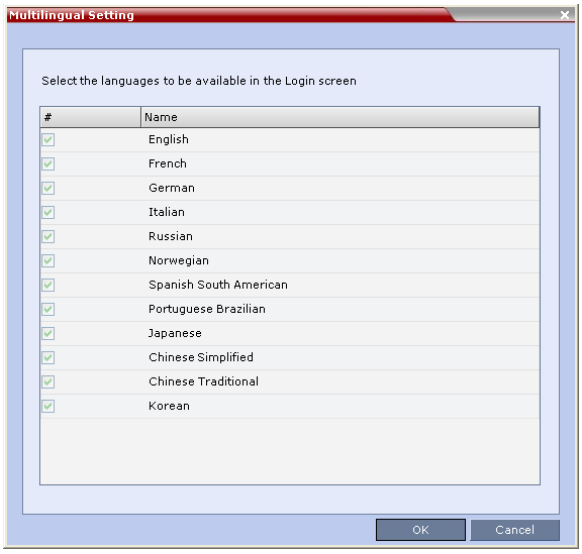
Customizing the Multilingual Setting

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.



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

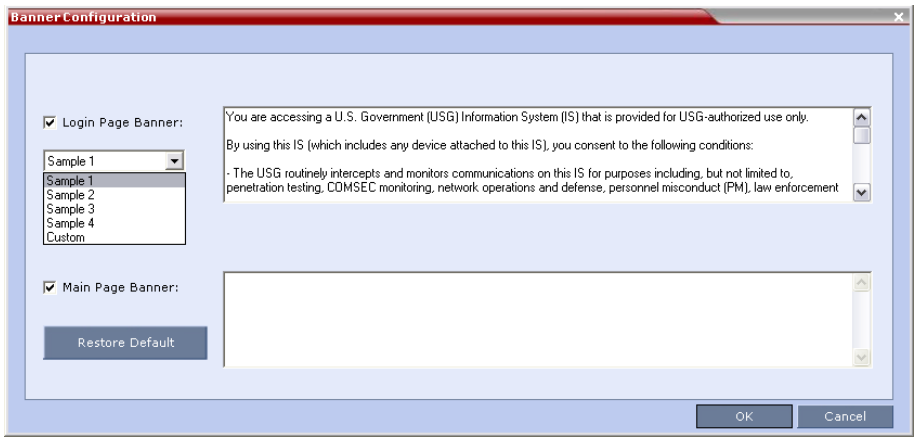
The Login Screen and Main Screen of the Collaboration Server Web Client and the RMX Manager can display informative or warning text banners. These banners can include general information or they can be cautioning users to the terms and conditions under which they may log into and access the system, as 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.



One of 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 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.

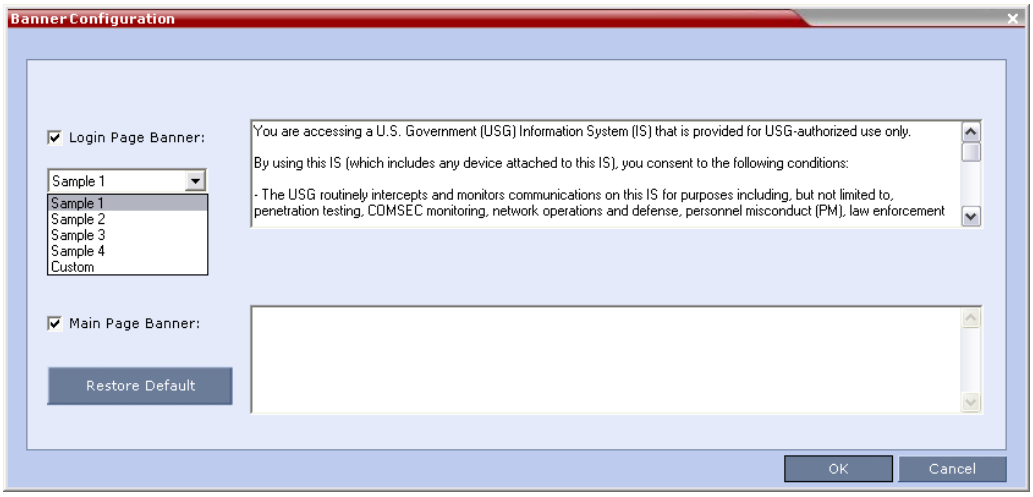
Customizing Banners

The Login and Main Screen banners can be customized to display conference information, assistance information or warning text.

To customize the banners:

1 In the Collaboration Server menu, click **Setup** > **Customize Display Settings > Banners Configuration**.

The **Banners Configuration** dialog box opens.



2 Customize the banners by modifying the following fields:

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.

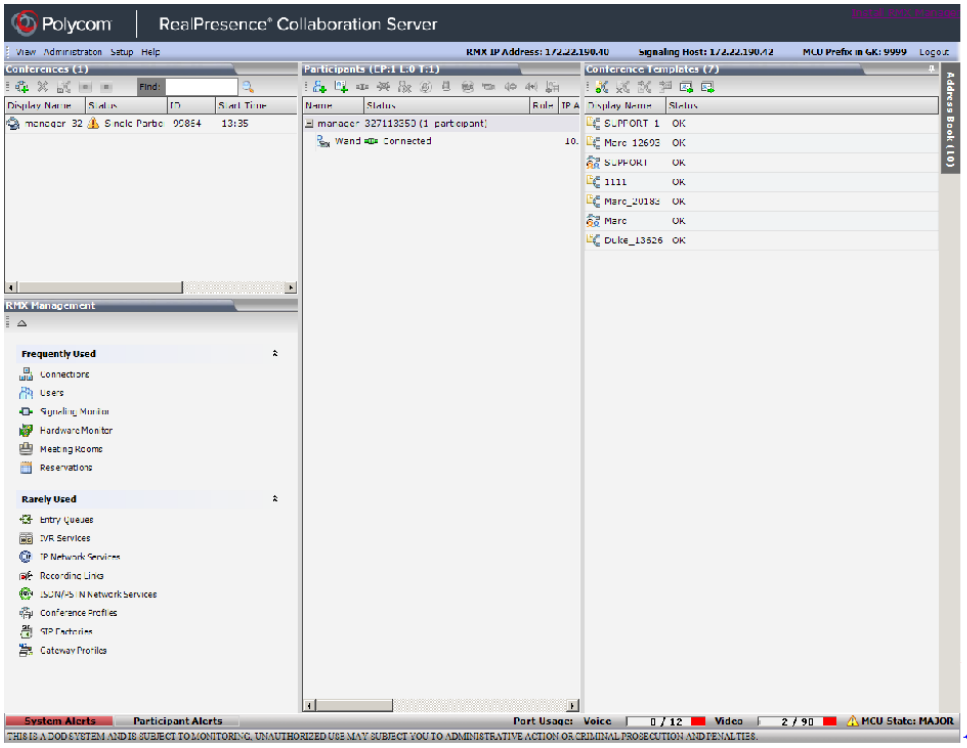
Banner Display

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:

Main Screen Banner

The Main Screen banner is displayed at the bottom of the screen:



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 Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition system to another. Do not use a backup file from the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition on any other model.

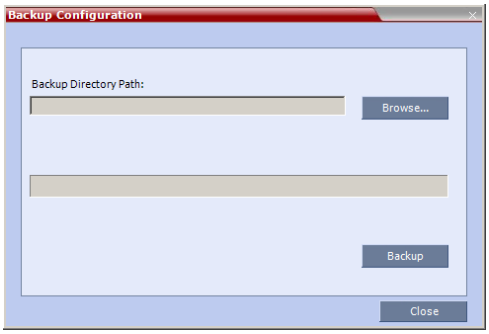
- To ensure file system consistency, do not perform any configuration changes as the system does not suspended them during the backup procedure.
- The following parameters, settings and files are backed up:
  - MCMS configuration files (/mcms/Cfg):
  - Network and service configurations,
  - Rooms,
  - Profiles
  - Reservations
  - 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 the Collaboration Server

The Ping administration tool enables the Collaboration Server Signaling Host to test network connectivity by Pinging IP addresses.

Guidelines

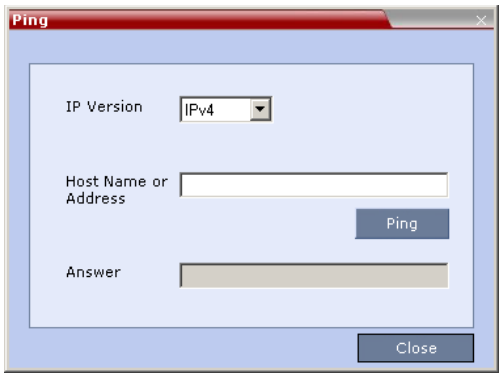
- The IP addressing mode can be either IPv4 or IPv6.
- Both explicit IP addresses and Host Names are supported.
- The Collaboration Server Web Client blocks any attempt to issue another Ping command before the current Ping command has completed. Multiple Ping commands issued simultaneously from multiple Collaboration Server Web Clients 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 fields:

Ping Parameters

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:

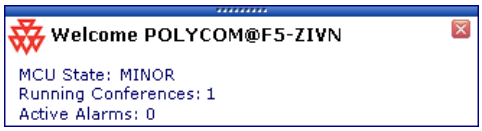
- OK, or
- FAILED

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 alarms list.

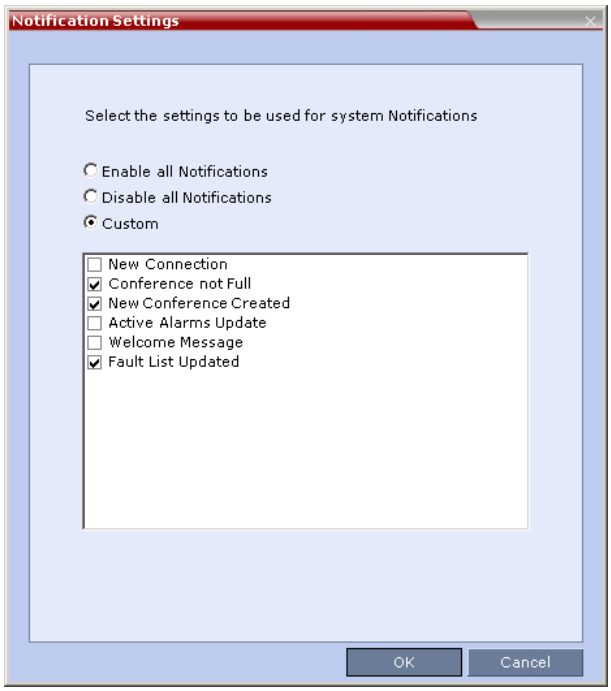
A welcome message is displayed to the Collaboration Server user upon connection.



To configure the notifications:

1 On the Collaboration Server menu, select **Setup** > **Notification Settings**.

The **Notification Settings** dialog box is displayed.



The following notification options are displayed.

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.

Notification Settings Parameters (Continued)

Field	Description
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

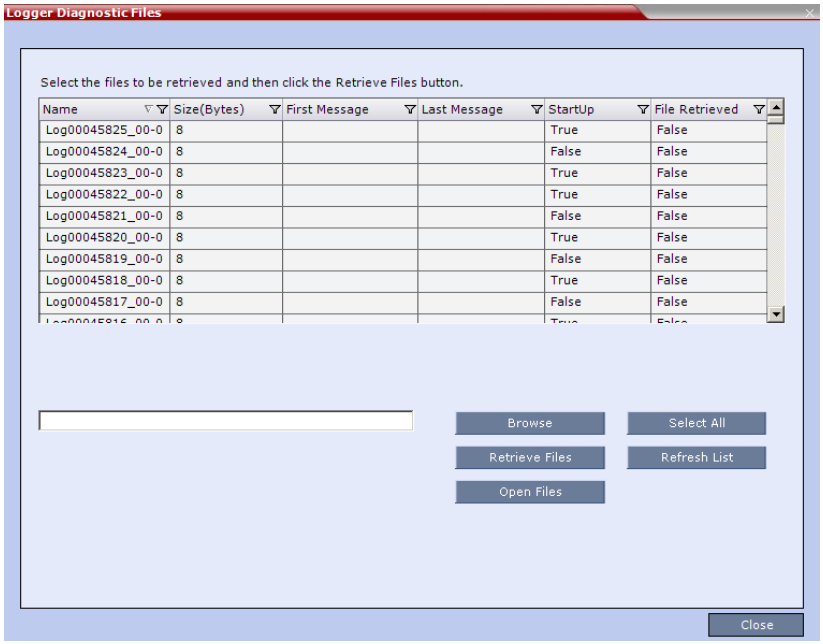
The Logger utility is a troubleshooting tool that continually records MCU system messages and saves them to files in the MCU hard drive. For each time interval defined in the system, a different data file is created. The files may be retrieved from the hard drive for off-line analysis and debugging purposes.

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**.



The following tasks can be performed:

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:

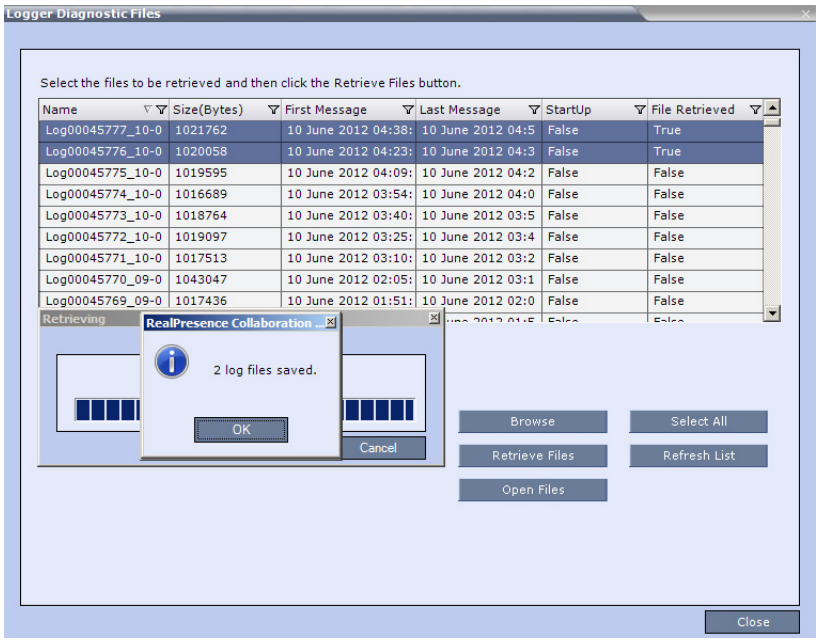
```
Log_SN0000000002_FMD06032007_FMT083933_LMD06032007_LMT084356_SZ184951_SUY.log.
```

To Retrieve 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.



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

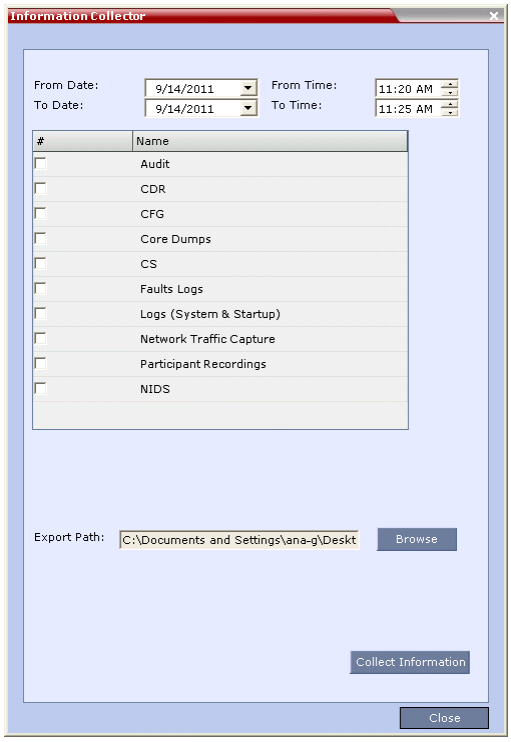
When the Information Collector is used the following steps are performed:

- **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.



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 Audit Events.

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

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.

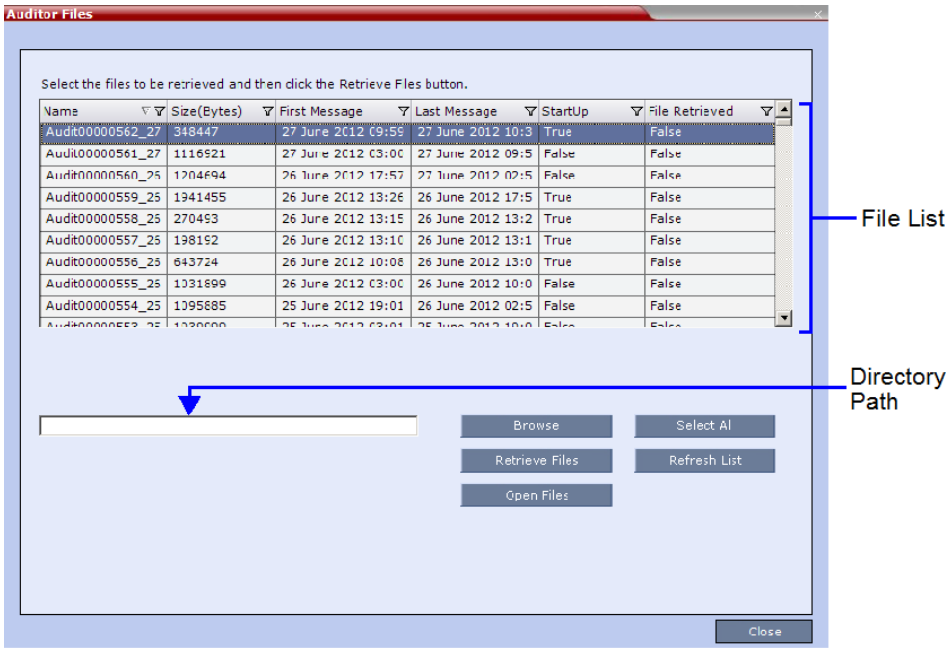
Retrieving Auditor Files

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.



  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.

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.

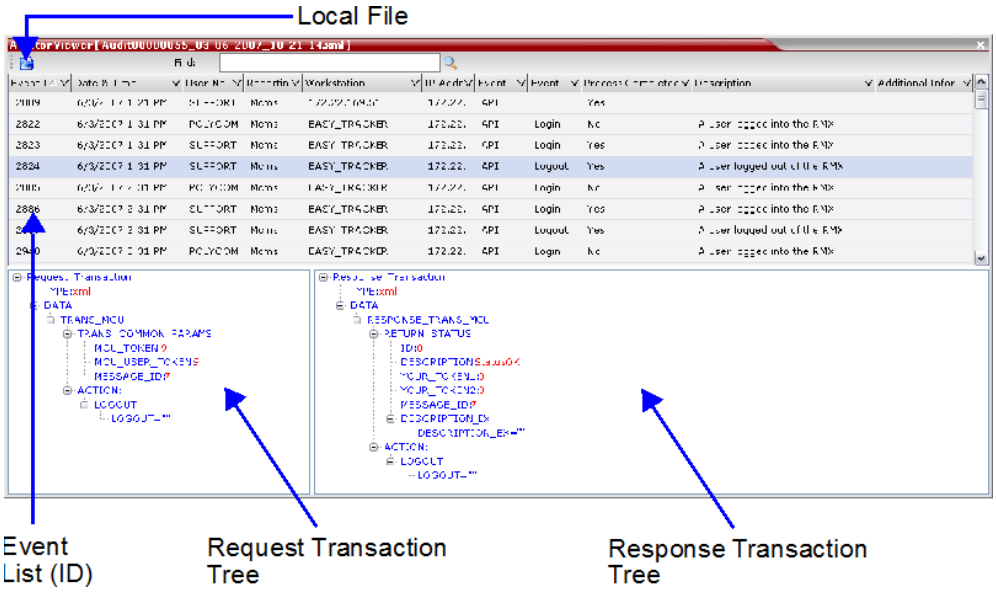
You can view an Auditor Event File directly from the Auditor Files list or by opening the file from the Auditor File Viewer.

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.



The following fields are displayed for each event:

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 • Video • MUX
Workstation	The name (alias) of the workstation used to send the request that triggered the event.

Auditor Event Columns (Continued)

Field	Description
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)
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

Alerts and Faults that are recorded by the Auditor.

Alerts and Faults recorded by the Auditor

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.
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.

Alerts and Faults recorded by the Auditor (Continued)

Event
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).
USB key used to change system configuration.
User closed the browser
User initiated MCU reset.

Transactions

Transactions that are recorded by the Auditor.

Transactions recorded by the Auditor

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

Transactions recorded by the Auditor (Continued)

Transaction
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
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.

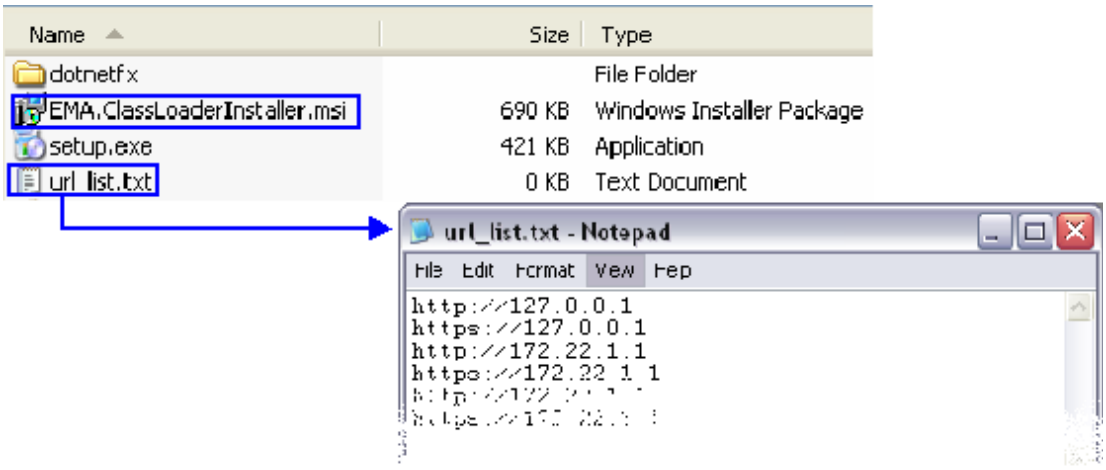
Installing ActiveX

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.



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.

Resetting the Collaboration Server 800s

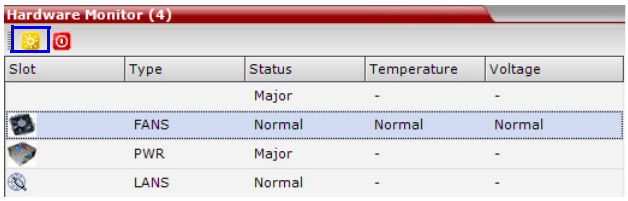
These instructions are applicable to the RealPresence Collaboration Server 800s only.

System Reset saves system configuration changes and restarts the system with the latest settings.

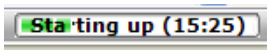
To reset the RMX:

1 In the **RMX Management** pane, click the **Hardware Monitor button**.
The **Hardware Monitor** pane is displayed.

2 Click the **Reset** ( ) button.



When the Collaboration Server system is reset, during Collaboration Server startup the Progress Bar appears at the bottom of the Collaboration Server *Status* pane, displaying the amount of time remaining for the reset process to complete:



The Startup progress is also indicated by a green bar moving from left to right.

The duration of the *Startup* depends on the type of activity that preceded the MCU reset. For example: Fast Configuration Wizard, New Version installation, Version Upgrade, Restore Last Configuration etc.

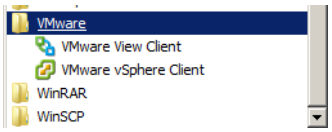
Resetting the RealPresence Collaboration Server Virtual Edition

To restart the MCU instance:

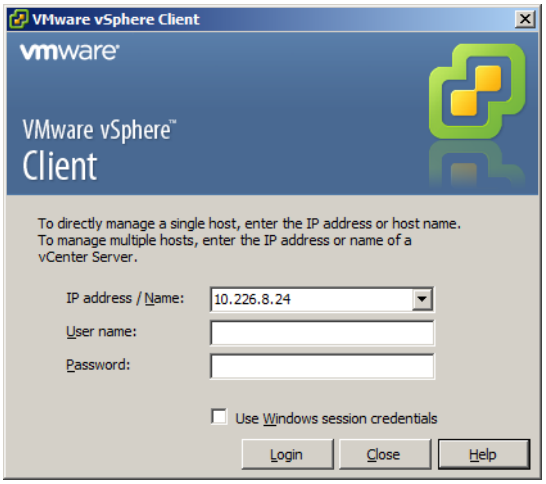
1 Click **Start > Programs.**

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.



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.

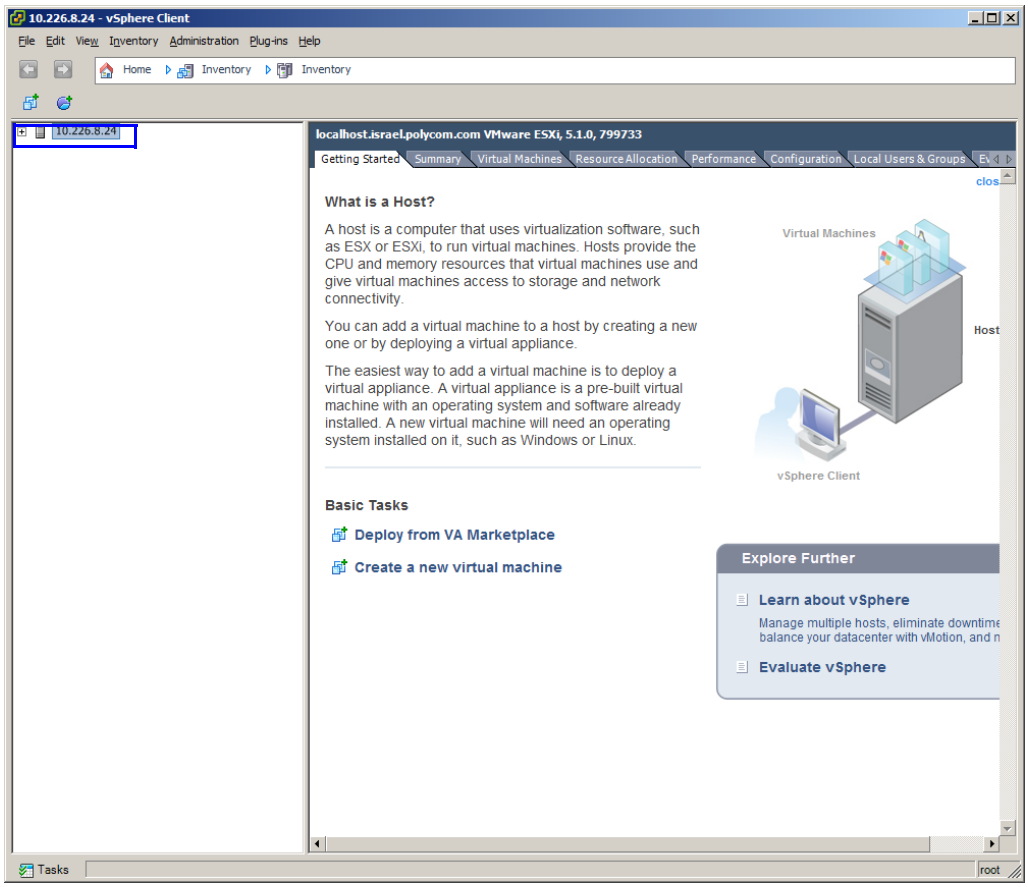
In the *Password* field, enter the password as defined for the user name with which you will log in to the **vSphere** host.

or

b Click the **Use Windows session credentials** check box.

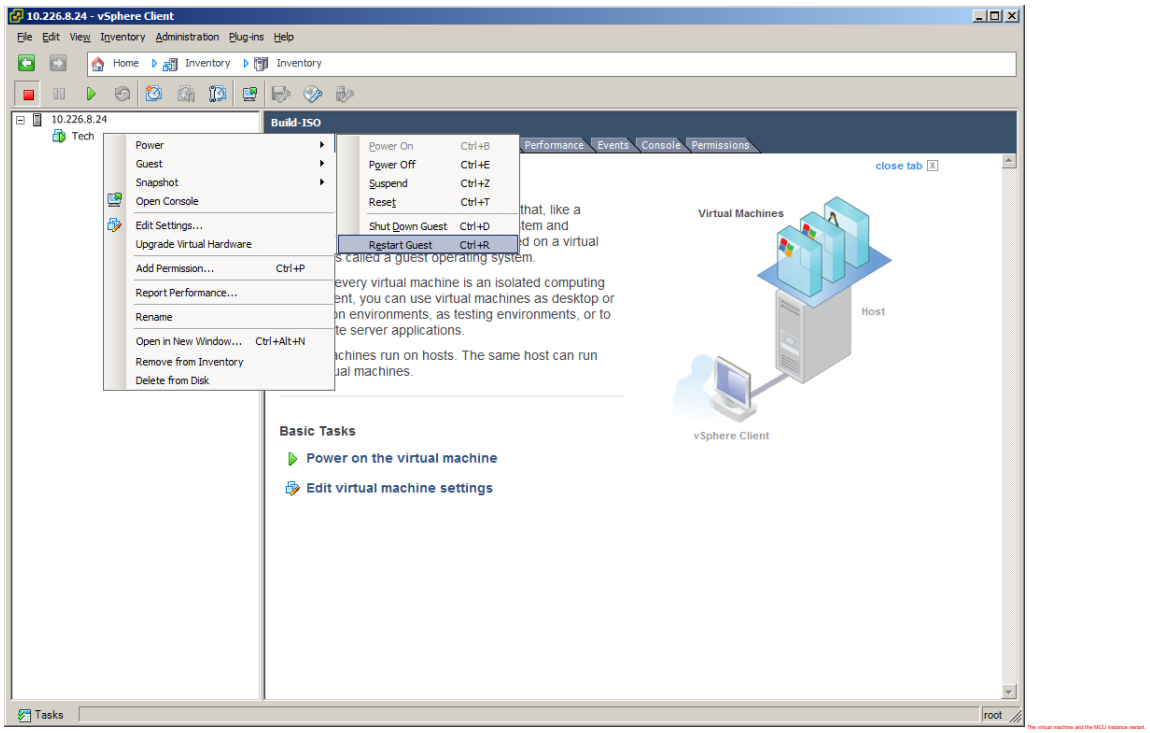
4 Click **Login.**

The **VMware vSphere Client** opens.



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.

Upgrading and Downgrading

This procedure allows an administrator to update the MCU instance without requiring the administrator to reregister the product.

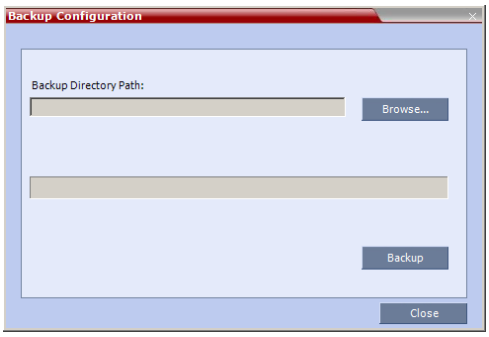
Updating the MCU instance requires the previously used activation key. If you no longer have the activation key, contact support before starting this procedure.

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.

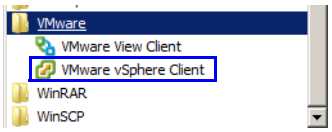
3 Select the *Backup Directory Path* and then click **Backup**.

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.

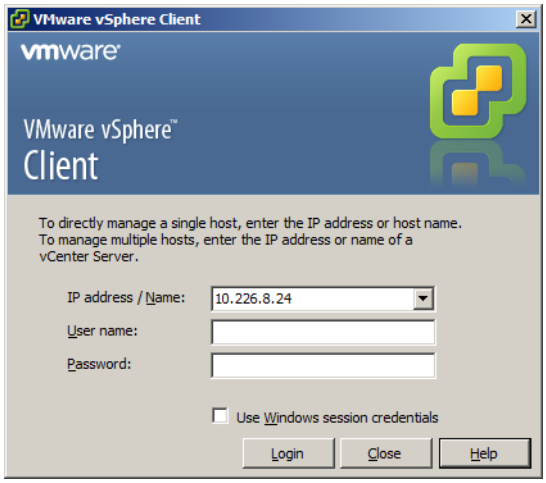
4 On the Windows taskbar, click the **Start > Programs.**

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.

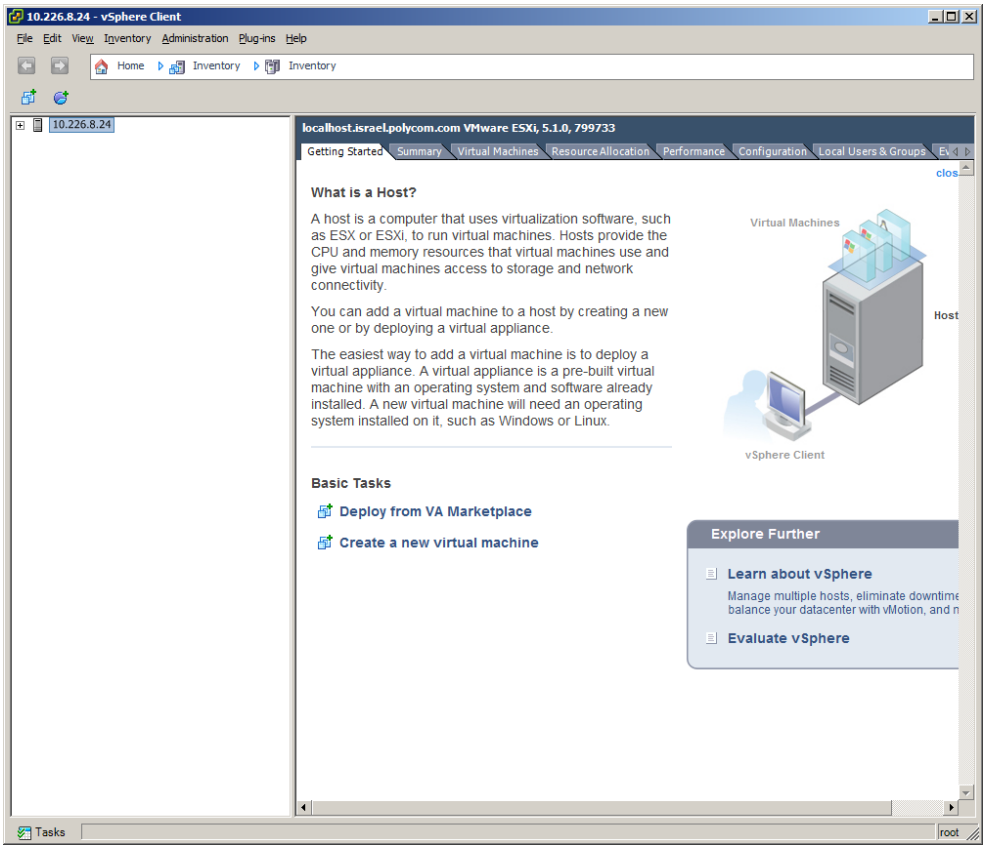


5 In the *IP address / Name* field, enter the IP Address or the name of the **vSphere** host.

6 Either type your **vSphere** *User Name* and *Password* or select **Use Windows sessions credentials**.

7 Click **Login.**

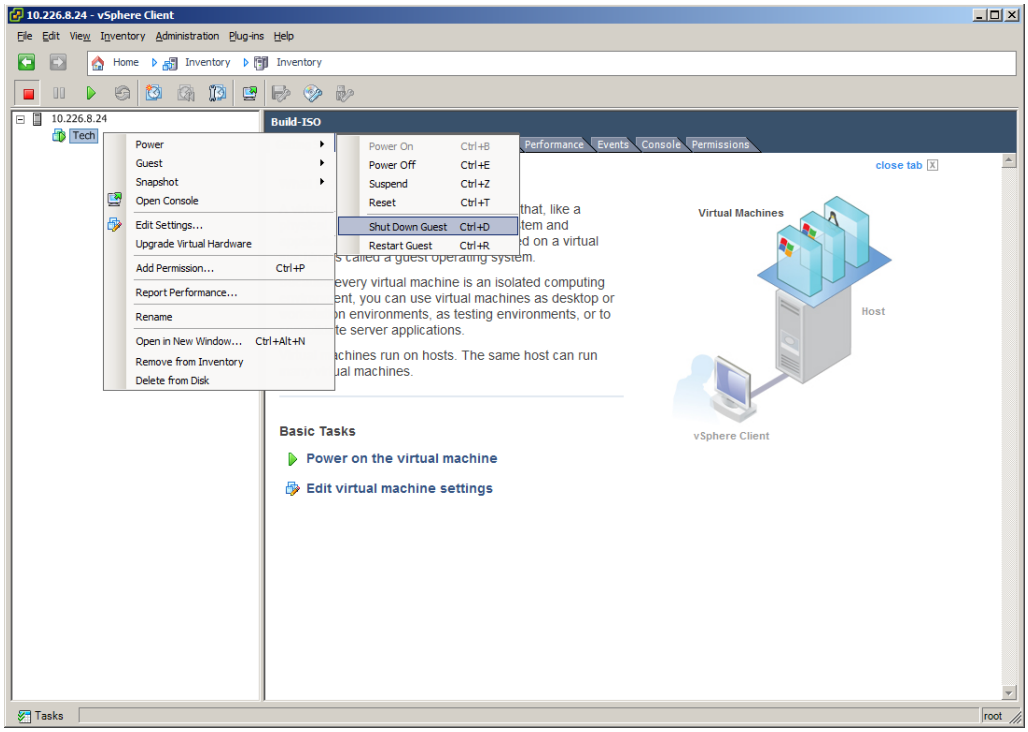
The *VMware vSphere Client* is displayed.



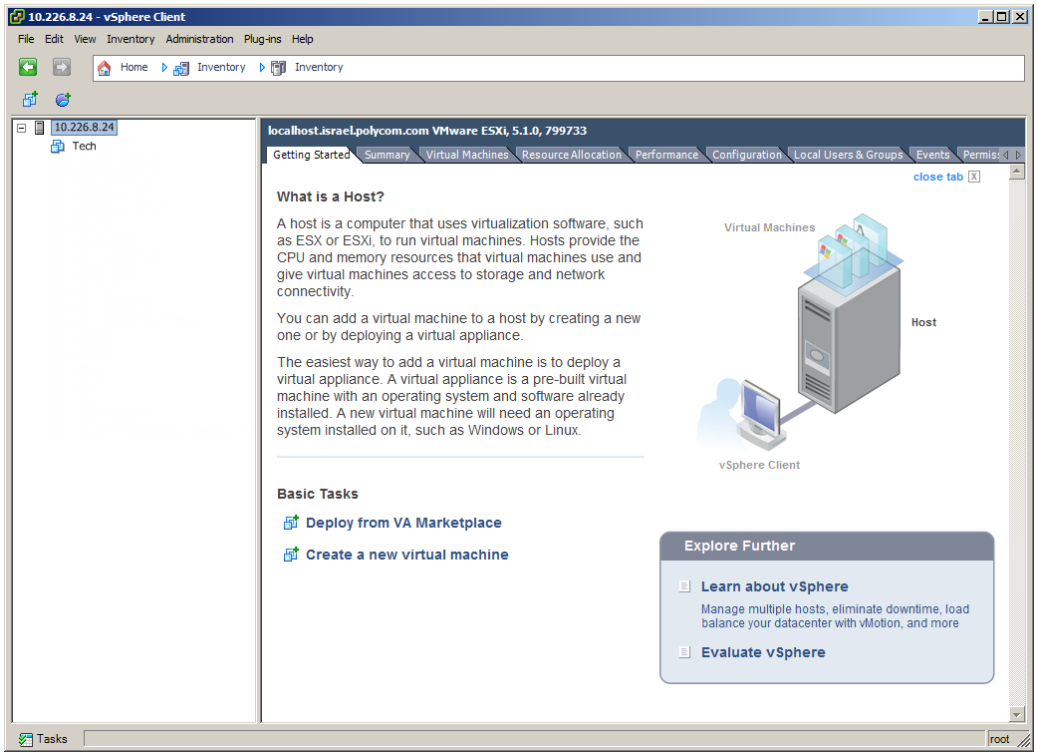
8 In the *Inventory Panel*, select the *Datastore* that houses the MCU.

The inventory of the Datastore appears.

9 Right-click the MCU virtual machine, then click **Power > Shut Down Guest**.

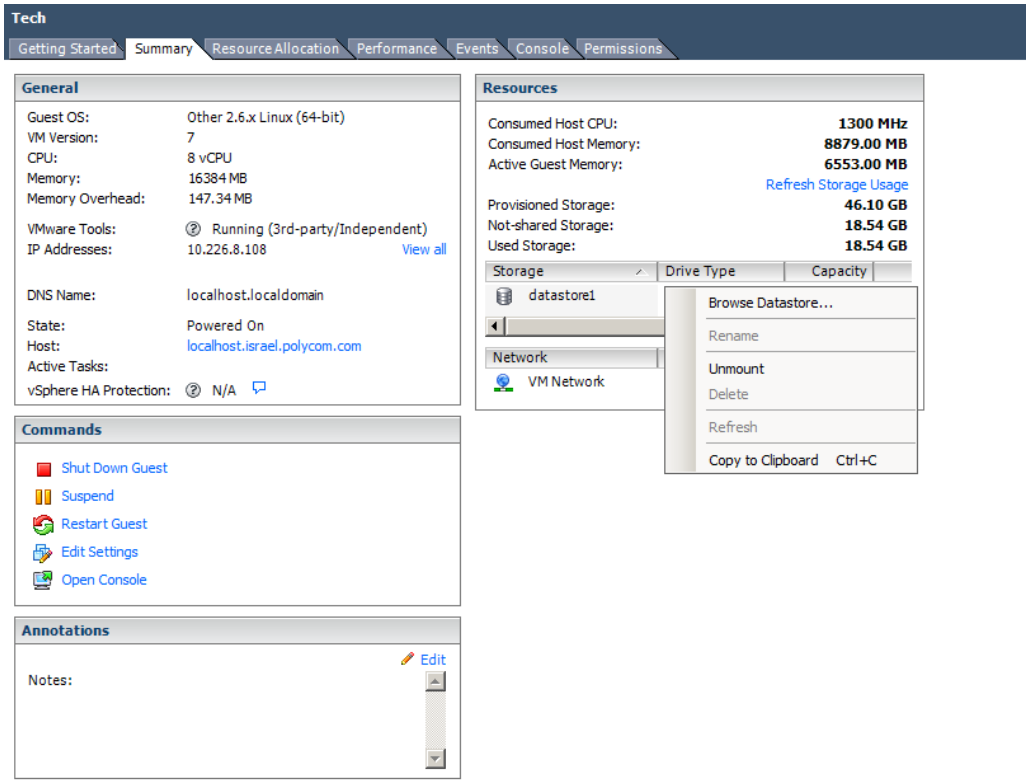


10 When VM turns blue, the virtual machine has shut down.

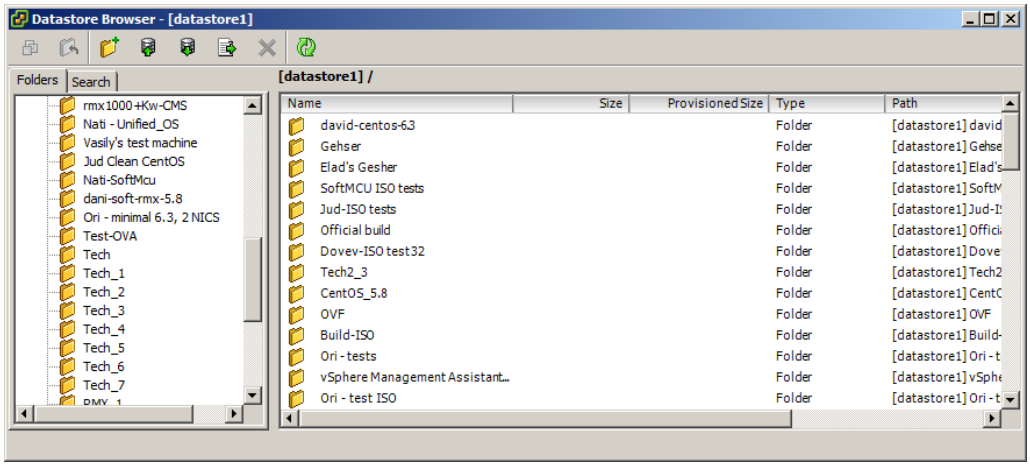


11 When the MCU has shut down, click the **Summary** tab.

12 Under *Resources*, right-click the datastore, and click **Browse Datastore**.



The *Browse Datastore* window appears.

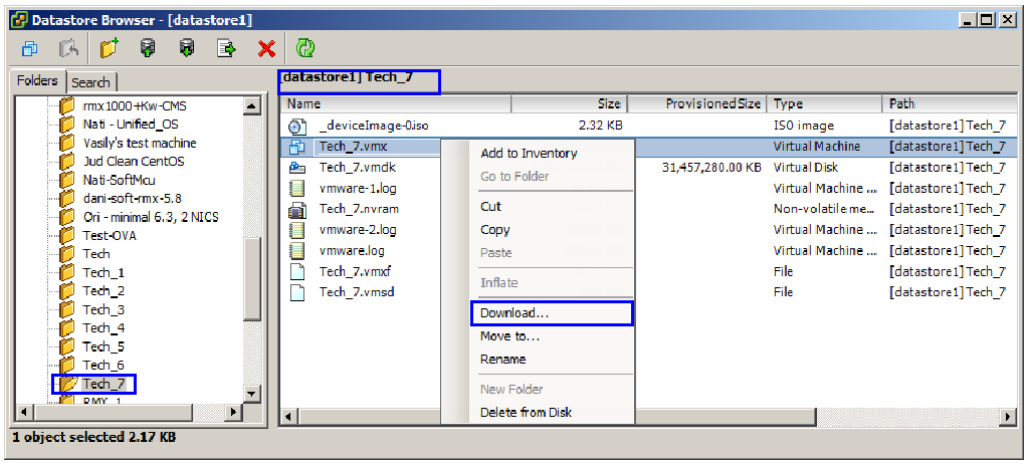


13 In the **Folders** tab, select the folder whose name matches that of the MCU.

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.

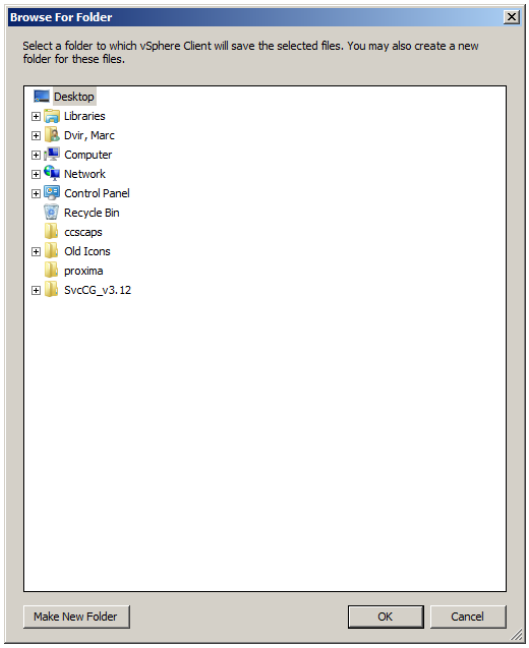
The contents of the folder are displayed.

14 Right-click the file ending with ".vmx" and click **Download.**



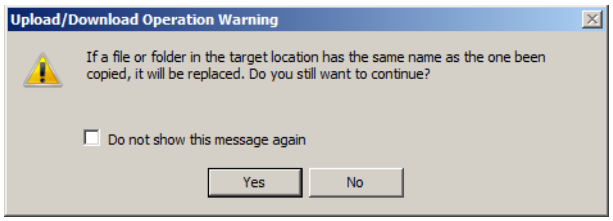
The *Browse For Folder* window appears.

15 Browse to a location and click **OK**.



The *Upload/Download Operation Warning* window may appear.

16 If the *Upload/Download Operation Warning* window appears, click **Yes**.



If it does not appear, proceed to step 17.

The file downloads.

17 Open the file in any plain text editor.

```
43  ethernet0.networkName = "VM Network"
44  ethernet0.addressType = "generated"
45  guestOS = "centos-64"
46  uuid.location = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
47  uuid.bios = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
48  vc.uuid = "52 b7 2d 0c 17 a8 af 14-ec 8b 97 49 bd ec 49 6a"
49  hpet0.present = "TRUE"
50  usb.vbluetooth.startConnected = "TRUE"
51  scsi0.pciSlotNumber = "16"
52  ethernet0.generatedAddress = "00:0c:29:59:3c:e9"
```

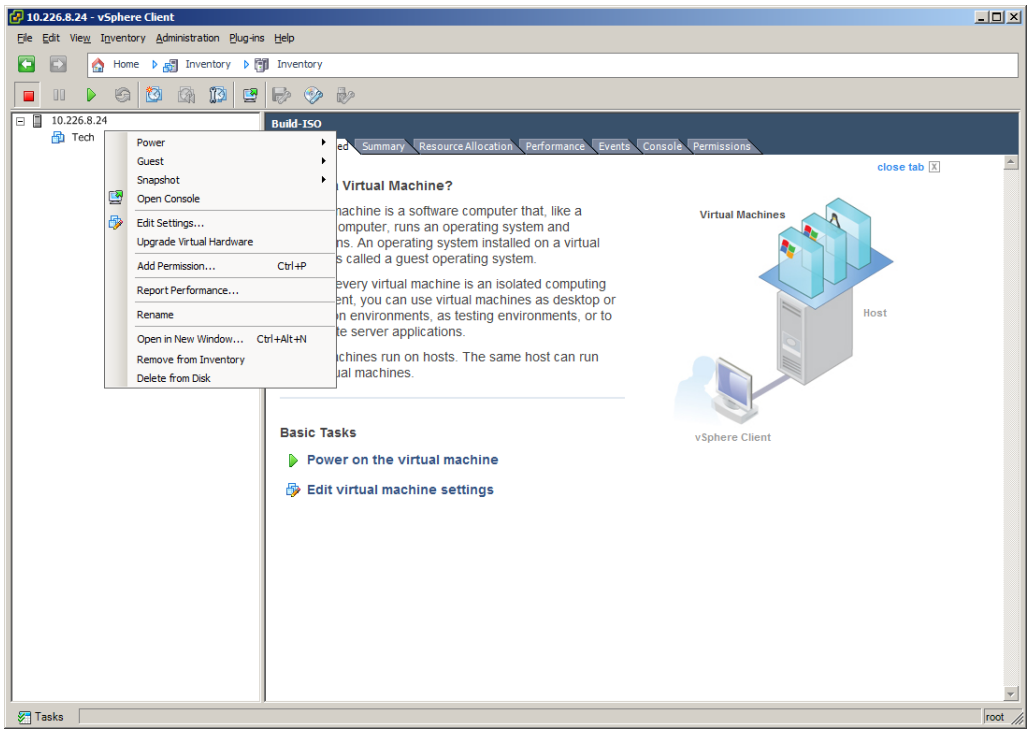
18 Locate the line that starts with *uuid.bios.*

19 **Copy** the entire line and paste it into another text file.

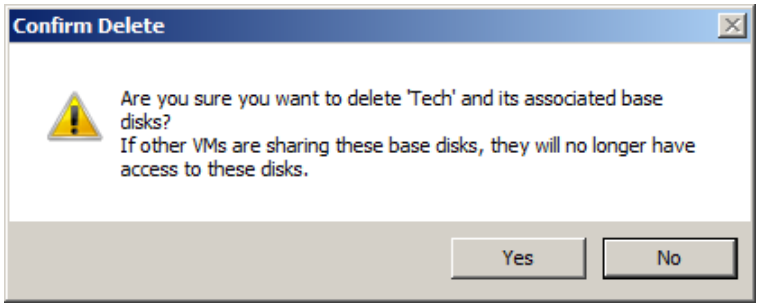
20 **Save** the text file.

21 In the *Inventory Panel*, click the Datastore that houses the MCU.

22 Right-click the MCU, and select **Delete from Disk.**

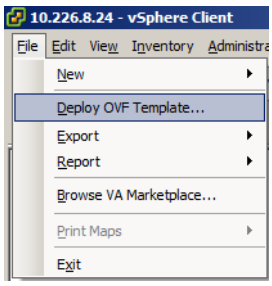


The **Confirm Delete** window appears.



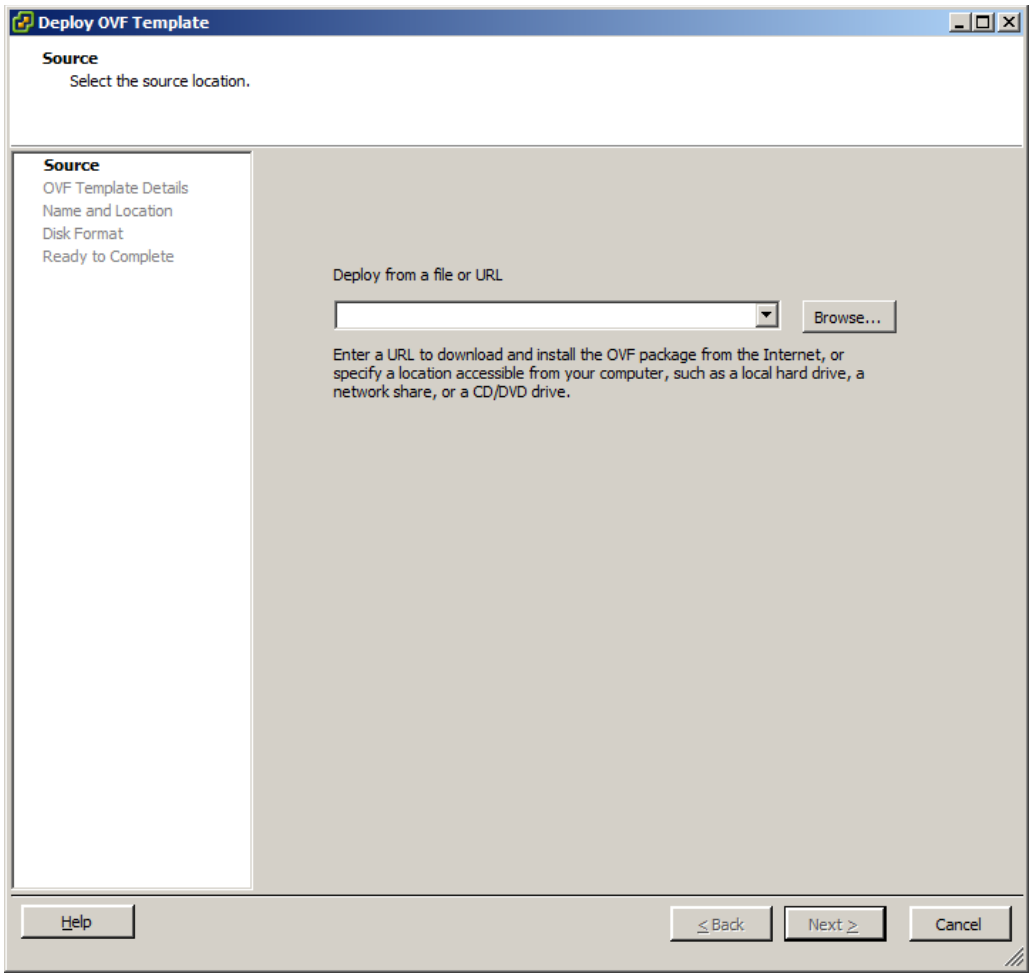
23 Click **Yes**.

24 On the *vSphere Client* menu, select **File > Deploy OVF Template.**



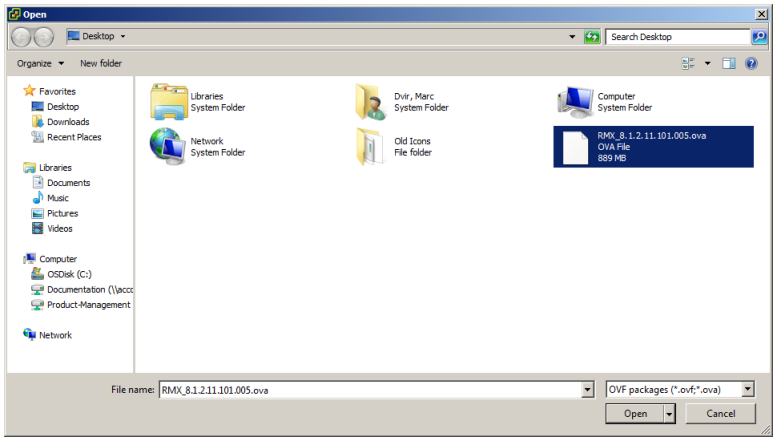
The *Deploy OVF Template* wizard opens to the *Source* page.

25 Click **Browse.**

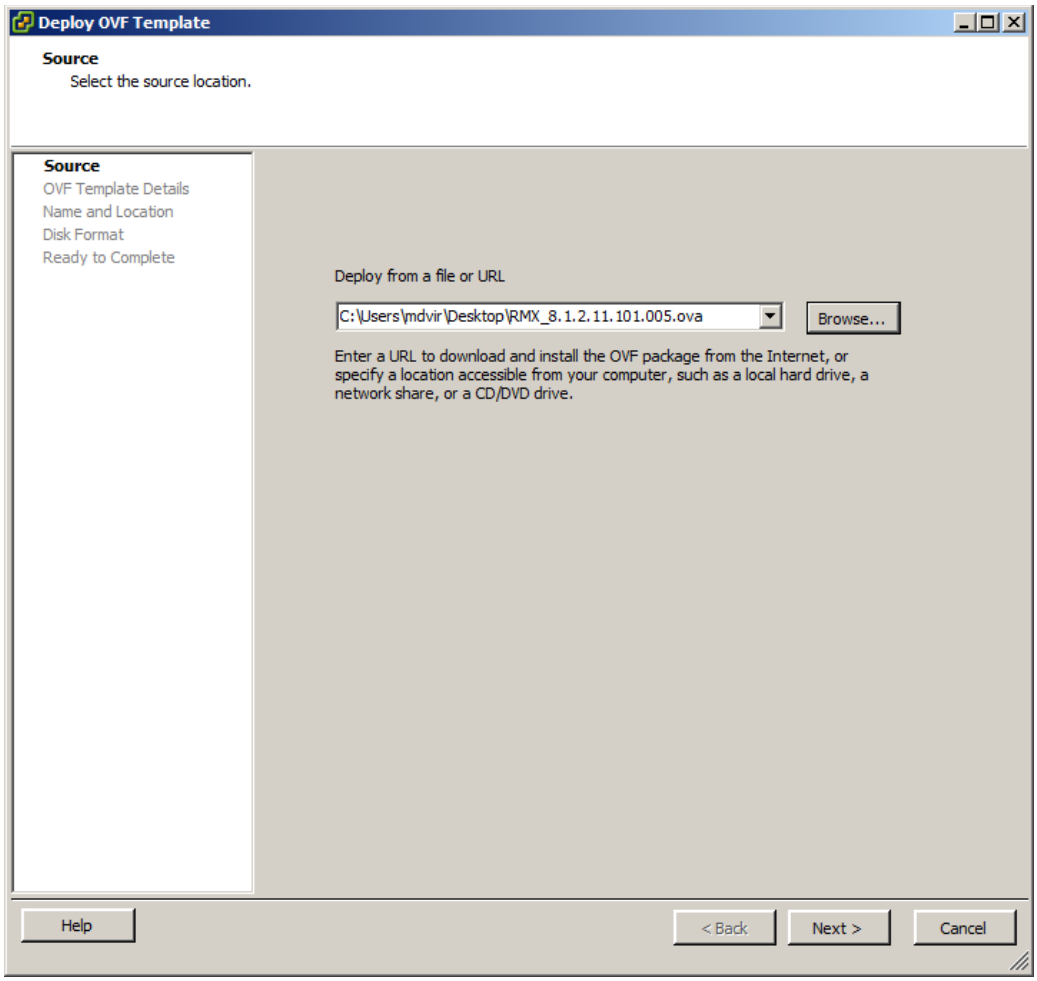


The **Open** dialog box appears.

26 Browse to the new OVA file.

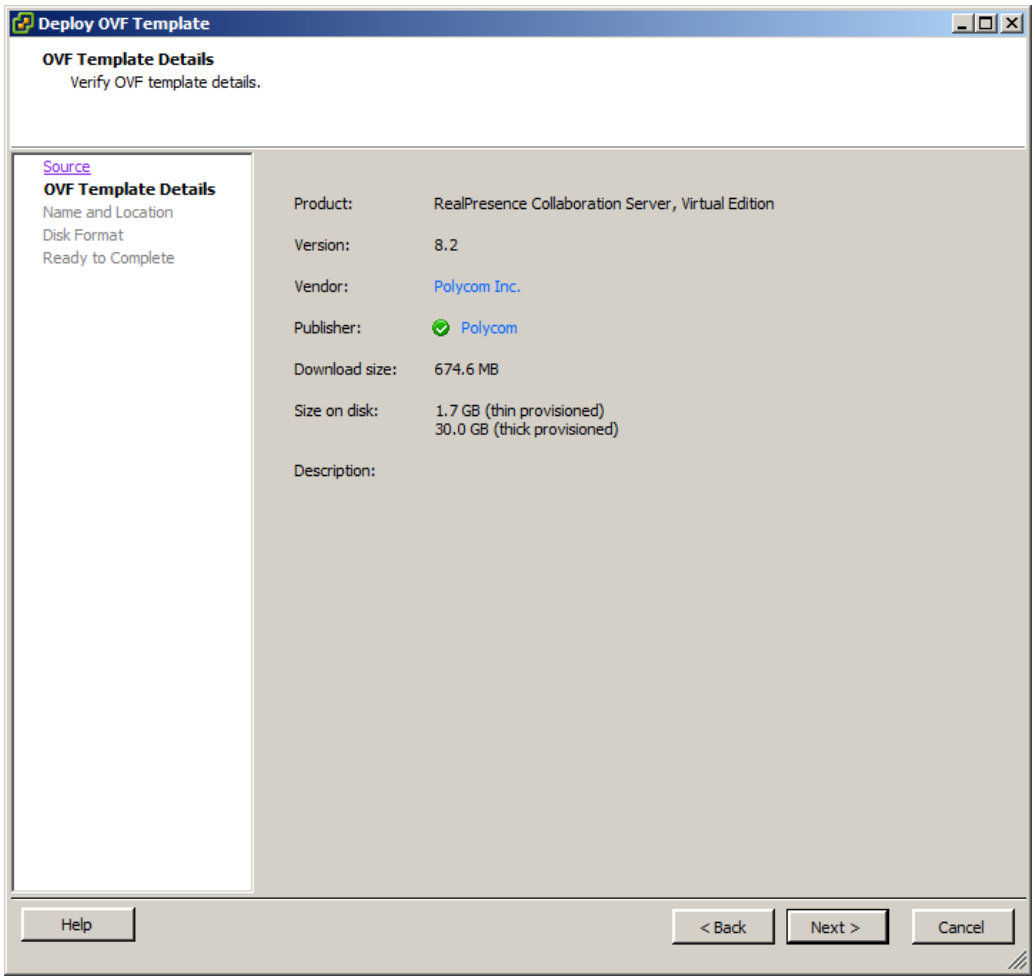


27 Either double-click on the OVA file or click on the file, then click **Open.**



28 Click **Next**.

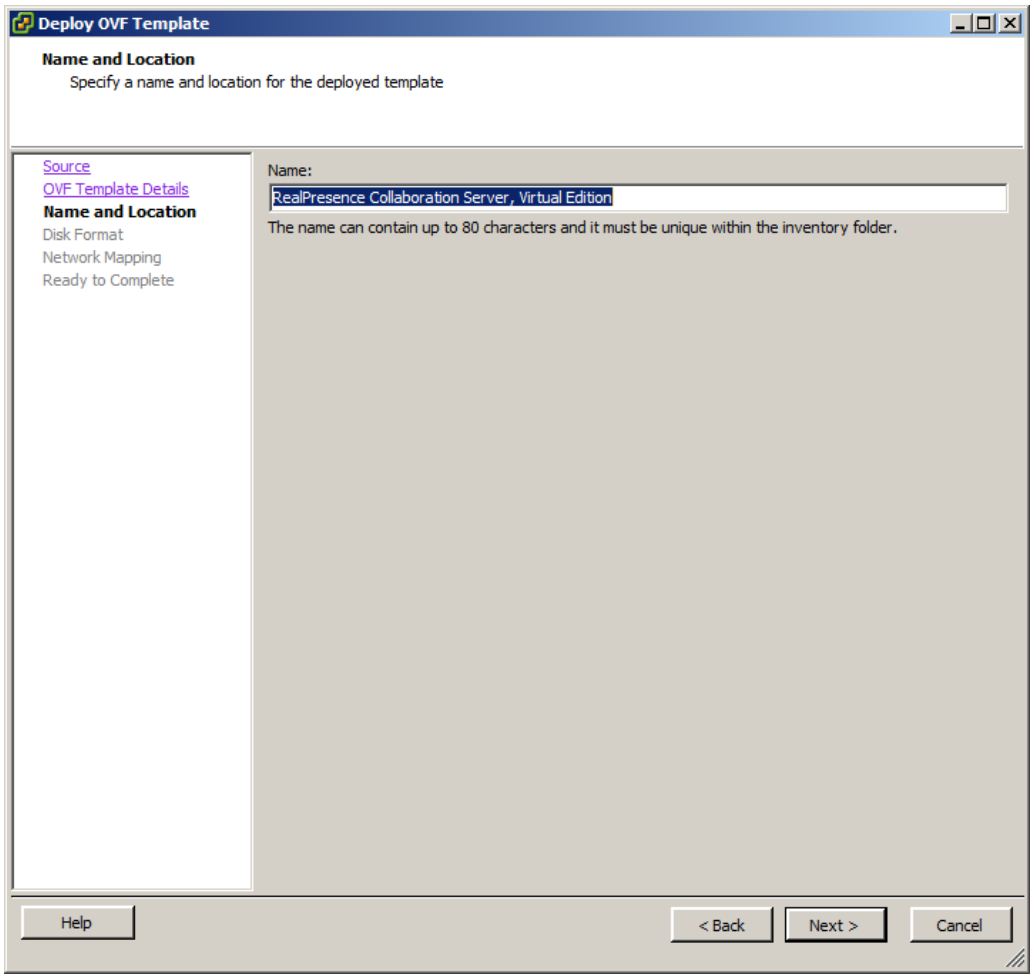
The *OVF Template Details* page is displayed.



29 Click **Next**.

The *Name and Location* page is displayed.

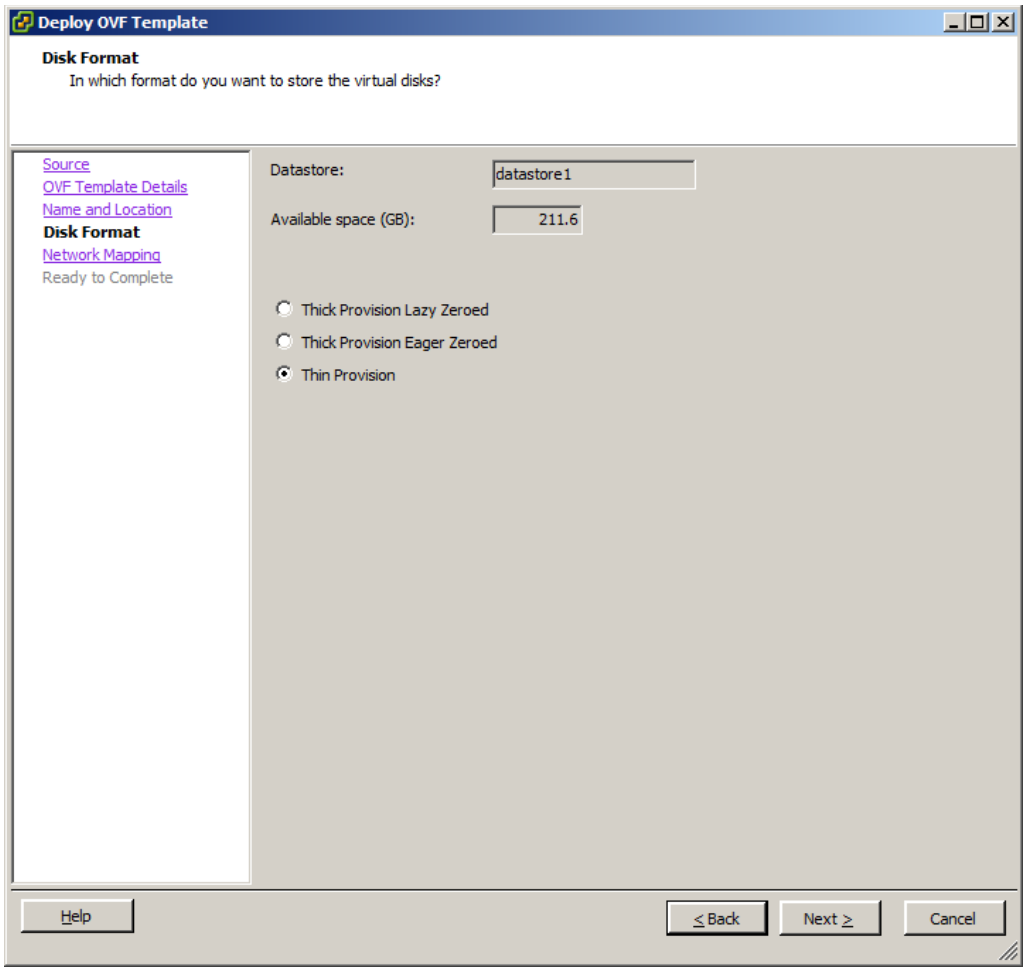
30 In the *Name* field, type the same name previously used for the MCU.



