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Patent Information
The accompanying product may be protected by one or more U.S. and foreign patents and/or pending patent applications held by Polycom, Inc.
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System Overview

The product names, Polycom® RealPresence® Collaboration Server 1500, 1800, 2000, 4000 and RMX® 1500, 1800, 2000, 4000 are used interchangeably throughout this Guide.

About the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Getting Started Guide


This guide will help you understand the Polycom video conferencing components, and provides a description of basic conferencing operations.

This guide will help you perform the following tasks:

- Install the MCU
  - Unpack the RealPresence Collaboration Server system and install it on a rack.
  - Connect the required cables to the RealPresence Collaboration Server.
- Perform basic configuration procedures.
- Connect to the MCU.
- Start a new conference directly on the MCU and connect participants/endpoints to it (optional).
- Monitor ongoing conferences.
- Perform basic operations and monitoring tasks.

The Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide provides more in-depth information on configuring and managing the system, and performing the following tasks:

- Configure the MCU to support special call flows and conferencing requirements, such as Cascading conferences.
- Advanced conference Management.
- Manage and troubleshoot the MCU performance.
Prerequisites

This guide assumes the user has the following knowledge:

- Familiarity with Windows® XP, Windows® 7, and Windows® 8 operating systems and interface.
- Familiarity with Microsoft® Internet Explorer® Version 7, 8, 9, and 10.
- Basic knowledge of video conferencing concepts and terminology.

Who Should Read This Guide?

System administrators and network engineers should read this guide to learn how to properly install and set up Polycom Collaboration Server systems. Chairpersons and system operators should read this guide to learn how to use the Collaboration Server Web Client/RMX Manager to run conferences and monitor them.

For more information on configuring and managing the system, refer to the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide.

How This Guide is Organized

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

<table>
<thead>
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<th>Description</th>
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<tr>
<td>Bold</td>
<td>Highlights interface items such as menus, soft keys, flag names, and directories. Also used to represent menu selections and text entry to the Collaboration Server Web Client or the RMX Manager.</td>
</tr>
<tr>
<td>Italics</td>
<td>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.</td>
</tr>
<tr>
<td>Underlined Blue</td>
<td>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.</td>
</tr>
<tr>
<td>Blue Text</td>
<td>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.</td>
</tr>
<tr>
<td>&lt;variable name&gt;</td>
<td>Indicates a variable for which you must enter information specific to your installation, endpoint, or network. For example, when you see &lt;IP address&gt;, enter the IP address of the described device.</td>
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The Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 is a high performance, scalable, IP (H.323 and SIP) and ISDN/PSTN (Collaboration Server 1500/2000/4000 only) MCU that provides feature-rich and easy-to-use multipoint voice and video conferencing.

The Collaboration Server 1500/2000/4000 meets International Telecommunication Union - Telecommunication Standardization Sector, (ITU-T, formerly CCITT) standards for multipoint multimedia bridging devices, and meets ETSI standards for telecommunication products. In addition, it has been designed in compliance with IETF (Internet Engineering Task Force).

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 telepresence and video conferencing systems, tablets and smartphones, or from your Web browser.
- **Polycom® RealPresence 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™ (RPAD)** - removes communication barriers and enables internal and external teams to collaborate more easily and effectively over video.

The following diagram describes the multipoint video conferencing configuration with the RealPresence Collaboration Server system as a standalone MCU system.
Multipoint Video Conferencing using a RealPresence Collaboration Server (RMX) 1500/2000/4000

Diagram showing the connections between ISDN Endpoints, PSTN Phones, E1/T1 PRI Lines, RealPresence Collaboration Server, Collaboration Server Web Client, LAN, IP Phone, and PC.
The RealPresence Collaboration Server system 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 (RMX) 1500/2000, MCU management and IP conferencing are performed via a single LAN port. The networks can be separated in Maximum Security Environments.

In the RealPresence Collaboration Server (RMX) 4000/1800, MCU management and IP conferencing are performed via two different LAN ports. The networks can be separated in Maximum Security Environments. Management and IP Service can be combined in one LAN port or separate to different ports.

The RealPresence Collaboration Server 1800 system is an IP-Only MCU and does not support ISDN connections.

RealPresence Collaboration Server (RMX) 1500 supports one ISDN card with up to 4 E1/T1 PRI lines.

RealPresence Collaboration Server (RMX) 2000 and RealPresence Collaboration Server (RMX) 4000 support a maximum of two RTM ISDN cards, each providing connection for up to either 7 E1 or 9 T1 PRI lines.

On RealPresence Collaboration Server (RMX) 1500/2000/4000, E1 and T1 connections cannot be used simultaneously.
RealPresence Collaboration Server Main Features

The RealPresence collaboration server offers:

Conferencing Modes

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)
- AVC-based Conferencing - Video Switching (Collaboration Server (RMX) 1500/2000/4000 only)
- SVC-based Conferencing (Media Relay) - SVC Only
- Mixed AVC and SVC Conferencing - CP and SVC

AVC-based 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 Video Session Type for AVC-based conferencing.

In the Collaboration Server (RMX) 1500/2000/4000, the MCU also offers the Video Switching conferencing Mode. Video Switching is not supported with Collaboration Server (RMX) 1800.

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).

CP Only (CP Transcoding) - AVC-based Conferencing

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

- Video protocols: H.261, H.263, H.264 Base and High profile and RTV
- Video Resolutions: from QCIF, CIF and up to 1080p60
- Frame rates up to 60fps

By default every conference, Entry Queue and Meeting Room has the ability to declare the maximum CP resolution as defined for the system.

The MCU processes the received video, transcodes it and sends the resulting video streams to the endpoints.

The Continuous Presence (CP) conference enables viewing flexibility by offering multiple viewing options and video cell layouts for video conferencing.
Video Switching Conferencing Mode (Collaboration Server 1500/2000/4000 only)

In Video Switching (VSW) mode all participants see the same video picture (full screen). Only one CIF video resource is used for each connection. Video Switching mode is not supported in Collaboration Server 1800.

In VSW conferences:

- All endpoints must connect to the conference at the same line rate. Line rates range from 192Kbps to 6Mb. The Collaboration Server will always connect participants at the highest possible video quality that is supported by the conference Line Rate.

- All endpoints must connect to the conference at the same video protocol and resolution. Video Protocol and Resolution ranges from H.261 CIF to H.264 1080p60.

- Compliant endpoints must connect to conferences at the conference line rate and resolutions. Endpoints that do not meet the conference requirements connect as Secondary (Audio Only) if there are resources defined for Voice connection or they will disconnect from the conference if there are not such resources.

AVC-based Video Resolutions

The resolution in which participants connect to the conference is determined by system according to their connection line rate and the endpoint’s capabilities.

Resolution Configuration for AVC-based CP Conferences

The minimum threshold line rates at which endpoints are connected to the conference at the various video resolutions are determined in the Resolution Configuration dialog box. Using the Resolution Configuration sliders, you can modify the video resolution decision matrix, effectively creating your own decision matrix.

AVC-based Video Switching Resolutions (Collaboration Server (RMX) 1500/2000/4000 only)

In Video Switching (VSW) conferences, participants must connect at the conference line rate, video protocol and resolution. Endpoints that cannot connect using these settings will be connected using Audio only and their status will appear as Secondary (no video).

The video quality for the conference is determined by the conference video protocol and resolution and it should be set based on the endpoint with the lowest capabilities that will be connected to the conference.

- Available resolutions in Video Switching mode are: H.261 CIF, H.263 CIF, H.264 CIF, H.264 SD 30, H.264 720p 30, H.264 720p 60 and H.264 1080p60.

SVC-based Conferencing (Media Relay)

Media Relay SVC Conferencing is based on the Scalable Video Coding (SVC) video protocol and Polycom’s proprietary Scalable Audio Coding (SAC) protocol. It offers high resolution video conferencing with low end-to-end latency, improved Error Resiliency and higher system capacities.
The Polycom multipoint media server serves as an integrated media relay engine that provides media streams for displaying conferences at low latency video experience in video conferences. For more details, see Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, SVC-based Conferencing.

SVC-based Video Resolutions

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

Using the SVC video protocol, SVC conferences relays the received video streams to the SVC-enabled endpoints at different resolutions, frame rates and line rates according to the endpoint’s display capabilities and layout configurations without sending the entire video layout to the endpoints.

For more details, see Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, AVC Basic Conferencing Parameters.

Mixed CP and SVC Conferencing

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

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

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

Rich Video Layout Display Options

Video Layout describes the arrangement of cells on the screen of an endpoint, with each cell showing the video of one of the conference participants. Usually, the participant cannot see himself/herself in the layout. A selection of layouts are available to accommodate different numbers of participants and conference settings.

The Collaboration Server (RMX) 1500/2000/4000 supports the VUI annex to the H.264 protocol for endpoints that transmit wide video format instead of 4CIF resolution.

The following layouts are available as both Conference (all participants in the conference see the same layout) and Personal (individually selected per participant) Layouts.
The following Overlay Layouts are available as Conference Layouts. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Overlay Layouts.

Telepresence Mode Layouts

TPX (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.

The RealPresence Collaboration Server enables Telepresence Rooms to connect to conferences where point-to-point connections cannot be used.
Additional video layouts have been created to give Telepresence operators more video layout options when configuring TPX room systems. These additional video layout options are available for selection when Telepresence is selected in the conference profile.

The following layouts are specifically intended for use in Telepresence Mode.

Multiple Switching Modes for Layout Display

If the number of participants is higher than the number of video windows in the selected layout, switching between video participants can be performed in one of these modes:

- Voice activation (default mode)
- The Collaboration Server user forces participants to selected video window
- Lecture Mode - The lecturer is viewed in full screen by all conference participants, while the audience is “time-switched” in the speaker’s view
- Presentation Mode - When the speaker’s presentation extends beyond a predefined time, he/she becomes the current lecturer and the conference switches to Lecture Mode

Various Methods for Creating Conferences

The collaboration Server offers various methods for creating, starting and scheduling conferences:

- On Demand Conferencing
- Scheduled Conferencing / Reservations (AVC-based Conferencing)
- Permanent Conference
- Polycom Conferencing for Microsoft Outlook® (AVC-based Conferencing)

On Demand Conferencing

The following options are available to set up conferences:

- New Conference – Set up once, use once.
  The conference is deleted from the MCU after it ends.
- **Meeting Rooms** – Set up once, use many times. Meeting Rooms are saved in memory (using no resources) and can be activated as many times as needed.
- **Ad Hoc Entry Queue** (AVC CP Only and Mixed CP and SVC Conferencing) – No setup, a new conference can be created when an AVC participant dials in and enters a conference ID that is not being used by an existing conference or Meeting Room.
- **Collaboration Server (RMX) 1500/2000/4000 only: Gateway calls** (AVC Only Conferencing) – From IP endpoints to other participants, using the direct dialing method, with up to 10 destination numbers contained in a single dial string.

**Scheduled Conferencing / Reservations (AVC-based Conferencing)**

Reservations provide calendar-based scheduling of single or recurring conferences. These conferences can be launched immediately or become ongoing, at a specified time on a specified date. For more details, see *Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide*, Scheduling Reservations.

**Permanent Conference**

A **Permanent Conference** is an ongoing conference with no predetermined End Time, continuing until it is terminated by an administrator, operator or chairperson. For more details, see *Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide*, Permanent Conference.

**Polycom Conferencing for Microsoft Outlook® (AVC-based Conferencing)**

*Polycom Conferencing for Microsoft Outlook* is implemented by installing the Polycom Conferencing Add-in for Microsoft Outlook on Microsoft Outlook -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.


**Operator Conference**

In **Continuous Presence** conferencing, a special conference that enables the MCU user, acting as an operator, to assist participants without disturbing ongoing conferences and without being heard by other conference participants. The operator can move a participant from an Entry Queue or ongoing conference to a private, one-on-one conversation in the Operator conference. For more details, see *Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide*, Operator Conferences.

**Cascading Conferences (AVC-based Conferencing)**

Cascading enables administrators to connect one conference directly to one or several conferences usually running on different MCUs, depending on the topology, creating one large conference. Supported topologies are:

- Basic Cascading of two MCUs.
- Star Topology.
- Multi Hierarchy Cascading (MIH).
Gateway Calls (Collaboration Server (RMX) 1500/2000/4000, AVC-based Conferencing only)

Using a special Gateway Profile, the Collaboration Server 1500/2000/4000 can be used as a gateway that provides connectivity across different physical networks such as H.323, SIP, ISDN and PSTN. The Gateway also provides connectivity between the ISDN/PSTN endpoints and the RealPresence DMA system.

Gateway calls are not supported with Collaboration Server (RMX) 1800.

For more details, see Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Gateway Calls.

Methods for connecting to conferences

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

Using Entry Queues to Access Conferences

An Entry Queue is a special routing lobby for video and audio participants. After dialing the Entry Queue ID or in Collaboration Server (RMX) 1500/2000/4000 a dial-in number (ISDN/PSTN), voice prompts from an IVR service are used to connect the participants to the appropriate conference.

This service can also be used (if required) to verify the participant’s right to start an Ad Hoc conference or to join an ongoing conference.


AVC-based Connections

In AVC-based connection, IPv4, IPv6, ISDN and PSTN (not in Collaboration Server 1800) H.323 and SIP communication protocols are supported for connection to the conference.

Endpoints can connect using the following Video Protocols: H.263, H.264 Base and High Profile and RTV.

- **Dial-out:** automatically, to pre-defined participants (line rate detection is automatic)
- **Dial-in:**
  - for participants defined in advance
  - for undefined participants connecting directly to a conference (IP and ISDN/PSTN)
  - for undefined participants via a single dial Entry Queue (IP and ISDN/PSTN)

SVC-based Connections

In SVC-based connections, IPv4 SIP communication protocol is supported for connection to the conference. Endpoints can connect using the SVC Video Protocol.

- **Dial-out:** not supported for SVC participants in SVC Only or mixed CP and SVC conferences
- **Dial-in:**
  - for participants defined in advance
for undefined participants connecting directly to a conference (SIP)

Content Sharing using H.239 / People+Content Protocols

Endpoints that are H.239 protocol compliant can share content. By default, all Conferences, Entry Queues, and Meeting Rooms launched on the Collaboration Server have H.239 capability. This protocol is also supported in MIH Cascading conferences.

Conferences can include a mix of endpoints that support H.239 or People+Content. People+Content is Polycom’s proprietary equivalent of H.239.

For more details, see Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Sharing Content During Conferences.

IVR-Enabled Conferencing

Interactive Voice Response (IVR) is a software module that automates the connection process and lets participants perform various operations during ongoing conferences. The participants use their endpoints’ keypads, remote controls and touch control devices to interact with the conference’s menu-driven scripts using DTMF codes.

Operations that can be performed by participants or chairpersons during a conference include:

- Initiate Polycom’s Click&View™ application to change the local screen layout (AVC-based conferencing).
- Mute or unmute the participant’s audio channel.
- Adjust the participant’s broadcasting and listening audio volume (AVC-based conferencing).
- Play the Help menu.
- Mute or unmute undefined dial-in participants upon their connection to the conference.
- Request a Roll Call and stop the Roll Call names review (AVC-based conferencing)
- Secure and unsecure a conference.
- Request individual and conference assistance (CP conferences).
- Manually terminate the conference.


PCM (CP AVC-based Conferences)

The Personal Conference Manager (PCM) interface enables the conference chairperson to control various conference features using his/her endpoint’s remote control device.

One PCM session per conference can be activated at any time during the conference.

The following conference operations can be performed:

- Initiate Polycom’s Click&View™ application to change the local screen layout.
- Invite participants to connect to the conference.
- View and control the audio and video of each connected endpoint.
- Camera Control - control the camera of a remote endpoint using (FECC).
- Control the camera of a connected endpoint.
● Video Force a specific participant to a specific window of the video layout.
● Initiate and control recording of the conference.
● Disconnect a participant.
● Terminate the conference.

**Video Clarity™ (AVC-based Conferences)**

The *Video Clarity* feature applies video enhancing algorithms to incoming video streams of resolutions up to and including SD. Clearer images with sharper edges and higher contrast are sent back to all endpoints at the highest possible resolution supported by each endpoint.

All layouts, including 1x1, are supported.

*Video Clarity* can only be enabled for *Continuous Presence* conferences.

**Security**

- In conferences Media Encryption is available at conference and participant levels, based on AES 128 Media Encryption and DH 1024 Key Exchange standards.
- Secured Communication Mode (SSL/TLS).
- Secured conferences via DTMF codes and limited monitoring of secured conferences.
- Auditor to analyze configuration changes and unusual or malicious activities in the Collaboration Server.
- Network security can be enhanced by separation of the Signaling and Management Networks.
- Collaboration Server users can be disabled by the administrator, or automatically when inactive. Disabled Users can be enabled by the administrator.
- Multiple Network Services (Collaboration Server (RMX) 1500/2000/4000)
- The following additional Security features can be implemented:
  - Password management:
    - Strong Passwords and password re-use / history rules,
    - password aging rules, password change frequency and forcing password change
    - Conference and Chairman Passwords
    - Locking out Users
    - Displaying the User Login record
  - Controlling the User Sessions includes:
    - Limiting the maximum number of concurrent user sessions
    - Connection Timeout
    - User session timeout
    - Limiting the maximum number of users that can connect to the system

**LAN Redundancy**

Enables the redundant LAN port connection to automatically replace the failed LAN port by using another physical connection and *NIC (Network Interface Card)* should a LAN port fail.
Conference Management and Monitoring Features

The Collaboration Server Web Client and RMX Manager application provide capabilities for management and monitoring of participants and conferences, including the following:

CP AVC-based and SVC-based Conferencing

- Active display of all conferences and participants
- Real-time monitoring of each participant’s connection status and properties.
- Automatic termination of idle (no participants) conferences.
- Automatic extension of conference duration.
- Control of listening and broadcasting audio volume for individual participants.
- Support for SNMP versions 1, 2 and 3.
- Easily accessed Call Detail Records (CDR) for administrator.
- Active display of all system resources.
- Hot Backup implements a high availability and rapid recovery solution.

CP AVC-based Conferencing

- Lecture Mode or Presentation Mode in Continuous Presence conferences.
- Far End Camera Control (FECC/LSD) in video conferences.
- Auto Gain Control (AGC) noise and audio volume regulation for individual participants.
- Conference control via DTMF codes from participant’s endpoint or telephone.
- Entry, exit and end-of-conference indications.
- Option to limit the properties display of conference participants in secured conferences.
- Multiple drag & drop of participants.
- Closed Caption provides real-time text transcriptions or language translations of the video conference (Collaboration Server (RMX) 1500/2000/4000).
- Message Overlay allows messages to be sent to all or specific participants in an ongoing conference.
- PCM enables the conference chairperson to control various conference features using his/her endpoint’s remote control device.
- Video Preview allows Collaboration Server users to preview video sent from the participants to the conference and from the conference to the participants.
- Auto Redial when Endpoint Drops instructs the Collaboration Server to automatically redial IP and SIP participants that have been abnormally disconnected from the conference.
- Operator Assistance & Participant Move for conferences in CP mode.
Card Configuration Modes (Collaboration Server (RMX) 1500/2000/4000 only)

The media card installed in the system determines the Card Configuration Mode. The Card Configuration Mode represents different generations of the media card. Each new generation provides additional functionality, higher video resolutions and higher resource capacity.

Only one Media Card type can be installed in any Collaboration Server, which sets the Card Configuration Mode for that Collaboration Server:


Workstation Requirements

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

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

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 Web Client/ RMX Manager Environment Interoperability Table**

<table>
<thead>
<tr>
<th>Web Browser</th>
<th>Operating System</th>
</tr>
</thead>
<tbody>
<tr>
<td>Internet Explorer 7</td>
<td>Windows Vista™</td>
</tr>
<tr>
<td></td>
<td>Windows 7</td>
</tr>
<tr>
<td>Internet Explorer 8</td>
<td>Windows 7</td>
</tr>
<tr>
<td>Internet Explorer 9</td>
<td>Windows 7 and Windows 8</td>
</tr>
<tr>
<td>Internet Explorer 10*</td>
<td>Windows 8</td>
</tr>
</tbody>
</table>

* Internet Explorer 10 has been tested on the RMX 1800. If for any reason it fails to run, right-click the Internet Explorer icon and select Run As Admin. .Net Framework 2.0 is required and installed automatically.

If ActiveX installation is blocked please see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, ActiveX Bypass.

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

When installing the Collaboration Server Web Client, Windows Explorer >Internet Options> Security Settings must be set to **Medium** or less.
It is not recommended to run Collaboration Server Web Client and Polycom CMA 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.

Windows 7™ Security Settings

If Windows 7 is installed on the workstation, Protected Mode must be disabled before downloading the software to the workstation.

To disable Protected Mode:

1. In the Internet Options dialog box, click the Security tab.
The **Security** tab is displayed.
2 Clear the **Enable Protected Mode** check box for each of the following tabs:

- Internet
- Local intranet
- Trusted sites

3 After successful connection to Collaboration Server, the **Enable Protected Mode** check boxes can be selected to enable **Protected Mode** for the following tabs:

- Internet
- Local intranet
Internet Explorer 8 Configuration

When using Internet Explorer 8 to run the Collaboration Server Web Client or RMX Manager applications, it is important to configure the browser according to the following procedure.

To configure Internet Explorer 8:

1. Close all browsers running on the workstation.
2. Use the Windows Task Manager to verify that no iexplore.exe processes are running on the workstation. If any processes are found, use the End Task button to end them.
3. Open Internet Explorer but do not connect to the MCU.
4. In the Internet Explorer menu bar select Tools >> Internet Options. The Internet Options dialog box is displayed with General tab open.

5. In the Browsing history section, click the Delete button. The Delete Browsing History dialog box is displayed.
6 Select the **Temporary Internet** files and **Cookies** check boxes.

7 Click the **Delete** button.

The **Delete Browsing History** dialog box closes and the files are deleted.

8 In the **Internet Options** dialog box, click the **Settings** button.
The **Temporary Internet Files and History Settings** dialog box is displayed.

9 Click the **View objects** button. The Downloaded Program Files folder containing the installed Program Files is displayed.
10 Select the **EMAClassLoader.dll** file and press the **Delete** key on the workstation or right-click the **EMA.ClassLoader.dll** file and then click **Delete**.

11 Close the Downloaded Program Files folder and the **Temporary Internet Files and History Settings** dialog box.

12 In the **Internet Options** dialog box, click the **OK** button to save the changes and close the dialog box.
First Time Installation and Configuration

First Time Installation and Configuration of the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 consists of the following procedures:

1 Preparation
   - Gather Network Equipment and Address Information - get the information needed for integrating the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 into the local network.
   - Unpack the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000.
   - Modify the Management Network parameters on the USB memory stick.

2 Hardware Installation and Setup
   - Mount the Collaboration Server in a rack.
   - Connect the necessary cables.

3 First Entry Power-up and Configuration
   - Power up the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000.
   - Register the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000.
   - Configure the Default IP Network Service.
   - Configure the ISDN/PSTN Network Service (not on supported on the RMX 1800).

Preparation

Gather Network Equipment and Address Information

IP Services
The IP addresses and network parameters which enable communication between the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000, its management application and the conferencing devices are organized in two IP services:

- Management Network (Control Unit)
- Default IP Service (Conferencing Service which includes the signaling and media)

During the First Entry Configuration, the parameters of these two network services are modified to comply with your local network settings.
Management Network

The Management Network enables communication between the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 and the Collaboration Server Web Client and is used to manage the MCU.

The Collaboration Server is shipped with default IP addresses as listed in the RMX 1500/2000/4000 Network Equipment and Address Information and the RealPresence Collaboration Server (RMX) 1800 Network Equipment and Address Information tables.

Management Network Definition

The configuration of the Management Network can be done by two methods:

- **USB Memory Stick (recommended method)** – The system is shipped with a USB Memory Stick containing the default IP addresses for the Control Unit and the shelf management. These IP addresses are first modified in the administrator’s PC and then uploaded to the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000.


For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Appendix G - Configuring Direct Connections to the Collaboration Server.

DHCP is not supported in the Management Network.

Default IP Service (Conferencing Service)

The Default IP Service (Conferencing Service) is used to configure and manage communications between the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 and conferencing devices.

IP Network Services Required Information

When installing the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000, these default IP addresses must be modified to your local network settings. Therefore it is important that before powering up the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 for the first time, that you obtain the information needed to complete the Local Network Settings section of the table from your network administrator.

Conferencing (Media and signaling) and Management networks can be logically separated on the Collaboration Server to provide enhanced security. A number of media and signaling networks can be defined on each MCU. For more information see “Multiple Networks” in the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide.

For each MCU, an IP address must be allocated in the local network for the:

- Control Unit
- Signaling Host
- **Shelf Management** (optional for RealPresence Collaboration Server (RMX) 1500)

An additional IP address is also required for any type of MPM card installed in the Collaboration Server.

Examples:
● **RealPresence Collaboration Server (RMX) 1500** - the network administrator should allocate:
  - **Three** IP addresses in the local network for an Collaboration Server.
  - **Four** IP addresses in the local network for an Collaboration Server, if a separate *Shelf Management IP Address* is required.

● **RealPresence Collaboration Server 1800** - the network administrator should allocate:
  - **Two** IP addresses are required one for management, another for signalling and media.

● **RealPresence Collaboration Server (RMX) 2000** - the network administrator should allocate:
  - **Four** IP addresses in the local network for an Collaboration Server with one *MPMx/MPMRx* card.
  - **Five** IP addresses for an Collaboration Server with two *MPMx/MPMRx* cards.

