



Polycom RMX™ 2000

Getting Started Guide

Version 4.0



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Catalog No. DOC2224A
Version 4.0

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PATENT PENDING

Regulatory Notices

United States Federal Communication Commission (FCC)

Part 15: Class A Statement. This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. Test limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio-frequency energy and, if not installed and used in accordance with the instruction manuals, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his or her own expense.

Part 68: Network Registration Number. This equipment is registered with the FCC in accordance with Part 68 of the FCC Rules. This equipment is identified by the FCC registration number.

If requested, the FCC registration Number and REN must be provided to the telephone company.

Any repairs to this equipment must be carried out by Polycom Inc. or our designated agent. This stipulation is required by the FCC and applies during and after the warranty period.

United States Safety Construction Details:

- All connections are indoor only.
- Unit is intended for RESTRICTED ACCESS LOCATION.
- Unit is to be installed in accordance with the National Electrical Code.
- The branch circuit overcurrent protection shall be rated 20 A for the AC system.
- This equipment has a maximum operating ambient of 40°C, the ambient temperature in the rack shall not exceed this temperature.

To eliminate the risk of battery explosion, the battery should not be replaced by an incorrect type.

Dispose of used batteries according to their instructions.

CE Mark R&TTE Directive

Polycom Inc., declares that the Polycom RMX™ 2000 is in conformity with the following relevant harmonized standards:

EN 60950-1:2001

EN 55022: 1998+A1:2000+A2:2003 class A

EN 300 386 V1.3.3: 2005

Following the provisions of the Council Directive 1999/CE on radio and telecommunication terminal equipment and the recognition of its conformity.

Canadian Department of Communications

This Class [A] digital apparatus complies with Canadian ICES-003.

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets telecommunication network protective, operational and safety requirements as prescribed in the appropriate Terminal Equipment Technical Requirements document(s). The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations. Repairs to certified equipment malfunctions, may give the telecommunications company causes to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

Regulatory Notices

Chinese Communication Certificate

声 明

此为 **A** 级产品，在生活环境中，该产品可能会造成无线电干扰。在这种情况下，可能需要用户对其干扰采取切实可行的措施。

Singapore Certificate

RMX 2000 complies with IDA standards G0916-07

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System Overview

This Getting Started Guide provides information on the installation and basic operation of your RMX system.



Chairpersons and Operators (users who start and manage conferences for other users) please read:

- *Chapter 1 – System Overview*
- *Chapter 3 – Basic Operation*

System Administrators please read:

- *Chapter 1 – System Overview*
- *Chapter 2 – First Time Installation and Configuration*
- *Chapter 3 – Basic Operation*

For more information on configuring and managing the system, refer to the *RMX 2000 Administrator's Guide* included with the system.

RMX 2000

The Polycom RMX 2000 Multipoint Control Unit (MCU) is a high performance, scalable, IP-network (H.323 and SIP) and PSTN solution that provides the user with feature-rich and easy-to-use multipoint voice and video conferencing.

The RMX MCU meets International Telecommunication Union - Telecommunication Standardization Sector, (ITU-T, formerly CCITT) standards for multipoint multimedia bridging devices, and meets ETSI standards for telecommunication products.

The RMX unit has, in addition, been designed in compliance with IETF (Internet Engineering Task Force) – a large open international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet.

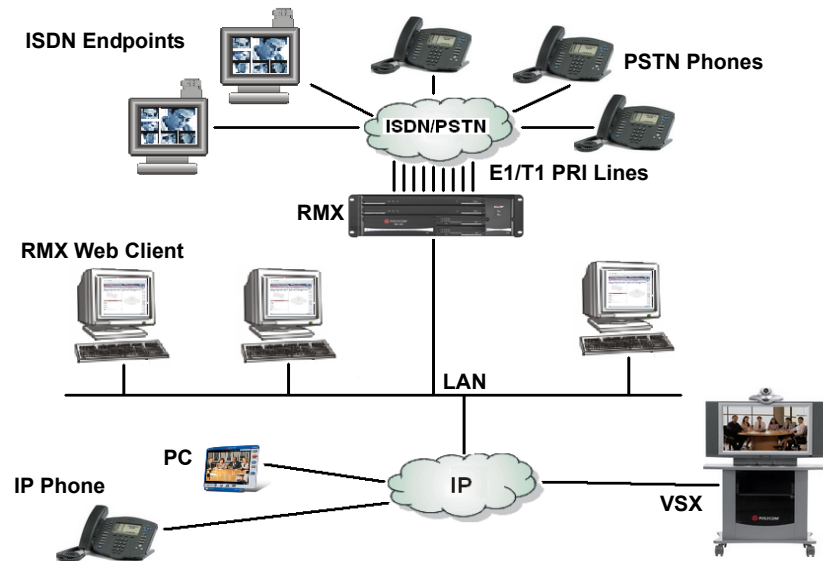


Figure 1-1 *Multipoint Video Conferencing using a Polycom RMX 2000*

The Polycom RMX 2000 unit is controlled, via the LAN, by the Polycom RMX 2000 Web Client application, using Internet Explorer® installed on the user's workstation.

Both RMX management and IP conferencing are performed via a single LAN port. A maximum of two RTM ISDN cards are supported, each providing connection for up to either 7 E1 or 9 T1 PRI lines (E1 and T1 connections cannot be used simultaneously).

RMX Main Features

Conferencing Modes

Dynamic Continuous Presence

The dynamic Continuous Presence (CP) capability of the RMX system enables viewing flexibility by offering multiple viewing options and window layouts for video conferencing. By default every conference, Entry Queue and Meeting Room has the ability to declare the maximum CP resolution as defined for the system. This includes conferences launched by the *RMX Web Client* and conferences started via the API.

CP mode conferencing is defined by:

- Conference profile settings:
 - Conference Line Rate.
 - Video Quality selection – Motion or Sharpness.
- Endpoint Capabilities – Participants can connect at differing line rates using endpoints with differing capabilities.