31 Click **Next.**

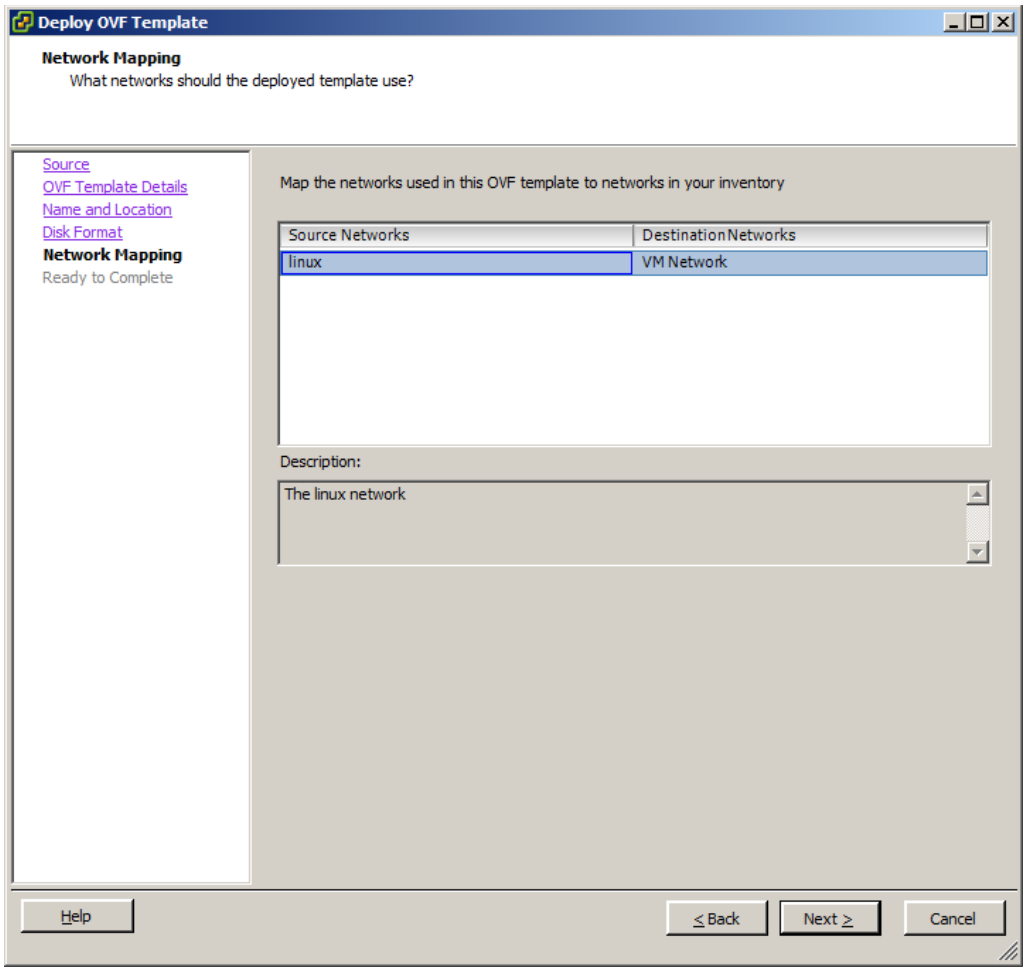
The **Disk Format** page is displayed.

32 Select **Thin Provision**, then click **Next**.

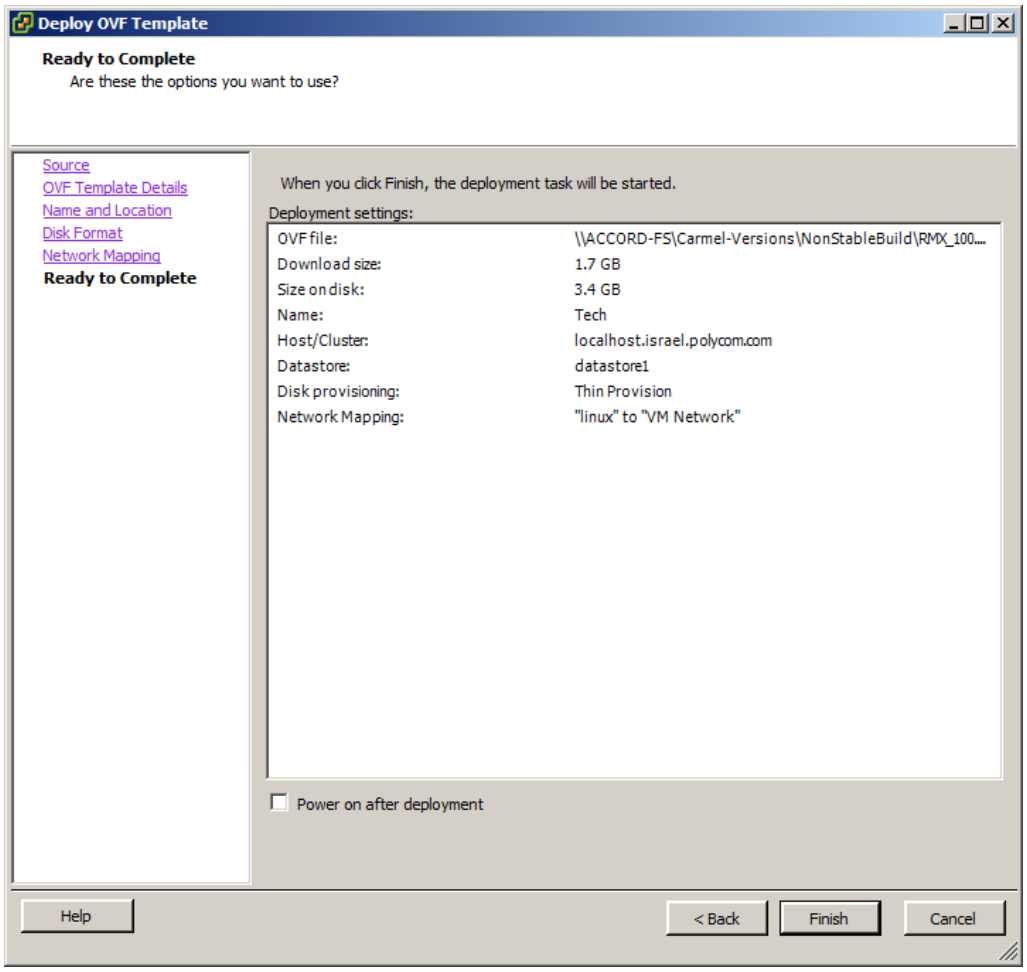


The *Network Mapping* page is displayed.

33 Select the appropriate network mappings, then click **Next.**



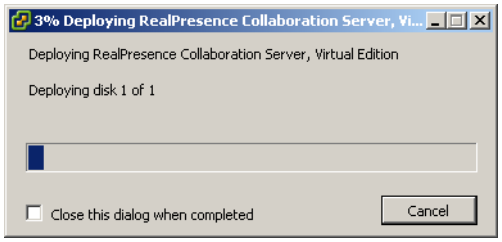
The *Ready to Complete* page is displayed.



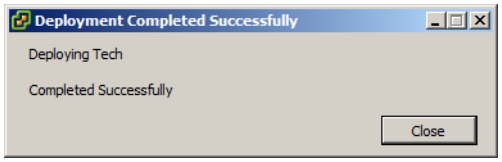
34 Verify that **Power on after deployment** is not selected.

35 Confirm that all the settings are correct, then click **Finish.**

The *vSphere Client* deploys the OVF file.



When the deployment is complete the following window appears:

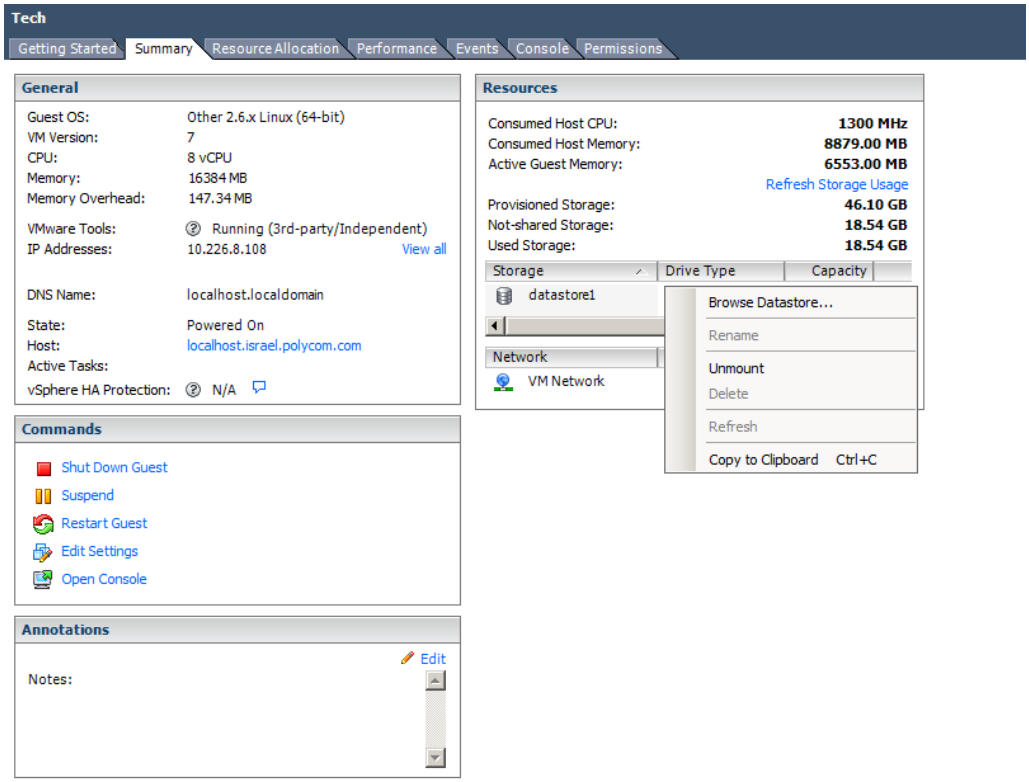


36 Click **Close.**

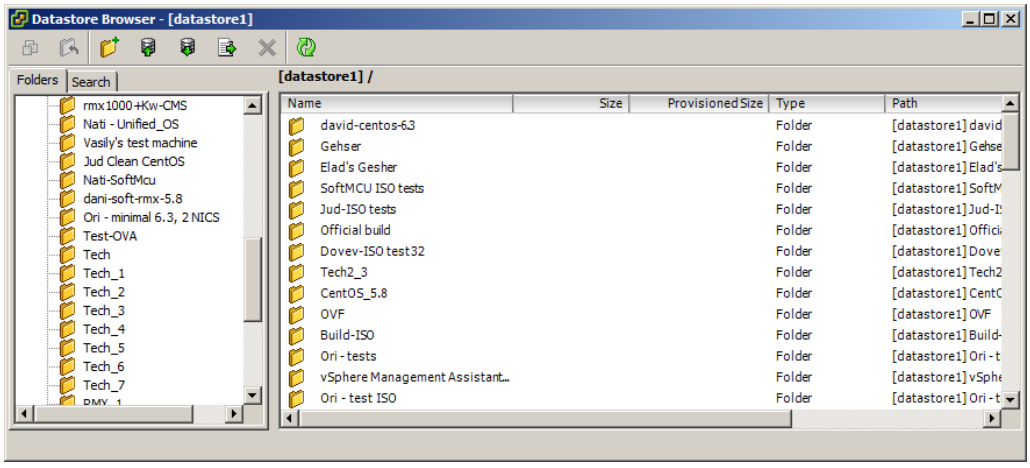
37 In the *Inventory Panel*, select the Datastore that used to house the MCU.

38 Click the **Summary** tab.

39 Under *Resources*, right-click the datastore, and click **Browse Datastore**.



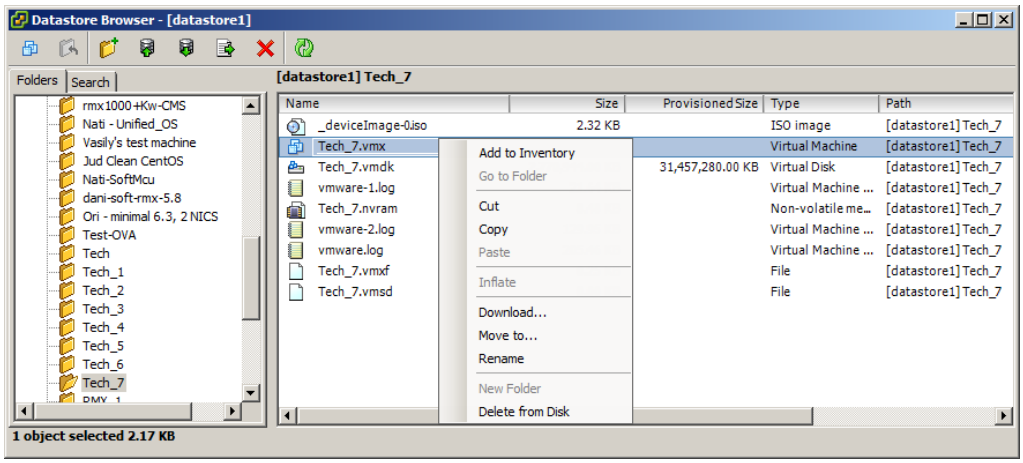
The *Browse Datastore* window appears.



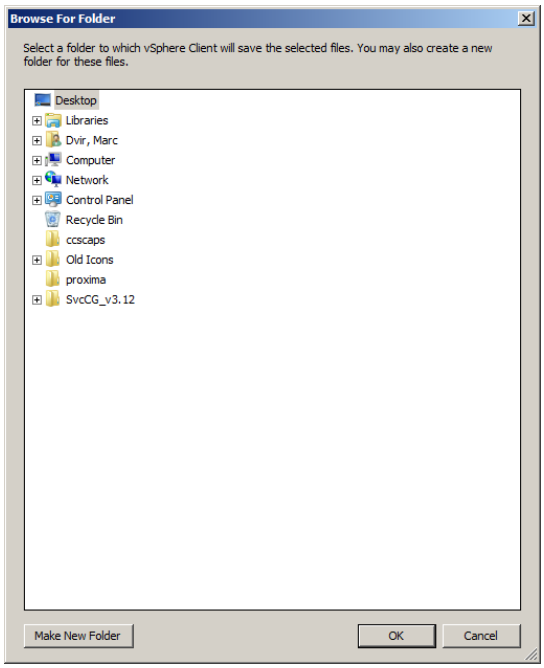
40 In the **Folders** tab, select the folder whose name matches that of the MCU. The folder name will have an underscore and a number at the end.

The contents of the folder are displayed.

41 Right-click the file ending with ".vmx" and click **Download.**

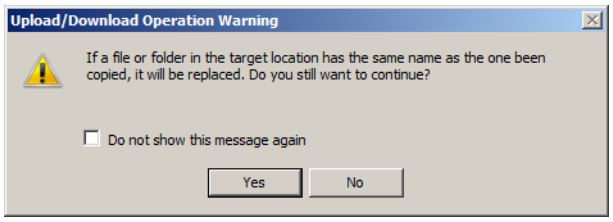


The *Browse For Folder* window appears.



42 Browse to a location and click **OK**.

The *Upload/Download Operation Warning* window may appear.



43 If the *Upload/Download Operation Warning* window appears, click **Yes**. If it does not appear, proceed to step 44.

The file downloads.

44 Open the file created in step 19.

45 Open the file in any plain text editor.

```
43  ethernet0.networkName = "VM Network"
44  ethernet0.addressType = "generated"
45  guestOS = "centos-64"
46  uuid.location = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
47  uuid.bios = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
48  vc.uuid = "52 b7 2d 0c 17 a8 af 14-ec 8b 97 49 bd ec 49 6a"
49  hpet0.present = "TRUE"
50  usb.vbluetooth.startConnected = "TRUE"
51  scsi0.pciSlotNumber = "16"
52  ethernet0.generatedAddress = "00:0c:29:59:3c:e9"
```

46 Locate the line that starts with, *uuid.bios.*

47 Replace that line with the line saved in the other text file.

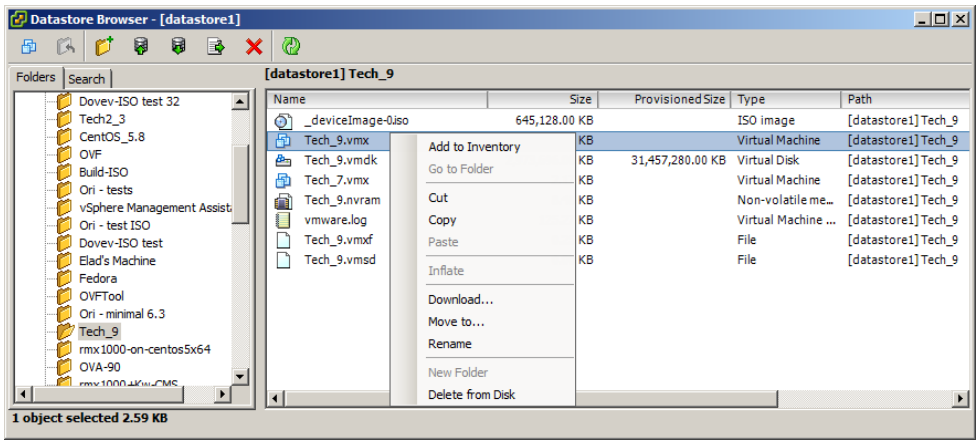
48 Add the following as a separate line to the file, including the quotation marks:
uuid.action = “keep”

```
43  ethernet0.networkName = "VM Network"
44  ethernet0.addressType = "generated"
45  guestOS = "centos-64"
46  uuid.location = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
47  uuid.bios = "56 4d c9 6c 5d 6d 52 b8-71 32 cb d4 6f 59 3c e9"
48  uuid.action = "keep"
49  vc.uuid = "52 b7 2d 0c 17 a8 af 14-ec 8b 97 49 bd ec 49 6a"
```

49 Save and close the file.

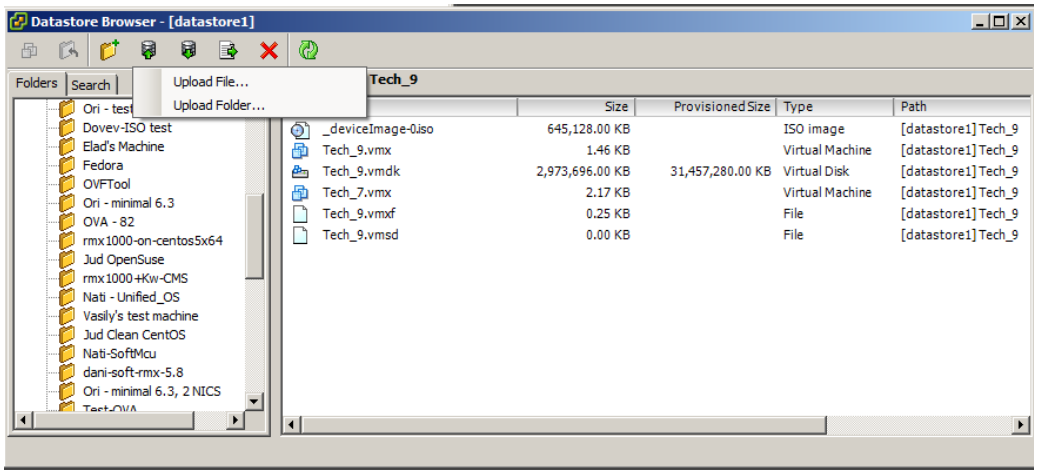
50 **Optional**. To back up the previous configuration:

a In the *Datastore Browser* window, right-click the .vmx file, then select **Rename**.



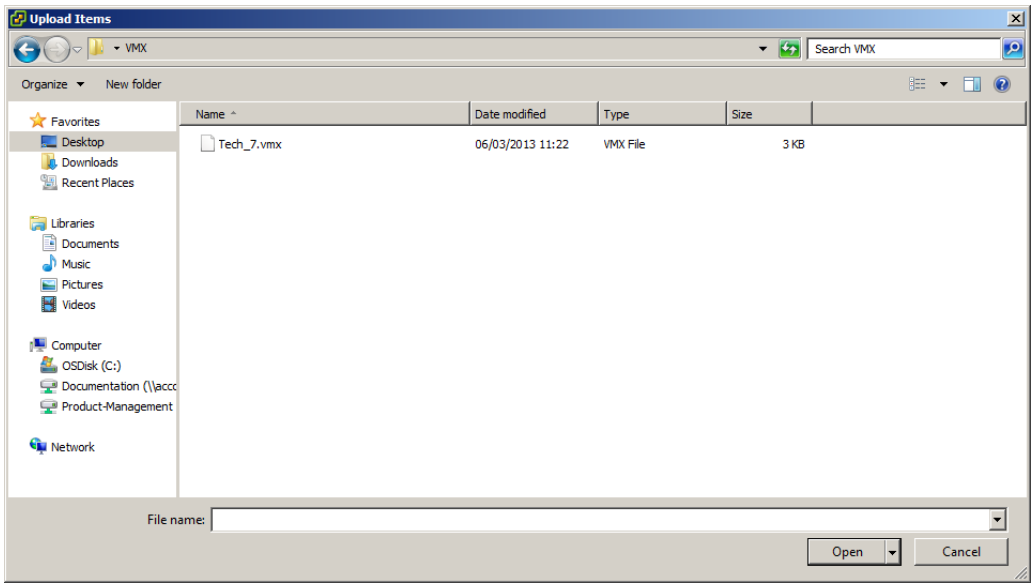
b Change the file extension to ".bak".

51 In the tool bar of the *Datastore Browser* window, click the **Upload files to this datastore** button.



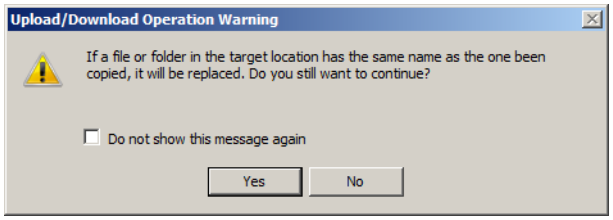
52 Click **Upload File.**

The *Upload Items* window appears.



53 Navigate to where you saved the ".vmx" file in step 49, select it, then click **Open**.

The *Upload/Download Operation Warning* window may appear.



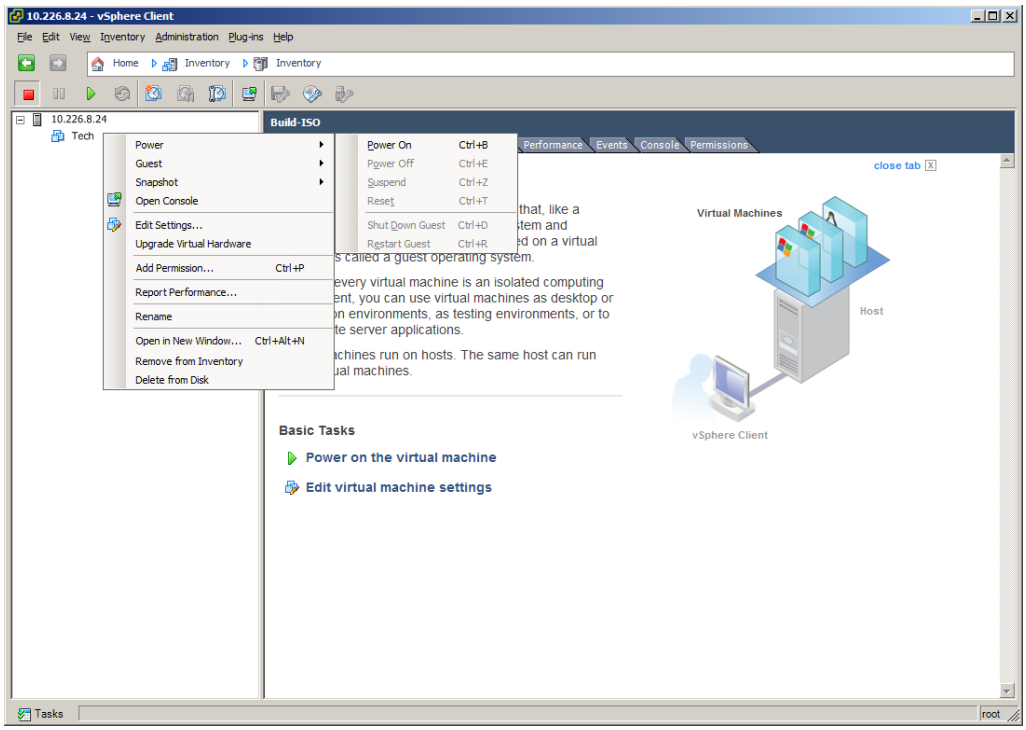
54 If the *Upload/Download Operation Warning* window appears, click **Yes**.

The file is uploaded.

55 Close the *Datastore Browser*.

56 In the *Inventory Panel*, click the Datastore that houses the MCU.

57 Right-click the MCU virtual machine, then click **Power > Power On**.



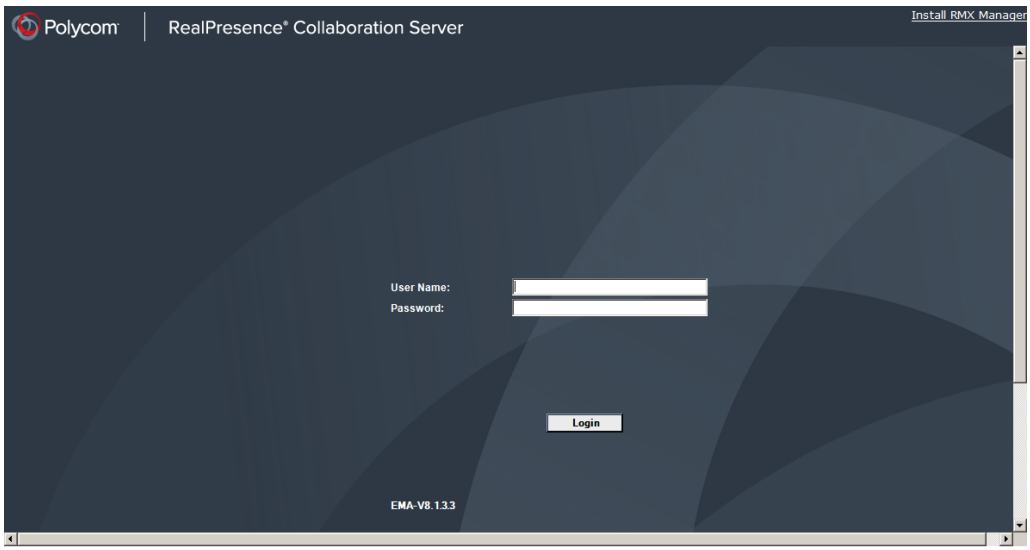
After a few minutes, the MCU turns on.

58 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`.

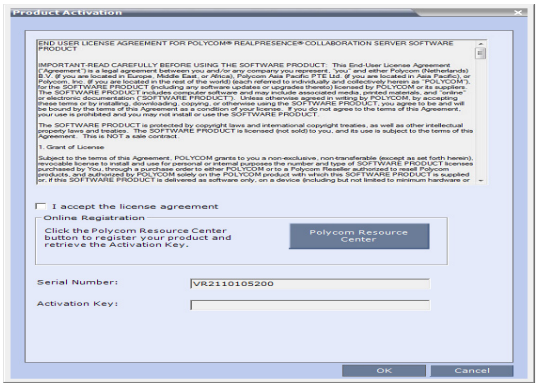
b Click **Enter**.

The *Collaboration Server Web Client* Login screen is displayed.



59 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:



60 In the *Activation Key* field, enter or **paste** the *Product Activation Key* that was used on the previous MCU.

61 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.

62 Click **OK**.

If the *Product Activation* dialog box does not appear, go to **Setup --> Product Activation** to display the dialog box.

63 On the *RealPresence Collaboration Server* menu, click **Administration** > **Software Management** > **Restore Configuration**.

64 **Browse** to the *Restore Directory Path* where the backed up configuration files are stored and then click **Restore**.

Upgrading or Downgrading the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition

These instructions are applicable to the RealPresence Collaboration Server 800s only.

To upgrade or downgrade the RealPresence Collaboration Server 800s:

1 Inset the Polycom USB key that came with the RealPresence Collaboration Server 800s into your computer.

The *Polycom Documentation* window is displayed.

In Windows XP:

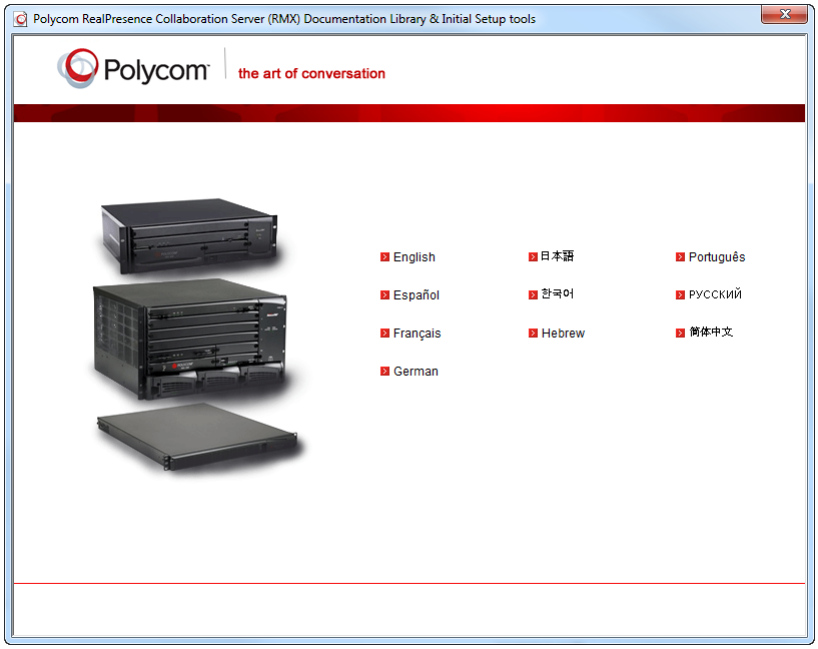
a The **Polycom Documentation** option is automatically selected. Click **OK**.

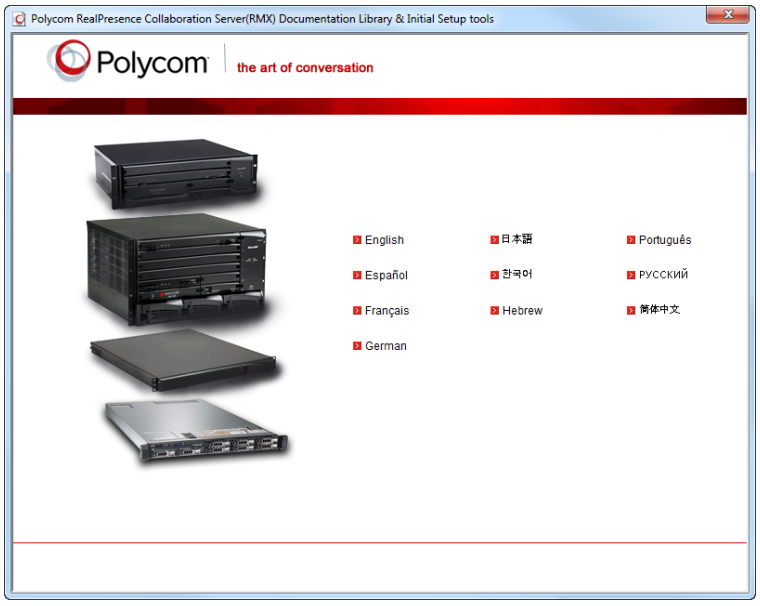
In Windows 7:

b Select **Open Folder to view files using Windows Explorer**.

c Double-click the **index.hta** file.

The *Language Menu* is displayed, offering a choice of several languages.



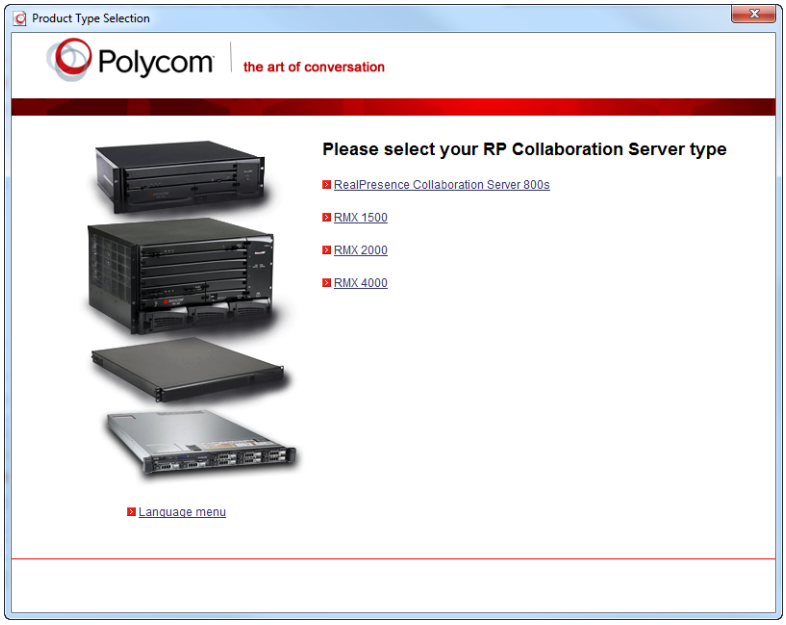


2 Click the documentation language of your choice.

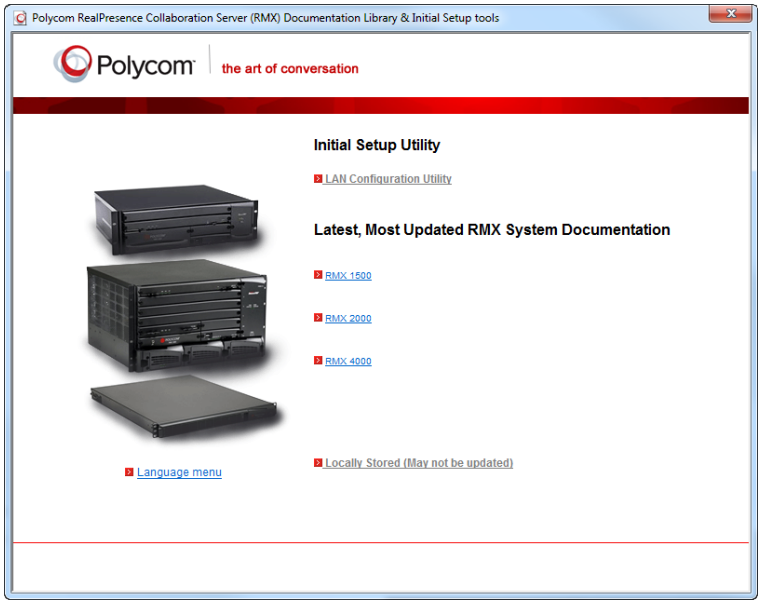
An *End-User Licence Agreement for Polycom Software* is displayed.

3 Read the agreement and click the **Accept Agreement** button.

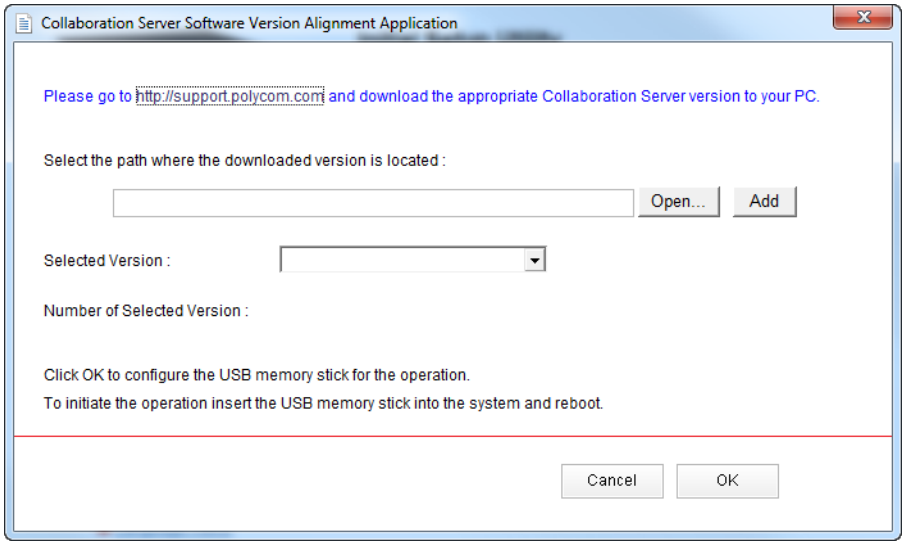
4 In the *Product Type* dialog box, select **RealPresence Collaboration Server 800s**.



5 In the *Initial Setup Utility* dialog box, click the **Upgrade / Downgrade** link.

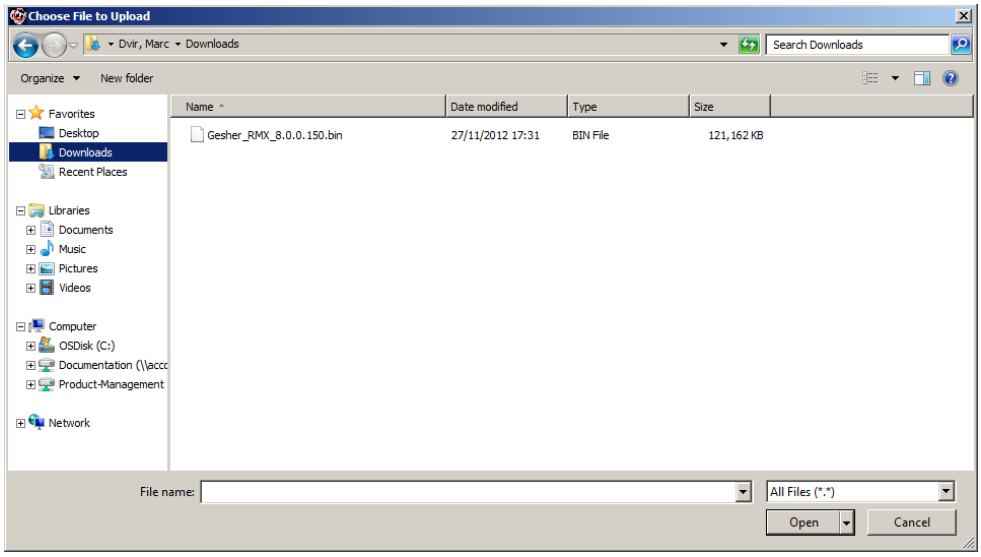


The *Collaboration Server Software Version Alignment Application* dialog box is displayed.

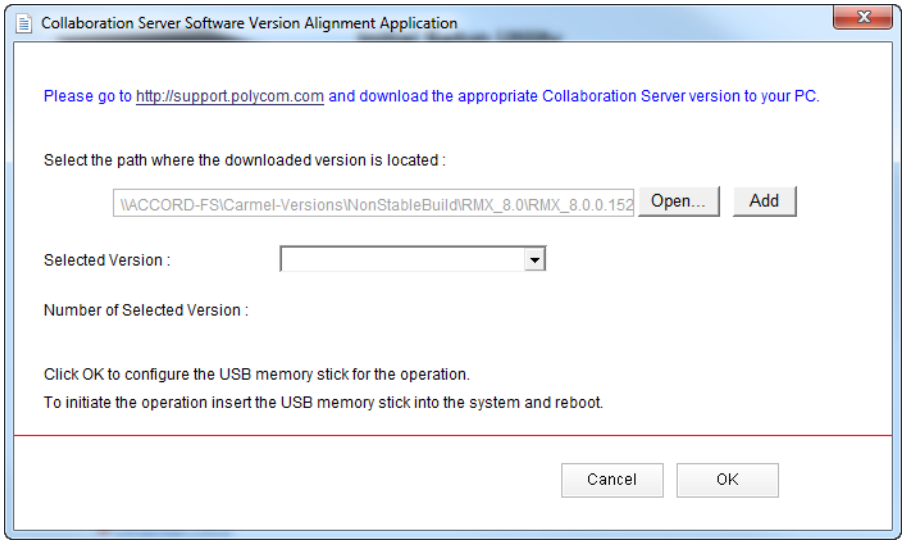


6 Click **Open**.

7 Navigate to the folder where the upgrade or downgrade software you have downloaded is saved and click **Open.**



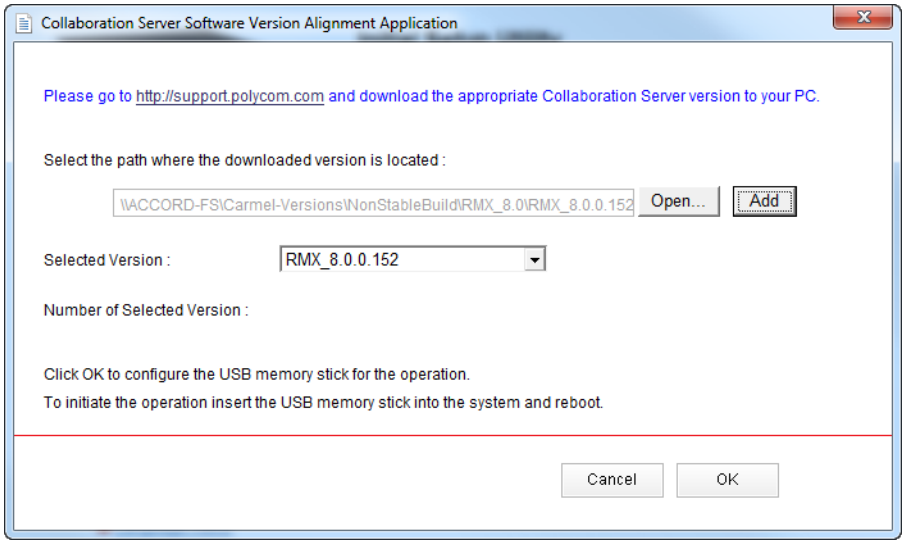
The software file is displayed in the *Select the path where the downloaded version is located* box.



8 Click **Add.**

This step may take a few minutes.

The version is added to the *Selected Version* dropdown box.



9 Make sure the correct version is selected in the *Selected Version* dropdown box and click **OK.**

10 Remove the *USB key* from the PC workstation.

11 Insert the *USB key* in any USB port of the RealPresence Collaboration Server 800s.

12 Turn off the Collaboration Server, then turn it On.

This step may take up to ten minutes.

13 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 following format: `http://<Control Unit IP Address>.`

b Click **Enter**.

When the *Collaboration Server Web Client* Login window is displayed, the version change was successful.

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 Resetting the Collaboration Server (RMX).

The following **System Flags** do not require an MCU reset:

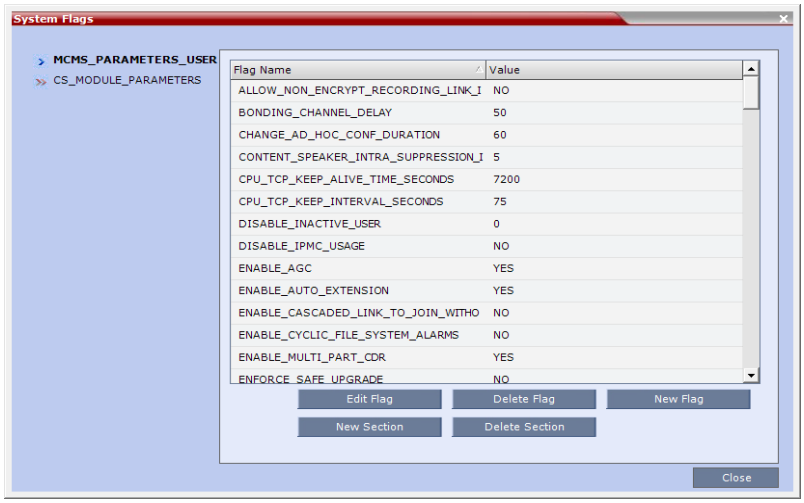
- IVR_MESSAGE_VOLUME
- IVR_MUSIC_VOLUME
- IVR_ROLL_CALL_VOLUME
- ENABLE_SELECTIVE_MIXING

Modifying System Flags

To modify system flags:

1 On the *Collaboration Server* menu, click **Setup > System Configuration**.

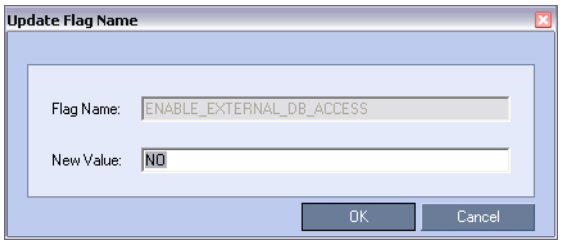
The **System Flags** dialog box opens.



2 In the *MCMS_PARAMETERS_USER* tab, the flags listed in the **MCMS_PARAMETERS_USER Flags** table can be modified.

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.



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 (RMX).

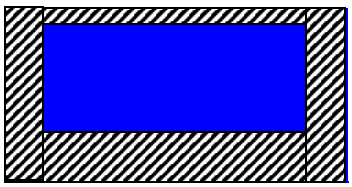
MCMS_PARAMETERS_USER Flags

Flag	Description
ALLOW_NON_ENCRYPT_PARTY_IN_ENCRYPT_CONF	If YES, allows non-encrypted participants to connect to encrypted conferences. Default: No
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

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. 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.
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

Flag	Description
ENFORCE_SAFE_UPGRADE	Applicable to the RealPresence Collaboration Server 800s only. 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 Polycom® RealPresence® Collaboration Server 800s and Polycom® 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, Changing a User’s Password. 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.
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.

Flag	Description
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)

Flag	Description
ITP_CROPPING	If the conference is set to TelePresence mode, cropping of the image is done according to this flag value: • **ITP (default)** - Cropping is done as follows: ⅄ Left/right sides: no cropping ⅄ Top/Bottom: the calculated area to be stripped will be split and cropped equally from the top and the bottom of the display area.  • **CP** - Cropping is done as follows: ⅄ Left/right sides: the calculated area to be stripped will be split and cropped equally from the top and bottom of the image ⅄ Top/Bottom: the calculated area to be stripped will be split and cropped equally from both sides.  • MIXED - Cropping is done as follows: ⅄ Left/right sides: the calculated area to be stripped will be split and cropped equally from the top and bottom of the image ⅄ Top/Bottom: the calculated area to be stripped will be cropped 84% of the calculated area to be stripped will be cropped from the bottom, and 16%will be cropped from the top.  **Note:** If the flag was added with no value, and the conference is set to TelePresence mode, cropping is done as follows: • Left/right sides: no cropping • Top/Bottom: the calculated area to be stripped will be cropped 84% of the calculated area to be stripped will be cropped from the bottom, and 16%will be cropped from the top.

Flag	Description
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.
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

Flag	Description
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: • HD1080 - High Definition at 30 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.
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

Flag	Description
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
MIN_TIP_COMPATIBILITY_LINE_RATE	This flag determines the minimum line rate at which conferencing entities such as an Entry Queue or Meeting Room can be TIP-enabled and TIP-enabled endpoints can connect to them. CTS version 7 requires a minimum line rate of 1024 kbps and will reject calls at lower line rates, therefore the System Flag value should be 1024 kbps or higher. **0** means that no minimum line rate is enforced on the conference for TIP connectivity. Default: 1024
MS_ENVIRONMENT	If YES, sets the Collaboration Server SIP environment to integrate with Microsoft OCS solution. Default: NO
MULTIPLE_SERVICES	Applicable to the RealPresence Collaboration Server (RMX) 800s only. 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.

Flag	Description
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.
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.

Flag	Description
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.
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	Applicable to the RealPresence Collaboration Server (RMX) 800s only. Enables/disables the Network Separation Default: NO.

Flag	Description
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
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

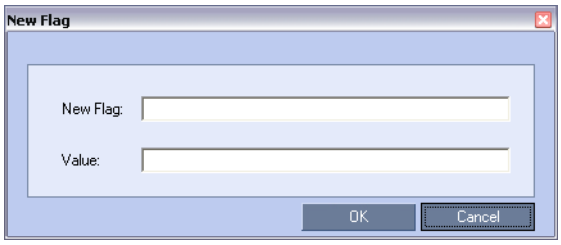
Flag	Description
TERMINATE_CONF_AFTER_CHAIR_DROPPED	From Version 8.1, this flag's functionality is replaced by the **Terminate Conference after Chairperson Drops** check box in the *Profile - IVR* dialog box. In versions prior to 8.1, if YES, sets conferences to automatically terminate if the Chairperson 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

Manually Adding and Deleting System Flags

To add a flag:

1 In the **System Flags** dialog box, click the **New Flag** button.

The **New Flag** dialog box is displayed.



2 In the **New Flag** field enter the flag name.

3 In the **Value** field enter the flag value.

The flags in the **Manually Added, Modified, Deleted System Flags** table can be manually added to the *MCMS_PARAMETERS_USERS* tab.

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.

Manually Added, Modified, Deleted System Flags

Flag	Description
802_1X_CERTIFICATE_MODE	Not supported with Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition.
802_1X_SKIP_CERTIFICATE_VALIDATION	Not supported with Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition.
802_FIPS_MODE	Not supported with Polycom® RealPresence® Collaboration Server 800s and Polycom® 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

Manually Added, Modified, Deleted System Flags

Flag	Description
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.
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 RealPresence 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.

Manually Added, Modified, Deleted System Flags

Flag	Description
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).
CP_REGARD_TO_INCOMING_SETUP_RATE	For use in the Avaya Environment. If set to YES, the RealPresence 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

Manually Added, Modified, Deleted System Flags

Flag	Description
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_H323_EP_DISCONNECT_TIMER	Allows you to specify the amount of time the MCU waits before disconnecting H.323 endpoint.
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.
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.

Manually Added, Modified, Deleted System Flags

Flag	Description
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 RealPresence 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 RealPresence 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).
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.
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

Manually Added, Modified, Deleted System Flags

Flag	Description
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_MCCF	Enables or disables the support of External IVR Services via the MCCF-IVR package is enabled. In Ultra Secure Mode and in secured environments where the External IVR Services via the MCCF-IVR package is not required and unused ports should be closed, this flag should be set to NO. **Range:** YES / NO **Default: YES** (in Standard security Mode) or **NO** (in Ultra Secure Mode) **Note**: Ultra Secure Mode is not supported by 800s and Virtual Edition MCUs.
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 RealPresence Collaboration Server and that he/she is being connected as Audio Only. Default: YES
ENABLE_RECORDING_OPERATION_VIA_SIPINFO	Enables or disables recording control operations to be performed using either DTMF tones or a SIP INFO request.

Manually Added, Modified, Deleted System Flags

Flag	Description
ENABLE_SELECTIVE_MIXING	Enables (default) or disables the Automatic muting of noisy AVC endpoints. For more details, see Automatic Suppression of Noisy Endpoints (AVC Endpoints). 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).
EXTERNAL_CONTENT_DIRECTORY	The Web Server folder name. Change this name if you have changed the default names used by the CMA application. Default: /PlcmWebServices

Manually Added, Modified, Deleted System Flags

Flag	Description
EXTERNAL_CONTENT_IP	Enter the IP address of the CMA 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 server.
EXTERNAL_CONTENT_PORT	The CMA 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 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 443. Default: 5005.
FADE_IN_FADE_OUT	Not supported from version 8.1
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

Manually Added, Modified, Deleted System Flags

Flag	Description
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.
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

Manually Added, Modified, Deleted System Flags

Flag	Description
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.
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. Default: NO.

Manually Added, Modified, Deleted System Flags

Flag	Description
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 conferences run on the RealPresence® Collaboration Server 800s and 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)* and *ping6 (IPv6)* commands. When set to **NO**, the Collaboration Server will not respond to *ping* and *ping6* 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	Applicable to the RealPresence Collaboration Server (RMX) 800s only. 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.
LIMIT_SD_AND_CIF_BW_MPMRX	When to YES (default), limits the maximum negotiated and opened bit rate for resolutions equal or lower than SD to 1Mbps. When set to NO no limitation is applicable to SD and CIF bit rates. Range: YES/NO. Default: YES.
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 Room Switch Telepresence Layouts.

Manually Added, Modified, Deleted System Flags

Flag	Description
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
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 conferences run on the RealPresence® Collaboration Server 800s and 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

Manually Added, Modified, Deleted System Flags

Flag	Description
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
NUM_OF_PCM_IN_MPMX	In Collaboration Server 1500/2000/4000 systems with MPMx cards, sets the maximum number of PCM sessions. The default value of this flag is set according to the SVC license: **1** - If SVC is enabled in the license (the only possible value). **4** - If SVC is disabled in the license Range: 1-4 (If SVC is disabled in the license).
NUMBER_OF_REDIAL	Enter the number re dialing attempts required. Dialing may continue until the conference is terminated. Default: 3
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)

Manually Added, Modified, Deleted System Flags

Flag	Description
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).
PASSWORD_FAILURE_LIMIT	The number of unsuccessful Logins permitted in Ultra Secure Mode. Default: 3 **Note**: Ultra Secure Mode is not supported by 800s and Virtual Edition MCUs.
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

Manually Added, Modified, Deleted System Flags

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 800s / Virtual Edition Getting Started Guide*, *Force Layout and Preserve Participant Call*.
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

Manually Added, Modified, Deleted System Flags

Flag	Description
REMOVE_EP_FROM_LAYOUT*f*n_ON_NO_VIDEO_TIMER	Enables the removal of empty video cells from the Video Layout. **Range:** 0 – 19 (seconds): The feature is disabled. 20 – 300 (seconds): The feature is enabled. **Default:** 20 For more information, see Remove Empty Cells From the Video Layout.
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.

Manually Added, Modified, Deleted System Flags

Flag	Description
RFC2833_DTMF	Controls the receipt of in-band and out-of-band DTMF Codes. When set to: • YES The RMX will receive DTMF Codes sent in-band. • NO The RMX receive DTMF Codes sent out-of-band. The RMX always sends DTMF Codes in-band (as part of the Audio Media stream). Range: YES/NO Default YES
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
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
RTV_MAX_BIT_RATE_FOR_FORCE_CIF_PARTICIPANT	Enables the removal of empty video cells from the Video Layout.
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

Manually Added, Modified, Deleted System Flags

Flag	Description
SEND_SIP_BUSY_UPON_RESOURCE_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
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.

Manually Added, Modified, Deleted System Flags

Flag	Description
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
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).

Manually Added, Modified, Deleted System Flags

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_OMIT_DOMAIN_FROM_PARTY_NAME	Provides option to remove Domain Names from SIP dial-in participants' Site Names. This prevents long domain names being appended to SIP participant names.
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.

Manually Added, Modified, Deleted System Flags

Flag	Description
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).
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_ULTRA_SECURED_SUPPORT	This flag must be set to YES when deploying a *Serial Gateway S4GW* in *Ultra Secure Mode*. **Note**: Ultra Secure Mode is not supported by 800s and Virtual Edition MCUs.
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.

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:

Manually Added CS_MODULE_PARAMETERS System Flags

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
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

Manually Added CS_MODULE_PARAMETERS System Flags

Flag	Description
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 the following **Default Auto Layouts** table.
The flag that controls the **Auto Layout** you wish to modify must be added to the **System Configuration** file. For more information see Manually Adding and Deleting System Flags.

The table below lists the available layouts.

Available Layouts

No. of Cells	Layout	Layout Flag Value
1		CP_LAYOUT_1X1
2		CP_LAYOUT_1X2
		CP_LAYOUT_1X2HOR
		CP_LAYOUT_1x2VER
		CP_LAYOUT_2X1
3		CP_LAYOUT_1P2HOR
		CP_LAYOUT_1P2HOR_UP
		CP_LAYOUT_1P2VER
4		CP_LAYOUT_2X2
		CP_LAYOUT_1P3HOR_UP
		CP_LAYOUT_1P3VER
5		CP_LAYOUT_1P4HOR_UP
		CP_LAYOUT_1P4HOR
		CP_LAYOUT_1P4VER
6		CP_LAYOUT_1P5
8		CP_LAYOUT_1P7
9		CP_LAYOUT_1P8UP
		CP_LAYOUT_1P8CENT
		CP_LAYOUT_1P8HOR_UP
		CP_LAYOUT_3X3
		CP_LAYOUT_1TOP_LEFT_P8
10		CP_LAYOUT_2P8
		CP_LAYOUT_2TOP_P8
13		CP_LAYOUT_1P12
16		CP_LAYOUT_4X4

The table below lists the default layouts for according to the number of participants.

Default Auto Layouts

Flag Name: PREDEFINED_AUTO_LAYOUT_n (n = Number of Participants)		
n	**Layout**	**Layout Flag Value**
0		CP_LAYOUT_1X1
1		CP_LAYOUT_1X1
2		CP_LAYOUT_1X1
3		CP_LAYOUT_1x2VER
4		CP_LAYOUT_2X2
5		CP_LAYOUT_2X2
6		CP_LAYOUT_1P5
7		CP_LAYOUT_1P5
8		CP_LAYOUT_1P7
9		CP_LAYOUT_1P7
10		CP_LAYOUT_2P8
11		CP_LAYOUT_2P8
12		CP_LAYOUT_1P12

Example:

The following table 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.

Example of Predefined Auto Layouts

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 _IP5	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.

LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag Values

The following table lists the value for each video layout that can be defined for the LEGACY_EP_CONTENT_DEFAULT_LAYOUT Flag. It allows the selection of video layout that will be displayed on the screen of the legacy endpoint when switching to Content mode.

For a list of available layouts see Available Layouts.

CS_ENABLE_EPC Flag

Endpoints that support **People+Content** 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, Reservations, 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 conferences, Reservations, and 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:

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.

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*. **Note**: Ultra Secure Mode is not supported by 800s and Virtual Edition MCUs.
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.

Hardware Monitoring

The status and properties of the MCU hardware components can be viewed and monitored in the *Hardware Monitor* list pane.

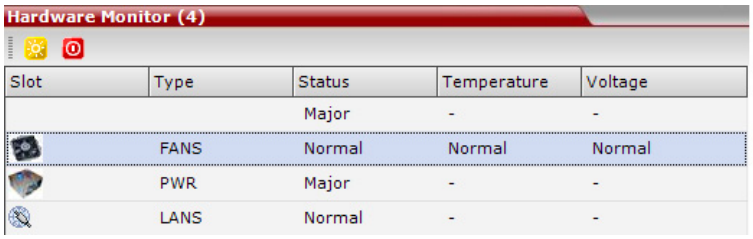
Viewing the Status of the Hardware Components

The *Hardware Monitor* pane displays the hardware component, its present status, temperature and voltage.

To view the status of the Hardware Components on the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition:

- In the *RealPresence Collaboration ServerManagement* pane, click the **Hardware Monitor** button.

  The *Hardware Monitor* pane is displayed.



  The *Hardware Monitor* pane displays the following information:

Hardware Monitor Pane Columns

Field	Description
Slot	Displays an icon according to the type of hardware component. The icon indicates the hardware status as an exclamation point (**!**) indicates errors in the hardware component.
Type	The type of the hardware component.
Status	The current status of the hardware component; *Normal, Major, or Critical*.
Temperature	Monitors the temperature of the hardware components; Normal, Major and Critical. **Note:** Critical condition invokes a system shut down.
Voltage	The voltage threshold of the hardware component; either *Normal* or *Major*.

Hardware Monitor Toolbar

The following buttons appear in the toolbar of the Hardware Monitor:

Button	Name	Description
	System Reset	Resets and restarts the system. Resetting saves settings and information that you changed in the system, i.e. IP Services, etc...
	System Shut Down	Shuts down the system into a standby mode. When the user in the *RMX Manager Manager/Web Client* presses the *System Shut Down* (red) button in the H*ardware Monitor* toolbar, the system should enter a standby mode and the LED turns ON. Only the media and control unit cards are in a standby mode. Shelf Manager remains active. Turn the system OFF/ON to exit the standby mode.

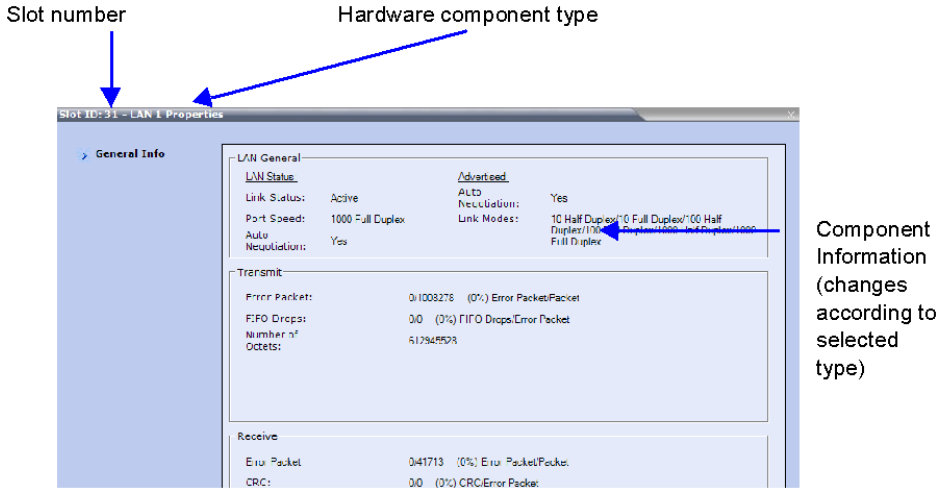
Viewing the Properties of Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition Hardware Components

The properties displayed for the hardware components will vary according to the type of component viewed. These component properties can be grouped as follows:

- MCU Properties (Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition)
- Supporting Hardware Components Properties (FANS, LAN)

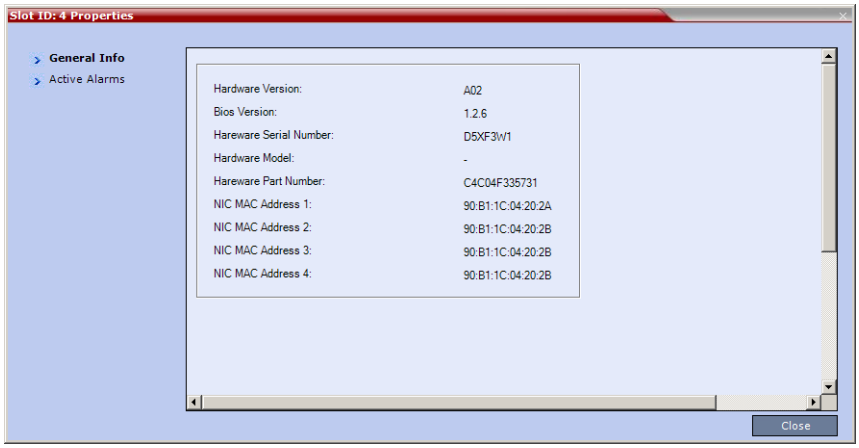
No properties are provided for Power Supply (PWR).

The Hardware Properties dialog box has the following structure:



To view the MCU Properties:

1 In the *Hardware Monitor* pane, either double-click or right-click and select **Properties** for *RealPresence Collaboration Server, slot 4*.



The following information is displayed:

MCU Properties - General Info

Field	Description
Hardware Version	The version of the server hardware.
Bios Version	The version of the BIOS installed on the server.
Hardware Serial Number	The hardware serial number.
Hardware Model	Always blank.
Hardware Part Number	The hardware part number.

Field	Description
Network Interface Cards (NIC)	
NIC number (1-4)	The number of the NIC.
NIC MAC Address	The physical address of the NIC.

2 Click the **Active Alarms** tab to view alarms related to the RealPresence Collaboration Server, i.e. temperatures, CPU usage, memory usage, voltages and main power sensors.

The *Active Alarms* dialog box displays fields that relate to faults and errors detected on the RealPresence Collaboration Server by sensors. The *Active Alarms* dialog box is divided into two sections: *Hardware Alarm List* and *Software Alarm List*.

The Hardware Alarms list and software alarm list can each be saved to an Excel file (*.xls) by clicking the **Save hardware Alarm List** and **Save SW Alarm List** buttons respectively. The severity of the alarms is color coded: Critical (RED), Major (ORANGE) and Normal (GREEN).

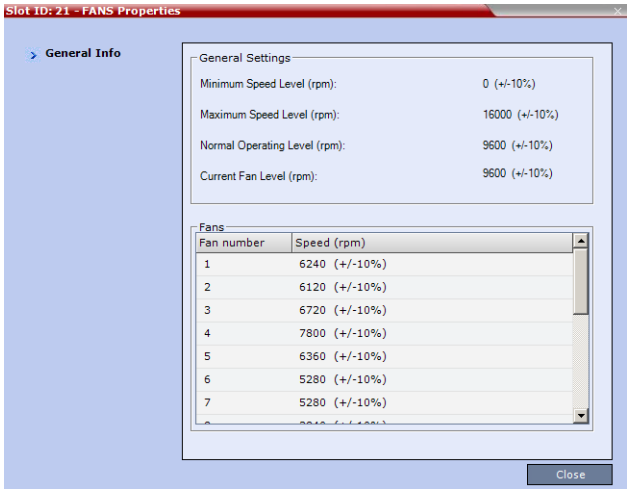
3 Click **Close** to return to the *Hardware Monitor* pane.

FAN Properties:

The RealPresence Collaboration Server chassis contains 14 fans that regulate the unit's temperature. If the temperature increases, the fans speed will increase and vice-versa. A "Critical" condition in the fans operation will result in a system shut down.

To access the Fan properties:

» In the *Hardware Monitor* pane, either double-click or right-click and select **Properties** for **FANS.**



FANS Properties - General Info

Field	Description
General Settings	
Min. Speed Level (rpm)	The minimum speed level of the fans.
Max. Speed Level (rpm)	The maximum speed level of the fans.
Normal Operating Level (rpm)	The normal operating level defined for the fans.
Current Fan Level (rpm)	The current operating level of the fans.
Fans	
Fan number (1-14)	Fan numbering
Speed (rpm)	Present speed of a fan (1-14).

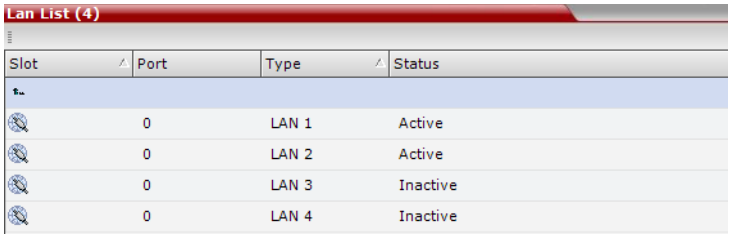
LAN 1, LAN 2, LAN 3, LAN4 Properties:

The Collaboration Server unit's chassis contains 4 external LAN connectors which register the following information listed below. The information will be refreshed every second and also contains a peek detector to log the maximal values, since the last peek values reset.

To access the properties of LAN connector:

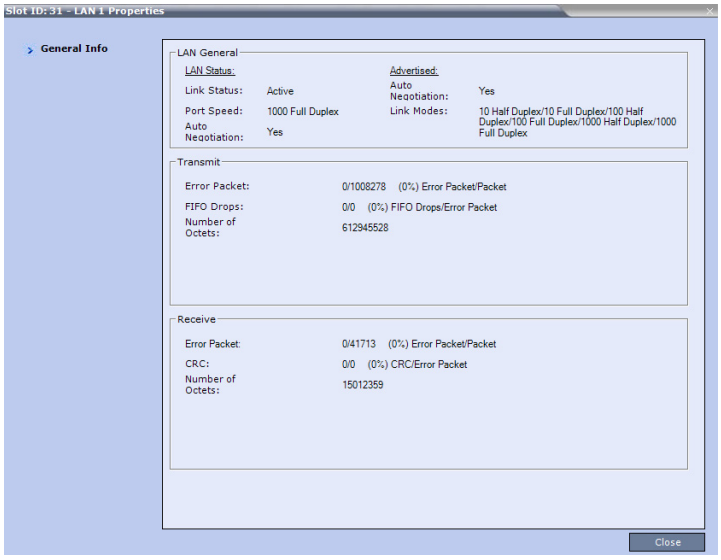
1 In the *Hardware Monitor* pane, double-click anywhere on the line for *LANS*.

The *Lan List* is displayed:



2 Either double-click or right-click and select **Properties** for the Lan connector.

The properties for the LAN connector are displayed:



Diagnostics

Your system comes with a Hardware Diagnostics USB memory stick. If required for troubleshooting, Polycom Global Services personnel may ask you to run hardware diagnostics.

The appliance will not be operational during the running of diagnostics.