● **RealPresence Collaboration Server (RMX) 4000** - the network administrator should allocate:
  - **Four** IP addresses in the local network for an Collaboration Server with one *MPMx/MPMRx* card.
  - **Up to seven** IP addresses for an Collaboration Server with up to four *MPMx/MPMRx* cards.

From *Version 8.1*, *MPM+* media cards are not supported.

### RMX 1500/2000/4000 Network Equipment and Address Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Factory Default</th>
<th>Local Network Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Control Unit IP Address</td>
<td>192.168.1.254</td>
<td></td>
</tr>
<tr>
<td>Control Unit Subnet Mask</td>
<td>255.255.255.0</td>
<td></td>
</tr>
<tr>
<td>Default Router IP Address</td>
<td>192.168.1.1</td>
<td></td>
</tr>
<tr>
<td>Shelf Management IP Address</td>
<td>192.168.1.252</td>
<td></td>
</tr>
<tr>
<td>Signaling Host IP address</td>
<td>–</td>
<td></td>
</tr>
<tr>
<td>Media Card 1 IP Address</td>
<td>–</td>
<td></td>
</tr>
<tr>
<td>Media Card 2 IP Address</td>
<td>–</td>
<td></td>
</tr>
<tr>
<td><strong>RealPresence Collaboration Server (RMX) 2000/4000 only</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Media Card 3 IP Address</td>
<td>–</td>
<td></td>
</tr>
<tr>
<td><strong>RealPresence Collaboration Server (RMX) 4000 only</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Media Card 4 IP Address</td>
<td>–</td>
<td></td>
</tr>
<tr>
<td>Gatekeeper IP address (optional)</td>
<td>–</td>
<td></td>
</tr>
<tr>
<td>DNS IP address (optional)</td>
<td>–</td>
<td></td>
</tr>
<tr>
<td>SIP Server IP address (optional)</td>
<td>–</td>
<td></td>
</tr>
</tbody>
</table>
Example: The network administrator should allocate two IP addresses in the local network for a RealPresence Collaboration Server (RMX) 1800: Control Unit and Signaling Host. Since the signaling and media are transferred on the same network, the same IP address is allocated to both.

### RealPresence Collaboration Server (RMX) 1800 Network Equipment and Address Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Factory Default</th>
<th>Local Network Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Control Unit IP Address</td>
<td>192.168.1.254</td>
<td></td>
</tr>
<tr>
<td>Control Unit Subnet Mask</td>
<td>255.255.255.0</td>
<td></td>
</tr>
<tr>
<td>Default Router IP Address</td>
<td>192.168.1.1</td>
<td></td>
</tr>
<tr>
<td>Signaling Host IP address</td>
<td>Same as the Media and defined by the Media IP address</td>
<td></td>
</tr>
<tr>
<td>Media Card 1 IP Address</td>
<td>Same as the Signaling Host IP address</td>
<td></td>
</tr>
<tr>
<td>Gatekeeper IP address (optional)</td>
<td>–</td>
<td></td>
</tr>
<tr>
<td>DNS IP address (optional)</td>
<td>–</td>
<td></td>
</tr>
<tr>
<td>SIP Server IP address (optional)</td>
<td>–</td>
<td></td>
</tr>
</tbody>
</table>

### ISDN/PSTN Services

ISDN/PSTN Services are not supported with Collaboration Server (RMX) 1800.

The ISDN/PSTN Network Service is used to define the properties of the ISDN/PSTN switch and the ISDN lines running from the ISDN/PSTN switch to the ISDN card installed in the Collaboration Server.

Before configuring the ISDN/PSTN Network Service, obtain the following information from your ISDN/PSTN Service Provider:

- Switch Type
- Line Coding and Framing
- Numbering Plan
- Numbering Type
- Dial-in number range

If the Collaboration Server is connected to the public ISDN Network, an external CSU or similar equipment is needed.

### Unpacking the RealPresence Collaboration Server

1500/1800/2000/4000 Hardware Guide

The following procedures have to be performed to install the RealPresence Collaboration Server (RMX) 1500 in your site.
Unpacking the RealPresence Collaboration Server (RMX) 1500

The following procedures have to be performed to install the RealPresence Collaboration Server (RMX) 1500 in your site.

To unpack and lift the RealPresence Collaboration Server (RMX) 1500:

1. When you receive the RealPresence Collaboration Server (RMX) 1500 packing case, inspect the equipment for damage and verify that the components match the packing slip.
2. Open the top cover of the packing case.
   Boxes are placed on the top Stratocell® and contain the following:
   - **Installation Accessories**. This kit contains the power cables, 3 ethernet cables, USB key and documentation.
   - **Rack Installation Accessories**. This kit contains the accessories for the 19” rack. For more details, see Telescopic Rail Runners Accessory Kit.
   - **Optional - ISDN Package**. Contains the ISDN card and ISDN Software License for ISDN/PSTN.

Write down the Collaboration Server’s serial number that is on a sticker on the back of the unit. It will be needed for product registration later in the process.

Make sure that boxes contain all the required parts.

Unpacking the RealPresence Collaboration Server (RMX) 2000

The following procedures have to be performed to install the RealPresence Collaboration Server (RMX) 2000 in your site.

To unpack and lift the RealPresence Collaboration Server (RMX) 2000:

1. When you receive the RealPresence Collaboration Server (RMX) 2000 packing case, inspect the equipment for damage and verify that the components match the packing slip.
2. Open the top cover of the packing case.
   Boxes are placed on the top Stratocell® and contain the following:
   - **Installation Accessories**. This kit contains the power cables, 1 ethernet cable, USB key and documentation.
   - **Rack Installation Accessories**. This kit contains the accessories for the 19” rack. For more details, see Telescopic Rail Runners Accessory Kit.
   - **Optional - ISDN Package**. Contains the ISDN card and ISDN Software License for ISDN/PSTN.

Write down the Collaboration Server’s serial number that is on a sticker on the back of the unit. It will be needed for product registration later in the process.

Make sure that boxes contain all the required parts.
Unpacking the RealPresence Collaboration Server (RMX) 4000

The following procedure has to be performed to ensure the safe unpacking of the RealPresence Collaboration Server (RMX) 4000 in your site.

To unpack and lift the RealPresence Collaboration Server (RMX) 4000:

1. When you receive the RealPresence Collaboration Server (RMX) 4000 packing case, inspect the equipment for damage and verify that the components match the packing slip.

2. The RealPresence Collaboration Server (RMX) 4000 is shipped in a packing case with Stratocell® packaging, and the top cover must be unlocked and lifted.
   Boxes are placed on the top Stratocell® and contain the following:
   - **Installation Accessories**. This kit contains the power cables, 4 ethernet cables, USB key and documentation.
   - **Rack Installation Accessories**. This kit contains the accessories for the 19” rack. For more details, see Telescopic Rail Runners Accessory Kit.
   - **Optional - ISDN Package**. Contains the ISDN card and ISDN Software License.

   Write down the Collaboration Server's serial number that is on a sticker on the back of the unit. It will be needed for product registration later in the process.

   Make sure that boxes contain all the required parts.

3. Remove the boxes and top Stratocell® and open the anti-static plastic bag wrapping the RealPresence Collaboration Server 1500/1800/2000/4000 Hardware Guide.
Holding the handle on each side, lift the RealPresence Collaboration Server (RMX) 4000 from the box, and place it on a flat surface or in a rack. Remove any packaging material prior to positioning the RealPresence Collaboration Server (RMX) 4000.
Hardware Installation and Setup

In a well-ventilated area, mount the MCU unit in a 19" rack. It is important to adhere to the Site Requirements as described in the RealPresence Collaboration Server 1500/1800/2000/4000 Hardware Guide, Safety Requirements.

The following procedures have to be performed to install the Collaboration Server in your site.

- Installing the Collaboration Server in a rack or as a standalone. When installing the Collaboration Server unit on a rack, this process is done in two stages:
  - Installing the telescopic rail runners on the rack. This stage is identical to all system types.
  - Mounting the Collaboration Server on the rack using the previously installed rail runners
- Connecting the Collaboration Server to the power source
- Connecting the network (LAN and ISDN) cables to the Collaboration Server.

To maximize conferencing performance, especially in high bit rate call environments, a 1Gb connection is recommended for all Collaboration Server types.
Installing the Telescopic Rail Runners on the Rack

Telescopic Rail Runners Accessory Kit

Before installing the telescopic rail runners in the rack, make sure that the kit has the following parts:

<table>
<thead>
<tr>
<th>Rail Runners Kit Contents</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Part/Kit no.</strong></td>
</tr>
<tr>
<td>ASY2716A-L0</td>
</tr>
<tr>
<td>Rail runner</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>Rack spacer assembly kit</td>
</tr>
<tr>
<td>Flat head screw - M5*10mm</td>
</tr>
</tbody>
</table>
### Rail Runners Kit Contents

<table>
<thead>
<tr>
<th>Part/Kit no.</th>
<th>Item</th>
<th>Item no.</th>
<th>Item Sample</th>
<th>Item Quantity</th>
</tr>
</thead>
<tbody>
<tr>
<td>Rail runner assembly kit</td>
<td>Flat head screw - M3*8mm</td>
<td>5</td>
<td><img src="image1" alt="Sample" /></td>
<td>4</td>
</tr>
<tr>
<td>Flat washer M3</td>
<td></td>
<td>6</td>
<td><img src="image2" alt="Sample" /></td>
<td>4</td>
</tr>
<tr>
<td>Nut spring M3</td>
<td></td>
<td>7</td>
<td><img src="image3" alt="Sample" /></td>
<td>4</td>
</tr>
<tr>
<td>Collaboration Server chassis assembly kit</td>
<td>Pan head screw - M5*12mm</td>
<td>8</td>
<td><img src="image4" alt="Sample" /></td>
<td>2</td>
</tr>
<tr>
<td>Flat washer M5</td>
<td></td>
<td>9</td>
<td><img src="image5" alt="Sample" /></td>
<td>2</td>
</tr>
</tbody>
</table>

### Telescopic Rail Runner Assembly

Rack Rail Runners require a minimum of 48cm and a maximum of 80cm within the rack for installation.

1. Determine the location of the Collaboration Server on the rack:
   - Allow for a 1U gap above and below the system for ventilation.
   - Use the Rack Spacer (item no. 3) to predetermine its position on the rack post, making sure that square studs of the spacer fit into the rack post's square/rounded mounting holes. Mark the spacer’s location on the rack post. Repeat this process for the 3 remaining vertical posts ensuring that the system can be horizontally seated.
2 Position the *Rack Spacer (3)* onto the marked rack post together with left rack rail runner (item no. 1 which is labeled LEFT) and fasten the flat head screws 3*10mm (4) as shown in the following figure.
Detail of Front Rack Spacer Assembly for all Collaboration Server types

**RMX 1500 rack assembly view**
- without rail runner clip
- with rail runner clip

**RMX 2000 rack assembly view**
- without rail runner clip
- with rail runner clip
Adjust the telescopic rack rail runner to the rack opening and mount it onto the marked position of the rear post as described in step 2.

- On the Collaboration Server 1500/4000 the center hole on the Rack Spacer must be left clear as it is required for fixing the Collaboration Server to the rack post.
- On the RealPresence Collaboration Server (RMX) 2000 the top hole on the Rack Spacer must be left clear as it is required for fixing the Collaboration Server to the rack post.

3 Adjust the telescopic rack rail runner to the rack opening and mount it onto the marked position of the rear post as described in step 2.

4 Repeat steps 2 and 3 for the right rack rail runner.

5 Install the flat head screw (5), flat washer (6) and nut spring (7) in the middle of the telescopic rack rail runner for added stability.
The number of screws to install depends on the rack width.

6 Repeat step 5 for the right rack rail runner.

Installing the RealPresence Collaboration Server (RMX) 1500

For detailed instructions, precautions and requirements for installing the RealPresence Collaboration Server (RMX) 1500 refer to the Polycom RealPresence Collaboration Server (RMX) 1500 Hardware Guide.

The following procedures have to be performed to install the RealPresence Collaboration Server (RMX) 1500 in your site:

- **Optional.** Installing the RTM ISDN card on the Collaboration Server (Optional)
- Installing the Collaboration Server in a rack or as a standalone
- Connecting the Collaboration Server to the power source
- Connecting the network (LAN, IP and ISDN) cables to the Collaboration Server.

**Optional. Installing the RTM ISDN 1500 Card on the RealPresence Collaboration Server (RMX) 1500**

If the ISDN option was purchased with your Collaboration Server, the ISDN card is shipped separately and must be manually installed into the rear of the RealPresence Collaboration Server (RMX) 1500. It is recommended to install the ISDN card before the RealPresence Collaboration Server (RMX) 1500 is placed in a rack.
Removing the blank cover from the rear of the RealPresence Collaboration Server (RMX) 1500

1. Ensure that the power switch on the Collaboration Server is turned OFF (O).
2. Remove the cover or RTM ISDN 1500 card by unscrewing the captive screws that fasten the card to the MCU.
3. Slide out the cover or RTM ISDN 1500 card.

Installing the RTM ISDN 1500 Card

1. Slide in the RTM ISDN 1500 card.

2. Insert the card into the slot and tighten the captive screws on each side of the rear panel of the card, securing the RTM ISDN card to Collaboration Server.

A Software License is included with the ISDN card. This license must be registered as part of the Product Registration and Product Activation process.

Mounting the RealPresence Collaboration Server (RMX) 1500 in a Rack

There are two methods for installing the Collaboration Server in a 19” rack:

- **Using the rack rail runners on the RealPresence Collaboration Server (RMX) 1500**
  - Install the telescopic rail runners, as described in Installing the Telescopic Rail Runners on the Rack.
  - Mount the RealPresence Collaboration Server (RMX) 1500 on top of the rail runners.
Fasten the Collaboration Server to the rack spacers using the flat head screw (8) with the flat washer (9) through the two holes in the Collaboration Server’s front mounting brackets.

Using a shelf

- Install the shelf, supplied by the rack manufacturer, in the rack.
- Mount the Collaboration Server unit on the shelf.
- Fasten the Collaboration Server unit to the rack with screws through the four holes in the Collaboration Server’s front mounting brackets.

Refer to Detail of Front Rack Spacer Assembly for all Collaboration Server types for installation instructions.

Connecting Cables to the RealPresence Collaboration Server (RMX) 1500

To connect the cables:

- Connect the Media cable to LAN 2 port.
  - Optional. If LAN Redundancy or Multiple Networks options are used, connect the LAN cable to LAN 1. For more information, see the RealPresence Collaboration Server 800s Administrator’s Guide, LAN Redundancy and Multiple Network Services.
● Connect the Network cables to:
  ➢ the MNG (Signaling) port
  ➢ the MNGB (Management Network) port.

● Optional. Connect the Shelf Management cable to the Shelf port.

● Optional. For ISDN/PSTN connections, connect the E1/T1 cables to their PRI (1-4) ports.

The LAN 1*, LAN3, LAN4 and Modem ports are not be used and the plastic caps covering those
ports should not be removed.
* With Multiple Networks and LAN Redundancy configurations, LAN 1 port is used. For more
information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000

Installing the RealPresence Collaboration Server (RMX) 2000

For detailed instructions, precautions and requirements for installing the RealPresence
Collaboration Server (RMX) 2000 refer to the Polycom RealPresence Collaboration Server
(RMX) 2000 Hardware Guide.

The following procedures have to be performed to install the RealPresence Collaboration Server
(RMX) 2000 in your site:

● Optional. Installing the RTM ISDN card on the Collaboration Server (Optional)
● Installing the Collaboration Server in a rack or as a standalone
● Connecting the Collaboration Server to the power source
● Connecting the network (LAN and ISDN) cables to the Collaboration Server
Optional. Installing the RTM ISDN Card on the RealPresence Collaboration Server (RMX) 2000

If the ISDN option was purchased with your Collaboration Server, the ISDN card is shipped separately and must be manually installed into the rear of the RealPresence Collaboration Server (RMX) 2000. It is recommended to install the ISDN card before the RealPresence Collaboration Server (RMX) 2000 is placed in a rack.

Removing the blank cover from the rear of the RealPresence Collaboration Server (RMX) 2000

Use the following procedure to remove the blank cover:

1. Ensure that the power switch/circuit switch on the Collaboration Server is turned OFF (O).
2. Unscrew the captive screws on the rear panel of the Collaboration Server that secure the blank panel.
3. Use the metal ejector levers to pull the blank panel.
Installing the RTM ISDN 2000 Card

1. On the new RTM ISDN card move the ejector levers to their fully open position
2. Slide in the RTM ISDN card in the Collaboration Server slot.

An RTM ISDN card must connect directly to an MPMx/MPMRx card in the opposite facing front slot.

3. Push the card into the slot until the ejector levers touch the front edge of the card cage.
4. Push the ejector levers to their fully closed position.

Tighten the captive screws on each side of the rear panel of the card, securing the RTM ISDN card to the Collaboration Server.

A Software License is included with the ISDN card. This license must be registered as part of the Product Registration and Product Activation process.

Mounting the RealPresence Collaboration Server (RMX) 2000 in a Rack

There are two methods for installing the RealPresence Collaboration Server (RMX) 2000 in a 19” rack:

- **Using rack rail runners on the RealPresence Collaboration Server (RMX) 2000:**
  - Install the telescopic rail runners, as described in Installing the Telescopic Rail Runners on the Rack.
  - Mount the RealPresence Collaboration Server (RMX) 2000 on top of the rail runners.
Fasten the Collaboration Server to the rack spacers using the flat head screw (item 8) with flat washer (item 9) through the two holes in the Collaboration Server’s front mounting brackets.

Using a shelf:
- Install the shelf, supplied by the rack manufacturer, in the rack.
- Mount the Collaboration Server on the shelf.
- Fasten the Collaboration Server to the rack with screws through the four holes in the Collaboration Server’s front mounting brackets.
Connecting Cables to the RealPresence Collaboration Server (RMX) 2000

Do not remove the protective caps from LAN1, LAN3 and ShMG ports.

Connect the following cables to the back panel:

- **Required.** Power cable.
- **Required.** On the RTM IP card connect the LAN cable to **LAN 2** Port.
- **Required with MPMRx Card(s).** An RTM LAN port card must be installed and a LAN cable must be connected to its **LAN 2** Port.

An MPMRx card on the front of the RMX must always be seated or connected opposite to either an RTM LAN or RTM ISDN card on the rear of the chassis.

- For MPMx card(s), any type of RTM LAN type card(s) is optional. However, when using an MPMx card with the RTM LAN - 4 ports, connect the LAN cable to **LAN 4**. With Multiple Networks and LAN Redundancy configurations, **LAN 3** port is used.
- An RTM LAN type card is always required with Multiple Networks and LAN Redundancy configurations.

- With Multiple Networks and LAN Redundancy configurations, **LAN 1** port is used. For more information, see *RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrators Guide Administrator’s Guide, LAN Redundancy and Multiple Network Services.*

  Connect the LAN cable to **LAN 1**.

  - When an **RTM LAN - 2 port** card is installed on the RealPresence Collaboration Server (RMX) 2000, connect the LAN cable to **LAN 2**.

    - On the RMX 2000 this type RTM LAN card(s) is not to be used with the MPMRx.
    - An MPMx card on the front of the RMX must always be seated or connected opposite to either an RTM LAN - 2 ports or RTM ISDN card on the rear of the chassis.
    - An RTM LAN type card is always required with Multiple Networks and LAN Redundancy configurations.

- With Multiple Networks and LAN Redundancy configurations, **LAN 1** port is used. For more information, see *RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrators Guide Administrator’s Guide, LAN Redundancy and Multiple Network Services.*

  Connect the LAN cable to **LAN 1**.

  - Optional. If an RTM ISDN card is installed on the RealPresence Collaboration Server (RMX) 2000, connect the E1/T1 Cables to **PRI** Ports.
Do not remove the protective caps from LAN1*, LAN 3 and ShMG ports on the RTM IP card.

Installing the RealPresence Collaboration Server (RMX) 4000

The following procedures have to be performed to install the RealPresence Collaboration Server (RMX) 4000 at your site:

- **Optional.** Installing the RTM ISDN card on the Collaboration Server
- Mounting the Collaboration Server in a rack
- Connecting the Collaboration Server to the power source
- Connecting the network (LAN and ISDN) cables to the Collaboration Server

Installing the RTM ISDN Card on the RealPresence Collaboration Server (RMX) 4000

If the ISDN option was purchased with your Collaboration Server, the ISDN card is shipped separately and must be manually installed into the rear of the RealPresence Collaboration Server (RMX) 2000. It is recommended to install the ISDN card before the RealPresence Collaboration Server (RMX) 2000 is placed in a rack.

Removing the RTM LAN Card or the blank cover from the rear of the RealPresence Collaboration Server (RMX) 4000

1. Ensure that the power switch on the Collaboration Server is turned OFF (O).
2. Remove the RTM LAN or blank cover by unscrewing the captive screws that fasten the card or the cover to the Collaboration Server. When removing a card, use the metal ejector levers to pull the RTM LAN card out of its slot from the backplane.
3. Slide out the RTM LAN or RTM ISDN card.
Installing the RTM ISDN 4000 Card

1. On the RTM ISDN card move the ejector levers to their fully open position.

2. Push the card into the slot until the ejector levers touch the front edge of the card cage. Push the ejector levers to their fully closed position.

3. Tighten the captive screws on each side of the rear panel of the card, securing the RTM ISDN card to the MCU.

A Software License is included with the ISDN card. This license must be registered as part of the Product Registration and Product Activation process.

Mounting the RealPresence Collaboration Server (RMX) 4000 in a Rack

Either place the RealPresence Collaboration Server (RMX) 4000 on a hard, flat surface such as a desktop or mount it on a 19” rack

To install the RealPresence Collaboration Server (RMX) 4000 in a 19” rack:

- Using rack rail runners on the RealPresence Collaboration Server (RMX) 4000
  - Install the telescopic rail runners, as described in Installing the Telescopic Rail Runners on the Rack.
  - Mount the RealPresence Collaboration Server (RMX) 4000 on top of the rail runners.
Fasten the Collaboration Server to the rack spacers using the flat head screw (item 8) with flat washer (item 9) through the two holes in the Collaboration Server’s front mounting brackets.

- **Using a shelf**
  - Install the shelf, supplied by the rack manufacturer, in the rack.
  - Mount the Collaboration Server on the shelf.
  - Fasten the Collaboration Server to the rack with screws through the eight holes in the Collaboration Server’s front mounting brackets.
Connecting the RealPresence Collaboration Server (RMX) 4000 to the Power Sources

The size of the protective earthing conductor & cable should be a minimum of 10AWG.

Connect the following power cables to the RealPresence Collaboration Server (RMX) 4000 back panel:

**AC Power Supply connections:**

1. Insert power cables to each of the three AC Power Entry Modules (PEMs)

   ![Diagram of AC Power Supply connections](image)

**DC Power Supply connections:**

1. On the DC Power Rail Modules set the two circuit breakers to OFF

   Two types of circuit breakers can be installed on the DC Power Rail Module (PRM). For more information, see the *RealPresence Collaboration Server (RMX) 4000 Hardware Guide*.

2. Ensure that the cables from the Main that supplies electricity to the DC power units are OFF or disconnected.

3. Remove the transparent plastic caps on the terminal block.
4 Using the two wires of a 10 AWG cable running from the DC power distribution unit, connect the black wire into the -48VDC terminal block and the red wire to the RTN terminal block.

5 Connect the green or green-yellow wire to the system single-point M6x15 “Ground” bolt.

The rating of the protective earthing conductor should be a minimum of 10AWG.

If the unit is rack mounted, the single-point ground on the MCU must be connected to the rack with a single conductor and fixed as to prevent loosening. When using bare conductors, they must be coated with an appropriate antioxidant compound before crimp connections are made. Tinned, solder-plated or silver plated connectors do not have to be prepared in this manner.

6 Replace the transparent plastic caps on the terminal block.
7 Turn ON the Main that supplies power to the Collaboration Server.
8 Turn ON the circuit breaker on each of the DC Power Rail Modules.

Connecting Cables to the RealPresence Collaboration Server (RMX) 4000

This section describes the cables connected on the RMX 4000 AC and DC systems.

To connect the cables (AC and DC systems):

- **RTM-IP 4000**:
  - Connect the Management Network cable to **LAN 2**.
  - Connect the Signaling cable to **LAN 3**. This port is also available for Signaling and Management Redundancy. For more information, see Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, **LAN Redundancy and Multiple Network Services**.
  - Connect the Shelf Management cable to **LAN 6**

  When an NTP Server is used for the **RMX Time**, the Shelf Management cable must be connected to the shelf port.

- **RTM LAN** type cards:
  - When an **RTM LAN - 4** port card is installed on the RealPresence Collaboration Server (RMX) 4000, connect the LAN cable to **LAN 2**:
    - An MPMRx card on the front of the RMX must always be seated or connected opposite to either an RTM LAN - 4 ports or RTM ISDN card on the rear of the chassis.
    - " When using an MPMx card with the RTM LAN - 4 ports, connect the LAN cable to **LAN 4**. With Multiple Networks and LAN Redundancy configurations, **LAN 3** port is used.
    - The RTM LAN type card is always required with Multiple Networks and LAN Redundancy configurations.

  - With Multiple Networks and LAN Redundancy configurations, **LAN 1** port is used. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, **LAN Redundancy and Multiple Network Services**.
    
    Connect the **LAN cable to LAN 1**.