Video Layouts in CP

Twenty-four layouts are available to accommodate different numbers of participants and conference settings. The VUI annex to the H.264 protocol for endpoints that transmit wide video format instead of 4CIF resolution is also supported.

Table 1-1 Continuous Presence – Video Layouts


















			
			

Table 1-1 Continuous Presence – Video Layouts (Continued)

Telepresence Mode

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 RMX 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.

Multiple Switching Modes

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
- RMX 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

High Definition Video Switching

In High Definition Video Switching (HD VSW) mode all participants see the same video picture (full screen). Only one CIF video resource is used for each connection.

HD VSW mode conferencing is defined by:

- Conference *Line Rate* setting in the conference profile
 - All endpoints must connect to the conference at the same *Line Rate*. Line rates range from 384kbp to 6Mb. The RMX will always connect participants at the highest possible video quality the is supported by the conference *Line Rate*.
- Endpoint capabilities
 - Compliant endpoints can connect to conferences at resolutions of up to 1920 x 1080 pixels (1080p).

The video quality for the conference is determined by the *Highest Common Mechanism* and is based on the endpoint with the lowest capabilities that is connected to the conference. The *Highest Common Mechanism* enables the system to dynamically select the best video quality as endpoints connect to and disconnect from the conference.

Video Resolutions

High Definition (HD) Video Resolution

HD is an ultra-high quality video resolution. Depending on the RMX's *Card Configuration mode*, compliant endpoints are able to connect to conferences at resolutions ranging from 720p (1280 x 720 pixels) to 1080p (1920 x 1080 pixels) (in *MPM+ Mode*) at bit rates ranging from 1024 Kbps to 4 Mbps (6 Mbps with HD VSW).

Standard Definition (SD) Video Resolution

SD is a high quality video protocol which uses the H.264 video algorithm. It enables HD compliant endpoints to connect to conferences at resolutions of 720x576 pixels for PAL systems and 720x480 pixels for NTSC systems. Bit rates for SD range from 256 Kbps to 2 Mbps.

4CIF Video Resolution

4CIF (704 x576 pixels) resolution in H.263 endpoints is supported for conferences in which the video quality is set to sharpness and for lines rates of 384 Kbps to 1920 Kbps.

Video Clarity™

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 in *MPM+ Mode*.

H.239 / People+Content

The H.239 protocol allows compliant endpoints to share content. By default, all Conferences, Entry Queues, and Meeting Rooms launched on the RMX have H.239 capability. This protocol is also supported in MIH Cascading conferences.

People+Content is Polycom's proprietary equivalent of H.239.

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 and remote control 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:

- Manually terminate the conference
- Mute or unmute the participant's audio channel
- Adjust the participant's broadcasting and listening audio volume
- 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
- Secure and unsecure a conference.

Entry Queue

An Entry Queue is a special routing lobby for video and audio participants. After dialing the Entry Queue ID or 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.

Conferencing Capabilities and Options

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 – no setup, a new conference can be created when a user dials in and enters a conference ID that is not being used by an existing conference or Meeting Room.

Scheduled Conferencing / Reservations

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.

Connection Methods

- Dial-out: automatically, to pre-defined participants (line rate detection is automatic)

- Dial-in:
 - for participants defined in advance (IP participants only)
 - for undefined participants directly to a conference (IP participants only)
 - for undefined participants via a single dial Entry Queue (IP and ISDN/PSTN)

Cascading Conferences

- Simple Cascading (Star Topology).
- Multi Hierarchy Cascading (MIH).

Security

- Media Encryption (IP only), 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 RMX system.

Conference Management and Monitoring Features

The Polycom RMX 2000 Web Client provides capabilities for management and monitoring of participants and conferences, including the following:

- Lecture Mode or Presentation Mode in Continuous Presence conferences.
- Far End Camera Control (FECC/LSD) in video conferences.
- Automatic termination of idle (no participants) conferences.
- Automatic extension of conference duration.
- Control of listening and broadcasting audio volume for individual participants.
- 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.
- Media encryption.
- Active display of all conferences and participants with option to limit display in secured conferences.
- Real-time monitoring of each participant's connection status and properties.
- Multiple drag & drop of participants.
- Easily accessed Call Detail Records (CDR) for administrator.
- Active display of all system resources.

Card Configuration Modes

Two *Card Configuration Modes* are supported:

- **MPM Mode** – Supported with *MPM cards* on the current, and all previous RMX versions.
- **MPM+ Mode** – Supported from Version 4.0, with *MPM+ cards* installed in the RMX. It offers:
 - Two *Video/Voice Resource Capacity* resource allocation modes for increased control over system resource allocation.
 - Enhanced *Resource Report* for more accurate system management.
 - Additional video resolutions and video quality.

Workstation Requirements

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

- **Minimum Hardware** – Intel® Pentium® III, 1 GHz or higher, 1024 MB RAM, 500 MB free disk space.
- **Workstation Operating System** – Microsoft® Windows® XP, Vista®
- **Network Card** – 10/100 Mbps.
- **Web Browser** – Microsoft® Internet Explorer® Version 6 or higher.



.Net Framework 2.0 is required and installed automatically.
If ActiveX installation is blocked please see the *RMX 2000 Administrator's Guide*, "ActiveX Bypass" on page [14-78](#).

Prerequisites

This manual assumes the user has the following knowledge:

- Familiarity with Windows® XP or Vista® operating systems and interface.
- Familiarity with Microsoft® Internet Explorer® Version 6 or higher.
- Basic knowledge of video conferencing concepts and terminology.

First Time Installation and Configuration

First Time Installation and Configuration of the RMX 2000 consists of the following procedures:

1 Hardware Installation and Setup

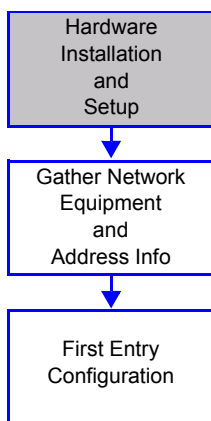
- Mount the RMX in a rack.
- Connect the necessary cables.