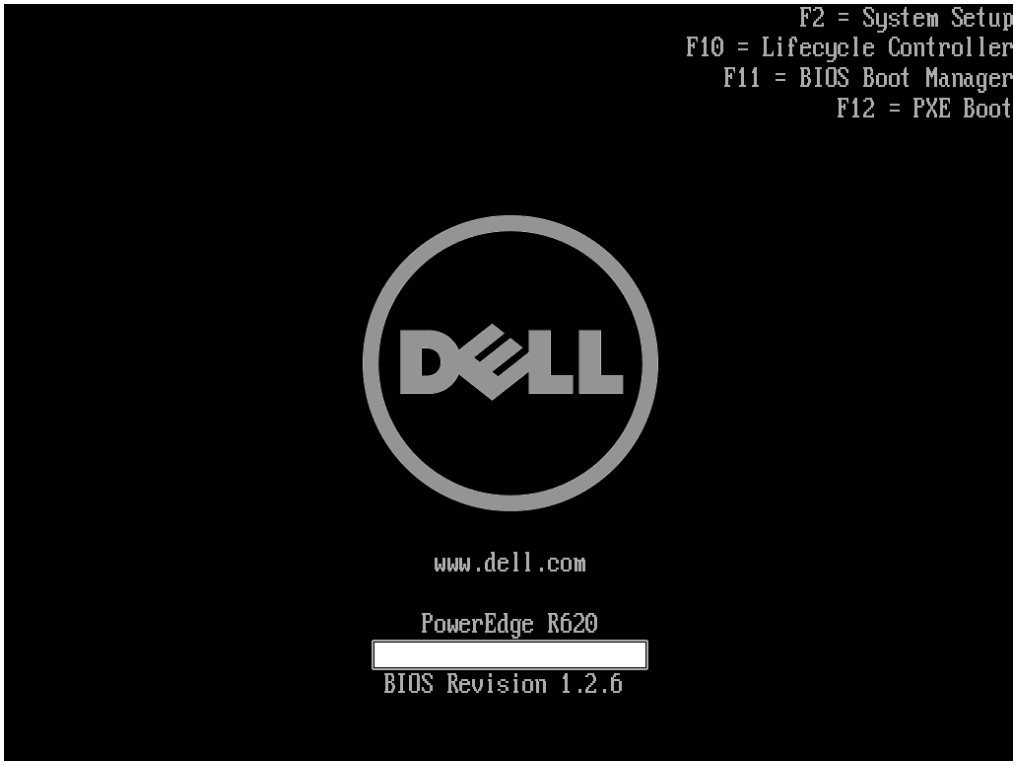
The following are required to run the diagnostics:

- Hardware diagnostics USB memory stick
- USB Keyboard
- VGA Monitor

To perform diagnostics:

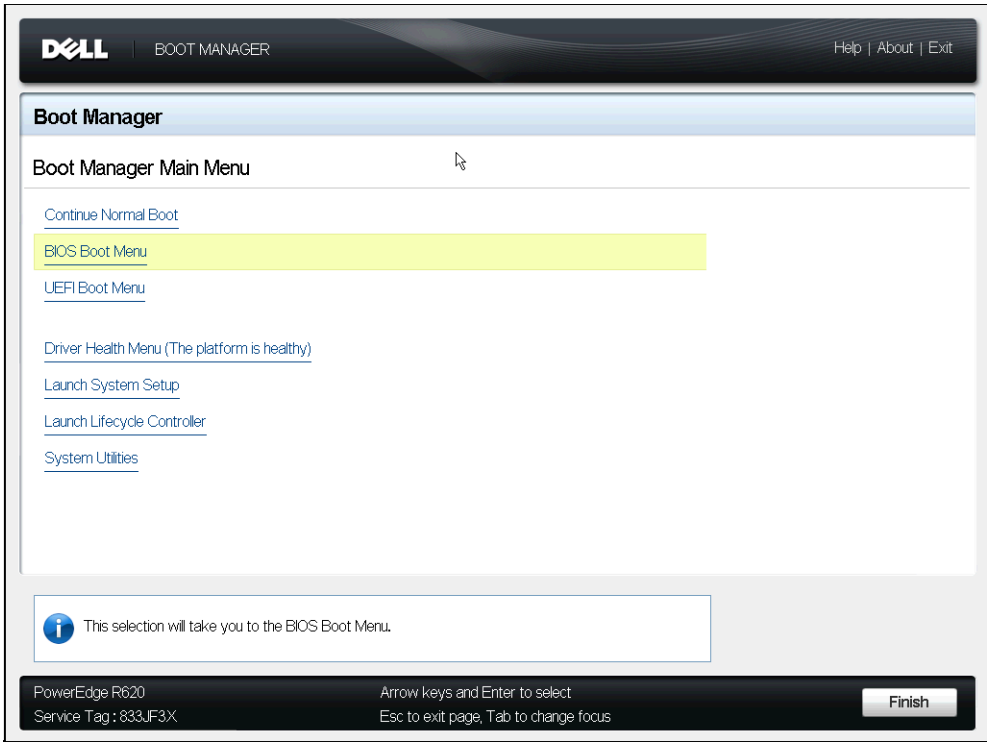
1. Turn off the *Collaboration Server*.
2. Connect a keyboard and a VGA monitor.
3. Insert the USB memory stick containing the hardware diagnostics utility into any of the system's USB ports.
4. Turn on the *Collaboration Server*.

   The server boots up.



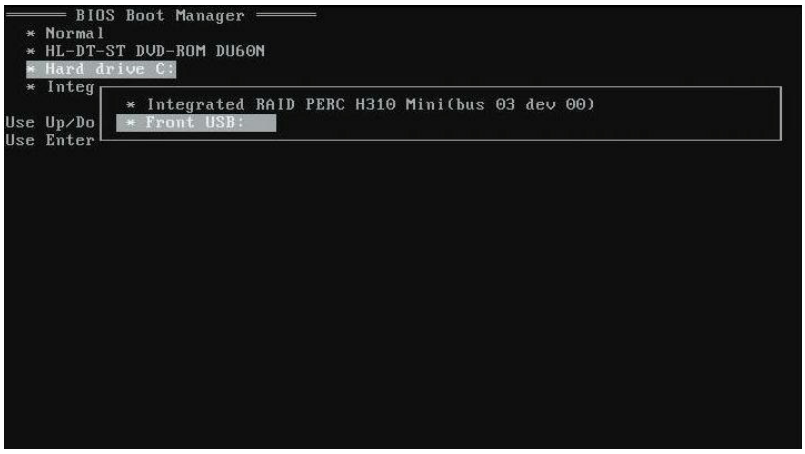
5 Press **F11**.

The *Boot Manager Main Menu* loads.



6 Select **Bios Boot Menu**.

The *Bios Boot Manager* is displayed.



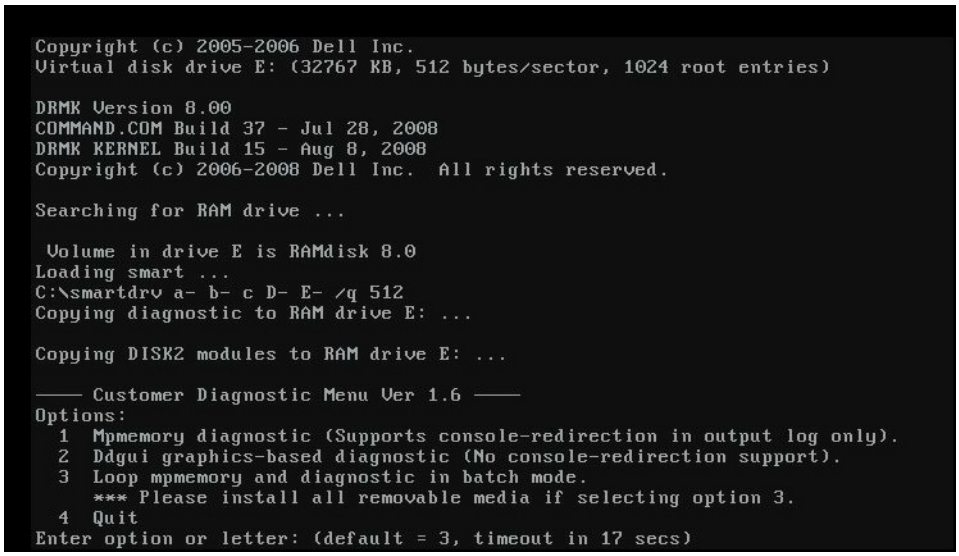
7 Using the Up/Down arrows go to **Hard drive C:**.

A window will display showing two options.

8 Select **Front USB**.

If the USB memory stick is inserted into the back of the server this option will be called *Back USB*.

The *Collaboration Server* loads the hardware diagnostics utility.



9 Select the desired diagnostic mode.
If none is selected the *Loop mpmemory and diagnostic in batch mode* tests will be performed automatically.

The diagnostics will continue running until stopped. One cycle of the diagnostics may take up to 30 minutes to complete.

The results are saved in a file called *auto.txt* to the USB memory stick.

10 Press **Escape** to stop the diagnostics.

11 Remove the USB memory stick.

12 **Optional:** Disconnect the keyboard and monitor.

If the *RealPresence Collaboration Server* is in secure mode the keyboard and monitor must be disconnected, otherwise the *RealPresence Collaboration Server* will revert to non-secure mode when the *RealPresence Collaboration Server* is started.

13 Turn off the *Collaboration Server*, then turn it On.

14 Contact Polycom Global Services for further instructions.

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

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.
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.

Disconnection Cause	Description
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.
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.

Disconnection Cause	Description
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.
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.

Disconnection Cause	Description
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.
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.

Disconnection Cause	Description
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.
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).

Disconnection Cause	Description
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

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.

Active Alarms

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	

Active Alarms

Alarm Code	Alarm Description
eUserMsgCode_Cs_SipTLS_CertificateSubjNameIsNotValid_Or_DnsFailed	
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	

Active Alarms

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	

Active Alarms

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.

Active Alarms

Alarm Code	Alarm Description
Hot backup: Network issue	
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

Active Alarms

Alarm Code	Alarm Description
Management Network not configured	Configure the Management Network.
Missing Central Signaling configuration	Configure the central signaling.
Missing Central Signaling IP configuration	
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	

Active Alarms

Alarm Code	Alarm Description
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	
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.	

Active Alarms

Alarm Code	Alarm Description
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.
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.

Active Alarms

Alarm Code	Alarm Description
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	
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	

Active Alarms

Alarm Code	Alarm Description
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

:Conference Summary Record Fields

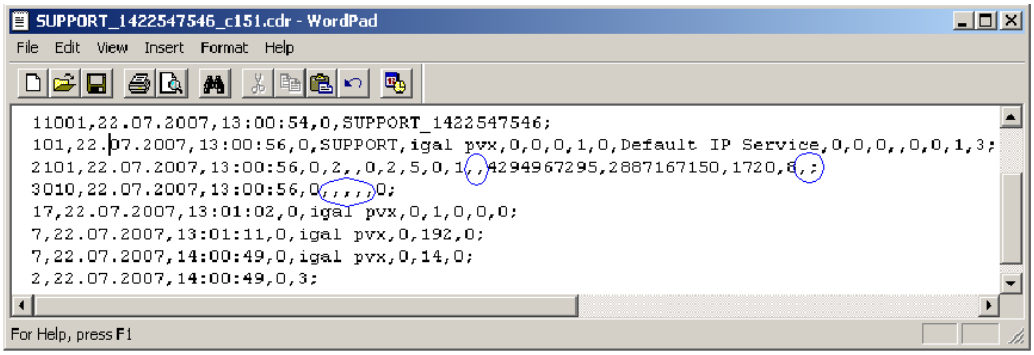
Field	Description
File Version	The version of the CDR utility that created the file.
Conference Routing Name	The Routing Name of the conference.
Internal Conference ID	The conference identification number as assigned by the system.
Reserved Start Time	The time the conference was scheduled to start in 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.

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:



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 CDR Event Types.
- 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.

CDR Event Types

Event Code	Event Name	Description
1	CONFERENCE START	The conference started. For more information about the fields, see Event Fields for Event 1 - CONFERENCE START. **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 Event Fields for Event 2 - CONFERENCE END. **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 Event Fields for Event 7 - PARTICIPANT DISCONNECTED.
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 Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT. **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 Event fields for Event 15 - H323 CALL SETUP.
17	H323 PARTICIPANT CONNECTED	An H.323 participant connected to the conference. For more information about the fields, see Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED.
18	NEW UNDEFINED PARTICIPANT	A new undefined participant joined the conference. For more information about the fields, see Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT.

Event Code	Event Name	Description
20	BILLING CODE	A billing code was entered by a participant using DTMF codes. For more information about the fields, see Event Fields for Event 20 - BILLING CODE.
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 Event Fields for Event 21 - SET PARTICIPANT DISPLAY NAME.
22	DTMF CODE FAILURE	An error occurred when a participant entered a DTMF code. For more information about the fields, see Event Fields for Event 22 - DTMF CODE FAILURE.
23	SIP PARTICIPANT CONNECTED	A SIP participant connected to the conference. For more information about the fields, see Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED.
26	RECORDING LINK	A recording event, such as recording started or recording resumed, occurred. For more information about the fields, see Event fields for Event 26 - RECORDING LINK.
28	SIP PRIVATE EXTENSIONS	Contains SIP Private Extensions information. For more information about the fields, see Event Fields for Event 28 - SIP PRIVATE EXTENSIONS.
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 Event Fields for Event 30 - GATEKEEPER INFORMATION.
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 Event fields for Event 31 - PARTICIPANT CONNECTION RATE.
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 Event fields for Event 33 - PARTY CHAIR UPDATE.
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 Event Fields for Event 35 - SVC SIP PARTICIPANT CONNECTED.

Event Code	Event Name	Description
100	USER TERMINATE CONFERENCE	A user terminated the conference. For more information about the fields, see Event Fields for Event 100 - USER TERMINATE CONFERENCE.
101	USER ADD PARTICIPANT	A user added a participant to the conference during the conference. For more information about the fields, see Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT.
102	USER DELETE PARTICIPANT	A user deleted a participant from the conference. For more information about the fields, see Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT.
103	USER DISCONNECT PARTICIPANT	A user disconnected a participant. For more information about the fields, see Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT.
104	USER RECONNECT PARTICIPANT	A user reconnected a participant who was disconnected from the conference. For more information about the fields, see Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT.
105	USER UPDATE PARTICIPANT	A user updated the properties of a participant during the conference. For more information about the fields, see Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT.
106	USER SET END TIME	A user modified the conference end time. For more information about the fields, see Event Fields for Event 106 - USER SET END TIME.
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 Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY.
108	OPERATOR MOVE PARTY TO CONFERENCE	The Collaboration Server User moved the participant from an ongoing conference to another conference. For more information, see Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE,OPERATOR ATTEND PARTY TO CONFERENCE.
109	*OPERATOR ATTEND PARTY*	The Collaboration Server User moved the participant to the Operator conference. For more information, see Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY.

Event Code	Event Name	Description
111	OPERATOR BACK TO CONFERENCE PARTY	The Collaboration Server User moved the participant back to his Home (source) conference. For more information, see Event Fields for Event 111 - OPERATOR BACK TO CONFERENCE PARTY.
112	OPERATOR ATTEND PARTY TO CONFERENCE	The Collaboration Server User moved the participant from the Operator conference to another conference. For more information, see Event Fields for Events 108, 112 - OPERATOR MOVE PARTY TO CONFERENCE,OPERATOR ATTEND PARTY TO CONFERENCE.
1001	NEW UNDEFINED PARTICIPANT CONTINUE 1	Additional information about a NEW UNDEFINED PARTICIPANT event. For more information about the fields, see Event Fields for Event 1001 - NEW UNDEFINED PARTY CONTINUE 1.
2001	CONFERENCE START CONTINUE 1	Additional information about a CONFERENCE START event. For more information about the fields, see Event Fields for Event 2001 - CONFERENCE START CONTINUE 1.
2007	PARTICIPANT DISCONNECTED CONTINUE 1	Additional information about a PARTICIPANT DISCONNECTED event. For more information about the fields, see Event Fields for Event 2007 - PARTICIPANT DISCONNECTED CONTINUE 1.
2010	DEFINED PARTICIPANT CONTINUE 1	Additional information about a DEFINED PARTICIPANT event. For more information about the fields, see Event Fields for Events 2010, 2011, 2015 - DEFINED PARTICIPANT CONTINUE 1,USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1.
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.
2012	RESERVED PARTICIPANT CONTINUE 2	Additional information about a DEFINED PARTICIPANT event. For more information about the fields, see Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2012 - USER ADD PARTICIPANT CONTINUE 2, Event 2016 - USER UPDATE PARTICIPANT CONTINUE 2.
2101	USER ADD PARTICIPANT CONTINUE 1	Additional information about a USER ADD PARTICIPANT event. For more information about the fields, see Event Fields for Events 2010, 2011, 2015 - DEFINED PARTICIPANT CONTINUE 1,USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1.
2102	USER ADD PARTICIPANT CONTINUE 2	Additional information about a USER ADD PARTICIPANT event. For more information about the fields, see Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2012 - USER ADD PARTICIPANT CONTINUE 2, Event 2016 - USER UPDATE PARTICIPANT CONTINUE 2.

Event Code	Event Name	Description
2105	USER UPDATE PARTICIPANT CONTINUE 1	Additional information about a USER UPDATE PARTICIPANT event. For more information about the fields, see Event Fields for Events 2010, 2011, 2015 - DEFINED PARTICIPANT CONTINUE 1,USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1.
2106	USER UPDATE PARTICIPANT CONTINUE 2	Additional information about a USER UPDATE PARTICIPANT event. For more information about the fields, see Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2012 - USER ADD PARTICIPANT CONTINUE 2, Event 2016 - USER UPDATE PARTICIPANT CONTINUE 2.
3010	PARTICIPANT INFORMATION	The contents of the participant information fields. For more information about the fields, see Event Fields for Event 3010 - PARTICIPANT INFORMATION.
5001	CONFERENCE START CONTINUE 4	Additional information about a CONFERENCE START event. For more information about the fields, see Event Fields for Event 5001 - CONFERENCE START CONTINUE 4. **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 Event Fields for Event 6001 - CONFERENCE START CONTINUE 5.
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 Event Fields for Event 11001 - CONFERENCE START CONTINUE 10.

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.

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*.

Event Fields for Event 1 - CONFERENCE START

Table 1-1

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

Table 1-1

Field	Description
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
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**.

Event Fields for Event 2001 - CONFERENCE START CONTINUE 1

Table 1-2

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. Soft MCU: AVC, 4; SVC, 5.
Operator Conference	Not supported. Always contains the value **0**.
Video Protocol	The video protocol. Currently the only value is: **255** - Auto

Table 1-2

Field	Description
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**.
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**.

Table 1-2

Field	Description
Video Board ID	Not supported. Always contains the value **65535**.
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**.

Event Fields for Event 5001 - CONFERENCE START CONTINUE 4

Table 1-3

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.
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.

Event Fields for Event 6001 - CONFERENCE START CONTINUE 5

Table 1-4

Field	Description
Encryption	Indicates the conference encryption setting, as follows: **0** - The conference is *not* encrypted. **1** - The conference is encrypted.

Event Fields for Event 11001 - CONFERENCE START CONTINUE 10

Table 1-5

Field	Description
Display Name	The Display Name of the conference.

Event Fields for Event 2 - CONFERENCE END

Table 1-6

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.

Event fields for Event 3 - ISDN/PSTN CHANNEL CONNECTED

Event fields for Event 4 - ISDN/PSTN CHANNEL DISCONNECTED
Event fields for Event 5 - ISDN/PSTN PARTICIPANT CONNECTED

Event Fields for Event 7 - PARTICIPANT DISCONNECTED

Table 1-9

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 Disconnection Cause Values.
Q931 Disconnect Cause	If the disconnection cause is “No Network Connection” or “Participant Hang Up”, then this field indicates the Q931 disconnect cause.

Event Fields for Event 2007 - PARTICIPANT DISCONNECTED CONTINUE 1

Table 1-10

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**.
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**.

Table 1-10

Field	Description
H.323 Board ID	Not supported. Always contains the value **0**.
H323 Unit ID	Not supported. Always contains the value **0**.

Event Fields for Events 10, 101, 105 - DEFINED PARTICIPANT, USER ADD PARTICIPANT, USER UPDATE PARTICIPANT

Table 1-11

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
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

Table 1-11

Field	Description
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.
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

Event Fields for Events 2010, 2011, 2015 - DEFINED PARTICIPANT CONTINUE 1,USER ADD PARTICIPANT CONTINUE 1, USER UPDATE PARTICIPANT CONTINUE 1

Table 1-12

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.

Table 1-12

Field	Description
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.
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.

Event Fields for Event 2011 - DEFINED PARTICIPANT CONTINUE 2, Event 2012 - USER ADD PARTICIPANT CONTINUE 2, Event 2016 - USER UPDATE PARTICIPANT CONTINUE 2

Table 1-13

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.

Event fields for Event 15 - H323 CALL SETUP

Field	Description
Participant Name	The name of the participant.
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

Event Fields for Events 17, 23 - H323 PARTICIPANT CONNECTED, SIP PARTICIPANT CONNECTED

Table 1-14

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.
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**.

Table 1-14

Field	Description
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

Event Fields for Event 18 - NEW UNDEFINED PARTICIPANT

Table 1-15

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**.

Table 1-15

Field	Description
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
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

Table 1-15

Field	Description
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.
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.

Event Fields for Event 1001 - NEW UNDEFINED PARTY CONTINUE 1

Table 1-16

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.

Event Fields for Event 20 - BILLING CODE

Table 1-17

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).

Event Fields for Event 21 - SET PARTICIPANT DISPLAY NAME

Table 1-18

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.

Event Fields for Event 22 - DTMF CODE FAILURE

Table 1-19

Field	Description
Participant Name	The name of the participant.
Participant ID	The identification number assigned to the participant by the MCU.

Table 1-19

Field	Description
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.

Event fields for Event 26 - RECORDING LINK

Table 1-20

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.
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*.

Event Fields for Event 28 - SIP PRIVATE EXTENSIONS

Table 1-21

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.

Event Fields for Event 30 - GATEKEEPER INFORMATION

Table 1-22

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.

Event fields for Event 31 - PARTICIPANT CONNECTION RATE

Field	Description
Participant Name	The participant name.
Participant ID	The identification number assigned to the participant by the MCU.
Participant Current Rate	The participant line rate in Kbps.

Event Fields for Event 32

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)

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.

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
Participant ID	The identification number assigned to the participant by the MCU.

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
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	

Event Fields for Event 100 - USER TERMINATE CONFERENCE

Table 1-24

Field	Description
Terminated By	The login name of the user who terminated the conference.

Event Fields for Events 102,103, 104 - USER DELETE PARTICIPANT, USER DISCONNECT PARTICIPANT, USER RECONNECT PARTICIPANT

Table 1-25

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.

Event Fields for Event 106 - USER SET END TIME

Table 1-26

Field	Description
New End Time	The new conference end time set by the user, in GMT.
User Name	The login name of the user who changed the conference end time.

Event Fields for Events 107 and 109 - OPERATOR MOVE PARTY FROM CONFERENCE and OPERATOR ATTEND PARTY

Table 1-27

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.

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**
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

Field	Description
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.
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

Field	Description
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.
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.

Field	Description
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.

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.

Event Fields for Events 2011, 2012, and 2016

Table 1-28

Field	Description
IP V6	IPv6 address of the participant’s endpoint.

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:* Appendix A - Disconnection Causes.

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
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

Value	Call Disconnection Cause
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
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

Value	Call Disconnection Cause
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
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

Value	Call Disconnection Cause
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 *Polycom® RealPresence® Collaboration Server 800s and Polycom® 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.

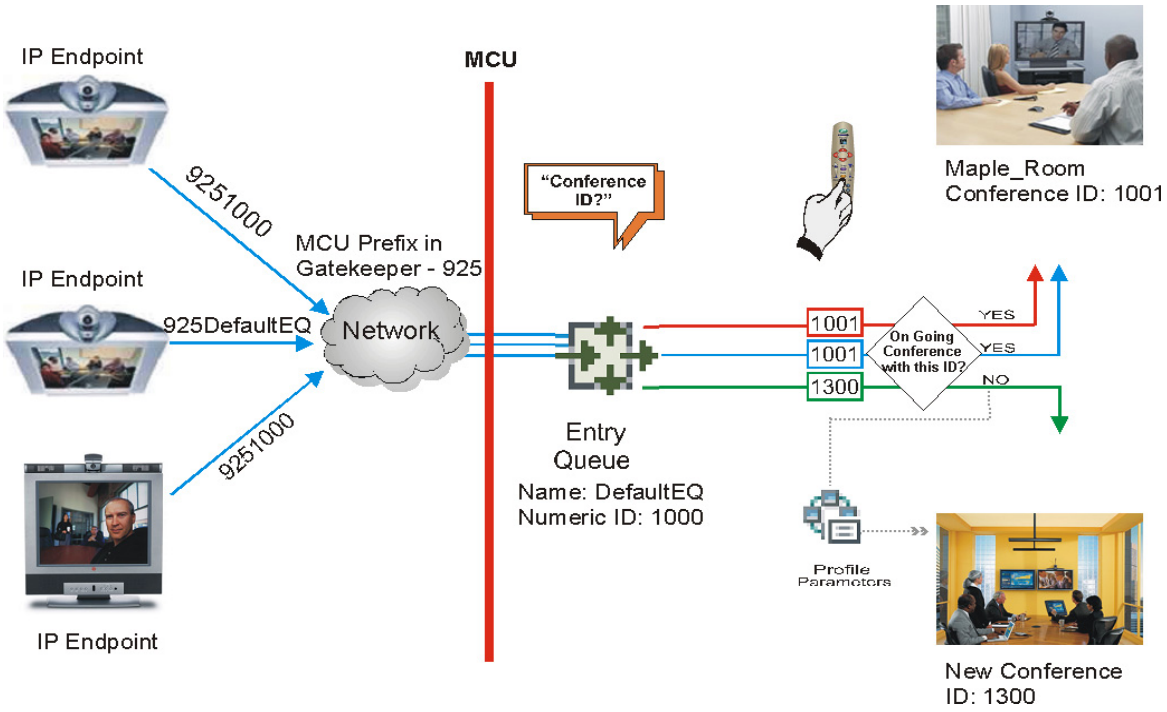
Ad Hoc Conferencing without Authentication

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.

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

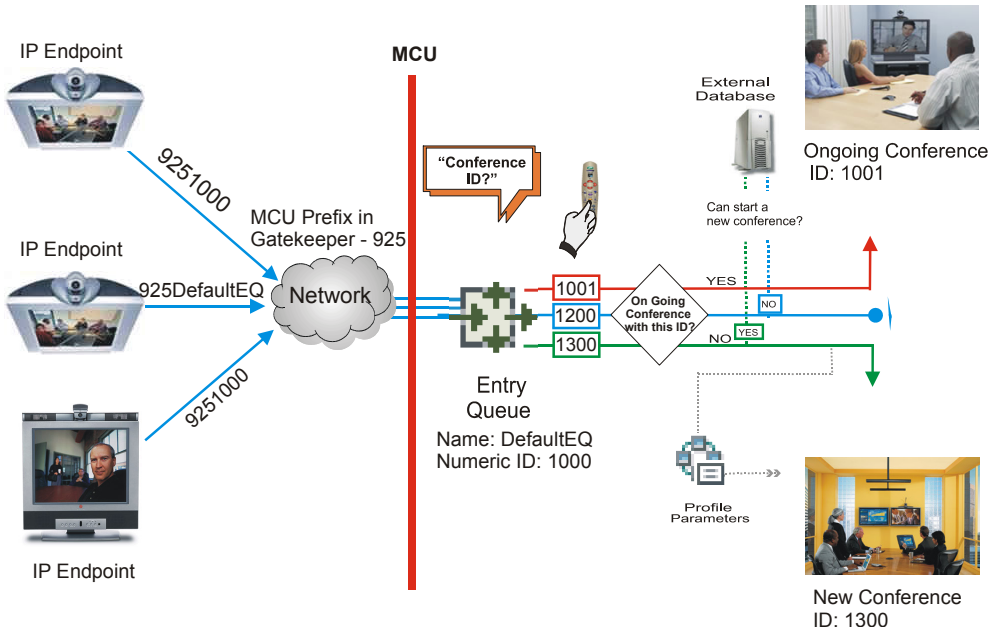
The MCU can work with an external database application to validate the participant's right to start a new conference. The external database contains a list of participants, with their assigned parameters. The conference ID entered by the participant is compared against the database. If the system finds 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:

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 MCU Configuration to Communicate with an External Database Application.

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:

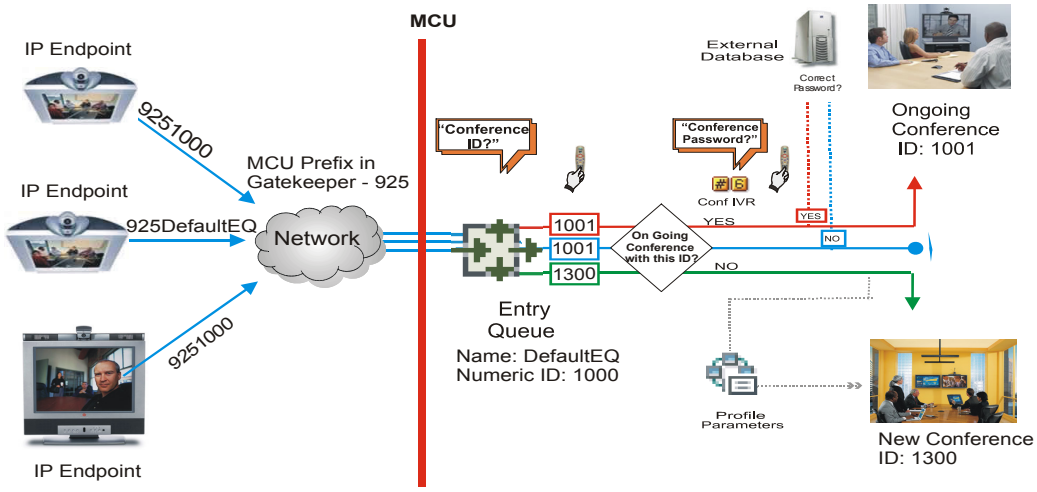
- Part of the Ad Hoc conferencing flow where the participants must be authorized before they can enter the conference created in the Ad Hoc flow
- Independent of Ad Hoc conferencing where conference access is validated for all conferences running on the MCU regardless of the method in which the conference 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.

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.

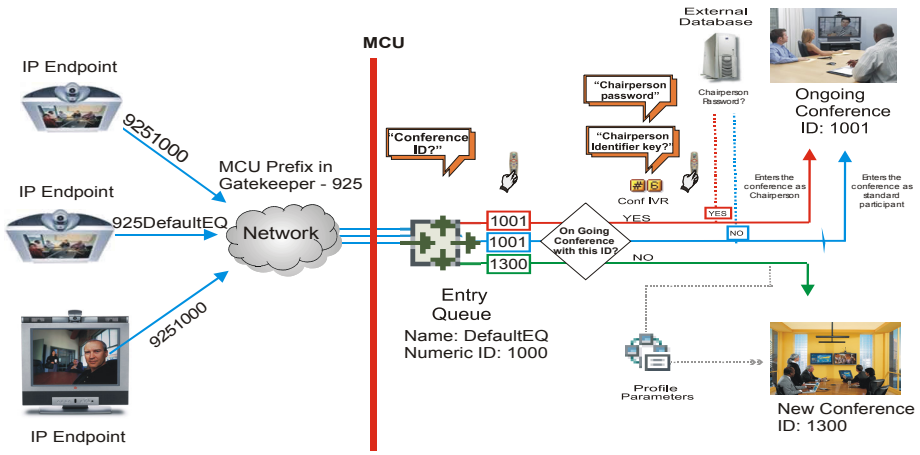
To enable conference access validation for all participants the following conferencing components are required:

- 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).

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 Using Conference Profiles.
- 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.

  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.

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.

- 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.

- 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.

- 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.

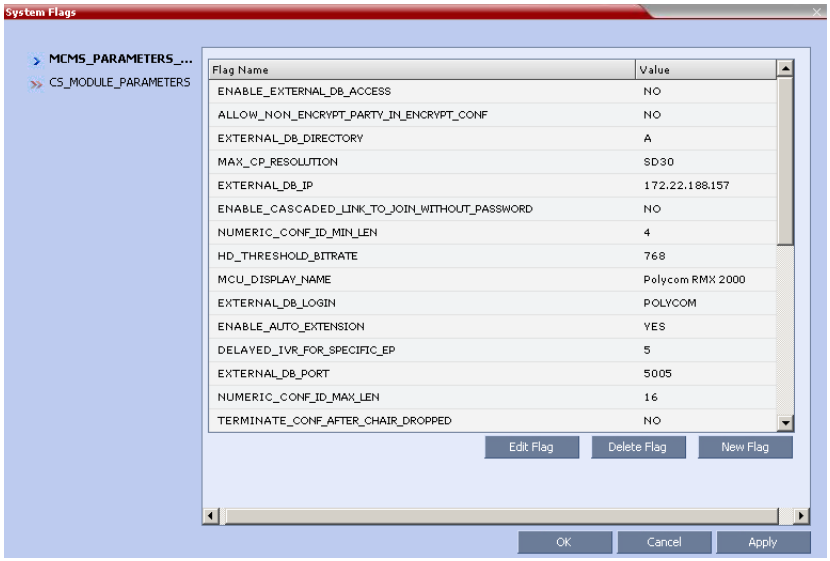
MCU Configuration to Communicate with an External Database Application

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.



2 Modify the values of the following flags:

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.

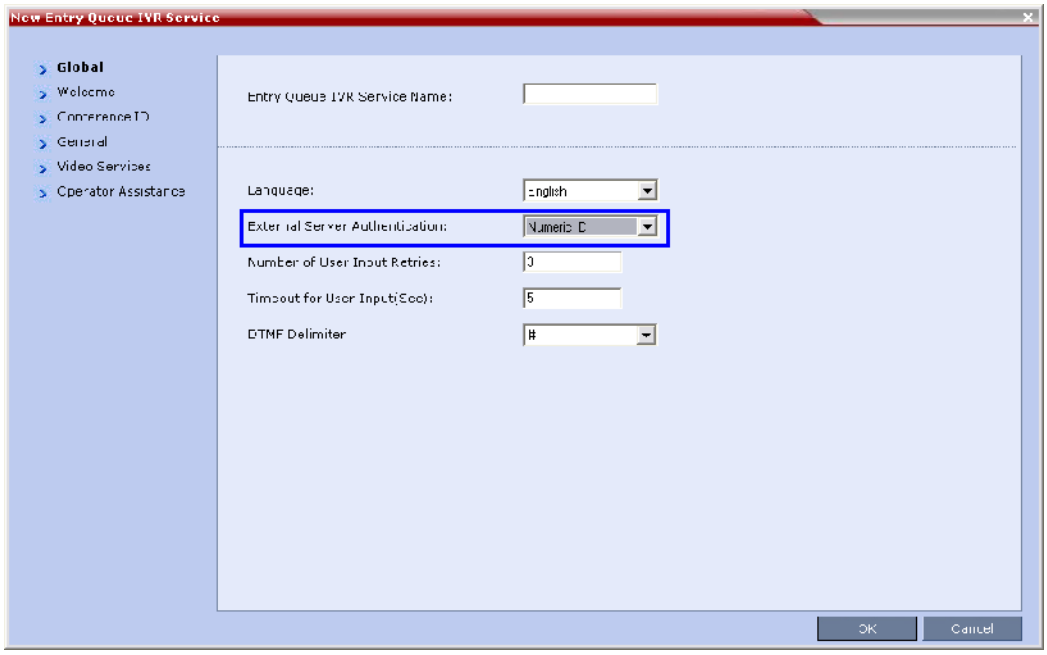
3 Click **OK**.

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**.



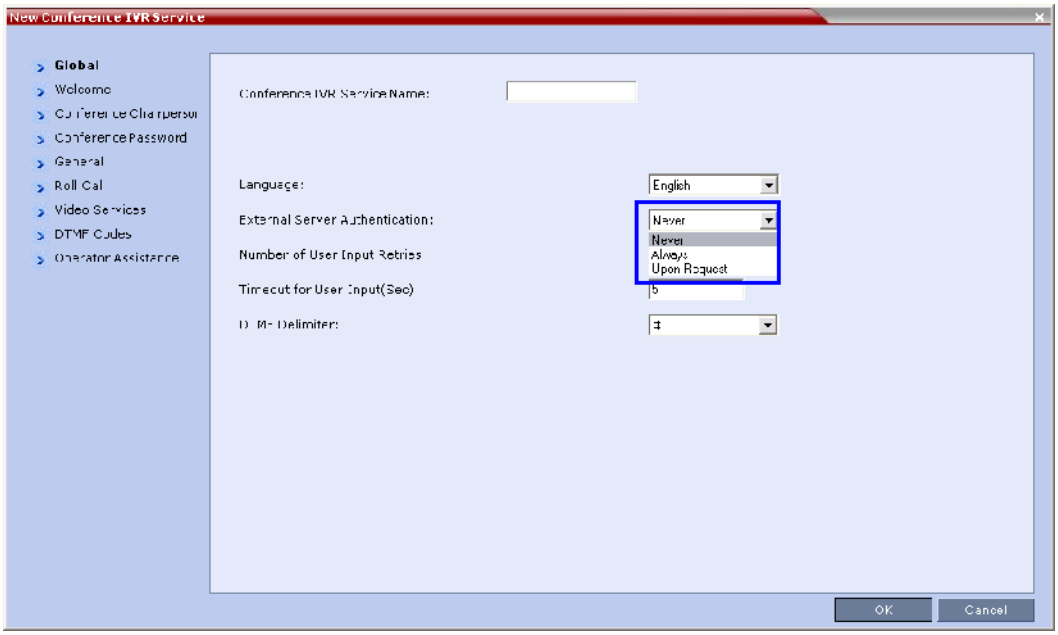
Enabling External Database Validation for Conferences Access

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



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.

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.
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.

Field	Description
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.
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 G - Configuring Direct Connections to the Collaboration Server

Direct connection to the Collaboration Server is necessary if you want to:

- Modify the Collaboration Server's *Factory Default Management Network* settings without using the USB memory stick.
- Connect to the Collaboration Server's *Alternate Management Network* for support purposes.
- Connect to the Collaboration Server via a modem.

Management Network (Primary)

If you do not want to use the USB memory stick method of modifying the Collaboration Server's *Management Network* parameters, it is necessary to establish a direct connection between a workstation and the Collaboration Server.

Configuring the Workstation

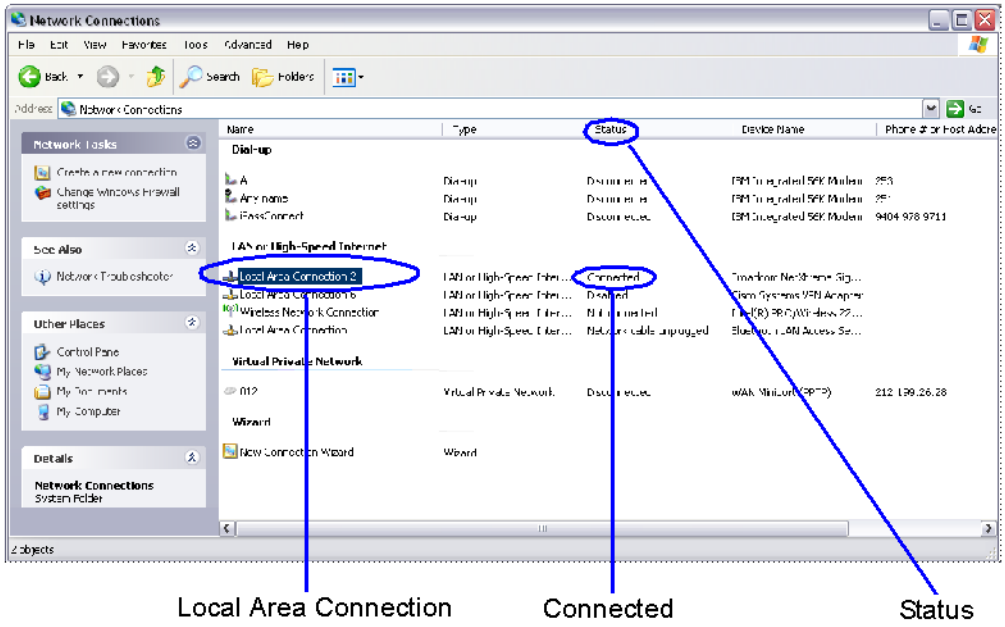
The following procedures show how to modify the workstation's networking parameters using the *Windows New Connection Wizard*.

For non-Windows operating systems an equivalent procedure must be performed by the system administrator.

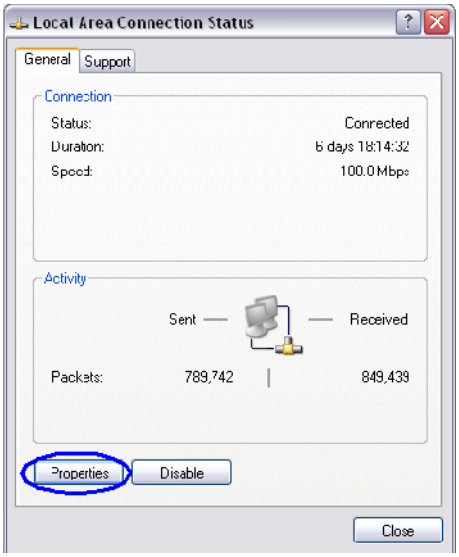
Before connecting directly, you must modify the *IP Address*, *Subnet Mask* and *Default Gateway* settings of the workstation to be compatible with either the Collaboration Server's *Default Management Network* or *Alternate Management Network*.

To modify the workstation's IP addresses:

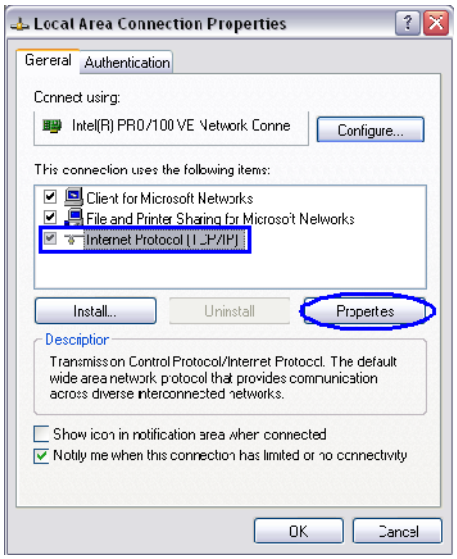
1. On the Windows *Start* menu, select **Settings** > **Network Connections.**
2. In the *Network Connections* window, double-click the **Local Area Connection** that has *Connected* status.



In the *Local Area Connection Status* dialog box, click the **Properties** button.

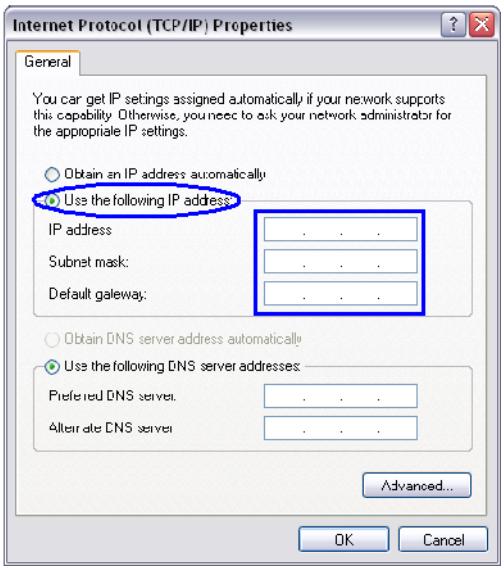


3 In the *Local Area Connection Properties* dialog box, select **Internet Protocol [TCP/IP]** > **Properties**.



4 In the *Internet Protocol (TCP/IP) Properties* dialog box, select **Use the following IP address**.

5 Enter the *IP address, Subnet mask* and *Default gateway* for the workstation.



The workstation's IP address should be in the same network neighborhood as the Collaboration Server's *Control Unit* IP address.

Example: *IP address* – near `192.168.1.nn`

None of the reserved IP addresses listed in *Reserved IP Addresses* should be used for the IP Address.

The *Subnet mask* and *Default gateway* addresses should be the same as those for the Collaboration Server's *Management Network*.

The addresses needed for connection to either the Collaboration Server's *Default Management Network* or *Alternate Management Network* are listed in the table below.

For more information about connecting to the *Alternate Management Network*, see Appendix G - Configuring Direct Connections to the Collaboration Server.

Reserved IP Addresses

Network Entity	IP Address	
	Management Network (Factory Default)	Alternate Network
Control Unit IP Address	192.168.1.254	169.254.192.10
Control Unit Subnet Mask	255.255.255.0	255.255.240.0
Default Router IP Address	192.168.1.1	169.254.192.1
Shelf Management IP Address	192.168.1.252	169.254.192.16
Shelf Management Subnet Mask	255.255.255.0	255.255.240.0

Network Entity	IP Address	
	Management Network (Factory Default)	Alternate Network
Shelf Management Default Gateway	192.168.1.1	169.254.192.1

6 Click the **OK** button.

Connecting to the Management Network

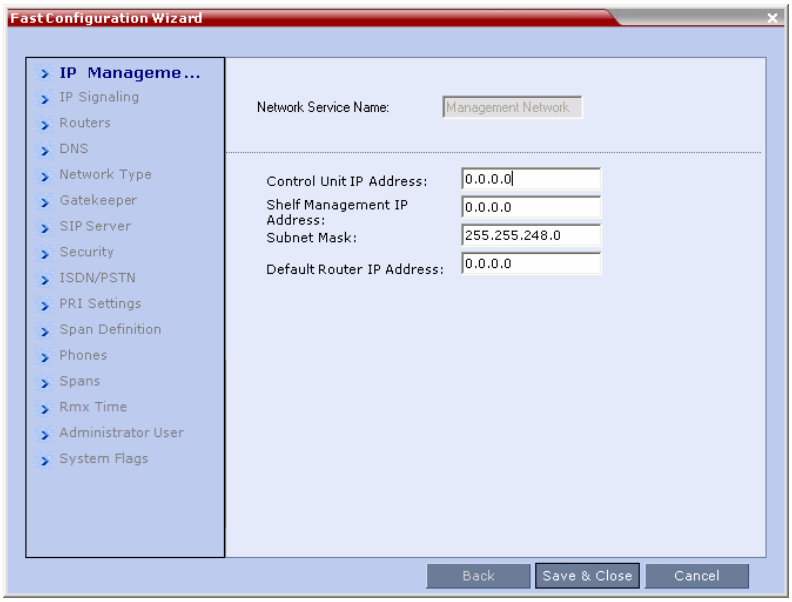
To connect directly to the Collaboration Server:

1 Connect one LAN cable between the PC and LAN 1 on the Collaboration Server's back panel.Connect the power cable and power the Collaboration Server **On**.

2 Start the *Collaboration Server Web Client* application on the workstation, by entering the factory setting *Management IP* address in the browser's address line and pressing **Enter**.

3 In the *Collaboration Server Web Client* Login screen, enter the default *Username* (**POLYCOM**) and *Password* (**POLYCOM**) and click the **Login** button.

The *Fast Configuration Wizard* starts.



If no *USB memory stick* is detected and **either**: this is the *First Time Power-up* **or** the *Default IP Service* has been deleted and the Collaboration Server has been reset, the following dialog box is displayed:

For more information about First-time Power-up and the *Fast Configuration Wizard* see the *Polycom® RealPresence Collaboration Server 800s / Virtual Edition Getting Started Guide*, Procedure 1: First-time Power-up.

4 Enter the following parameters using the information supplied by your network administrator:

- Control Unit IP Address

- Shelf Management IP Address
- Control Unit Subnet Mask
- Default Router IP Address

5 Click the **Save & Close** button.

The system prompts you to sign in with the new *Control Unit IP Address*.

6 Disconnect the LAN cable between the workstation and the LAN 1Port on the Collaboration Server's back panel.

7 Connect LAN 1Port on the Collaboration Server's back panel to the local network using a LAN cable.

8 Enter the new *Control Unit IP Address* in the browser's address line, using a workstation on the local network, and press **Enter** to start the *Collaboration Server Web Client* application.

9 In the *Collaboration Server Web Client* Login screen, enter the default *Username* (**POLYCOM**) and *Password* (**POLYCOM**) and click the **Login** button.

Connecting to the Collaboration Server via Modem

Remote access to the Collaboration Server's *Alternate Management Network* is supported via an external PSTN <=> IP modem.

To connect via modem to the *Alternate Management Network* the following procedures must be performed:

1 **Procedure 1: Install the RMX Manager** – the web client enables direct access to the Collaboration Server for support purposes.

2 **Procedure 2: Configure the modem** – by assigning it an IP address on a specific subnet in the *Alternate Management Network*.

3 **Procedure 3: Create a dial-up connection –** using the *Windows New Connection Wizard.*

4 **Procedure 4: Connect to the Collaboration Server** – via the *RMX Manager*.

Procedure 1: Install the RMX Manager

Before installing the *RMX Manager*, verify that you have at least 150Mb of free space on your workstation.

For more information see Installing the RMX Manager Application.

Procedure 2: Configure the Modem

Configure the modem as follows:

- **IP address** – near `169.254.192.nn`

- **Subnet Mask** – `255.255.240.0`

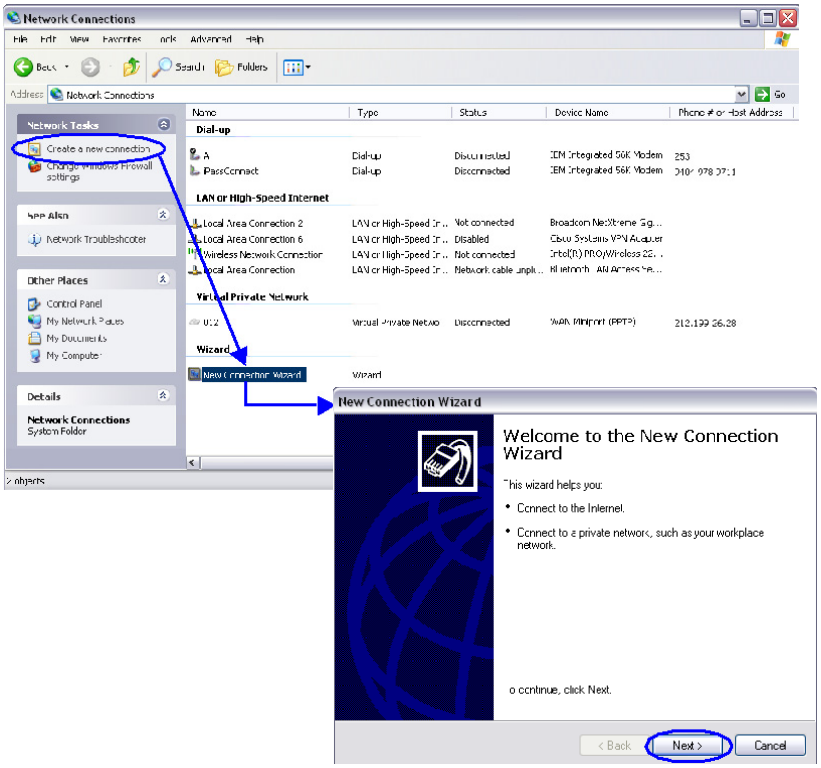
None of the reserved IP addresses listed in Table 5-127 on page **5-10** should be used for the IP Address.

Procedure 3: Create a Dial-up Connection

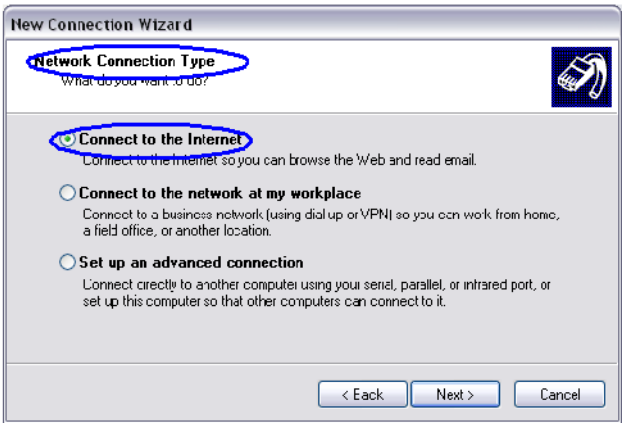
To create a dial-up connection:

This procedure is performed once. Only the *Dial* field in the *Connect* applet (see step 10 on Click the Dial button to establish a connection to LAN 3 Port via the modem.) is modified for connection to different modems.

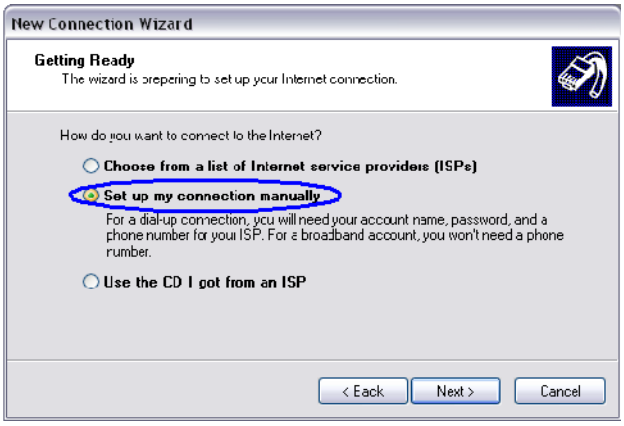
1. In *Windows*, navigate via the *Control Panel* to the *Network Connections* applet and select **Create a new connection.**
2. When the *New Connection Wizard* is displayed, click the **Next** button.



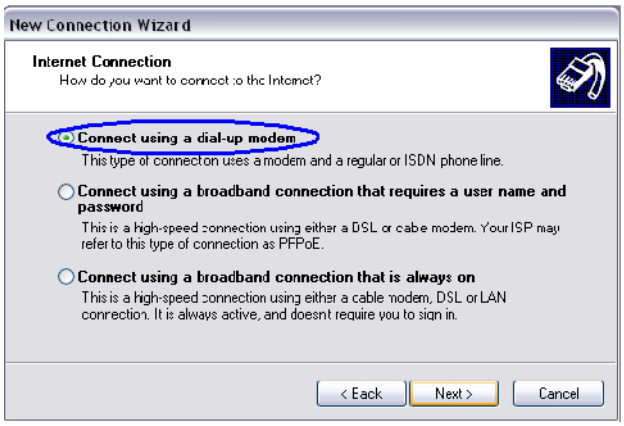
3 In the *Network Connection Type* box, select **Connect to the Internet** and click the **Next** button.



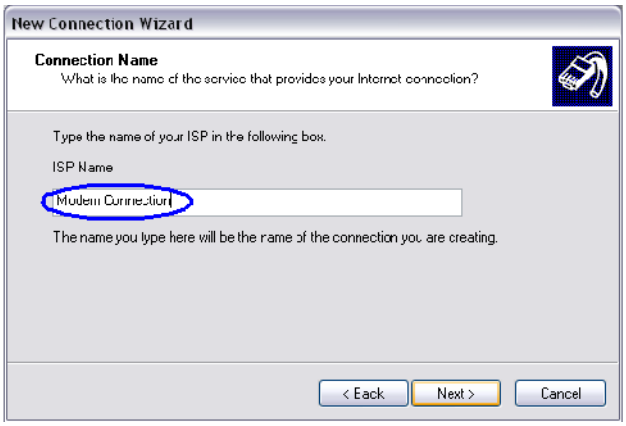
4 In the *Getting Ready* box, select **Set up my connection manually** and click the **Next** button.



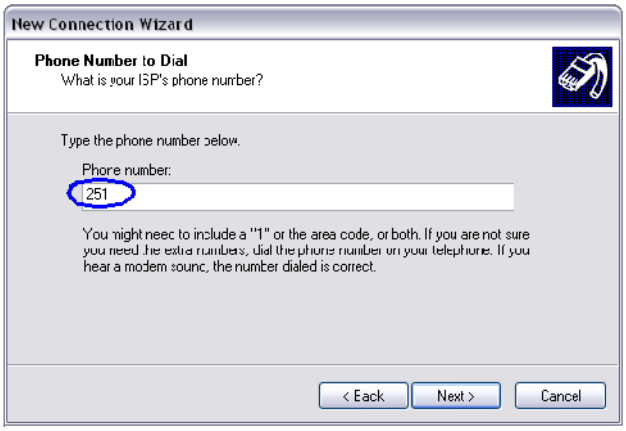
5 In the *Internet Connection* box, select **Connect using dial-up modem** and click the **Next** button.



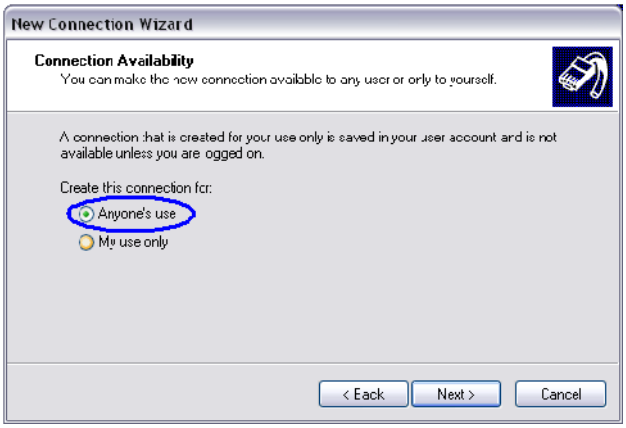
6 In the *Connection Name* box, enter a **Name** for the modem connection (e.g. *Modem Connection*) and click the **Next** button.



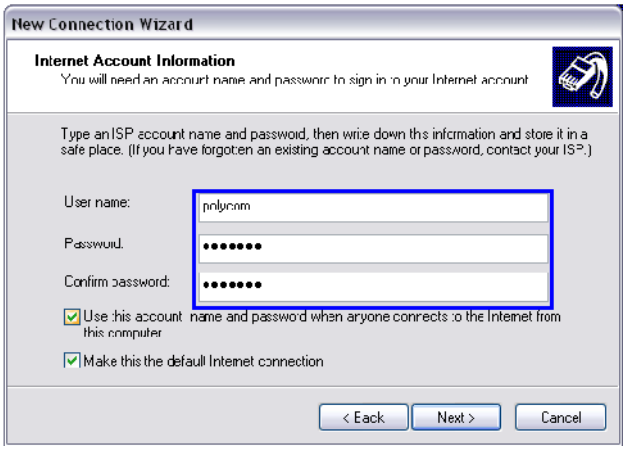
7 In the *Phone Number to Dial* box, enter the **Phone Number** for the modem and click the **Next** button.



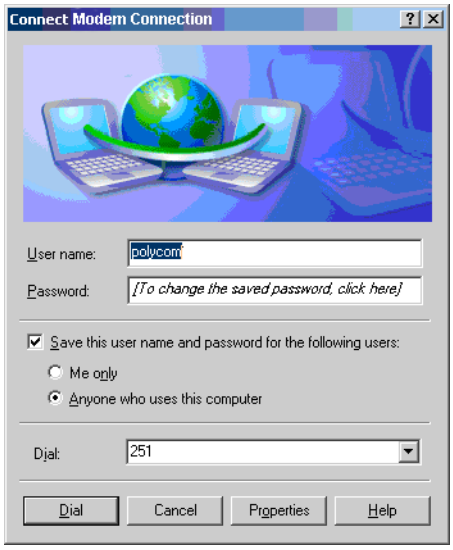
8 In the *Connection Availability* box, select **Anyone's use** and click the **Next** button.



9 In the *Internet Account Information* box, complete the *Username*, *Password* and *Confirm Password* fields and click the **Next** button.

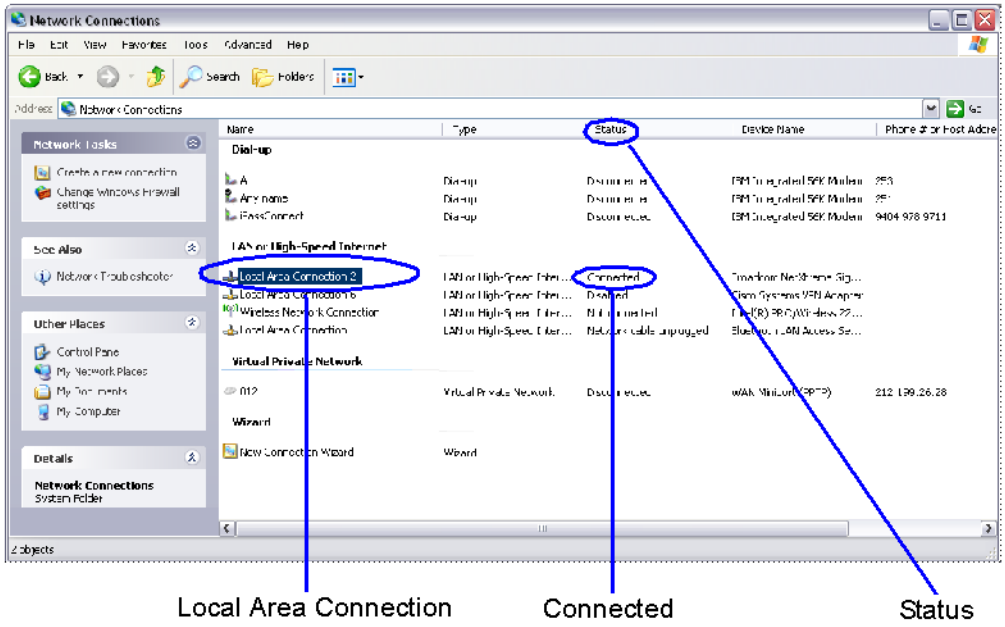


10 The *Connection* applet is displayed with the field values filled in as specified by the *New Connection Wizard*.



11 Click the **Dial** button to establish a connection to *LAN 3 Port* via the modem.

The *Windows – Network Connections* applet displays *Connected* status for the new connection.



Procedure 4: Connect to the Collaboration Server

To Connect using the RMX Manager:

To use the browser:

- In the browser's command line, enter `http://<MCU Control Unit IP Address>/RmxManager.html` and press **Enter**.

To use the Windows Start menu:

1 Click **Start.**

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.**

The *All Programs* list is displayed.

c Select **Polycom** and then select **RMX Manager**.

The *RMX Manager – Welcome* screen is displayed.

Appendix H - Integration Into Microsoft Environments

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 the following Microsoft environments:

- Office Communications Server 2007 environment (Microsoft R2, Wave 13)
- Lync Server 2010 environment (Microsoft Wave 14).

Point-to-point and multipoint audio and video meetings can be initiated from Office Communicator/ 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 Office Communications Server/Lync Server either by:

- *Matched URI dialing* - using the SIP URI address.(both Office Communications Server and Lync Server)
- *Numerical dialing* - enables a common dialing plan for Meeting Rooms across Office Communications Server and H.323 infrastructures (not available in Lync server environment).

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. The table below lists the *SIP TLS Encryption Key* length support for the various system components.

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			

Conferencing Entities Presence

Conferencing entities (Meeting Rooms, Entry Queues and SIP Factories) can be registered with the SIP server (Office Communication Server or Lync server) enabling the addition of these conferencing entities to the buddy list while displaying their presence (availability status: Available, Offline, or Busy). Office Communication Server client or 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.

Multiple Networks

A more complex configuration, in which two Microsoft SIP servers are used (one Lync Server and one Office Communications Server) is also supported using the Collaboration Server Multiple Networks configuration.

In this configuration, each Microsoft SIP Server is defined in a Network Service of its own (in this case two IP Network Services are defined). A DNS server can be specified for each IP Network Service and for the RMX Management Network Service.

Collaboration Server Multiple Networks Topology

Guidelines

- If *ICE* initialization fails in a *Network Service*:
  - The *Network Service* remains functional but without *ICE* capability.
  - *ICE* capability on media cards that share the same *Network Service* also remain functional but without *ICE* capability.
  - Other *Network Services* with *ICE* capability on other media cards are unaffected.
- A *DNS* server can be specified for each *IP Network Service* and for the Collaboration Server *Management Network Service*.

Interactive Connectivity Establishment (ICE)

Interactive Connectivity Establishment (ICE) provides a structure/protocol to unify the various NAT Traversal techniques that are used to cross firewalls.

It enables SIP based endpoints to connect while traversing a variety of firewalls that may exist between the calling endpoint (local) and the MCU or called endpoint (remote). It is the only way for remote Microsoft Office Communicator/Lync users to call into the enterprise without a VPN.

ICE Guidelines

- Collaboration Server ICE implementation complies with Microsoft ICE implementation.
- ICE is available only in IPv4 environment.
- ICE can be implemented in an environment that includes a STUN server and a Relay server (for example, Microsoft AV Edge server).
- The firewall must be UDP enabled.

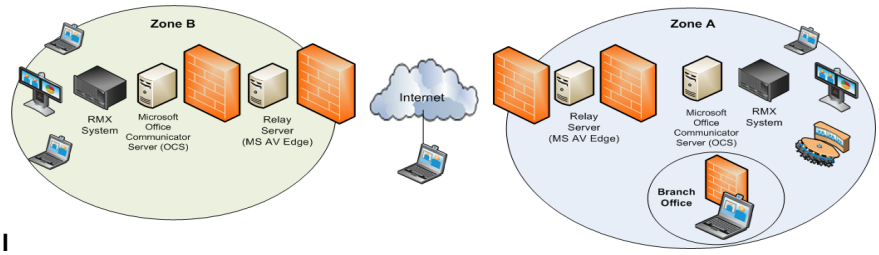
When ICE over UDP is blocked in the firewall UDP port, the ICE connection through the TCP protocol is automatically used instead of UDP for fallback.

- The Collaboration Server must have a unique account in the Active Directory and must be registered with the Office Communications/Lync server.
- ICE is supported with Collaboration Server Multiple Networks.
- Ensure that the Collaboration Server system SIP signaling domain has been allowed on the Lync Server edge server to which you are federating (if your deployment does not include a DMA system).
- Content sharing (BFCP protocol) is not supported in ICE environment.

Connecting to the Collaboration Server in ICE Environment

The dialing methods that can be used by an endpoint to connect to another endpoint depends on the ICE environment: Local, Remote or Federation.

ICE Environmen



I

Local connection - a connection between the Collaboration Server and endpoints that reside within the same organization. For example, an endpoint in Zone A calls the Collaboration Server in Zone A.

Branch Office - a connection between an endpoint that is behind a firewall and the Collaboration Server that reside in the same zone. The user in the Branch Office can also place and receive calls from other enterprises and remote users. For example, Enterprise A also contains a branch office, which in this example is a Polycom HDX user who is behind more than one firewall.

Remote - a connection between Collaboration Server that resides within the organization and an endpoint that resides outside of the organization (on a WAN). For example, an endpoint on the internet that calls the Collaboration Server in Zone A. In such a case, the call has to traverse at least one firewall.

Federation - a connection between Collaboration Server that resides within one organization and an endpoint that resides within another organization. For example, an endpoint in Zone A calls the Collaboration Server in Zone B. The call has to traverse two or more firewalls.

Dialing Methods

The ICE protocol enables remote and federation connections using the registered user name for dialing. The endpoint connects to the Collaboration Server by entering the Collaboration Server registered user name in the following format:

```
[Collaboration Server registered user name]@[OCS/Lync server domain name]
```

For example:

```
rmx111@ilsnd.vsg.local
```

The call reaches the Transit Entry Queue of the Collaboration Server and via IVR is routed to the destination conference.

This method is added to the local connections and *Matched URI* and *Numerical Dialing* methods available in Microsoft Office Communication environment and the *Numerical Dialing* method available in the Lync server environment.

The following table summarizes the dialing methods and its availability in the various configurations.

Available dialing methods per Connection Type

	Matched URI Routing	Numerical Dialing	Registered User Name
Local	✓	✓	✓
Branch office	✓*	×	✓
Remote	✓*	×	✓
Federation	✓*	×	✓

***** To enable the *Matched URI dialing* in the federated environment to be able to connect to the Collaboration Server SIP signaling domain, you must also configure the Office Communications Server/Lync Server.

When federating an Office Communications Server/Lync Edge server with another Office Communications Server/Lync server environment, you need to include the FQDN of the Office Communications Server/Lync Edge server as well as the SIP signaling domain for federated environment. The SIP signaling domain is the FQDN of the Polycom DMA system or a Polycom Collaboration Server system (when your deployment does not include a DMA system).

For example, if company B wants to set up federation with company A and receive and send SIP calls that will be handled by the Polycom SIP signaling domain in company A, you need to add the FQDN of the company A Office Communications Server domain as well as the SIP signaling domain of company A to the list of internal SIP Server domains supported by the company B Office Communications Server/Lync Server environment.

Integrating the Collaboration Server into the Microsoft Office Communications Server Environment

When the Collaboration Server is integrated into the Office Communications Server environment, calls to conferences running on the Collaboration Server can be routed using Matched URI dialing and/or Numerical dialing.

Both routing methods (numerical dialing and Matched URI dialing) can be enabled simultaneously in the Office Communications Server and the Collaboration Server system or you can enable one of these methods, depending on your environment requirements.

In both methods, the Collaboration Server configuration is the same.

Setting the Matched URI Dialing Method

To enable the Matched URI dialing method the following tasks have to be completed:

Office Communications Server side:

1. Set the Static Route & Trusted Host for Collaboration Server in the Office Communications Server.
2. **Optional if Load Balancer Server is present.** Set the Static Route & Trusted Host for Collaboration Server in the Load Balancer server.

Collaboration Server side:

The following tasks are detailed in Configuring the Collaboration Server for Microsoft Integration.

1. Modify the Management Network Service to include the DNS server and set the Transport Type to TLS.
2. Create the security certificate (using one of the two available methods)
3. Define a SIP Network Service in the Collaboration Server and install the TLS certificate.
4. Modify and add the required system flags in the Collaboration Server System Configuration.
5. **Optional.** Defining additional Entry Queues and Meeting Rooms in the Collaboration Server environment. For more information see Defining a New Entry Queue and Creating a New Meeting Room.

For a detailed description of the configuration of the Polycom conferencing components for the integration in Microsoft Office Communications Server 2007 see the *Polycom® HDX and Collaboration Server™ Systems Integration with Microsoft Office Communications Server 2007 Deployment Guide*.

In an ICE environment, to enable the Matched URI dialing in the federated environment to be able to connect to the Collaboration Server SIP signaling domain, you must also configure the Office Communications Server. When federating an Office Communications Server edge server with another Office Communications Server environment, you need to include the FQDN of the Office Communications Server edge server as well as the SIP signaling domain for federated environment. The SIP signaling domain is the FQDN of the Polycom DMA system or a Polycom Collaboration Server system (when your deployment does not include a DMA system).

Note: The RMX does not support working with multiple edge servers.

For example, if company B wants to set up federation with company A and receive and send SIP calls that will be handled by the Polycom SIP signaling domain in company A, you need to add the FQDN of the company A Office Communications Server domain as well as the SIP signaling domain of company A to the list of Internal SIP Server domains supported by the company B Office Communications Server environment.

For more information, see the *Microsoft documentation and the Visual Communications Deployment Administration Guide*.

Configuring the Office Communications Server for Collaboration Server Systems

To be able to work with the Office Communications Server, the Collaboration Server unit must be configured as a Trusted Host in the OCS. This is done by defining the IP address of the signaling host of each Collaboration Server unit as Trusted Host.

Meeting Rooms are usually not registered to the OCS, and Static Routes are used instead. Setting Static Routes in the OCS enables SIP entities / UAs to connect to conferences without explicit registration of conferences with the OCS.

Routing is performed by the OCS based on the comparison between the received URI and the provisioned static route pattern. If a match is found, the request is forwarded to the next hop according to the defined hop's address.