  - When an **RTM LAN - 2 port** card is installed on the RealPresence Collaboration Server (RMX) 4000, connect the LAN cable to **LAN 2**.
    - An MPMx card on the front of the RMX must always be seated or connected opposite to either an RTM LAN - 2 ports or RTM ISDN card on the rear of the chassis.
    - The RTM LAN type card is always required with Multiple Networks and LAN Redundancy configurations.
With Multiple Networks and LAN Redundancy configurations, **LAN 1** port is used. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, LAN Redundancy and Multiple Network Services.

Connect the **LAN** cable to **LAN 1**.

- **Optional.** If RTM ISDN is installed, for each installed RTM ISDN:
  - Connect the E1/T1 cables to their PRI Ports.
  - Connect the LAN cable to **LAN 1**.

When LAN redundancy is enabled, LAN 1 is used for both management, media and signaling network connection. Connect the media and signaling cable to LAN 2 port. By default this port is used for signaling, but when LAN redundancy is enabled, LAN 2 is the backup of LAN 1 port.

Attention: Two people are required to lift the MCU out of the box and when installing it in a rack.

Write down the serial number of the Collaboration Server (RMX), located on a sticker on the back of the unit. It will be needed for product registration later in the process.
Installing the RealPresence Collaboration Server 1800

The following procedures have to be performed to install the RealPresence Collaboration Server 1800 in your site:

- Unpacking the Collaboration Server
- Installing the Collaboration Server in a rack or as a standalone system
- Connecting the Collaboration Server to the power source
- Connecting the network (LAN and IP) cables to the Collaboration Server
- Modifying the Factory Default Management Network Settings on the USB Memory Stick

Unpacking the RealPresence Collaboration Server 1800

To unpack and lift the RealPresence Collaboration Server 1800:

1. When you receive the RealPresence Collaboration Server 1800 packing case, inspect the equipment for damage and verify that the components match the packing slip.
2. Open the top cover of the packing case.
   - Boxes are placed on the top Stratocell® and contain power cables, 2 ethernet cables, USB key and documentation.
   - Write down the Collaboration Server’s serial number that is on a sticker on the back of the unit. It will be needed for product registration later in the process.

Make sure that boxes contain all the required parts.

Mounting the RealPresence Collaboration Server 1800 on a Shelf

- Install the shelf supplied by the rack manufacturer, in the rack.
- Mount the Collaboration Server on the shelf.
- Fasten the Collaboration Server to the rack with screws through the four holes in the Collaboration Server’s front mounting brackets.
Connecting the RealPresence Collaboration Server 1800 to a Power Source

The following restrictions apply to the conductors and connectors that may be used to ground the unit when rack mounted:

- When using bare conductors, they must be coated with an appropriate antioxidant compound before crimp connections are made. Tinned, solder-plated or silver-plated connectors do not have to be prepared in this manner.
- The same bolt assemblies should not secure multiple connectors.
- Listed fastening hardware must be compatible with the materials being joined and must be preclude loosening, deterioration and electrochemical corrosion of the hardware and joint materials.

Connecting the RealPresence Collaboration Server 1800 to AC Power

- Only the AC Power cable supplied by Polycom should be used.
- The size of the protective earthing conductor should be a minimum of 10 AWG.
- The outlet intended for connecting the power cord must be protected with an external overcurrent protection device either in building or in the rack with the rating not higher than 20 AMP.
- Do not use an extension cord with the cable.

» Insert the power cable into the power connector on the rear panel of the RealPresence Collaboration Server 1800 system.

Connecting Cables to the RealPresence Collaboration Server 1800

This section outlines the way to connect the cables to the Collaboration Server 1800.

To connect the cables:

1. Connect the management network cable to LAN 1 port.
   When LAN redundancy is enabled, LAN 1 is used for both management, media and signaling network connection.

2. Connect the media and signaling cable to LAN 2 port.
   By default this port is used for signaling, but when LAN redundancy is enabled, LAN 2 is the backup of LAN 1 port.
Modifying the Factory Default Management Network Settings on the USB Memory Stick

The *USB memory stick* contains a text file, *lan.cfg*, which holds the factory default IP address parameters. These parameters must be modified to your local network settings using the *LAN Configuration Utility*, but are also available on the USB memory stick.

On the RMX1800 must use the LAN configuration Utility on a USB memory stick to change the IP address.

To modify the USB memory stick settings:

1. Take the *USB memory stick* from the *Installation Accessories* kit and insert it into the PC workstation.
   In Windows XP:
   ➢ The *Polycom Documentation* option is automatically selected. Click **OK**.
   In Windows 7:
   ➢ Select **Open Folder to view files using Windows Explorer**.

2. Double-click the *index.hta* file.
   The *Language Menu* opens, offering a choice of several languages.
   In Windows XP:
   ➢ The *Polycom Documentation* option is automatically selected. Click **OK**.
   In Windows 7:
3 Click the documentation language of your choice. 
   An End-User Licence Agreement for Polycom Software is displayed.
4 Read the agreement and click the Accept Agreement button.
5 In the Product Type Selection window, click the RealPresence Collaboration Server hyperlink.

![Product Type Selection Window]

6 In the Initial Setup Utility screen, click the LAN Configuration Utility link.

![Initial Setup Utility Screen]

The LAN Configuration Utility dialog box opens.
7 Modify the following parameters in the utility’s dialog box using the information supplied by your network administrator.

- Control Unit IP Address (the Management IP address for the MCU)
- Shelf Management IP Address
- Subnet Mask
- Default Router IP Address

8 Click **OK**.

9 Remove the **USB memory stick** from the PC workstation.

The **USB memory stick** is required for **First Entry Power-up and Configuration** of the MCU.
First Entry Power-up and Configuration

There are four procedures necessary for setup of the new RealPresence Collaboration Server 1500/1800/2000/4000 Hardware Guide. It is important that they are performed in the following sequence:

1. First-time Power-up.
2. Product Registration.
3. Connection to MCU.
4. Modifying the Default IP and ISDN/PSTN Service Settings (*Fast Configuration Wizard*).

Procedure 1: First-time Power-up

To power-up for the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 first time

For first entry installation, you must insert the *USB key* containing the modified IP addresses in *USB slot* on the Collaboration Server’s back panel.

1. Turn on the power by pressing on the ON/OFF button located on the front panel of the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000. You are required to register all *Polycom Software Licenses* that you have purchased when retrieving the *Activation Key*. For example, Encryption and Network Separation each have different *Polycom Software Licenses*.

   The parameters in the lan.cfg file are uploaded from the USB key to the Collaboration Server’s memory and applied during the power-up sequence.

   System power-up sequence may take up to five minutes.

   When the Collaboration Server's configuration is completed (including the *Management* and *IP Network Services*), and if there are no *System Errors*, the STATUS LED (on the Collaboration Server’s front panel) turns green.

2. Remove the *USB key*.
To power-up for the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 first time using the USB memory stick:

1. Insert the *USB memory stick* containing the modified IP addresses in USB port on the RealPresence Collaboration Server (RMX) 1500 front panel and the RealPresence Collaboration Server (RMX) 2000/4000 back panel.

2. Power the Collaboration Server **ON**.
   - **AC System** - Turn ON the power by pressing on the power switch located on the rear panel of the RealPresence Collaboration Server (RMX) 1500/2000/4000.
     - On the RealPresence Collaboration Server (RMX) 1500, the ON/OFF button is lit (ON).
     - On the RealPresence Collaboration Server (RMX) 2000/4000, the FAN STATUS and PWR STATUS LEDs turn ON.
   - **DC System (RealPresence Collaboration Server (RMX) 4000)** - Turn ON the Main that supplies power to the Collaboration Server and then turn ON each of the DC power rail modules.

3. Press the power button on the system. The power indicators should light.
System power-up sequence may take approximately 105 minutes. During this time, the parameters in the lan.cfg file are uploaded from the USB memory stick to the MCU’s memory and applied during the power-up sequence.

Wait for the upload process to complete. Initially, all the READY/IN USE/ERROR LEDs on the RealPresence Collaboration Server (RMX) 1500 or ERR/RDY/ACT LEDs on the RealPresence Collaboration Server (RMX) 2000/4000 flicker and flash. Upload is completed when all the LEDs turn off and only the red ERROR/ERR LED (on the CTNL unit of the RealPresence Collaboration Server (RMX) 2000/4000) remains ON. It remains ON until the Default IP Network Service is configured. For more details, see Fast Configuration Wizard.

Do not remove the USB memory stick from the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 until the connection with the MCU is established and the Login screen of the Collaboration Server Web Client is displayed. For more details, see Procedure 3: Connection to MCU.

Procedure 2: Product Registration

Before the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 can be used, it is necessary to register the product and the various software licenses and obtain an Activation Key.

During first-time power-up, the Product Activation dialog box is displayed in the Collaboration Server Web Client, requesting you to enter an Activation Key.

From version 8.1 onwards, a license is required for SVC conferencing.

Obtaining the Activation Key

1. Access the Service & Support page of the Polycom website at http://support.polycom.com
2. Log in with your Email Address and Password or register as a new user.
3. Select Product Registration.
4. Follow the on-screen instructions for Product Registration and Product Activation.
   (The MCU’s serial number on the product sticker on the back of the unit.) For more information, refer to the Collaboration Server Software Licence document you received with your shipment.
Register all *Polycom Software Licences* that you have purchased when obtaining the *Activation Key*. For example, ISDN, Encryption and Multiple Networks each have different *Polycom Software Licenses*.

5 When the *Product Activation Key* is displayed, write it down or **copy** it for later pasting into the *Activation Key* field of the *Product Activation* dialog box.

**Procedure 3: Connection to MCU**

If Windows™ is installed on the workstation, **Protected Mode** must be disabled before connecting to the MCU running Version 7.0 software. For more information, see *Windows 7™ Security Settings*.

1 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>`, as defined in the USB memory stick.
   
   b Click **Enter**.

   The *Collaboration Server Web Client* Login screen is displayed.
2 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:

3 Select the "**I accept the license agreement**" check box to enable **Online Registration** of your product.

4 In the **Activation Key** field, enter or **paste** the **Product Activation Key** obtained earlier.

If you did not register your product earlier in the process and you do not have an **Activation Key**, click the **Polycom Resource Center** button to access the **Service & Support** page of the Polycom Support website.

5 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.

6 Click **OK**.
On the MPMx/MPMRx card of the RealPresence Collaboration Server (RMX) 2000/4000, initially, the ERR/RDY/ACT LEDs all flicker and flash, until only the RDY LED turns ON on the media card, while the ERR LED on the CTNL unit is still ON.

As no Default IP Network Service is defined, the system automatically starts the Fast Configuration Wizard.

**Procedure 4: Modifying the Default IP Service and ISDN/PSTN Network Service Settings**

The Fast Configuration Wizard is automatically started when no Default IP Network Service is defined and it assists you in configuring the Default IP Network Service. This happens during First Time Power-up, before the service has been defined or if the Signaling 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 IPv6 addressing is required, complete the Fast Configuration Wizard for IPv4 and then:

1. Modify the Management Network to use IPv6 addressing or IPv4 and IPv6 addressing.
2. Restart the Collaboration Server.
3. Modify the properties of the Default IP Network Service, which will now include IPv6 addressing or IPv4 & IPv6 addressing options to configure the Network Service.


**Fast Configuration Wizard**

1. Enter the required IP information in the dialog box.

In certain cases the actual RMX 1800 dialog boxes are not shown as the user interface (UI) is nearly identical to the RealPresence Collaboration Server (RMX) 1500/2000/4000. Tabs may be missing or in certain cases not shown, as with the PRI Settings, ISDN tabs which are not supported on the 1800.
Fast Configuration Wizard – IP Signaling

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Service Name</td>
<td>The name Default IP Service is assigned to the IP Network Service by the Fast Configuration Wizard. This name can be changed. <strong>Note:</strong> This field is displayed in all IP Signaling dialog boxes and can contain character sets that use Unicode encoding.</td>
</tr>
<tr>
<td>Signaling Host IP Address</td>
<td>Enter the address to be used by IP endpoints when dialing into the MCU. Dial out calls from the Collaboration Server are initiated from this address. This address is used to register the Collaboration Server with a Gatekeeper or a SIP Proxy server. In the RealPresence Collaboration Server 1800, this field is disabled as only one IP address is used for the Signaling Host and the Media Card and it is defined by the Media Card IP address.</td>
</tr>
<tr>
<td>Media Card 1-4 IP Addresses</td>
<td>Enter the IP address(es) of the media card(s) (MPMx/MPMRx 1 and MPMx/MPMRx 2-4 (if installed)) as provided by the network administrator. Endpoints connect to conferences and transmit call media (video, voice and content) via these addresses. In the RealPresence Collaboration Server 1800, one IP address is used for the signaling and media. Enter the address to be used by IP endpoints when dialing into the MCU. Dial out calls from the Collaboration Server are initiated from this address. This address is used to register the Collaboration Server with a Gatekeeper or a SIP Proxy server.</td>
</tr>
<tr>
<td>Subnet Mask</td>
<td>Enter the subnet mask of the MCU. Default value: 255.255.255.0.</td>
</tr>
</tbody>
</table>

- To set the Collaboration Server to Secured Communication, first complete the Fast Configuration Wizard and reset the Collaboration Server. After Login, install the Certificate and then enable Secured Communication Mode.
- The IP Network Service configured using the Fast Configuration Wizard will be saved only if Media cards are installed in the Collaboration Server.

2 Click **Next**.
3. Enter the required **Routers** information in the dialog box.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default Router IP Address</td>
<td>Enter the IP address of the default router.</td>
</tr>
</tbody>
</table>

4. Click **Next**.
5 Enter the required **DNS** information in the dialog box.

![Fast Configuration Wizard – DNS](image)

**Fast Configuration Wizard – DNS**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>MCU Host Name</td>
<td>Enter the name of the MCU on the network. Default name is Polycom MCU.</td>
</tr>
<tr>
<td>DNS</td>
<td>Select:</td>
</tr>
<tr>
<td></td>
<td>• <strong>Off</strong> – if DNS servers are not used in the network.</td>
</tr>
<tr>
<td></td>
<td>• <strong>Specify</strong> – to enter the IP addresses of the DNS servers.</td>
</tr>
<tr>
<td></td>
<td><strong>Note:</strong> The IP address fields are enabled only if <strong>Specify</strong> is selected.</td>
</tr>
<tr>
<td>Register Host Names Automatically to DNS Server</td>
<td>Select this option to automatically register the MCU Signaling Host and Shelf Management with the DNS server.</td>
</tr>
<tr>
<td>Local Domain Name</td>
<td>Enter the name of the domain where the MCU is installed.</td>
</tr>
<tr>
<td>Primary DNS Server IP Address</td>
<td>The static IP address of the primary DNS server.</td>
</tr>
</tbody>
</table>

6 **Click Next.**

7 Select the **IP Network Type:** **H.323, SIP** or **H.323 & SIP.**
The Collaboration Server supports SVC-based conferencing, which is based on the SIP protocol. If SVC-based conferencing is required in your organization, select SIP as one of your Network Type options.

8 Click **Next**.

9 If you selected **SIP only**, go to **Step 13**.

10 Enter the required **Gatekeeper** information in the dialog box.
11 Click Next.

12 If you selected H.323, click Next and go to Step 14.

13 Enter the required SIP Server information in the dialog box.
### Fast Configuration Wizard – SIP Server

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Server</td>
<td>Select:</td>
</tr>
<tr>
<td></td>
<td>• Specify – to manually configure SIP servers.</td>
</tr>
<tr>
<td></td>
<td>• Off – if SIP servers are not present in the network.</td>
</tr>
<tr>
<td>SIP Server IP Address or Name</td>
<td>Enter either the IP address of the preferred SIP server or its host name (if a DNS server is used).</td>
</tr>
<tr>
<td>Server Domain Name</td>
<td>Enter the name of the <strong>SIP domain</strong>.</td>
</tr>
<tr>
<td>Transport Type</td>
<td>Select the transport type and protocol that is used for signaling between the MCU and the SIP Server or the endpoints according to the protocol supported by the SIP Server:</td>
</tr>
<tr>
<td></td>
<td>• <strong>UDP</strong> – Select this option to use UDP for signaling.</td>
</tr>
<tr>
<td></td>
<td>• <strong>TCP</strong> – Select this option to use TCP for signaling.</td>
</tr>
<tr>
<td></td>
<td>• <strong>TLS</strong> – The <strong>Signaling Host</strong> 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.</td>
</tr>
<tr>
<td></td>
<td>The following protocols are supported:</td>
</tr>
<tr>
<td></td>
<td>• TLS 1.0</td>
</tr>
<tr>
<td></td>
<td>• SSL 2.0</td>
</tr>
<tr>
<td></td>
<td>• SSL 3.0.</td>
</tr>
</tbody>
</table>

**14 Click Next.**
15 Enter the required **Security** information in the dialog box.

### Default IP Network Service – Security

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Authentication</td>
<td>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 Collaboration Server, calls will fail to connect to the conferences. Leave this check box cleared if the authentication option is disabled on the SIP proxy.</td>
</tr>
<tr>
<td>User Name</td>
<td>Enter the user name the Collaboration Server will use to authenticate itself with the SIP proxy. This name must be defined in the gatekeeper. These fields can contain up to 20 ASCII characters.</td>
</tr>
<tr>
<td>Password</td>
<td>Enter the password the Collaboration Server will use to authenticate itself with the gatekeeper. This password must be defined in the SIP proxy.</td>
</tr>
</tbody>
</table>
Default IP Network Service – Security

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.323 Authentication</td>
<td>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 Collaboration Server, calls will fail to connect to the conferences. Leave this check box cleared if the authentication option is disabled on the gatekeeper.</td>
</tr>
<tr>
<td>User Name</td>
<td>Enter the user name the Collaboration Server will use to authenticate itself with the gatekeeper. This name must be defined in the gatekeeper.</td>
</tr>
<tr>
<td>Password</td>
<td>Enter the password the Collaboration Server will use to authenticate itself with the gatekeeper. This password must be defined in the gatekeeper.</td>
</tr>
</tbody>
</table>

16 Click **Save & Continue**.

The IP Network Service is created and confirmed.

![IP Network service created.]

17 Click **OK**.

The IP Network Service cannot be saved if no Media cards are installed in the Collaboration Server.

During the initial Collaboration Server setup, if the system detects the presence of the RTM ISDN card, the ISDN/PSTN Network Service definition screens of the Fast Configuration Wizard are enabled.

If there is no RTM ISDN card in the Collaboration Server or if you do not want to define an ISDN/PSTN Network Service, go to Step 33.

A new ISDN/PSTN Network Service can be defined even when there is no RTM ISDN card is installed in the system using the ISDN/PSTN Network Service -> Add New Service dialog box.
The Fast Configuration Wizard’s ISDN/PSTN configuration sequence begins with the ISDN/PSTN dialog box.

![Fast Configuration Wizard](image)

18 Define the following parameters

### Fast Configuration Wizard – ISDN Service Settings

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Service Name</td>
<td>Specify the service provider’s (carrier) name or any other name you choose, using up to 20 characters. The Network Service Name identifies the ISDN/PSTN Service to the system. Default name: ISDN/PSTN Service</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong>: This field is displayed in all ISDN/PSTN Network Properties tabs and can contain character sets that use Unicode encoding.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Span Type</th>
<th>Select the type of spans (ISDN/PSTN) lines, supplied by the service provider, that are connected to the Collaboration Server. Each span can be defined as a separate Network Service, or all the spans from the same carrier can be defined as part of the same Network Service. Select either:</th>
</tr>
</thead>
<tbody>
<tr>
<td>T1</td>
<td>(U.S. – 23 B channels + 1 D channel)</td>
</tr>
<tr>
<td>E1</td>
<td>(Europe – 30 B channels + 1 D channel)</td>
</tr>
<tr>
<td></td>
<td>Default: T1</td>
</tr>
</tbody>
</table>

**Note**: Only one Span Type (E1 or T1) is supported on the Collaboration Server. If you define the first span as type E1 all other spans that you may later define must also be of type E1.
The PRI Settings dialog box opens.

19 Click Next.

20 Define the following parameters:

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Default Num Type | Select the Default Num Type from the list. The Num Type defines how the system handles the dialing digits. For example, if you type eight dialing digits, the Num Type defines whether this number is national or international. If the PRI lines are connected to the Collaboration Server via a network switch, the selection of the Num Type is used to route the call to a specific PRI line. If you want the network to interpret the dialing digits for routing the call, select Unknown. Default: Unknown  
**Note:** For E1 spans, this parameter is set by the system. |
| Num Plan        | Select the type of signaling (Number Plan) from the list according to information given by the service provider. Default: ISDN  
**Note:** For E1 spans, this parameter is set by the system. |
21 Click Next.

The Span Definition dialog box opens.

22 Define the following parameters:

**Fast Configuration Wizard – Spans Definition**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Framing</td>
<td>Select the Framing format used by the carrier for the network interface from the list.</td>
</tr>
<tr>
<td></td>
<td>• For T1 spans, default is SFSF.</td>
</tr>
<tr>
<td></td>
<td>• For E1 spans, default is FEBE.</td>
</tr>
</tbody>
</table>
The Phones dialog box opens.

23 Click Next.

The Phones dialog box opens.

24 Click Add to define dial-in number ranges.

The Add Phone Number dialog box opens.
25 Define the following parameters:

**Fast Configuration Wizard – Add Phone Numbers**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>First Number</td>
<td>The first number in the phone number range.</td>
</tr>
<tr>
<td>Last Number</td>
<td>The last number in the phone number range.</td>
</tr>
</tbody>
</table>

- A range must include at least two dial-in numbers.
- A range cannot exceed 1000 numbers.

26 Click **OK**.

The new range is added to the *Dial-in Phone Numbers* table.

27 **Optional**. Repeat steps 24 to 25 to define additional dial-in ranges.

28 In the *Phones* tab enter the **MCU CLI** (Calling Line Identification).

With dial-in connections, the **MCU CLI** indicates the MCU’s number dialed by the participant. In a dial-out connection, indicates the MCU (CLI) number as seen by the participant.

29 Click **Save & Continue**.

After clicking **Save & Continue**, you cannot use the **Back** button to return to previous configuration dialog boxes.

The **ISDN/PSTN Network Service** is created and is added to the ISDN/PSTN Network Services list.

If the system cannot create the **ISDN/PSTN Network Service**, an error message is displayed indicating the cause and allowing you access the appropriate dialog box in the **Fast Configuration Wizard** for corrective action.

30 Click **OK** to continue the configuration.

The **Spans** dialog box opens displaying the following read-only fields:
ID – the connector on the RTM ISDN card (PRI1 to PRI12).

Slot – the MPMx/MPMRx card that the RTM ISDN / RTM ISDN 1500 card is connected to (RealPresence Collaboration Server (RMX) 2000: MPM 1/MPM2 RealPresence Collaboration Server (RMX) 4000: MPM1/MPM2/MPM3/MPM4).

On the RealPresence Collaboration Server (RMX) 1500, the Slot field does not appear.

Service – the ISDN/PSTN Network Service to which the span is assigned.

Clock Source – indicates if ISDN signaling synchronization is being supplied by the Primary or Secondary clock source.

State – the System Alert level of the span (Major, Minor). If there are no span related alerts, this column contains no entries.

Click the check boxes in the Attached field to attach spans (E1 or T1 PRI lines) to the network service named in the Network Service Name field.

The Spans Table displays the configuration of all spans and all ISDN network services in the system.

When using the Fast Configuration Wizard during First Entry Configuration, you are defining the first ISDN/PSTN Network Service in the system. Spans can only be attached to this service.

Additional ISDN/PSTN Network Services can be defined by using the ISDN/PSTN Network Services > New PSTN Service button in the Collaboration ServerWeb Client.

Spans can be attached to, or moved between ISDN network services by using the ISDN/PSTN Network Services > ISDN Properties > Spans tab in the Collaboration Server Web Client.

- **RMX 2000/4000** - Each ISDN RTM card can support either 7 E1 or 9 T1 PRI lines.
- **RMX 1500** - Either 4 E1 or 4 T1 PRI lines are supported.

E1 and T1 connections cannot be used simultaneously.
32 Click Next.
The RMX Time dialog box is displayed.

![RMX Time dialog box]

33 Set the RMX Time using one of the three available options: setting the RMX Time manually, clicking the Retrieve Client Time button, or using the NTP Servers options.

**Fast Configuration Wizard - Collaboration Server Time**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>GMT Date</td>
<td>The date at Greenwich, UK.</td>
</tr>
<tr>
<td>Local Time</td>
<td>The MCU's local time settings, are calculated from the GMT Time and the GMT Offset.</td>
</tr>
</tbody>
</table>
| GMT Time         | Displays the MCU's current GMT Time settings. **Option 1**: Manually setting the Collaboration Server time:  
|                  | • Using the Up or Down arrows alter the GMT Time and the GMT Offset to set the Collaboration Server time. |
| GMT Offset       | The time zone difference between Greenwich and the MCU's physical location.  
|                  | • Using the Up or Down arrows manually modify the GMT Offset time on the Collaboration Server. |
| Retrieve Client Time | **Option 2**: Automatically setting the MCU time:  
|                  | • Click this button to automatically update the MCU's GMT Date, Time and Offset to match that of the workstation. |
The Administrator User dialog box is displayed.

34 Click Next.

35 Enter the required Administrator User information:

Fast Configuration Wizard - Administrator User

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>New User Name</td>
<td>Enter the new user name of the new administrator user.</td>
</tr>
<tr>
<td>New Password</td>
<td>Enter the password for the new administrator user.</td>
</tr>
<tr>
<td>Confirm Password</td>
<td>Enter the same password again to confirm its accuracy.</td>
</tr>
</tbody>
</table>

36 Click Next.
The **System Flags** dialog box is displayed.
37 Enter the required **System Flags** information in the dialog box.