2 Gather Network Equipment and Address Information

- Get the information needed for integrating the RMX into the local network.

3 First Entry Configuration

- Register the RMX.
- Power up the RMX.
- Modify the *Management Network*.
- Configure the *Default IP Network Service*.
- Configure the *ISDN/PSTN Network Service*.



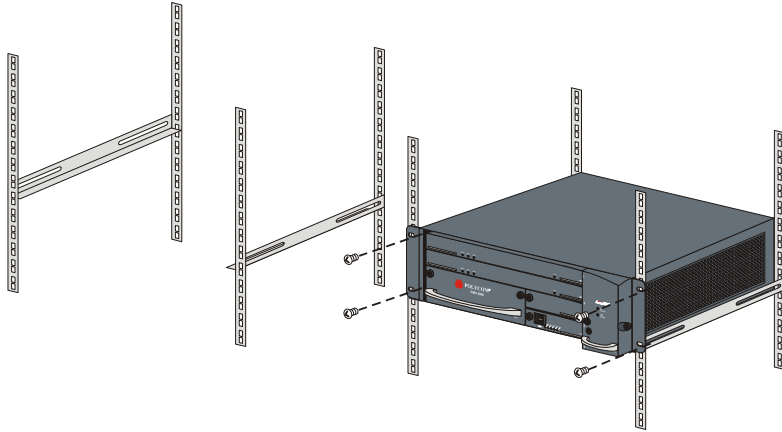
Hardware Installation and Setup

The RMX unit should be mounted in a 19" rack in a well ventilated area. It is important to adhere to the *Site Requirements* as described in the *RMX 2000 Hardware Guide*, "Site Requirements" on page **1-3**.

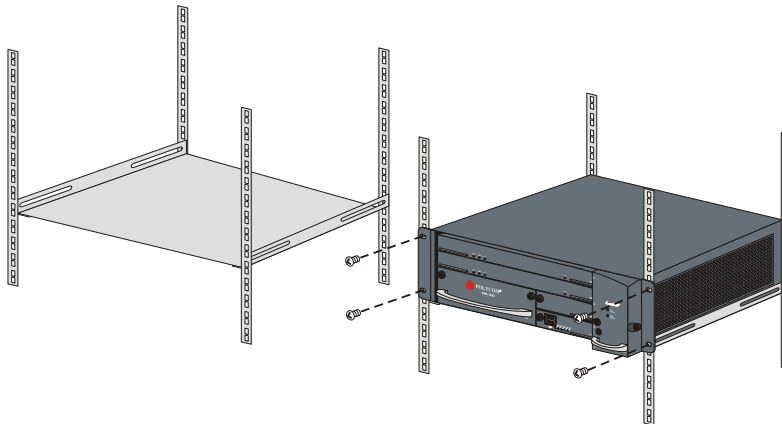
Mounting the RMX in a Rack

There are two methods for installing the RMX in a rack:

- **Using rack brackets** – Install rack brackets, supplied by the rack manufacturer, in the rack. Mount the RMX on top of the rack brackets. Fasten the RMX to the rack with screws through the four holes in the RMX's front mounting brackets.



- **Using a shelf** – Install the shelf, supplied by the rack manufacturer, in the rack. Mount the RMX on the shelf. Fasten the RMX to the rack with screws through the four holes in the RMX's front mounting brackets.



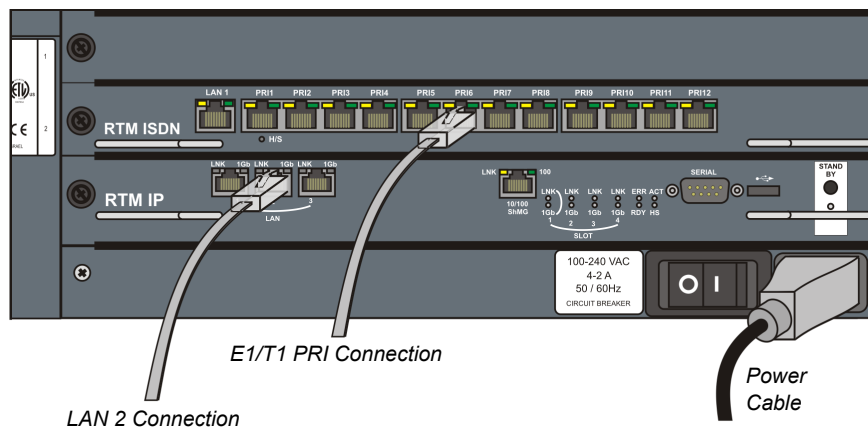
Connecting Cables



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

Connect the following cables to the back panel:

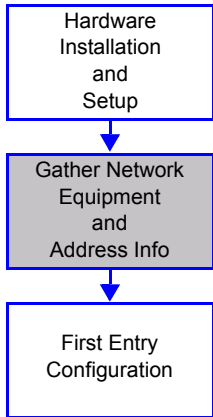
- Power cable
- LAN cable to **LAN 2** Port
- E1/T1 Cables to **PRI** Ports



To maximize conferencing performance, especially in high bit rate call environments, a 1Gb connection is recommended.

Gather Network Equipment and Address Information

IP Services



The IP addresses and network parameters which enable communication between the RMX, its management application and the conferencing devices are organized in two IP services:

- **Management Network (Control Unit)**
- **Default IP Service (Conferencing Service)**

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 RMX *Control Unit* and the RMX *Web Client* and is used to manage the RMX.

The RMX is shipped with default IP addresses as listed in Table 2-1.

Default IP Service (Conferencing Service)

The *Default IP Service (Conferencing Service)* is used to configure and manage communications between the RMX and conferencing devices.