This is the recommended working method. It alleviates the need to create a user account in the OCS for each Meeting Room and Entry Queue. This also allows users to join ongoing conferences hosted on the MCU without registering all these conferences with OCS.

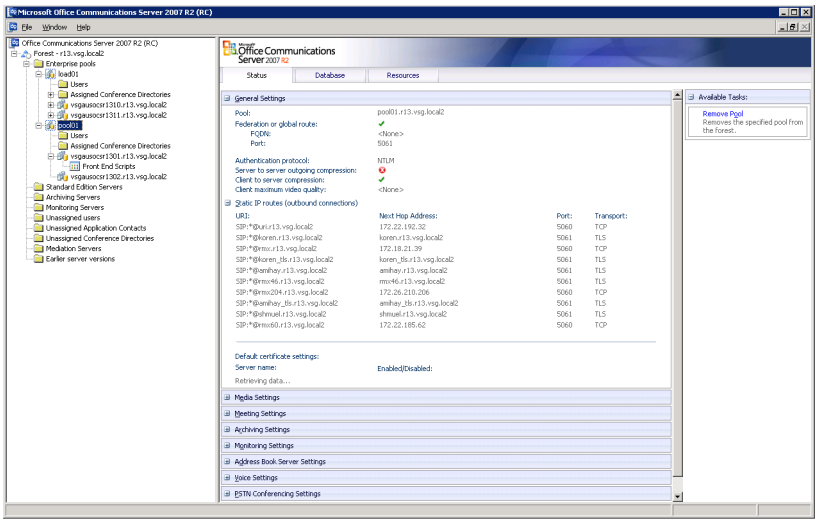
Entry Queues can also be for Ad-hoc conferencing enabling Office Communicator clients to dial to the Entry Queue and create a new ongoing conference using DTMF codes to enter the target conference ID. In such a case, other OC users will have to use that ID to join the newly created conference.

Setting the Trusted Host for Collaboration Server in the Office Communications Server

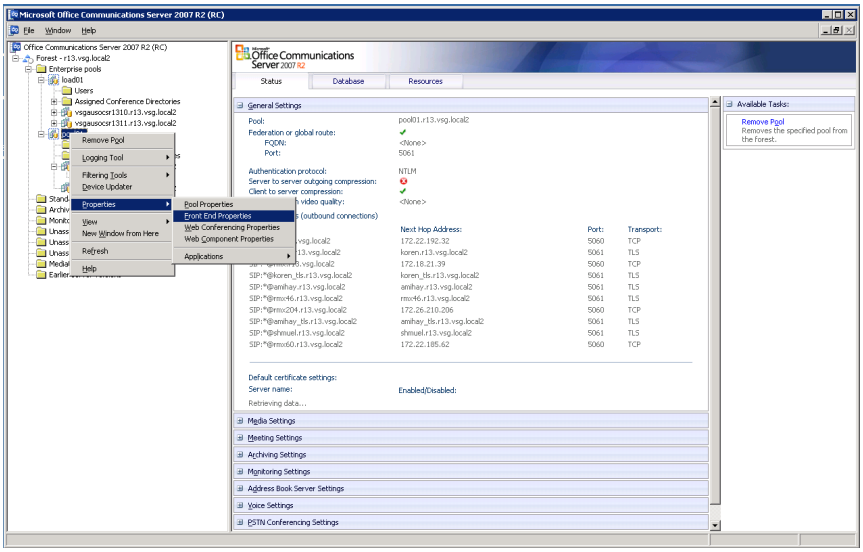
To set the Collaboration Server as trusted in OCS:

1. Open the OCS Management application.

2 Expand the *Enterprise Pools* list.

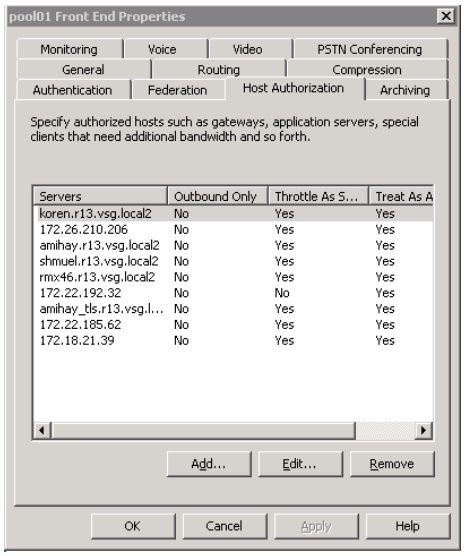


3 Right-click the *server pool* icon, click **Properties > Front End Properties**.

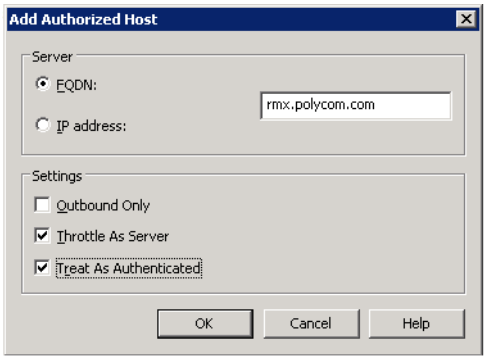


The *Pool Front End Properties* dialog box opens.

4 Click the **Host Authorization** tab.



5 Click the **Add** button to add the Collaboration Server as trusted host.
The *Add Authorized Host* dialog box opens.



6 In the *Add Authorized Host* dialog box, enter the Collaboration Server *FQDN* name as defined in the DNS and will be used in the Static Routes definition.

7 In the *Settings* section, select the **Throttle as Server** and **Treat As Authenticated** check boxes.

8 Click **OK**.
The defined Collaboration Server is displayed in the trusted servers list in the server *Front End Properties—Host Authorization* dialog box.

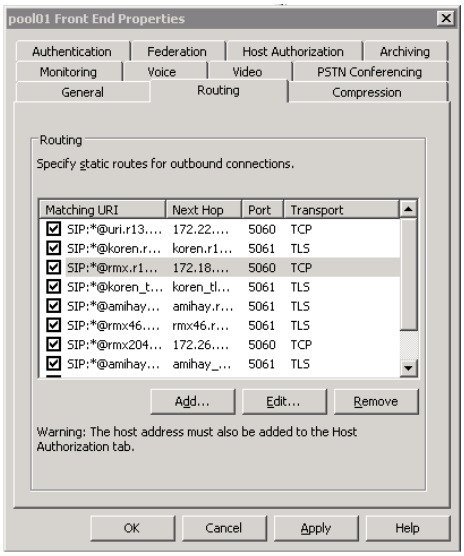
If routing between the Collaboration Server and the OCS using Static Routes is required, do not close this dialog box, and continue with the following procedure. If you do not want to define Static Routes, click OK to close this dialog box.

Setting the Static Route for Collaboration Server in the OCS

To add Collaboration Server to the Routing Roles:

1 In the *Front End Properties* dialog box, click the **Routing** tab.

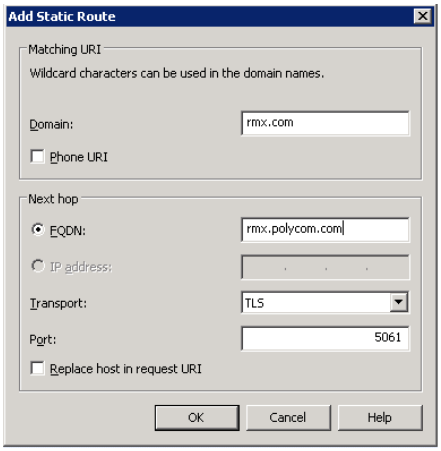
2 Click the **Add** button.



The *Add Static Routes* dialog box opens.

3 In the *Matching URI* section, enter the *Domain* name for the Collaboration Server. Any domain name can be used.

4 In the *Next hop* section enter the Collaboration Server *FQDN* name as defined in the DNS and is used in the *Host Authorization* definition.



5 In the *Transport* field, select **TLS** to enable the dial-out from conferences to SIP endpoints.

6 Click **OK**.
The new Route is added to the list of routes in the *Front End Properties—Routes* dialog box.

7 Click **OK**.

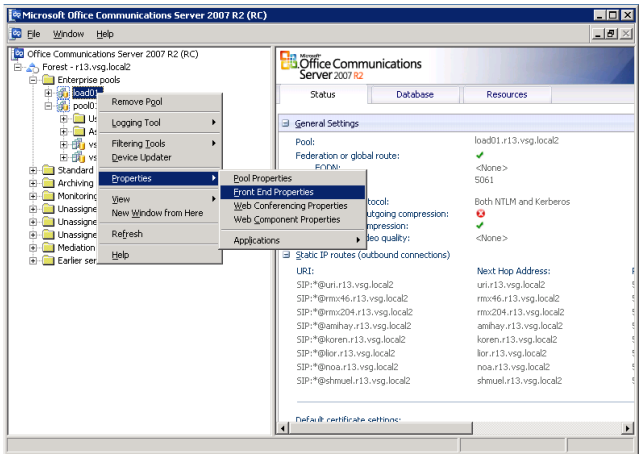
Setting the Static Route & Trusted Host for Collaboration Server in the Load Balancer Server (Optional)

If your network includes a Load Balancer server, the Collaboration Server unit must be configured as a trusted host in the Load Balancer server in the same way it is configured in the OCS. In addition, Static Routes must also be defined in the Load Balancer server in the same way it is configured in the OCS,

however, the Load Balancer should be pointed to the OCS pool and not to the Collaboration Server directly. This configuration procedure is done in addition to the configuration in the OCS.

To set the Collaboration Server as trusted and define Static routes in the Load Balancer Server:

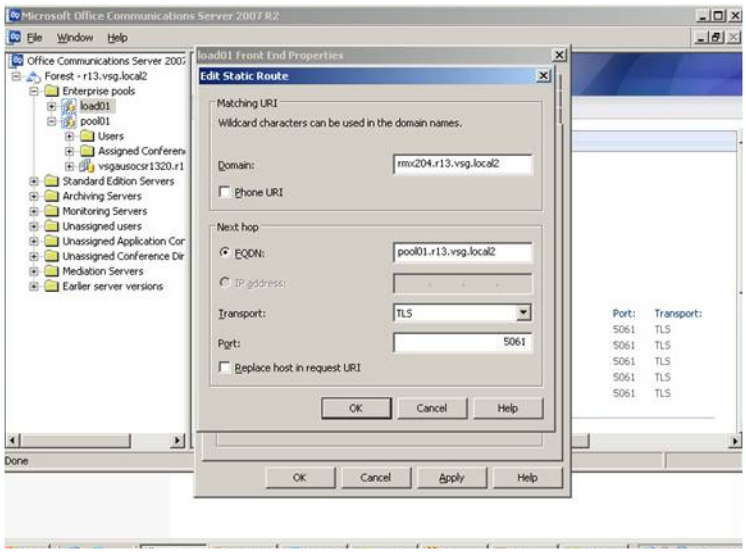
1. Open the OCS Management application.
2. Expand the *Enterprise Pools* list.
3. Right-click the *Load* icon, click **Properties > Front End Properties**.



The *Load Front End Properties* dialog box opens.

The definition procedure is the same as for setting the Collaboration Server as trusted and define Static routes in the OCS.

For details, see Setting the Trusted Host for Collaboration Server in the Office Communications Server.



Make sure that when defining the Static Route it is pointing to the OCS pool and not to the Collaboration Server directly.

Configuring the Collaboration Server System

The required tasks are detailed in Configuring the Collaboration Server for Microsoft Integration.

Dialing to an Entry Queue, Meeting Room or Conference Using the Matched URI Method

Once the Collaboration Server is configured for integration in the OCS environment, the preferred dialing mode to the conferencing entities such as Meeting Rooms, conferences and Entry Queues is direct dial in using the domain name defined in the OCS Static Routes. This eliminates the need to register the conferencing entities with the SIP server and to define a separate user for each conferencing entity in the Active Directory.

In such a case, after the first dial in, the conferencing entity will appear in the OC client directory for future use.

To dial in directly to a conference or Entry Queue:

Enter the conferencing entity SIP URI in the format:
conferencing entity routing name@domain name

The domain name is identical to the domain name defined in the OCS Static Routes.

For example, if the domain name defined in the OCS static routes is lcs2007.polycom.com and the Routing Name of the Meeting Room is 4567, the participant enters 4567@lcs2007.polycom.com.

Another dialing method is to register the Entry Queues with the SIP Server and create a user for each Entry Queue in the Active Directory. In such a case, OC clients can select the Entry Queue from the Contacts list and dial to the Entry Queue.

Setting the Numerical Dialing Method

The Collaboration Server can be configured as a Voice Gateway in the OCS environment, enabling dialing in to meeting rooms using numbers instead of or in addition to SIP URI addresses which are long strings.

In such configuration, HDX or MOC users dial a number rather than a full SIP URI, simplifying the dialing, which is especially beneficial with the HDX remote control.

Such configuration also enables a common dialing plan for meeting rooms across OCS and H.323 infrastructures. In an integrated environment that also includes Microsoft Exchange Server and Polycom Conferencing Add-in for Microsoft Outlook, a single number can be inserted into a calendar invitation and it will be valid for OC client endpoints and H.323 endpoints.

This dialing method can be configured in parallel to the matching URI dialing method (using Static Routes).

Setting the Numerical Dialing for Collaboration Server Meeting Rooms

The following processes are required to set up the numerical dialing for the Collaboration Server Meeting Rooms in the OCS infrastructure:

OCS side:

- Configuring the Collaboration Server as a Routable Gateway - The Collaboration Server (or DMA) must be set as a trusted voice gateway in the OCS infrastructure. This does not restrict Collaboration Server to just voice operation, rather it means that the Collaboration Server (or DMA) can be set as a destination for a voice route using the OCS management console.
  Setting the Collaboration Server as a trusted voice gateway also enables it to be used as a trusted gateway for static routes using URI matching.
- Establishing a Voice Route to the Collaboration Server “Voice” Gateway - The Voice Route to the Collaboration Server (or DMA) must be configured in the OCS infrastructure.

If the Collaboration Server was previously defined as a Trusted Host for matching URI dialing method, this definition must be removed before configuring the Collaboration Server as a voice gateway. It will be defined as trusted host as part of the voice gateway configuration. For more details, see Optional. Removing the Collaboration Server from the Host Authorization List.

- Configure Office Communicator Users for Enterprise Voice.

Collaboration Server side:

The following tasks are detailed in Configuring the Collaboration Server for Microsoft Integration.

1. Modify the Management Network Service to include the DNS server and set the Transport Type to TLS.
2. Create the security certificate (using one of the two available methods)
3. Define a SIP Network Service in the Collaboration Server and install the TLS certificate.
4. Modify and add the required system flags in the Collaboration Server System Configuration.
5. **Optional.** Defining additional Entry Queues and Meeting Rooms in the Collaboration Server environment. For details see Meeting Rooms and Entry Queues.

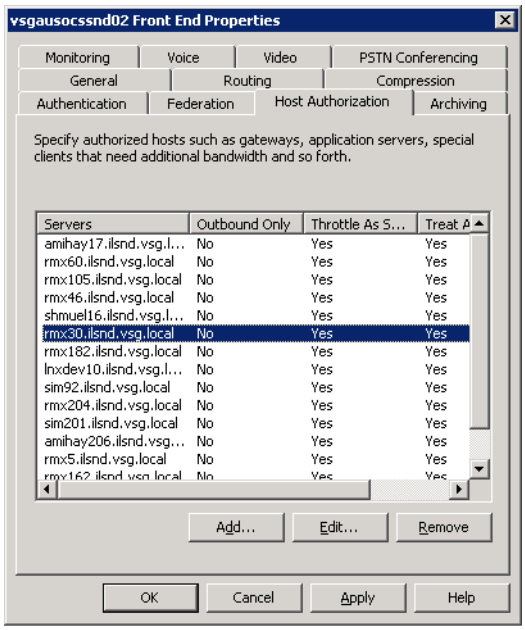
For a detailed description of the configuration of the Polycom conferencing components for the integration in Microsoft Office Communications Server 2007 see the Polycom® HDX and Collaboration Server™ Systems Integration with Microsoft Office Communications Server 2007 Deployment Guide.

Optional. Removing the Collaboration Server from the Host Authorization List

If you have defined the Collaboration Server as Trusted Host to enable dialing using the Static Routes and you want to use numerical dialing in addition or instead of SIP URI dialing, you need to remove the current definition of the Collaboration Server and redefine it as a voice gateway.

To remove the definition of the Collaboration Server as trusted host from the Front End Properties:

1. In the OCS application, display the *Front End Properties* (right-click the Front End and select Properties).
2. Click the **Host Authorization** tab.
3. In the *Trusted Hosts* list, click the Collaboration Server entry and then click the **Remove** button.



4 Click **OK**.

Configuring the Collaboration Server as a Routable Gateway

The Collaboration Server must be set as a routable voice gateway in the Office Communications Server infrastructure. This does not restrict the Collaboration Server to just voice operation, rather it means that the Collaboration Server can be set as a destination for a voice route in the Office Communications Server infrastructure.

The Office Communications Server infrastructure uses the WMI class `MSFT_SIPTrustedAddInServiceSetting` to store information for each voice gateway in the infrastructure. Typically, these gateways are Office Communications Server Mediation Servers, but in this case, the Collaboration Server is set as a voice gateway by creating a new instance of the class `MSFT_SIPTrustedAddInServiceSetting`.

Polycom recommends using the Office Communications Server 2007 R2 Resource Kit Tools to accomplish this.

To set up the Collaboration Server/DMA as a Voice Gateway:

1 Download and install the Office Communications Server 2007 R2 Resource Kit Tools from the following URL:

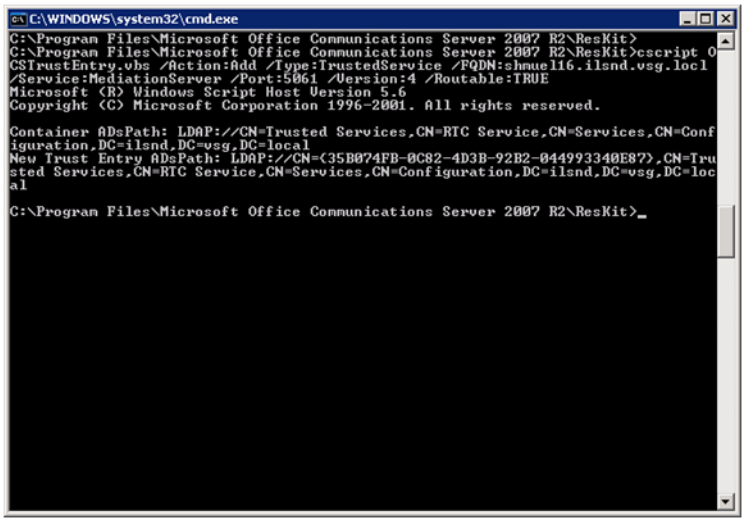
http://www.microsoft.com/downloads/details.aspx?familyid=9E79A236-C0DF-4A72-ABA6-9A9602A93ED0&displaylang=en

2 Open a command prompt and navigate to where you installed the resource kit. For example, `C:\Program Files\Microsoft Office Communications Server 2007 R2\ResKit\`.

3 Run the following command:

```
cscript OCSTrustEntry.vbs /action:add /type:trustedservice /fqdn:<your
FQDN> /service:MediationServer /port:5061 /version:4 /routable:TRUE
```

Where *<your FQDN>* is the FQDN of your Collaboration Server system. The script automatically generates the GUID discover the proper Active Directory container to store the object.



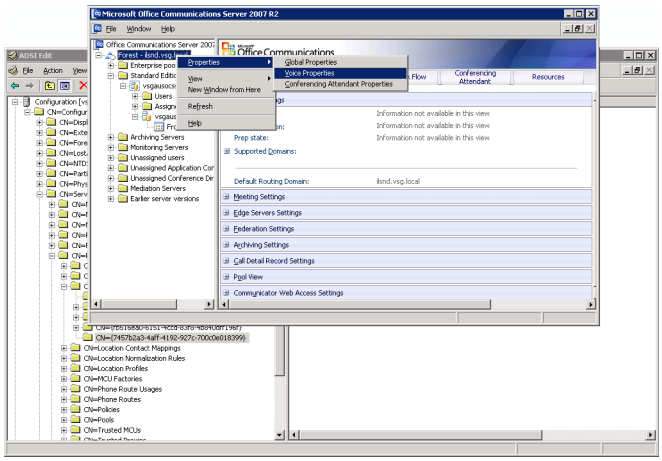
Your Collaboration Server system is now established as a trusted gateway by all Office Communications Server pools in the domain. It is displayed in the list of voice gateways when you establish a voice route.

Establishing a Voice Route to the Collaboration Server “Voice” Gateway

The OCS infrastructure enables you to establish a voice route to a voice gateway. Typically, this means that all SIP INVITEs to phone numbers which match a particular pattern will be routed to a specific gateway. In this example, all INVITEs to numbers which start with “11” will be routed to Collaboration Server11 (DNS name rmx11.r13.vsg.local2).

To establish the voice route:

1 Open the OCS R2 management Console and right click on **Forest** and then click **Properties > Voice Properties**.

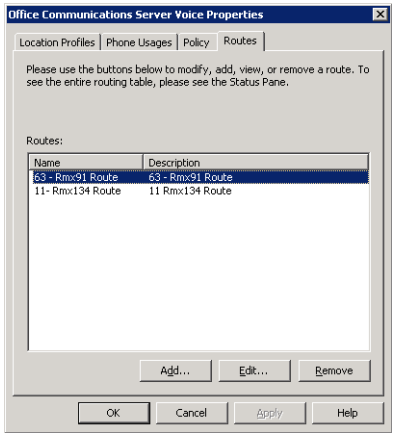


The *Office Communications Server Voice Properties* dialog box opens.

2 Click the **Routes** tab.

Office Communications Server Voice Properties - Routes dialog box opens.

3 Click the **Add** button.

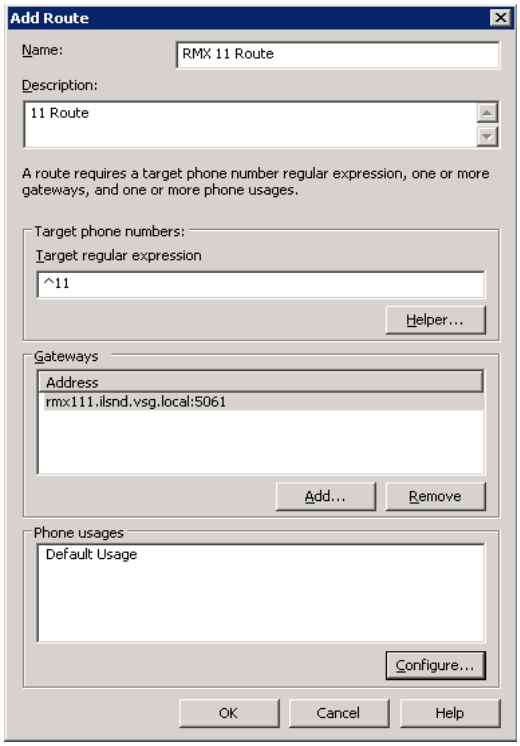


The *Add Route* dialog box opens.

4 In the *Name* field, enter a name that will identify this voice route.

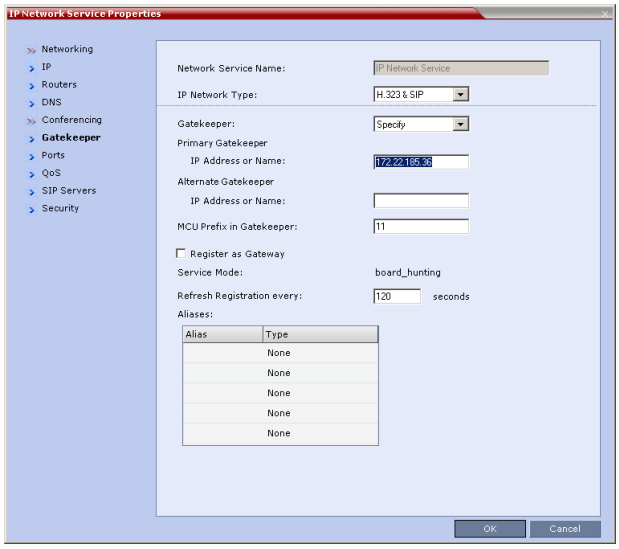
5 Optional. In the *Description* field, enter a description.

6 In the *Target Regular Expression* field enter **^** and the MCU prefix as defined in the gatekeeper. This prefix is also defined in the *Collaboration Server IP Network Service*.



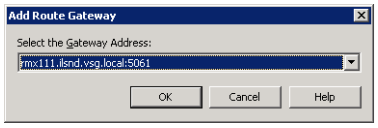
For example, if 11 is the Collaboration Server prefix defined in gatekeeper, enter **^11**. The circumflex expression "^11" causes this route to be applied to all numbers starting with "11".

If you have not defined such a prefix in the IP Network Service in the Collaboration Server configuration, you can add it later, using value entered here.



7 In the *Gateways - Addresses* box, click the **Add** button.

The *Add Route Gateway* dialog box opens.

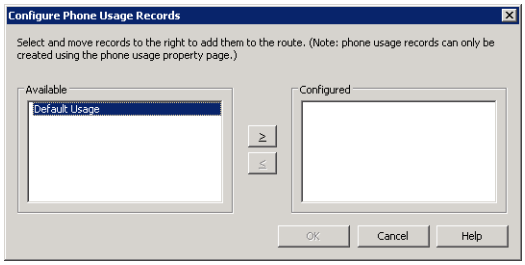


8 Select the Collaboration Server gateway address that was set up in Configuring the Collaboration Server as a Routable Gateway that is displayed in the drop down list of gateways.

9 Click **OK** to save the address and return to the *Add Route* dialog box.

10 In the *Phone Usage* box, click the **Configure** button.
The *Configure Phone Usage Records* dialog box opens.

11 In the *Available* box, click **Default Usage** and then click the **>** button.



The *Default Usage* option is displayed in the *Configured* box.

12 Click **OK**.

13 In the *Add Route* dialog box, click **OK** to save the new route.

Configuring Office Communicator Users for Enterprise Voice

Each of the endpoints in the OCS environment must be set to use the voice route.

The setting is done in the Office Communications Server management console for all required users (endpoints) simultaneously or in the Active Directory for each of the Users (endpoints).

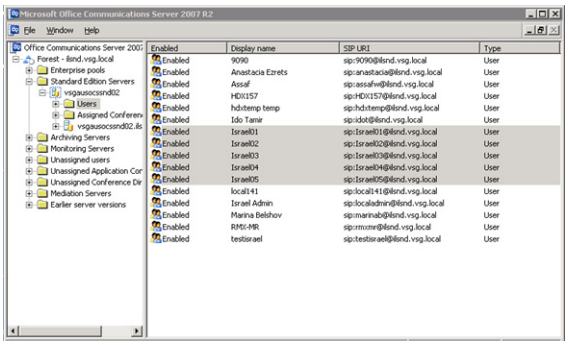
To Configure Office Communicator Users for Enterprise Voice in the Office Communications Server management console:

1 Navigate to **Start > All Programs > Administrative Tools > Office Communications Server 2007 R2** to open the Office Communications Server management console.

2 Expand the Enterprise pool or Standard Edition server node where your users reside.

3 Expand the pool or server where your users reside, and then click the **Users** node.

4 In the right pane, right-click one or more users whom you want to configure, and then select **Configure users**.



The *Welcome to the Configure Users Wizard* opens.

5 On the *Welcome to the Configure Users Wizard* dialog box, click **Next**.

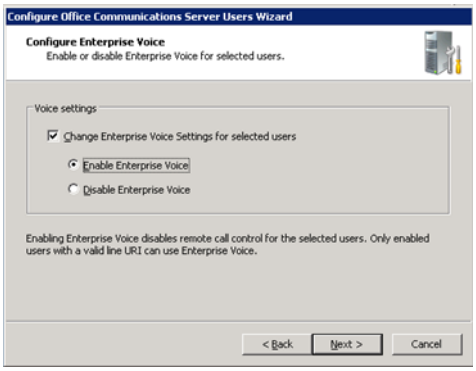
6 On the *Configure User Settings* dialog box, click **Next**.

7 On the *Configure Meeting Settings* dialog box, click **Next**.

8 On the *Configure User Settings specify meeting policy* dialog box, click **Next**.

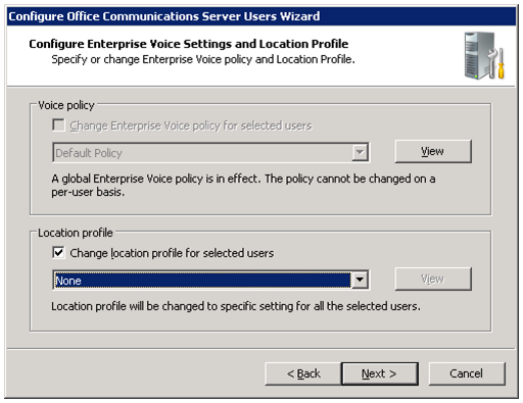
9 On the *Configure Enterprise Voice* dialog box, select **Change Enterprise Voice Settings for selected users**, and then click **Enable Enterprise Voice**.

10 Click **Next**.



11 On the *Configure Enterprise Voice Settings and Location Profile* dialog box, select **Change Enterprise Voice Policy for selected users**.

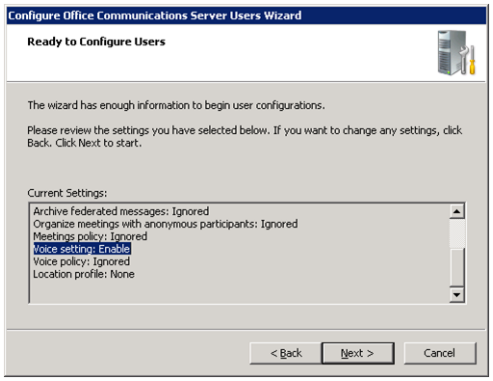
12 Select an Enterprise Voice policy from the list.



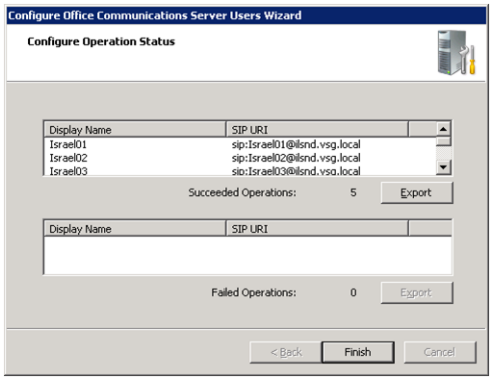
13 Select **Change location profile** for selected users.

14 Select a location profile from the list, and then click **Next**.

15 On the *Ready to Configure Users* dialog box, review the settings, and then click **Next**.



16 On the *Configure Operation Status* dialog box, verify that the operation succeeded, and then click **Finish**.

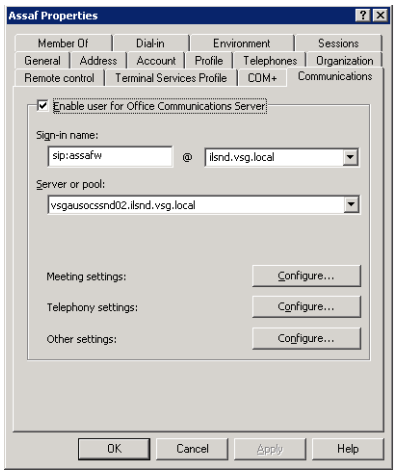


To Configure Office Communicator Users for Enterprise Voice in the in the Active Directory:

1 Open the *Active Directory* and navigate to the endpoint whose properties require changing.

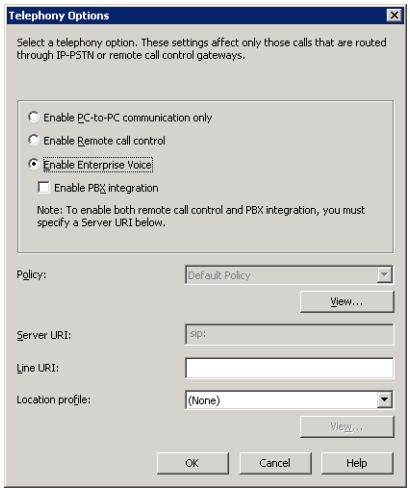
2 Right-click the endpoint and select **Properties**.
The *Properties* dialog box opens.

3 Click the **Communications** tab.



4 Click the **Telephony Settings** - **Configure** button.
The *Telephony Options* dialog box opens.

5 Select the **Enable Enterprise Voice** option.

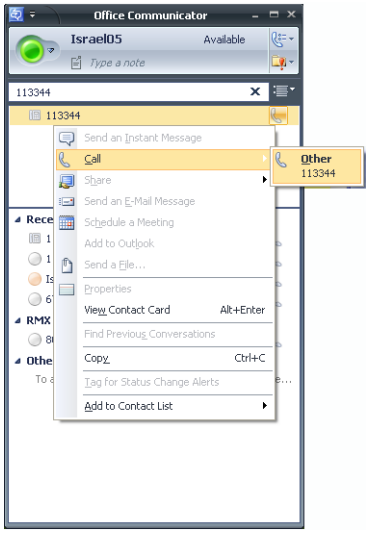


6 Click **OK** to return to the *Properties* - *Communications* dialog box**.**

7 Click **OK**.

Starting a Conferencing Call from the MOC

1 In the Office Communicator application, enter the number to dial, for example, 113344. This number is composed of the Collaboration Server Prefix in the Gatekeeper (for example, 11) and the Meeting Room ID, as defined on the Collaboration Server (for example, 3344).



2 Click **Call**, and then click **Other**.
The call is routed to the Meeting Room on the Collaboration Server, and the caller that initiated the call connects as the conference chairperson.

3 The MOC User can then add video to the call, by selecting **Add Video** in the *Office Communicator* window.

Setting Simultaneous Numerical Dialing and Matched URI Routing

You can simultaneously set up an Collaboration Server for both numerical and Matched URI dialing. If you want to do this, follow these instructions:

1 Set the Collaboration Server as a trusted service (MediationServer) and a voice gateway using the instructions in Setting the Numerical Dialing Method.

2 Set up a matching URI route to the Collaboration Server/DMA by right-clicking the **OCS Pool**, selecting **Properties > Front End Properties > Routing Tab** and follow the instructions in Setting the Static Route for Collaboration Server in the OCS.

- When defining both routing methods, you **cannot** add an Collaboration Server as an Authorized Host using the **Front End Properties > Host Authorization** tab. There can only be one trusted service entry for the Collaboration Server even though there are two different routes to the Collaboration Server (i.e., Matched URI and numerical dialing). If the Matched URI routing method was previously defined and the Collaboration Server was set as trusted host, and you are adding the numerical dialing method, you have to remove the Collaboration Server from the Trusted Hosts list. For more details, see Optional. Removing the Collaboration Server from the Host Authorization List.
- Only TLS connections to the Collaboration Server will work, TCP connections will not work.

PFX Method - Creating the Security (TLS) Certificate in the OCS and Exporting the Certificate to the Collaboration Server Workstation

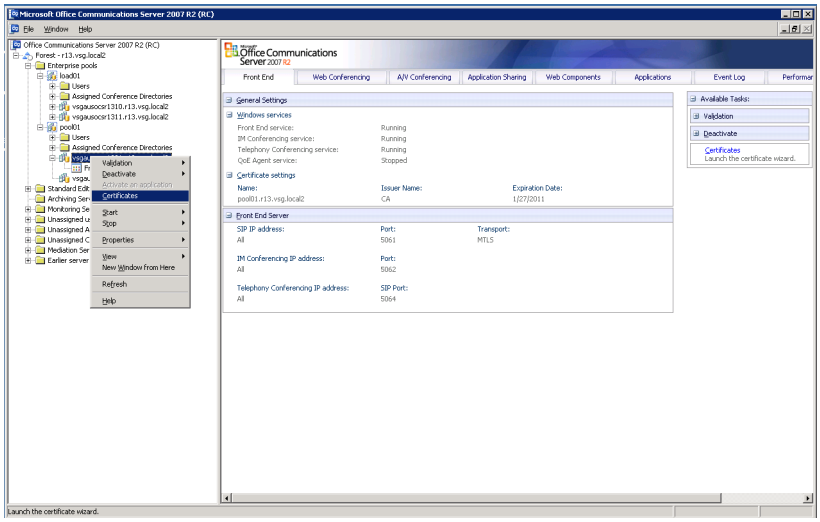
If you are using the PFX method to create and send the security certificate to the Collaboration Server, certificate files *rootCA.pem*, *pkey.pem* and *cert.pem* must be sent to the Collaboration Server unit. These files can be created and sent to the Collaboration Server in two methods:

- The files *rootCA.pem*, *pkey.pem* and *cert.pem* are provided by a Certificate Authority and are sent independently or together with a password file to the Collaboration Server. This is the recommended method.
- Alternatively, the TLS certificate files are created internally in the OCS and exported to the Collaboration Server workstation from where the files can be downloaded to the Collaboration Server. If the certificate is created internally by the OCS, one **.pfx* file is created. In addition, a text file containing the password that was used during the creation of the **.pfx* file is manually created. Both files can then be sent from the Collaboration Server workstation to the Collaboration Server unit. When the files are sent to the Collaboration Server, the **.pfx* file is converted into three certificate files: *rootCA.pem*, *pkey.pem* and *cert.pem*.
  Sometimes, the system fails to read the *.pfx file and the conversion process fails. Resending *.pfx file again and then resetting the system may resolve the problem.

The following procedure describes how to create the *.PFX file in the OCS and export it so it can be sent to the Certificate Authority or to the Collaboration Server.

To create the TLS certificate in the Office Communications Server:

1. In the OCS *Enterprise Pools* tree, expand the Pools list and the *server pool* list.
2. Right-click the pool *Front End* entity, and click **Certificate**.



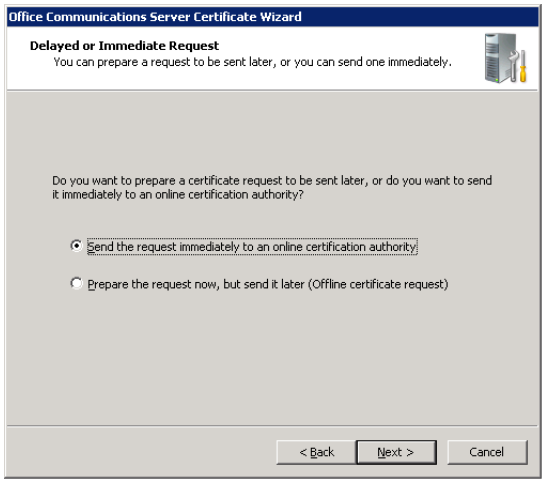
The *Office Communicator Server Wizard Welcome* window is displayed.

3. Click **Next**.
   The *Available Certificate Tasks* window is displayed.

4 Select **Create a New Certificate** and click **Next**.

The *Delayed or Immediate Request* window is displayed.

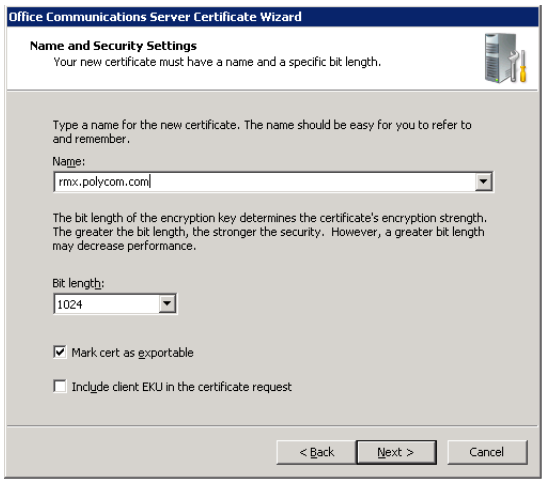
5 Select **Send the Request immediately to an online certificate authority** and click **Next**.



The *Name and Security Settings* window is displayed.

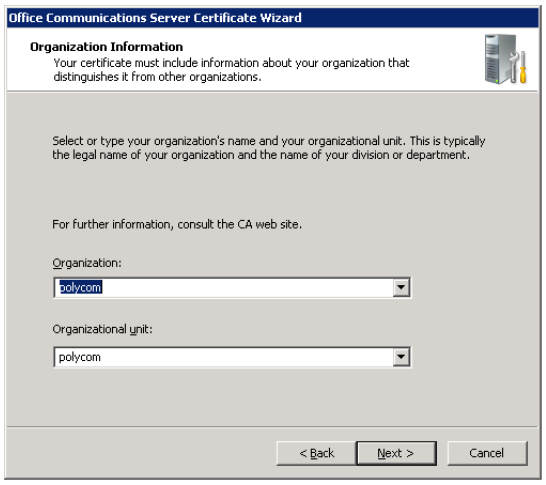
6 In the *Name* field, select the Collaboration Server name you entered in the *FQDN* field when defining the trusted host or as defined in the DNS server.

7 Select the **Mark cert as exportable** check box.



8 Click **Next**.
The *Organization Information* window is displayed.

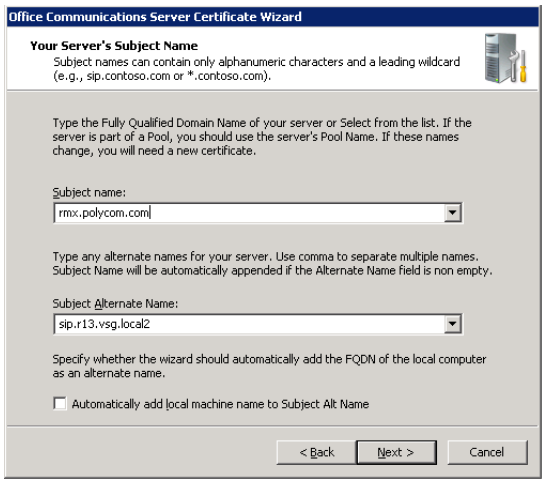
9 Enter the name of the *Organization* and the *Organization Unit* and click **Next**.



Your Server's Subject Name window is displayed.

10 In the *Subject name* field, select the *FQDN* name of the Collaboration Server from the list or enter its name.

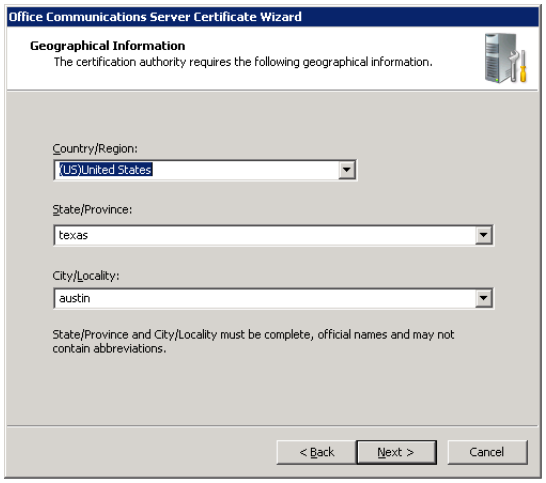
11 Keep the default selection in the *Subject alternate name* field and click **Next**.



12 If an error message is displayed, click **Yes** to continue.

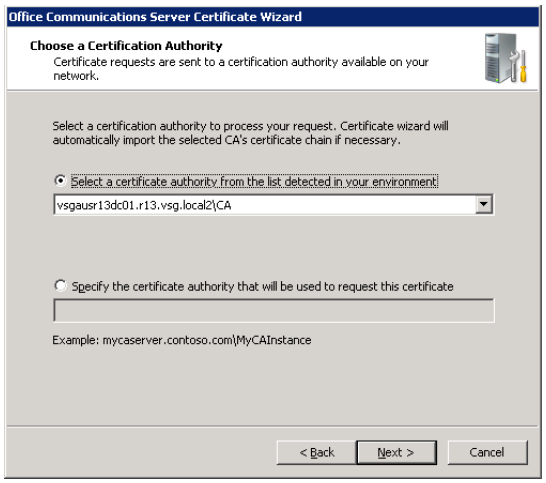
The *Geographical Information* window is displayed.

13 Enter the geographical information as required and click **Next**.



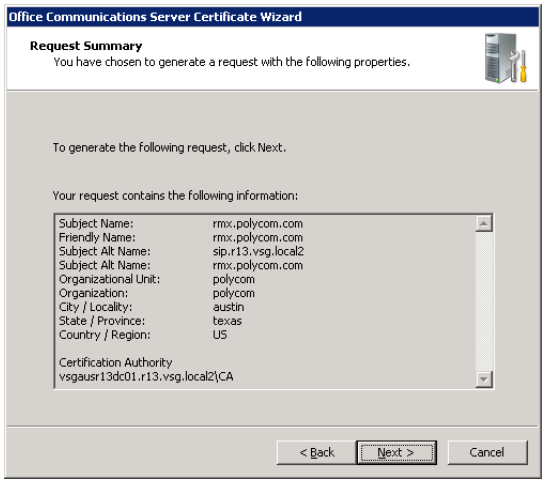
The *Choose a Certification Authority* window is displayed.

14 Ensure that the **Select a certificate authority from the list detected in your environment** option is selected and that the local OCS front end entity is selected.



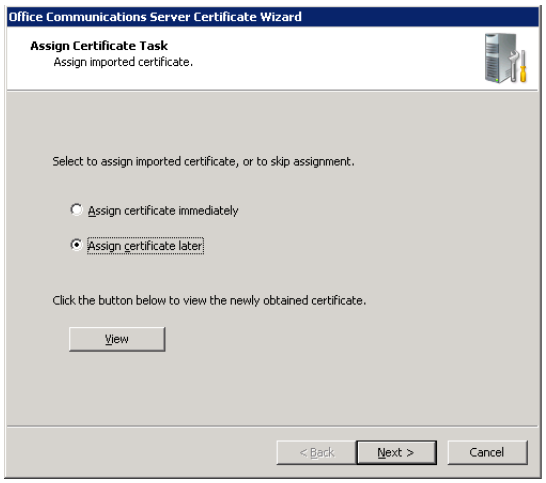
15 Click **Next**.
The *Request Summary* window is displayed.

16 Click **Next** to confirm the listed parameters and create the requested certificate.



The *Assign Certificate Task* window is displayed.

17 Select **Assign certificate later** and click **Next** (MS R2).



The *Certificate Wizard Completed* window is displayed (MS R2).

18 Click **Finish** (MS R2).

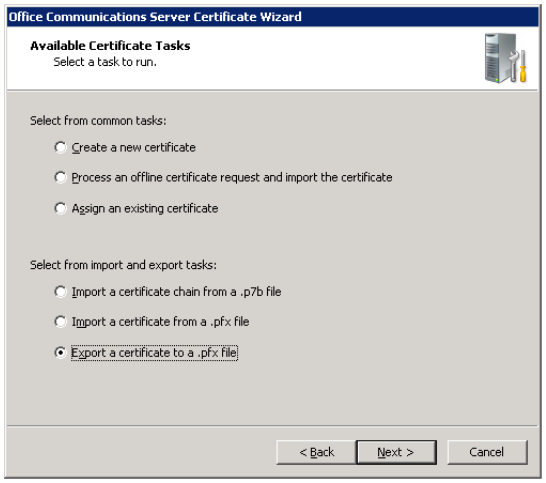
Retrieving the Certificate from the OCS to be sent to the Collaboration Server Workstation

1 In the OCS *Enterprise Pools* tree, expand the *Pools* list and the *Server Pool* list.

2 Right-click the *pool Front End* entity, and select **Certificate**.

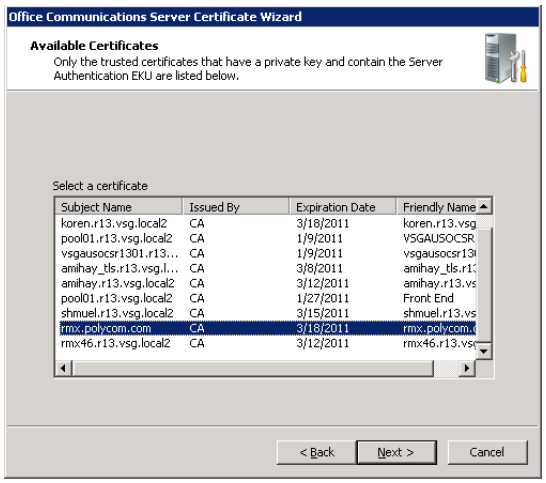
The *Available Certificate Tasks* window is displayed.

3 Select **Export a certificate to a *.pfx file** and click **Next**.



The *Available Certificates* window is displayed.

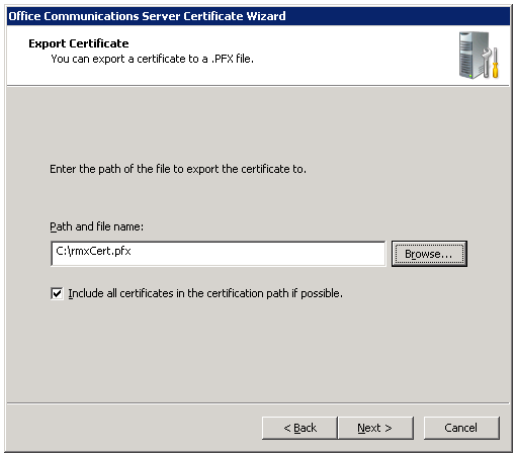
4 Select the certificate *Subject Name* of the Collaboration Server and click **Next.**



The *Export Certificate* window is displayed.

5 Enter the path and file name of the certificate file to be exported or click the **Browse** button to select the path from the list.

The new file type must be ***.pfx** and its name must include the **.pfx** extension.

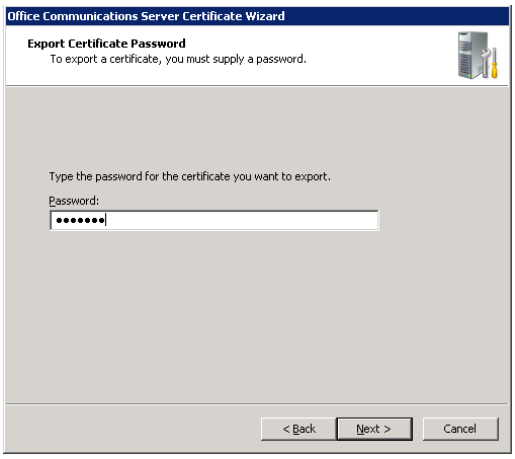


6 Select the **Include all certificates in the certification path if possible** check box and then click **Next**.

The *Export Certificate Password* window is displayed.

7 If required, enter any password. For example, *Polycom*.

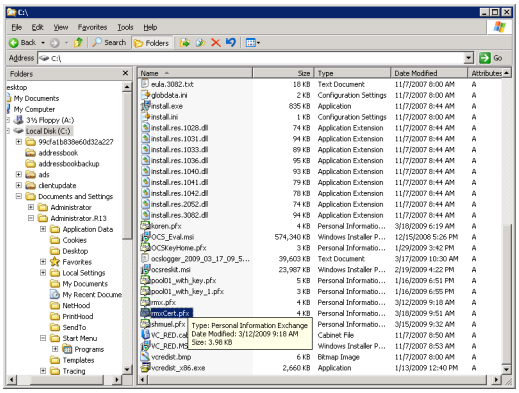
8 Write down this password as you will have to manually create a password file in which this password will be displayed.



Click **Next**.
The *Certificate Wizard Completed* window is displayed.

9 Click **Finish**.

The created **.pfx* file is added in the selected folder.



Optional. Creating the Certificate Password File (certPassword.txt)

If you have used a password when creating the certificate file (**.pfx*), you must create a **certPassword.txt** file. This file will be sent to the Collaboration Server together with the **.pfx* file.

To create the certPassword.txt file:

1 Using a text editor application, create a new file.

2 Type the password as you have entered when creating the certificate file. For example, enter *Polycom*.

3 Save the file naming it **certPassword.txt** (file name must be exactly as show, the Collaboration Server is case sensitive).

Supporting Remote and Federated Users in Office Communications Server ICE Environment

To enable the remote and Federation connections the following operations must be performed:

- Create an Active Directory account for the Collaboration Server that will be used for registering and operating in the MS ICE environment
- Enable the Collaboration Server User Account for Office Communication Server
- Configure the Collaboration Server for ICE dialing for more details, see Configuring the Collaboration Server for Federated (ICE) Dialing.

To place federated calls between Domain A and Domain B in ICE environment sub domains must be federated to the main domain or the Collaboration Server system must be installed on a main domain.

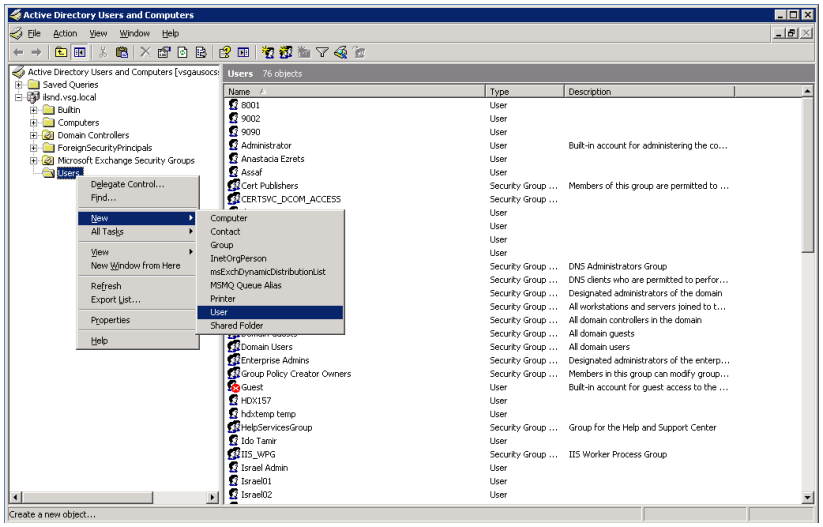
The Collaboration Server can also be set for Matched URI Routing and/or Numerical Dialing to Meeting Rooms. For more details, see Setting the Matched URI Dialing Method and Setting the Numerical Dialing Method.

Creating an Active Directory Account for the Collaboration Server

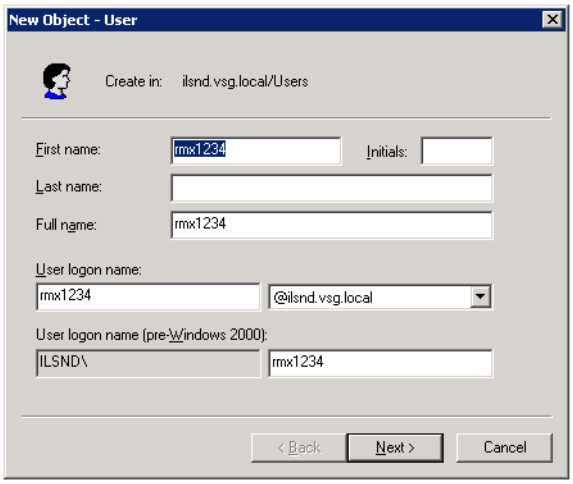
The User account created for the Collaboration Server is used for registration in the Office Communication Server and to automatically synchronize with the STUN and relay (Edge) servers.

To add the Collaboration Server 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**.



3 In the *New User* wizard, define the following parameters:

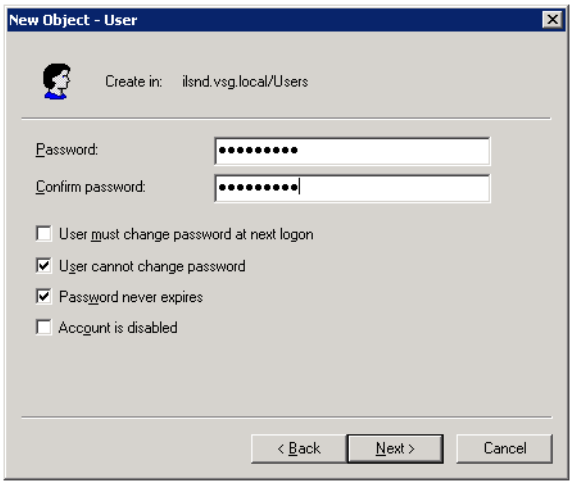


Active Directory - New User Parameters for the Collaboration Server

Field	Description
First Name	Enter the name for the Collaboration Server user. This name will be used in the configuration of the ICE environment in the Collaboration Server.
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.

4 Click **Next**.

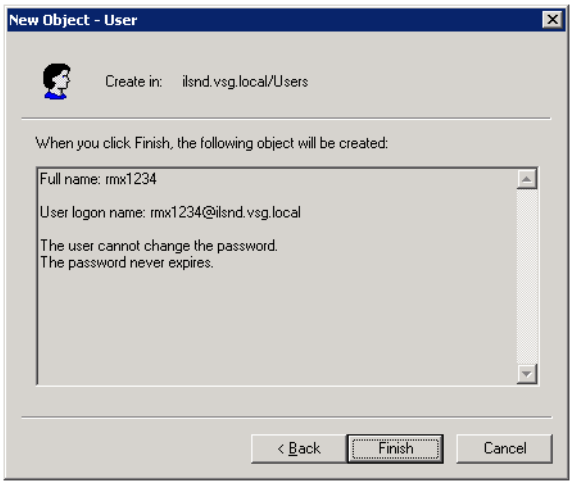
5 Enter the password that complies with the Active Directory conventions and confirm the password.



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**.

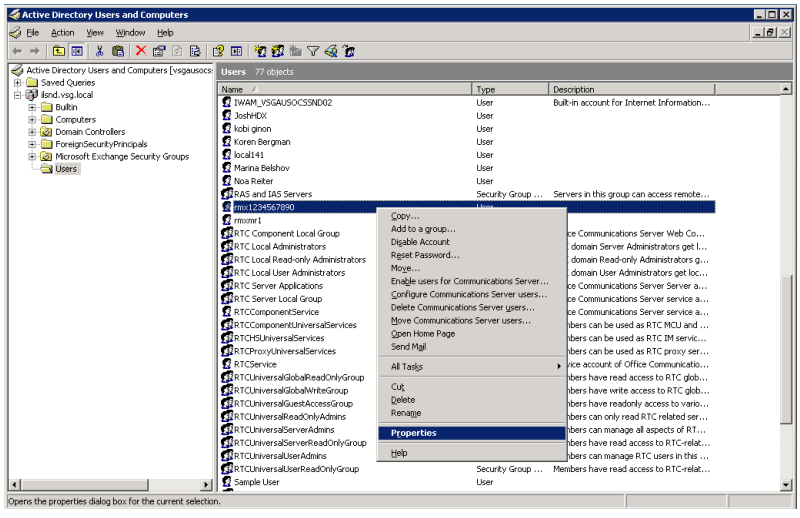
The new User is added to the Active Directory *Users* list.

Enabling the Collaboration Server User Account for Office Communication Server

The new Collaboration Server user must be enabled for registration with the Office Communications Server.

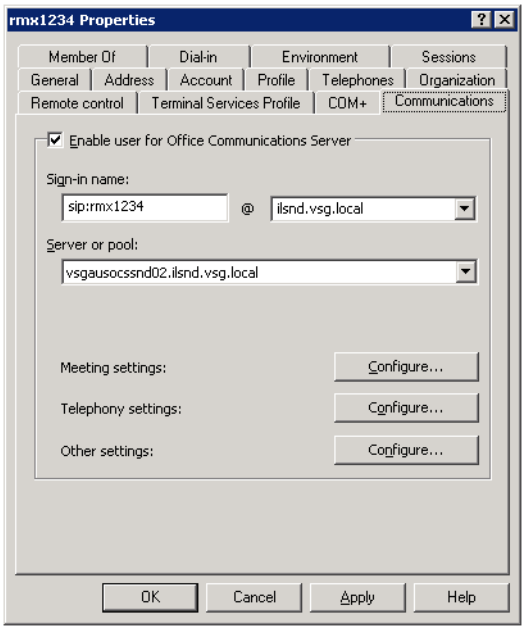
To enable the Collaboration Server User Account for Office Communication Server:

1 In the *Active Directory Users and Computers* window, right-click the Collaboration Server user and then click **Properties**.



2 In the *Properties* dialog box, click the **Communications** tab.

3 In the *Sign in name* field, enter the Collaboration Server user name in the format **SIP:rmx user name** (for example sip:rmx1234) and select the domain name (for example, ilsnd.vsg.local) as entered in the *New User* dialog box.



4 Select the *Server or Pool* from the list.

5 Click **Apply** and then **OK**.

Configure the Collaboration Server for ICE dialing

For details, see Configuring the Collaboration Server for Federated (ICE) Dialing.

Registering the Collaboration Server as a Trusted Application for Lync 2010/2013

The following procedures are mandatory to register the Collaboration Server to Lync 2010/2013.

In the following procedures Domain Names, Server Names and Collaboration Server Names are examples for syntax purposes only and must be adapted to local network requirements.

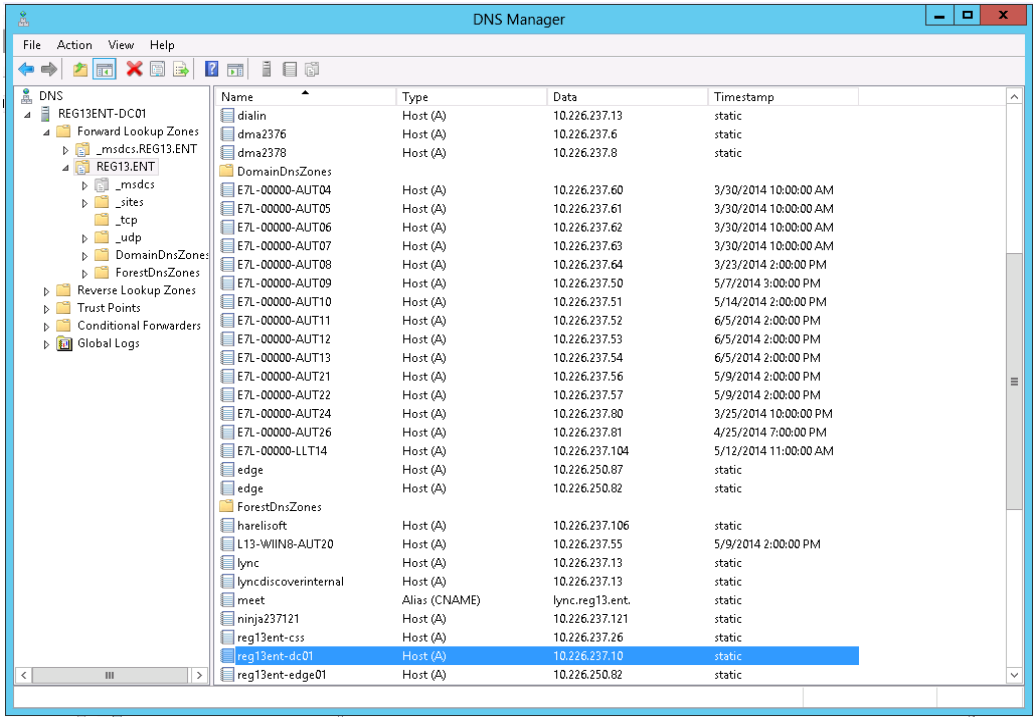
Configure the Collaboration Server FQDN in the DNS

Perform the following steps on the Lync Server to configure the Collaboration Server FQDN in the DNS.

To Configure the Collaboration Server FQDN in the DNS:

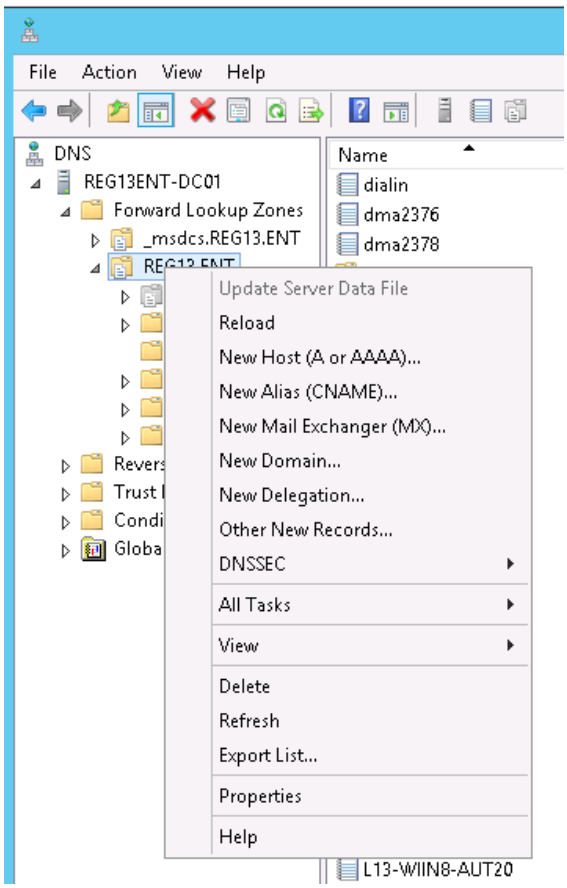
1 Connect to the DNS Server.

2 Open the DNS Manager.

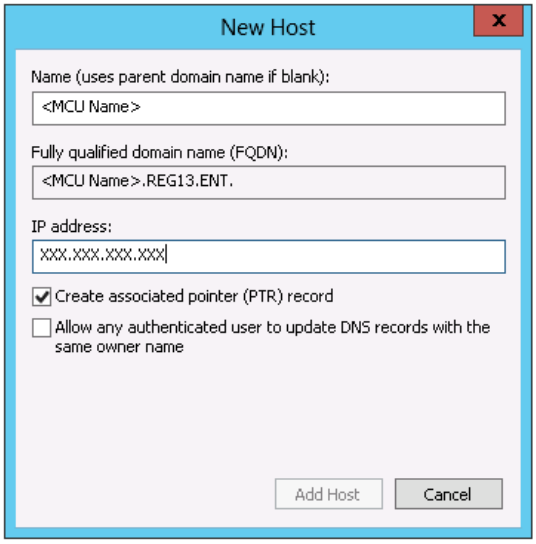


3 Navigate to and expand the **Forward Lookup Zones** folder.

4 Right click on the **Zone** for your Domain.



5 Select **New Host** (A or AAAA).



6 Enter the Collaboration Server **Name** and the Signaling **IP address**.

7 Click **Add Host**.

Configure Collaboration Server Static Route and Trusted Application

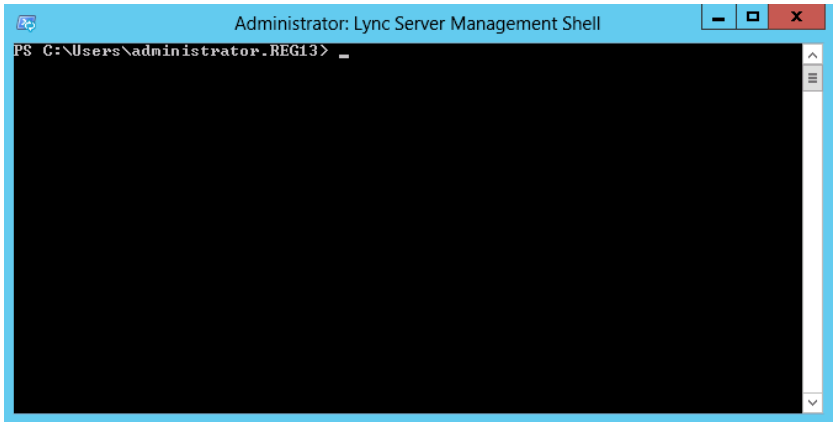
Perform the following steps on the Lync Server to configure the Collaboration Server Static Route and Trusted Application.

To Configure the Collaboration Server Static Route and Trusted Application:

1 Connect to Lync Front End server

2 In the Start Menu Search field, enter **Lync** and click the **Search** button.

3 Select the **Lync Server Management Shell**.



4 Add a **Static Route**:

a Type: **$route = New-CsStaticRoute -TLSRoute -destination "<RMX FQDN>" -port 5061 -matchuri "<Collaboration Server>" -usedefaultcert $true** and press **Enter**.

<Collaboration Server> is the Collaboration Server name.

b Type: **Set-CsStaticRoutingConfiguration -identity global -route @{Add=$route}** and press **Enter**

(To check the Static Route configuration enter the following command: **Get-CsStaticRoutingConfiguration** and press **Enter.**)

5 Create a **Trusted Application Pool** and a **Trusted Application**:

a Type: **New-CsTrustedApplicationPool -Identity <Collaboration Server FQDN> -Registrar Registrar:<Lync pool> -site 1 -ComputerFqdn <Collaboration Server FQDN> -ThrottleAsServer $true -TreatAsAuthenticated $true** and press **Enter**.

b Enter **YES**.

c Type: **New-CsTrustedApplication -ApplicationId <FQDN> -TrustedApplicationPoolFqdn <FQDN> -Port 5061** and press **Enter**.

ApplicationId is the name of the application. This must be a string that is unique within the pool that is specified in the **TrustedApplicationPoolFqdn** parameter.

TrustedApplicationPoolFqdn is the FQDN of the Trusted Application Pool in which the application will reside.

6 Add a **Trusted Application Endpoint** by typing: **New-CsTrustedApplicationEndpoint -sipaddress sip:<name>@<domain> -ApplicationId <FQDN> -TrustedApplicationPoolFqdn <FQDN>** and pressing **Enter**.

7 Enable the changes, by typing: **Enable-CsTopology**.

8 Get the **GRUU** (Globally Routable User-Agent URI):

a Type: **Get-CsTrustedApplication -Filter "<MCU>*" | fl ServiceGruu** and press **Enter**.