If you need to modify these settings later, select **System Configuration** from the **Setup** menu. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/2000/4000 Administrator’s Guide, Modifying System Flags.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description / Default</th>
<th>Note:</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference ID Length (MCU)</td>
<td>The number of digits of the Conference ID to be assigned by the MCU. Range: 2-16 (Default: 5)</td>
<td></td>
</tr>
<tr>
<td>Minimum Conference ID Length (User)</td>
<td>The minimum number of digits that the user must enter when manually assigning a numeric ID to a conference. Range: 2-16 (Default: 4)</td>
<td></td>
</tr>
<tr>
<td>Maximum Conference ID Length (User)</td>
<td>The maximum number of digits that the user can enter when manually assigning a Numeric ID to a conference. Range: 2-16 (Default: 8)</td>
<td></td>
</tr>
<tr>
<td>MCU Display Name</td>
<td>The MCU name is displayed on the endpoint's screen. Default name: <strong>RMX 1500/2000/4000</strong></td>
<td></td>
</tr>
<tr>
<td>Terminate Conference when Chairperson Exits</td>
<td>When <strong>Yes</strong> is selected (default), the conference ends when the chairperson exits even if there are other participants connected. When <strong>No</strong> is selected, the conference automatically ends at the predefined end time, or when all the participants have disconnected from the conference.</td>
<td></td>
</tr>
<tr>
<td>Auto Extend Conferences</td>
<td>When <strong>Yes</strong> is selected (default), allows conferences running on the Collaboration Server to be automatically extended as long as there are participants connected and there are available resources. The maximum extension time allowed by the MCU is 30 minutes.</td>
<td></td>
</tr>
</tbody>
</table>

38 Click **Save & Close**. The system confirms successful configuration.

39 In the **Success Message** box, click **OK**.

40 In the **Reset Confirmation** dialog box, click Yes.

41 In the **Please wait for system reset** message box, click **OK**. System restart may take up to five minutes.
42 Refresh the browser periodically until the Login screen is displayed.

43 When the Login screen is displayed, enter your Username and Password and click Login. In the Collaboration Server Web Client’s Main Screen an MCU State indicator displays a progress indicator showing the time remaining until the system start-up is complete.

If the default User (POLYCOM) remains or the Collaboration Server time was not set, the active alarm is not deleted and the system status remains in Major. For system security reasons the system is not fully configured until the default user is deleted.

44 The system is now fully configured and if there are no other System Errors, the green READY/RDY LED (on the 2000/4000 the LEDs are on the CNTL module) turns ON and the ERROR/ERR LED is OFF.

The Fast Configuration Wizard configures the Default IP Network Service with common parameters. Specific or additional settings (e.g. for ICE, or Secured Mode) should be performed once the initial configuration is complete. For detailed description of the IP Network Services, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, IP Network Services.

Selecting the Collaboration Server Web Client Languages

By default, the Collaboration Server Web Client interface is displayed only in English. However, the system administrator can choose the languages available for selection on the Login screen. These languages are represented by flags.

To choose the languages for selection in the Login screen:


If the selected language is not supported by the browser or the workstation’s Operating System, the Collaboration Server Web Client is displayed in English.

3. Click OK.

4. Log out and reconnect to the Collaboration Server. The Login screen will display the flags of the selected languages.

RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Default Conferencing Settings

The Collaboration Server is shipped with default pre-configured conferencing entities set to CP (AVC) Conferencing Mode, which allow the MCU users and participants to start CP AVC ongoing conferences without further configuration.
The default conferencing entities are:

Default AVC-based Conferencing Entities

<table>
<thead>
<tr>
<th>Entity</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Meeting Rooms</td>
<td>Conferences saved on the MCU without using resources. They are activated when the first participant dials in. There are four Meeting Rooms ready for use: NameID Maple_Room 1001 Oak_Room 1002 Juniper_Room 1003 Fig_Room 1004 Each Meeting Room uses the default Conference Profile called Factory_Video_Profile set to CP (AVC) Only Conferencing Mode, running at 384Kbps and has a default duration of one hour.</td>
</tr>
<tr>
<td>Conference Profile</td>
<td>Name: Factory_Video_Profile A CP (AVC) Only Conference Profile is assigned to a Meeting Room to define its Conferencing Mode, conferencing parameters, such as line rate and video resolution. Factory_Video_Profile contains AVC video parameters with a bit rate of 384Kbps, Auto Layout and Polycom Skin. The Profile uses an IVR Service called Conference IVR Service. In addition to the default Conference Profile, the system is shipped with the following default Conference Profiles: • Factory_SVC_Video_Profile - Contains the parameters of an SVC-based conference Factory_Video_Profile - Contains the parameters of an AVC-CP only conference</td>
</tr>
<tr>
<td>Conference IVR Service</td>
<td>Name: Conference IVR Service The Conference IVR Service includes an optional video slide and all the voice messages played during the participant's connection process and during the conference. The Conference IVR Service contains a set of voice prompts in English and an optional video slide. It automates the participant's connection to a conference.</td>
</tr>
</tbody>
</table>
### Default AVC-based Conferencing Entities

<table>
<thead>
<tr>
<th>Entity (AVC endpoints)</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Entry Queue</td>
<td>NameID</td>
</tr>
<tr>
<td></td>
<td>DefaultEQ 1000</td>
</tr>
<tr>
<td></td>
<td>Using an Entry Queue enables one dial-in number to be used for all AVC-based connections. In the Entry Queue, AVC participants are prompted for information to enable routing to their destination conferences.</td>
</tr>
<tr>
<td></td>
<td>A default Entry Queue called DefaultEQ is provided.</td>
</tr>
<tr>
<td></td>
<td>The default Entry Queue is also set to Ad Hoc conferencing which allows participants to start new conferences without prior definition by entering a Conference or Meeting Room ID that is not used by any ongoing conference currently running on the MCU. It uses an Entry Queue IVR Service called Entry Queue IVR Service.</td>
</tr>
<tr>
<td></td>
<td>The default Welcome Slide displayed at the participants endpoint upon connection to the Entry Queue and lists the default Meeting Rooms. The participant can select one of these Meeting Rooms or enter another ID to start a new conference.</td>
</tr>
<tr>
<td></td>
<td>If no Transit Entry Queue is defined, DefaultEQ is the default Transit Entry Queue. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Entry Queues.</td>
</tr>
<tr>
<td></td>
<td>Note: An ISDN/PSTN dial-in number is not assigned to the Entry Queue as the number depends on the dial-in numbers range defined in the Network Service. It must be manually assigned to enable ISDN or PSTN participant connections to this Entry Queue. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Defining ISDN/PSTN Network Services.</td>
</tr>
<tr>
<td>Entry Queue IVR Service</td>
<td>Name: Entry Queue IVR Service</td>
</tr>
<tr>
<td></td>
<td>Includes all the voice messages and video slides used to guide AVC participants through their connection process to the MCU and route them to their destination conference.</td>
</tr>
<tr>
<td></td>
<td><em>Entry Queue IVR Service</em> is the default Entry Queue IVR Service provided for the default Entry Queue.</td>
</tr>
</tbody>
</table>

### Customizing the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Default Conferencing Settings

You can customize the conferencing entities to your organization’s requirements:

- To define SVC-based, Mixed CP and SVC or CP Only conferences, a new Conference Profile must be created, setting the appropriate Conferencing Mode, the video parameters for the conference.

This Profile can be used for defining new ongoing conferences and Meeting Rooms. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Defining New Profiles.
● To modify the properties of an existing CP AVC-based conference Profile, such as the conference
line rate, or specific video layout for the conference or the background that is used for the video
display (skin), create a new Conference Profile.

You can create a new CP AVC-based conference Profiles, to define additional conferencing
parameters and video session types.

This Profile can be used for defining new ongoing conferences, Meeting Rooms and Single-dial Entry
Queues.

For more information, see the Polycom® RealPresence Collaboration Server (RMX)

● To customize the Voice Prompts and Video Slides to different organizations, users, languages etc.
First record the required messages and create the video slides and then create the appropriate
conference IVR Service or Entry Queue IVR Service.

These services must be assigned to the appropriate conference profile or Entry Queue. For more
information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000

● To allow ISDN participants to connect to a single dial Entry Queue a dial in number must be
assigned to the pre-configured Entry Queue.

To allow AVC participants to connect to a single dial Entry Queue at a line rate other than 384 Kbps
(as in the default Entry Queue) or play voice messages in different languages – create a new Entry
Queue.

For more information, see the Polycom® RealPresence Collaboration Server (RMX)

● You can personalize Meeting Rooms for people in your organization with predefined conference and
chairperson passwords (for added security) and allow only authorized people to start on going
conferences. For more information, see the Polycom® RealPresence Collaboration Server (RMX)

● To allow ISDN participants to connect directly to Meeting Rooms a dial in number must be
assigned to the pre-configured Meeting Room.

The conferencing entities are designed mainly for dial in participants without prior definition of
participants. You can create your own Address Book containing a list of AVC participants to be dialed
by the MCU. Once defined, these participants can be added to ongoing conferences saving the need
to define them again.

For more information, see the Polycom® RealPresence Collaboration Server (RMX)

● You can schedule conferences to start in the future. For more details, see the Polycom®
Reservations.
Basic Operation

The MCU can be controlled using the Collaboration Server Web Client and the RMX Manager application. The most common operations performed via these applications are:

- Starting, monitoring and managing conferences
- Monitoring and managing participants and endpoints as individuals or groups.
  - Participant – A person using an endpoint to connect to a conference. When using a Room System, several participants use a single endpoint.
  - Endpoint – A hardware device, or set of devices, that can call, and be called by an MCU or another endpoint. For example, an endpoint can be a phone, a camera and microphone connected to a PC or an integrated Room System (conferencing system).
  - Group – A group of participants or endpoints with a common name.

Starting the Collaboration Server Web Client

You start the Collaboration Server Web Client by connecting to the MCU system. To connect to the MCU you need to get the following information from your system administrator:

- User name
- Password
- MCU Control Unit IP Address

To start the Collaboration Server Web Client:

1. In your browser address line, enter http://<Control Unit IP Address> and press the Enter key.
The Login screen is displayed.

1. Enter your **User Name** and **Password** and click the **Login** button.

   If the default User name and password were not changed on first entry, the default Username and Password are both **POLYCOM**.

   The Collaboration Server Web Client main screen is displayed.

2. **Link to RMX Manager Installer**

   The Login screen contains a link to the RMX Manager installer.

   Using the RMX Manager application, a single user can control a single or multiple Collaboration Server units as well as conferences from multiple Collaboration Servers.

   When upgrading from a previous version, it is recommended to install the RMX Manager latest version.

   For more information see the *Polycom® RealPresence Collaboration Server (RMX)* 1500/1800/2000/4000 Administrator’s Guide, RMX Manager Application.
Collaboration Server Web Client Screen Components

The Collaboration Server Web Client’s main screen consists of the following panes:

- Conference List
- Status Bar
- List Pane
- Address Book
- Collaboration Server Management
- Conference Templates Tab

You can login as a user with Chairperson, Operator or Administrator authorization. Your Authorization Level determines your viewing and system functions.

The Administrator’s view is shown below.

The main screen can be customized. For more information, see Customizing the Main Screen.

Viewing and System Functionality Permissions

Your User’s Authorization Level determines the viewing and system functionality of the Collaboration Server Web Client as summarized in the following table.
### Viewing and System Permissions

<table>
<thead>
<tr>
<th>Authorization Level</th>
<th>Chairperson</th>
<th>Operator</th>
<th>Administrator</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Viewing Permissions</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Conference List</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>List pane</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Address Book pane</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Conference Templates pane</td>
<td>✓</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Status Bar</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>RealPresence Collaboration Server Management pane</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Conference Alarms</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Conference Status</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Configurations dialog boxes</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td><strong>System Functionality</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Start Conferences</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Monitor Conferences</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Monitor Participants</td>
<td>✓</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Solve Basic Problems</td>
<td>✓</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Modify MCU Configuration</td>
<td></td>
<td></td>
<td>✓</td>
</tr>
</tbody>
</table>

In addition to Chairpersons, Operators and Administrators, additional user types are:

- **Auditor**: A user that can view Auditor Files and audit the system. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Auditor.
- **Administrator - Read Only**: A user with the viewing and monitoring permissions of a regular Administrator, but can only create system backups and cannot perform any other functional operation. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Administrator Read-only.

Conferences List

If you are logged in as a User with Operator or Administrator permissions:

The Conferences pane lists all the conferences currently running on the MCU along with their 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:

- **New Conference** – to start a new ongoing conference.
- **Delete Conference** – delete the selected conference(s).
- **Save Conference to Template** - to save the conference with its participants to a template for future use.
- If Conference Recording is enabled for AVC Only conferences, the following buttons are displayed in color:
  - **Start/Resume Recording** – start/resume recording.
    A Recording Indication is displayed to all conference participants informing them that the conference is being recorded.
  - **Stop Recording** – stop recording.
  - **Pause** – toggles with the **Start/Resume** button.
    A Paused Indication is displayed to all conference participants informing them that conference recording has been paused.

If you are logged in as a User with Chairperson permissions:

- You can list and monitor conferences you have started or for which you have entered the password or that don’t have a Chairperson Password assigned.
- A **Chairperson Password** field and a **Refresh** button are displayed.
  The **Refresh** button does not change the **Chairperson Password**; it refreshes the **Conferences** list to display all ongoing conferences with the requested password.
  For more information **Using the Chairperson Password for Filtering**.
A Chairperson Password column is included in the conference data.

List Pane

The List pane displays details of the item selected in the Conferences pane or RMX Management pane. The title of the pane changes according to the selected item.

Example: When an ongoing conference is selected in the Conferences pane, the list and parameters of the connected participants is displayed.

Selecting an item in the RMX Management pane lists the items currently defined.

Example: If the Users item is selected, a list of system Users defined for the MCU is displayed.

RMX Management Pane

This view is available to Operators and Administrators.

The RMX Management pane lists the entities that need to be configured to enable the Collaboration Server to run conferences. Only users with Administrators permission can modify these parameters.

The RMX 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.

Items can be moved between these two sections to customize the management tasks per system User. For more details, see Customizing the RMX Management Pane.

Status Bar

This view is available to Operators and Administrators.

The Status Bar at the bottom of the Collaboration Server Web Client contains System and Participant Alerts tabs as well as Port Usage Gauges and an MCU State indicator.
System Alerts

Starting from version 8.0/8.1, the basic unit used for reporting resource usage in the Port Gauges his HD720p30. Usage numbers are rounded to the nearest integer.

This view is available to Operators and Administrators.
This is a list of system problems. 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 pane is opened and closed by clicking the System Alerts button in the left corner of the Status Bar.

Active Alarms
Faults List

For more information about Active Alarms and Faults List, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, System and Participant Alerts.

Participant Alerts

This view is available to Operators, Chairpersons and Administrators.
This is a list of participants that are experiencing connection problems. It is sorted by conference.
The Participant Alerts pane is opened and closed by clicking the Participant Alerts button in the left corner of the Status Bar.
Port Usage Gauges

The Port Usage Gauges are displayed in the Status Bar at the bottom of the Collaboration Server Web Client screen.

In the RealPresence Collaboration Server (RMX) 1500/2000/4000, the Port Usage gauges displays for the selected MCU:

- The total number of Video or Voice ports in the system according to the Video/Voice Port Configuration. The Audio gauge is displayed only if Audio ports were allocated by the administrator, otherwise only the Video port gauge is displayed.
- The number of Video and Voice ports in use.
- The High Port Usage threshold.

In the RealPresence Collaboration Server 1800 and the RealPresence Collaboration Server (RMX) 2000/4000 with an MPMRx card, 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.
Port Gauges - RealPresence Collaboration Server (RMX) 1500/2000/4000 with MPMx cards

The basic unit used for reporting resource usage in the Port Gauges is HD720p30. Results are rounded to the nearest integer.

**MCU State**

This view is available to Chairpersons, Operators and Administrators.

The MCU State indicator displays one of the following:

- **Starting up (15:25)**: The MCU is starting up. The time remaining until the system start-up is complete is displayed between brackets while a blue progress indicator bar indicates the start-up progress.

- **MCU State: NORMAL**: The MCU is functioning normally.

- **MCU State: MAJOR**: The MCU has a major problem. MCU behavior could be affected and attention is required.

**Address Book**

This view is available to Chairpersons, Operators and Administrators.
The Address Book is a list of Participants and Groups that have been defined on the MCU. The information in the Address Book can be modified only by an administrator. All system users can, however, view and use the Address Book to assign participants to conferences.

The Address Book toolbar contains a **Quick Search** field and the following six buttons:

- New Participant
- New Group
- Delete Participant
- Delete Group
- Import Address Book
- Export Address Book

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** — when clicked, displays the single unique entity of all the participants in a single level as in previous versions. 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 Participants List in the Address Book lists entries according to:

- **Type** – whether an individual Participant or a Group of participants
- **Name** – name of the participant or group
- **Dialing Direction** – Dial-in or Dial-out
- **IP Address/Phone** – IP Address/Phone of the participant
- **Encryption** - whether the participant is encrypted, not encrypted or the system automatically selects the encryption according to the conference settings

**Displaying and Hiding the Address Book**

The first time you access the Collaboration Server Web Client, the Address Book pane is displayed. You can hide it by clicking the anchor pin button.
The Address Book pane closes and a tab appears at the right edge of the screen. Click the tab to re-open the Address Book.

Conference Templates

*Conference Templates* enable administrators and operators to create, save, schedule and activate identical conferences.

A Conference Template:

- Saves the conference Profile.
- Saves all participant parameters including their *Personal Layout* and *Video Forcing* settings.
- Simplifies the setting up *Telepresence* conferences where precise participant layout and video forcing settings are crucial.

Displaying and Hiding Conference Templates

The *Conference Templates* list pane 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 pane.

Hide the Conference Templates list pane by clicking the anchor pin button in the top right corner of the pane. The Conference Templates list pane closes and a tab appears in the top right corner of the screen.

**Customizing the Main Screen**

You can customize the main screen according to your preferences. Pane sizes can be changed, column widths can be adjusted and data lists can be sorted.

Customization settings are automatically saved for each logged-in user. The next time the Collaboration Server Web Client is opened, the main screen settings appear as they were when the user exited the application.

**To re-size a pane:**

» Move the pointer over the pane border and when the pointer becomes a ‡ click and drag the pane border to the required size and release the mouse button.

**To adjust column width:**

1 In the column header row, place the pointer on the vertical field- separator bar of the column.

2 When the pointer becomes a ‡, click and drag the field separator bar to the required column size and release the mouse button.

**To sort the data by any field (column heading):**

1 In the Conference list or List view pane, click on the column heading of the field to be used for sorting.

   A ▼ or ▲ symbol appears in the column heading indicating that the list is sorted by this field, as well as the sort order.

2 Click on the column heading to toggle the column’s sort order.

**To change the order of columns in a pane:**

» Click the column heading to be moved and drag it to its new position. When a set of red arrows appears indicating the column’s new position, release the mouse button.
To restore the RealPresence Collaboration Server display window to its default configuration:

» On the system menu, click View > Restore RMX Display Defaults.

Increasing and Decreasing the Text Size
You can increase or decrease the text size of the Collaboration Server Web Client windows for easier readability.

To increase or decrease the text size:

1 On the system menu, click View.
2 Select the drop-down arrow on the Zoom option and click the desired text size percentage (default is 100%).

In some cases, the text in the dialog boxes in the Collaboration Server Web Client might appear not aligned or incomplete. This issue is not caused by setting the Zoom option. Rather, the text size in the Display settings in Windows Control Panel may affect the display of the dialog boxes. Check the Windows Display settings by clicking the Windows Start button and selecting Control Panel > Display. For Windows XP, click the Appearance tab, select Normal for the Font size and click OK. For Windows 7, select the Smaller - 100% option and click OK.

Customizing the RMX Management Pane
The RMX Management pane can be viewed either as a list or as a toolbar.

To switch between Toolbar and List Views:

1 In the RMX Management pane, click the Toolbar View button to switch to Toolbar view.
2 In Toolbar view, click the List View button to switch back to List view.

In List view, you can move items between the Frequently Used and Rarely Used sections depending on the operations you most commonly perform and the way you prefer to work with the Collaboration Server Web Client. This option does not work in Toolbar view as all items are represented by icons.
To expand or collapse the Frequently Used and Rarely Used sections:
The Frequently Used and Rarely Used sections can be expanded or collapsed by clicking the and buttons.

To move items within and between the Frequently Used and Rarely Used sections:
1. In the RMX Management pane click and drag the icon of the item that you wish to move. An indicator line (→) appears indicating the new position of the icon.
2. Release the mouse button when the icon is in the desired position.
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 an AVC CP Conference from the Conferences Pane and Starting a New SVC Conference.

- Dialing in to a Meeting Room.

  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 the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Meeting Rooms.

- Dialing in to an Ad Hoc Entry Queue which is used as the access point to the MCU. This option is valid to AVC participants dialing into an AVC-based or a Mixed CP and SVC Entry Queue.


- Start a Reservation (AVC CP):

  - 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.

  Changes made to this information once the conference is running are not saved to the CDR.

- Start any Conference Template saved in the Conference Templates list.

  For more information, see Starting an Ongoing Conference From a Template.

- Clicking on the link included in a Microsoft Outlook Polycom Meeting Invitation or by manually dialing the numbers displayed in the invitation using the endpoint's numeric input device.

  This option is valid to AVC participants only.

  Attendees that have received an invitation to a Polycom Meeting via the Polycom Conferencing Add-in for Microsoft Outlook can start a conference by being the first invited attendee to click on a link in the Meeting Invitation displayed on his/her workstation or calendaring enabled endpoint, or manually dial in to the meeting using the Polycom Conference information included in the Meeting Invitation.

Starting an AVC CP Conference from the Conferences Pane

To start an AVC CP conference from the Conference pane:

1. In the Conferences pane, click the New Conference ( ) button. The New Conference – General dialog box opens.
The system displays the conference default Name, Duration and the default Profile, which contains the conference parameters and media settings.

The system 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 Collaboration Server 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.
General Tab

1 Define the following parameters:

New Conference – General Options

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| Display Name       | The Display Name is the conferencing entity name in native language character sets to be displayed in the RP 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.  
                    | **Note:** This field is displayed in all tabs.                                                                                                                                                                |
| Duration           | Define the duration of the conference in hours using the format HH:MM (default 01:00).  
                    | Default minimum duration is 11 minutes for the RealPresence Collaboration Server (RMX) 1500 and 20 minutes for RealPresence Collaboration Server (RMX) 2000/4000 when the Automatic Extension of conference duration mechanism is enabled. To define a shorter duration, the Automatic Extension of conference duration mechanism is disabled, by setting the system flag ENABLE_AUTO_EXTENSION to NO. For more information about system flags, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Modifying System Flags.  
                    | **Note:** This field is displayed in all tabs.                                                                                                                                                                |
| Permanent Conference | Click this check box to make this conference a Permanent Conference: an ongoing conference with no pre-determined End Time, continuing until it is terminated by an administrator, operator or chairperson. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Audio Algorithm Support.  
                    | **Note:** This field is displayed in all tabs.                                                                                                                                                                |
### New Conference – General Options (Continued)

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| 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 enter a different name. |
| Profile                | The system displays the name of the default Conference Profile. Select the required Profile from the list. The **Conference Profile** includes the conferencing mode, conference line rate, media settings and general settings.  
| 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.  
  **Note**: If setting the Conference ID to the digits that are used for MCU prefix in Gatekeeper (for example gatekeeper prefix is set to 10 and the conference ID is 1001), the system will not be able to dial to the destination conference as the prefix digits are truncated from the conference ID, preventing the system from locating it.  
  **Note**: If **SIP Factories** are used do not use the number 7001 as an ID. 7001 is the default **SIP Factory** ID. |
| Conference Password    | Enter a password to be used by participants to access the conference. If left blank, no password is assigned to the conference.  
  This password is valid only in conferences that are configured to prompt for a conference password.  
  This field is numeric and has a default length of 4 characters. The administrator can modify it in the Setup - System Configuration settings. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Modifying System Flags.  
  The Collaboration Server can be configured to automatically generate conference (and chairperson) passwords when these fields are left blank. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Automatic Password Generation Flags. |
### New Conference – General Options (Continued)

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Chairperson Password</td>
<td>Enter a password to be used by the Collaboration Server to identify the Chairperson and grant him/her 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. This field is numeric and has a default length of 4 characters. The administrator can modify it in the Setup - System Configuration settings. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Modifying System Flags. The Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide can be configured to automatically generate chairperson (and conference) passwords when these fields are left blank. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Manually Adding and Deleting System Flags.</td>
</tr>
</tbody>
</table>
| Reserve Resources for Video Participants | Collaboration Server 1500/2000/4000 only. Enter the number of video participants for which the system must reserve resources. Default: 0 participants. Maximum:  
  • x Mode: 60 participants (Double card assembly)  
  • x Mode: 30 participants (Single card assembly) |
| Reserve Resources for Audio Participants | Collaboration Server 1500/2000/4000 only. Enter the number of audio participants for which the system must reserve resources. Voice resources must be allocated in the Video/Voice Port Allocation dialog box. Default: 0 participants. Maximum:  
  • x Mode: 720 participants (Double card assembly, with all ports are set to Audio Only).  
  • x Mode: 360 participants (Single card assembly, with all ports are set to Audio Only). |
| Maximum Number of Participants    | Indicate the total number of participants that can be connected to the conference. The automatic setting indicates that the maximum number of participants that can be connected to the conference is determined according to resource availability.  
  **Note:** If a number is specified, it should be large enough to accommodate the participants specified in the Reserve Resources for Video/Audio Participants fields. |
| Enable ISDN/PSTN Dial-in          | Collaboration Server 1500/2000/4000 only. Select this check box if you want ISDN and PSTN participants to be able to connect directly to the conference. |
| ISDN/PSTN Network Service         | Collaboration Server 1500/2000/4000 only. The default Network Service is automatically selected. A different ISDN/PSTN Network Service can be selected from the Network Services list. |
2 If all participants are undefined, dial-in and no additional information is required for the new conference, click **OK**.