IP Network Services Required Information

When installing an RMX unit, these default IP addresses must be modified to your local network settings. Therefore it is important that before powering the RMX unit up for the first time, that you obtain the information needed to complete the **Local Network Settings** section of the table from your network administrator. The network administrator should allocate four IP addresses in the local network for an MCU with one MPM card and five IP addresses for an MCU with two MPM cards.

Table 2-1 Network Equipment and Address Information

Parameter	Factory Default	Local Network Settings
Control Unit IP Address	192.168.1.254	

Table 2-1 Network Equipment and Address Information (Continued)

Parameter	Factory Default	Local Network Settings
<i>Control Unit Subnet Mask</i>	255.255.255.0	
<i>Default Router IP Address</i>	192.168.1.1	
<i>Shelf Management IP Address</i>	192.168.1.252	
<i>Signaling Host IP address</i>	—	
<i>Media Board IP address (MPM 1)</i>	—	
<i>Media Board IP address (MPM 2)</i>	—	
<i>Gatekeeper IP address (optional)</i>	—	
<i>DNS IP address (optional)</i>	—	
<i>SIP Server IP address (optional)</i>	—	

ISDN/PSTN Services

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 RMX.

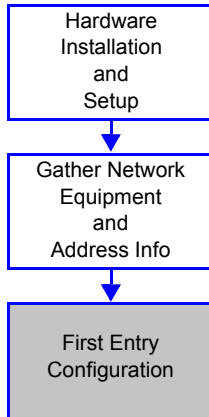
Before configuring the ISDN/PSTN Network Service, obtain the following information from your ISDN/PSTN Service Provider:

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



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

First Entry Configuration



There are four procedures necessary for setup of the new RMX. It is important that they are performed in the following sequence:

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

Procedure 1: Product Registration

Before the RMX can be used, it is necessary to register the product and obtain an *Activation Key*.

During first-time power-up, the *Product Activation* dialog box is displayed, requesting you to enter an *Activation Key*.

Obtaining the Activation Key

- 1 Access the *Service & Support* page of the Polycom website at: <http://portal.polycom.com>
- 2 Login 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 RMX's serial number is on a sticker on the back of the unit, if needed.)
- 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 2: Modifying the Factory Default Management Network Settings

Management Network Definition

Management Network Definition can be done by two methods:

- **USB key (recommended method)** – The system is shipped with a *USB key* containing the default IP addresses for the control unit and the shelf management.

These defaults are first modified in the PC and then uploaded to the RMX.

- **Direct connection** – Creating a private network between the RMX and the computer and modifying the management network parameters using *Fast Configuration Wizard* in the *RMX Web Client*.

For more information, see the *RMX 2000 Administrator's Guide*, "Configuring Direct Connections to RMX" on page [H-1](#).

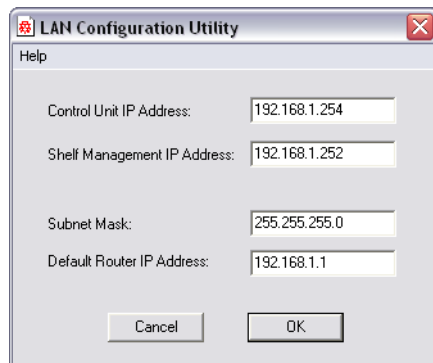
Modifying the USB Key Settings

The *USB key* 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*, also on the *USB key*.

To modify the USB key settings:

- 1 Insert the *USB key* into the PC workstation and double-click **LanConfigUtility.exe** to start the utility.

The *LAN Configuration Utility* dialog box opens.

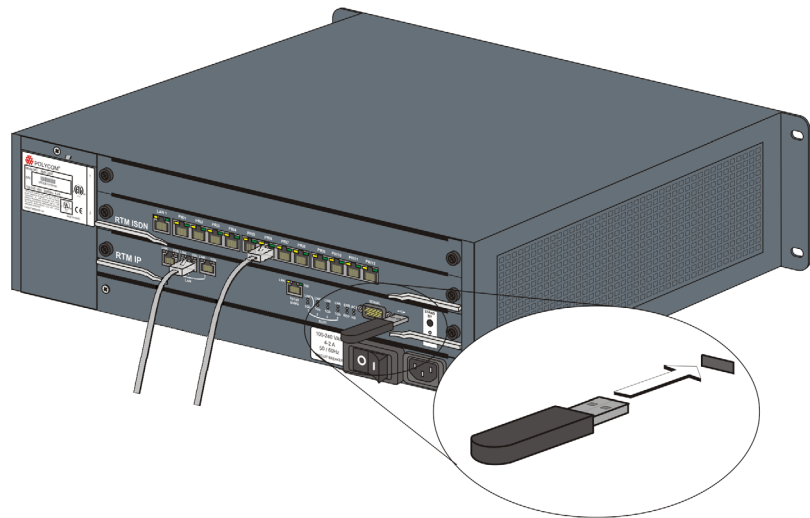


- 2 Modify the following parameters in the utility's dialog box using the information supplied by your network administrator.
 - Control Unit IP Address
 - Shelf Management IP Address
 - *Subnet Mask*
 - *Default Router IP Address*
- 3 Click **OK**.

Procedure 3: First-time Power-up and Connection to MCU

To power-up for the first time using the USB key:

- 1 Insert the *USB key* containing the modified IP addresses in USB port on the RMX's back panel.



- 2 Power the RMX **On**.

The parameters in the *lan.cfg* file are uploaded from the USB key to the RMX's memory and applied during the power-up sequence.

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

During the First-time Power-up the red **ERR** LED on the RMX's front panel remains **ON** until both the *Management* and *IP Network Services* have been defined.