The Lync Server will reply with a string similar to the following:

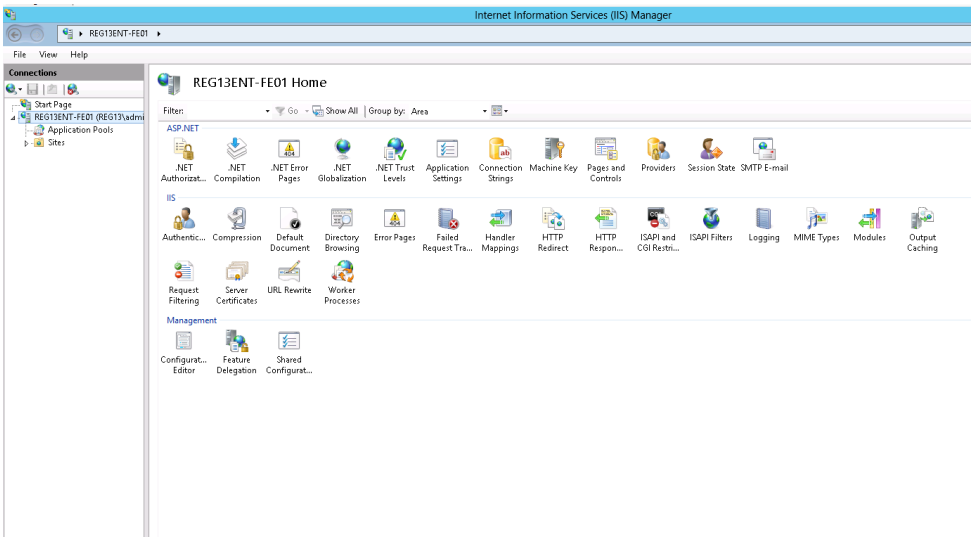
ServiceGruu:sip:rmx23772.reg13.ent@reg13.ent;gruu;opaque=srvr:rmx23772.reg13.ent:0l2d_trGDFSQv4FntDXH-wAA

b Select, Copy, and transfer the **ServiceGruu** string into the workstation that is connected to the Collaboration Server. The SIP prefix (**sip:**) can be omitted from the copied string.

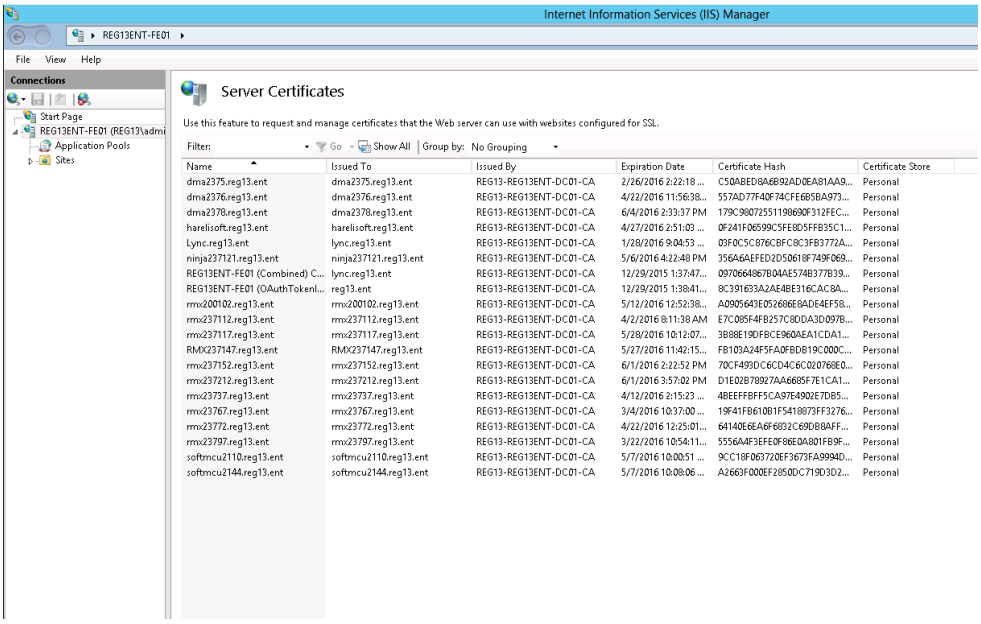
9 Create and export a **Certificate** for the Collaboration Server:

a Type: **Request-CsCertificate -New -Type Default -KeyAlg RSA -CA <CA FQDN> -City PT -State Israel -ComputerFqdn <Collaboration Server FQDN> -Country IL -DomainName <Domain name> -FriendlyName <Collaboration Server FQDN> -Organization 'Polycom ' -PrivateKeyExportable $true** and press **Enter**.

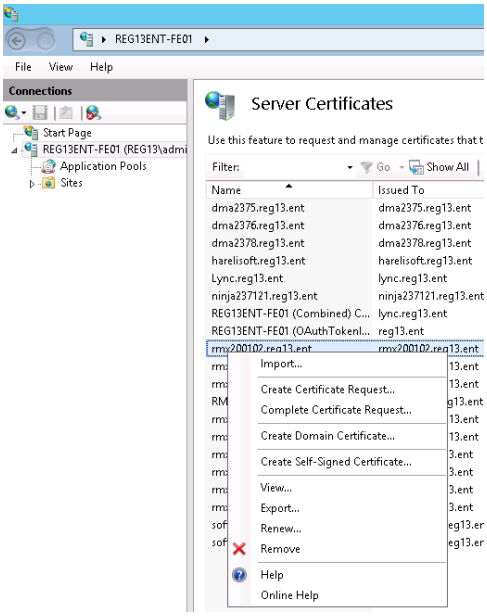
b In the Start Menu, select **Administrative Tools-> Internet Information Services (IIS) Manager**.



c Click **Server Certificates**.

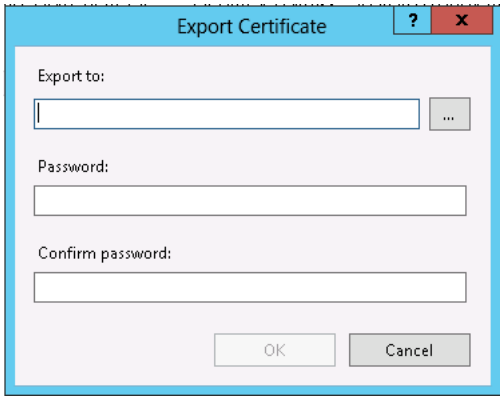


d In the Certificate List, locate the Collaboration Server's certificate.



e Right click the Collaboration Server's certificate.

f Click **Export**.



g In the **Export to** field, enter the file location for exported certificate.

h In the **Password** and **Confirm Password** fields, enter the password.

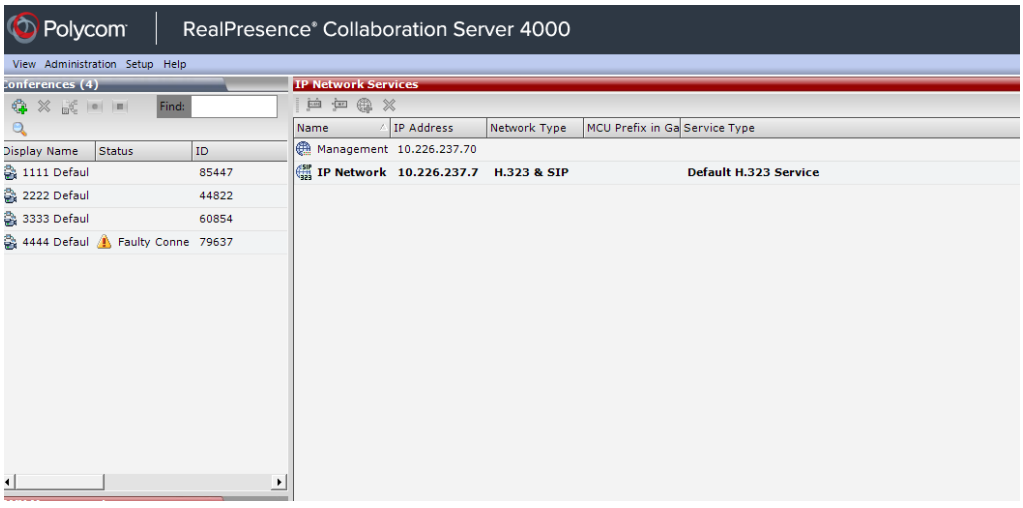
i Click **OK**.

Configure the Collaboration Server for Lync 2010/2013

Perform the following steps on the Collaboration Server to configure it for Lync 2010/2013.

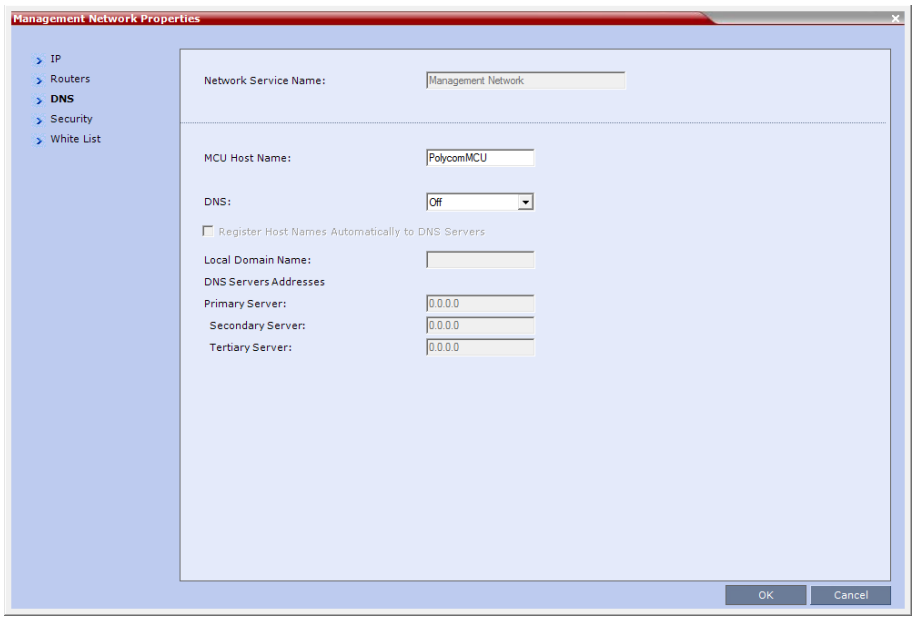
To Configure the Collaboration Server for Lync 2010/2013:

1 Open the IP Network Management Services list.



2 Double click **Management Network**.

3 Click the **DNS** tab.



4 In the MCU Host Name field, enter the name of the DNS (as listed in the DNS Manager).

5 In the Local Domain Name field, enter the Domain Name (as listed in the DNS Manager).

6 In the DNS Servers Primary Server field, enter the DNS IP Address (as listed in the DNS Manager).

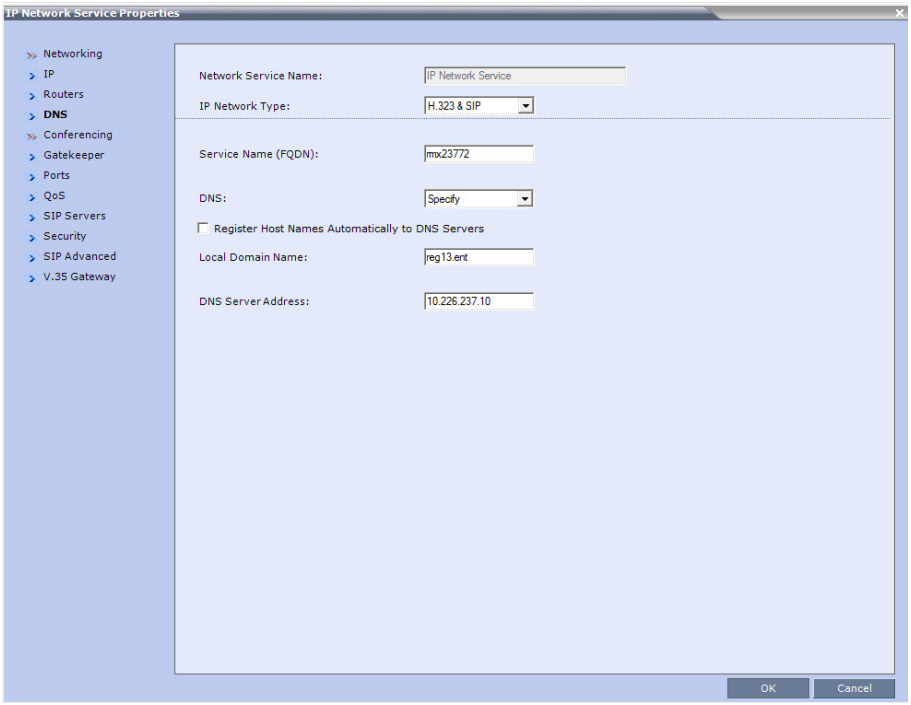
7 Click **OK**

8 Restart the Collaboration Server.

9 Login again and open the IP Network Management Services list.

10 Double click **IP Network Service**.

11 Click the **DNS** tab.



12 In the IP Network Type drop down menu, select **H.323 &SIP**.

13 In the Service Name (FQDN) field, enter the DNS Name (as listed in the DNS Manager).

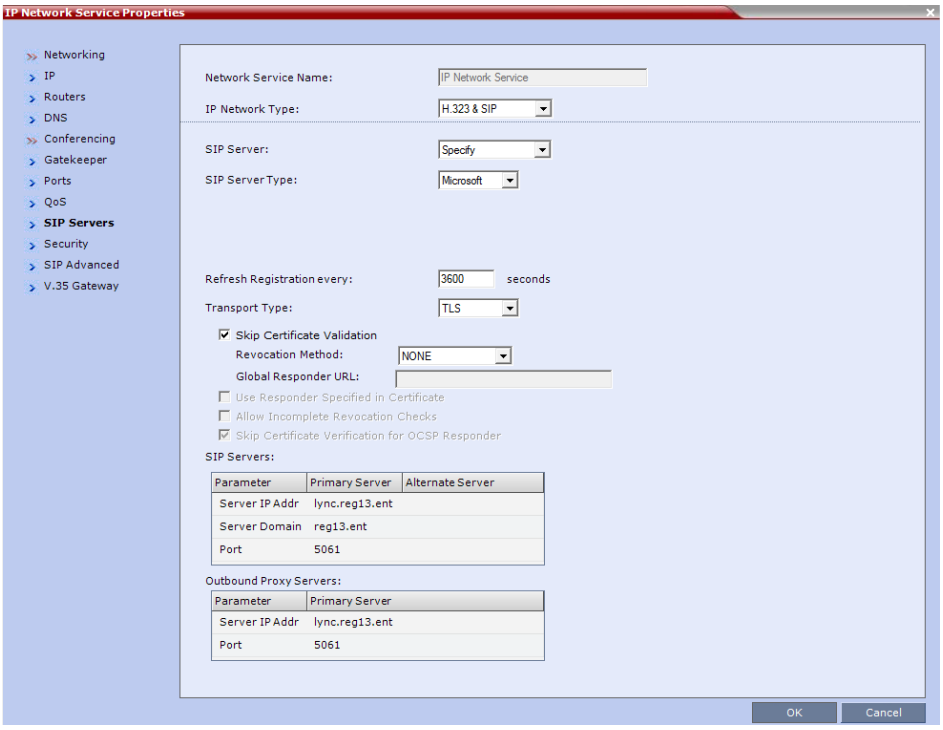
This should not be the FQDN.

14 In the DNS drop down menu, select **Specify.**

15 In the Local Domain Name field, enter the Domain Name (as listed in the DNS Manager).

16 In the DNS Server Address field, enter the IP Address of the DNS server (as listed in the DNS Manager).

17 Click the **SIP Servers** tab.



18 In the IP Network Type drop down menu, select **H.323 &SIP**.

19 In the SIP Server drop down menu, select **Specify.**

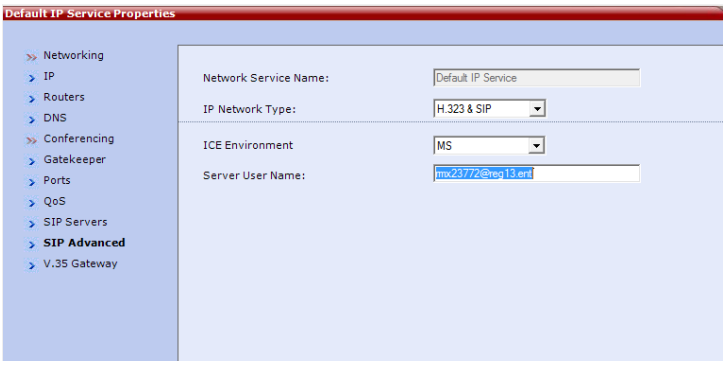
20 In the SIP Server Type drop down menu, select **Microsoft.**

21 In the Transport Type drop down menu, select **TLS.**

22 In the SIP Servers and Outbound Proxy Servers Parameter tables:

- Set the Server IP Address to the DNS Pool name (as listed in the DNS Manager).
- Set the Server Domain to the Domain Name (as listed in the DNS Manager).
- Set Port to 5061

23 Click the **SIP Advanced** tab.



24 In the Server User Name field, enter the Collaboration Server's SIP Address.

This is the first segment of the **ServiceGruu** string with the an "@" replacing the "."

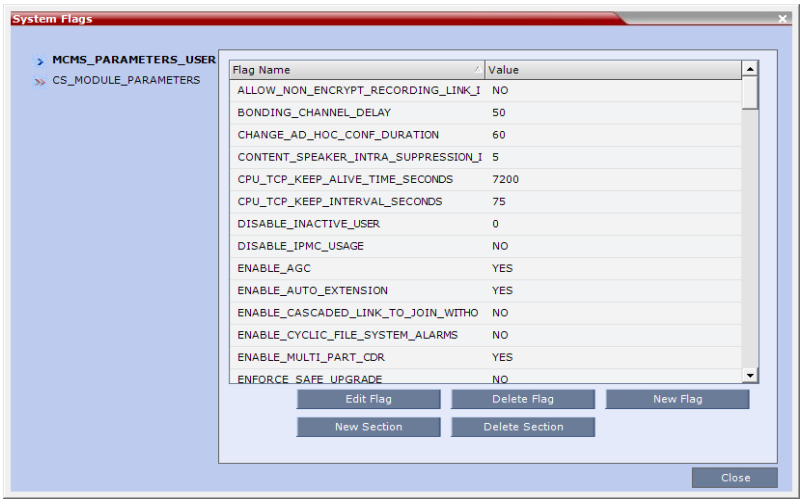
For example if the **ServiceGruu** string was:
rmx23772.reg13.ent@reg13.ent;gruu;opaque=srvr:rmx23772.reg13.ent:0I2d_trGDFSQv4FntDXH-wAA,

the Server User Name would be **rmx23772@reg13**.

25 Add the SIP_CONTACT_OVERRIDE_STR system flag with the ServiceGruu string as its value.

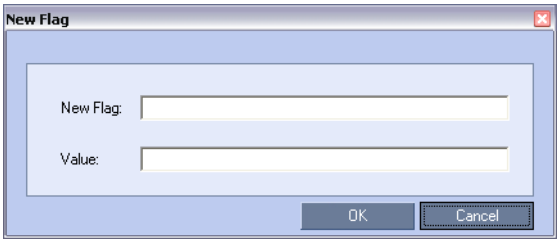
a On the workstation that is connected to the Collaboration Server, on the RMX Menu, click **Setup > System Configuration**.

The System Flags dialog opens.



b In the System Flags dialog, click the **New Flag** button.

The New Flag dialog box is displayed.



c In the New Flag field, enter **SIP_CONTACT_OVERRIDE_STR**.

d Locate, Copy and Paste the **ServiceGruu** string, that was transferred to the workstation in **Step 8b** of the **Configure the Collaboration Server Static Route and Trusted Application** above, into the Value field.

e Click **OK** to close the New Flag dialog.

26 Add the **LIMIT_CIF_SD_PORTS_PER_MPMX_CARD** System Flag and set its value to **YES**.

Setting this flag's value to YES will limit the number of CIF and SD ports (including non Lync clients) to 45 per media card. The flag value and must be set to YES when ICE is active because ICE ports consume more system resources. HD participants are not affected.

27 Repeat Steps **25b - 25e** above with:
- New Flag field: **LIMIT_CIF_SD_PORTS_PER_MPMX_CARD**.
- Value field: **YES**.

28 Reset the Collaboration Server for the new flag settings to take effect.
- **a** Click **OK** to close the New Flag dialog.
- **b** Click OK to close the System Flags dialog.
- **c** Reset the Collaboration Server.

29 After the rest has completed, Login to the Collaboration Server again.

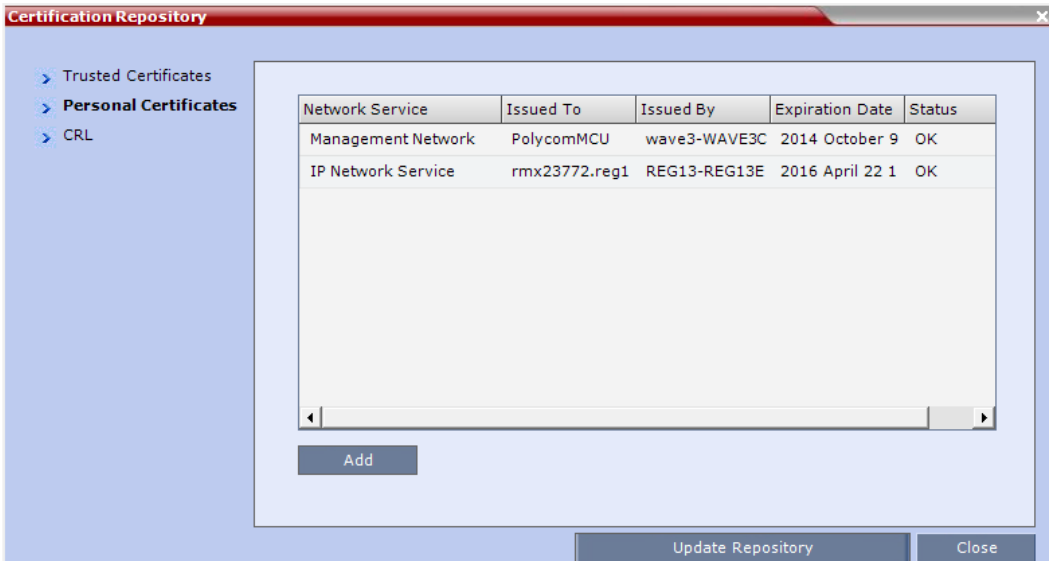
Import and install the Certificate on the Collaboration Server

The Certificate created in **Step 9** of the **Configure Collaboration Server Static Route and Trusted Application** procedure above must be imported an installed on the Collaboration Server.

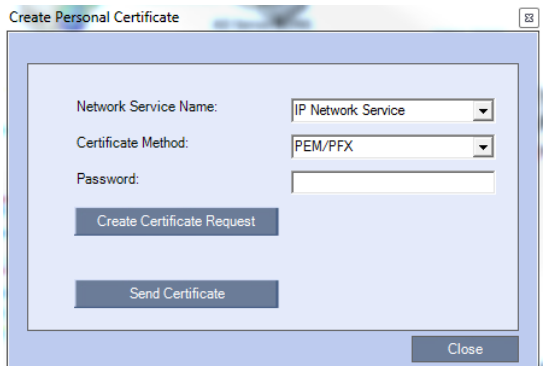
To import and install a certificate:

1 In the Collaboration Server menu, click **Setup > RMX Secured Communications > Certificate Repository**.

2 Click the **Personal Certificates** tab.



3 Click **Add**.

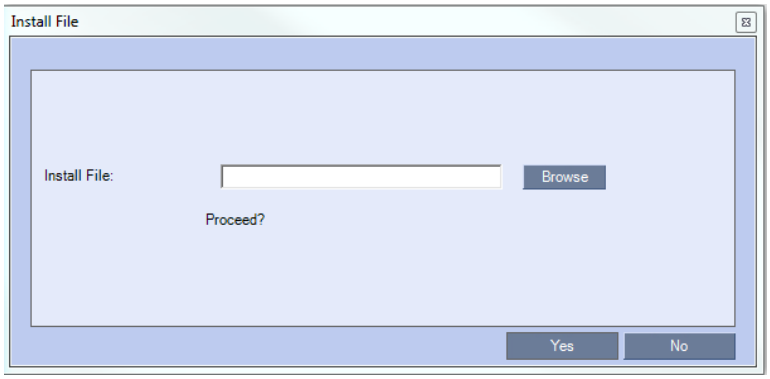


4 In the **Network Service Name** drop down menu, select **IP Network Service.**

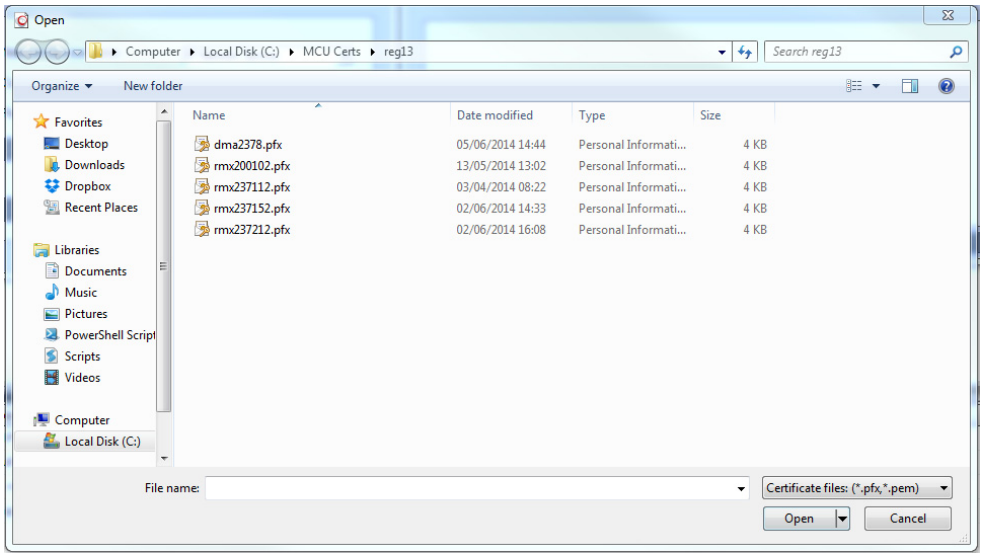
5 In the **Certificate Method** drop down menu, select **PEM/PFX.**

6 In the **Password** field, enter the password entered in **Step 9h** of the **Configure Collaboration Server Static Route and Trusted Application** procedure.

7 Click **Send Certificate**.

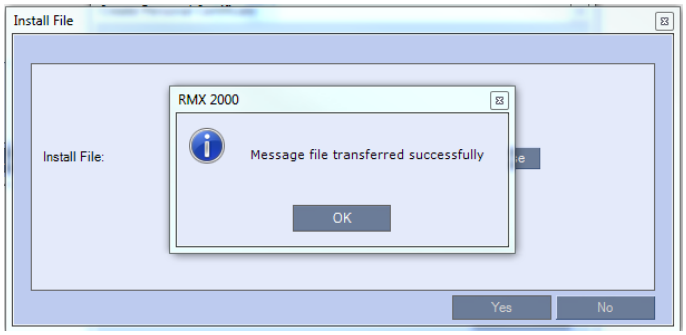


8 Click **Browse,** and then navigate to the folder where the certificate was saved.



9 Select the certificate and click **Open**.

The certificate is installed and a Confirmation Message is displayed.



10 Click **OK** in the confirmation message.

A System Reset confirmation message is displayed.

11 If you are ready to reset the system, click **YES**.

The System Reset completes the Registration of the Collaboration Server to Lync 2010/2013.

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:

Additional MS 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, Microsoft Office Communicator R2 or 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 *Microsoft Office Communicator R2* or *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 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.

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

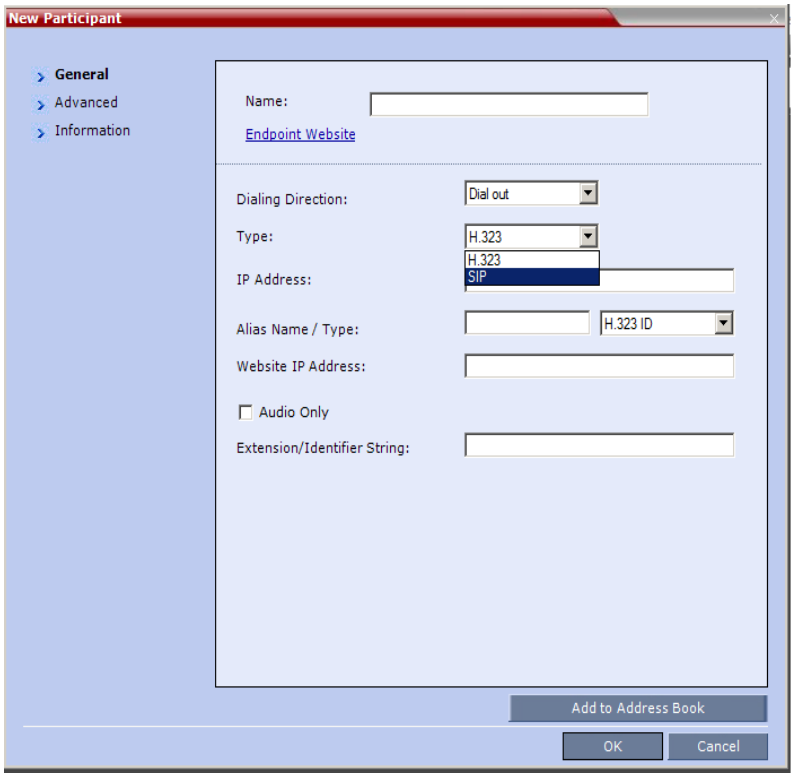
- *System Resource* usage is the same as for the *H.264* protocol. The table below summarizes *System Resource* usage for each of the supported resolutions.

RTV - Resolution by Resolution

Resolution	HD Video Resources Used
QCIF / CIF30	0.5
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:

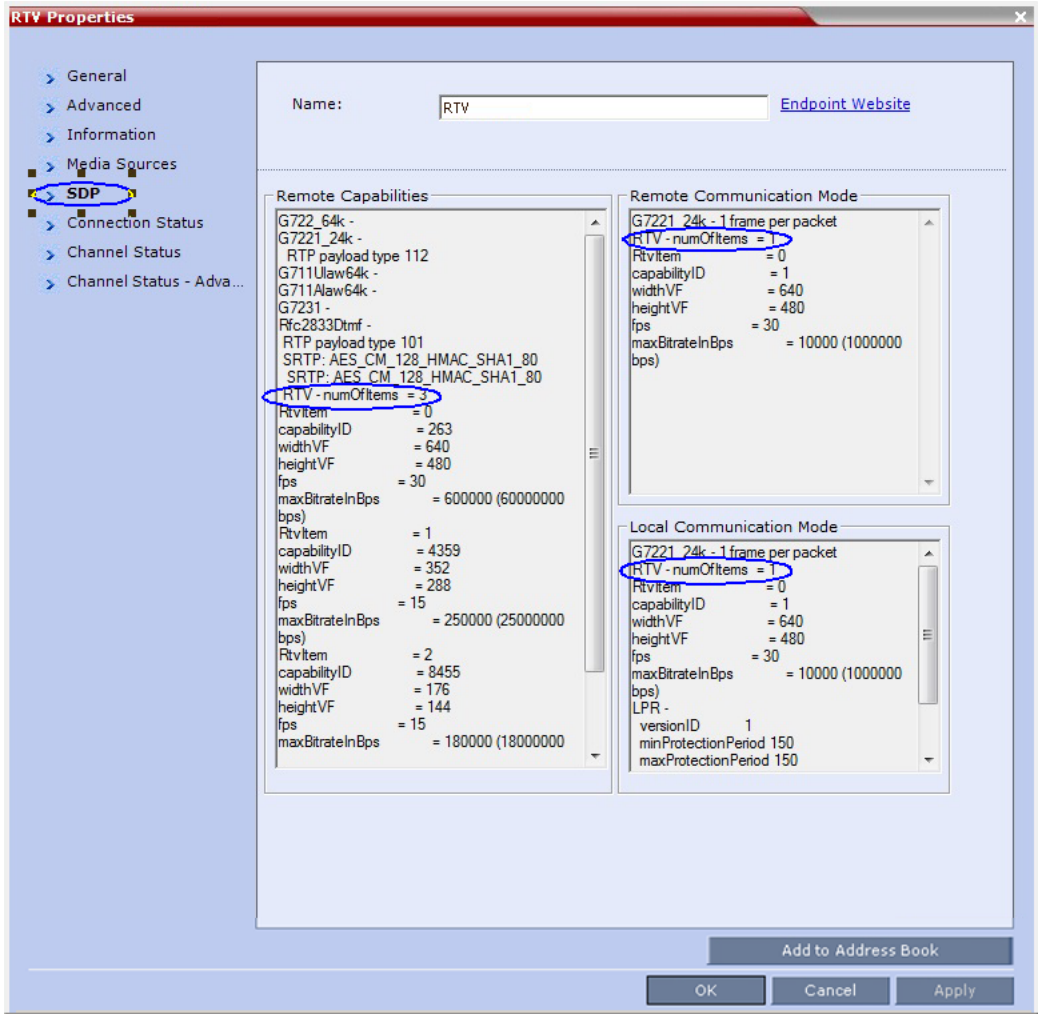
- *OCS* endpoints and *Lync Server* clients connect to the conference using the *RTV* protocol.
- Other endpoints negotiate the video protocol in the following sequence: *H.264*, followed by *RTV,* followed by *H.263* and finally *H.261*.

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).

Monitoring RTV

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



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*.

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 *<Italic>RealPresence Collaboration Server 800s Administrator's Guide*, the *<Italic>RealPresence Collaboration Server 800s Administrator's Guide*, Modifying System Flags.

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.

Selected video resolution based on flag setting and conference line rate and core processor The following table describes the number of allocated video resources for each video resolution when using the RTV protocol.

Allocated video resolutions per video resolution

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
- 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

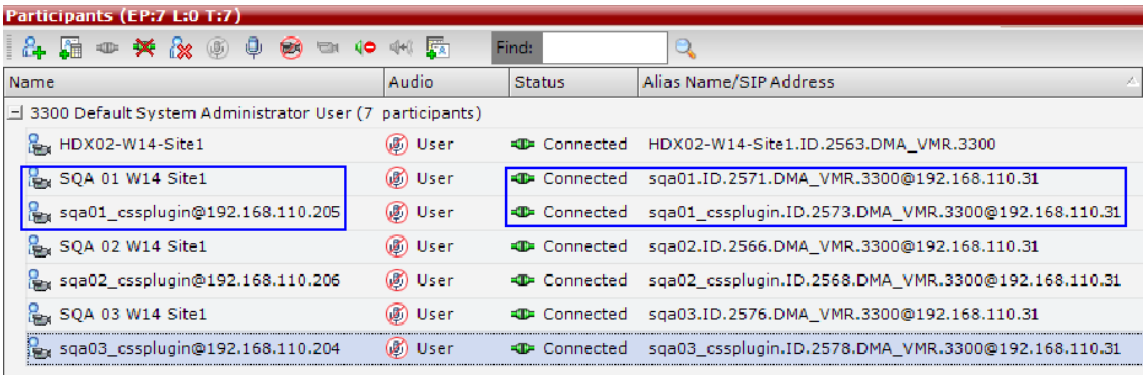
Sharing Content via the Polycom CSS Plug-in for Lync Clients

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.



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.

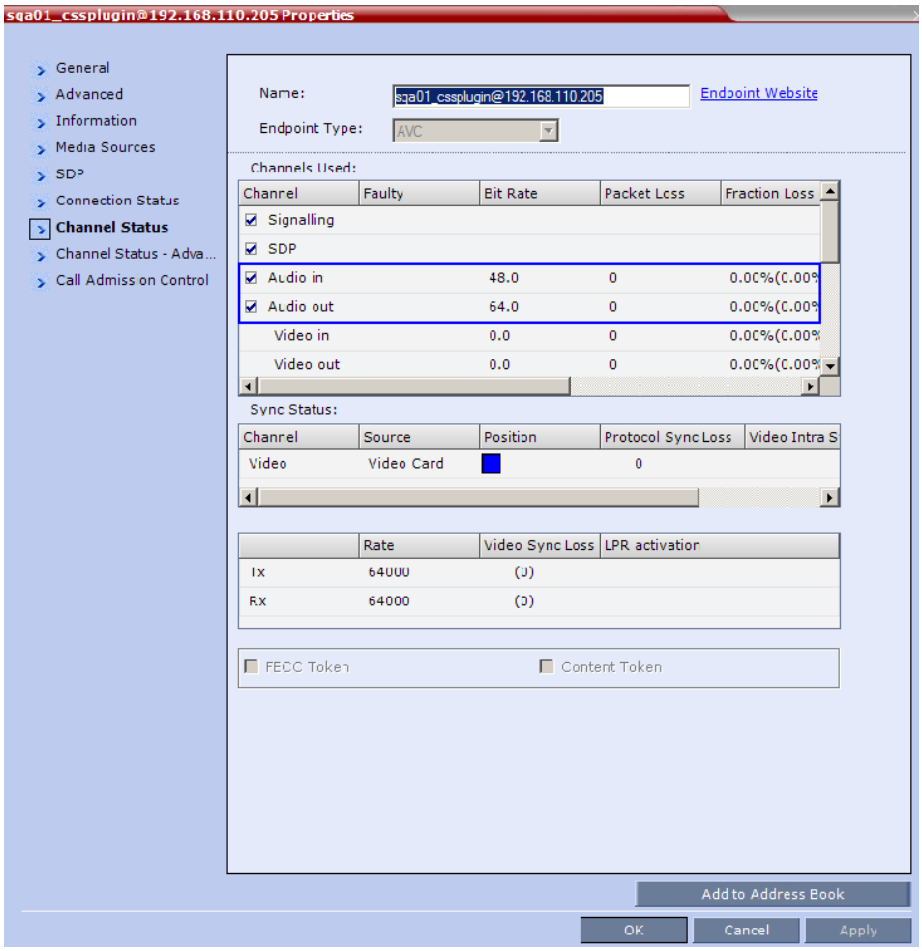
- When set to **NO** (default), *Content* is sent to all Lync clients over the video channel, including those with the Polycom CSS plug-in installed, even when the *Send Content to Legacy Endpoints* is disabled.
  Other, non-Lync legacy endpoints will not be affected by this flag and will receive content 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

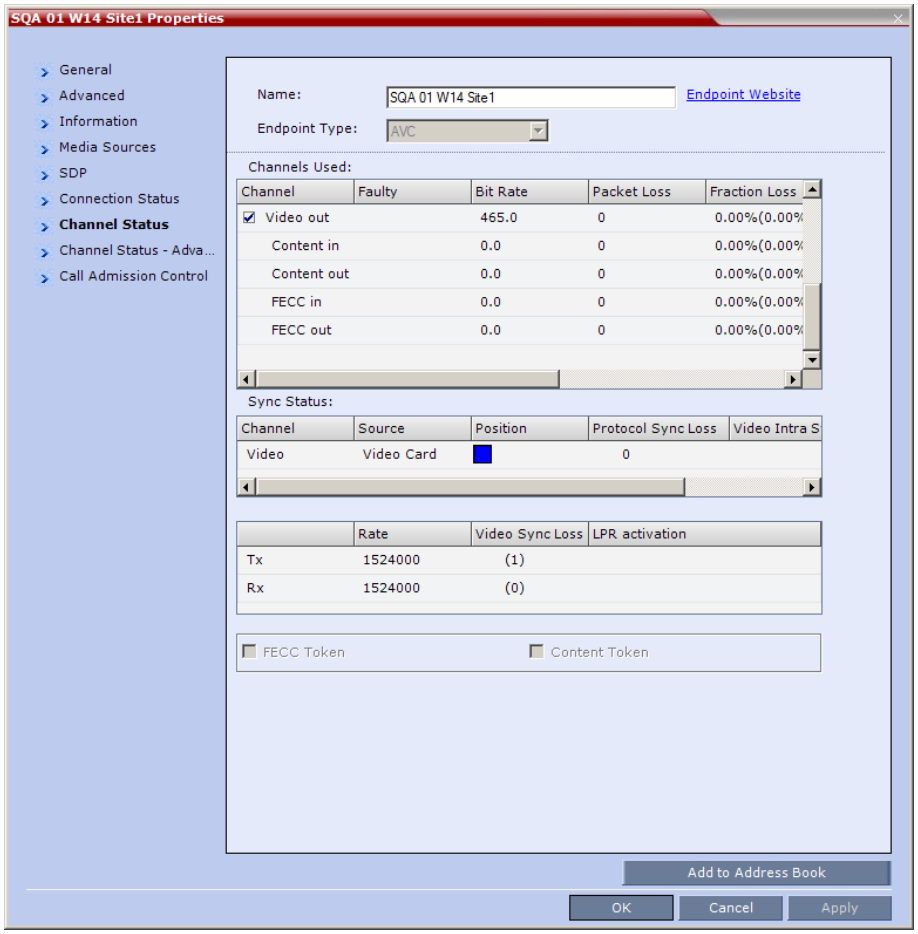
Content is shared in a video switching mode. Therefore, when a Lync client connects to the conference via the Polycom CSS plug-in, the content resolution will be adjusted to the maximum content rate possible by the Lync client, up to a maximum of **720p 5fps** in all line rates, even if you select a 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.

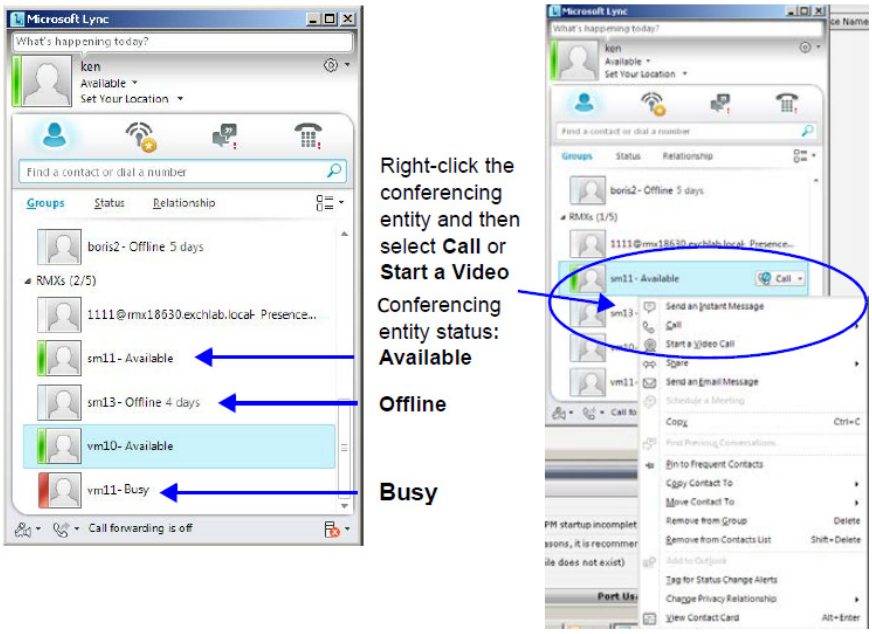


- The properties of the Lync client are those of a video participant. However, the *Content* channel will show 0 as there is no content channel.



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 Office Communication Server client or 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.



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
- Entry Queues, Ad Hoc Conferences and SIP Factories

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 Office Communication Server or Lync Server
- Defining the Microsoft SIP Server in the IP Network Service
- Enabling Registration in the Conference Profile

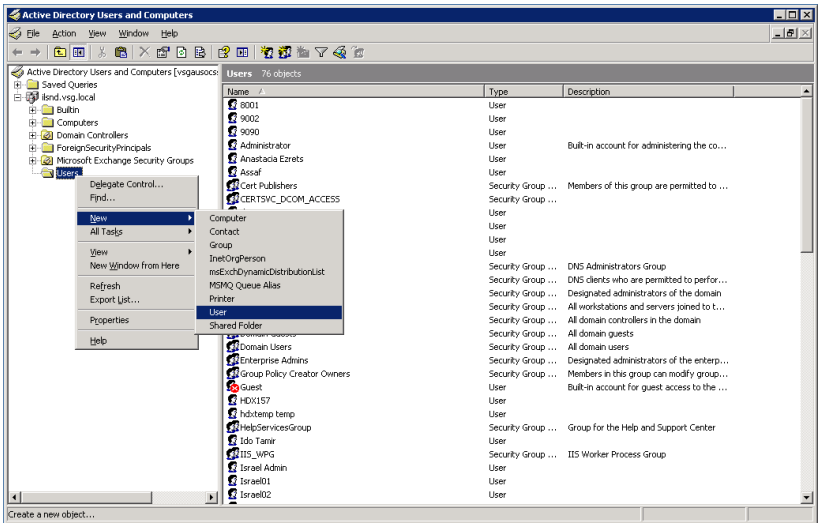
Creating an Active Directory Account for the Conferencing Entity

The User account created for the Conferencing entity is used for registration with the Office Communication Server or Lync server and to automatically synchronize with the STUN and relay (Edge) servers.

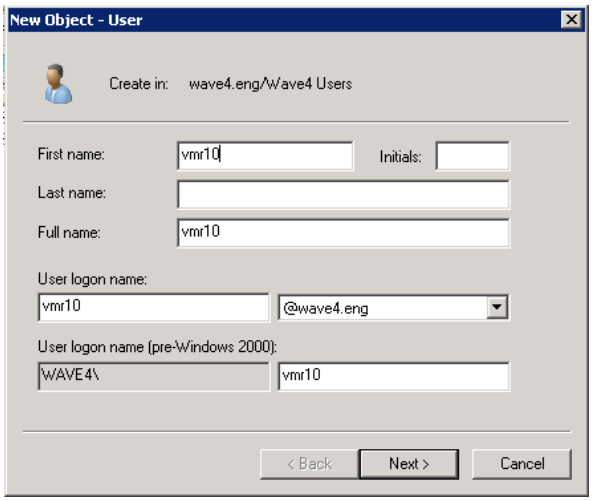
To add the conferencing entity user to the Active Directory:

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**.



3 In the *New User* wizard, define the following parameters:



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 Office Communication Server or Lync Server

The new Conferencing Entity user must be enabled for registration with the Office Communications Server or Lync Server.

To enable the Conferencing Entity User Account for Office Communication Server:

1. In the *Active Directory Users and Computers* window, right-click the conferencing entity user and then click **Properties**.
2. In the *Properties* dialog box, click the **Communications** tab.
3. In the *Sign in name* field, enter the conferencing entity user name in the format **SIP:conferencing entity user name** (for example sip:vm10) and select the domain name (for example, lab.vsg.local) as entered in the *New User* dialog box.
4. Select the *Server or Pool* from the list.
5. Click **Apply** and then **OK**.

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**.

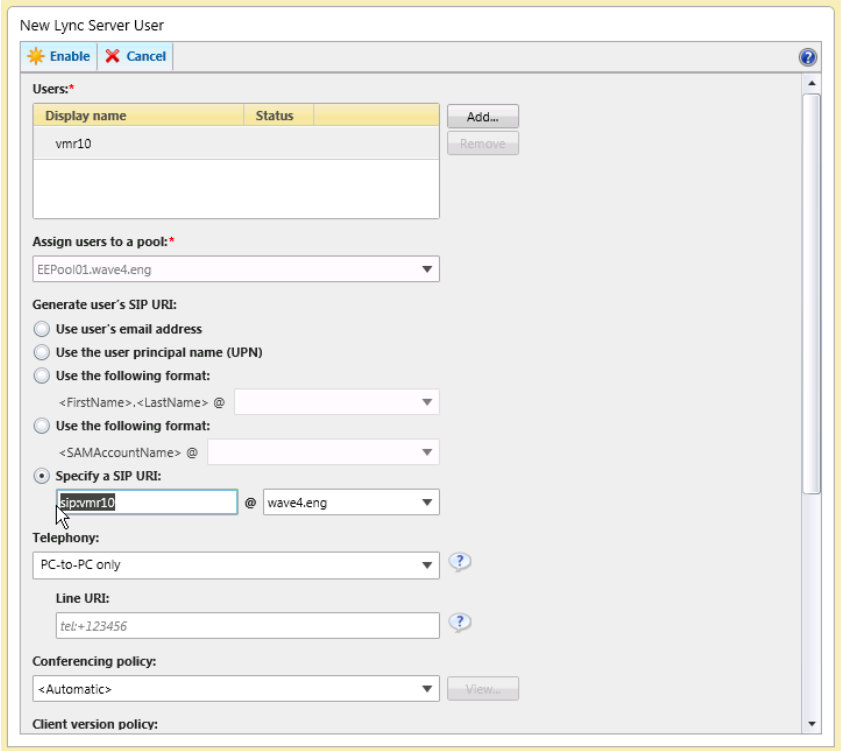
   Windows Security window opens.
2. Enter your User name and Password as configured in the Lync Server and click OK.
   The *Microsoft Lync Server 2010 Control Panel* window opens.
3. Click the **Users** tab.
4. In the *User Search* pane, click the **Enable Users** heading.

   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 is displayed in the *New Lync Server User* pane.

8 Select the following parameters:



- 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 is displayed as enabled in the *User Search* pane.

Defining the Microsoft SIP Server in the IP Network Service

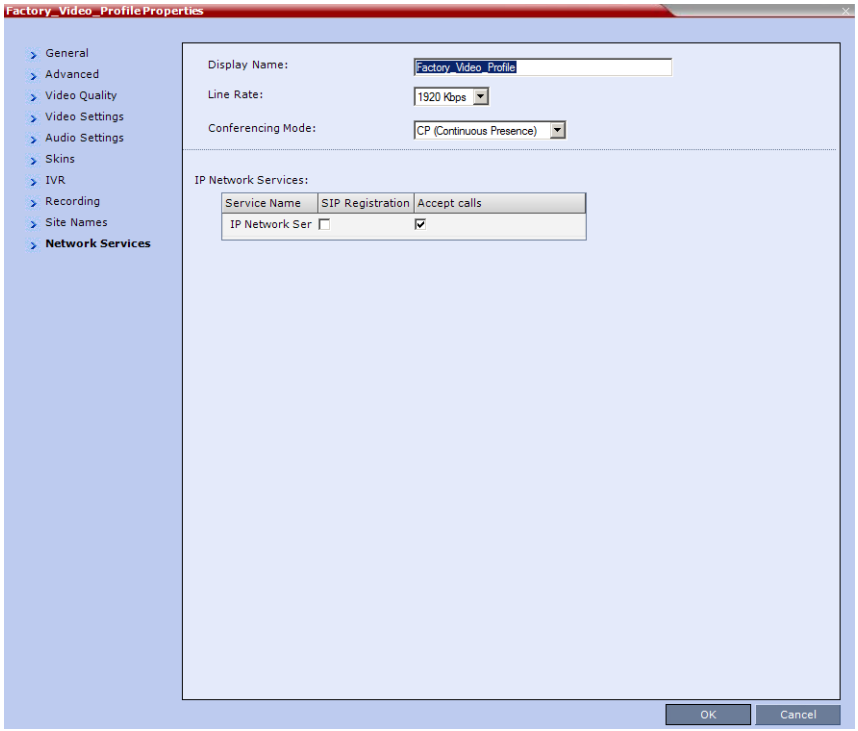
To enable the registration of the conferencing entities the *SIP Server Type* must be set to **Microsoft** and the Office Communication Server or Lync Server properties in the *IP Network Service - SIP Servers* dialog box.

For more details, see Configuring the Collaboration Server IP Network Service.

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.



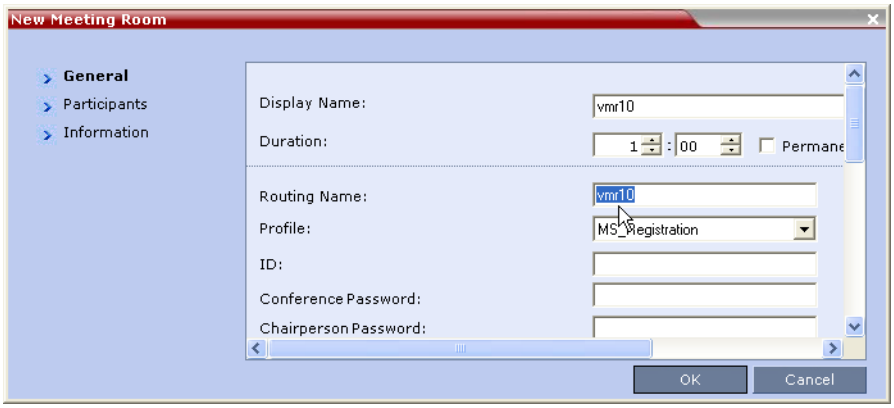
Profile Properties - Network Services

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 Office Communications client or 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.



  - 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.

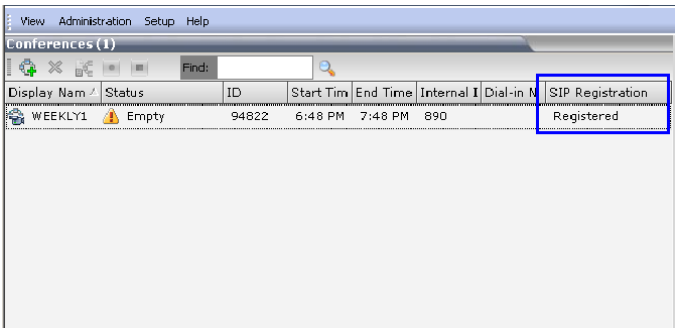
Conferencing Entity List

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:

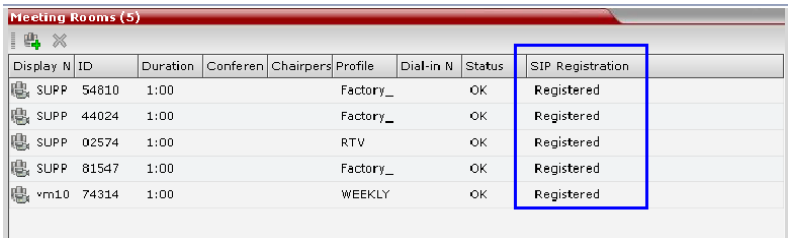
- **Not configured** - Registration with the SIP Server was not enabled in the Conference Profile assigned to this conferencing Entity. In Multiple Networks configuration, If one service is not configured while others are configured and registered, the status reflects the registration with the configured 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.

Ongoing Conferences list - SIP Registration



Meeting Rooms list - SIP Registration



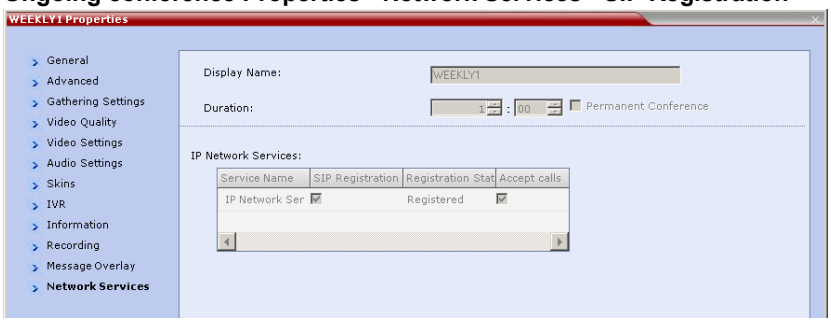
Entry Queues list - SIP Registration

SIP Factories list - SIP Registration

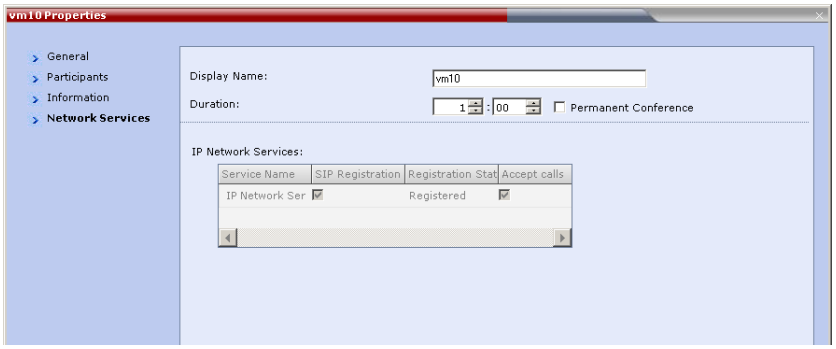
Conferencing Entity Properties

Registration status is reflected in the *Properties - Network Services* dialog box:

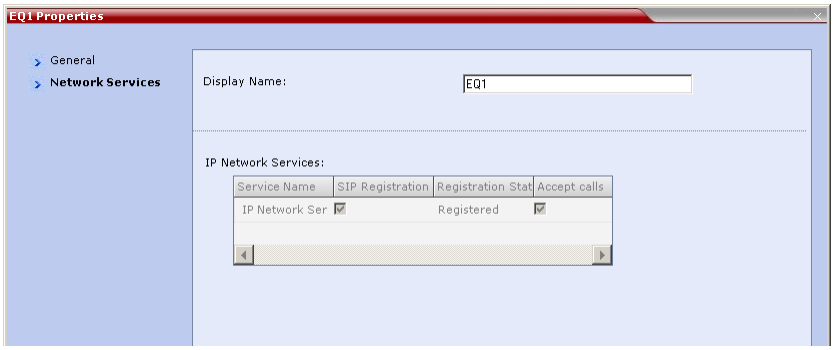
Ongoing conference Properties - Network Services - SIP Registration



Meeting Room Properties - Network Services - SIP Registration



Entry Queue Properties - Network Services - SIP Registration



Collaboration Server Configuration for CAC Implementation

CAC is enabled by manually adding the flags to the system Configuration and setting their values as follows:

- To enable the Call Admission Control implementation in the Collaboration Server:
  - CAC_ENABLE=YES
- In addition, to ensure that endpoints such as HDX remain connected to the conference for its duration when the Collaboration Server is configured with FQDN address and the Lync server is working with load balancing and holds more than one address, the following two flags must be manually added and set to:
  - MS_KEEP_ALIVE_ENABLE = YES

    Note: Since the keep alive is only required when the Lync server is working with load balancing and holds more than one address, the default value is NO.
  - SIP_TCP_PORT_ADDR_STRATEGY = 1 (default setting)

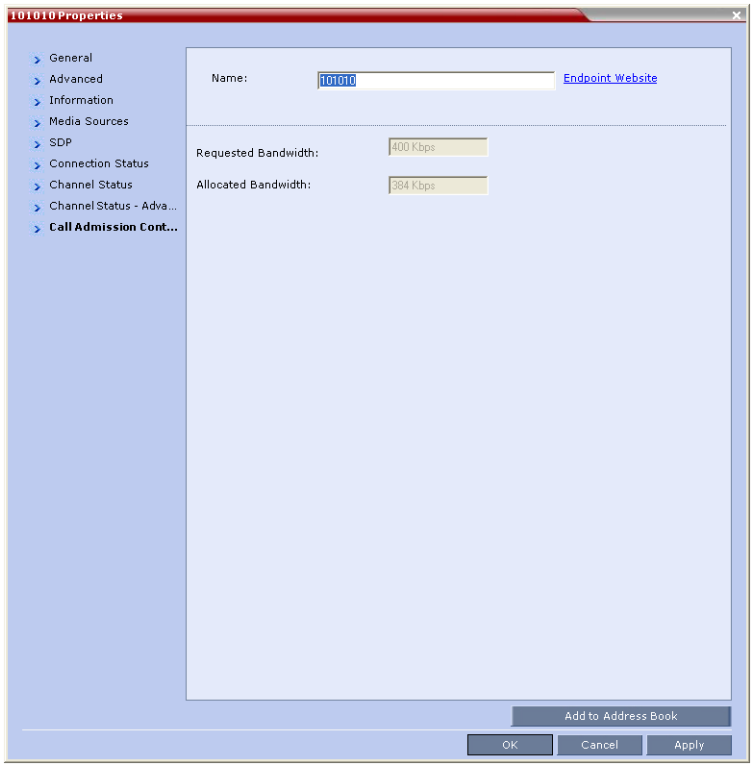
- When Call Admission Control is enabled in the local network, by default the local the ICE channel is closed after applying CAC bandwidth management.
  To change and preserve the ICE channel open throughout the call:
  - **PRESERVE_ICE_CHANNEL_IN_CASE_OF_LOCAL_MODE=YES**.

Conferencing Behavior in CP Conferences

In Continuous Presence conference, Lync clients connect with any allocated bandwidth.

Monitoring Participant Connections

Activation of the Call Admission Control for a call can be viewed in the *Participant Properties - Call Admission Control* dialog box.



This information applies only to dial-in participants.

The following information is available:

Participant Properties - Call Admission Control Parameters

Field	Description
Requested Bandwidth	Indicates the bandwidth requested by the Lync client (usually the line rate set for the conference). NA - indicates that *Call Admission Control is disabled.*
Allocated Bandwidth	The actual bandwidth allocated by the Lync Policy Server. NA - indicates that *Call Admission Control is disabled.*

Connecting a Collaboration Server Meeting Room to a Microsoft AV-MCU Conference

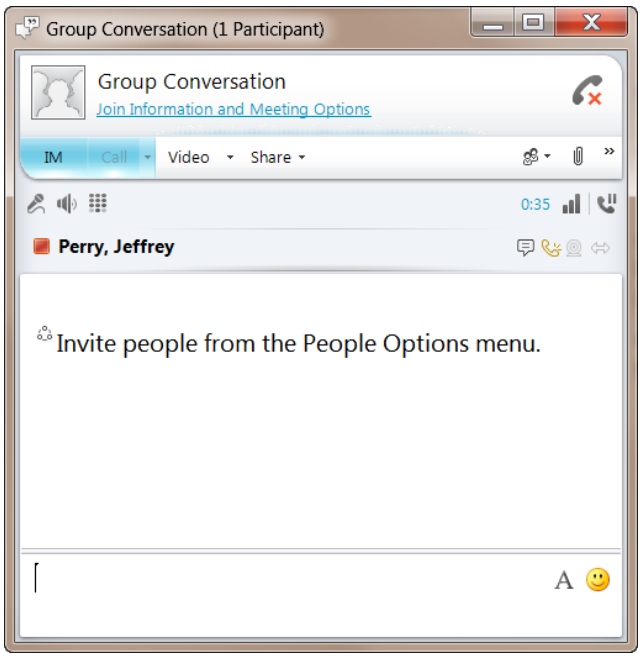
Microsoft Lync users can connect an Collaboration Server Meeting Room to a conference running on the Microsoft A/V MCU. This allows Collaboration Server Lync users to connect with a conference in progress on the A/V MCU and be an active participant in the conference.

The connection to the A/V MCU uses the same configuration as in a cascading conference between multiple Collaboration Server MCUs, with the only difference that both the MCU and the AV-MCU should reside on the same network subnet, so that no firewall or other barriers will trigger the NAT Traversal functionality.

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.



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.

Configuring the Collaboration Server for Federated (ICE) Dialing

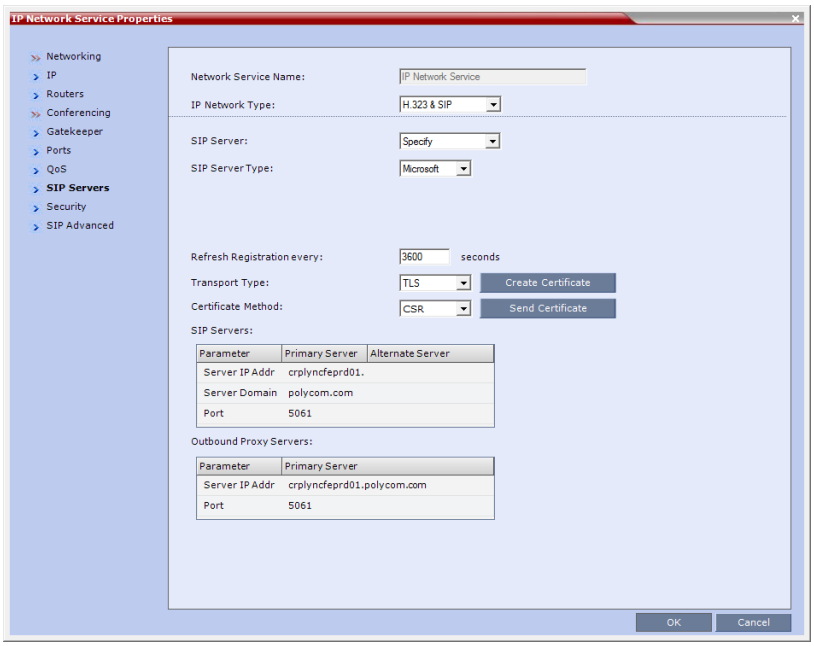
The Collaboration Server *Default IP Network Service* must be configured to work with the Office Communication Server/Lync Server as the SIP Server and the Collaboration Server user defined in the Active Directory must also be defined in the Collaboration Server ICE environment parameters to enable remote dialing in a federated (ICE) environment, .

The procedure described here assumes that the Collaboration Server is configured to work in Microsoft environment as described in Configuring the Collaboration Server for Microsoft Integration.

To configure the Collaboration Server for ICE Dialing:

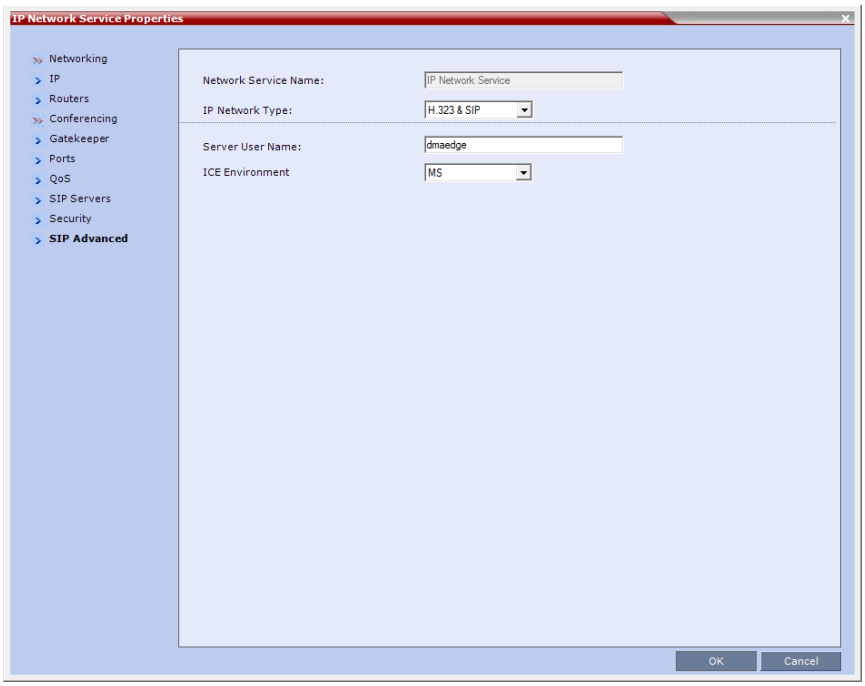
1. In the RealPresence Collaboration Server Web browser, in the *RealPresence Collaboration Server RMX Management* pane, expand the **Rarely Used** list and click **IP Network Services**.
2. In the *IP Network Services* pane, double-click the **Default IP Service** entry.

   The *Default IP Service - Networking IP* dialog box opens.
3. Click the **SIP Servers** tab.



4. Make sure that the *SIP Server* is set to **Specify.**
5. Make sure that the *SIP Server Type* is set to **Microsoft**.
6. Make sure that the IP address of the Office Communications Server 2007 or Lync Server 2010 is specified and the *Server Domain Name* is the same as defined in the OCS/Lync Server and in the *Management Network* for the DNS.

7 Click the **SIP Advanced** tab.



8 In the *ICE Environment* field, select **MS** (for Microsoft ICE implementation) to enable the ICE integration.

9 In the *Server User Name* field, enter the Collaboration Server User name as defined in the Active Directory. For example, enter **rmx111**.
This field is disabled if the *ICE Environment* field is set to **None**.

10 Multiple Lync Registrations can be configured on the Collaboration Server. For more information see Multiple Lync Registrations on the Collaboration Server.

11 **Optional** if the **Fixed Ports** options was selected previously.
Click the **Ports** tab to modify the number of UDP Ports allocated to the calls to accommodate the number of ports required for ICE dialing.

12 In the *UDP Port Range*, modify the number of UDP ports by enter the first and last port numbers in the range. When ICE environment is enabled, the number of ports defined in the range should be **2048**.

13 Click **OK**.
The Collaboration Server will register with the OCS/Lync Server enabling automatic retrieval of the STUN server and Relay server parameters for ICE dialing.

These parameters can be viewed in the *Signaling Monitor - ICE Servers* dialog box.

Monitoring the Connection to the STUN and Relay Servers in the ICE Environment

1 In the Collaboration Server Web browser, in the *Collaboration Server Management* pane, click **Signaling Monitor**.

2 In the *Signaling Monitor* pane, click the **IP Network Service** entry.

3 Click the **ICE Servers** tab.

The system lists the ICE servers to which it is connected.It addition, the system indicates the status of the firewall detection in the Collaboration Server.

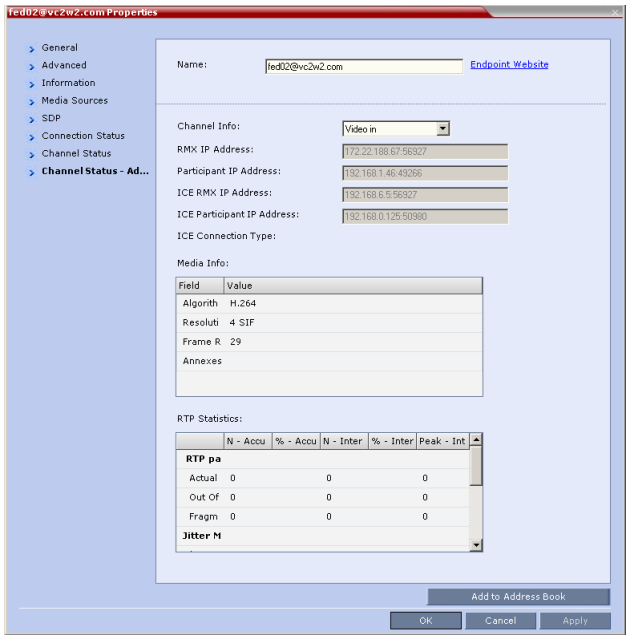
Monitoring the Participant Connection in ICE Environment

For each participant in the conference running in ICE environment, you can view the local and the external IP addresses and the type of connection between the Collaboration Server and the participant (remote).

The ICE information is displayed only for the media channels and not the signaling channel.

To view the channel properties of the participant:

1. In the participants pane, double-click the participant entry or right-click the participant entry and then click **Properties**.
2. Click the **Channel Status - Advanced** tab.