3 To add participants from the **Participants Address Book** or to define participants (mainly dial-out participants) click the **Participants** tab.

### Participants Tab

This procedure is optional.

The **Participants** tab is used to add participants to the conference from the **Address Book** or by defining them. Defined dial-out participants are connected to the conference automatically when the conference is launched, unless the **Dial Out Manually** option is selected.

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dial-in Number (1)</td>
<td>Leave this field blank to let the system automatically assign a number from the dial-in range defined for the selected ISDN/PSTN Network Service. To manually define a dial-in number, enter a <strong>unique</strong> number from the dial-in number range defined for the selected Network Service. This number cannot be assigned to another Conference/Reservation/Meeting Room/Gateway Profile.</td>
</tr>
<tr>
<td>Dial-in Number (2)</td>
<td>By default, the second dial-in number is not defined. To define a second dial-in number, enter a required number from the dial-in number range defined for the selected Network Service. This procedure is optional.</td>
</tr>
</tbody>
</table>

When defining a new conference, the **Participants List** is empty.

The following table describes the information displayed in the **Participants List** and the operations that can be performed.
### New Conference – Participants Tab

<table>
<thead>
<tr>
<th>Column / Button</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Participants List</strong></td>
<td></td>
</tr>
<tr>
<td>Name</td>
<td>A Unicode field that displays the participant’s name and an icon representing the endpoint type: <strong>Voice</strong> or <strong>Video</strong>.</td>
</tr>
<tr>
<td>IP Address/Phone</td>
<td>Indicates the IP address or phone number of the participant’s endpoint.</td>
</tr>
<tr>
<td>• For dial-out connection, displays the IP address or phone number of the endpoint called by the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000.</td>
<td></td>
</tr>
<tr>
<td>• For dial-in connection, displays the participant’s IP address or phone number used to identify and route the participant to the appropriate conference.</td>
<td></td>
</tr>
<tr>
<td>Alias Name/SIP Address (IP Only)</td>
<td>Displays the alias name of an H.323 endpoint or the SIP URL.</td>
</tr>
<tr>
<td>Network</td>
<td>The network communication protocol used by the endpoint to connect to the conference: <strong>IP</strong> (<strong>H.323</strong> or <strong>SIP</strong>) or <strong>ISDN/PSTN</strong> (Collaboration Server 1500/2000/4000 only).</td>
</tr>
<tr>
<td>Dialing Direction</td>
<td>Select the dialing direction:</td>
</tr>
<tr>
<td>• <strong>Dial-in</strong> – The participant dials in to the conference. This field applies to IP participants only.</td>
<td></td>
</tr>
<tr>
<td>• <strong>Dial-out</strong> – The MCU dials out to the participant. <strong>Note</strong>: Dial-out is forced when defining an ISDN/PSTN participant (Collaboration Server 1500/2000/4000 only).</td>
<td></td>
</tr>
<tr>
<td>Encryption</td>
<td>Displays whether the endpoint uses encryption for its media. The default setting is <strong>Auto</strong>, indicating that the endpoint must connect according to the conference encryption setting.</td>
</tr>
<tr>
<td>Lecturer</td>
<td>This option is used to activate the <strong>Lecture Mode</strong>. Select the participant you want to designate as <strong>Lecturer</strong> from the list of conference participants. The Lecturer can be selected once the conference is running and participants are connected.</td>
</tr>
<tr>
<td>Dial Out Manually</td>
<td>Select this option to designate a Collaboration Server User-controlled dial-out conference connection. When checked, the user must connect each of the dial-out participants who remain on standby until they are connected to the conference. This option is disabled in SVC conferences.</td>
</tr>
<tr>
<td><strong>Buttons</strong></td>
<td></td>
</tr>
<tr>
<td>New</td>
<td>Click to define a new participant. For more information, see the <em>Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide</em>, <em>Adding a New participant to the Address Book Directly</em>.</td>
</tr>
<tr>
<td>Remove</td>
<td>Click to remove the selected participant from the conference.</td>
</tr>
<tr>
<td>Add from Address Book</td>
<td>Click to add a participant from the Address Book to the conference.</td>
</tr>
</tbody>
</table>
Participants can be added to the conference in the following methods:

- Defining a new participant during the definition of the conference (clicking the **New** button).
- Adding pre-defined participants from the *Address Book* by either selecting the participants from the list or dragging and dropping the participants from the *Address Book* to the Participants list.
- Dial-in participants can connect to the conference after it was started (without using the *New Conference - Participants* dialog box).
- Once the conference has started, participants can be added to a conference directly from the Participants *Address Book* without having to use the *New Conference – Participants* tab. For more details, see *Adding Participants from the Address Book*.

**To add participants from the Address Book:**

1. In the *Participants List*, click the **Add from Address Book** button to open the *Participants Address Book*.

   ![Participants Address Book](image)

   - The *All Participants* list opens.

   2. In the *Participants Address Book*, select the participants that you want to add to the conference and click the **Add** button.

      Standard Windows multiple selection techniques can be used in this procedure.

   3. The selected participants are assigned to the conference and appear in the *Participant List*.

   4. Select additional Participants or click the **Close** button to return to the *Participants* tab.
Information Tab

In the *Info* fields, you can add general information about the conference, such as contact person name, company name, billing code, etc.

This information is written to the *Call Detail Record (CDR)* when the conference is launched. Changes made to this information once the conference is running are **not** saved to the *CDR*.

This procedure is optional.
The information entered into these fields does not affect the conference.

To add information to the conference:

1. Click the **Information** tab.

   The **Information** tab opens.

   ![Information Tab](image)

2. Enter the following information:

   **New Conference – Info Options**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Info1, 2, 3</td>
<td>There are three information fields that allow you to enter general information for the conference such as company name, contact person etc. Unicode can be used in these fields. The maximum length of each field is 80 characters.</td>
</tr>
<tr>
<td>Billing</td>
<td>Enter the conference billing code if applicable.</td>
</tr>
</tbody>
</table>
3 Click OK.

An entry for the new conference appears in the Conferences pane.

If no participants were defined for the conference or as long as no participants are connected, the indication Empty and a warning icon (▲) appear in the Status column in the Conferences pane.

The status changes when participants connect to the conference.

If no participant connects within the time specified in the Conference Profiles > Auto Terminate > Before First Joins field, the conference is automatically terminated by the system.

**Starting a Mixed CP and SVC or SVC Only Conference from the Conferences Pane**

**Starting a New SVC Conference**

Starting from version 8.1:
- A license is required for SVC conferencing.
- In mixed AVC/SVC conferences, participants with SVC-enabled endpoints and AVC endpoints can participate in the same conference.
- During mixed SVC & CP conferences, PSTN (Audio Only) calls are also supported (Collaboration Server 1500/2000/4000 only).

**To start a SVC conference from the Conference pane:**

1 In the Conferences pane, click the New Conference (▲) button.
The New Conference – General dialog box opens.

The system displays the conference default *Name*, *Duration* and the default *Profile*, which contains the conference parameters and media settings.

The system automatically allocates the conference *ID*, when the conference starts.

By default, the system assigns the factory default profile to the conference, which is set to AVC conferencing Mode.

2 To start a mixed CP and AVC conference or an SVC-based conference, you must select a *Profile* set to *SVC* or Mixed CP and SVC Conferencing Mode.

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 **Collaboration Server 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.

If no defined participants are to be added to the conference, or you do not want to add additional information, click **OK**.

3 In *SVC* or mixed CP and SVC Conferencing Mode, only the following parameters can be defined:
### New SVC Conference – General Parameters

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| 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.  
  **Note:** This field is displayed in all tabs. |
| Duration            | Define the duration of the conference in hours using the format HH:MM (default 01:00).  
  Default minimum duration is 11 minutes for the RealPresence Collaboration Server (RMX) 1500 and 20 minutes for RealPresence Collaboration Server (RMX) 1800/2000/4000 when the Automatic Extension of conference duration mechanism is enabled. To define a shorter duration, the Automatic Extension of conference duration mechanism is disabled, by setting the system flag `ENABLE_AUTO_EXTENSION` to **NO**. For more information about system flags, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Modifying System Flags.  
  **Note:** This field is displayed in all tabs. |
| Permanent Conference| Click this check box to make this conference a Permanent Conference: an ongoing conference with no pre-determined End Time, continuing until it is terminated by an administrator, operator or chairperson. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Audio Algorithm Support.  
  **Note:** This field is displayed in all tabs. |
### New SVC Conference – General Parameters (Continued)

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
</table>
| 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 system displays an error message and requests that you enter a different name. |
| Profile                | The system displays the name of the default *Conference Profile*. The default Profile is set to mixed **CP and SVC** conferencing Mode. Select the required Profile from the list. The Conference Profile includes the Conferencing Mode, conference line rate, media settings and general settings. For a detailed description of Conference Profiles, see the *Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Using Conference Profiles*. |
| 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.  
**Note:** If setting the Conference ID to the digits that are used for MCU prefix in Gatekeeper (for example gatekeeper prefix is set to 10 and the conference ID is 1001), the system will not be able to dial to the destination conference as the prefix digits are truncated from the conference ID, preventing the system from locating it.  
**Note:** If **SIP Factories** are used do not use the number 7001 as an ID. 7001 is the default **SIP Factory** ID. |
| Maximum Number of Participants | Indicate the total number of participants that can be connected to the conference. The automatic setting indicates that the maximum number of participants that can be connected to the conference is determined according to resource availability.  
**Note:** If a number is specified, it should be large enough to accommodate the participants specified in the *Reserve Resources for Video/Audio Participants* fields. |
Participants Tab

This procedure is optional.

SVC-based conferencing allows dial-in connections only, and it is designed for ad-hoc conferencing where participants just call in without defining their properties in advance. However, you can use the Participants tab to add dial-in participants to the conference from the Address Book or by defining them directly.

In mixed CP and SVC conferencing, dial-out AVC (H.323 and SIP) connections are allowed.

Information Tab

In the Info fields, you can add general information about the conference, such as contact person name, company name, billing code, etc.

This information is written to the Call Detail Record (CDR) when the conference is launched.

Changes made to this information once the conference is running are not saved to the CDR.

This procedure is optional.
The information entered into these fields does not affect the conference.

Scheduling an AVC-based Reservation

AVC-based Reservations are started by selecting an AVC-based Profiles.

To start a conference from the Reservation Calendar:

1. In the RMX Management pane, click the Reservations entry ( ). The Reservation Calendar is displayed.
2 Click the **New Reservation** button.

The **New Reservation – General** tab dialog box opens.
This dialog box is identical to the AVC CP New Conference - General dialog box. For a field description, see General Tab.

3 Optional (Collaboration Server 1500/2000/4000 only). Select the Enable ISDN/PSTN Dial-in check box if you want ISDN and PSTN participants to be able to connect directly to the conference.

4 If Enable ISDN/PSTN Dial-in option is selected, either enter a dial-in number, or leave the Dial-in Number field blank to let the system automatically assign a number from the dial-in range defined for the selected ISDN/PSTN Network Service.

5 Click the OK button.

A confirmation box is displayed stating that the Reservation time is past due and that the conference will become ongoing.

6 Click OK. The conference is started. If applicable, an ISDN/PSTN dial-in number was assigned to the conference either automatically or manually, this number can be viewed in the Conferences pane. For more information about Reservations, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Scheduling Reservations.

Starting an Ongoing Conference From a Template

An ongoing conference can be started from any Conference Template saved in the Conference Templates list.

SVC-based Conference Templates cannot be saved with dial-out participants, only dial-in SIP participants may be defined.

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 exist in the system, the conference will not be started.


**Starting a Meeting from Microsoft Outlook using the Polycom Conferencing Add-in to Microsoft Outlook (AVC CP Only Conferencing)**

*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®**. For more information see the *Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide*, *Polycom Conferencing for Microsoft Outlook®*.

This option is not valid for SVC Only or Mixed CP and SVC conferencing.

**Starting an Audio Meeting from a Microsoft Outlook Polycom Meeting Invitation**

Using the *Polycom Conferencing Add-in to Microsoft Outlook*, a meeting can be created as any other Outlook meeting. For more details, see **Appendix H - Integration Into Microsoft Environments**.
Connecting to a Conference

Direct Dial-in to the MCU

Direct dial-in connection to conferences and Meeting Rooms is available to all endpoints. Participants must be provided with a dialing string which can vary according to the network type, conference password and chairperson password.

Participants dial the conference dial-in string and are connected to the conference IVR Service. Once the correct information, such as the conference password and chairperson password are entered, the participants are connected to the conference.

Dial-in Connection via IVR System (Collaboration Server 1500/2000/4000)
In SVC Only Conferencing Mode, only SIP dial-in using the SVC protocol is available as follows:

- AVC-capable endpoints (participants) can only connect to an AVC 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 Only conferences, they connect as AVC endpoints.

In CP AVC or Mixed CP and SVC Conferencing Mode, the MCU can be configured to enable the chairperson to use the chairperson password as the conference password without the need to enter the conference password.

In a Mixed CP and SVC conference, only dial-in participants (AVC and SVC-based) can connect to the conference.
H.323 Participants (AVC CP Only and Mixed CP and SVC Conferencing)

For H.323 participants, the dialing string is composed of the MCU prefix in the Gatekeeper and the conference ID.

Example:

Prefix in gatekeeper: 925
Conference ID: 1001
Conference Name: Maple_Room

» The participant dials 9251001 or 925Maple_room

If there is no gatekeeper defined for the network, H.323 participants dial the MCU's signaling host IP address and the conference ID, separated by ##.

Example:

MCU (Signaling Host) IP address: 172.22.30.40
Conference ID: 1001

» The participant dials 172.22.30.40##1001

SIP Participants (All Conferences)

For SIP participants the dialing string is composed of:

- The conference routing name and domain name in the following format:
  conference_routing_name@domain_name
- The conference routing name and the IP address of the MCU signaling in the following format:
  conference_routing_name@IP Address of the MCU Signaling

Example:

Conference routing name: 1001
MCU domain name: polycom.com
MCU (Signaling Host) IP address: 172.22.20.42

» The participant dials 1001@polycom.com
  or
  The participant dials 1001@172.22.20.42

ISDN/PSTN Participants (AVC Only Conferencing)

ISDN/PSTN calls are not supported on the Collaboration Server (RMX) 1800.
Dial-in ISDN and PSTN participants dial one of the dial-in numbers assigned to the conference/Meeting Room/Reservation/Conference Template, including the country and area code (if needed). They are routed to their conference according to the dial-in number.
Example:

Assigned dial-in number: 4631111

The participant dials: 4631111

When adding a defined dial-out participant to the conference, and that participant dials in as an undefined participant before she or he is called by the system, the Collaboration Server is unable to identify that it is the same participant and will have two separate participant entries in the conference for the same participant. In such a case, since the dial-in participant is already connected, the dial-out participant will be displayed as disconnected (line is busy).

For MCUs registered to a gatekeeper, the RMX 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. Direct IP dialing is enabled or disabled by the GK_MANDATORY_FOR_CALLS_OUT and GK_MANDATORY_FOR_CALLS_IN System Flags. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Direct IP Dialing.

Entry Queue Access

Access via an Entry Queue allows all participants to dial the same entry point that acts as a routing lobby. Once in the Entry Queue, participants are guided to the conference according to the conference ID they enter.

Move between the Entry Queue and the destination conference is enabled when both conferencing entities have the same conferencing parameters (it is recommended that the same Profile is used for both entities). For example, participants connected to a mixed CP and SVC conference can only be moved to a mixed CP and SVC conference.
Dial-in Connection via Entry Queue (Collaboration Server 1500/2000/4000)

Dial-in Connection via IVR System (Collaboration Server 1800)
**H.323 Participants (AVC Participants)**

Dialing is executed in the same way as for conferences, where the Entry Queue ID/Name replaces the Conference ID/Name.

» H.323 participants dial \[Gatekeeper Prefix\][Entry Queue ID/Name].

**Example:**

Prefix in gatekeeper: 925
Entry Queue ID: 1000

» The participant dials: 9251000

H.323 participants can bypass the Entry Queue IVR voice messages by adding the correct Conference ID of destination conference to the initial dial string:

\[Gatekeeper Prefix\][EQ ID][\#Destination Conference ID]

**Example:**

Conference ID: 1001

» H.323 participants dial: 9251000##1001

H.323 participants can also bypass the conference IVR voice messages by adding the Conference Password to the initial dial string:

\[Gatekeeper Prefix\][EQ ID][\#Destination Conference ID][\#Password]

**Example:**

Conference ID: 1001
Conference Password: 34567

» H.323 participants dial: 9251000##1001##34567

**SIP Participants (All Participants)**

Using an Entry Queue or Meeting Room minimizes the number of conferences that require registration with the SIP server and enables using one URI address for all dial-in connections, using the format:

- `<Entry Queue routing name>@<domain name>` or
- `<Entry Queue routing name>@<MCU signaling IP Address>` or
- `<Meeting_Room_Name>*<password>@MCU signaling IP Address`

**Examples:**

**Entry Queue Routing Name:** DefaultEQ
**Domain Name:** polycom.com
**MCU Signaling IP address:** 172.22.20.42

» SIP participants dial: DefaultEQ@polycom.com
or

» DefaultEQ@172.22.20.42

Meeting Room Name: Maple_Room
Password: 1234
MCU Signaling IP address: 172.22.20.42

» SIP participants dial Maple_Room**1234@172.22.20.42

**ISDN and PSTN Participants (AVC only Participants)**

ISDN/PSTN calls are not supported on the Collaboration Server (RMX) 1800.

Up to two dial-in numbers can be allocated to an Entry Queue for use by ISDN and PSTN participants.

Calls to numbers within the ISDN and PSTN Dial-in Range that are not allocated to an Entry Queue are routed to the Transit Entry Queue.

Dial-in ISDN and PSTN participants dial one of the dial-in numbers assigned to the Entry Queue, including the country and area code (if needed).

They are routed to their conference according to the conference ID.

**Example:**

Entry Queue ID: 1000
Assigned Dial-in number: 4631000

» ISDN/PSTN participants dial 4631000
Connecting to a Polycom Conference from an Outlook Meeting Invitation (AVC Only Conferencing)

Attendees that have received a *Meeting Invitation* via *Outlook* using the *Polycom Conferencing Add-in for Microsoft Outlook*, connect by clicking on the link included in the invitation or by manually dialing the numbers displayed using the endpoint's numeric input device.

For more information see the *Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide*, *Polycom Conferencing for Microsoft Outlook®*.

Dial-out Participants (AVC CP Only Conferencing)

Dial-out participants are participants that the MCU dials to their endpoint to connect them to the conference. These participants must be defined in the conference when the conference is started (they are usually added to the conference from the Address Book.)

Dial-out is not supported for SVC participants.

From version 8.1, H.323 Alias or SIP address can be used instead of the IP address for a participant when a scheduled conference or conference from a template is started.
Automatic Dial Out

Dial-out participants are defined with their dial-out number. Once they are added to the ongoing conference, the MCU automatically calls them at a rate of 1 dial-out per second, using the default IP or ISDN/PSTN Network Service defined for them.

Manual Dial Out

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 telephone number) and added to the conference. This mode can only be selected at the conference definition stage and cannot be changed once the conference is running.

Gathering Phase (AVC Only Conferencing)

The Gathering Phase of a conference is the time period during which participants are connecting to a conference. It is enabled for the conference in the Conference Profile - Gathering Settings dialog box.

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

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.
Gathering Phase Guidelines

- The Gathering Phase slide can be displayed at any time during the conference by entering the Show Participants DTMF code, *88.
- The Gathering Phase is not supported in Video Switching Conferences.
- 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
  - Korean
  - French
  - Japanese
  - German
  - Simplified Chinese
  - International Spanish
- **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.

For more information see, the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Auto Scan and Customized Polling in Video Layout (CP Conferences Only).
Audio and Visual Indications (AVC CP Conferencing)

During CP conferences, visual and audio indications are used. Various tones and voice messages may be played during the conference, depending on the IVR Service configuration. For more details, see Audio and Visual Indications (AVC CP Conferencing).

Visual indications can be displayed on the screen of the endpoint participating in the meeting, usually as an overlay layer on top of the video layout that includes the participants video.

These indications include:

- Site Names displaying the names of the endpoints that are connected to the conference
- Text messages via closed captions to provide real-time text transcriptions or language translations
- Text messages sent by the meeting organizer to all conference participants or selected participants
- Network Quality icon indicating the quality of the network used by a participant to connect to the conference

Site Names

During conferences you can view the names of the endpoints that are connected to the conference in your endpoint’s video layout windows. The MCU can display up to 33 characters of the endpoint’s name, depending on the window’s layout (size).

The following is an example of endpoint name display in the endpoint screen:

![Example of endpoint name display](image)

The display of the site names is enabled or disabled in the conference Profile for CP Only or Mixed CP and AVC conferences for AVC-based endpoints.

SVC-based endpoints handle the display of the site names themselves and are not dependant on the conference Profile. The Site Names option is not supported in Video Switching conferences (Collaboration Server 1500/2000/4000).

The displayed name is determined as follows:

- The system displays the name that is defined at the endpoint.
- If the endpoint does not send its name:
  - For a defined H.323 or SIP participant:
    - The system displays the name from the participant definition.
  - For an undefined H.323 participant:
Display the **H.323 ID** alias.
or
Display the **E.164** alias.
or
Display nothing if all the fields are empty.

- For a SIP undefined participant:
  - Display the SIP **DisplayName** field.
or
  - Display the SIP **Address** (SIP application server).
or
  - Display the SIP **ContactDisplay** field.
or
  - Display nothing if all the fields are empty.

- For a defined H.320 participant (Collaboration Server 1500/2000/4000):
  - The system displays the name from the participant definition.

- For an undefined H.320 participant:
  - Display the **Terminal Command String** (TCS-2) to identify the participant.
or
  - Display nothing if the string is not received or empty.

- If the endpoint’s **Display Name** is changed in the Collaboration Server Web Client, it overrides all the above.

**To change the Display Name:**

1. In the **Participants** list, double-click the participant or right-click the participant and then select **Participant Properties**.
2. Click the **Media Sources** tab.
3 Enter the new Display Name in the Name field.
4 Click OK.

Displaying and Hiding Site Names

Site Names display are enabled or disabled display characteristics are controlled in the Conference Properties - Site Names dialog box.


Transparent Site Names

Endpoint name backgrounds are 50% transparent, and while maintaining contrast, do not completely obscure the overlaid video.

Site Name Transparency is controlled using the Site Names tab in the New Profile and Conference Properties dialog boxes.


Permanent Display of Site Names

Site Names can be permanently displayed on the screens of the endpoints.

Permanent display of Site Names is controlled using the Site Names tab in the New Profile and Conference Properties dialog boxes.
For more information see Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide.

Location of Site Names
The default location (top left) of the Site Names in the video layout can be changed.

All Site Names display characteristics are controlled using the Site Names tab in the New Profile and Conference Properties dialog boxes.


Obtaining the Display Name from the Address Book
The MCU can be configured to replace the name of the dial-in IP 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/RRM 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.

For more details, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Obtaining the Display Name from the Address Book.

Closed Caption (Collaboration Server 1500/2000/4000 AVC Only)
When enabled, IP endpoints that support FECC (Far End Camera Control) can also be configured to provide real-time text transcriptions or language translations of the video conference by displaying closed captions.

The captions for a conference may be provided by the captioner who is present in the conference, or the captioner may use a telephone or web browser to listen to the conference audio. When the captioner sends a unit of text, all conference participants see it on the main monitor for 15 seconds. The text then disappears automatically.

Endpoint Name display is not affected by Closed Captions display.

For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Packet Loss Compensation (LPR and DBA) AVC CP Conferences.

Closed Captions option is enabled by a system flag in the system configuration. For more details about system flags, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Modifying System Flags.

Message Overlay for Text Messaging (AVC Only)
Message Overlay allows messages to be sent during a conference to a single participant, selected number of participants or all participants in to an ongoing conference.
The number of characters that can be included in a message varies according to the language and can differ due to the type of font used. For example, the available number of characters in Chinese is 50 (32 for Collaboration Server 1800) for English and 50 (48 for Collaboration Server 1800) and for Russian.

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.

The message can be set to be displayed at various positions on the screen in different colors, static or scrolling.

For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide Packet Loss Compensation (LPR and DBA) AVC CP Conferences.

**Network Quality Indication**

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

The display of the *Network Quality* icon can be customized for the following:

- The participant’s own endpoint
- 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 modifying the values of the relevant *System Flags*. 
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.

The default Major and Critical indication threshold values can be manually modified by changing the system flag values. Major and Critical states are indicated with yellow and red indicator bars respectively.

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.