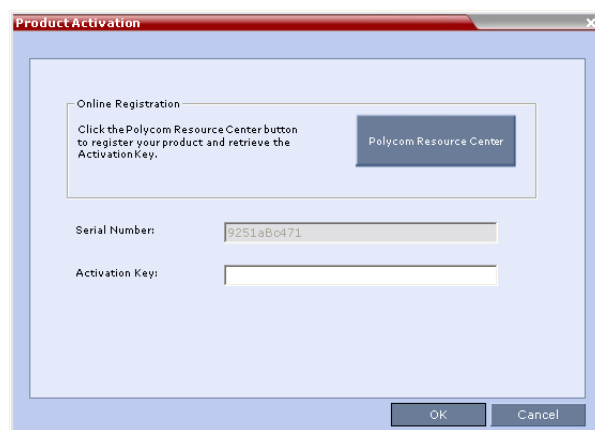
When the RMX's configuration is completed (including the *Management* and *IP Network Services*), and if there are no *System Errors*, the green RDY LED on the CNTL module (on the RMX's front panel) turns ON.

- 3 Remove the *USB key*.
- 4 Start the *RMX 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 key.
 - b Press **Enter**.

The *RMX Web Client* Login screen is displayed.

- 5 In the *RMX Web Client* Login screen, enter the default *Username* (**POLYCOM**) and *Password* (**POLYCOM**) and click **Login**.

The *RMX Web Client* opens and the *Product Activation* dialog box appears with the serial number filled in:

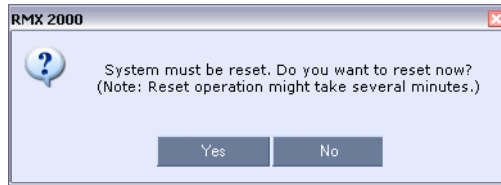


- 6 In the *Activation Key* field, enter or **paste** the *Product Activation Key* obtained earlier.
- 7 Click **OK**.

If you do not have an *Activation Key*, click **Polycom Resource Center** to access the *Service & Support* page of the Polycom website.

For more information, see "*Obtaining the Activation Key*" on page 2-7.

The system prompts with a restart dialog box:



- 8** In the dialog box, click **No**.

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* enables you to configure the *Default IP Service*. It starts automatically if no *Default IP Network Service* is defined. This happens during *First Time Power-up*, before the service has been defined or if the *Default IP Service* has been deleted, followed by an RMX restart.

The *IP Management Service* tab in the *Fast Configuration Wizard* is enabled only if the factory default Management IP addresses were not modified.

Fast Configuration Wizard

1 Enter the required IP information in the dialog box.

The screenshot shows the 'Fast Configuration Wizard' window with the 'IP Signaling' tab selected. On the left is a tree view with options: IP Management Service, IP Signaling (selected), Routers, DNS, Network Type, Gatekeeper, SIP Server, Security, ISDN/PSTN, PRI Settings, Span Definition, Phones, Spans, and System Flags. The main area contains the following fields: 'Network Service Name' with a text box containing 'Default IP Service'; 'Signaling Host IP Address' with a text box containing '0.0.0.0'; 'MPM 1 IP Address' with a text box containing '0.0.0.0'; 'MPM 2 IP Address' with a text box containing '0.0.0.0'; and 'Subnet Mask' with a text box containing '255.255.255.0'. At the bottom right are 'Back', 'Next', and 'Cancel' buttons.

Table 2-2 Fast Configuration Wizard – IP Signaling

Field	Description
<i>Network Service Name</i>	The name <i>Default IP Service</i> is assigned to the IP Network Service by the Fast Configuration Wizard. This name can be changed. Note: This field is displayed in all IP Signaling dialog boxes and can contain character sets that use Unicode encoding.
<i>Signaling Host IP Address</i>	Enter the address to be used by IP endpoints when dialing in to the MCU. Dial out calls from the RMX are initiated from this address. This address is used to register the RMX with a Gatekeeper or a SIP Proxy server.
<i>MPM/MPM+ 1 IP Address</i>	Enter the IP address(es) of the media card (s) (MPM/MPM+ 1 and MPM/MPM+ 2 (if installed)) as provided by the network administrator. Endpoints connect to conferences and transmit call media (video, voice and content) via these addresses.
<i>MPM/MPM+ 2 IP Address</i>	

Table 2-2 Fast Configuration Wizard – IP Signaling (Continued)

Field	Description
<i>Subnet Mask</i>	Enter the subnet mask of the MCU. Default value: 255.255.255.0.



If *Secured Communication* is required on the RMX: complete the *Fast Configuration Wizard*, Login, install the *Certificate* and then enable *Secured Communication Mode*.

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

The screenshot shows the 'Fast Configuration Wizard' window. On the left is a tree view with the following items: IP Management Service, IP Signaling, **Routers** (selected), DNS, Network Type, Gatekeeper, SIP Server, Security, ISDN/PSTN, PRI Settings, Span Definition, Phones, Spans, and System Flags. The main pane on the right contains two fields: 'Network Service Name' with the value 'Default IP Service' and 'Default Router IP Address' with the value '0.0.0.0'. At the bottom right of the window are three buttons: 'Back', 'Next', and 'Cancel'.

Table 2-3 Fast Configuration Wizard – Routers

Field	Description
<i>Default Router IP Address</i>	Enter the IP address of the default router.

- 4 Click **Next**.

5 Enter the required DNS information in the dialog box.

The screenshot shows the 'Fast Configuration Wizard' window. On the left is a tree view with the following items: IP Management Service, IP Signaling, Routers, DNS (highlighted), Network Type, Gatekeeper, SIP Server, Security, ISDN/PSTN, PRI Settings, Span Definition, Phones, Spans, and System Flags. The main area on the right contains the following fields and options:

- Network Service Name: Default IP Service
- MCU Host Name: PolycomMCU
- DNS: Off (selected from a dropdown menu)
- ☐ Register Host Names Automatically to DNS Server
- Local Domain Name: (empty text box)
- Primary DNS Server IP Address: 0.0.0.0

At the bottom right are three buttons: Back, Next, and Cancel.