The following connection information is displayed:

Participant Properties - ICE Connection Parameters

Field	Description
Collaboration Server IP Address	The local IP address and port (in the format IP address:Port) of the Collaboration Server.
Participant IP Address	The local IP address and port (in the format IP address:Port) of the endpoint.
ICE Collaboration Server IP Address	The IP address and the Port number of the Collaboration Server used to pass through the media. This information changes according to the *ICE connection type:* • When *ICE connection type* is **local**, it is identical to the IP address:Port displayed in the *Collaboration Server IP Address.* • When *ICE connection type* is **relay**, the system displays the IP address and port number of the relay server used to pass the media from the Collaboration Server to the participant. • When *ICE connection type* is **firewall**, the system displays the public IP address and port of the Collaboration Server as seen outside the private network.
ICE Participant IP Address	The IP address and the Port number of the endpoint used to pass through the media. This information changes according to the *ICE connection type:* • When *ICE connection type* is **local**, it is identical to the IP address:Port displayed in the *Participant IP Address.* • When *ICE connection type* is **relay**, the system displays the IP address and port number of the relay server used to pass the media from the participant to the Collaboration Server. • When *ICE connection type* is **firewall**, the system displays the public IP address and port of the endpoint as seen outside the private network.

Field	Description
ICE Connection Type	Indicates the type of connection between the Collaboration Server and the participant in the ICE environment: • **Local** (or Host) - The endpoint (Remote) is on the same network as the Collaboration Server and the media connection is direct, using local addresses. • **Relay** - Media between the Collaboration Server and the participant passes through a media relay server. • **Firewall** - Media connection between the Collaboration Server and the participant is done using their external IP addresses (the IP addresses as seen outside of the local network).

For a detailed description of ICE Active alarms, see ICE Active Alarms.

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:

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.

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.

ICE Active Alarms

When ICE environment is enabled in the Collaboration Server, failure to communicate with a required component triggers the display of an Active Alarm in the System Alerts pane.

The following table lists these active alarms:

ICE Environment - Collaboration Server Active Alarms

Active Alarm	Phase	Alarm Displayed When	Troubleshooting
ICE failure: Failed to register with OCS. Check the Collaboration Server Server Name.	Registration	The Collaboration Server did not receive a confirmation response from the OCS to the Registration request.	• Check that the Collaboration Server Name in IP Network Service - SIP Advanced is identical to the User name defined for the Collaboration Server in the OCS Active Directory. • Make sure that the Collaboration Server user is defined in the OCS Active Directory.
ICE failure: Failed to subscribe with the OCS, therefore the A/V Edge Server URI was not received.	Subscribe	The Collaboration Server did not receive a confirmation response from the OCS to the Subscription request. The Subscription is required for obtaining the A/V Edge Server URI which is followed by the notify message containing the credentials).	
ICE failure: The Notify message containing the A/V Edge Server URI was not received	Notify	The Notify message containing the A/V Edge Server URI was not received by the Collaboration Server.	
ICE failure: Received Notification does not contain URI.	Notify	The notify message that was sent from the A/V Edge Server does not contain the A/V Edge server URI.	Verify the A/V Edge server is configured in the OCS.
ICE failure: No response from the A/V Edge Server to the Collaboration Server Service Request	Service	The Collaboration Server did not receive a confirmation response from the A/V Edge Server to the Service request.	
ICE failure: Received Service message does not contain the Credentials.	Service	The Service message response does not contain the Credentials.	

Active Alarm	Phase	Alarm Displayed When	Troubleshooting
ICE failure: A/V Edge server URI cannot be resolved	Service	The Collaboration Server failed to resolve The remote address of the Edge server URI.	
ICE failure: Service credential denied. A/V Edge server credentials rejected by the OCS.	Service	This alarm indicates that the OCS does not configure with the. Generated by the ICE stack.	

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.

Lync 2013 SVC Connectivity to Polycom MCU

The Microsoft H.264 SVC codec replaces the H.263 codec previously used with Lync 2013 clients. Although similar Polycom's standards-based H.264 and SVC implementation, it is proprietary to Microsoft, enabling video calls between Lync 2013 clients (endpoints) and Polycom endpoints to be established.

The Collaboration Server considers Lync 2010 and H.264 SVC Lync 2013 clients to be AVC endpoints. The administrator must set the Conferencing Mode in the Conference Profile to **CP (Continuous Presence)** to enable H.264 SVC Lync clients to connect to the conference.

Deployment Architectures

Two Deployment Architectures are presented as examples. Both require that a Polycom RealPresence Distributed Media Application (DMA) System 7000 be configured as part of the solution.

- `Deployment Architecture 1 - Collaboration Server Hosted` (Direct)

  Lync 2013 clients connect to a conference hosted on Polycom RealPresence Collaboration Server.

- `Deployment Architecture 2 - MS AV MCU Cascade`

  Lync 2013 clients connect to a conference on Microsoft AVMCU which is connected to a Polycom RealPresence Collaboration Server.

The following table summarizes current and legacy (non DMA) conferencing modes within the deployment architectures:

Conference Modes by Deployment Architecture

Conference Mode	Deployment Architecture 1 Direct Dial In/Out (No AV MCU cascade) With or without DMA	Deployment Architecture 2 AV MCU Cascade Call (Indirect Dial In)	
		With DMA	Non DMA (Backward Compatibility)
AVC Only	Supported. Dial Out out from DMA is not supported. For backward compatibility, Dial out from the RMX Web Client or RMX Manager can be used. When using backward compatibility mode for Dial out, H.264 SVC Lync 2013 clients are not be supported. If the Video Protocol field in the Participant dialog - Advanced tab is set to **Auto**, RTV protocol is used.	Supported. Simulcast transmitted. Multiple participants received.	Supported. One transmitted RTV stream. One received RTV stream.
Mixed Mode	Supported	Not Supported	
SVC Only	Not Supported		

Backward compatibility to Lync 2010

All Lync 2013 functionality can be disabled by adding the **BLOCK_NEW_LYNC2013_FUNCTIONALITY** System Flag and setting its value to **YES**.

The flag's default value is **NO**, and when set to yes, all Lync 2013 new functionality is disabled. All Lync 2013 clients, whether connected Directly or by MS AV MCU cascade, will connect using the RTV codec, not the MS SVC codec.

Video Resource Requirements and Implications

Lync 2013 SVC clients may not all connect to a VMR with the same stream layout. They are therefore considered H.264 AVC participants and transcoding resources are allocated to them as summarized in the following table.

Bandwidth and Resource Consumption by Video Codec

Video Codec	Resolution and Aspect Ratio	Maximum Video Payload Bit rate (Kbps)	Minimum Video Payload Bit rate (Kbps)	Resources
H.264	320x180 (16:9) Coded as 320x192 212x160 (4:3)	250	15	CIF
H.264/RTVideo	424x240 (16:9) Coded as 432x240 320x240 (4:3)	350	100	CIF
H.264	480x270 (16:9) Coded as 480x272 424x320 (4:3) Coded as 432x320	450	200	SD
H.264/RTVideo	640x360 (16:9) Coded as 640:368 640x480 (4:3)	800	300	SD
H.264	848x480 (16:9)	1500	400	SD
H.264	960x540 (16:9) Coded as 960:544	2000	500	SD
H.264/RTV	1280x720 (16:9)	2500	700	HD720p30
H.264	1920x1080 (16:9)	4000	1500	HD1080p30

Support for HD1080p Resolution

The Collaboration Server Hosted deployment supports HD1080p30 video resolution symmetrically for direct calls.

The MS AV MCU Cascade deployment supports HD1080p30 video resolution only if `Video Optimized` mode is selected and according to the settings of the **LYNC_AVMCU_1080p30_ENCODE _RESOLUTION** System Flag:

NO (Default) Video streams sent to and received from the MS AV MCU are HD720p30, SD, and CIF.

YES Video streams sent to the MS AV MCU are HD1080p30, SD, CIF. Video streams received from the MS AV MCU are 720p30,SD, and CIF.

Limit Maximum Resolution for MS SVC Using a System flag

The **MAX_MS_SVC_RESOLUTION** System Flag can be used to minimizing the resource usage by overriding the default resolution selection and limiting it to a lower resolution.

Range: AUTO, CIF, VGA, HD720, HD1080

Default: AUTO

The **MAX_MS_SVC_RESOLUTION** System Flag operates independently from the **MAX_RTV_RESOLUTION** System Flag allowing differing maximum resolutions to be selected for the MS SVC and RTV protocols.

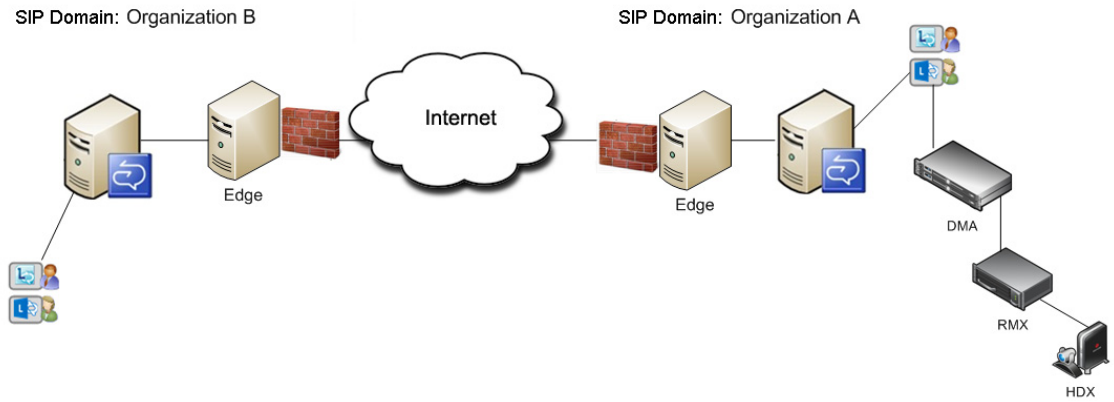
If you want to modify System Flag values, the flags must be added to the System Configuration file. For more information see: `Modifying System Flags` and `Controlling Resource Allocations for Lync Clients Using RTV Video Protocol` in the *Collaboration Server (RMX) Administrator's Guide*.

ICE Configuration

The Collaboration Server should be configured with ICE regardless of whether an EDGE Server is part of the configuration or not.

Federation Configuration

A secondary SIP domain can be added to the Lync 2013 environment. It must be configured as Federated.



For more information see: `Appendix H - Integration Into Microsoft Environments` in the *Collaboration Server (RMX) Administrator's Guide*.

System Flags for Cropping Control

Cropping occurs when the video source (endpoint) aspect ratio is different from the video cell aspect ratio in the Polycom Video Layout.

For all endpoints other than ITP endpoints and panoramic cells the Collaboration Server calculates the mismatch percentage between the source video aspect ratio and Polycom video layout cell aspect ratio. The mismatch percentage is used to determine whether cropping or striping will be applied to the video cell in the Polycom video layout.

For non-panoramic layouts, cropping and striping can be controlled by adding the **CROPPING_PERCENTAGE_THRESHOLD_GENERAL** System Flag and setting its value accordingly.

For panoramic layouts, cropping and striping can be controlled by adding the **CROPPING_PERCENTAGE_THRESHOLD_PANORAMIC** System Flag and setting its value accordingly.

For both System Flags:

Range: -1-100.

Default: -1

If the calculation result is less than or equal to the value of the CROPPING_PERCENTAGE_THRESHOLD_GENERAL System Flag, cropping will be applied.

If the calculation result is greater than the value of the CROPPING_PERCENTAGE_THRESHOLD_GENERAL System Flag, striping will be applied.

If the CROPPING_PERCENTAGE_THRESHOLD_GENERAL System Flag value is set to 0, cropping will not be applied.

If the CROPPING_PERCENTAGE_THRESHOLD_GENERAL System Flag value is set to -1 cropping will always be applied.

If the CROPPING_PERCENTAGE_THRESHOLD_GENERAL System Flag value is set to 100, always apply cropping with the exception of mobile aspect ratios. Mobile aspect ratios are those with a larger vertial aspect than horizontal for example 3:4.

Example: An iPhone Lync client sends 288x352 (3:4) aspect ratio. Cropping this resolution to 16:9 would require the cropping of 190 pixels on the vertical aspect. The mismatch calculation would yield 190/352 ~54. If the applicable System Flag's value is set to >= 54, striping, rather than cropping, is applied.

For more information see, `Modifying System Flags` in the *Collaboration Server (RMX) Administrator's Guide*.

Sharing Content during a Conference

Using Polycom CSS (Content Sharing Suite) plug in, Lync 2013 clients are able to share content in Polycom Content sessions in both Collaboration Server Hosted calls and Cascaded MS AV MCU calls, with both Lync 2010 & 2013 clients.

Content Sharing behavior is summarized in the following tables:

Content Sharing by Lync Version and Deployment Architecture

Lync Version	Collaboration Server Hosted	Cascaded MS AV MCU
Lync 2010	Supported, as in previous versions.	Not supported
Lync 2013	Supported.	Supported

Content Sharing Behavior by Lync Connection and MCU Type

Connection Type	Content Sharing Behavior
Point to point: Lync client to Lync client.	Microsoft Content is used for the entire session.
Lync calls MS AV MCU conference with no Collaboration Server cascade participant.	Microsoft content is used until Collaboration Server will joins, from which time it will switch to Polycom content
Lync calls VMR directly	Polycom Content is used for the entire session.
Collaboration Server joins an MS AV MCU conference before Content sharing is initiated.	Polycom Content is used until Collaboration Server leaves the conference.
Collaboration Server joins an MS AV MCU conference while Content is being shared.	When the CSS plug in of the Content speaker detects that Microsoft Content is being sent by the Lync client, it automatically stops the Microsoft Content and switches to Polycom Content and starts sending Polycom Content. Polycom Content is used until the Collaboration Server leaves the MS AV MCU conference.

CSS Behavior by Lync Content Type

Lync Content Type	CSS Behavior
Desktop Sharing	Send the desktop In cases where the is more than one monitor, the Lync client asks which monitor to use and the CSS will comply.
Program Sharing	Only the application is sent.
Power Point Sharing	Not supported. CSS should not send the Power Point to the Collaboration Server but the Lync clients will be able to send/receive the Power Point. CSS issues a notification in the Lync Content presenter device stating that the Power Point cannot be shared with non Lync devices.
Whiteboard Sharing	Not supported. CSS should not send the whiteboard to the Collaboration Server but the Lync clients will be able to send/receive the whiteboard. CSS issues a notification in the Lync Content presenter device stating that the whiteboard cannot be shared with non Lync devices.

Cisco TIP Support

Polycom's solution that allows the Collaboration Server to natively inter-operate with Cisco TelePresence Systems using Cisco TIP protocol is supported.

MLA (Multipoint Layout Application) is required for managing Cisco TelePresence layouts (whether Polycom ITP Systems are deployed or not. MLA is a Windows® application that allows conference administrators to configure and control video layouts for multipoint calls involving Polycom Immersive Telepresence (ITP) systems. For more information see the *Polycom® Multipoint Layout Application (MLA) User's Guide for Use with Polycom Telepresence Solutions*.

System behavior can be controlled by adding the **MS_AV_MCU_ MONITORING** System Flag and setting its value accordingly as summarized in the following table.

System Behavior by MS_AV_MCU_MONITORING System Flag Value and MLA Mode

MS_AV_MCU_MONITORING=	MLA Mode	Collaboration Server Side	MS AV MCU Side
MAIN_AND_IN_SLAVE (Default)	Room Switch	Sees the AV MCU current speaker.	Sees the Collaboration Server hosted current speaker in a 1x1 layout.
	CP Layout	Sees all connected Lync Clients in the layout.	
NO (Not recommended)	Room Switch	Sees a Lync Client which may or may not be the current speaker.	
	CP Layout		
YES	Room Switch	Sees the MS AV MCU current speaker.	
	CP Layout (Not recommended)	Sees cascaded slave MS AV MCUs as empty cells in the layout.	

For more information seeCollaboration With Cisco's Telepresence Interoperability Protocol (TIP)and Collaboration with Microsoft and Ciscoin the *Collaboration Server (RMX) Administrator's Guide.*

Lync 2013 Participant monitoring

Lync Clients connected to a conference using the Collaboration Server Hosted architecture will experience normal monitoring, with the addition of the MS SVC codec.

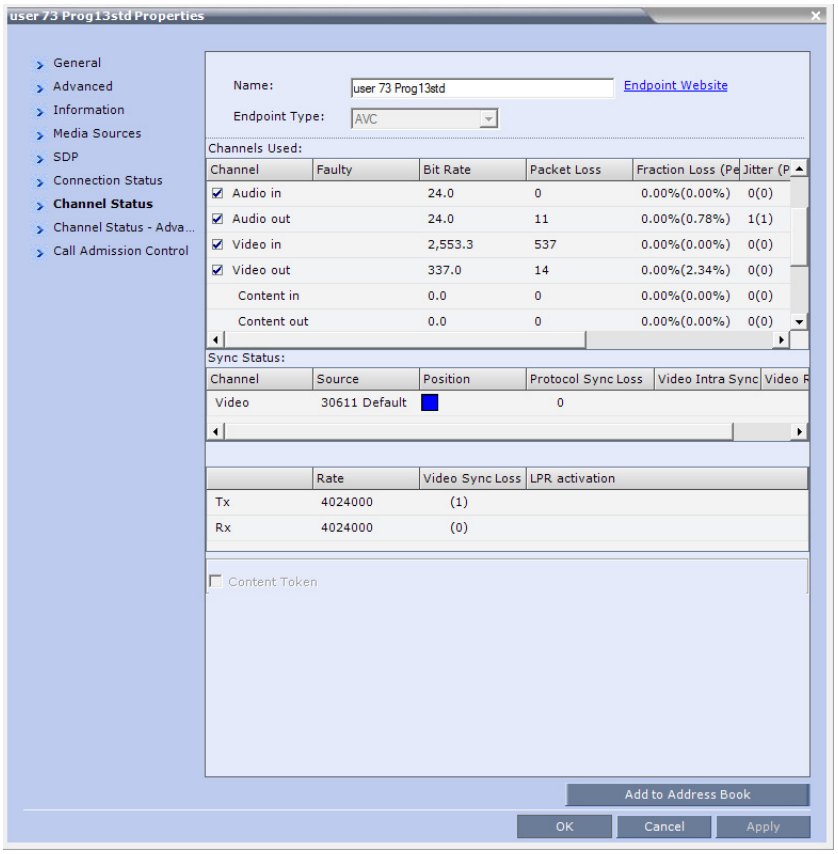
Lync Clients connected to a conference using a MS AV MCU Cascade link will be monitored as a single participant. In the Conference list the MS AV MCU is listed as *Lync AV MCU_x,* where *x* is an incrementing number, should multiple be multiple conferences connected using MS AV MCU Cascade links.

For more information see Microsoft RTV Video Protocol Support in CP Conferencesin the *Collaboration Server (RMX) Administrator's Guide.*

Monitoring Participant Properties - Channel Status Tab

Two Channel parameters for each MS AV MCU Cascade link are displayed: **Video in** and **Video out**.

For both Video in and Video out, the Bit Rate and Packet Loss parameters are displayed as aggregate values. For all other Channel parameters (Jitter, Latency, etc.) the highest values ar displayed for each video stream.



Monitoring Participant Properties - Channel Status - Advanced Tab

Media Info of each media stream sent by the MS AV MCU is displayed:

Stream name: The Display Name of the Lync client.

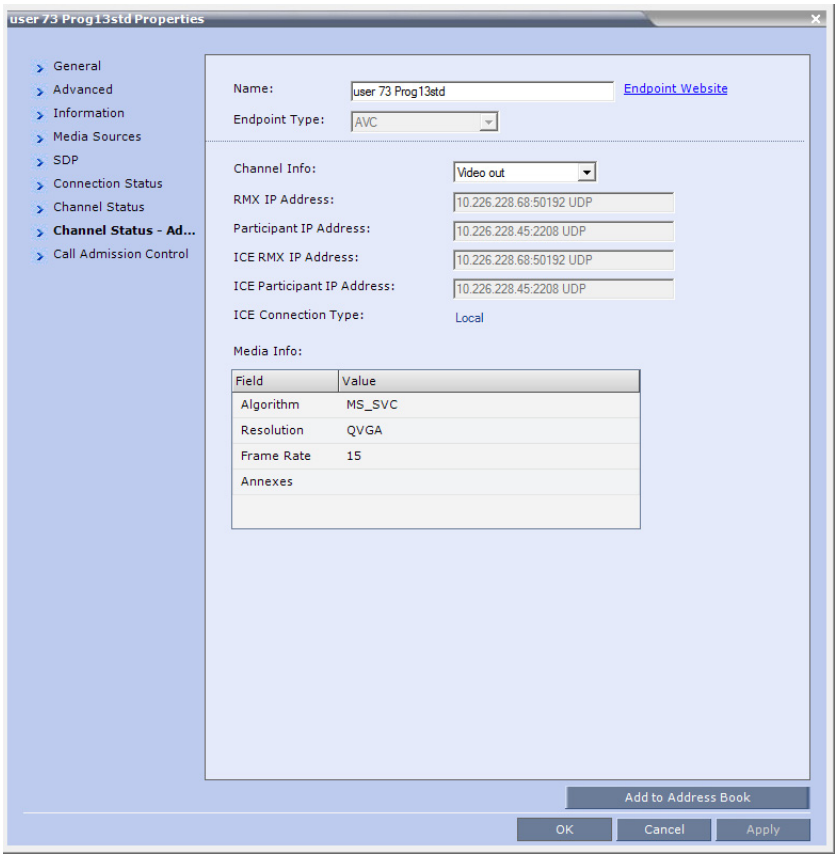
Algorithms: H.264 or RTV.

Resolution: CIF, SD, VGA, HD720 etc.

Frame Rate: 7.5, 15, 30 etc.

Annexes: Used for H.263 only

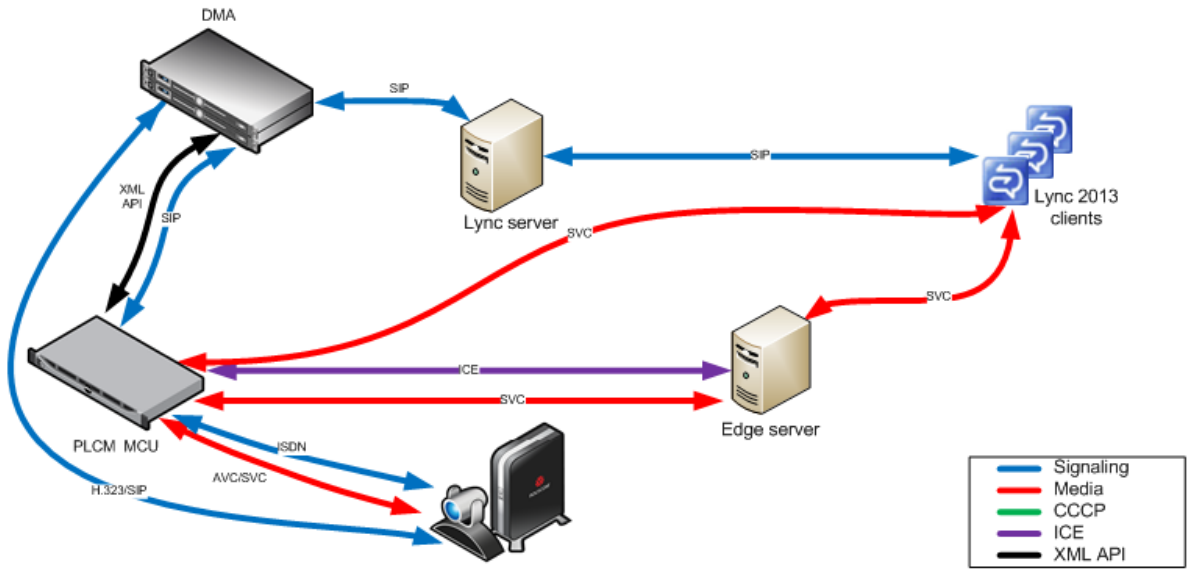
RTP Statistics are aggregated and are not detailed per stream.



For more information see Participant Level Monitoringin the Collaboration Server (RMX) Administrator's Guide.

Deployment Architecture 1 - Collaboration Server Hosted

Lync 2013 clients connect to a conference hosted on a Polycom Collaboration Server.



- H.264 SVC Lync clients are connected using UCConfig Mode 1:
  - The SVC Codec's Temporal Scaling capability is used to send one video stream to and from the Collaboration Server for each resolution at multiple frame rates.
  - H.264 SVC uses H.264 SEI messages to send stream layout information rather than SDP messages.
  - There is one audio stream per direction.
- Lync clients place calls to a Virtual Meeting Room provisioned on the DMA, for example, 1234@dma.example.com
- The Collaboration Server can connect Lync 2013 participants to either mixed AVC/SVC or to AVC only conferences.
- Lync 2013 clients dialing to a VMR, where the type of the conference is SVC/AVC mixed and AVC CP only have their video decoded. The Collaboration Server sends encoded video to Lync 2013 participants.

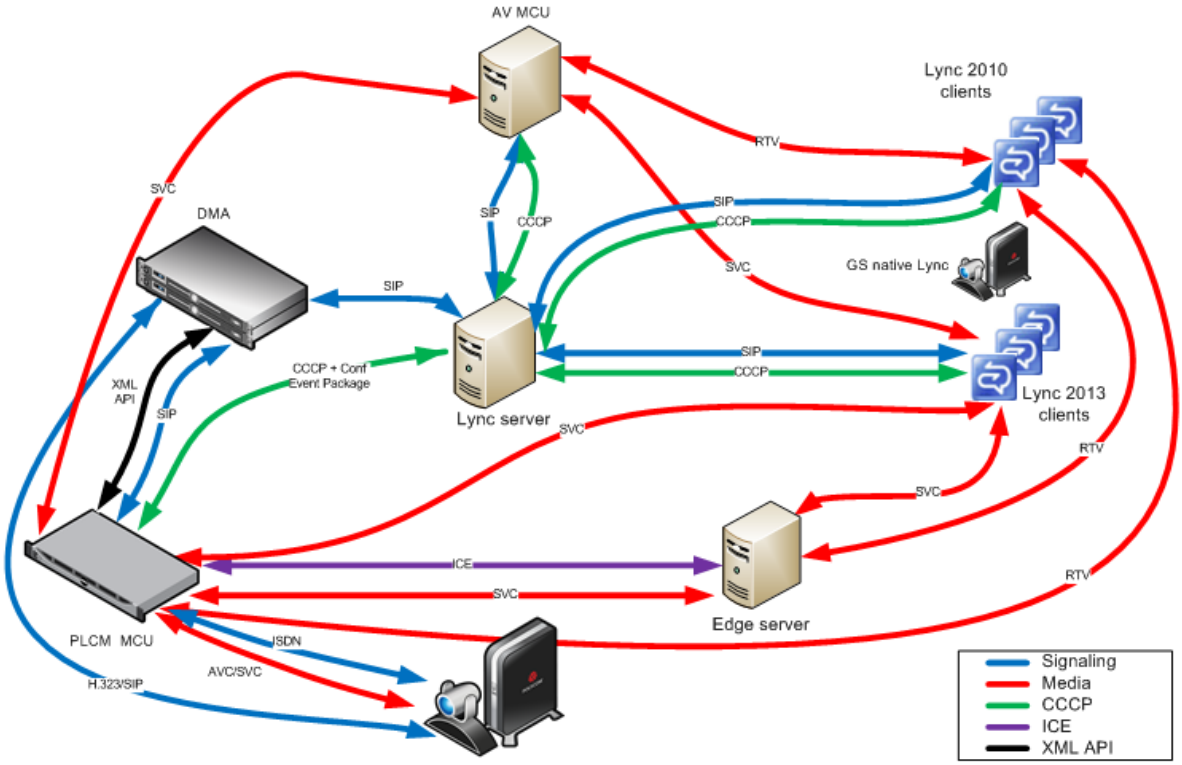
Look and Feel

All participants experience a Microsoft Point to Point conference with Polycom video layouts.

Only Classic Skin is supported. For more information about Skins, see `Defining AVC-Based Conference Profiles` in the *Collaboration Server (RMX) Administrator's Guide*.

Deployment Architecture 2 - MS AV MCU Cascade

Cascaded VMR Participants (Lync 2013 clients) connect to a conference on a Microsoft AV MCU which is cascaded with a Collaboration Server.



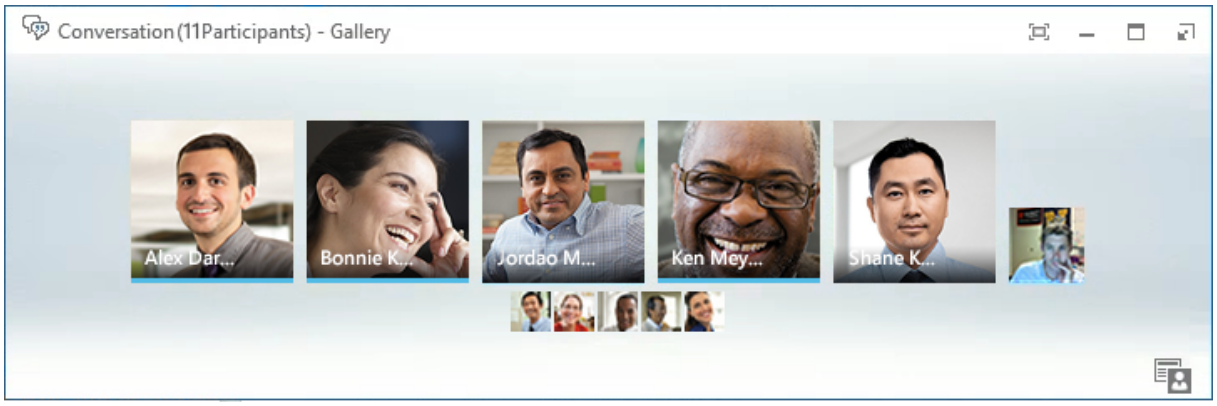
In this deployment architecture, participants connecting to the conference by way of the Collaboration Server are referred to as Cascaded VMR Participants.

- Lync clients place calls to a Virtual Meeting Room provisioned on the DMA, for example, 1234@dma.example.com
- The Cascaded VMR connects as a Lync client to the AV MCU.
  - The Collaboration Server utilizes the SVC Codec's Temporal Scaling capability to send up to three simulcast video streams to the AV MCU, each at multiple frame rates.
  - The Collaboration Server receives media from up to five different Lync clients from the AV MCU.
- Lync clients experience a Microsoft Point to Point conference with Cascaded VMR Participants appearing as Lync clients.
- Cascaded VMR Participants experience a conference with Polycom video layouts.
- Lync clients will see the active speaker from the
- Cascaded VMR while Cascaded VMR Participants will see up to 5 Lync clients in addition to other participants.

- Lync 2013 clients, connected by means of the Lync AV MCU, in point to point calls, can connect to Collaboration Server VMR participants by escalating the Lync call to a multipoint conference, including the Collaboration Server VMR meeting room and its participants.
  - The Buddy List can be used to select participants followed by, right-clicking and selecting Start a Video Call.
  - An ad-hoc (Meet Now) Lync conference can be started; a drag-and-drop operation can then be used in the Buddy List to add a Cascaded VMR to the conference.
- Cascaded VMR Participants, in point to point calls, can connect to Lync AV MCU participants by escalating the call to a multipoint Lync Conference.
  - A re-INVITE is issued to escalate the conference from point-to-point to multipoint.
- A re-INVITE can be issued from an ongoing Audio conference to escalate it, enabling connected participants to start sending video.
- Conferences hosted on a Collaboration Server can connect in cascade to only one AV MCU hosted conference.
- The Collaboration Server can host multiple conferences, each connected in cascade to a different AV MCU hosted conference. Conferences, connected in cascade to an AV MCU hosted conference cannot be connected in cascade to other Collaboration Servers.

Look and Feel for Lync clients and Group Series Endpoints

Lync clients and Group Series endpoints with native Lync capability connect to the AV MCU directly and experience a Lync look and feel conference and can see all Lync clients; up to 5 simultaneously in Gallery View.



If the active speaker is a Cascaded VMR Participant, the participant is seen by Lync Clients in a video window in the Gallery View.

By default, Cascaded VMR Participants are forced to a 1x1 layout by the default setting of the FORCE_1X1_LAYOUT_ON_CASCADED_LINK_CONNECTION System Flag. The flag's default setting is YES, which prevents a VMR layout being displayed within a Gallery View video window. If required, alternative Personal Layouts can be forced only after this System Flag has been added and its value set to **NO**.

For more information see, `Modifying System Flags` in the *Collaboration Server (RMX) Administrator's Guide*.

Look and Feel for Legacy Endpoints

Legacy endpoints will connect to the Collaboration Server and will be able to see all Lync clients; up to 5 simultaneously in Polycom video layouts. Only Classic Skin is supported. For more information about Skins, see `Defining AVC-Based Conference Profiles` in the *Collaboration Server (RMX) Administrator's Guide*.

Video Resource Requirement Selection in MS AV MCU Cascade

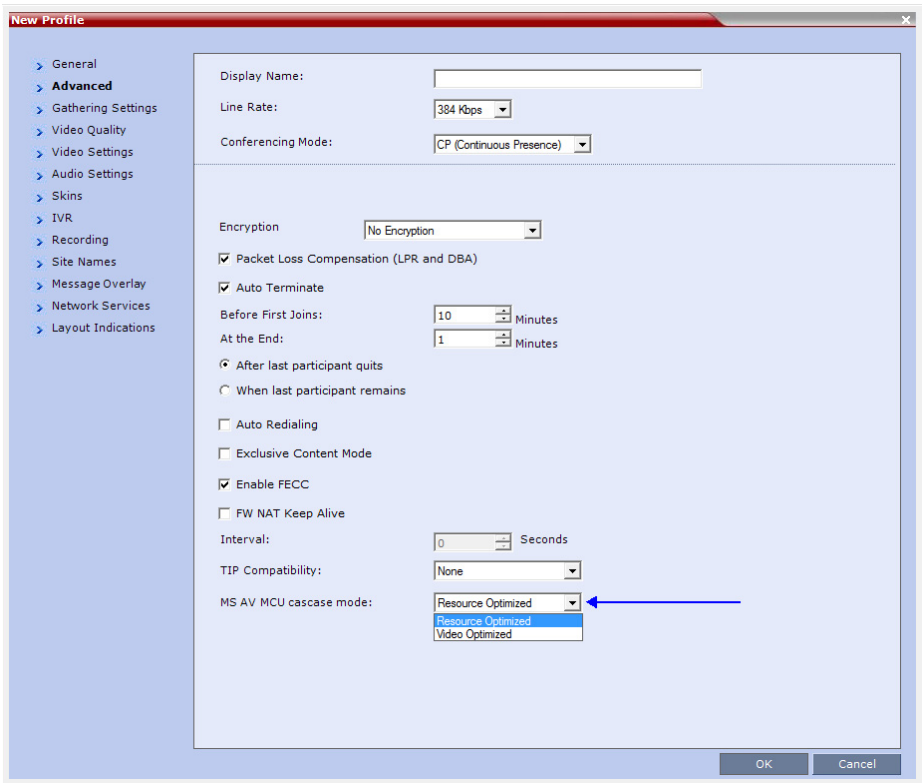
Collaboration Server resource usage in MS AV MCU Cascade can be configured in the conference Advanced tab of the Profile dialog by selecting either **Resource Optimized** or **Video Optimized**.

Resource Optimized

The Collaboration Server's Resolution Configuration menu, resolutions of up to HD540p30 (SD30) are supported, depending on the conference's profile setting.

Video Optimized

The Collaboration Server's Resolution Configuration menu, resolutions of up to HD720p30 are supported, depending on the conference's profile setting.



If the Collaboration Server has insufficient resources, endpoints will be connected at the lowest resolutions possible: CIF or SD. If the Collaboration Server has no available resources, endpoints will not be connected.

Calls that are initially connected as Audio Only will only have video resources allocated to them if they are escalated to video calls.

Video Forcing and Changing Layout in MS AV MCU Cascade

MS AV MCU Cascade behaves in the same manner as Collaboration Server to Collaboration Server Cascading.

The Conference Layout as well as the Personal Layouts of participants can be changed. Participants can be forced to appear in specific video cells of the layouts.

If Lync 2013 video streams are to be included in a Polycom Conference Layout, the Collaboration Server will remove these streams from the layout sent over the AV MCU cascade link.

If Lync 2013 video streams are to be included in a Personal Conference Layout, the Collaboration Server will not remove these streams from the layout to be sent over the AV MCU cascade link.

Handle Low Bit Rate Calls From the AV MCU

If the Collaboration Server, or a Group Series endpoint is connected to the AV MCU at a bit rate of 256kbps, the AV MCU transmits only one video stream even if it receives multiple video source requests.

At bit rates lower than256kbps (128kbps and 192kbps) the AV MCU does not transmit video.

This limitation can be controlled using the **DISABLE_LYNC_AV_MCU_128_AND_192_KBPS** System Flag. The flag must be manually added to the System Configuration and its value modified as required:

NO (Default)—The Collaboration Server sends 128kbps or 192kbps (according to the call rate) but will receive 256kbps for each incoming video stream.

YES—The Collaboration Server will not send or receive video from the Lync AV MCU. The connection is audio only.

For more information see, `Modifying System Flags` in the *Collaboration Server (RMX) Administrator's Guide*.

Remove Empty Cells From the Video Layout

Empty cells in the Video Layout can occur as result of the following causes:

Case 1 — A camera connected a PC that is hosting a Lync client is switched off, the cell in which the Lync client was displayed remains in the video layout and is empty.

Case 2 — A Lync 2013 Client is connected using a CIF port at a bit rate that exceeds 192kbps.

Case 1

The empty cell can be removed from the video layout by adding the **REMOVE_EP_FROM_LAYOUT_ON_NO_VIDEO_TIMER** System Flag and setting its value as required.

Range: 0 – 19 (seconds): The feature is disabled.

20 – 300 (seconds): The feature is enabled.

Default: 20

When enabled (flag value 20 - 300), the endpoint is removed when the empty cell is detected, and the cell is used for another participant if:

- No video RTP messages are received from the EP for the defined timer value in addition to one of the following timers, depending on the call type:
  - DETECT_SIP_EP_DISCONNECT_TIMER
  - DETECT_H323_EP_DISCONNECT_TIMER

- Either the PRESERVE_PARTY_CELL_ON_FORCE_LAYOUT System Flag =NO

  or

  The endpoint is not forced in the layout.

In Lync environments that do not include ICE, the empty cell will remain in the layout for direct Lync calls.

Case 2

The **RTV_MAX_BIT_RATE_FOR_FORCE_CIF_PARTICIPANT** System Flag has been added to the system with a default value of 192 (kbps). This ensures that the Lync Client sends the correct resolution and that its cell in the Video Layout is displayed correctly.

Configuring the Collaboration Server as a Trusted Application for Lync 2013

Registering the Collaboration Server as a Trusted Application for Lync 2010/2013

The following procedures are mandatory to register the Collaboration Server to Lync 2010/2013.

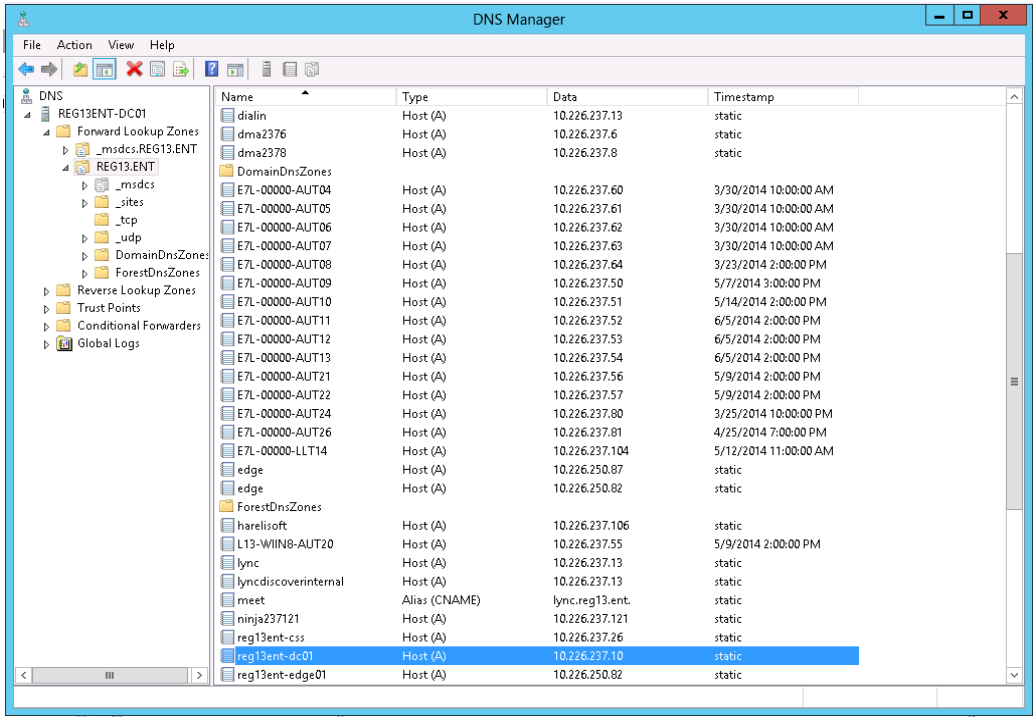
In the following procedures Domain Names, Server Names and Collaboration Server Names are examples for syntax purposes only and must be adapted to local network requirements.

Configure the Collaboration Server FQDN in the DNS

Perform the following steps on the Lync Server to configure the Collaboration Server FQDN in the DNS.

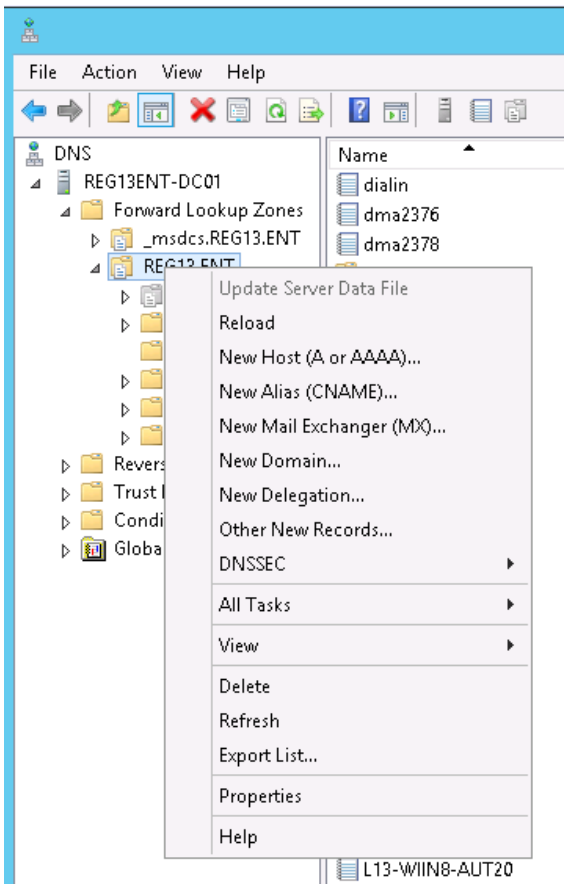
To Configure the Collaboration Server FQDN in the DNS:

1. Connect to the DNS Server.
2. Open the DNS Manager.

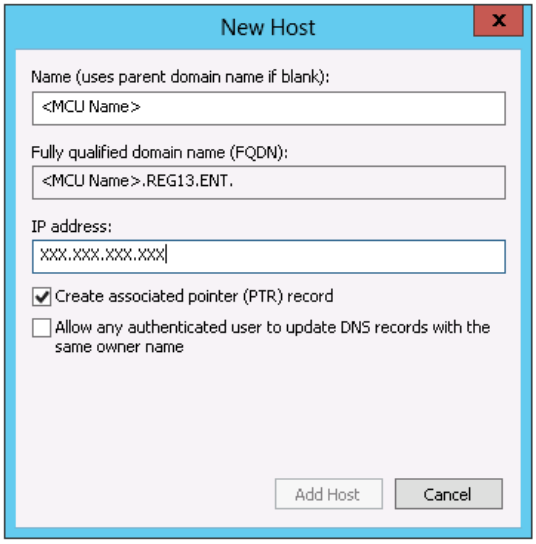


3 Navigate to and expand the **Forward Lookup Zones** folder.

4 Right click on the **Zone** for your Domain.



5 Select **New Host** (A or AAAA).



6 Enter the Collaboration Server **Name** and the Signaling **IP address**.

7 Click **Add Host**.

Configure Collaboration Server Static Route and Trusted Application

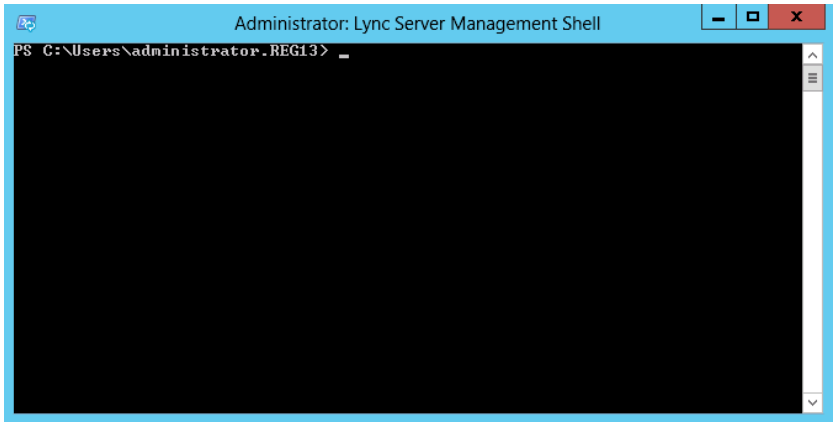
Perform the following steps on the Lync Server to configure the Collaboration Server Static Route and Trusted Application.

To Configure the Collaboration Server Static Route and Trusted Application:

1 Connect to Lync Front End server

2 In the Start Menu Search field, enter **Lync** and click the **Search** button.

3 Select the **Lync Server Management Shell**.



4 Add a **Static Route**:

a Type: **$route = New-CsStaticRoute -TLSRoute -destination "<RMX FQDN>" -port 5061 -matchuri "<Collaboration Server>" -usedefaultcert $true** and press **Enter**.

<Collaboration Server> is the Collaboration Server name.

b Type: **Set-CsStaticRoutingConfiguration -identity global -route @{Add=$route}** and press **Enter**

(To check the Static Route configuration enter the following command: **Get-CsStaticRoutingConfiguration** and press **Enter.**)

5 Create a **Trusted Application Pool** and a **Trusted Application**:

a Type: **New-CsTrustedApplicationPool -Identity <Collaboration Server FQDN> -Registrar Registrar:<Lync pool> -site 1 -ComputerFqdn <Collaboration Server FQDN> -ThrottleAsServer $true -TreatAsAuthenticated $true** and press **Enter**.

b Enter **YES**.

c Type: **New-CsTrustedApplication -ApplicationId <FQDN> -TrustedApplicationPoolFqdn <FQDN> -Port 5061** and press **Enter**.

ApplicationId is the name of the application. This must be a string that is unique within the pool that is specified in the **TrustedApplicationPoolFqdn** parameter.

TrustedApplicationPoolFqdn is the FQDN of the Trusted Application Pool in which the application will reside.

6 Add a **Trusted Application Endpoint** by typing: **New-CsTrustedApplicationEndpoint -sipaddress sip:<name>@<domain> -ApplicationId <FQDN> -TrustedApplicationPoolFqdn <FQDN>** and pressing **Enter**.

7 Enable the changes, by typing: **Enable-CsTopology**.

8 Get the **GRUU** (Globally Routable User-Agent URI):

a Type: **Get-CsTrustedApplication -Filter "<MCU>*" | fl ServiceGruu** and press **Enter**.

The Lync Server will reply with a string similar to the following:

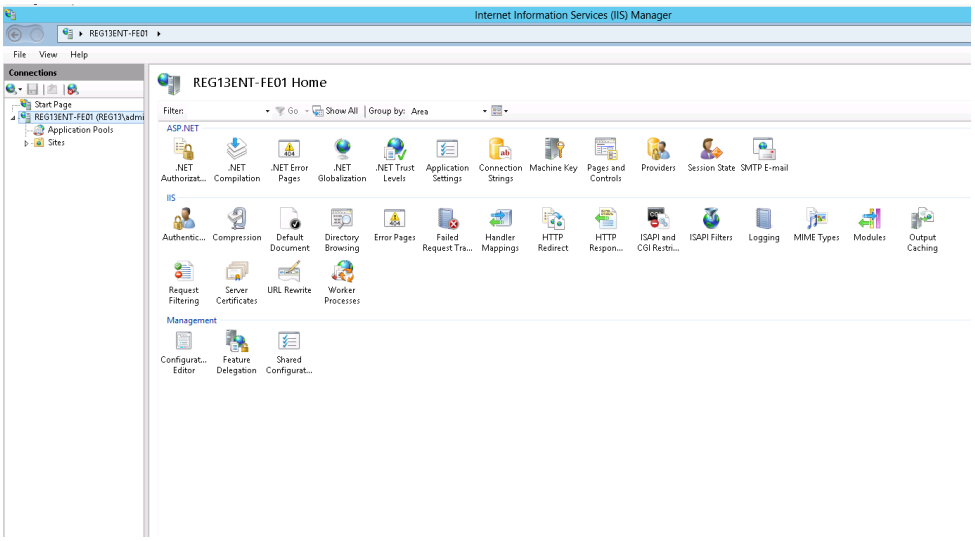
ServiceGruu:sip:rmx23772.reg13.ent@reg13.ent;gruu;opaque=srvr:rmx23772.reg13.ent:0l2d_trGDFSQv4FntDXH-wAA

b Select, Copy, and transfer the **ServiceGruu** string into the workstation that is connected to the Collaboration Server. The SIP prefix (**sip:**) must be omitted from the copied string.

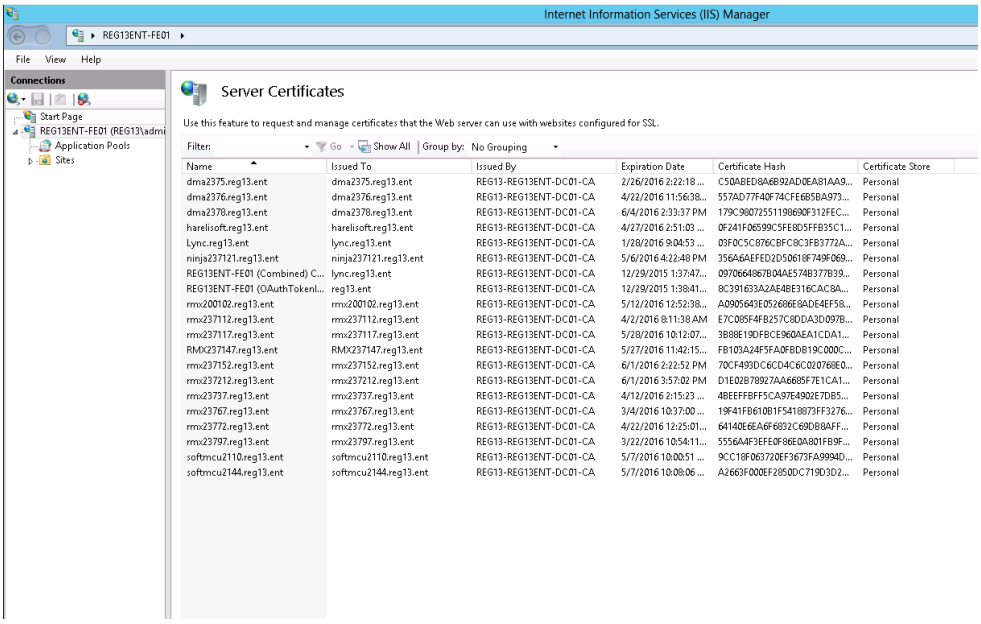
9 Create and export a **Certificate** for the Collaboration Server:

a Type: **Request-CsCertificate -New -Type Default -KeyAlg RSA -CA <CA FQDN> -City PT -State Israel -ComputerFqdn <Collaboration Server FQDN> -Country IL -DomainName <Domain name> -FriendlyName <Collaboration Server FQDN> -Organization 'Polycom ' -PrivateKeyExportable $true** and press **Enter**.

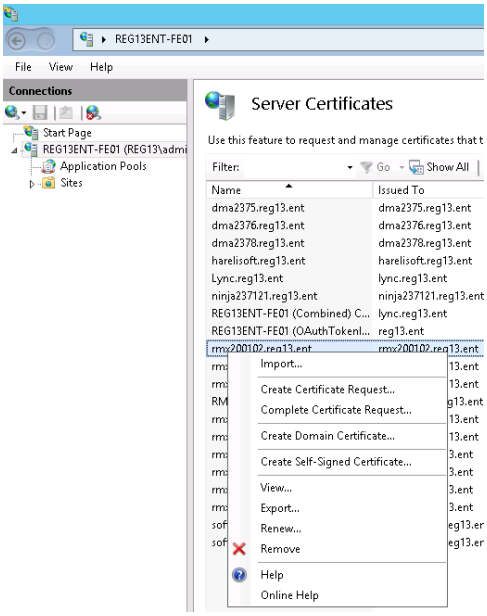
b In the Start Menu, select **Administrative Tools-> Internet Information Services (IIS) Manager**.



c Click **Server Certificates**.

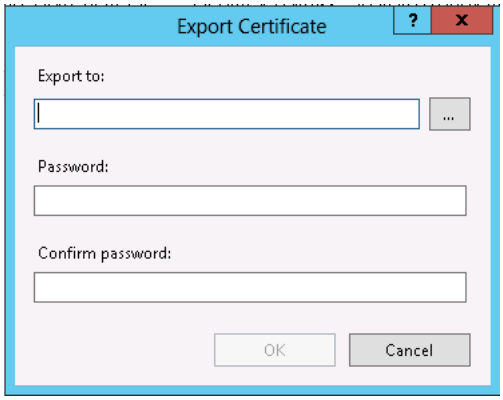


d In the Certificate List, locate the Collaboration Server's certificate.



e Right click the Collaboration Server's certificate.

f Click **Export**.



g In the **Export to** field, enter the file location for exported certificate.

h In the **Password** and **Confirm Password** fields, enter the password.

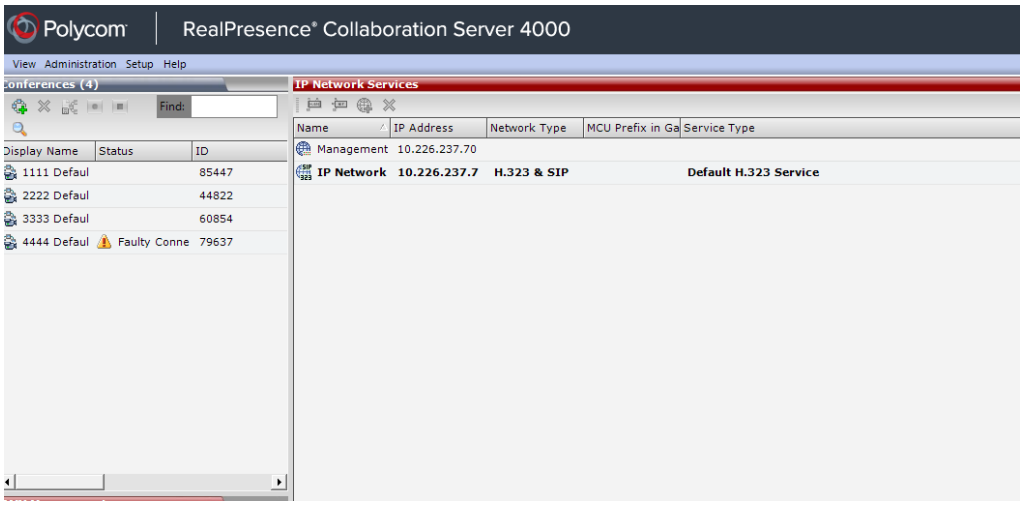
i Click **OK**.

Configure the Collaboration Server for Lync 2010/2013

Perform the following steps on the Collaboration Server to configure it for Lync 2010/2013.

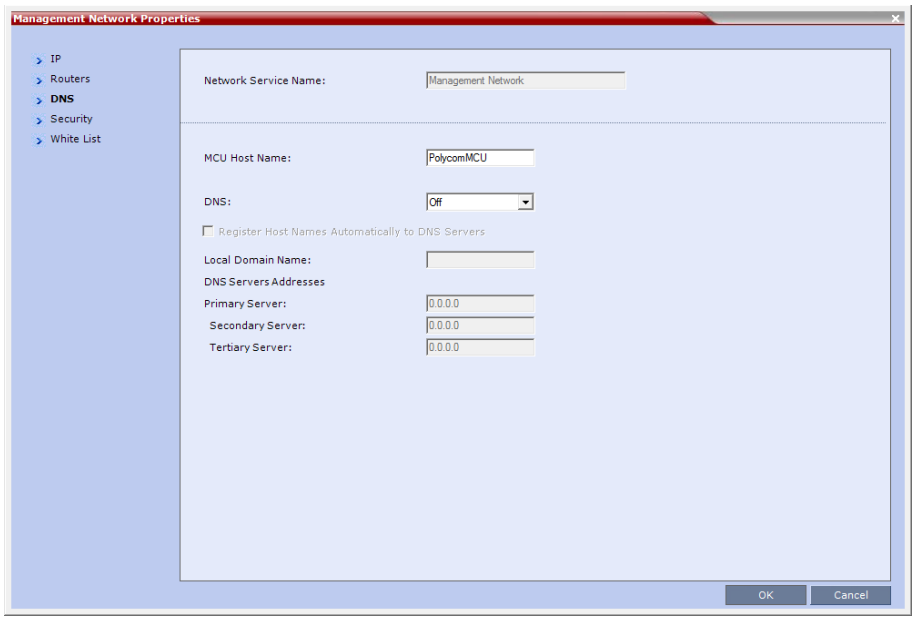
To Configure the Collaboration Server for Lync 2010/2013:

1 Open the IP Network Management Services list.



2 Double click **Management Network**.

3 Click the **DNS** tab.



4 In the MCU Host Name field, enter the name of the DNS (as listed in the DNS Manager).

5 In the Local Domain Name field, enter the Domain Name (as listed in the DNS Manager).

6 In the DNS Servers Primary Server field, enter the DNS IP Address (as listed in the DNS Manager).

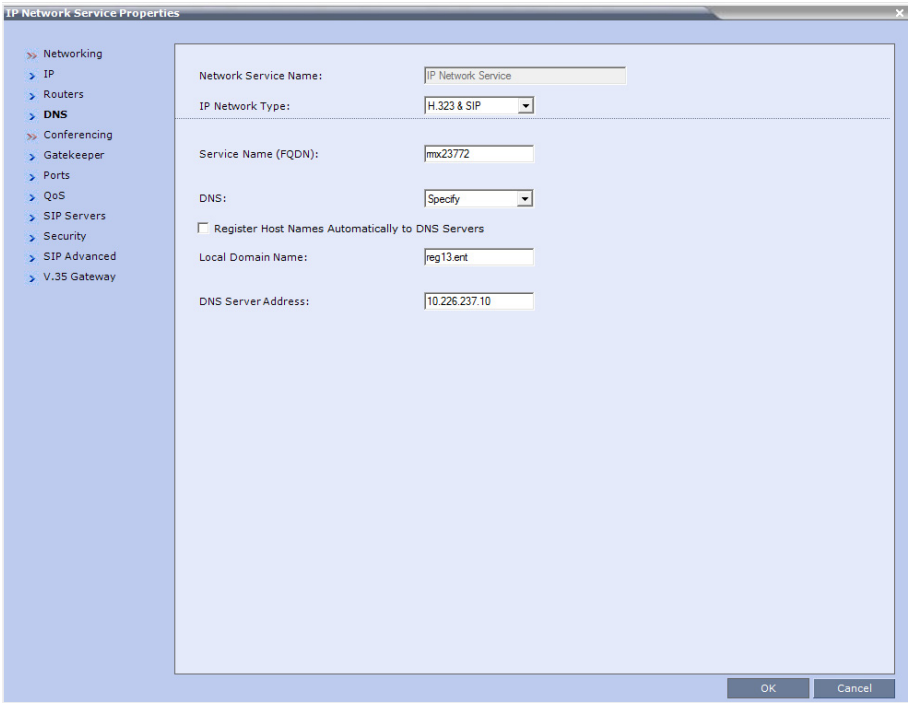
7 Click **OK**

8 Restart the Collaboration Server.

9 Login again and open the IP Network Management Services list.

10 Double click **IP Network Service**.

11 Click the **DNS** tab.



12 In the IP Network Type drop down menu, select **H.323 &SIP**.

13 In the Service Name (FQDN) field, enter the DNS Name (as listed in the DNS Manager).

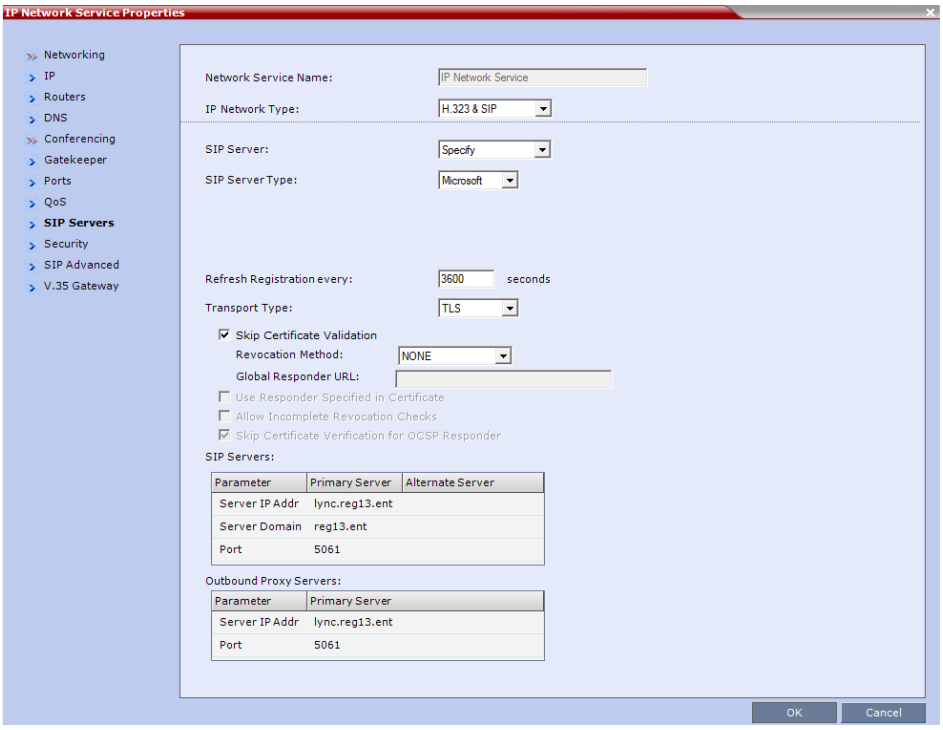
This should not be the FQDN.

14 In the DNS drop down menu, select **Specify.**

15 In the Local Domain Name field, enter the Domain Name (as listed in the DNS Manager).

16 In the DNS Server Address field, enter the IP Address of the DNS server (as listed in the DNS Manager).

17 Click the **SIP Servers** tab.



18 In the IP Network Type drop down menu, select **H.323 &SIP**.

19 In the SIP Server drop down menu, select **Specify.**

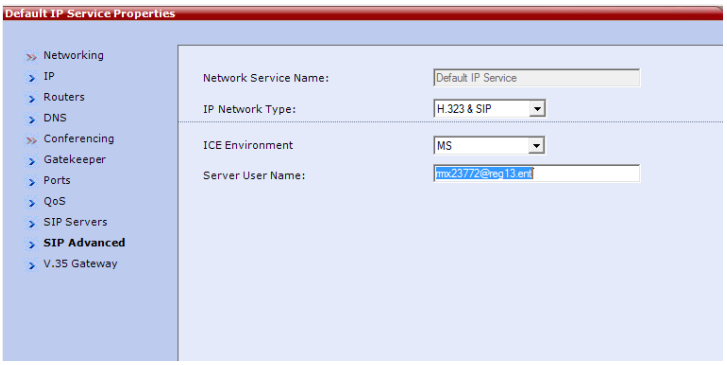
20 In the SIP Server Type drop down menu, select **Microsoft.**

21 In the Transport Type drop down menu, select **TLS.**

22 In the SIP Servers and Outbound Proxy Servers Parameter tables:

- Set the Server IP Address to the DNS Pool name (as listed in the DNS Manager).
- Set the Server Domain to the Domain Name (as listed in the DNS Manager).
- Set Port to 5061

23 Click the **SIP Advanced** tab.



24 In the Server User Name field, enter the Collaboration Server's SIP Address.

This is the first segment of the **ServiceGruu** string with the an "@" replacing the "."

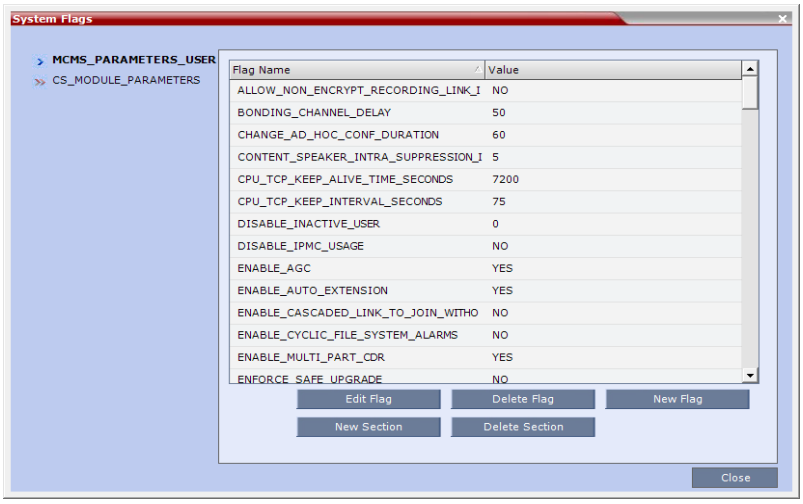
For example if the **ServiceGruu** string was:
rmx23772.reg13.ent@reg13.ent;gruu;opaque=srvr:rmx23772.reg13.ent:0I2d_trGDFSQv4FntDXH-wAA,

the Server User Name would be **rmx23772@reg13.ent**

25 Add the SIP_CONTACT_OVERRIDE_STR system flag with the ServiceGruu string as its value.

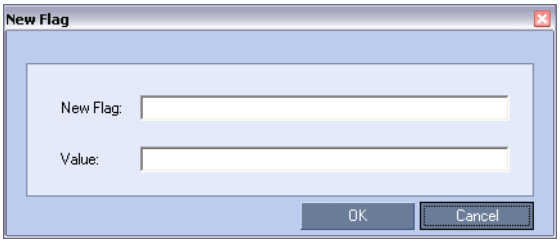
a On the workstation that is connected to the Collaboration Server, on the RMX Menu, click **Setup > System Configuration**.

The System Flags dialog opens.



b In the System Flags dialog, click the **New Flag** button.

The New Flag dialog box is displayed.



c In the New Flag field, enter **SIP_CONTACT_OVERRIDE_STR**.

d Locate, Copy and Paste the **ServiceGruu** string, that was transferred to the workstation in **Step 8b** of the **Configure the Collaboration Server Static Route and Trusted Application** above, into the Value field.

e Click **OK** to close the New Flag dialog.

26 Add the **LIMIT_CIF_SD_PORTS_PER_MPMX_CARD** System Flag and set its value to **YES**.

Setting this flag's value to YES will limit the number of CIF and SD ports (including non Lync clients) to 45 per media card. The flag value and must be set to YES when ICE is active because ICE ports consume more system resources. HD participants are not affected.

27 Repeat Steps **25b - 25e** above with:
- New Flag field: **LIMIT_CIF_SD_PORTS_PER_MPMX_CARD**.
- Value field: **YES**.

28 Reset the Collaboration Server for the new flag settings to take effect.
- **a** Click **OK** to close the New Flag dialog.
- **b** Click OK to close the System Flags dialog.
- **c** Reset the Collaboration Server.

29 After the rest has completed, Login to the Collaboration Server again.

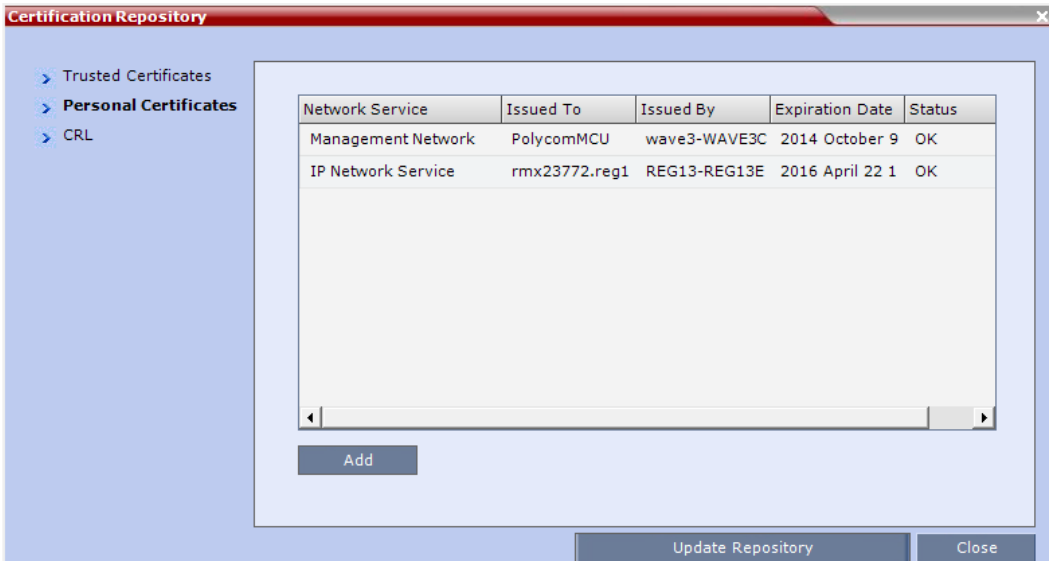
Import and install the Certificate on the Collaboration Server

The Certificate created in **Step 9** of the **Configure Collaboration Server Static Route and Trusted Application** procedure above must be imported an installed on the Collaboration Server.