Network Quality Indicators Guidelines

Network Quality Indicators 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 Indicators are displayed by default and can be disabled
  - For media transmitted to and received from the RMX (Video in / Video out).
- Participants displayed in the cells of the conference Video Layout:
  - Network Quality Indicators are not displayed by default and can be enabled
  - The media transmitted from the endpoint to the RMX (Video in).

Network Quality Indicators:

- Are supported with both x and Rx cards.
- Are not supported in:
  - SVC Conferencing Mode
  - AVC - Video switched conferences

Audio Indications

During the definition of the IVR Service which is assigned to the conference, you can enable any of the audio tones (Entry tone and Exit tone) or voice messages (when a participant joins the conference or leaves the conference) to be heard during an ongoing conference. If Roll Call is enabled for the conference, a roll call can be requested. In addition, various other massages may be heard during the conference, if enabled. For example, when the conference is being recorded, or when the conference is locked.
Noisy Line Detection and Automatic Muting of Noisy Endpoints

The Collaboration Server can detect AVC-enabled endpoints with a noisy audio channel and automatically mute them, reducing the noise heard by other conference participants. When the auto muted endpoint becomes the “speaker” the endpoint is automatically un-muted by the system. If the speaker halts his/her conversation and the line still emits noises, the endpoint will be automatically muted again.

When the endpoints are automatically muted by the MCU, no indication is displayed in the Collaboration Server Web Client or RMX Manager as the system does not consider it as a “real” mute.

Monitoring Ongoing Conferences

Conference monitoring enables you to keep track of conferences and their participants: if all its participants are correctly connected and whether errors or faults have occurred.

Monitoring can be done at three levels:

● General Monitoring
● Conference Level Monitoring
● Participant Level Monitoring

General Monitoring

All monitoring and operations procedures performed during ongoing conferences can be performed by either of two methods:

● Using the buttons in the toolbars.
- **Right-clicking** an entity in the *Conferences or Participants* pane and selecting an operation from the menu.

**Multi Selection**

Using multiple selection, you can monitor and perform simultaneous operations on multiple participants in multiple conferences.

The selected conferences are displayed as sub-lists in the *Participants* list pane. The sub-lists can be expanded and collapsed by clicking the and sublist control buttons that appear next to the conference name in the sub-list headings.

**Using the Chairperson Password for Filtering**

If you are logged in as a chairperson, the *Chairperson Password* field is displayed. It enables you to search for, and display a list of, ongoing conferences for which you have the password.
To search for an ongoing conference by Chairperson Password:

1. Click in the Chairperson Password field.
2. Enter the password to be searched for.
3. Click the Refresh button.

The Conferences list is refreshed and ongoing conferences with the requested password are displayed.

Conference Level Monitoring

Conference level monitoring is available to the administrator, operator and chairperson. In Collaboration Server 1500/2000/4000, monitoring of VSW conferences is done in the same way as for CP conferences.

The Conference List pane displays information about ongoing conferences. No status indicator display in the Status column means that the conference is running without problems. One or more of the status indicators may appear in the Status column.

Conferences – Monitoring Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display Name</td>
<td>Displays conference name and type of conference:</td>
</tr>
<tr>
<td></td>
<td>• 🎤 – AVC Conference running in CP mode.</td>
</tr>
<tr>
<td></td>
<td>• 🎤 – AVC Conference running in Video Switching mode (Collaboration Server 1500/2000/4000).</td>
</tr>
<tr>
<td></td>
<td>• 🎤 – The AVC conference has been secured using the *71 DTMF code. For details, see Secured Conference Monitoring (AVC CP Only).</td>
</tr>
<tr>
<td></td>
<td>• 🎤 – AVC Operator Conference</td>
</tr>
<tr>
<td></td>
<td>• 🎤 – Media Relay - SVC-based Conference.</td>
</tr>
<tr>
<td></td>
<td>• 🎤 – Mixed CP and SVC Conference</td>
</tr>
</tbody>
</table>
Conferences – Monitoring Information (Continued)

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status</td>
<td>Displays the status of the ongoing conference. If there is no problem with the participant's connection no indication is displayed. If one of the following statuses occur, the appropriate indication is displayed, proceeded by a warning icon ( ).</td>
</tr>
<tr>
<td></td>
<td>• Audio – There is a problem with the participant's audio.</td>
</tr>
<tr>
<td></td>
<td>• Empty – No participants are connected.</td>
</tr>
<tr>
<td></td>
<td>• Faulty Connection – Participants are connected, but the connection is problematic.</td>
</tr>
<tr>
<td></td>
<td>• Not Full – Not all the defined participants are connected.</td>
</tr>
<tr>
<td></td>
<td>• Partially Connected – The connection process is not yet complete; the video channel has not been connected.</td>
</tr>
<tr>
<td></td>
<td>• Single Participant – Only one participant is connected.</td>
</tr>
<tr>
<td></td>
<td>• Video – There is a problem with the participant’s video.</td>
</tr>
<tr>
<td></td>
<td>• Content Resource Deficiency – Content will not be sent to legacy endpoints.</td>
</tr>
<tr>
<td></td>
<td>• Awaiting Operator – A participant has requested operator assistance.</td>
</tr>
<tr>
<td>ID</td>
<td>The Conference ID assigned to the conference.</td>
</tr>
<tr>
<td>Start Time</td>
<td>Conference start time.</td>
</tr>
<tr>
<td>End Time</td>
<td>The time the conference is expected to end.</td>
</tr>
<tr>
<td>Dial in Number(1)</td>
<td>The Conference dial in number for ISDN/PSTN participants (Collaboration Server 1500/2000/4000).</td>
</tr>
<tr>
<td>SIP Registration</td>
<td>The status of registration with the SIP server:</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• 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 MCU or the SIP Server to the network.</td>
</tr>
<tr>
<td></td>
<td>• Registered - The conferencing entity is registered with the SIP Server.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
</tbody>
</table>

Additional information about the conference can be viewed when accessing the conference properties.

**To monitor a conference:**

» In the Conference List pane, double click the name of the conference you wish to monitor, or right-click the conference and then click Conference Properties.
The **Conference Properties - General** dialog box opens.

You can view all the conference properties but those that appear with a gray background cannot be modified.

**Roll Call/Audible Tones During an Ongoing AVC Only Conference**

If **Roll Call** is enabled, when a participant joins or leaves a conference, the system plays a voice message to all participants. The audio message consists of the joining or leaving participant’s **Roll Call** name followed by either “has joined the conference” or “has left the conference”.

These voice messages can be replaced with audio tones. The use of tones requires that the appropriate tone files in *.wav format be uploaded and configured to replace the **Roll Call Joined** and **Roll Call Left** message files.

These options are enabled or disabled by selection in the **Conference IVR Service Properties - Roll Call/Notifications** dialog box.

For more information see the *Defining a New Conference IVR Service*. 
Audio Only Message (AVC CP Only)

H.323 and SIP video participants that are connected as Secondary (Audio Only) because of a lack of video resources receive an audio message: “All video resources are currently in use. Connecting using audio only” indicating why their video has not connected.

Secured Conference Monitoring (AVC CP Only)

When Secured Conference Mode is enabled on the Collaboration Server, it locks the conference and prevents participants and Collaboration Server Users from joining the conference. A Secured conference cannot be monitored or controlled in any way. While a conference is in the Secure Mode, the Collaboration Server User with Administrator authorization cannot view the participants list or any other conference properties, but can manually terminate it.

The conference chairperson can enable or disable the Secure mode from the DTMF input device (touch-tone telephone or the endpoint’s remote control), using the appropriate DTMF code (the default code is *71). During the secure conference, the chairperson and participants can perform various operations, such as muting, via their DTMF input device using the appropriate DTMF codes. However, because Secure conferences cannot be monitored, these operations or status changes are not displayed in the Conferences or Participants panes.

A special icon 🗝 is used to indicate that the conference is in Secure mode.

Monitoring Ongoing Gateway Sessions (AVC Only)

Not applicable to the RMX 1800.

Ongoing Gateway Sessions that are created when calling the Gateway Profile, are listed in the ongoing Conferences list pane and are monitored in the same way as the conferences.

For more details, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Monitoring Ongoing Conferences.

Participant Level Monitoring

Participant Connection Monitoring (AVC and SVC-based Connections)

When a conference is selected in the Conference list, the Participants list header displays a summary of participant connection status.
These numbers include:

- **EP** = the number of Endpoints currently connected to the conference (both defined and undefined participants). This number includes participants whose status is connected with problem, connected partially or connected as secondary. Connected Cascading Links are not included and are detailed separately.

- **L** = the number of Cascading Links currently connected to the conference.

- **T** = the total number of all:
  - connected Participants - both defined and undefined participants
  - defined participants that are currently disconnected
  - Cascading Links - both connected and disconnected

If more than one conference is selected, the **EP:n L:n T:n** numbers reflect the cumulative connection status information of all the selected conferences.

If no conference is selected, the **EP:n L:n T:n** numbers are all zeroed.

If the **T**(otal) number of participants is higher than the number of connected participants, it indicates that additional participants defined for the conference have not yet connected. For example, if the **EP:** is **12** and the **T:** is **15**, it means that 12 out of 15 of the expected participants have connected and that **3** defined participants have not yet connected.

The *Participant* list displays the following participant indicators and properties:

**Participant Monitoring – Indicators and Properties**

<table>
<thead>
<tr>
<th>Column</th>
<th>Icon/Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
<td>Displays the name and type (icon) of the participant:</td>
</tr>
<tr>
<td></td>
<td><img src="image" alt="Audio Participant" /> – Connected via IP phone or ISDN/PSTN.</td>
</tr>
<tr>
<td></td>
<td><img src="image" alt="Video Participant" /> – Connected with audio and video channels.</td>
</tr>
<tr>
<td>Status</td>
<td>Displays the connection status (text and icon) of the participant. If there is no problem with the participant's connection no indication is displayed.</td>
</tr>
<tr>
<td></td>
<td><img src="image" alt="Connected" /> – The participant is successfully connected to the conference.</td>
</tr>
<tr>
<td></td>
<td><img src="image" alt="Disconnected" /> – The participant is disconnected from the conference. This status applies only to defined participants.</td>
</tr>
<tr>
<td></td>
<td><img src="image" alt="Waiting for Dial-in" /> – The system is waiting for the defined participant to dial into the conference.</td>
</tr>
<tr>
<td></td>
<td><img src="image" alt="Partially Connected" /> – The connection process is not yet complete; the video channel has not been connected.</td>
</tr>
<tr>
<td></td>
<td><img src="image" alt="Faulty Connection" /> – The participant is connected, but problems occurred in the connection, such as synchronization loss.</td>
</tr>
<tr>
<td></td>
<td><img src="image" alt="Secondary Connection" /> – The endpoint’s video channel cannot be connected to the conference and the participant is connected only via audio.</td>
</tr>
<tr>
<td></td>
<td><img src="image" alt="Awaiting Individual Assistance" /> (AVC-based connection) – The participant has requested the user’s (operator’s) assistance.</td>
</tr>
</tbody>
</table>
### Participant Monitoring – Indicators and Properties

<table>
<thead>
<tr>
<th>Column</th>
<th>Icon/Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Awaiting Conference Assistance (AVC-based connection) – The participant has requested the operator’s assistance for the conference. This usually means that the user (operator) has been requested to join the conference.</td>
<td></td>
</tr>
<tr>
<td>Role</td>
<td>Displays the participant’s role or function in the conference:</td>
</tr>
<tr>
<td>Chairperson – The participant is defined as the conference chairperson. The chairperson can manage the conference using touch-tone signals (DTMF codes).</td>
<td></td>
</tr>
<tr>
<td>Lecturer (AVC-based connection) – The participant is defined as the conference Lecturer.</td>
<td></td>
</tr>
<tr>
<td>Lecturer and Chairperson – The participant is defined as both the conference Lecturer and Chairperson.</td>
<td></td>
</tr>
<tr>
<td>Cascade-enabled Dial-out Participant (AVC-based connection) – A special participant functioning as a link in a cascaded conference.</td>
<td></td>
</tr>
<tr>
<td>Recording (AVC-based connection) – A special participant functioning as a Recording Link. Note: The Recording participant does not support H.264 High Profile. If recording a conference set to H.264 High Profile, the Recording participant connects as Audio Only and records the conference Audio while displaying the recording icon for the conference.</td>
<td></td>
</tr>
<tr>
<td>Request to speak (AVC-based connection) - Participants that were muted by the conference organizer/system operator can indicate that they want to be unmuted by entering the appropriate DTMF code (default 99). The icon is displayed for 30 seconds.</td>
<td></td>
</tr>
<tr>
<td>IP Address/Phone</td>
<td>The IP participant’s IP address or the ISDN/PSTN participant’s phone number.</td>
</tr>
<tr>
<td>Alias Name/ SIP Address</td>
<td>The participant’s Alias Name or SIP URI. The alias of an RSS 4000 Recording System if the participant is functioning as a recording link.</td>
</tr>
<tr>
<td>Network</td>
<td>The participant’s network connection type – H.323, or SIP or ISDN/PSTN.</td>
</tr>
<tr>
<td>Dialing Direction</td>
<td>Dial-in – The participant dialed the conference.</td>
</tr>
<tr>
<td></td>
<td>Dial-out – The MCU dialed the participant.</td>
</tr>
</tbody>
</table>
Participant Monitoring – Indicators and Properties

<table>
<thead>
<tr>
<th>Column</th>
<th>Icon/Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio</td>
<td>Displays the status of the participant’s audio channel. If the participant’s audio is connected and the channel is neither muted nor blocked, no indication is displayed.</td>
</tr>
<tr>
<td></td>
<td><strong>Disconnected</strong> – Participant’s audio channel is disconnected. This is a defined participant who is waiting to be connected to the conference.</td>
</tr>
<tr>
<td></td>
<td><strong>Muted</strong> – Participant’s audio channel is muted. Indicates who initiated the Mute: participant, Collaboration Server User or MCU. The participant can still hear the conference.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong>: If the participant muted his/her audio channel from the endpoint, the system displays the mute icon only for H.323. This icon is not displayed for SIP participant due to SIP standard limitation.</td>
</tr>
<tr>
<td></td>
<td><strong>Blocked</strong> – Transmission of audio from the conference to the participant is blocked.</td>
</tr>
<tr>
<td></td>
<td><strong>Muted and Blocked</strong> - Audio channel is muted and blocked.</td>
</tr>
<tr>
<td>Video</td>
<td>Displays the status of the participant’s video channel. If there is no problem with the participant’s video connection and the channel is neither suspended nor secondary, no indication is displayed.</td>
</tr>
<tr>
<td></td>
<td><strong>Disconnected</strong> – Participant’s video channel is disconnected. This is a defined participant who is waiting to be connected to the conference.</td>
</tr>
<tr>
<td></td>
<td><strong>Suspended</strong> – Video transmission from the endpoint to the conference is suspended.</td>
</tr>
<tr>
<td></td>
<td><strong>Secondary</strong> – Participant is connected only through the audio channel due to problems with the video channel.</td>
</tr>
<tr>
<td>Encryption</td>
<td>(AVC-based connection) Indicates that the endpoint is connected to the conference using encryption.</td>
</tr>
<tr>
<td>Service Name</td>
<td>Displays the IP Network Service used to connect this participant to the conference.</td>
</tr>
<tr>
<td>FECC Token</td>
<td>Participant is the holder of the FECC token and has Far End Camera Control capabilities. The FECC token can be allocated to only one participant at a time and remains un-allocated if no participant requests it.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong>: FECC is not supported with ISDN.</td>
</tr>
<tr>
<td>Content Token</td>
<td>Participant is the holder of the Content token and has content sharing permission. The Content token can be allocated to only one participant at a time and remains un-allocated if no participant requests it.</td>
</tr>
</tbody>
</table>

For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Participant Level Monitoring.
Video Preview (AVC-based Connection)

Video Preview enables the system users to monitor the quality of the video sent and received by the participant and identify possible quality degradation.

Guidelines

- Video preview is supported with both MPMx and MPMRx cards.
- 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 media card on different workstations (one per workstation and one per conference).


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. For more information, including workstation minimum requirements, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator's Guide.
Operations Performed During On Going Conferences

During ongoing conferences, operators and administrators can perform various operations, at the conference level affecting the entire conference or at a participant level affecting individual participants.

Conference Level operations

The following operations can be performed during an ongoing conference, affecting all the participants in the conference:

- Changing the Duration of a Conference
- Adding Participants from the Address Book
- Moving Participants Between Conferences (AVC-based Connection)
- Saving an Ongoing Conference as a Template
- Copy and Paste Conferences
- Changing the Video Layout of a Conference (AVC-Based CP and Mixed CP and SVC Conferences)
- Video Forcing (AVC-Based CP and Mixed CP and SVC Conferences)
- Enabling and Disabling Video Clarity™ (AVC-based Conferences)
- Muting all Participants Except the Lecturer (AVC-Based CP Conferences)
- Sending Text Messages to All Participants Using Message Overlay (AVC-based Conferences)
- Auto Scan (AVC-based CP Only Conferences)
- Customized Polling

Changing the Duration of a Conference

The duration of each conference is set when the new conference is created. The default duration of a conference is 1 hour. All conferences running on the MCU are automatically extended as long as there are participants connected to the conference.

A conference's *Duration* can be extended or shortened while it is running, by modifying its scheduled *End Time*.

To extend or shorten a conference manually:

1. In the Conference List pane, double-click the conference Name.
2 In the General tab, modify the **End Time** fields and click OK.

![Image of General tab with End Time fields modified]

The **End Time** is changed and the **Duration** field is updated.

**To terminate a conference manually:**

1. In the **Conferences** list, select the conference you wish to delete and click the **Delete Conference** button.
   
   You are prompted for confirmation.

2. Click **OK** to terminate the conference.
Adding Participants from the Address Book

Once the conference has started, you can add participants to a conference directly from the Participants Address Book without having to use the New Conference – Participants tab.

In SVC-based conferences, only dial-in participants can be added from the address book.

To drag & drop participants into the Participants List:

1. Open the Address Book.
2. Select, drag and drop the participant that you wish to add to the conference directly from the Participant Address Book into the Participant List.

Standard Windows multiple selection techniques can be used in this procedure.

Moving Participants Between Conferences (AVC-based Connection)

Collaboration Server users can assist participants by performing the following operations:

- Move participant from one ongoing conference to another
- Move a participant to an Operator conference (Attend a participant).
- Move a participant to the Home (destination) conference.

The move to the Operator conference is enabled only when an Operator conference is running. For more details about Operator conferences and moving participants to and from the Operator conference, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Operator Assistance & Participant Move.

You can move participants 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 to be moved.
2 In the Participants list, right-click the icon of the participant to move, and select **Move to Conference** to move the participant to any ongoing conference.

![Move to Conference dialog box](image)

The **Move to Conference** dialog box opens.

3 Select the name of the destination conference from the ongoing conferences list.

![Move to Conference dialog box](image)

**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.

**Saving an Ongoing Conference as a Template**

Any conference that is ongoing can be saved as a template.

**To save an ongoing conference as a template:**

1 In the **Conferences List**, select the conference you want to save as a Template.
2. Click the **Save Conference** 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_.

**Copy and Paste Conferences**

The Collaboration Server user can **Copy**, and **Paste** conferences. When using the *Collaboration Server Web Client*, conferences can be copied and pasted on the same MCU, however when using the *RMX Manager*, with its ability to manage multiple MCUs, conferences can be copied and pasted between different MCUs.

**Copy Conference**

The **Copy** command copies all the conference’s properties including connected participants and makes these properties available for pasting, starting a new conference. The copied conference remains active until it terminates or is deleted.

**To copy a conference:**

1. In the **Conferences List** pane, right-click the conference you want to copy.

2. In the drop-down menu select **Copy Conference**.
Paste Conference

The **Paste Conference** command starts the new conference on the same MCU or on a different MCU.

**To paste a conference:**

- Right-click in the *Conferences List* pane and in the drop-down menu select **Paste Conference**.
  - or
  - If you are using the *RMX Manager* and you want to paste the conference to a different MCU:
    a. In the *MCUs* list pane, click the MCU that is to receive the conference.
    b. In the *Conferences* list pane, right-click, and in the drop-down menu select **Paste Conference**.

![Paste Conference](image)

The conference is pasted to the MCU with a *Display Name* assigned by the system.

**Paste Conference As**

The **Paste Conference As** command allows the system user to create a new conference using the copied conference’s properties as a template. It automatically opens the *Conference Properties* dialog box allowing the user to modify the *General*, *Participants* and *Information* tabs to create the new conference. When the **OK** button in the *Conference Properties* dialog box is clicked the new conference is started.

**To paste a conference as a new conference:**

1. Right-click in the *Conferences List* pane and in the drop-down menu select **Paste Conference As**.
   - or
   - If you are using the *RMX Manager* and you want to paste the conference to a different MCU:
     a. In the *MCUs* list pane, click the MCU that is to receive the conference.
In the Conferences list pane, right-click, and in the drop-down menu select Paste Conference As.

The Conference Properties dialog box is displayed.

2 Modify the conference information as required.
3 Click the OK button to paste and start the new conference.

Changing the Video Layout of a Conference (AVC-Based CP and Mixed CP and SVC Conferences)

In SVC-base conferences, the video is automatically set to Auto Layout and cannot be modified from the Web Client. However, each SVC-based participant can select its own layout and the participants to display in the video layout cells.
In AVC-based CP and Mixed CP and SVC conferences, while the conference is running, you can change the video layout and select one of the video layouts supported by the MCU.

Video Layout selection can be done in two levels:

- **Conference Level** – Applies to all conference participants. All participants have the same video layout.
- **Participant Level** – The participant’s video layout is changed. The video layout of all other conference participants is unaffected.

The initial video layout is selected for the conference in the *Conference Profile*.

Participant level video layout selection overrides conference level video layout settings. Participants cannot change their *Personal Layouts* when *Lecturer Mode* is enabled.

**To change the video layout of a conference:**

1. In the *Conference Properties* dialog box, select *Video Settings*.
2. If the *Auto Layout* check box is selected, clear it.
3 From the Video Layout options, select the Number of Windows to display and the Video Layout thumbnail required and click OK.

Video Forcing (AVC-Based CP and Mixed CP and SVC Conferences)

You can select which participant appears in each of the video layout windows using Video Forcing. When a participant is forced to a layout window, switching between participants is suspended for that window and only the assigned participant is viewed. Video Forcing works on Conference Level or Participant Level:

- **Conference Level** – When forcing a participant to a window, all conference participants will see that participant in the selected window.
- **Participant Level** – When forcing a participant to a window, only the participant’s video layout display is affected. All other participants see the conference layout.

**Video Forcing Guidelines:**

- A participant cannot appear in two or more windows at the same time.
- Participant level video forcing overrides conference level video forcing.
- In Same Layout mode, participant can view him/herself in a layout window.
- When different size video windows are used in video layouts such as 1+2, 1+3, 1+4, etc., a participant can only be forced, in Personal Layout, to a video window of the same size as that selected for him/her in Conference Layout.
- When changing the Video Layout at the conference level, the video forcing settings are not applied to a new layout, and switching between participants is audio-activated. The video forcing setting is saved and applied the next time that layout is selected.
- Windows that are not assigned any participant display the current speaker and last speakers., but appear at random.
To video force a participant to a window:

1. In the *Conference Properties* dialog box, select the *Video Settings* tab.
2. If *Auto Layout* check box is selected, clear it.
3. Select the required video layout.
4. In the window to which you want to force a participant, select the participant’s name from the list of conference participants.
5. Repeat step 3 to force participants to other windows.
6. Click *OK*.

To cancel Video Forcing for a window:

1. In the *Conference Properties* dialog box, select the *Video Settings* tab.
2. In the video layout window, in the *Participants* list, select *Auto*.
3. Click *OK*.
4. Switching between participants is renewed and is audio activated.

**Force Layout and Preserve Participant Call**

Cells in a forced layout that were assigned to endpoints that have disconnected, have paused their video, or have been deleted from the conference will remain black until the endpoint reconnects, a new layout is used, or the conference ends.

This version adds the option to prevent the MCU from reassigning cells in a forced layout that were assigned to endpoints that have disconnected, have paused their video, or have been deleted from the conference. The cell will remain black until the endpoint reconnects, a new layout is used, or the conference ends.

A new flag, `PRESERVE_PARTY_CELL_ON_FORCE_LAYOUT`, has been added to control this option. This flag must be manually added to change its value. The values of this flag are:

- **NO** (Default) - Cells for dropped endpoints are reassigned. Endpoints that reconnect will be treated as new endpoints.
● YES - Cells for dropped endpoints will not be reassigned, but will be reserved until the endpoint reconnects.

Forced Layout Guidelines

● When a new forced layout is sent to the MCU, the MCU no longer preserves the cells for disconnected participants. The layout is redrawn using the currently connected participants only.

● If the dropped endpoint was forced to use a particular cell, and that cell is switched from forced layout to automatically assigned, the MCU no longer preserves the cell. Any other endpoint can be assigned that particular cell.

● This feature works the same way in Telepresence conferences, even where the layouts are controlled by MLA.

For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Manually Adding and Deleting System Flags.

Enabling and Disabling Video Clarity™ (AVC-based Conferences)

When enabled (default), Video Clarity applies video enhancing algorithms to incoming video streams of resolutions up to and including SD. Clearer images with sharper edges and higher contrast are sent back to all endpoints at the highest possible resolution supported by each endpoint. Video Clarity™ can disabled during an ongoing conference.

To disable or enable Video Clarity:

1. In the Conference List pane, double-click the name of the conference or right-click the conference name and then click Conference Properties.
2. Click the Video Quality tab.
3. Clear the Video Clarity check box to disable it or select it to enable it as required.
4. Click OK.