Table 2-4 Fast Configuration Wizard – DNS

Field	Description
MCU Host Name	Enter the name of the MCU on the network. Default name is RMX.
DNS	Select: <ul style="list-style-type: none">• Off – if DNS servers are not used in the network.• Specify – to enter the IP addresses of the DNS servers. Note: The IP address fields are enabled only if Specify is selected.
Register Host Names Automatically to DNS Server	Select this option to automatically register the MCU Signaling Host and Shelf Management with the DNS server.
Local Domain Name	Enter the name of the domain where the MCU is installed.
Primary DNS Server IP Address	The static IP address of the primary DNS server.

- 6 Click **Next**.
- 7 Select the *IP Network Type*: **H.323**, **SIP** or **H.323 & SIP**.

The Fast Configuration Wizard dialog box is shown with the 'IP Network Type' section selected in the left-hand tree. The 'Network Service Name' is set to 'Default IP Service'. The 'IP Network Type' dropdown menu is open, showing the following options: 'H.323 & SIP' (selected), 'H.323', 'SIP', and 'H.323 & SIP'. The 'Back', 'Next', and 'Cancel' buttons are at the bottom right.

- 8 Click **Next**.
- 9 If you selected **SIP** only, go to **Step 13**.
- 10 Enter the required **Gatekeeper** information in the dialog box.

The Fast Configuration Wizard dialog box is shown with the 'Gatekeeper' section selected in the left-hand tree. The 'Network Service Name' is set to 'Default IP Service'. The 'Gatekeeper' dropdown menu is set to 'Off'. The 'Primary Gatekeeper' section contains the following fields: 'IP Address or Name:' and 'MCU Prefix in Gatekeeper:'. The 'Aliases' section contains a table with the following data:

Alias	Type
	None
	None
	None
	None
	None

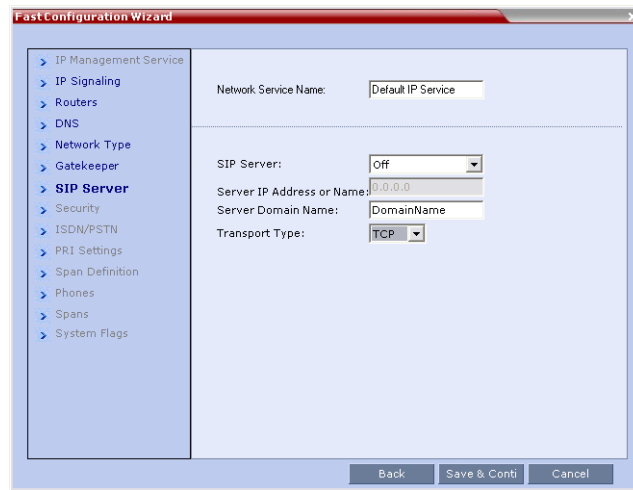
The 'Back', 'Next', and 'Cancel' buttons are at the bottom right.

Table 2-5 Fast Configuration Wizard – Gatekeeper

Field	Description
<i>Gatekeeper</i>	Select Specify to enable configuration of the gatekeeper IP address. When Off is selected, all gatekeeper options are disabled.
Primary Gatekeeper	
<i>IP Address or Name</i>	Enter either the gatekeeper's host name (if a DNS Server is used) or IP address.
<i>MCU Prefix in Gatekeeper</i>	Enter the string with which the MCU registers itself with the gatekeeper. The gatekeeper uses this string to identify the MCU when forwarding calls to it. H.323 endpoints use this number as the first part of their dial-in string when dialing the MCU.
Aliases	
<i>Alias</i>	The alias that identifies the RMX's Signaling Host within the network. Up to five aliases can be defined for each RMX. Note: When a gatekeeper is specified, at least one prefix or alias must be entered in the table.
<i>Type</i>	The type defines the format in which the card's alias is sent to the gatekeeper. Each alias can be of a different type: <ul style="list-style-type: none"> • H.323 ID (alphanumeric ID) • E.164 (digits 0-9, * and #) • Email ID (email address format, e.g. abc@example.com) • Participant Number (digits 0-9, * and #) Note: Although all types are supported, the type of alias to be used depends on your gatekeeper's capabilities.

11 Click **Next**.

12 If you selected **H.323** only, go to **Step 15**.

13 Enter the required **SIP Server** information in the dialog box.


The image shows a screenshot of the 'Fast Configuration Wizard' window, specifically the 'SIP Server' configuration step. On the left is a tree view with the following items: IP Management Service, IP Signaling, Routers, DNS, Network Type, Gatekeeper, **SIP Server** (highlighted), Security, ISDN/PSTN, PRI Settings, Span Definition, Phones, Spans, and System Flags. The main area contains the following fields:

- Network Service Name: Default IP Service
- SIP Server: Off (dropdown menu)
- Server IP Address or Name: 0.0.0.0
- Server Domain Name: DomainName
- Transport Type: TCP (dropdown menu)

At the bottom right are three buttons: Back, Save & Continue, and Cancel.

Table 2-6 Fast Configuration Wizard – SIP Server

Field	Description
<i>SIP Server</i>	Select: <ul style="list-style-type: none"> • Specify – to manually configure SIP servers. • Off – if SIP servers are not present in the network.
<i>SIP Server IP Address</i>	Enter either the IP address of the preferred SIP server or its host name (if a DNS server is used).
<i>Transport Type</i>	Select the 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: UDP – Select this option to use UDP for signaling. TCP – Select this option to use TCP for signaling. TLS – The <i>Signaling Host</i> 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.

Table 2-6 Fast Configuration Wizard – SIP Server (Continued)

Field	Description
<i>Transport Type (cont.)</i>	The following protocols are supported: <ul style="list-style-type: none">• TLS 1.0• SSL 2.0• SSL 3.0.

14 Click **Next**.

Enter the required **Security** information in the dialog box.

The screenshot shows the 'Fast Configuration Wizard' window. On the left is a tree view with the following items: IP Management Service, IP Signaling, Routers, DNS, Network Type, Gatekeeper, SIP Server, Security (selected), ISDN/PSTN, PRI Settings, Span Definition, Phones, Spans, and System Flags. The main area on the right is titled 'Security' and contains the following fields: 'Network Service Name' with a dropdown menu showing 'Default IP Service', 'Authentication User Name' with a text input field, and 'Authentication Password' with a text input field. At the bottom of the window are three buttons: 'Back', 'Next', and 'Cancel'.