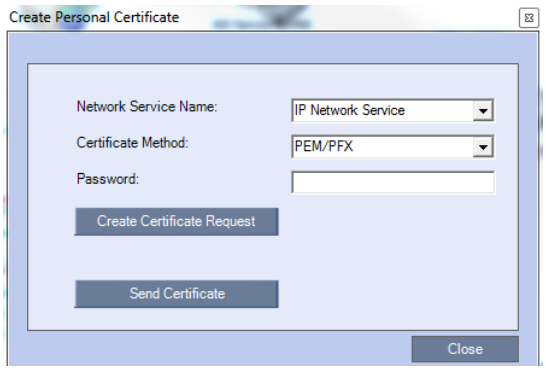
To import and install a certificate:

1 In the Collaboration Server menu, click **Setup > RMX Secured Communications > Certificate Repository**.

2 Click the **Personal Certificates** tab.



3 Click **Add**.

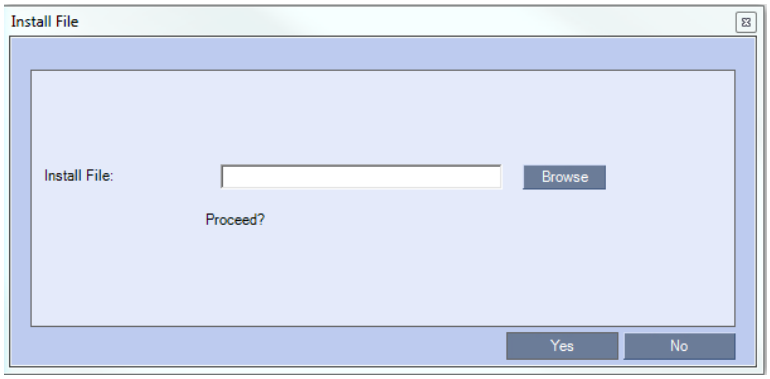


4 In the **Network Service Name** drop down menu, select **IP Network Service.**

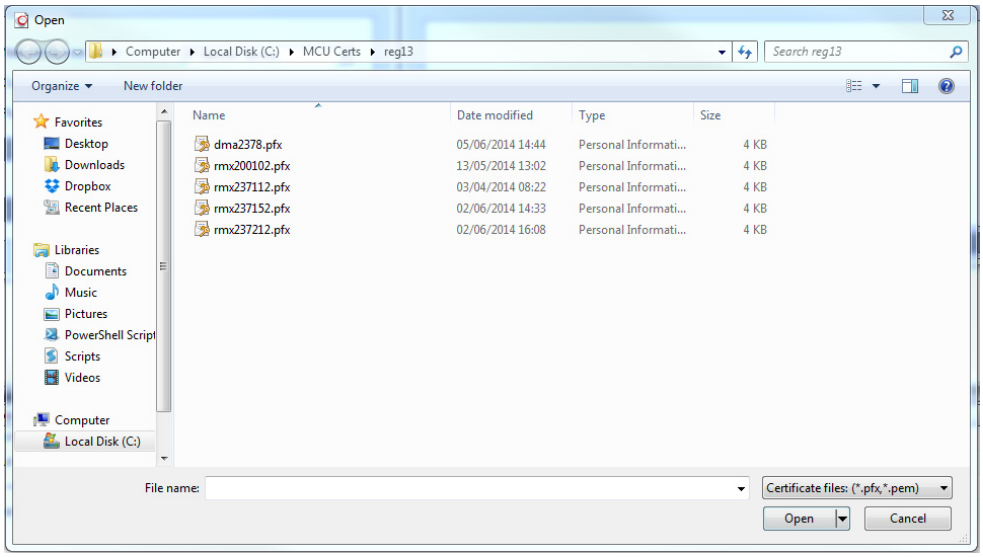
5 In the **Certificate Method** drop down menu, select **PEM/PFX.**

6 In the **Password** field, enter the password entered in **Step 9h** of the **Configure Collaboration Server Static Route and Trusted Application** procedure.

7 Click **Send Certificate**.

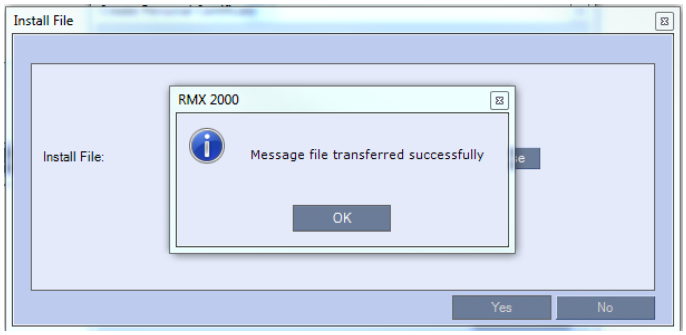


8 Click **Browse,** and then navigate to the folder where the certificate was saved.



9 Select the certificate and click **Open**.

The certificate is installed and a Confirmation Message is displayed.



10 Click **OK** in the confirmation message.

A System Reset confirmation message is displayed.

11 If you are ready to reset the system, click **YES**.

The System Reset completes the Registration of the Collaboration Server to Lync 2010/2013.

Appendix I - Polycom Open Collaboration Network (POCN)

Working in the Open Collaboration Server and TIP protocol are supported in AVC Conferencing Mode only.

Collaboration With Cisco's Telepresence Interoperability Protocol (TIP)

TIP is a proprietary protocol created by Cisco for deployment in Cisco TelePresence systems (CTS). Since TIP is not compatible with standard video communication systems, interoperability between Cisco and other vendors' Telepresence systems was initially impossible.

Gateways were developed to provide interoperability but were subject to the inherent problems of additional latency (delay) in connections and low video quality resulting from the reformatting of video and audio content.

Polycom's solution is to allow the Collaboration Server to natively inter-operate with Cisco TelePresence Systems, ensuring optimum quality multi-screen, multipoint calls between:

- Polycom Immersive Telepresence Systems (ITP) Version 3.1.1:
  - RPX 200
  - RPX 400
  - OTX 300

    (At Telepresence Licence is required on the Collaboration Server.)
- Polycom video conferencing endpoints
  - Standalone HDX
  - Polycom Group Series 300/500
- Microsoft
  - MS Lync (using MS-ICE)
  - RTV 720p
- Cisco TelePresence® System (CTS) Versions 1.10Collaboration ServerCollaboration ServerCollaboration Server
  - CTS 1300
  - CTS 3010

Conferences hosted on the Collaboration Server can include a mix of existing end points (that do not support TIP) and CTS endpoints.

TIP-enabled endpoints must support TIP Version 7 or higher. Calls from endpoints supporting older versions of TIP will be rejected.

Deployment Architectures

The following multipoint topologies are given as examples. Actual deployments will depend on user requirements and available infrastructure:

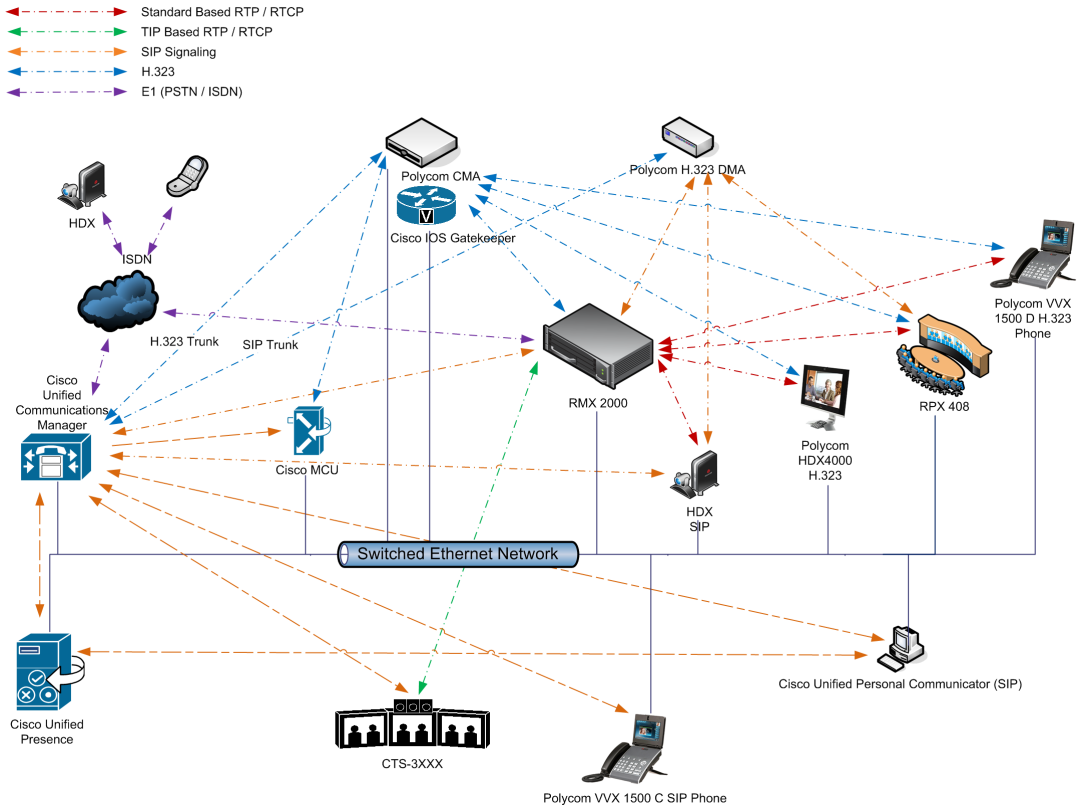
- Single company with Polycom and Cisco Infrastructure
  - CTS and Polycom Telepresence Rooms in a corporate environment.
- Company to company via Service Provider
  - **Model 1:** Mixed Polycom and Cisco infrastructure at one of the companies, Cisco only infrastructure at the other.
  - **Model 2:** Polycom only infrastructure at one of the companies, Cisco only infrastructure at the other.

Single Company Model - Polycom and Cisco Infrastructure

The deployment architecture in Single company with Polycom and Cisco Infrastructure - Polycom endpoints using SIP shows a company that has a mixture of Polycom and Cisco endpoints, room systems and telephony equipment that needs to enable multipoint calls between all its video and audio endpoints using the Collaboration Server as the conference bridge.

As shown in Single company with Polycom and Cisco Infrastructure - Polycom endpoints using SIP, prior to Version 8.1.1, Cisco Telepresence endpoints could connect to conferences using the TIP protocol, Polycom endpoints connected to the same conferences using SIP protocol.

Single company with Polycom and Cisco Infrastructure - Polycom endpoints using SIP



Polycom endpoints can also connect to Entry Queues, Meeting Rooms and conferences using all protocols, including TIP and SIP.

The following table lists components and versions of the Collaboration Server and Cisco Telepresence Systems (CTS) Integration Solution Architecture.

Solution Architecture Components

Component	Version	Description
CISCO Equipment		
CUCM	8.5.1, 8.6.2	Cisco Unified Communication Manager: CUCM must be configured to: • Route calls to DMA (if present). • Route all H.323 calls to the gatekeeper, which can be either CMA or IOS.
IOS	15.1T	Cisco Internetwork Operating System - Gatekeeper
Endpoints (CTS)	1.7.2 (ATT), 1.8.1	Telephony, desktop and room systems. • CTS endpoints must register to CUCM.

Component	Version	Description
Cisco Unified Video Conferencing 5230	7.2	MCU.
Cisco Unified Presence	8.5, 8.6	Network-based Presence and Instant Messaging.
Cisco Unified Contact Center Express	8.0, 8.5	Call distributor (ACD), interactive voice response (IVR) and computer telephony integration (CTI).
Cisco IP Communicator	7.0,8.6	Windows PC-based softphone application.
Cisco Unified Personal Communicator	8.5(2),8.5(5)	Web client for Presence and Instant Messaging.
Cisco Unified Video Advantage	2.2(2)	Video telephony functionality for Cisco Unified IP phones.
Cisco Unified IP Phones 7960, 7961, 7962, 7965, 7975	CUCM 8.5.1 / CUCM 8.6.1 compatible	IP Phones.
Cisco Unified IP Phones 9971	CUCM 8.5 / CUCM 8.6(2) compatible	
CTMS	1.7.3, 1.8.2	Cisco TelePresence Multipoint Switch.
Cisco Unified Border Element	15.1T	SBC - Voice and video connectivity from enterprise IP network to Service Provider SIP trunks.
Telepresence Server	2.2(1.54)	Telepresence Server.
VCS	X7.1	Video Communication Server / Session Manager.
Polycom Equipment		
DMA 7000	4.0	Polycom Distributed Media Application • *DMA* is an optional component but is essential if *Content* sharing is to be enabled. • All *SIP* endpoints register to *DMA* as a *SIP Proxy*. • DMA should be configured to route SIP calls (with CTS destination) to CUCM. If *DMA* is not present in the solution architecture, *SIP* endpoints must register to *CUCM* as gatekeeper. • *DMA* must be configured with a *VMR* (*Virtual Meeting Room*). Incoming calls are then routed to the Collaboration Server.

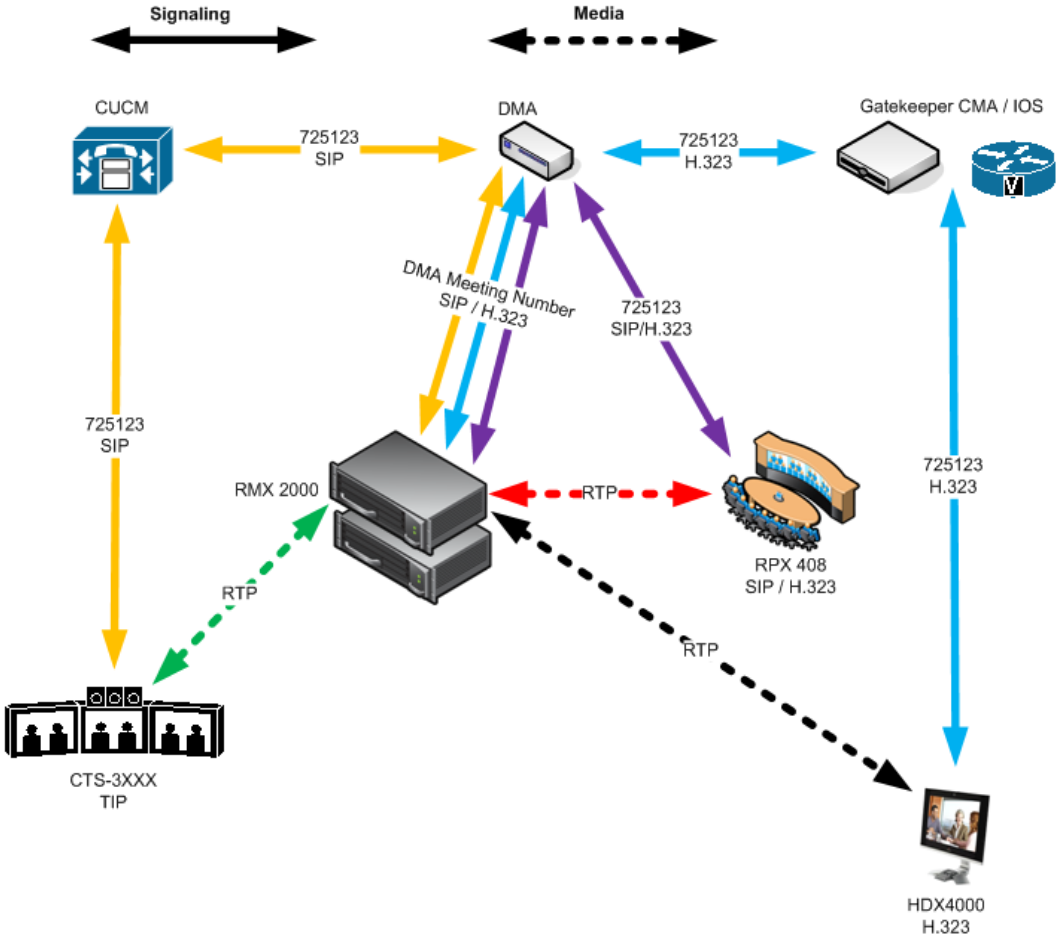
Component	Version	Description
Collaboration Server	7.6 and higher	MCU: • Functions as the network bridge for multipoint calls between *H.323*, *SIP* and *TIP* endpoints. • The Collaboration Server can be interfaced to *CUCM* using a *SIP* trunk, enabling *CTS* to join multipoint calls on Collaboration Server. Signaling goes through the *CUCM* while the media in *TIP* format goes directly between the *CTS* and Collaboration Server. • The Collaboration Server must be configured to route outbound SIP calls to DMA. • The H.323 Network Service of the Collaboration Server should register it's dial prefix with the *CMA* gatekeeper. • When DMA is not used an *Ad-hoc Entry Queue*, designated as *Transit Entry Queue*, must be pre-defined on the Collaboration Server.
MLA	3.0.3	Multipoint Layout Application *R*equired for managing multi-screen endpoint layouts for *Cisco CTS 3XXX, Polycom TPX*, *RPX* or *OTX* systems.
CMA	5.5	Polycom Converged Management Application - Gatekeeper • The gatekeeper must route calls to Collaboration Server based on the Collaboration Server prefix registration on the gatekeeper.
Endpoints		Telephony, desktop and room systems. • H.323 endpoints must register to the *CMA* or *IOS* gatekeeper. • *Polycom SIP* endpoints must register to *DMA* as *SIP Proxy* when *DMA* is used. • *H.323* endpoints must register to the *CMA* or *IOS* gatekeeper.

Call Flows

Multipoint call with DMA

In this example:

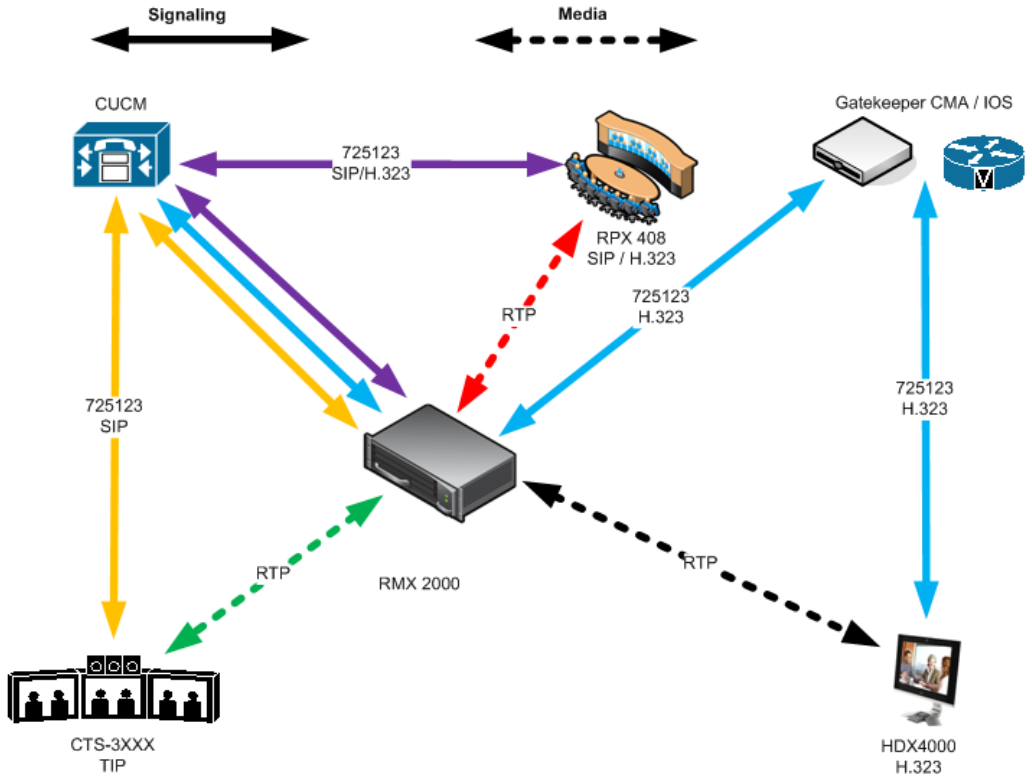
- Collaboration Server prefix in the gatekeeper: 72
- Virtual Meeting Room in DMA: 725123
- DMA Meeting Number: Generated by DMA



Multipoint call without DMA

In this example:

- Collaboration Server prefix in the gatekeeper: 72
- CUCM: According to its Dial Plan forwards calls with prefix 72 to the Collaboration Server



Company to Company Models Using a Service Provider

Using this topology, both companies connect to a Service Provider via a Cisco Session Border Controller (SBC). The Service Provider functions as a B2B Telepresence Exchange, enabling multipoint calls between the two companies and their respective video and audio endpoints using the Collaboration Server as the conference bridge.

The SBC functions as a firewall that the Service Provider can configure according to Trust Relationships between two or several companies. By using this method, companies do not have to open their corporate firewalls and administer connectivity with the many companies they may need to communicate with.

Two topology models are discussed:

- **Model 1:**
  - Company A has a Polycom only environment.
  - Company B has a Cisco only Environment.
- **Model 2:**
  - Company A has a mixed Polycom and Cisco environment.
  - Company B has a Cisco only Environment.

Model 1

The deployment architecture in Call Flow shows two companies: Company A and Company B.

Company A - has deployed a Polycom solution including:

- DMA
- Collaboration Server
- MLA
- CMA Gatekeeper
- Polycom telephony and desktop endpoints.

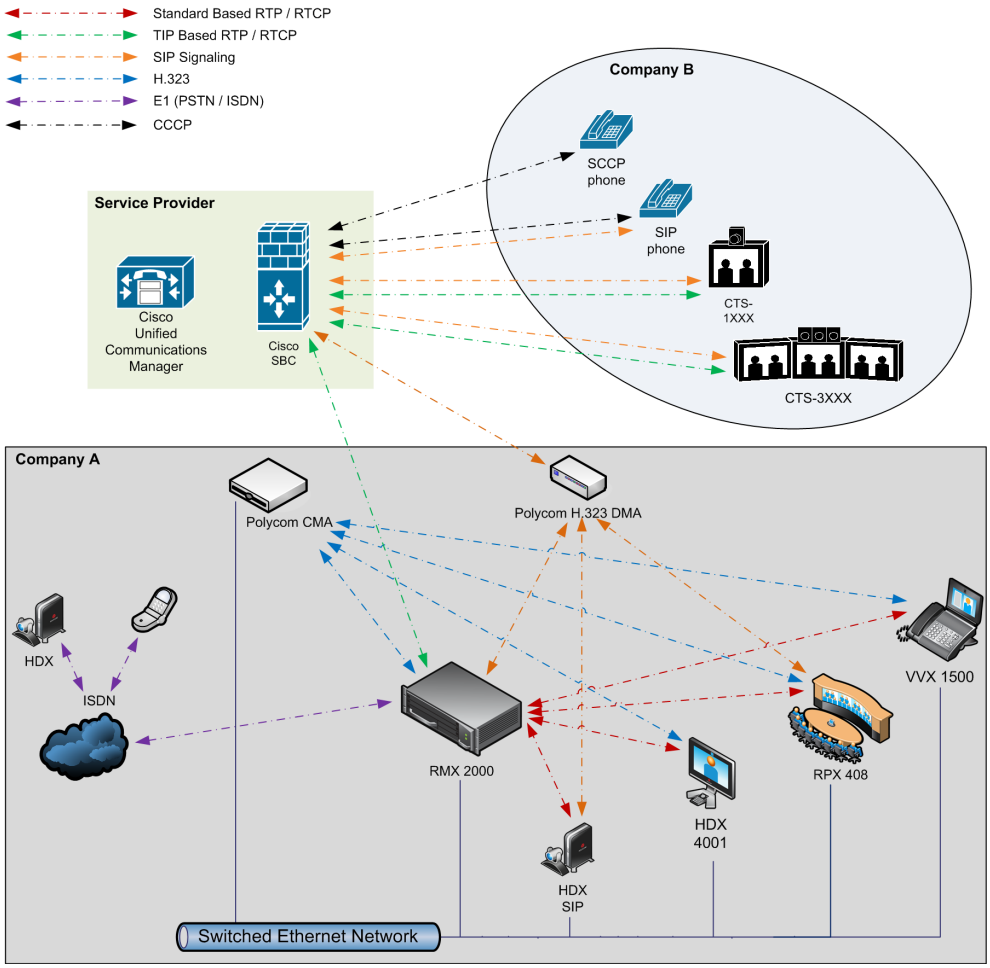
The roles of the Polycom components are described in the Polycom Equipment section of the Solution Architecture Components table.

Company B - has deployed a Cisco solution including:

- CTS 1000
- CTS 3000

- Cisco telephony and desktop endpoints

Company to Company via Service Provider - Model 1

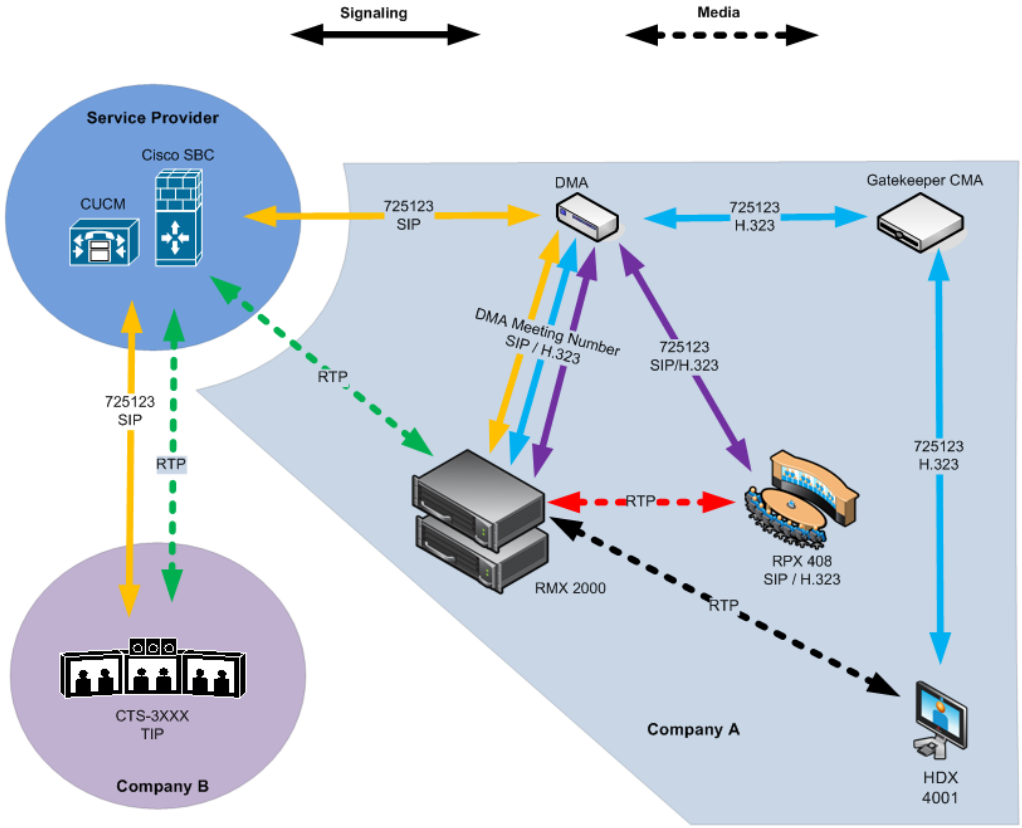


Call Flow

Multipoint call via Service Provider - Model 1

In this example:

- Collaboration Server prefix in the gatekeeper: 72
- *Virtual Meeting Room* in *DMA:* 725123
- *DMA Meeting Number* Generated by *DMA*



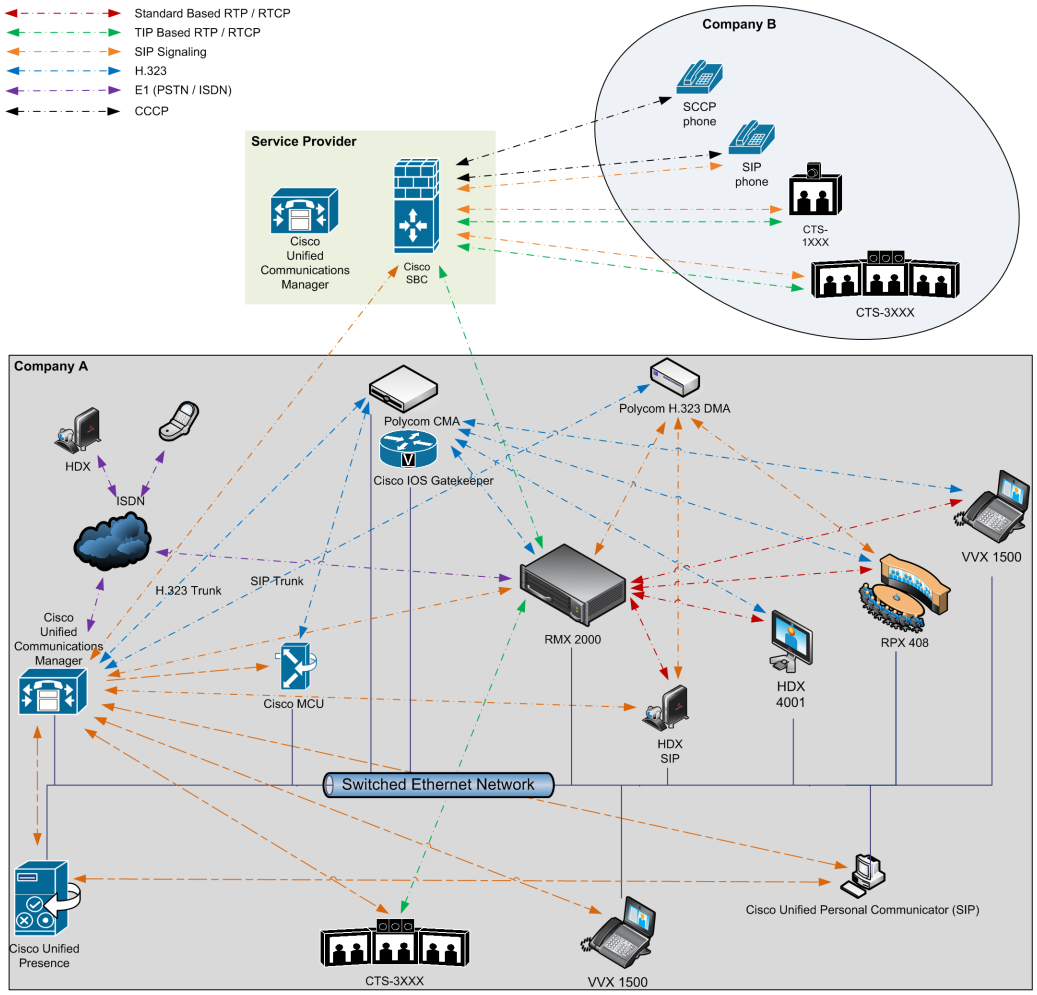
Multipoint call via Service Provider - Model 2

The deployment architecture in *The deployment architecture includes:* shows two companies: *Company A* and *Company B*.

Company A - has the same deployment architecture as shown in *Single Company Model - Polycom and Cisco Infrastructure*.

Company B - has deployed a *Cisco* solution including:

- CTS 1000
- CTS 3000
- *Cisco* telephony endpoints.



The deployment architecture includes:

Company A

For a full description of Company A's deployment, see Single Company Model - Polycom and Cisco Infrastructure.

Differing or additional configuration requirements for each element of this deployment model are listed below:

Company A Solution Architecture Components

Component	Version	Description
CISCO Equipment		
CUCM	8.5	Cisco Unified Communication Manager: CUCM must be configured with a SIP trunk to the Service Provider's SBC.
Polycom Equipment		
Collaboration Server	7.6.x	MCU: Collaboration Server must be configured to send and receive RTP streams to and from the Service Provider's SBC.

Company B

Company B Solution Architecture Components

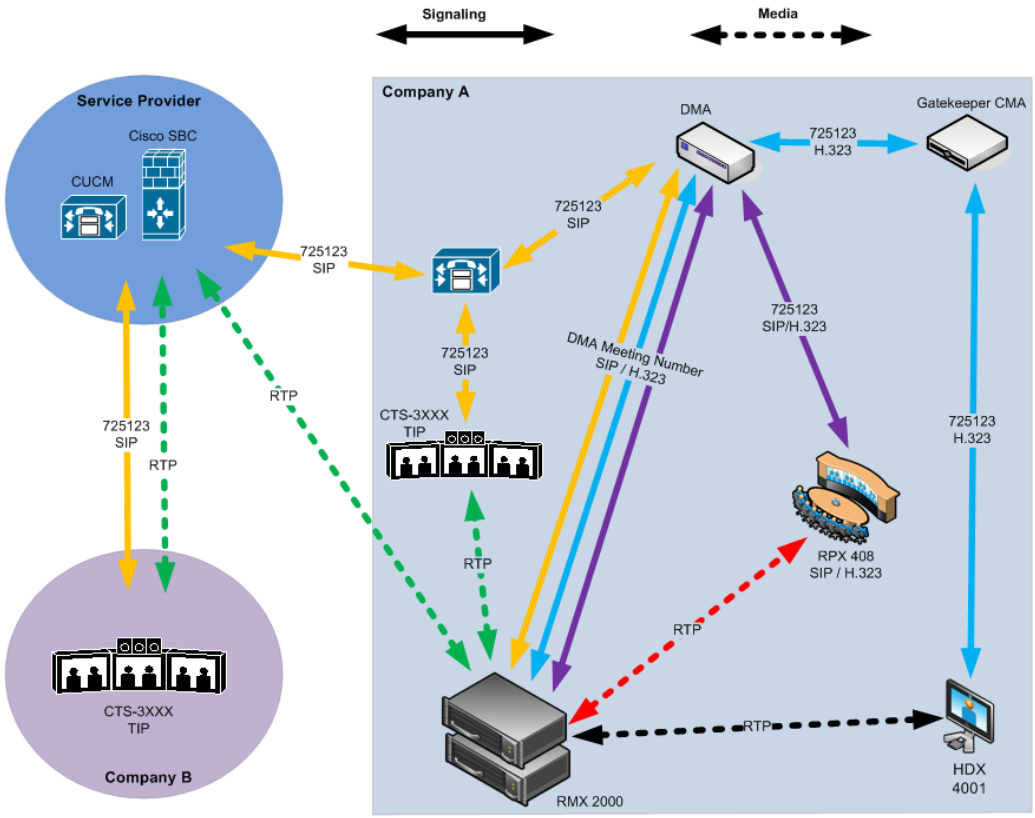
CISCO Equipment	
Endpoints	Endpoints should register with the *Service Provider's CUCM* (or the local CUCM, if present).

Call Flow

Multipoint call via Service Provider - Model 2

In this example:

- Collaboration Server prefix in the gatekeeper: 72
- Virtual Meeting Room in DMA: 725123
- CUCM: According to its Dial Plan forwards calls with prefix 72 to the Collaboration Server



Administration

The various deployment combinations and settings within the various Deployment Architectures affects the administration of the system.

Gatekeepers

Standalone Polycom CMA/DMA System as a Gatekeeper

The Polycom CMA/DMA system can be used as the only gatekeeper for the network. Bandwidth and call admission control of endpoints registered with the CMA system is split between the CMA system and the CUCM.

For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Using a Polycom CMA System as a Gatekeeper.

Standalone Cisco IOS Gatekeeper

The Cisco IOS Gatekeeper can be used as the only gatekeeper for the network if the management capabilities of the Polycom CMA system are not required.

For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Using a Standalone Cisco IOS Gatekeeper.

Neighbored Cisco IOS and Polycom CMA/DMA Gatekeepers

Neighbored gatekeepers make it easier to create a common dial plan and should be considered when integrating an existing Cisco telephony environment with an existing Polycom network. Neighbored Gatekeepers allow number translation while maintaining the existing environments.

For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Neighbored Cisco IOS and Polycom CMA Gatekeepers.

DMA

The Polycom DMA system can be configured as a SIP proxy and registrar for the environment. When used as a SIP peer, the DMA system can host video calls between Cisco endpoints that are registered with the CUCM and Polycom SIP endpoints that are registered with the DMA system.

For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Using a Polycom DMA System as SIP Peer.

CUCM

When Polycom SIP endpoints (voice and video) are registered directly with CUCM you can take advantage of supported telephone functions. CUCM may not support the full range of codecs and features available on the Polycom equipment. CUCM supported codecs and features will be used in such cases.

For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Direct Registration of Polycom Endpoints with the Cisco Unified Communications Manager Participants.

Configuring the Cisco and Polycom Equipment

MLA (Multipoint Layout Application) is required for managing CTS 3XXX layouts whether Polycom TPX, RPX or OTX systems are deployed or not. MLA is a Windows® application that allows conference administrators to configure and control video layouts for multipoint calls involving Polycom Immersive Telepresence (ITP) systems.

Call Detail Records (CDR) are generated on both the CMA Gatekeeper and the CUCM for reporting and billing purposes.

Cisco Equipment

To configure the various Cisco entities the following procedures are required.

CUCM

1. Configure the CUCM to send and receive calls from the H.323 network.
   a. With Neighbored IOS and CMA Gatekeepers

      For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Configuring Cisco Unified Communications Manager for H.323.

   b. With CMA Gatekeeper

      For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Configuring Cisco Unified Communications Manager for H.323.

   c. With IOS Gatekeeper

      For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Configuring Cisco Unified Communications Manager for H.323.

IOS Gatekeeper

- Set up zones and gateway type prefixes to enable dialing to DMA and Collaboration Server systems.

  For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Configuring the Cisco IOS Gatekeeper .

IOS and CMA Gatekeepers (Neighbored)

- Configure the Cisco IOS Gate*keeper for two separate zones.*

  For more information see the Polycom Unified Communications Deployment Guide for Cisco Environments, Configure the Cisco IOS Gatekeeper for use with a CMA System.

Polycom Equipment

The following table lists the Polycom products supported within the various Deployment Architecture.

Only Collaboration Server configurations are described in detail in this document.

Configuration procedures for all other solution components are described in the *Polycom Unified Communications Deployment Guide for Cisco Environments*.

Supported Polycom products

Polycom TIP and SIP	Version(s)
Polycom DMA 7000 system	V4.0
Polycom RealPresence Collaboration Server (RMX) 2000 and RealPresence Collaboration Server (RMX) 4000 systems	V7.6 and higher
Immersive Telepresence Systems: • RPX 200 and 400 systems • OTX 300 system • TPX HD 306 system • ATX HD 300 system	V3.0.3 Requires TIP option key. Requires Polycom Touch Control.
HDX Systems: • 7000 HD Rev C • 8000 HD Rev B • 9006 • 4500	V3.0.3 Requires TIP option key.
The following Polycom peripheral: • Polycom Touch Control	1.3.0
SIP ONLY (no TIP support)	**Version(s)**
Spectralink wireless phones 8020/8030	
Polycom VVX 1500	V4.0
Polycom VVX 1500 C	V3.3.1
KIRK Wireless Server 300/600v3/6000	

The following procedures **1** -**16** are a summary of the configuration procedures.

The detailed procedures **1** - **16** begin with Procedure 1: Set the MIN_TIP_COMPATIBILITY_LINE_RATE System Flag.

Configuring the Collaboration Server

1. Set the **MIN_TIP_COMPATIBILITY_LINE_RATE** System Flag
2. Configuring the Collaboration Server to statically route outbound SIP calls to DMA or CUCM
3. Configuring the Collaboration Server's H.323 Network Service to register with CMA gatekeeper
4. Configuring a TIP enabled Profile on the Collaboration Server
5. Configuring an Ad Hoc Entry Queue on the Collaboration Server if DMA is not used
6. Configuring a Meeting Room on the Collaboration Server
7. Configuring Participant Properties for dial out calls

Configuring DMA

If *DMA* is present in the configuration perform procedures **Configuring DMA to route SIP calls to CUCM** and **Configuring a Virtual Meeting Room (VMR)**, otherwise skip to procedure **Configuring CMA to route H.323 calls to Collaboration Server**.

8 Configuring DMA to route SIP calls to CUCM

9 Configuring a Virtual Meeting Room (VMR)

The procedures for configuring DMA are described in detail in the *Polycom Unified Communications Deployment Guide for Cisco Environments*.

Configuring CMA

10 Configuring CMA to route H.323 calls to Collaboration Server

11 Configuring CMA for use with Cisco IOS Gatekeeper (Neighbored)

12 Configuring CMA to route H.323 calls to CUCM

13 Configuring CMA to route non-H.323 calls to CUCM

The procedures for configuring CMA are described in detail in the *Polycom Unified Communications Deployment Guide for Cisco Environments*.

Configuring Endpoints

14 Configuring H.323 endpoints to register to the CMA or IOS gatekeeper

The procedures for configuring H.323 endpoints are described in detail in the *Polycom Unified Communications Deployment Guide for Cisco Environments*.

15 Configuring SIP endpoints to register to:

a DMA as SIP Proxy

b CUCM as SIP Proxy

The procedures for configuring SIP endpoints are described in detail in the *Polycom Unified Communications Deployment Guide for Cisco Environments*.

16 Configuring TIP endpoints to register to:

a DMA

b CUCM

The procedures for configuring TIP- enabled endpoints are described in detail in the *Polycom Unified Communications Deployment Guide for Cisco Environments*.

Configuring Entry Queues and IVR Services

Conference IVR and Entry Queue/Virtual Entry Queues are supported with AVC TIP protocol in conferences that include both TIP-enabled and non-TIP-enabled endpoints.

A Virtual Entry Queue can be configured to either *IVR Only Service Provider* or *External IVR Control* mode.

TIP-enabled endpoints can be moved from the Entry Queue to the destination conference if the TIP Compatibility Modes settings in the Profile are identical for both conferencing entities (it is recommended to use the same Profile for both entities).

TIP IVR users can access the conference directly or enter the Entry Queue/Virtual Entry Queue and provide a password to access the conference.

The IVR services can be enabled for all TIP Compatibility Modes:

- Video only
- Video and Content
- Prefer TIP

IVR media files, WAV for voice messages and JPG for video slides, are all stored on the Collaboration Server.

Guidelines

- IVR default audio files are enabled for all TIP Compatibility Modes.
- Only Polycom default Welcome slides are available. Custom Welcome slides are not supported.
- TIP-enabled endpoints can send DTMF digits to MCU.
- In an mixed TIP environment there is no support for content in cascaded conferences.

Entry Queue and Virtual Entry Queue Access

TIP endpoints can dial-in to conferences directly using the IVR, Entry Queue/Virtual Entry Queue and IVR Only Service Provider.

For more information on Multipoint see the Multipoint Call Flows.

Configuring the Conference and Entry Queue IVR Services

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 configuration process is the same for TIP and non-TIP enabled Conferences and Entry Queues.

Content

Polycom and *Cisco* endpoints can share *Content* within a *Cisco TelePresence* environment. The content sharing experience depends on whether the endpoints are registered with the *DMA* or *CUCM*.

Endpoint Registration Options - Content Sharing Experience

Multipoint Calls on Collaboration Server	Content Sharing	People + Content
Endpoints Registered to DMA		
HDX/ITP to HDX/ITP	Yes	Yes
HDX/ITP to Cisco CTS	Yes	Yes
Cisco CTS to HDX/ITP	Yes	No
Endpoints Registered to CUCM		
HDX/ITP to HDX/ITP	Yes	No
HDX/ITP to Cisco CTS	Yes	No
Cisco CTS to HDX/ITP	No	No

- H.239
  - A variety of resolutions and frame rates are supported.
    For more information see H.239 / People+Content.
  - Can be used with SIP and H.323 endpoints, desktop (CMAD), room systems (HDX) and ITP (OTX, RPX).
  - Not supported by Lync clients, IBM clients and Cisco CTS endpoints.
  - Cannot be used when HDX endpoints are registered to CUCM.
- TIP
  - The resolution is fixed at XGA at 5fps.
  - Supported on HDX, Polycom ITP and Cisco CTS systems.
- The following content compatibility options are available:
  - Tip not enabled – CTS cannot join the conference, all other endpoints can share H.239 content.
  - TIP video compatibility – CTS receives people video, all other endpoints can share H.239 content.
  - TIP video and content compatibility – CTS and HDX can share TIP content, all other endpoints receive only the people video.

    For more information see Procedure 4: Configuring a TIP Enabled Profile on the Collaboration Server.

Procedure 1: Set the MIN_TIP_COMPATIBILITY_LINE_RATE System Flag

The **MIN_TIP_COMPATIBILITY_LINE_RATE** System Flag determines the minimum line rate at which an Entry Queue or Meeting Room can be TIP enabled.

CTS version 7 requires a minimum line rate of 1024 kbps and will reject calls at lower line rates, therefore the SysCollaboration Servertem Flag value must be 1024 or higher.

HD Video Resolutions for TIP calls are determined according to the following table:

TIP HD Video Resolution by Line Rate

Line Rate	Video Resolution
Line Rate >=3Mbps	HD1080p30
3Mbps > Line Rate >= 936kbps	HD720p30
Line Rate < 936kbps	Call is dropped.

For more information see Modifying System Flags.

Procedure 2: Configuring Collaboration Server to statically route outbound SIP calls to DMA or CUCM

1. In the IP Network Services Properties dialog box, click the **SIP Servers** tab.
2. In the SIP Server field, select **Specify**.
3. In the SIP Server Type field, select **Generic**.
4. Set Refresh Registration every **3600** seconds.

5 If not selected by default, change the Transport Type to **TCP**.

6 In the SIP Servers table:

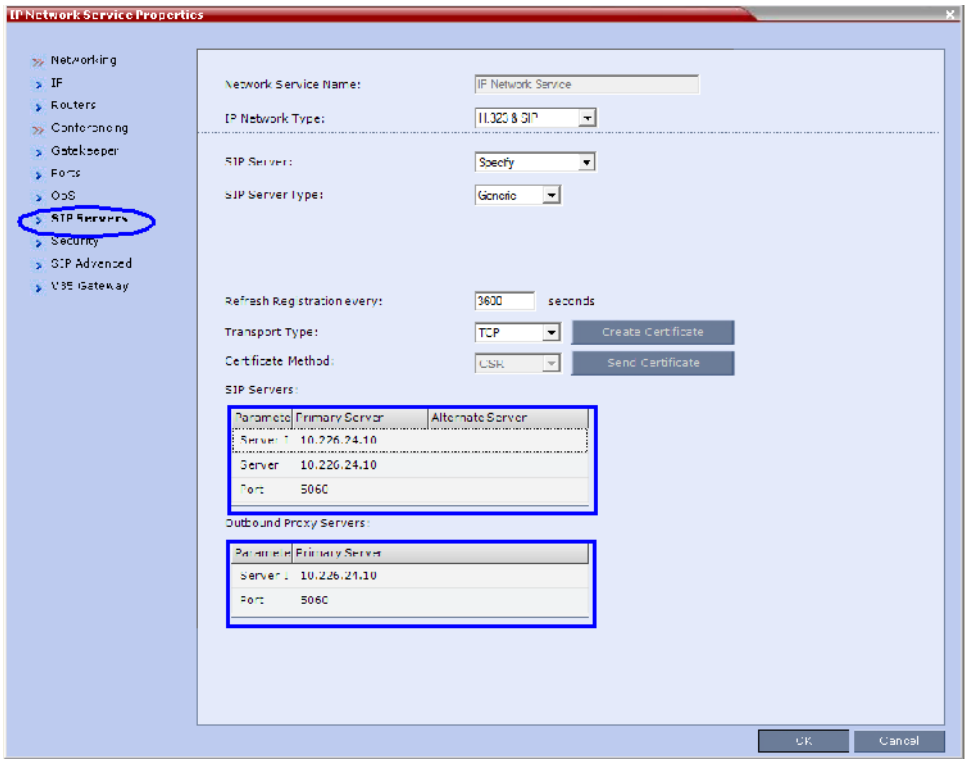
a Enter the IP address of the DMA or CUCM in both the Server IP Address or Name and Server Domain Name fields.

b The Port field must be set to it's default value: **5060**. DMA and CUCM use this port number by default.

7 In the Outbound Proxy Servers table:

a Enter the IP address in the Server IP Address or Name field. (The same value as entered in Step 6a.)

b The Port field must be set to it's default value: **5060**.
(By default, the Outbound Proxy Server is the same as the SIP Server.)

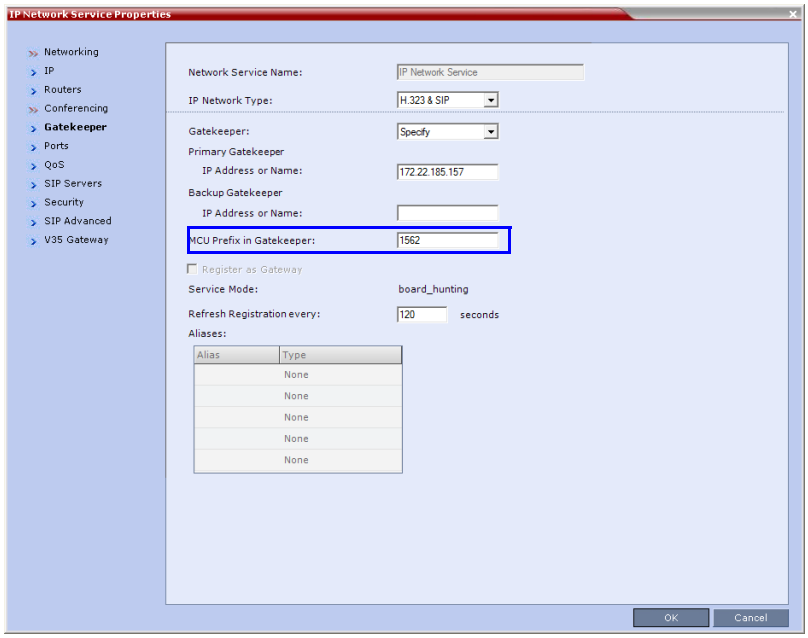


When configuring Collaboration Server to statically route SIP calls to DMA or CUCM, it is important to also configure the Collaboration Server's H.323 Network Service to register with CMA gatekeeper. For more information see Procedure 3: Configuring the Collaboration Server's H.323 Network Service to register with CMA gatekeeper.

Procedure 3: Configuring the Collaboration Server's H.323 Network Service to register with CMA gatekeeper

1 In the IP Network Services Properties dialog box, click the **Gatekeeper** tab.

2 In the MCU Prefix in Gatekeeper field, enter the prefix that the Collaboration Server uses to register with the gatekeeper.

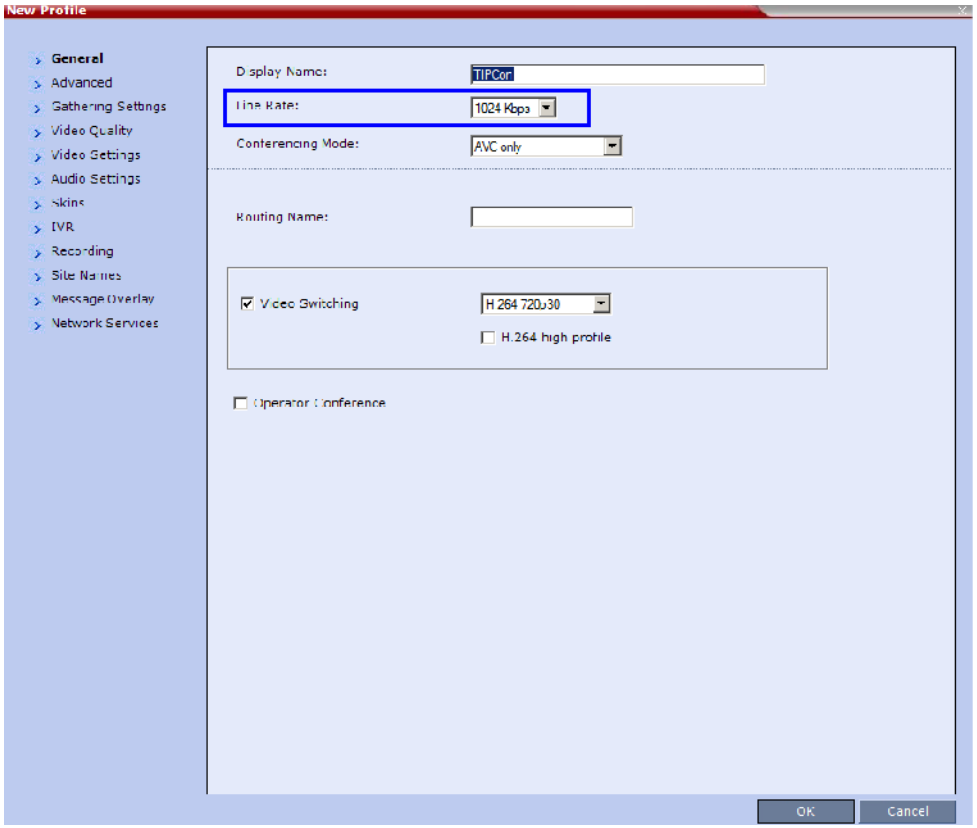


Procedure 4: Configuring a TIP Enabled Profile on the Collaboration Server

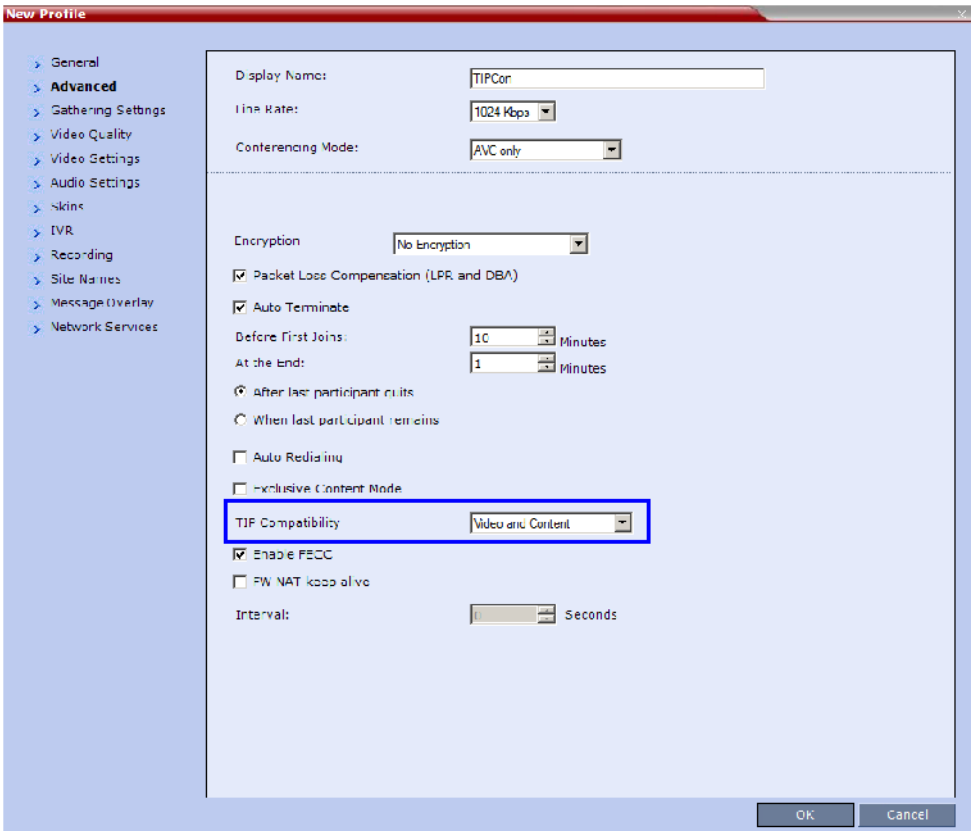
TIP enabled profiles must be used for the Entry Queues and Meeting Rooms defined on the Collaboration Server. (Different Profiles can be assigned to Entry Queues and Meeting Rooms, however they must be TIP enabled.) When TIP is enabled in the Profile, Gathering Settings and Message Overlay options are disabled.

1 Create a New Profile for the Meeting Room. For more information see Defining a CP Conference Profile.

2 In the New Profile - General tab, set the Line Rate to a value of at least that specified for the **MIN_TIP_COMPATIBILITY_LINE_RATE** System Flag in Procedure 1.

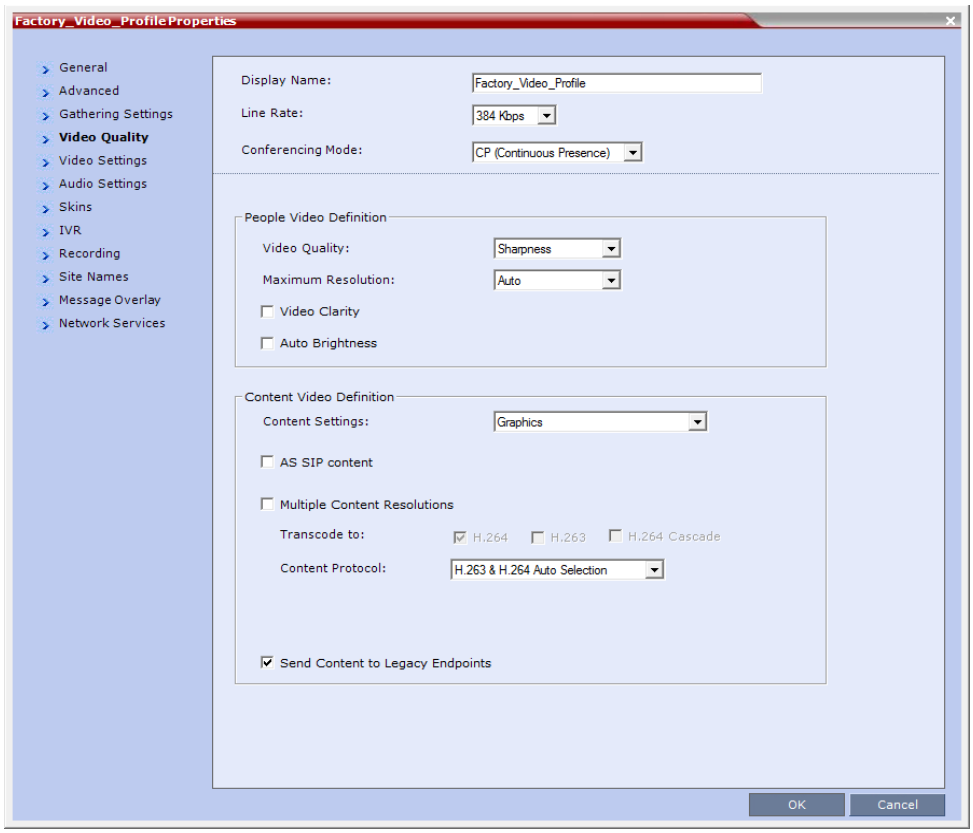


3 Click the **Advanced** tab.



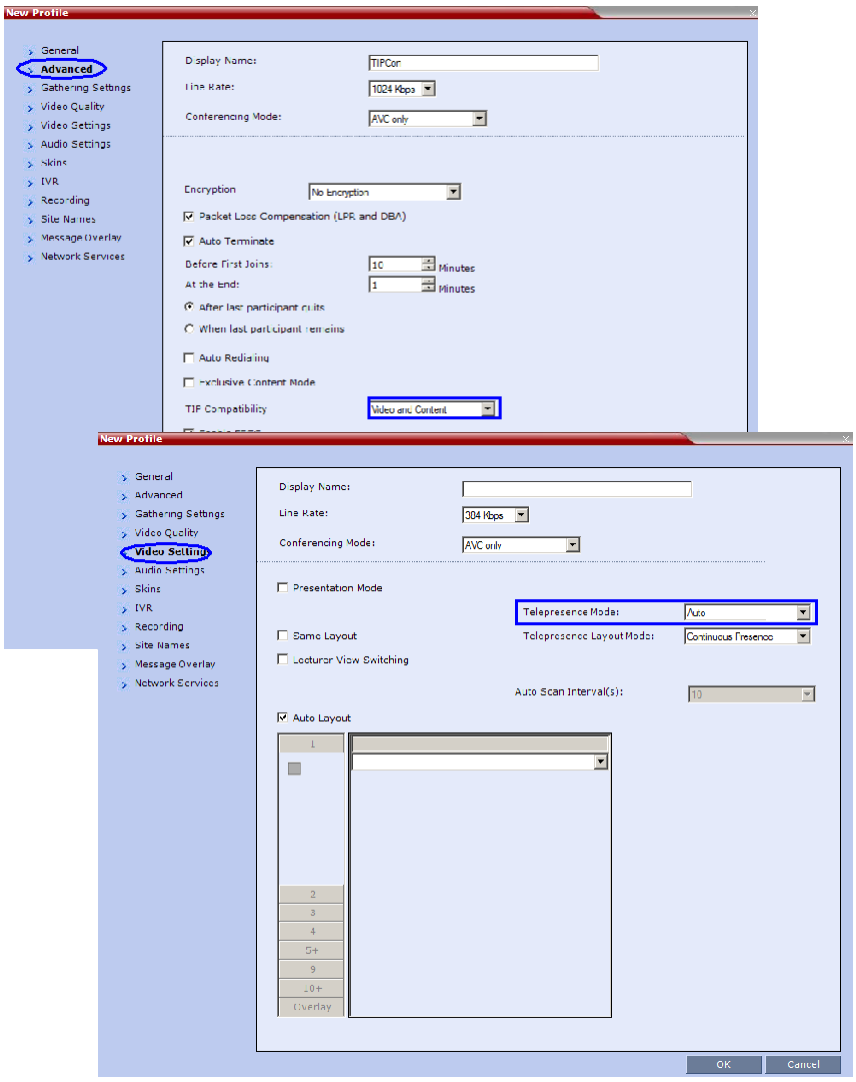
4 Select the TIP Compatibility mode according to the Content Sharing Behaviortables that are listed at the end of this procedure: **Video and Content** is recommended.

5 Click the **Video Quality** tab.



Content Settings is disabled if TIP Compatibility is set to **Video and Content** in the Advanced tab.

6 Click the **Video Settings** tab.



7 Set the Telepresence Mode to **Auto**.

8 Assign the New Profile to the Meeting Room. For more information see Creating a New Meeting Room.

Content Sharing Behavior

The following tables list the system's Content sharing behavior for the various combinations of TIP Compatibility mode settings and the following endpoints:

Polycom Immersive Telepresence Systems (ITP) Version 3.0.3:

- RPX 200
- RPX 400
- OTX 300
- TPX HD 306
- ATX HD 300

Polycom video conferencing endpoints (HDX) Version 3.0.3:

- 7000 HD Rev C
- 8000 HD Rev B
- 9006
- 4500

Cisco TelePresence® System (CTS) Versions 1.7 / 1.8:

- CTS 1000
- CTS 3000

TIP Compatibility - None

None		Content Receiver	
		HDX / ITP	CTS
Content Sender	HDX / ITP	Media: H.264 Flow Control: H.323 via H.239 SIP via BFCP	Not Connected
	CTS	Not Connected	Not Connected

TIP Compatibility - Video Only

Video Only		Content Receiver	
		HDX / ITP	CTS
Content Sender	HDX / ITP	Media: H.264 Flow Control: H.323 via H.239 SIP via BFCP	None
	CTS	None	None

TIP Compatibility - Video & Content

Video & Content		Content Receiver	
		HDX / ITP	CTS
Content Sender	HDX* / ITP	Media: H.264 Flow Control: H.323 via H.239 SIP via BFCP TIP via Auto Collaboration	
	CTS		

* If HDX supports TIP Content.

Selecting *TIP Compatibility* as **Video and Content** disables *Content Settings* in the *Video Quality* tab.

TIP Compatibility - Prefer TIP

Prefer TIP		Content Receiver	
		HDX / ITP	CTS
Content Sender	HDX / ITP	Media: H.264 Flow Control: H.323 via H.239 SIP via BFCP TIP via Auto Collaboration	
	CTS*		

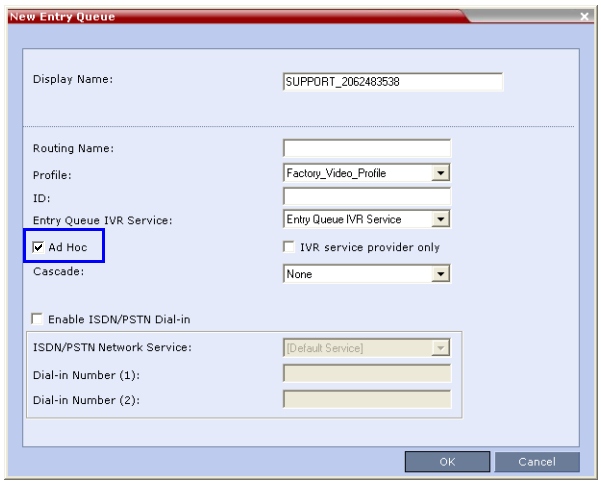
* CTS Version 1.9.1 and higher support H.264 Content.

In **Prefer TIP** mode, it is pre-requisite that the *CTS* and *CUCM* versions support *H.264* base profile content without restrictions and that the *CTS* version be 1.9.1 or higher and that *CUCM* version be version 9.0 or higher.

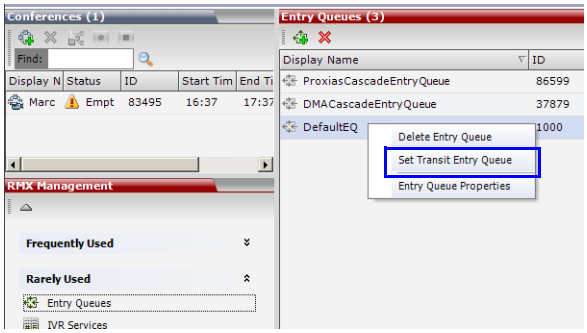
Procedure 5: Configuring an Ad Hoc Entry Queue on the Collaboration Server if DMA is not used

1 Create or select the **Entry Queue** as described in Entry Queues.

2 In the New Entry Queue or Entry Queue Properties dialog box, ensure that **Ad Hoc** is selected.



3 Ensure that the Entry Queue is designated as the **Transit Entry Queue** as described in Setting a Transit Entry Queue.



Procedure 6: Configuring a Meeting Room on the Collaboration Server

The Profile for the Meeting Room must be TIP enabled as described in Procedure 4.

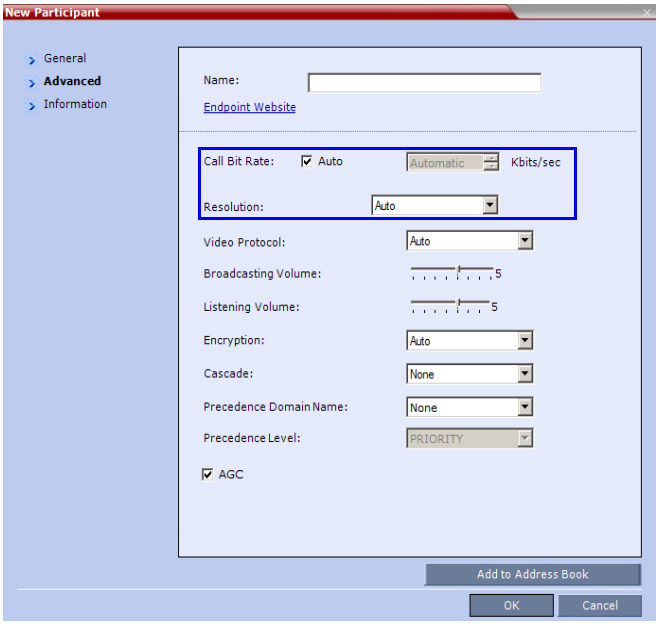
For more information see Creating a New Meeting Room.

Procedure 7: Configuring Participant Properties for dial out calls

Participant Properties must be configured to ensure that defined participants inherit their TIP settings from the Profile assigned to the Meeting Room.

a Define the New Participant's General settings. For more information see Adding a Participant to the Address Book.

b Click the **Advanced** tab.



c Ensure that:

- Call Bit Rate is set to Automatic or at least equal to or greater than the value specified by the **MIN_TIP_COMPATIBILITY_LINE_RATE** System Flag.
- Resolution is set to **Auto** or at least **HD 720**.
- Video Protocol is set to **Auto** or at least **H.264**.

Collaboration with Microsoft and Cisco

This solution enables Polycom, Microsoft and Cisco users, each within their own environment, to participate in the same conference running on an Collaboration Server.

The Collaboration Server natively inter-operates with Microsoft Lync and Cisco TelePresence Systems, ensuring optimum quality multi-screen, multipoint calls between:

- Polycom Immersive Telepresence Systems (ITP) Version 3.1.1:
  - RPX 200
  - RPX 400
  - OTX 300
- Polycom video conferencing endpoints
  - Standalone HDX
  - Polycom Group Series 300/500
- Microsoft
  - MS Lync (using MS-ICE)
  - RTV 720p
- Cisco TelePresence® System (CTS) Versions 1.10

  - CTS 1300
  - CTS 3010

The deployment architecture in Single company with Polycom and Cisco Infrastructure - Polycom endpoints using SIP shows a company that has a mixture of Polycom, Cisco and Microsoft endpoints, room systems and telephony equipment that needs to enable multipoint calls between all its video and audio endpoints using the Collaboration Server as the conference bridge.

This solution enables Polycom, Microsoft and Cisco users, each within their own environment, to participate in the same conference running on an MCU.