Muting all Participants Except the Lecturer (AVC-Based CP Conferences)

The Mute Participants Except Lecturer option can be enabled or disabled during the ongoing conference (in addition to its setting in the conference Profile). When 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.

Muting Participants Guidelines

● Both Administrators and Operators (users) are allowed to set the Mute Participants Except Lecturer option. It can be enabled at any time after the start of the conference to allow the conference participants to converse before the lecturer joins the conference or before they are muted.

● 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 function.
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 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 you disable the Mute Participants Except Lecturer option. When you replace a lecturer, the MCU automatically mutes the previous lecturer and unmutes the new lecturer.

- When The Mute Participants Except Lecturer option is disabled during the ongoing conference, all the participants in the conference are unmuted.
- 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.

To enable or disable the Mute Participants Except the Lecturer option:

The Mute Participants Except Lecturer option is enabled or disabled in the Conference Properties - Audio Settings dialog box:
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.

Sending Messages to All Conference Participants Using Message Overlay (AVC-based Conferences)

Messages can be sent during a conference to a single participant, selected number of participants or all participants in an ongoing conference. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide Packet Loss Compensation (LPR and DBA) AVC CP Conferences.

To send messages to all conference participants using Message Overlay:

1. In the Conferences List pane, double click the name of the conference to have Message Overlay enabled or right-click the conference name and then click Conference Properties.
2. Click the Message Overlay tab.
3. Select the Enable check box.
4. In the Content field, enter the text to be displayed to all the conference participants.
   The number of characters that can be included in a message varies according to the language and can differ due to the type of font used. For example, the available number of characters in Chinese is 32 and 48 for English and Russian.
5. Modify the remaining fields in the dialog box to display the message as required.
6. Click OK

When sending text messages using Message Overlay and stopping the text messaging in the middle, you must change (and then cancel the change) one of the message overlay properties in the Conference Properties - Message Overlay dialog box before you can re-send the same message.

To cancel the Message Overlay display:

1. Repeat Step 1 and Step 2 in the Sending Messages to All Conference Participants Using Message Overlay (AVC-based Conferences)
2 Clear the **Enable** check box.

3 Click **OK**.

**Auto Scan (AVC-based CP Only Conferences)**

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

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

For details, see *Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide*, *Auto Scan and Customized Polling in Video Layout (CP Conferences Only)*.

**Sending Text Messages to All Participants Using Message Overlay (AVC-based Conferences)**

Text messages can be sent to all participant in the conference using the **Message Overlay** options in the **Conference Properties – Message Overlay** dialog box.

To send text messages to all participants in a conference using Message Overlay:

1 In the Conferences List pane, double-click the conference entry or right-click the conference entry and then click **Conference Properties**. The **Conference Properties – General** dialog box is displayed.

2 Click the **Message Overlay** tab.

3 Click the **Enable** check box.

5 Click the OK button.

To cancel the Message Overlay display:

1 Repeat Step 1 and Step 2 in the Sending Text Messages to All Participants Using Message Overlay (AVC-based Conferences).

2 Clear the Enable check box.

3 Click OK.

Participant Level Operations

Various operations can be performed during an ongoing conference, affecting only the selected participants in the conference. These operations enable you to modify and control the connections and statuses of participants in ongoing conferences, as described in the following table.

<table>
<thead>
<tr>
<th>Participant Level Operations</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Menu Option</strong></td>
</tr>
<tr>
<td>New Participant</td>
</tr>
<tr>
<td>Add Participant From Address Book</td>
</tr>
<tr>
<td>Connect Participant</td>
</tr>
<tr>
<td>Disconnect Participant</td>
</tr>
<tr>
<td>Delete Participant</td>
</tr>
<tr>
<td>Mute Audio</td>
</tr>
<tr>
<td>Unmute Audio</td>
</tr>
<tr>
<td>Suspend Video</td>
</tr>
</tbody>
</table>
### Participant Level Operations (Continued)

<table>
<thead>
<tr>
<th><strong>Menu Option</strong></th>
<th><strong>Button</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Resume Video</td>
<td><img src="image" alt="Resume Video" /></td>
<td>Resume the participant’s video transmission to the conference. The <strong>Suspend Video</strong> button becomes active ( <img src="image" alt="Suspend Video" /> ).</td>
</tr>
<tr>
<td>Block Audio</td>
<td><img src="image" alt="Block Audio" /></td>
<td>Block the audio transmission from the conference to the participant. When blocked, the participant can still be heard by the conference. The <strong>Audio Blocked</strong> indicator appears in the Participants List and the <strong>Unblock Audio</strong> button ( <img src="image" alt="Unblock Audio" /> ) becomes active.</td>
</tr>
<tr>
<td>Unblock Audio</td>
<td><img src="image" alt="Unblock Audio" /></td>
<td>Resume the audio transmission from the conference to the participant. The <strong>Block Audio</strong> button ( <img src="image" alt="Block Audio" /> ) becomes active.</td>
</tr>
<tr>
<td>Change to Chairperson</td>
<td><img src="image" alt="Change to Chairperson" /></td>
<td>Define the selected participant as the conference leader/chairperson.</td>
</tr>
<tr>
<td>Change to Regular Participant</td>
<td><img src="image" alt="Change to Regular Participant" /></td>
<td>Define the chairperson as a regular participant without chairperson privileges.</td>
</tr>
<tr>
<td>Change To Content Token Owner</td>
<td><img src="image" alt="Change To Content Token Owner" /></td>
<td>Initiate <strong>Content Broadcast Control</strong> to prevent the accidental interruption or termination of H.239 Content that is being shared by this participant. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Content Settings.</td>
</tr>
<tr>
<td>Cancel Content Token Owner</td>
<td><img src="image" alt="Cancel Content Token Owner" /></td>
<td>Cancel <strong>Content Broadcast Control</strong>. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Content Settings.</td>
</tr>
<tr>
<td>Add Participant to Address Book</td>
<td><img src="image" alt="Add Participant to Address Book" /></td>
<td>Add selected participant’s details to the <strong>Participant Address Book</strong>.</td>
</tr>
<tr>
<td>Move to Conference (AVC Only)</td>
<td><img src="image" alt="Move to Conference (AVC Only)" /></td>
<td>Move an AVC participant to another ongoing AVC CP or Mixed CP and SVC conference. The destination conference is selected from a displayed list.</td>
</tr>
<tr>
<td>View Participant Sent Video (AVC Only)</td>
<td><img src="image" alt="View Participant Sent Video (AVC Only)" /></td>
<td>Preview the video sent from the AVC participant to the conference.</td>
</tr>
<tr>
<td>View ParticipantReceived Video (AVC Only)</td>
<td><img src="image" alt="View ParticipantReceived Video (AVC Only)" /></td>
<td>Preview the video sent from the conference to the AVC participant.</td>
</tr>
<tr>
<td>Copy Participant</td>
<td><img src="image" alt="Copy Participant" /></td>
<td>Copy the all participant's parameters in preparation for Pasting into another conference or back into the current conference.</td>
</tr>
<tr>
<td>Cut Participant</td>
<td><img src="image" alt="Cut Participant" /></td>
<td>Copy the all participant’s parameters and delete the participant from the current conference. The participant can be Pasted into another conference or back into the current conference.</td>
</tr>
<tr>
<td>Paste Participant As</td>
<td><img src="image" alt="Paste Participant As" /></td>
<td>Paste the participant into the selected conference as a new participant with parameters modified via the Address Book Participant - Properties dialog box.</td>
</tr>
<tr>
<td>Abort H.239 Session</td>
<td><img src="image" alt="Abort H.239 Session" /></td>
<td>Withdraw the Content Token from the participant back to the MCU for re-assignment.</td>
</tr>
</tbody>
</table>
### Participant Level Operations (Continued)

<table>
<thead>
<tr>
<th>Menu Option</th>
<th>Button</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connect to Website</td>
<td></td>
<td>Connect directly to the internal website of the participant’s endpoint to perform administrative, configuration and troubleshooting activities.</td>
</tr>
<tr>
<td>AGC</td>
<td></td>
<td>Enable AGC for the participant with weak audio signal during ongoing conferences. Notes:</td>
</tr>
<tr>
<td>AGC</td>
<td></td>
<td>• To implement AGC for the participant audio, the ENABLE_AGC System Flag in system.cfg must be set to YES. The flag’s default value is NO. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Managing the Address Book.</td>
</tr>
<tr>
<td>AGC</td>
<td></td>
<td>• Enabling AGC may result in amplification of background noise.</td>
</tr>
<tr>
<td>Participant Properties</td>
<td></td>
<td>View all Participant Properties. For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Participant Level Monitoring</td>
</tr>
</tbody>
</table>

### Copy Cut and Paste Participant

The Collaboration Server user can **Copy**, **Cut** and **Paste** participants between different conferences running on the MCU, including his/her current conference. These functions, when used via the RMX Manager, with its ability to manage multiple MCUs, participants, allows the MCU user to **Copy**, **Cut** and **Paste** participants between conferences running on different MCUs.

**Copy Participant**

The **Copy** command copies all the participant’s properties and makes them available for pasting. The participant remains connected to his/her current conference.

**To copy a participant:**

1. In the *Participants List* pane, right-click the participant you want to copy.
2. In the drop-down menu select **Copy Participant**.
Cut Participant

The **Cut** command copies all the participant’s properties and makes them available for *pasting*. The participant is deleted from his/her current conference.

To cut a participant:

1. In the **Participants List** pane, right-click the participant you want to cut.
2. In the drop-down menu select **Cut Participant**.

Paste Participant

The **Paste** command connects the *copied or cut* participant to the selected conference.

If the participant was copied, he/she should be deleted from the conference he/she was copied from, unless it is required that the participant is connected to two (or more) conferences. (There are endpoints that permit a participant to be connected to multiple conferences).

To paste a participant:

1. In the Conferences List pane, click the conference you want to paste the copied/cut participant into.
2. Right-click in the Participants List pane of the selected conference and in the drop-down menu select **Paste Participant**.
   
   or

   If you are using the RMX Manager and you want to paste the participant to a conference to a different MCU:
   
   a. In the MCUs list pane, click the MCU that is hosting the conference that is to receive the participant.
   
   b. In the **Conferences** list pane, click the conference you want to paste the copied/cut participant into.
   
   c. Right-click the **Participants** list pane, and in the drop-down menu select **Paste Participant**.
The participant is connected to the conference.

**Paste Participant As**

The **Paste Participant As** command allows the MCU user to create a new participant using the copied participant’s properties as a template. It automatically opens the **Address Book - Participant Properties** dialog box allowing the MCU user to modify the participant’s properties effectively creating a new participant. When the **OK** button in the **Participant Properties** dialog box is clicked the new participant is connected to the selected conference.

To paste a participant as a new participant:

1. In the **Conferences List** pane, click the conference you want to paste the copied/cut participant into.
   - Right-click in the **Participants List** pane of the selected conference and in the drop-down menu select **Paste Participant As**.
   - or
   - If you are using the **RMX Manager** and you want to paste the participant to a conference on another MCU:
     a. In the **MCUs** list pane, click the MCU that is hosting the conference that is to receive the participant.
     b. In the **Conferences** list pane, click the conference you want to paste the copied/cut participant into.
     c. Right-click the **Participants** list pane, and in the drop-down menu select **Paste Participant As**.

The **Address Book - Participant Properties** dialog box is displayed.
2 Modify the participant information as required. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Modifying Participants in the Address Book.

Optional. If not already in the Address Book, the copied/cut participant can be added to the Address Book.

Optional. The new participant can be added to the Address Book.

3 Click the OK button to connect the new participant to the selected conference.

Sending Messages to Selected Participants Using Message Overlay (AVC-based Conferences)

During an ongoing conference, text messages can be sent to selected participants (a single participant or a number of participants) using the Send Text Message to Participant right-click menu option.

To send text to selected participants:

1 In the Participant List pane, choose a participant or a number of participants.

2 In the Participant List pane, right-click a participant or a number of participants and then select Send Text Message to Participant.
The Send Text Message to Participant dialog box is displayed.

3 In the Content field, enter the text to be displayed to all the conference participants.

   The number of characters that can be included in a message varies according to the language and can differ due to the type of font used.


5 Click the OK button.

Conference Control Using DTMF Codes (AVC based Conferences)

Participants and chairpersons can manage their connection to ongoing conferences from their endpoints, using touch-tone signals (DTMF codes) from their endpoints. The following table lists the DTMF Codes that can be used.

Chairpersons can also control an ongoing conference using DTMF codes.

Permissions for DTMF actions to be performed by all conference participants or by chairperson only are configured in the Conference IVR Service assigned to the conference.

DTMF codes are not supported with SVC-based conferences.

For more information, see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Defining a New Conference IVR Service.

To use the DTMF codes to control the conference, the DTMF input must be first enabled on the endpoint remote control (for example, entering #).
Default DTMF Codes to Manage Conferences and Participants

<table>
<thead>
<tr>
<th>Operation</th>
<th>DTMF String</th>
<th>Permission</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference help (CP Only conference)</td>
<td>00</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Initiate PCM</td>
<td>##</td>
<td>Everyone</td>
</tr>
<tr>
<td>Start Click&amp;View</td>
<td>**</td>
<td>Everyone</td>
</tr>
<tr>
<td>Individual help (CP Only conference)</td>
<td>*0</td>
<td>Everyone</td>
</tr>
<tr>
<td>Pause Recording</td>
<td>*1</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Stop Recording</td>
<td>*2</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Start/Resume Recording</td>
<td>*3</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Enable Roll Call</td>
<td>*42</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Disable Roll Call</td>
<td>#42</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Roll Call Review Names</td>
<td>*43</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Roll Call Stop Review Names</td>
<td>#43</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Mute All Except Me</td>
<td>*5</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Cancel Mute All Except Me</td>
<td>#5</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Mute My Line</td>
<td>*6</td>
<td>Everyone</td>
</tr>
<tr>
<td>Unmute My Line</td>
<td>#6</td>
<td>Everyone</td>
</tr>
<tr>
<td>Increase Broadcast Volume</td>
<td>*9</td>
<td>Everyone</td>
</tr>
<tr>
<td>Decrease Broadcast Volume</td>
<td>#9</td>
<td>Everyone</td>
</tr>
<tr>
<td>Secure Conference</td>
<td>*71</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Unsecure Conference</td>
<td>#71</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Invite Participant</td>
<td>*72</td>
<td>Everyone</td>
</tr>
<tr>
<td>Disconnect Last Invited Participant</td>
<td>#72</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Increase Listening Volume</td>
<td>*76</td>
<td>Everyone</td>
</tr>
<tr>
<td>Decrease Listening Volume</td>
<td>#76</td>
<td>Everyone</td>
</tr>
<tr>
<td>Change Password</td>
<td>*77</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Change To Chairperson</td>
<td>*78</td>
<td>Everyone</td>
</tr>
<tr>
<td>Play Help Menu</td>
<td>*83</td>
<td>Everyone</td>
</tr>
<tr>
<td>Mute Incoming Participants</td>
<td>*86</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Unmute Incoming Participants</td>
<td>#86</td>
<td>Chairperson</td>
</tr>
<tr>
<td>Terminate Conference</td>
<td>*87</td>
<td>Chairperson</td>
</tr>
</tbody>
</table>
Default DTMF Codes to Manage Conferences and Participants (Continued)

<table>
<thead>
<tr>
<th>Operation</th>
<th>DTMF String</th>
<th>Permission</th>
</tr>
</thead>
<tbody>
<tr>
<td>Show Participants</td>
<td>*88</td>
<td>Everyone</td>
</tr>
<tr>
<td>Request to speak</td>
<td>99</td>
<td>Everyone</td>
</tr>
<tr>
<td>Override Mute All</td>
<td>Configurable</td>
<td>Everyone</td>
</tr>
<tr>
<td>Touch Control Prefix</td>
<td>*#</td>
<td>Everyone</td>
</tr>
</tbody>
</table>

Requesting Help (AVC CP Only Conferences)

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).

Participant who is waiting for User (Operator) Assistance is displayed with the following icons in the Status column of the Participants pane.

User Assistance Status Icons

- **Awaiting Individual Assistance** – The participant has requested the user’s (operator’s) assistance.
- **Awaiting Conference Assistance** – The participant has requested the user’s (operator’s) assistance for the conference. This usually means that the Collaboration Server user (operator) has been requested to join the conference.

Request to Speak (AVC CP Conferences)

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.

**Invite Participant (AVC-based 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.

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 RMX connects the participant when the call is answered.

**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**

- 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.

For details on enabling the Invite Participants option, see Inviting Participants using DTMF.

**Personal Conference Manager (PCM) (AVC CP Conferences)**

The Personal Conference Manager (PCM) interface enables the conference chairperson to control various conference features using his/her endpoint’s remote control device.

The following conference operations can be performed:

- Initiate Polycom’s Click&View™ application to change the local screen layout.
- Invite participants to connect to the conference.
- View and control the audio and video of each connected endpoint.
- Camera Control - control the camera of a remote endpoint using Far End Camera Control (FECC).
- Control the camera of a connected endpoint.
- Video Force a specific participant to a specific window of the video layout.
- Initiate and control recording of the conference.
- Disconnect a participant.
- Terminate the conference.

**PCM Usage Guidelines**

- PCM is supported with Collaboration Server 1500/2000/4000 MPMx and Collaboration Server 1500/1800/2000/4000 MPMRx cards.
- PCM is supported in H.323 and SIP network environments.
- PCM is only available in AVC CP conferences.
- Although the Collaboration Server can host up to a maximum of 800 conferences (depending on the RMX model) PCM can only be activated from up to 4 conferences per card on systems with MPMRx cards or with MPMx cards and a License without SVC (SVC is disabled). In systems with MPMx cards and SVC-enabled License, only 1 PCM session can be activated for one MCU.

For example, Collaboration Server 2000 with 2 MPMx media cards installed and the SVC option is disabled in the license can host a total of 8 PCM sessions.
In systems with MPMx cards, the maximum number of PCM sessions is set by a system flag `NUM_OF_PCM_IN_MPMX`. The default value of this flag is set according to the SVC license:

- If SVC is enabled in the license the default value (and the only possible value) is 1.
- If SVC is disabled in the license, the default value is 4 and the range is between 1 and 4.

If all possible PCM sessions are active on a media card and a chairperson of another conference, hosted on the same media card, attempts to start an additional PCM session, a message is displayed indicating that the requested PCM session cannot be activated because the maximum number of concurrent PCM sessions are already activated. FECC is then activated.

When PCM is active, FECC can only be performed by the chairperson using PCM’s Camera Control menu.

- FECC is available to all IP participants in the conference with the exception of the chairperson that has an active PCM session.
- If enabled, Message Overlay is not displayed while PCM is active.
- Normal conference video is resumed after 4 seconds of inactivity within the PCM menu.
- If a chairperson with active PCM session is moved to another conference, the PCM session is automatically terminated.
- PCM cannot be activated from the chairperson’s endpoint when connected with QCIF resolution.
- The `PCM_LANGUAGE` System Flag determines the language of the PCM interface.
  
  **Possible Values:** ENGLISH, CHINESE_SIMPLIFIED, CHINESE_TRADITIONAL, JAPANESE, GERMAN, FRENCH, SPANISH, KOREAN, PORTUGUESE, ITALIAN, RUSSIAN, NORWEGIAN

  **Default:** Current RMX Web Client language.

- The Start PCM DTMF code for initiating PCM is configured in the DTMF Codes tab of the New Conference IVR Service or Conference IVR Service Properties dialog box. The default DTMF Code is `##` and can be modified or if required.

The default value of the **Permission** field is **Everyone**. It can be modified to **Chairperson**.
• Although PCM is successfully initiated on Microsoft Office Communications client or Lync client, since FECC feature is not available on these clients none of the PCM options are accessible as the only available keys are #,* ,1..9 and not the navigation keys ("LEFT", "RIGHT", etc.).

System Flags

Two system flags can be used to affect PCM behavior: PCM_FECC and NUM_OF_PCM_IN_MPMX.

These System Flag must be added to the System Configuration file before it can be modified. For more information see the Polycom® RealPresence Collaboration Server (RMX) 1500/1800/2000/4000 Administrator’s Guide, Modifying System Flags.

Modifying the PCM_FECC System Flag

The PCM_FECC System Flag determines whether the DTMF Code, ##, the Arrow Keys (FECC) or both will activate the PCM interface. In addition, this flag can be used to disable the PCM.

Possible Values: YES / NO

Default: YES.

The table below summarizes the combined effect of the DTMF Code and PCM_FECC System Flag settings on the system.

<table>
<thead>
<tr>
<th>PCM_FECC</th>
<th>DTMF Code</th>
<th>Effect on System</th>
</tr>
</thead>
<tbody>
<tr>
<td>YES</td>
<td>Defined</td>
<td>Both DTMF Codes and Arrow Keys will activate PCM.</td>
</tr>
<tr>
<td></td>
<td>Deleted</td>
<td>Only the Arrow Keys can activate PCM. Once PCM is activated the Arrow Keys lose their FECC functionality.</td>
</tr>
<tr>
<td>NO</td>
<td>Defined</td>
<td>Only the DTMF Code can activate PCM. Once PCM is activated the Arrow Keys lose their FECC functionality.</td>
</tr>
<tr>
<td></td>
<td>Deleted</td>
<td>PCM is disabled.</td>
</tr>
</tbody>
</table>

Modifying the NUM_OF_PCM_IN_MPMX System Flag

The NUM_OF_PCM_IN_MPMX system flag determines the maximum number of PCM sessions that can be activated from conferences per MPMx card installed in Collaboration Server (RMX) 1500/2000/4000, depending on the system's SVC license.

• In systems with SVC-enabled license, the only possible (and default) flag value is 1.
• In systems with SVC-disabled license, possible flag values are 1 to 4. Default value is 4.

PCM Interface

Initiating PCM

For IP endpoints: PCM is activated by pressing the Far key ( ) followed by any of the Arrow Keys on the Remote Control Device.
For ISDN/PSTN endpoints: PCM is activated by pressing ## on the Remote Control Device. If, after activation, no keys are pressed on the Remote Control Device within a period of 10 seconds, PCM is automatically deactivated.

When PCM becomes active, the top level of the PCM Main Menu is displayed on the conference chairperson’s endpoint:

Remote Control Device Keys

PCM facilitates user interaction with the RMX using DTMF Codes and the Arrow (FECC) keys of the endpoint’s remote control device.
Menu Navigation - Arrow and Zoom Keys

The PCM Menu is navigated using the Remote Device’s Arrow Zoom Keys which are defined as follows:

**PCM - Menu Navigation**

<table>
<thead>
<tr>
<th>Arrow Key</th>
<th>Description</th>
</tr>
</thead>
</table>
| Left      | • Go up one menu level.  
            • Delete the character to the left of the cursor when entering data.  
            • Exit the PCM menu and return to normal conference video (from PCM the top level menu.)  
            • Select the video window to the left when navigating a video layout.  
            • Initiate PCM from a conference. |
| Right     | • Go down one menu level.  
            • Confirm current selection in the PCM menu.  
            • Select the video window to the right when navigating a video layout.  
            • Initiate PCM from a conference. |
| Up        | • Cycle upward through the displayed menu options.  
            • Select the video window above when navigating a video layout.  
            • Initiate PCM from a conference. |
| Down      | • Cycle downward through the displayed menu options.  
            • Select the video window below when navigating a video layout.  
            • Initiate PCM from a conference. |
| Zoom in (+)| • Confirm current selection in the PCM menu.  
            • Zoom in on the remote camera when using FECC. |
PCM - Menu Navigation (Continued)

<table>
<thead>
<tr>
<th>Arrow Key</th>
<th>Description</th>
</tr>
</thead>
</table>
| Zoom out (-) |  • Exit PCM Menu.  
|           |  • Zoom out on the remote camera when using FECC. |

**DTMF Codes - Numeric Keys**

Before using the shortcut number keys 0-9, enable the DTMF function of the endpoint according to that endpoint's configurations.

DTMF codes are entered using the Remote Device Numerics Keys and are defined as follows:

**PCM - DTMF Codes**

<table>
<thead>
<tr>
<th>Numeric Key</th>
<th>Description</th>
</tr>
</thead>
</table>
| 0           |  • Number input.  
|             |  • Shortcut key to a numbered menu option - when conference video or the PCM menu is displayed. |
| 1-9         |  • Number input.  
|             |  • Shortcut key to a numbered menu option - when the conference video or the PCM menu is displayed.  
|             |  • Initiate PCM session on ISDN/PSTN endpoint (1 is the default for the Start PCM DTMF Code). |
| *           |  • Initiate DTMF mode (with Polycom endpoints) - during a conference or when the PCM menu is displayed.  
|             |  • Enter a period "." - while entering an IP address. |
| #           | Confirm selection and send information to the RMX. |

**PCM Main Menu - Level 1**

**Click & View**

With the *Click&View* application, participants can change their Personal Layouts using the Arrow Keys or DTMF codes entered using the Numeric Keys of their endpoints.

For a full description of *Click&View* see *[Defining a New Conference IVR Service]*.
Invite Participant

To invite a participant to connect to the conference:

1. Use the Up/Down arrow keys to select Invite Participant in the Main Menu and then press the Right arrow or # key to confirm your selection.
   or Press the 2 key on the Remote Device.
   The Invite Participant sub menu is displayed.