Table 2-7 Fast Configuration Wizard – Security

Field	Description
<i>Authentication User Name</i>	Enter the conference, Entry Queue or Meeting Room name as registered with the proxy. This field can contain up to 20 ASCII characters.
<i>Authentication Password</i>	Enter the conference, Entry Queue or Meeting Room password as defined in the proxy. This field can contain up to 20 ASCII characters.

15 Click **Next**.

The IP Network Service is created and confirmed.



16 Click **OK**.

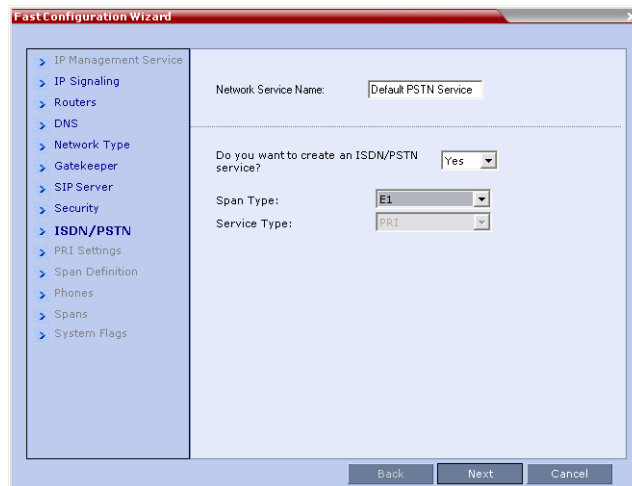
During the initial RMX setup, if the system detects the presence of the RTM ISDN card, the ISDN /PSTN Network Service definition screens of the *Fast Configuration Wizard* are enabled.

If there is no RTM ISDN card in the RMX 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 if no RTM ISDN card is installed in the system **but** only via 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:

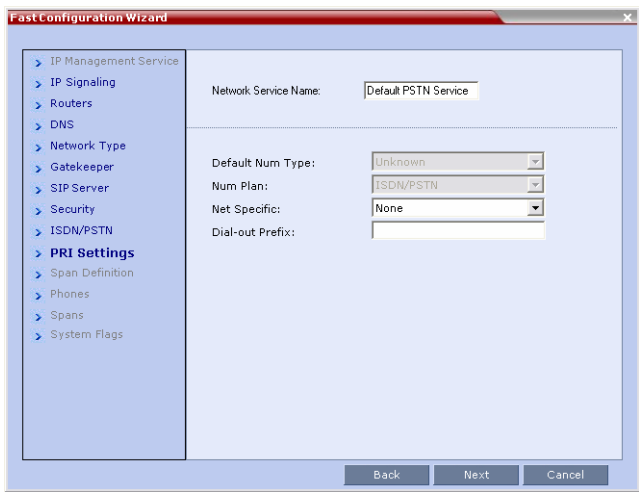


17 Define the following parameters:**Table 2-8** Fast Configuration Wizard – ISDN Service Settings

Field	Description
<i>Network Service Name</i>	<p>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.</p> <p>Default name: ISDN/PSTN Service</p> <p>Note: This field is displayed in all ISDN/PSTN Network Properties tabs and can contain character sets that use Unicode encoding.</p>
<i>Span Type</i>	<p>Select the type of spans (ISDN/PSTN) lines, supplied by the service provider, that are connected to the RMX. 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.</p> <p>Select either:</p> <ul style="list-style-type: none">• T1 (U.S. – 23 B channels + 1 D channel)• E1 (Europe – 30 B channels + 1 D channel) <p>Default: T1</p> <p>Note: Only one <i>Span Type</i> (E1 or T1) is supported on the RMX. If you define the first span as type E1 all other spans that you may later define must also be of type E1.</p>
<i>Service Type</i>	<p>PRI is the only supported service type. It is automatically selected.</p>

18 Click Next.

The *PRI Settings* dialog box opens.



19 Define the following parameters:

Table 2-9 Fast Configuration Wizard – PRI Settings

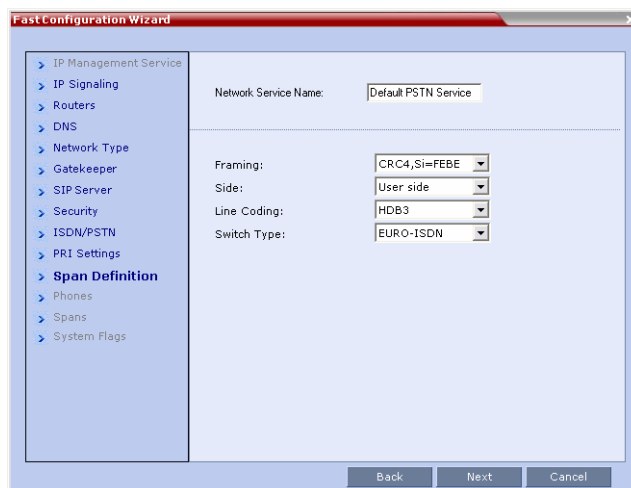
Field	Description
<i>Default Num Type</i>	<p>Select the Default Num Type from the list.</p> <p>The Num Type defines how the system handles the dialing digits. For example, if you type eight dialing digits, the Num Type defines whether this number is national or international.</p> <p>If the PRI lines are connected to the RMX 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.</p> <p>Default: Unknown</p> <p>Note: For E1 spans, this parameter is set by the system.</p>

Table 2-9 Fast Configuration Wizard – PRI Settings (Continued)

Field	Description
<i>Num Plan</i>	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.
<i>Net Specific</i>	Select the appropriate service program if one is used by your service provider (carrier). Some service providers may have several service programs that can be used. Default: None
<i>Dial-out Prefix</i>	Enter the prefix that the PBX requires to dial out. Leave this field blank if a dial-out prefix is not required. The field can contain be empty (blank) or a numeric value between 0 and 9999 . Default: Blank

20 Click **Next**.

The *Span Definition* dialog box opens.