In the solution described in Single company with Polycom and Cisco Infrastructure - Polycom endpoints using SIP:

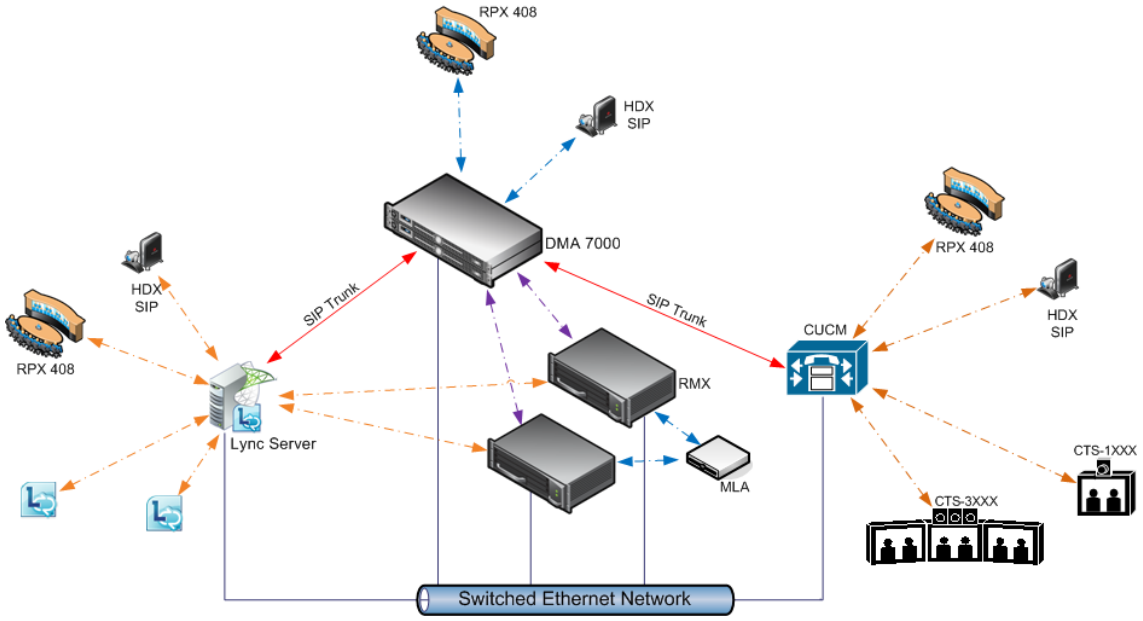
- DMA is required as all calls are dial-in to Virtual Meeting Rooms (VMR) provisioned on the DMA.
- Microsoft and Cisco clients dial the same VMR number to connect to the conference.
- Dial- out calls directly from the RMX are not supported.
- Lync Clients cannot share content with CTS
- SIP trunks are required to the DMA from:
  - MS Lync as a Static Route.
  - CUCM

Deployment Architecture

- DMA is required as all calls are dial-in to Virtual Meeting Rooms (VMR) provisioned on the DMA.
- Microsoft and Cisco clients dial the same VMR number to connect to the conference.
- Dial- out calls are not supported
- Lync Clients can not share content with CTS
- SIP trunks are required to the DMA from:
  - MS Lync as a Static Route.
  - CUCM

For more information, see Cisco TIP Support.

POCN Polycom, Microsoft and Cisco Infrastructure. Solution Architecture components.



.POCN Polycom, Microsoft and Cisco Infrastructure. Solution Architecture components

Component	Version
Polycom	
HDX	3.0.5
RSS	8.0
DMA	5.0
CMA	6.0.1
CMAD	5.2.3
ITP (OTX, RPX, ATX, TPX)	
Conferencing for Outlook (PCO)	1.0.7
Touch Control	1.3
Microsoft	
Microsoft Lync 2010 Server	4.0.7577.223(CU10)
Microsoft Lync 2013 Server	5.0.8308.556 (CU3)
Microsoft Lync 2010 client	4.0.7577.4051 CU4
Exchange 2007 R2 SP3	8.3.213.1

Component	Version
Exchange 2010 SP2	14.2.247.5
Outlook 2007	12.0.6557.5001 SP2
Outlook 2010	14.0.6112.5000
Cisco	
CUCM	8.5, 8.6.2
Cisco Unified Personal communicator	8.5(2),8.5(5)
Cisco Unified IP Phones 7960, 7961, 7962, 7965, 7975	CUCM 8.5 / CUCM 8.6(2) Compatible
CTS	1.7.4, 1.8.1
C90, C20	TC5.0

The following are not supported:

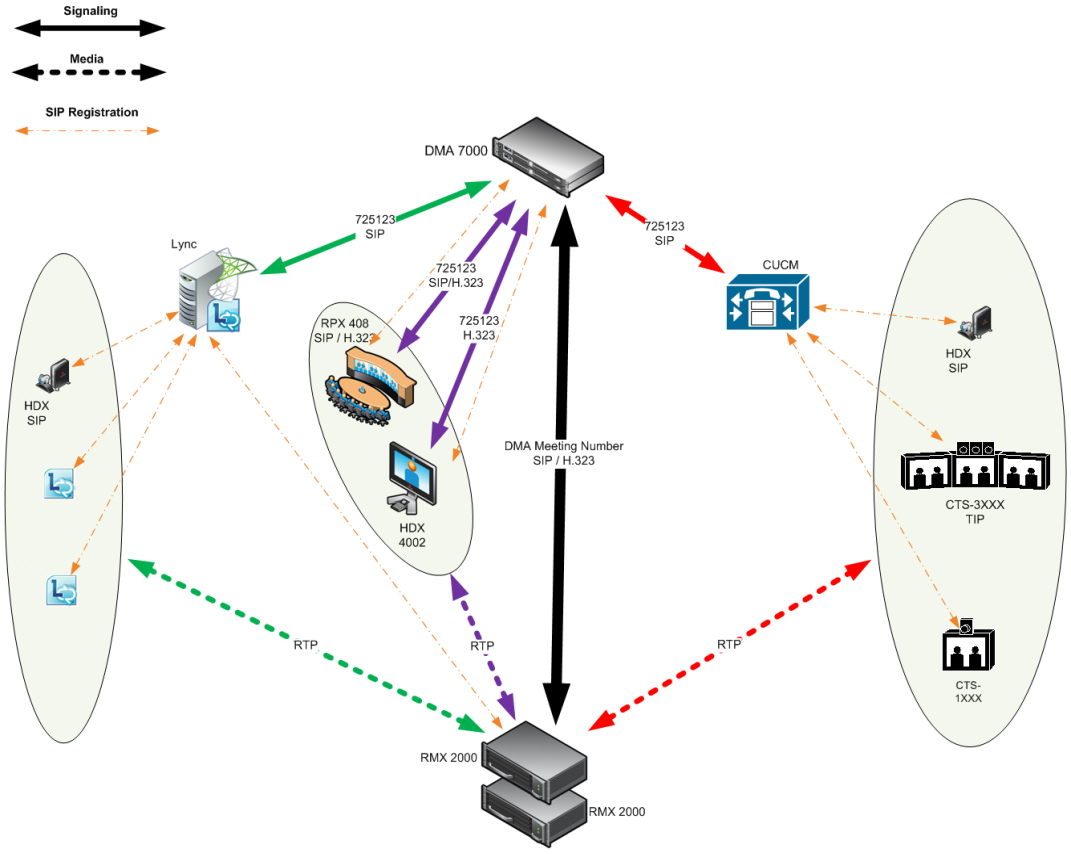
- In the Lync environment:
  - Sending or receiving Content.
  - Dial-out to Lync clients.
  - Presence of VMRs
- In the Cisco environment:
  - TLS and SRTP
  - OBTP

Call Flow

Multipoint Calls using DMA

In this example:

- Endpoint registration: To either DMA, Lync or CUCM.
- DMA dial in Prefix: 72
- Virtual Meeting Room in DMA: 725123
- DMA Meeting Number: Generated by DMA



Administration

The various deployment combinations and settings within the Deployment Architecture affects the administration of the system.

DMA

The DMA system can be configured as a SIP proxy and registrar for the environment as well as a Gatekeeper for dial in H.323 calls. When configured as a Gateway for dial in H.323 calls, it enables H.323 endpoints to connect to the same VMR as SIP clients.

When used as a SIP peer, the DMA system can host video calls between Cisco endpoints that are registered with the CUCM, Lync Clients that are registered with the Lync Server and Polycom endpoints that are registered with the DMA system.

For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments, Using a Polycom DMA System as SIP Peer*.

Microsoft Lync Server

Microsoft Lync Server manages Presence for each registered Polycom endpoint and enables video calls between Lync Clients and Polycom endpoints allowing Lync contacts to be called without needing to know their addresses.

RTV video, MS-ICE and Lync-hosted conferencing are supported when Polycom endpoints are registered to Lync Server. Polycom endpoints use H.264, while Lync Clients use the RTV protocol.

CUCM

When Polycom SIP endpoints (voice and video) are registered directly with CUCM you can take advantage of supported telephone functions. CUCM may not support the full range of codecs and features available on the Polycom equipment. CUCM supported codecs and features will be used in such cases.

For more information see the Polycom Unified Communications Deployment Guide for Cisco Environments, *Direct Registration of Polycom Endpoints with the Cisco Unified Communications Manager Participants*.

Solution Interoperability Table

The following table lists components and versions of the Collaboration Server, Microsoft and Cisco Telepresence Systems (CTS) Integration Solution Architecture.

Solution Architecture Components

Component	Version	Description
CISCO Equipment		
CUCM	9.0.1	Cisco Unified Communication Manager: • CUCM must be configured to route calls to ASR/SBC. CUCM must be configured with a SIP trunk to the Service Provider's SBC. • All endpoints must register once with the CUCM • SIP trunks from *CUCM* to *Polycom* system components (eg. *DMA*) should be configured with *Music on Hold* disabled.

Component	Version	Description
ASR (Cisco SBC)	100x	The Cisco Aggregation Services Routers (ASR) Series includes Cisco IOS XE Software Internetwork Operating System - Gatekeeper. It controls and manages real-time multimedia traffic flows between IP/SIP network borders, handling signaling, data, voice, and video traffic.
Polycom Equipment		
DMA	6.0.0_ATT_Build_25	Polycom Distributed Media Application • *DMA* is an optional component but is essential if *Content* sharing is to be enabled. • All *SIP* endpoints register to *DMA* as a *SIP Proxy*. • DMA should be configured to route SIP calls (with CTS destination) to CUCM. • *DMA* can be configured with a *VMR* (*Virtual Meeting Room*). Incoming calls are then routed to the Collaboration Server.
Collaboration Server	8.1.1	MCU: • Functions as the network bridge for multipoint calls between *H.323*, *SIP* and *TIP* endpoints. • The Collaboration Server can be interfaced to *CUCM* using a *SIP* trunk, enabling *CTS* to join multipoint calls on Collaboration Server. Signaling goes through the *CUCM* while the media in *TIP* format goes directly between the *CTS* and Collaboration Server. • The Collaboration Server must be configured to route outbound SIP calls to DMA. • Collaboration Server must be configured to send and receive RTP streams to and from the Service Provider's SBC.
MLA Server	3.0.5	Multipoint Layout Application Required for managing multi-screen endpoint layouts for *Cisco CTS 3XXX, Polycom TPX*, *RPX* or *OTX* systems.
HDX and ITP Endpoints	3.1.1.1	Telepresence, desktop and room systems. • *Polycom SIP* endpoints must register to *DMA* as *SIP Proxy*.
Microsoft		
Lync 2010	4.0.7577.183 CU4	
Lync 2010 client	4.0.7577.4051 CU4	
Exchange 2007 R2 SP3	8.3.213.1	
Exchange 2010 SP2	14.2.247.5	

Component	Version	Description
Outlook 2007	12.0.6557.5001 SP2	
Outlook 2010	14.0.6112.5000	

TIP Layout Support & Resource Usage

Cisco Telepresence endpoints using TIP protocol support only one (CTS 1000) or three (CTS 3000) display screens. Therefore, Polycom Telepresence endpoints will adjust their display to use one or three screens as follows:

- **OTX system** - works with three screens, therefore no adjustment is required and it should be set to work in *room switch* Telepresence Layout Mode (while avoiding zooming in/out)
- **RPX 2xx** - This endpoint works with two screens, therefore it will adjust to use only **one** screen.
- **RPX 4xx** - This endpoint works with four screens, therefore it will adjust to use only **three** screens.
- **Standalone HDX** - behaves as the CTS 1000 and uses **only** one screen.
- **Group system 300/500** - behaves as the CTS 1000 and uses **only** one screen.

The Polycom MLA Server manages the conference template layouts for Telepresence systems.

The number of screens used by each TIP-enabled endpoint is determined during the capabilities exchange phase of the dial-in connection. It affects the usage and allocation of resources used with TIP-enabled endpoints.

Supported TIP Resolutions and Resource Allocation

Supported Resolutions

In a Telepresence TIP-enabled environment, only two video resolutions are available: 720p30 & 1080p30.

Supported resolution per conference line rate

Conference Line Rate	Selected Resolution
3Mb or higher	1080p 30 fps
963kbps to 3Mb	720p 30 fps
Up to 936kbps	Call is disconnected.

Resource Allocation

The MCU media processor (ART) supports up to three TIP-enabled screens as follows:

- One TIP-enabled endpoint with three screens
- Up to three TIP-enabled endpoint with one screen

TIP-enabled endpoint with three screens must be handled by the same media processor. This endpoint may fail to connect if there is no one fully free media (ART) processor available.

The MCU will always try to fill up one media processor with up to three TIP-enabled endpoint with one screen, to save free media processors for TIP-enabled endpoint with three screens.

When monitoring an ongoing Telepresence conference with TIP-enabled endpoints (Cisco and Polycom), virtual participants are used to indicate the additional screens in the in the Web Client. For example, if the endpoint has three screens, the system will display three participants, one for each screen.

An additional virtual Audio Only participant is used for the audio only telephone connected to the TIP endpoint.

Configuring the Microsoft, Cisco and Polycom Components

1 Configure a SIP Trunk connection between the Polycom DMA system and the Cisco Unified Communications Manager (CUCM).

For more information see the *Polycom Unified Communications Deployment Guide for Cisco Environments*, Using a Polycom DMA System as SIP Peer.

2 Register the Collaboration Server to the Lync Server

a Install a Security Certificate on the Collaboration Server.

The Certificate is obtained from the System Administrator and saved on the Workstation.

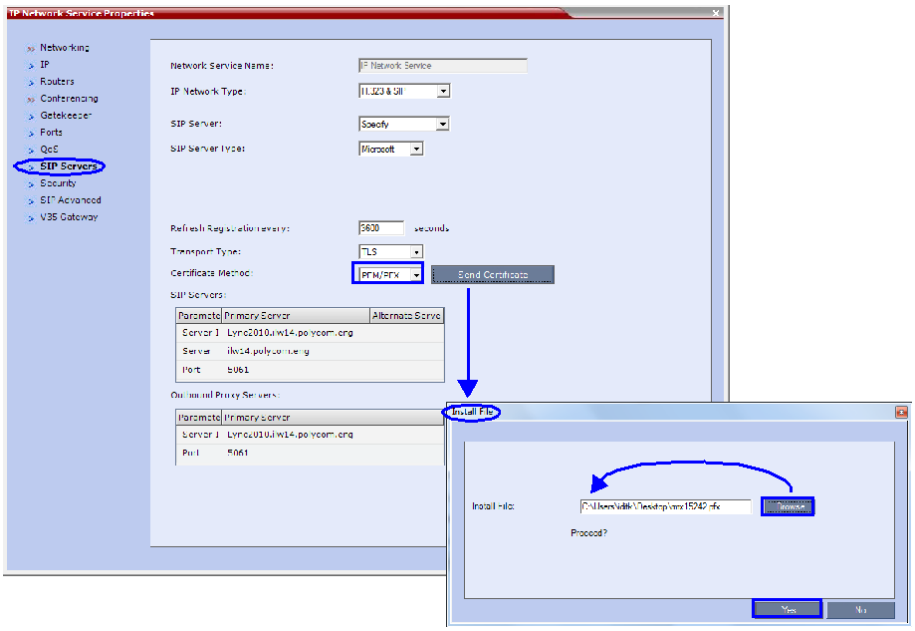
b In the SIP Servers tab of the IP Network Services Properties dialog box:

1 In the Certificate Method drop-down menu, select **PEM/PFX**.

2 Click the **Send Certificate** button.

The Install File dialog box is displayed.

iii Browse to the saved Certificate on the Workstation and click the **Yes** button to install the certificate.



For more information see:

- Appendix H - Integration Into Microsoft Environments.
- *Polycom Unified Communications Deployment Guide for Microsoft Environments*, Configuring Your Collaboration Server System for use with the Lync Server.

3 Register the Collaboration Server with the *Lync Server*.

a In the IP Network Services Properties dialog box, click the **SIP Servers** tab.

b In the SIP Server field, select **Specify**.

c In the SIP Server Type field, select **Microsoft**.

d Set Refresh Registration every **3600** seconds.

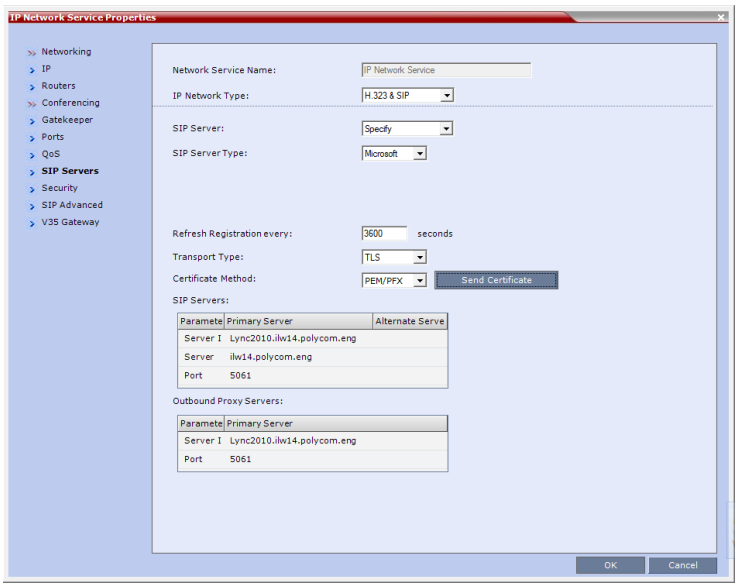
e If not selected by default, change the Transport Type to **TLS**.

f In the SIP Servers table, enter the IP address of the Lync Server in both the Server IP Address or Name and Server Domain Name fields.

g In the SIP Servers table, the Port field must be set to **5061**.

h In the Outbound Proxy Servers table, enter the IP address in the Server IP Address or Name field. (The same value as entered in **Step f**.)

i In the Outbound Proxy Servers table, the Port field must be set to 5061. (The same value as entered in Step g.)



For more information see the *Polycom Unified Communications Deployment Guide for Microsoft Environments*.

4 Set the **ITP_CERTIFICATION** System Flag to **YES**.
When set to **NO** (default), this flag disables the Telepresence features in the Conference Profile.

5 Set the **MIN_TIP_COMPATIBILITY_LINE_RATE** System Flag.

The **MIN_TIP_COMPATIBILITY_LINE_RATE** System Flag determines the minimum line rate at which a Profile can be TIP enabled.

CTS version 1.7 requires a minimum line rate of 1024 kbps and will reject calls at lower line rates, therefore the System Flag value must be **1024** or higher.

For more information see Modifying System Flags.

6 If required, manually add and set the **FORCE_720P_2048_FOR_PLCM_TIP** System Flag using one of the following values:

FORCE_720P_2048_FOR_PLCM_TIP (Default) - Forces HD 720p video resolution and a line rate of 2048kbps for all Polycom TIP-enabled endpoints that connect to the TIP-enabled Telepresence conference. This setting is the recommended setting.

FORCE_2048_FOR_PLCM_TIP - Forces a line rate of 2048kbps for all Polycom TIP-enabled endpoints connecting to the TIP-enabled Telepresence conference.

NO_FORCE - No forcing is applied and Polycom TIP-enabled endpoints can connect to the TIP-enabled Telepresence conference at any line rate or resolution.

7 Reset the Collaboration Server.

8 For more information see .

9 Register the DMA to the Lync server

For more information see the *Polycom Unified Communications Deployment Guide for Microsoft Environments*, Configure a DMA System SIP Peer for the Lync Server.

10 Register the ITP endpoints to the Lync server

For more information see the Polycom Unified Communications Deployment Guide for Microsoft Environments, *Deployment Process for Polycom Immersive Telepresence Systems*.

11 Register Lync Clients to the Lync server.

For more information see the relevant Lync documentation.

12 Register DMA to the CUCUM server

For more information see the Polycom Unified Communications Deployment Guide for Cisco Environments, *Using a Polycom DMA System in a Cisco Environment*.

13 Register CTS1000 and CTS3000 endpoints to the CUCUM server

For more information see the relevant Cisco documentation.

14 Register ITP endpoints to the CUCM server.

For more information see the Polycom Unified Communications Deployment Guide for Cisco Environments, *Direct Registration of Polycom Telepresence Systems with the Cisco Unified Communications Manager*.

15 Register HDX endpoints to the DMA as Gatekeeper

For more information see the *Polycom® DMA™ 7000 System Operations Guide*.

16 Open MLA to configure ITP Layouts

MLA (Multipoint Layout Application) is required for managing CTS 3XXX layouts whether Polycom TPX, RPX or OTX systems are deployed or not. MLA is a Windows® application that allows conference administrators to configure and control video layouts for multipoint calls involving Polycom Immersive Telepresence (ITP) systems.

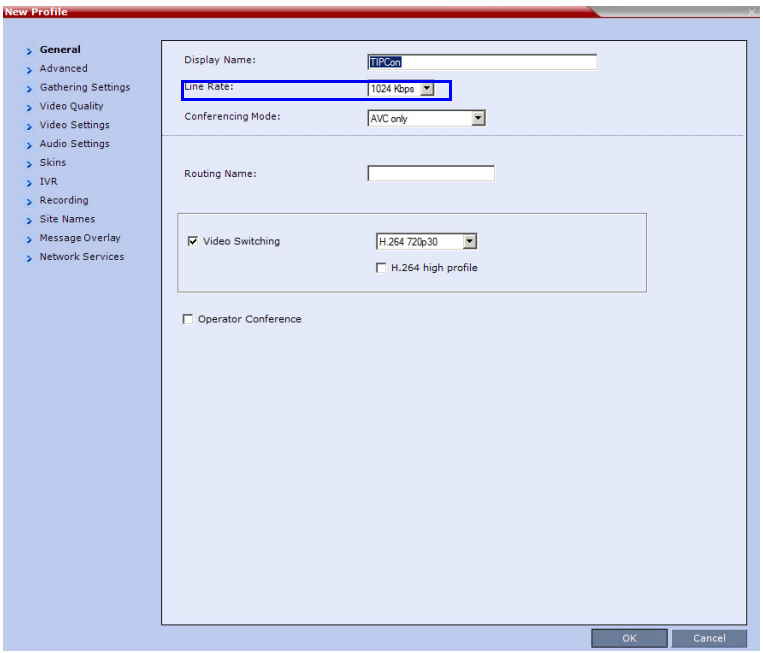
For more information see the *Polycom Multipoint Layout Application (MLA) User's Guide for Use with Polycom Telepresence Solutions*.

17 Configure a TIP Enabled Profile on the Collaboration Server.

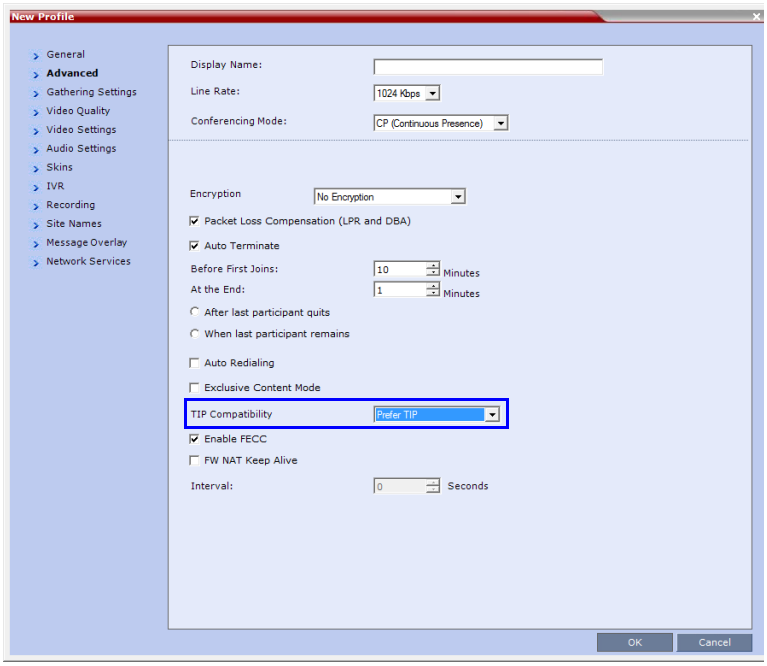
a Create a New Profile for the Meeting Room.

For more information see Defining New Profiles.

b In the New Profile - General tab, set the Line Rate to a value of at least that specified for the **MIN_TIP_COMPATIBILITY_LINE_RATE** System Flag in Procedure 1: Set the MIN_TIP_COMPATIBILITY_LINE_RATE System Flag.



c Click the *Advanced* tab.



d Select the **TIP Compatibility** mode according to the Content Sharing Behavior tables that are listed below.

Prefer TIP is recommended if *Polycom* endpoints are to connect using *TIP,* otherwise select **Video and Content**.

When *Prefer TIP* is selected *Gathering Settings, Skins, Message Overlay, Site Names* and *Network Indication*(*s*) are disabled.

Content Sharing Behavior

The following tables list the system's Content sharing behavior for the various combinations of TIP Compatibility mode settings and the following endpoints:

Polycom Immersive Telepresence Systems (ITP) Version 3.0.3:

- RPX 200
- RPX 400
- OTX 300
- TPX HD 306
- ATX HD 300

Polycom video conferencing endpoints (HDX) Version 3.0.3:

- 7000 HD Rev C
- 8000 HD Rev B
- 9006
- 4500

Cisco TelePresence® System (CTS) Versions 1.7 / 1.8:

- CTS 1000
- CTS 3000

TIP Compatibility - None

None		Content Receiver	
		HDX / ITP	CTS
Content Sender	HDX / ITP	Media: H.264 Flow Control: H.323 via H.239 SIP via BFCP	Not Connected
	CTS	Not Connected	Not Connected

TIP Compatibility - Video Only

Video Only		Content Receiver	
		HDX / ITP	CTS
Content Sender	HDX / ITP	Media: H.264 Flow Control: H.323 via H.239 SIP via BFCP	None
	CTS	None	None

TIP Compatibility - Video & Content

Video & Content		Content Receiver	
		HDX / ITP	CTS
Content Sender	HDX* / ITP	Media: H.264 Flow Control: H.323 via H.239 SIP via BFCP TIP via Auto Collaboration	
	CTS		

* If HDX supports TIP Content.

Selecting *TIP Compatibility* as **Video and Content** disables *Content Settings* in the *Video Quality* tab.

TIP Compatibility - Prefer TIP

Prefer TIP		Content Receiver	
		HDX / ITP	CTS
Content Sender	HDX / ITP	Media: H.264 Flow Control: H.323 via H.239 SIP via BFCP TIP via Auto Collaboration	
	CTS*		

* CTS Version 1.9.1 and higher support H.264 Content.

In **Prefer TIP** mode, it is pre-requisite that the *CTS* and *CUCM* versions support *H.264* base profile content without restrictions and that the *CTS* version be 1.9.1 or higher and that *CUCM* version be version 9.0 or higher.

Encryption

Encryption between the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition and a CISCO environment is supported. Media is encrypted using SRTP, while control is encrypted using SRTCP. TIP is encrypted using SRTCP. SIP is be encrypted using TLS. When upgrading, the Collaboration Server automatically creates a self-signed certificate to support encrypted communications with CISCO endpoints.

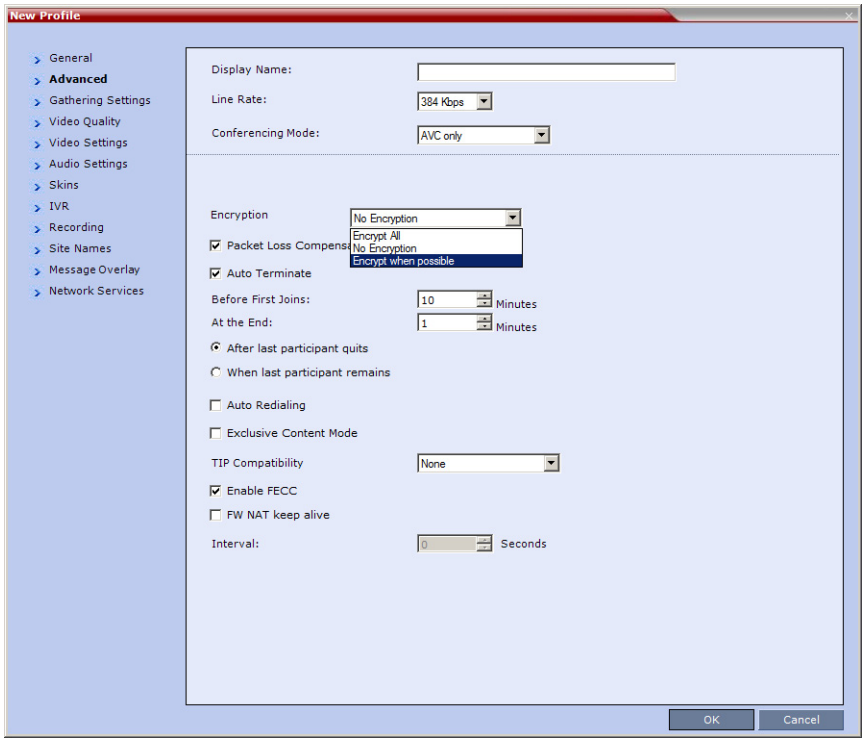
For media encryption. the Collaboration Server will first attempt to exchange keys using DTLS. If the Collaboration Server fails to exchange keys using DTLS, SIP TLS encrypted with SDES is used to exchange media encryption keys.

Guidelines

- This feature is not supported in *Ultra Secure Mode*.
- Voice activity metrics and RTP are not encrypted.
- In the event that DTLS negotiation fails, SIP will be encrypted using TLS if enabled in the IP Management Network properties, SIP Servers tab. DTLS negotiation does not require SIP TLS.
  - In a mixed CISCO and Microsoft Lync environment, in order to assure encrypted communications with both CISCO endpoints and Microsoft Lync in the event of DTLS negotiation failure, the certificate defined in the IP Management Network Services properties dialog box, SIP Servers tab, must have been issued by the same certificate authority that issued the certificates used by both the Microsoft Lync server and the CUCM server.
- The flag, **SIP_ENCRYPTION_KEY_EXCHANGE_MODE**, is used to control this feature. The possible values are:
  - AUTO (default): Normal encryption flow
  - DTLS: Only use DTLS for encryption
  - SDES: Only use SDES (SRTP) for encryption
  - NONE: Encryption is disabled
- The feature was tested using the following CISCO components:
  - Cisco CUCM Version 9.0
  - Cisco TPC Version 2.3
  - Cisco endpoints running Version 1.9.1
    - C20, C40, C60, and C90 running TC5
    - CTS500
    - CTS1310
    - CTS3010

To enable DTLS negotiation for content encryption:

1 In a new or existing **Profile**, click the **Advanced** tab.



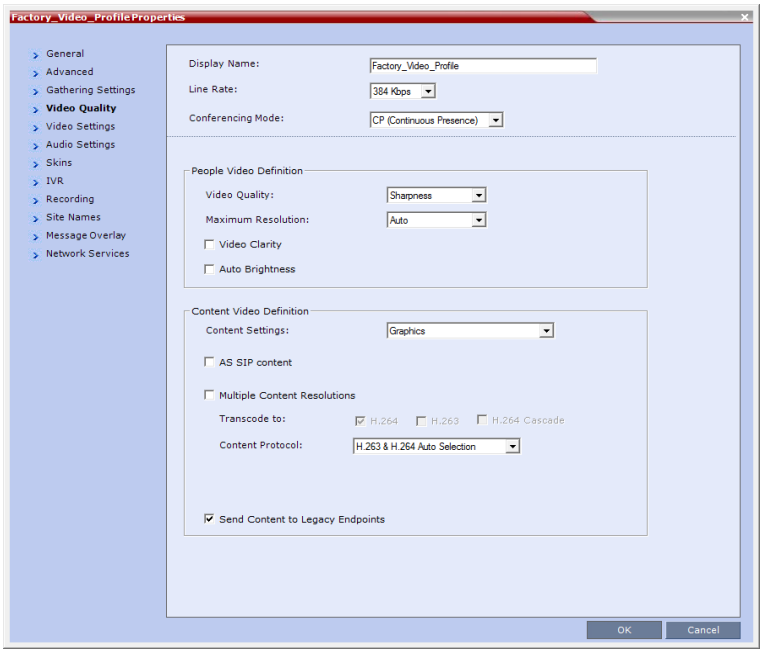
2 Set **Encryption** to either **Encrypt All** or **Encrypt when possible**.

3 Set the FORCE_ENCRYPTION_FOR_UNDEFINED_PARTICIPANT_IN_WHEN_AVAILABLE_MODE *System Flag* to **NO**

These setting will enable encrypted and non-encrypted *H.323* participants to connect to encrypted or non-encrypted conferences.

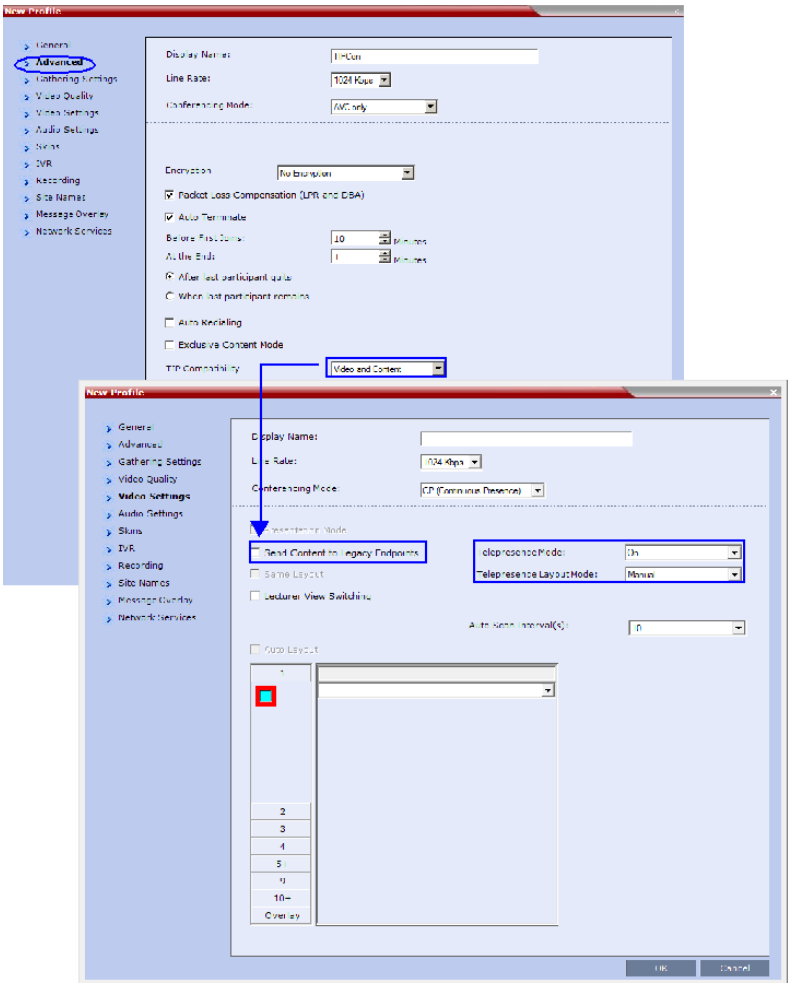
For more information see Encryption.

a Click the *Video Quality* tab.



Content Settings is disabled if *TIP Compatibility* is set to **Video and Content** in the *Advanced* tab.

b Click the *Video Settings* tab.



c Set the *Telepresence Mode* to **Auto/On** and select the *Telepresence Layout Mode*.

4 Assign the *New Profile* to the *Meeting Room.* For more information see Creating a New Meeting Room.

5 Configure a *Virtual Meeting Room* (*VMR*) on the *DMA.*

The procedures for configuring *DMA* are described in detail in the *Polycom Unified Communications Deployment Guide for Cisco Environments*.

Resolution Configuration

The resolution configuration dialog box is not applicable to TIP-enabled conferences as it uses fixed settings:

HD Video Resolutions for *TIP* calls are determined according to the following table:

TIP HD Video Resolution by Line Rate

Line Rate	Video Resolution
Line Rate >=3Mbps	HD1080p30
3Mbps > Line Rate >= 936kbps	HD720p30
Line Rate < 936kbps	Call is dropped.

Endpoints

1. Configure *HDX* endpoints to register to *Lync Server.*

   The procedures for configuring *HDX* endpoints are described in detail in the *Polycom Unified Communications Deployment Guide for Microsoft Environments*.

2. Configure *H.323* endpoints to register to *DMA* as *SIP Proxy*

   The procedures for configuring *SIP* endpoints are described in detail in the *Polycom Unified Communications Deployment Guide for Cisco Environments.*

3. Configure *SIP* endpoints to register to:

   - *DMA* as *SIP Proxy*
   - *Lync Server* as *SIP Proxy*
   - *CUCM* as *SIP Proxy*

   The procedures for configuring *SIP* endpoints are described in detail in the *Polycom Unified Communications Deployment Guide for Cisco Environments.*

4. Configure *TIP* endpoints to register to:

   - DMA
   - CUCM

   The procedures for configuring *TIP- enabled* endpoints are described in detail in the *Polycom Unified Communications Deployment Guide for Cisco Environments.*

Content

Endpoint Registration and *Dialing Method* affect the *Video* and *Content Sharing* characteristics of the conference as detailed in Table 5-9.

Video and Content

Dialing Method	Endpoint Registration		
	Lync	**CUCM**	**DMA**
	ITP /HDX RTV Key is required for HDX and ITP	**ITP /HDX** **TIP Key is required for HDX**	**ITP /HDX** **TIP Key is required for HDX**
HDX to Collaboration Server	• HD H.264 Video • SIP P+C • Content: XGA,5fps • ICE	• HD H.264 Video • No Content • ICE not supported	• HD H.264 Video • SIP P+C • Content: XGA,5fps • ICE not supported
Lync to Collaboration Server	• HD Video (RTV) • No Content Sharing • Content sent to Lync using Content for Legacy Endpoints		
CTS to Collaboration Server	• HD1080p30 • TIP Content Sharing • Content: XGA,5fps		

Operations During Ongoing Conferences

Moving participants between TIP enabled meetings and non TIP enabled meetings is not possible.

Monitoring

CTS Participants

1. In the *Participant List* pane double-click the participant entry. Alternatively, right-click a participant and then click **Participant Properties**.

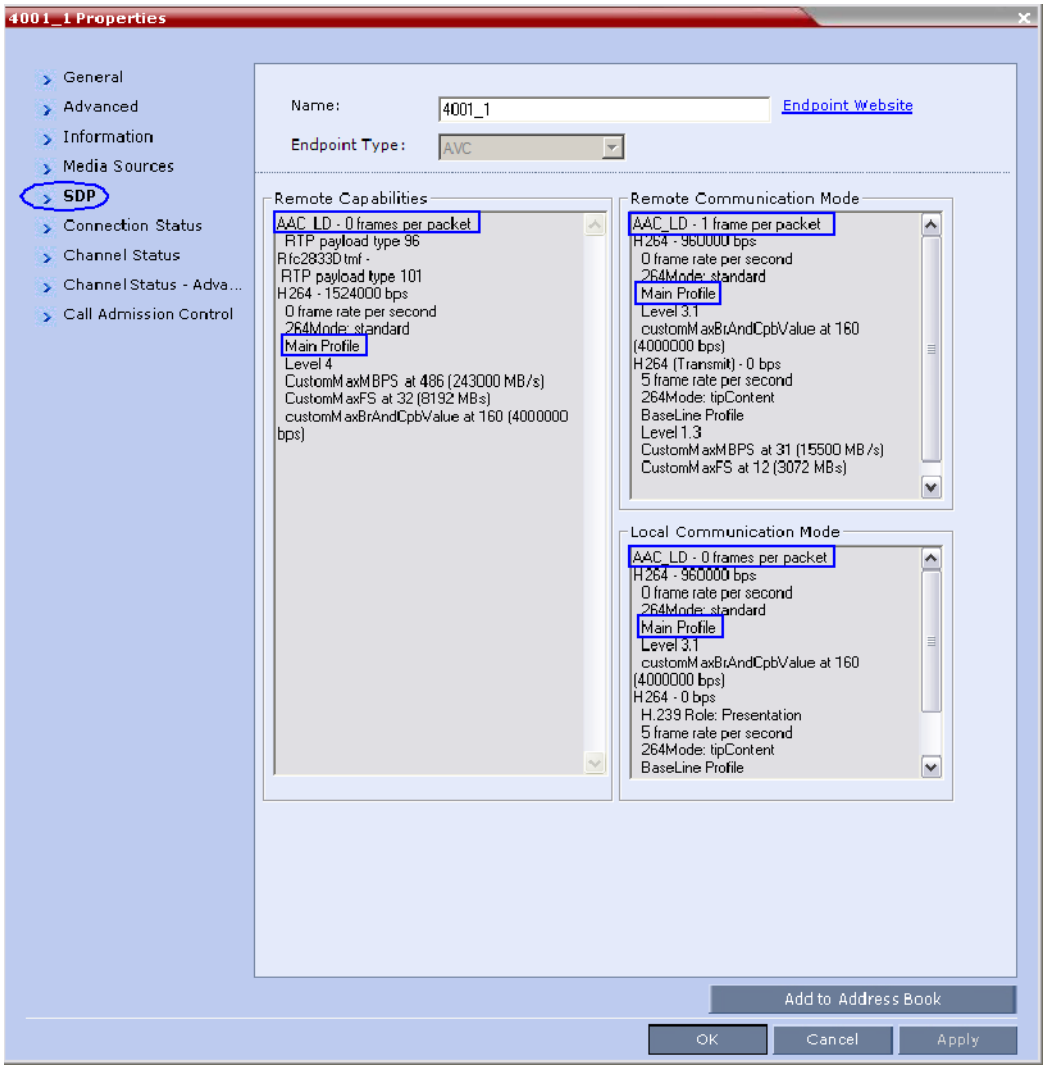
   The *Participant Properties - General* dialog box opens.

2. Click the **SDP** tab.

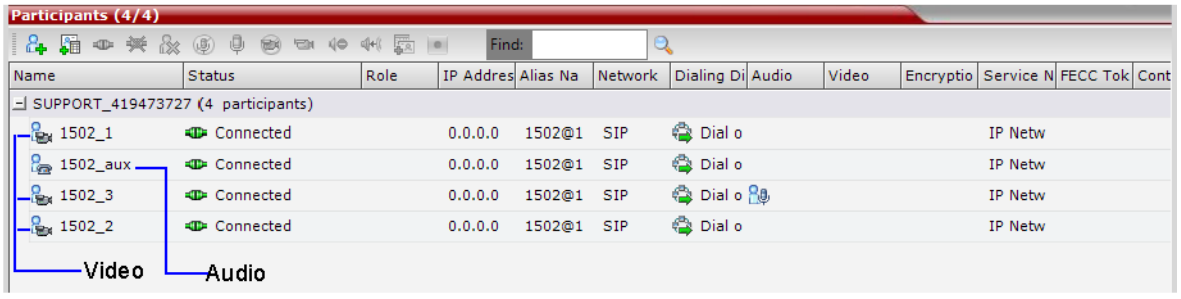
   The following are indicated in the *Remote Capabilities*, *Remote Communication Mode* and *Local Communication Mode* panes:

   - AAC_LD - Audio Protocol

- Main Profile - Video protocol



When viewing *CTS* systems in the *Participants* list, the individual video screens and the *Audio Channel* (*AUX*) of the *CTS* system are listed as separate participants. The *Participant* list below shows a connected *CTS 3000,* a 3-screen system.



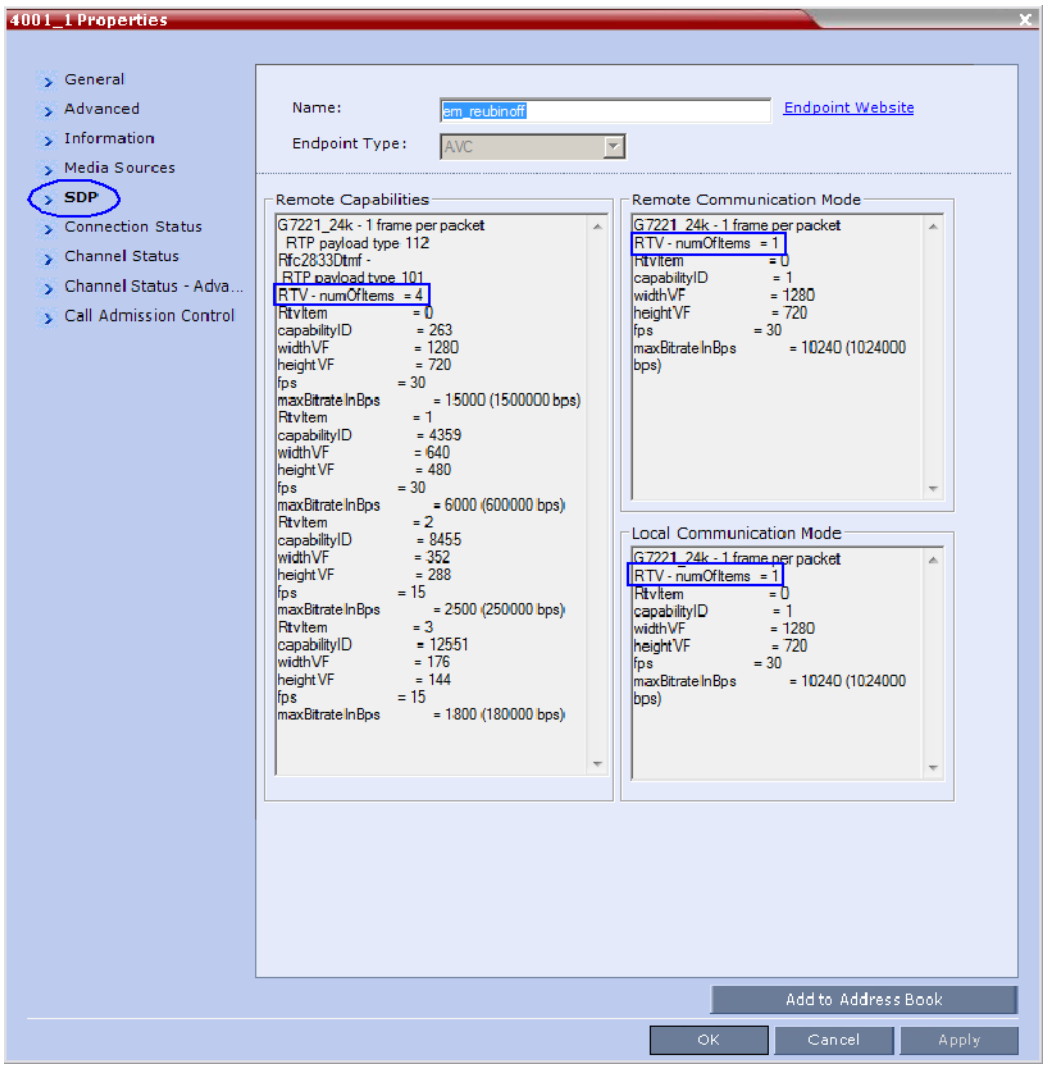
Lync Participants (RTV)

1 In the *Participant List* pane double-click the participant entry. Alternatively, right-click a participant and then click **Participant Properties**.

The *Participant Properties - General* dialog box opens.

2 Click the **SDP** tab.

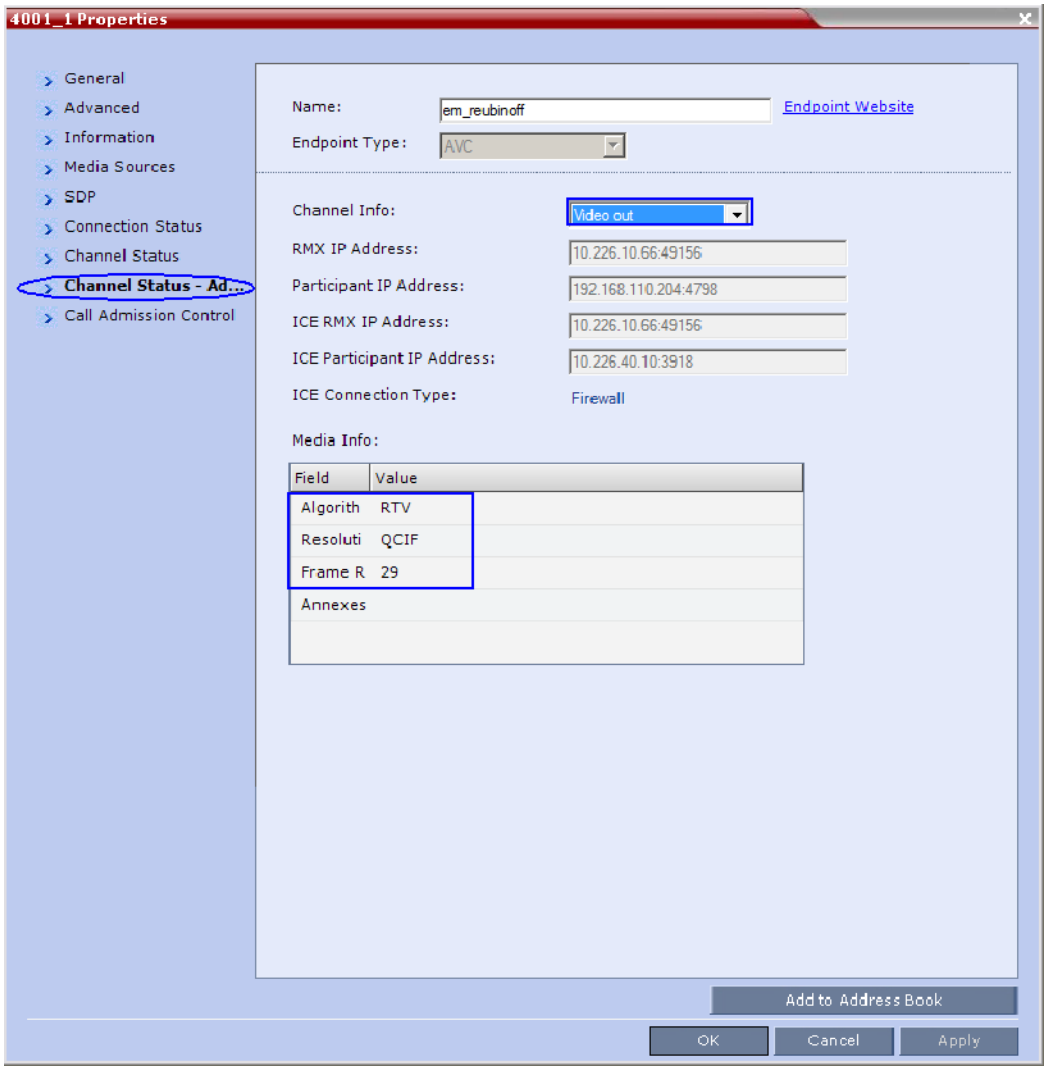
RTV is indicated in the *Remote Capabilities*, *Remote Communication Mode* and *Local Communication Mode* panes:



3 Click the **Channel Status - Advanced** tab

4 In the *Channel Info* drop-down menu select **Video Out.**

Media Info displays *RTV* ***Channel Status*** parameters:



Known Limitations

The following may occur in the collaborative environment:

- Artifacts and ghosting may appear when *Lync Clients* and *CTS* endpoints connect to the *VMR*.
  Frequency: Seldom.
- *Lync Client* receives fast updates (*Intra*) from *CTS 500* endpoints causing the screen to refresh repeatedly.
  Frequency: Often.
- Audio volume and video quality decreases on *CTS* endpoints.
  Frequency: Seldom.
- *CTS* endpoint connects and then disconnects after a few seconds.
  Frequency: Seldom.

- *Lync Clients* always connect *encrypted* to *non-encrypted* conferences.
- *Auto Layout* sometimes ignored for *CTS* and *Lync Clients* calling through *DMA*.
  Frequency: Rarely.
- *Content* sent from *HDX* endpoint is received by all endpoints for 1 second before stopping.
  Conference is *Content to Legacy* enabled and *TIP Compatibility* is *Video Only*.
  Frequency: Often.

Restoring Defaults and System Recovery

The *USB* port of a RealPresence Collaboration Server 800s in secure mode can be used to:

- Restore the RealPresence Collaboration Server 800s to *Factory Security Defaults* mode *(https → http).*
- Perform a Comprehensive Restore to Factory Defaults
- Perform an *Emergency CRL (Certificate Revocation List) Update*

Restore to Factory Security Defaults

Restore to Factory Security Defaults can be performed by either:

- Inserting a *USB* device such as a mouse or a keyboard into any USB port of the RealPresence Collaboration Server 800s causing it to exit secure mode and return to *Factory Security Defaults* mode. After performing this procedure, logins to the RealPresence Collaboration Server 800s use the **http** command and not the **https** command.

  or

- Using the Polycom USB key that came with the RealPresence Collaboration Server.

To restore the RealPresence Collaboration Server 800s to Factory Security Defaults using a USB device:

1. Insert a USB device into any *USB* port of the RealPresence Collaboration Server 800s.
2. Power the RealPresence Collaboration Server 800s **Off** and then **On**.
3. Login using **http**://<Control Unit IP Address>.

To restore the RealPresence Collaboration Server 800s to Factory Security Defaults using the Polycom USB key:

1. Inset the Polycom USB key that came with the RealPresence Collaboration Server into your computer.

   The *Polycom Documentation* window is displayed.

   a. Select **Open Folder to view files using Windows Explorer**.

b Double-click the **index.hta** file.

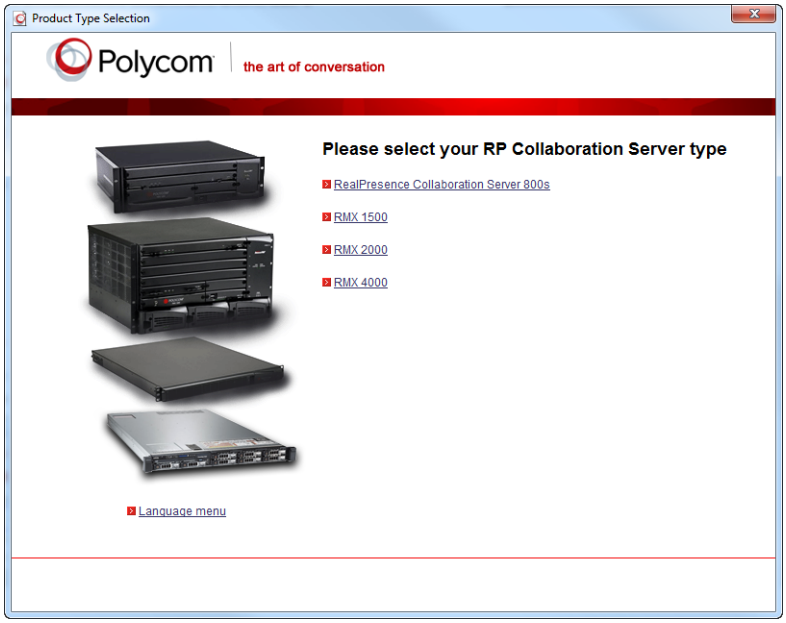
The *Language Menu* is displayed, offering a choice of several languages.

2 Click the documentation language of your choice.

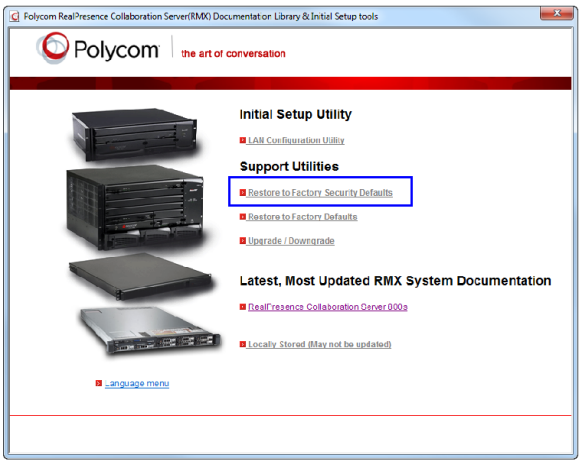
An *End-User Licence Agreement for Polycom Software* is displayed.

3 Read the agreement and click the **Accept Agreement** button.

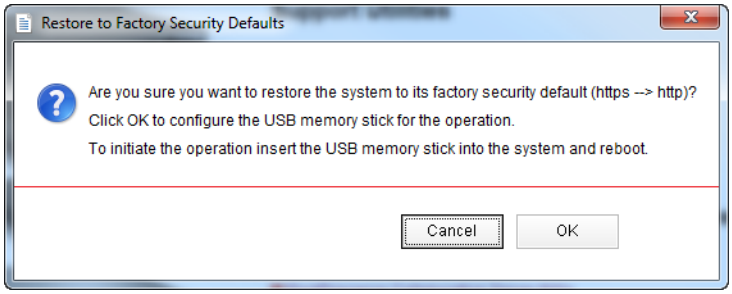
4 In the *Product Type* window, select **RealPresence Collaboration Server 800s**.



5 In the *Initial Setup Utility* dialog box, click the **Restore to Factory Security Defaults** link.



The *Restore to Factory Security Defaults* dialog box is displayed.



6 Click **OK.**

7 Remove the *USB key* from the PC workstation.

8 Insert the *USB key* in any USB port of the RealPresence Collaboration Server 800s.

9 Turn off the RealPresence Collaboration Server 800s, then turn it on.

10 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>`**.

b Click **Enter**.

When the *Collaboration Server Web Client* Login window is displayed, the system was successfully restored to the factory security mode.

11 Remove the *USB key* from the RealPresence Collaboration Server 800s.

Comprehensive Restore to Factory Defaults

The *Comprehensive Restore to Factory Defaults* deletes the following files:

- CDR
- Address Book
- Log Files
- Faults
- Dump Files
- Notes

In addition, all the conferencing entities are deleted:

- Entry Queues
- Profiles
- Meeting Rooms
- IVR Services
- Default Network IP Service
- Log Files
- CFS license information
- Management Network Service

The RealPresence Collaboration Server 800s is restored to the settings it had when shipped from the factory. The *Product Activation Key* is required to re-configure the *Management Network Service* during the *First Entry Configuration.*

Comprehensive Restore to Factory Defaults Procedure

Restoring the *RealPresence Collaboration Server 800s* to *Factory Defaults* consists of the following procedures:

A: Backup Configuration Files

- These files will be used to restore the system in Procedure **C**.

B: Restore to Factory Defaults

- Restart the system with the configured Polycom USB key plugged into any *USB* port.

C: Optional. Restore the System Configuration From the Backup

- Apply the backup file created in procedure **A**.
- Restart the RealPresence Collaboration Server 800s.

(If the RealPresence Collaboration Server 800s is unresponsive after these procedures, a further restart may be necessary.)

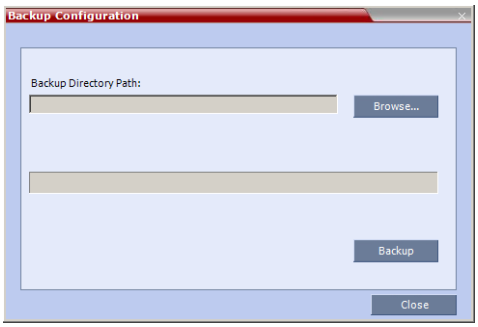
Procedure A: Backup Configuration Files

The *Software Management* menu is used to backup and restore the configuration files of the RealPresence Collaboration Server 800s and to download MCU software.

To backup configuration files:

1. On the *Collaboration Server Menu*, click **Administration > Software Management > Backup Configuration**.

   The *Backup Configuration* dialog box is displayed.



2. Click Browse.
3. Browse to the *Backup Directory Path* and then click **Ok.**
4. Click **Backup**.

Procedure B: Restore to Factory Defaults

To perform a Comprehensive Restore to Factory Default perform the following steps:

1. Insert the Polycom USB key that came with the RealPresence Collaboration Server into your computer.

   The *Polycom Documentation* window is displayed.

   a. Select **Open Folder to view files using Windows Explorer**.

   b. Double-click the **index.hta** file.

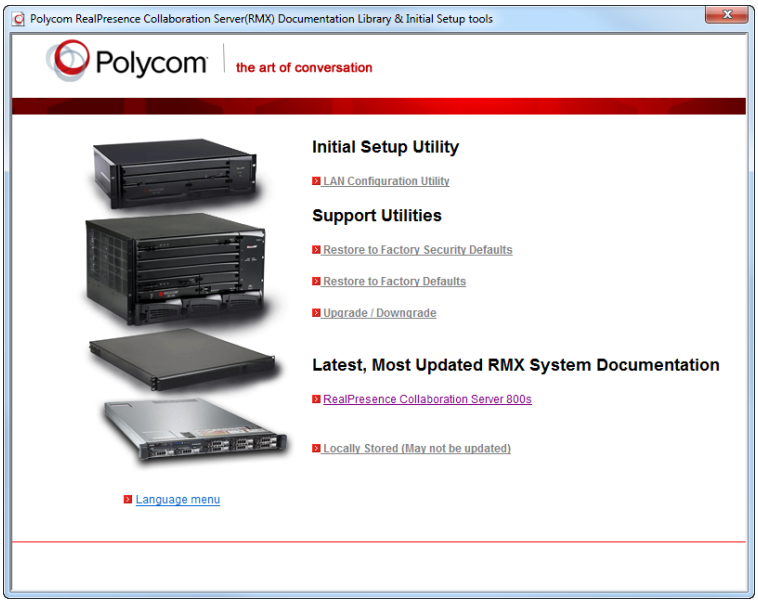
   The *Language Menu* is displayed, offering a choice of several languages.

2. Click the documentation language of your choice.

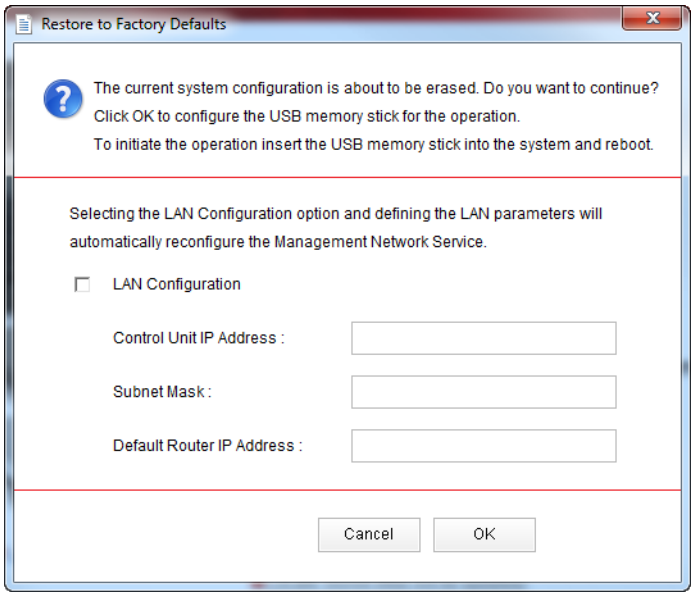
   An *End-User Licence Agreement for Polycom Software* is displayed.

3. Read the agreement and click the **Accept Agreement** button.
4. In the *Product Type* window, select **RealPresence Collaboration Server 800s**.

5 In the *Initial Setup Utility* window, click the **Restore to Factory Defaults** link.



The *Restore to Factory Security Defaults* dialog box is displayed.



6 **Recommended:** Click the **Lan Configuration** check box and modify the following parameters in the utility's dialog box using the information supplied by your network administrator.

- Control Unit IP Address
- Subnet Mask
- Default Router IP Address

The restore to Factory Defaults removes the Management Network Service. If the above fields addresses are not set, the *Control Unit IP Address* of the Management Network Service will be 192.168.1.254.

7 Click **OK.**

8 Remove the *USB key* from the PC.

9 Insert the *USB key* in any USB port of the RealPresence Collaboration Server 800s.

10 Turn off the RealPresence Collaboration Server 800s, then turn it On.

11 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 following format:
`http://<Control Unit IP Address>,`
as defined in the USB key. If no *Control Unit IP Address* was defined, the IP address will be 192.168.1.254.

b Click **Enter**.

When the *Collaboration Server Web Client* Login window is displayed, the system was successfully restored to *factory defaults*.

12 Optional. Restore the system using *Procedure C: Restore the System Configuration From the Backup* on page 873 below.

Procedure C: Restore the System Configuration From the Backup

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.
3. Click the **Restore** button.
4. When the **Restore** is complete, restart the RealPresence Collaboration Server 800s.

   RealPresence Collaboration Server 800s system settings, with the exception of User data, are restored.
5. Restore *User* data by repeating **step** 1 to **step** 3 of this procedure.

System Recovery Using the Recovery DVD

You can use the system recovery disk to re-install the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition application in case of system crashes or server hard disk damage.

Two DVDs are shipped with the Polycom® RealPresence® Collaboration Server 800s and Polycom® RealPresence® Collaboration Server Virtual Edition, the Polycom Recovery DVD, and the Dell Diagnostics DVD.

Preparation for System Recovery

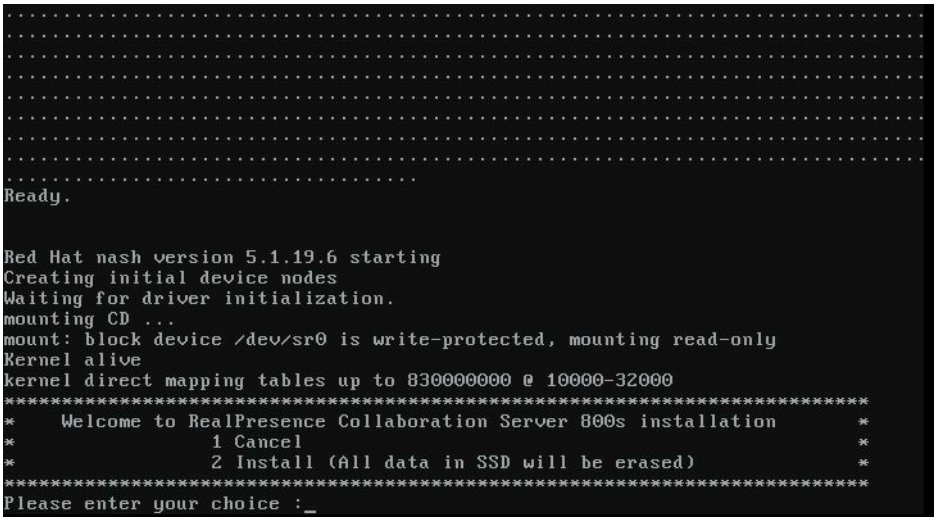
Before performing the recovery procedure it is necessary to modify the Factory Default Management Network Settings on the USB Utilities and Documentation memory stick. For details, see the *<Italic>RealPresence Collaboration Server 800s Getting Started Guide*.

Performing the Recovery Process

To re-install the system using the System Recovery DVD:

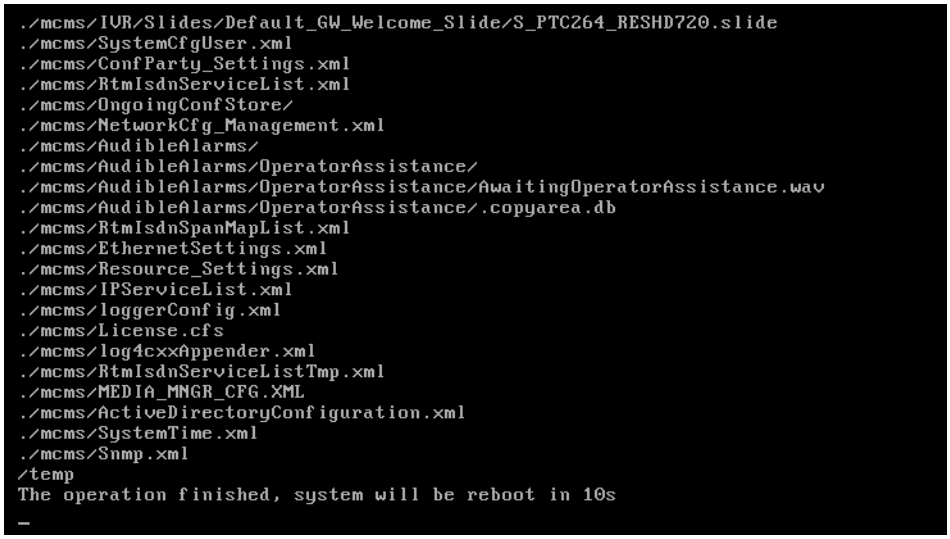
1. Turn off the RMX™.
2. Insert the USB memory stick into any USB slot.
3. Connect a USB keyboard and monitor.
4. Turn on the RMX™.
5. Insert the *Polycom Recovery DVD* that comes with your system.

   The RMX™ loads the *Polycom Recovery DVD*.



6. Press 2 and then Enter.
7. The RMX™ installation starts.

   The whole procedure may take 15 minutes. When the installation completes, the system reboots.



   When the system reboots the DVD is ejected.

8 Remove the DVD from the server and close the DVD drive.

Completing the System Configuration

To complete the system configuration:

1 Connect to the MCU using the default Polycom user name and Polycom password. For details, see the <*Italic*>*RealPresence Collaboration Server 800s Getting Started Guide,* Procedure 3: Connecting to the MCU..

2 Enter the activation key.

3 Define the IP Network Service for the media and signaling using the Fast Configuration Wizard. For details, see the <*Italic*>*RealPresence Collaboration Server 800s Getting Started Guide,* Fast Configuration Wizard.

4 **Optional:** Restore the system configuration. For details, see Software Management.

5 Remove the USB memory stick from the MCU.

The system recovery is complete.

Appendix K - SIP RFC Support

SIP RFC Support in RealPresence Collaboration Server (RMX) 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)	
4568	SDP Security Descriptions	
4573	H.224 RTP Payload (FECC)	

SIP RFC	Description	Note
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	