2. Use the Numeric Keys to enter the number of the participant's endpoint.

3. Press the # key to initiate the call.

4. Optional.
   a. Use the Down arrow to browse to and change the selection of the dialing method from Auto to one of the following:
      ♦ H.323
      ♦ SIP
      ♦ VoIP
      ♦ ISDN
      ♦ PSTN
   b. Press the # key to initiate the call.

5. Optional.
   a. Use the Arrow Keys to browse to the Directory button and select the Alphabetic Grouping sub menu.
The Local Directory is displayed.

- Use the Numeric Keys to enter the number of the participant’s endpoint.
- Press # or the Zoom In key to initiate the call.

Optional. To display all participants:
- Use the Down arrow and the to select the ALL button.
- Use the Zoom In key to display the global directory.
- Use the Up/Down and Left/Right arrow keys to select a Participant.
- Press # or the Zoom In key to initiate the call.

Participants Mute/Status
The chairperson can view and control the audio and video status of each participant’s endpoint. The chairperson can:
- View the audio and video status of participants.
- Mute, block, unmute, unblock participant audio.
- Suspend or allow participant video.
- Mute all (except chairperson and lecturer).

To view and change the Participants Mute/Status:
1 Use the Up/Down arrow keys to browse to Participants Mute/Status entry in the Main Menu and then press the Right arrow or # key to confirm your selection.

or

Press the 3 key on the Remote Device.
The Participants Mute/Status sub menu is displayed.

![Participants Mute/Status sub menu](image)

The sub menu displays the Muted, Blocked and Video Suspended status of all participants’ endpoints. ALL can be selected to mute and unmute all participant endpoints with the exception of the conference chairperson and lecturer.

2 Use the Up/Down/Left/Right arrow keys to select a Mute, Block or Suspend status icon.

3 Use the Zoom In key to toggle the state of the audio and video channels of the participants’ endpoints.

The status icons are summarized in the table below.

**Participants Mute/Status - Icons**

<table>
<thead>
<tr>
<th>Status Icon</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><img src="image" alt="Audio output" /></td>
<td>Audio output from this endpoint is allowed. Other conference participants can hear the audio from this endpoint.</td>
</tr>
<tr>
<td><img src="image" alt="Mute" /></td>
<td>The endpoint is muted. Other conference participants will not hear audio from this endpoint.</td>
</tr>
<tr>
<td><img src="image" alt="Audio input" /></td>
<td>The audio input to this endpoint is allowed. This endpoint can hear the conference audio.</td>
</tr>
<tr>
<td><img src="image" alt="Audio output blocked" /></td>
<td>The audio output to this endpoint is blocked. This endpoint cannot receive the conference audio.</td>
</tr>
<tr>
<td><img src="image" alt="Video output" /></td>
<td>Video output from the endpoint is allowed. Other conference participants can see video from this endpoint.</td>
</tr>
<tr>
<td><img src="image" alt="Video output suspended" /></td>
<td>Video output from the endpoint is suspended. Other conference participants cannot see video from this endpoint.</td>
</tr>
<tr>
<td><img src="image" alt="Neither muted nor blocked" /></td>
<td>The endpoint is neither muted nor blocked.</td>
</tr>
<tr>
<td><img src="image" alt="Both muted and blocked" /></td>
<td>The endpoint is both muted and blocked.</td>
</tr>
</tbody>
</table>

**Far End Camera Control**

This function is not available to chairpersons using PCM with ISDN endpoints. The endpoint to be controlled must support Far End Camera Control (FECC).
Using the arrow keys of the remote control device, the chairperson can control the direction, zoom and focus of a remote endpoint’s camera.

To control a far camera:

1. Use the **Up/Down** arrow keys to select **Camera Control** in the *Main Menu* and then press the **Right** arrow or # key to confirm your selection.
   
   or
   
   Press the 4 key on the *Remote Device*.
   
   The **Camera Control** sub menu is displayed.

2. Use the **Up/Down/Left/Right** arrow keys to select the far camera to be controlled.
   
   A colored frame is displayed around the selected cell in the video layout.

3. Use the **Zoom In** key to activate **FECC**.
   
   Following a 10 second delay, the **FECC** active icon ( ) is displayed in the video image of the selected site.

4. Use the **Up/Down/Left/Right/Zoom In/Zoom Out** arrow keys to control the remote camera.

5. Use the **Zoom Out** key on the remote control to return to the *PCM Main Menu*. 

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Video Forcing

This function is not available to chairpersons using PCM with ISDN endpoints.

Video Forcing enables the chairperson to force the video of a specific participant to a specific window of the video layout.

To Video Force a participant's video:

1. Use the Up/Down arrow keys to select Video Force in the Main Menu and then press the Right arrow or # key to confirm your selection.
   or
   Press the 5 key on the Remote Device.
   The Video Force sub menu is displayed.

2. Use the Up/Down/Left/Right arrow keys to select the window of the video layout that you want the specific participant to be displayed in.

3. Press the Zoom In key to cycle the display of all participants in the selected video window until the participant you want to be displayed appears.

4. Press the Right arrow or # key to confirm your selection.

Recording

This function is not available to chairpersons using PCM with ISDN endpoints.

If a Recording Link has been set up in the Conference Profile, the conference chairperson can use the PCM interface to start, pause, resume and stop recording. If no Recording Link exists, this function appears grayed out and cannot be selected.
In x mode aA Recording or Recording Paused indication is displayed in the upper left corner of the video layout of all conference participants.

To use the recording functions:

1. Use the Up/Down arrow keys to select Recording in the Main Menu and then press the Right arrow or # key to confirm your selection.

   or

   Press the 6 key on the Remote Device.

   The Recording sub menu is displayed.

To start recording:

   » Use the Up/Down arrow keys to select Start Recording and then press the Right arrow or # key to confirm your selection.

   or

   Press the 1 key on the Remote Device.
To pause recording:

» While recording is in progress, use the Up/Down arrow keys to select **Pause** and then press the Right arrow or # key to confirm your selection.

or

Press the 2 key on the *Remote Device*.

To resume recording:

» While recording is paused, use the Up/Down arrow keys to select **Resume** and then press the Right arrow or # key to confirm your selection.

or

Press the 2 key on the *Remote Device*.

To stop recording:

» While recording is in progress, use the Up/Down arrow keys to select **Stop Recording** and then press the Right arrow or # key to confirm your selection.

or

Press the 3 key on the *Remote Device*.

**PCM Main Menu - Level 2**

**Drop Participant**

To disconnect a participant from the conference:

1. Use the Up/Down arrow keys to navigate to the second page of the *Main Menu*.

2. Use the Up/Down arrow keys to select **Drop Participant** and then press the Right arrow or # key to confirm your selection.

or

Press the 7 key on the *Remote Device*.

The **Drop Participant** sub menu is displayed.
3 Using the **Up/Down** arrow keys, select the participant to be disconnected and then press the **Right** arrow or **#** key to confirm your selection.

or

Press the *Numeric Key* on the *Remote Device* corresponding to the participant entry.

**Terminating a Conference**

> Although this function can be accessed and viewed by chairpersons using *PCM with ISDN* endpoints, conference termination cannot be executed.

**To terminate a conference:**

1 Use the **Up/Down** arrow keys to navigate to the second page of the *Main Menu*

2 Use the **Up/Down** arrow keys to select **Terminate Conference** and then press the **Right** arrow or **#** key to confirm your selection.

or

Press the **8** key on the *Remote Device*.

The *Terminate Conference* sub menu is displayed.

3 Press the **Right** arrow key to confirm termination of the conference.

or

Press the **Left** arrow key to cancel termination of the conference.
This appendix lists the terms and abbreviations that are related to the RealPresence Collaboration Server (RMX) 1500/1800/2000/4000, and are commonly used in the Collaboration Server documentation.

<table>
<thead>
<tr>
<th>Abbreviation/Term</th>
<th>Explanation</th>
</tr>
</thead>
<tbody>
<tr>
<td>AGC</td>
<td>Auto Gain Control. A mechanism that regulates noise and audio volume by keeping the received signal of all participants balanced.</td>
</tr>
<tr>
<td>Bandwidth</td>
<td>Defines the information carrying capacity of a channel. In analog systems, it is the difference between the highest frequency that a channel can carry and the lowest, measured in hertz. In digital systems, bandwidth is measured in bits per second. The larger a connection’s bandwidth, the more data can be transmitted in a given amount of time, allowing for greater video resolution and more sites in a conference. For more information, see Line Rate.</td>
</tr>
<tr>
<td>Bonding</td>
<td>Bandwidth ON Demand INterpolarity Group. A transmission protocol that aggregates two 64 Kbps B channels to function as one 128 Kbps channel. When using several BRI channels, Bonding means that only one D-channel serves all BRI channels, while the remaining D-channels are used for data transfer. See also: BRI.</td>
</tr>
<tr>
<td>Bps, Kbps</td>
<td>Bits and kilobits per second; a unit of bandwidth, which is the amount of data that can flow during one second over a communications line (using a transmission medium). 1 Kbps = 1000 Bps</td>
</tr>
<tr>
<td>BRI</td>
<td>Basic Rate Interface. A type of ISDN connection for transmitting data, consisting of 3 channels: two B-channels (each of 64 Kbps) and one D-channel (16 Kbps).</td>
</tr>
<tr>
<td>Carrier</td>
<td>A telephone or other company that provides telecommunication transmission services.</td>
</tr>
<tr>
<td>CIF, 4CIF, QCIF</td>
<td>Common Intermediate Format, an optional part of the ITU-T’s H.261 and H.263 standards. CIF specifies 288 non-interlaced luminance lines, that contain 176 pixels. CIF can be sent at frame rates of 7.5, 10, 15, or 30 per second. When operating with CIF, the amount of data to transmit cannot exceed 256 K bits (where K equals 1024). The CIF video format has the capacity to transmit video images of 352x288 pixels at 36.45 Mbps and 30 frames per second. A 4CIF format has four times the capacity of CIF; QCIF has quarter the capacity of CIF.</td>
</tr>
<tr>
<td>Codec</td>
<td>Coder-decoder. A device that converts voice and video into digital code, and vice versa. Refers to the endpoint video camera and video board that are used for video conferencing.</td>
</tr>
<tr>
<td>Abbreviation/Term</td>
<td>Explanation</td>
</tr>
<tr>
<td>-------------------</td>
<td>-------------</td>
</tr>
<tr>
<td>Conference</td>
<td>Connection between two or more endpoints exchanging video and audio information. If only two endpoints are involved, a conference is called point-to-point and no MCU is required. If more than two endpoints are involved, it is called a multipoint conference, and an MCU (Multipoint Control Unit) is required as the management system. For more information, see MCU.</td>
</tr>
<tr>
<td>CSU</td>
<td>Channel Service Unit. Customer-provided equipment that is used as an interface between a communication network and the data terminal.</td>
</tr>
<tr>
<td>DBA</td>
<td>Dynamic Bandwidth Allocation. Used to allocate the bandwidth needed to transmit the additional packets for LPR.</td>
</tr>
<tr>
<td>DTMF</td>
<td>Dual Tone Multi Frequency. A system of coded signals used by touch-tone telephones in which a specific sound, frequency or tone is assigned to each key so that the signal can be easily recognized by a computer. The codes enable data input and control of voice-processing systems. DTMF signals can pass through the entire connection to the destination device and therefore are used for remote control after the connection with the MCU is established.</td>
</tr>
<tr>
<td>E1 Line</td>
<td>A 2Mb digital switched line used in Europe.</td>
</tr>
<tr>
<td>Endpoint</td>
<td>A hardware device, or set of devices, that can call, and be called by an MCU or another endpoint. For example, an endpoint can be a phone, a camera and microphone connected to a PC or an integrated Room System (conferencing system).</td>
</tr>
<tr>
<td>FECC</td>
<td>Far End Camera Control. In certain video cameras, the accompanying software that enables a participant to control a remote camera. Used in Continuous Presence video conferences in conjunction with the LSD option. For more information, see LSD.</td>
</tr>
<tr>
<td>Frame</td>
<td>A group of bits that make up an elementary block of video data for transmission by certain protocols.</td>
</tr>
<tr>
<td>Frame Rate</td>
<td>The number of video frames displayed on-screen during one second, measured in fps (frames per second).</td>
</tr>
<tr>
<td>G.711</td>
<td>ITU-T audio algorithm, 64Kbps, 3.4 kHz.</td>
</tr>
<tr>
<td>G.722</td>
<td>ITU-T audio algorithm, 64Kbps, 7 kHz.</td>
</tr>
<tr>
<td>G.728</td>
<td>ITU-T audio algorithm, 16Kbps, 3.4 kHz.</td>
</tr>
<tr>
<td>Gatekeeper</td>
<td>A type of server that performs two main functions: translates LAN alias addresses of terminals and gateways to IP addresses and provides bandwidth management.</td>
</tr>
<tr>
<td>H.221</td>
<td>ITU-T standard that defines how to multiplex video, audio, and user data into one serial bit stream.</td>
</tr>
<tr>
<td>H.230</td>
<td>ITU-T standard that defines simple multipoint control systems procedures and describes network maintenance functions.</td>
</tr>
<tr>
<td>H.231</td>
<td>ITU-T standard that defines a set of MCU functions and operational requirements.</td>
</tr>
<tr>
<td>H.242</td>
<td>ITU-T standard that defines initiation of communications between systems and capabilities negotiation procedures.</td>
</tr>
<tr>
<td>H.243</td>
<td>ITU-T standard that defines initiation of communications between systems and capabilities negotiation procedures in multipoint conferences.</td>
</tr>
<tr>
<td>Abbreviation/Term</td>
<td>Explanation</td>
</tr>
<tr>
<td>------------------</td>
<td>--------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>H.261</td>
<td>ITU-T standard that defines the Px64 video coding algorithm.</td>
</tr>
<tr>
<td>H.263</td>
<td>ITU-T standard that provides improved compression and quality of video images at a line rate lower than 384 Kbps. This standard is not supported by all codecs.</td>
</tr>
<tr>
<td>H.264*</td>
<td>A proprietary Polycom Video compression standard.</td>
</tr>
<tr>
<td>H.264</td>
<td>ITU-T standard that provides improved compression and quality of video images in lower line rate connections and is part of the Highest Common mechanism in Video Switching conferences.</td>
</tr>
<tr>
<td>H.320</td>
<td>ITU-T standard that defines how the H-series video conferencing recommendations work together.</td>
</tr>
<tr>
<td>H.323</td>
<td>ITU-T standard for audio, video and data communications across IP-based (LAN) networks, including the Internet.</td>
</tr>
<tr>
<td>ICE</td>
<td>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).</td>
</tr>
<tr>
<td>IP</td>
<td>Internet Protocol. The working protocol that forms the basis of the internet.</td>
</tr>
<tr>
<td>ISDN</td>
<td>Integrated Services Digital Network. A set of protocol and interface standards (voice, video and data) that comprise a telephone network. There are two types of ISDN lines: BRI and PRI.</td>
</tr>
<tr>
<td>ITU-T Standard</td>
<td>International Telecommunications Union, Telecommunication Standardization Sector (formerly CCITT). An international group that produces official standards for telecommunications.</td>
</tr>
<tr>
<td>LAN</td>
<td>Local Area Network. A group of computers and other devices linked via a network’s operating system.</td>
</tr>
<tr>
<td>Line Rate</td>
<td>The amount of bandwidth used by a communication device, measured in Kbps (kilobits per second).</td>
</tr>
<tr>
<td>LPR</td>
<td>Lost Packet Recovery. An algorithm that creates additional packets that contain recovery information necessary to reconstruct lost packets.</td>
</tr>
<tr>
<td>MCU</td>
<td>Multipoint Control Unit. Device which allows more than two sites to be connected in a video conference.</td>
</tr>
<tr>
<td>Null modem cable</td>
<td>A serial cable designed to eliminate the need for communication equipment when two digital devices are directly connected to each other.</td>
</tr>
<tr>
<td>Participant</td>
<td>A person using an endpoint to connect to a conference. When using a Room System, several participants use a single endpoint.</td>
</tr>
<tr>
<td>PRI</td>
<td>Priority Rate Interface. An ISDN interface designed for high volume data communication. Consists of 23 B channels of 64 Kbps each and one D channel of 64 Kbps. In Europe, the PRI line provides 30 B channels + one D channel.</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network.</td>
</tr>
<tr>
<td>Abbreviation/Term</td>
<td>Explanation</td>
</tr>
<tr>
<td>-------------------</td>
<td>-------------</td>
</tr>
<tr>
<td>QCIF</td>
<td>Quarter CIF. A video format with image size of 176x144 pixels that transmits 9.115 Mbps at 30 frames per second (a quarter of the capacity of CIF). For more information, see CIF.</td>
</tr>
<tr>
<td>QoS</td>
<td>Quality of Service. QoS defines the performance of a network service, such as the average delay between packets.</td>
</tr>
<tr>
<td>RS-232</td>
<td>A standard for serial interface connection.</td>
</tr>
<tr>
<td>RTV</td>
<td>Real Time Video protocol provides high quality video conferencing capability to Microsoft OCS (Office Communicator Server) endpoints.</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol. An application-layer protocol designed to work over IP networks. A SIP service defines the properties and the IP addresses of the SIP network components.</td>
</tr>
<tr>
<td>Span</td>
<td>An ISDN line or leased line. A span may be of either T1 (United States) or E1 (Europe) type. Also called a circuit.</td>
</tr>
<tr>
<td>SRTP</td>
<td>Encryption of SIP Media is supported using SRTP (Secured Real-time Transport Protocol) and the AES key exchange method.</td>
</tr>
<tr>
<td>T1 Line</td>
<td>A 1.5Mb digital switched line used in the United States.</td>
</tr>
<tr>
<td>ToS</td>
<td>Type of Service. ToS defines optimization tagging for routing audio and video packets.</td>
</tr>
<tr>
<td>WAN</td>
<td>Wide Area Network. A communications network that services a geographical area larger than the LAN.</td>
</tr>
<tr>
<td>Whiteboard</td>
<td>An on-screen shared notebook for placement of shared documents.</td>
</tr>
</tbody>
</table>
Troubleshooting

Collaboration Server Web Client Installation - Troubleshooting Instructions

Use of the RMX Web Client is not recommended in Maximum Security Environments. Management using the RMX Manager is the recommended method.

If a Browser Environment Error occurs, close all the Internet Explorer sessions and reconnect to the MCU. If the problem persists, you can run the Automatic Troubleshooting Utility or perform the Troubleshooting Procedures manually.

The Manual Troubleshooting Procedures include several procedures that can be performed in order to solve the connection error. At the end of each procedure, check if you can connect to the MCU and if the problem persists, perform the next procedure.

In Secured Mode (https://), the DNS name specified in the Collaboration Server’s Certificate must correspond with that of the DNS Server used by the Client that is connecting to the RMX.
The following troubleshooting procedures can be performed manually:

- Procedure 1: Ending all Internet Explorer Sessions
- Procedure 2: Deleting the Temporary Internet Files, Collaboration Server Cookie and Collaboration Server Object
- Procedure 3: Managing Add-ons Collisions
- Procedure 4: Add the Collaboration Server to the Internet Explorer Trusted Sites List
- Procedure 5: Browser Hosting Controls (Optional)

Procedure 1: Ending all Internet Explorer Sessions

In some cases, although all the Internet Explorer sessions were closed, the system did not end one or several IE processes. These processes must be ended manually.

To end all Internet Explorer sessions:

Start the Task Manager and click the Processes tab.

1. Select an `iexplore` process and click the End Process button.

2. Repeat this process for all `iexplore` processes that are currently active.

3. Close the Windows Task Manager dialog box.

4. Open the Internet Explorer and connect to the MCU.

5. If the problem persists, continue with Procedure 2.

Procedure 2: Deleting the Temporary Internet Files, Collaboration Server Cookie and Collaboration Server Object

If at the end of Procedure 1 the error message is still displayed, and you cannot connect to the MCU, perform the following operations:
Delete the Temporary Internet files
Delete the RMX/Collaboration Server Cookie
Delete the RMX/Collaboration Server ActiveX Object

Deleting the Temporary Internet Files

To delete the Temporary files:

1. In the Internet Explorer, click Tools > Internet Options. The Internet Options dialog box opens.
2. In the Browsing history pane, click the Delete button.

The Delete Browsing History dialog box opens.
3 It is recommended to delete only the **Temporary Internet files**. By default, the **Cookies** option is also selected. Clear it if you do not want to clear the cookies from your computer.

4 Click the **Delete** button.

5 When the process is complete, the system return to the **Internet Options** dialog box.
Deleting the RMX/Collaboration Server Cookie

To delete the Collaboration Server Cookie:

1. In the Internet Options dialog box - Browsing History pane, click the Settings button.

The Temporary Internet Files and History Settings dialog box is displayed.
2 Click the View files button.

![Image of Windows Explorer]

The Windows Explorer screen opens, listing Windows Temporary Internet Files.

3 Browse to the RMX/ Collaboration Server cookie. The cookie is listed in the format: cookie:user name@Collaboration Server/RMX IP address. For example: cookie:valerie@172.22.189.110.

4 Right-click the Collaboration Server cookie and click Delete.

![Image of right-clicking and deleting]

The system prompts for confirmation.

5 Click Yes. The cookie is deleted.

6 Close the Windows Explorer screen.
Deleting the RMX/Collaboration Server ActiveX Object

To delete the RMX/Collaboration Server ActiveX Object:

1. In the Temporary Internet Files and History Settings dialog box, click the View objects button.

The Windows Explorer screen opens, listing the Windows Downloaded Program Files.

2. Right-click the EMA.ClassLoader.dll and then click Delete.

The system prompts for confirmation.

3. Click Yes.

The Collaboration Server object is deleted.
4 Close the Windows Explorer screen.
5 In the **Temporary Internet Files and History Settings** dialog box, click **OK**.
6 In the **Internet Options** dialog box, click **OK** to close it.
7 Close the Internet Explorer session and reopen it.
8 Connect to the Collaboration Server.

If the problem persists, continue with Procedure 3.

**Procedure 3: Managing Add-ons Collisions**

In some cases, previously installed add-ons, such as anti virus programs can create collisions between applications and prevent the installation of a new add on. Disabling these add-ons may be required in order to install the Collaboration Server Web Client.

**To disable an add-on:**

1 In the Internet Explorer, click **Tools > Manage Add-ons**. The **Manage Add-ons - Toolbars and Extensions** dialog box opens.
2 Scroll to the add-on to disable (for example, the anti virus add-on), right-click it and then click **Disable**.
   Alternatively, select the add-on and click the **Disable** button.
3 Click the **Close** button to close this dialog box.
4 Connect to the Collaboration Server.

If the problem persists, continue with the Procedure 4.
Procedure 4: Add the Collaboration Server to the Internet Explorer Trusted Sites List

In some cases, local security settings may prevent Internet Explorer from accessing the Collaboration Server.

To add the Collaboration Server to the Internet Explorer Trusted Sites list:

1. In the Internet Options dialog box, click the Security tab. The Security tab is displayed.

2. Click the Trusted Sites tab.

3. Click the Sites button. The Trusted sites dialog is displayed.

4. If the Collaboration Server is using Secure Mode:
In the **Add this website to the zone:** field, enter, “https://” followed by the IP address or the DNS name of the Collaboration Server.

**b** Click the **Add** button.

**c** Click the **Close** button.

**5** If the Collaboration Server is using Standard Security Mode:

**a** In the **Add this website to the zone:** field, enter, “https://” followed by the IP address or the DNS name of the Collaboration Server.

**b** Click the **Add** button.

**c** Clear the **Require server verification (https:)** for all sites in this zone checkbox.

**d** Click the **Close** button.

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**Procedure 5: Browser Hosting Controls (Optional)**

If the **Collaboration Server Web Client** does not load and run after Procedures 1-4 have been performed, the reason may be that .NET Framework 4 or higher is running on the workstation with **Managed Browser Hosting Controls** disabled.

**Managed Browser Hosting Controls** is an Internet Explorer operating mode required by the **Collaboration Server Web Client**. By default, **.NET Framework 4** and higher are not enabled to support **Managed Browser Hosting Controls**.

Perform Procedure 5 to:

- Determine whether **.NET Framework 4** or higher is running on the workstation.

- Determine whether a **32-bit or 64-bit version of Windows** is running on the workstation.

- Enable **Managed Browser Hosting Controls** if **.NET Framework 4** or higher is running on the workstation.

**To enable Managed Browser Hosting Controls:**

1. Determine whether **.NET Framework 4** or higher is running on the workstation.

   **a** On the **Windows Desktop**, click **Start**.

   **b** In the **Start Menu**, click **Control Panel**.

   **c** In the **Control Panel**, click **Programs and Features**.

   **d** Inspect the **Programs and Features** list for the version of **Microsoft .NET Framework Client Profile** that is installed.

2. Determine whether a **32-bit or 64-bit version of Windows** is running on the workstation:

   **a** On the **Windows Desktop**, click **Start**.

   **b** In the **Start Menu**, click **Computer**.

   **c** In the **Computer Menu, System properties** and inspect the value of the System type field in the **System section**.

3. Enable **Managed Browser Hosting Controls** if **.NET Framework 4** or higher is running on the workstation.

   **a** Open the **Registry**.

   **b** Navigate to the **Subkey**:

   ♦ **32-bit System**: HKEY_LOCAL_MACHINE\SOFTWARE\MICROSOFT\.NETFramework
64-bit System:
HKEY_LOCAL_MACHINE\SOFTWARE\Wow6432Node\Microsoft\.NETFramework

- Add the **Dword Value: EnableIHosting**
- Set value of **EnableIHosting** to 1.
- Close the **Registry**.
- Close and re-open **Internet Explorer**.