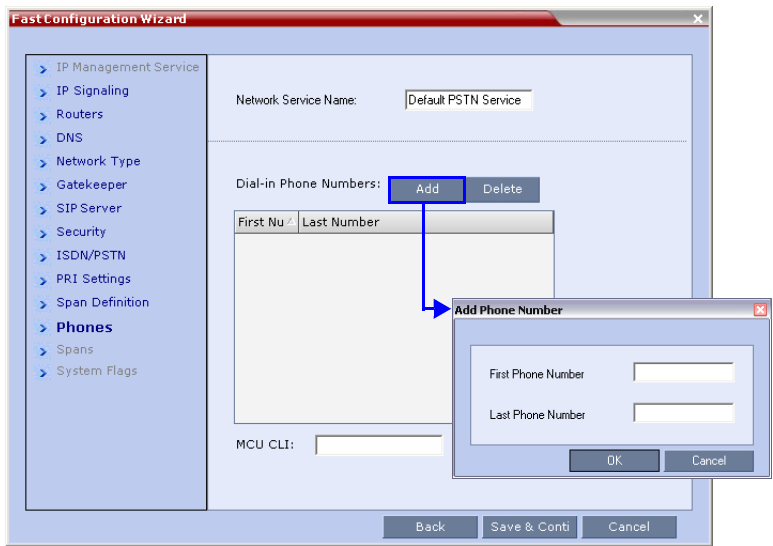


21 Define the following parameters:**Table 2-10** Fast Configuration Wizard – Spans Definition

Field	Description
<i>Framing</i>	Select the Framing format used by the carrier for the network interface from the list. <ul style="list-style-type: none"> For T1 spans, default is SFSF. For E1 spans, default is FEBE.
<i>Side</i>	Select one of the following options: <ul style="list-style-type: none"> User side (default) Network side Symmetric side <p>Note: If the PBX is configured on the network side, then the RMX unit must be configured as the user side, and vice versa, or both must be configured symmetrically.</p>
<i>Line Coding</i>	Select the PRI line coding method from the list. <ul style="list-style-type: none"> For T1 spans, default is B8ZS. For E1 spans, default is HDB3.
<i>Switch Type</i>	Select the brand and revision level of switch equipment installed in the service provider's central office. <ul style="list-style-type: none"> For T1 spans, default is AT&T 4ESS. For E1 spans, default is EURO ISDN.

22 Click Next.

The *Phones* dialog box opens.



23 Click **Add** to define dial-in number ranges.

The *Add Phone Number* dialog box opens.

24 Define the following parameters:

Table 2-11 *Fast Configuration Wizard – Add Phone Numbers*

Field	Description
<i>First Number</i>	The first number in the phone number range.
<i>Last Number</i>	The last number in the phone number range.



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

25 Click **OK**.

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

26 **Optional.** Repeat steps **23** to **24** to define additional dial-in ranges.

27 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.

28 Click **Save & Continue**.

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

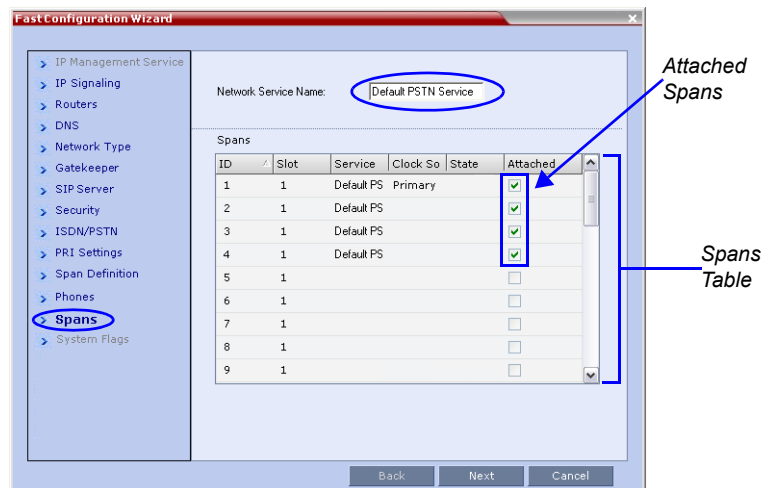
The *ISDN/PSTN Network Service* is created and is added to the ISDN/PSTN Network Services list.

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.



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

The *Spans* dialog box opens displaying the following read-only fields:



— *ID* – the connector on the ISDN RTM card (PRI1 to PRI12).

- *Slot* – the MPM card that the ISDN RTM card is connected to (MPM 1 or MPM 2).
- *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. The first span to synchronize becomes the *Primary* clock source.
- *State* – the *System Alert* level of the span (*Major*, *Minor*). If there are no span related alerts, this column contains no entries.

- 30** 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 *RMX Web 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 *RMX Web Client*.

Each ISDN RTM card can support either 7 E1 or 9 T1 PRI lines (E1 and T1 connections cannot be used simultaneously).

- 31** Click **Next**.

The *System Flags* dialog box is displayed.

Fast Configuration Wizard

- > IP Management Service
- > IP Signaling
- > Routers
- > DNS
- > Network Type
- > Gatekeeper
- > SIP Server
- > Security
- > ISDN/PSTN
- > PRI Settings
- > Span Definition
- > Phones
- > Spans
- > **System Flags**

Network Service Name: Default PSTN Service

Conference ID Length (MCU side): 5

Minimum Conference ID Length (user side): 4

Maximum Conference ID Length (user side): 4

MCU Display Name: POLYCOM RMX 2000

Terminate Conference when Chairperson Exits: No

Auto Extend Conferences: Yes

Back Save & Close Cancel

32 Enter the required **System Flags** information in the dialog box.

Table 2-12 Fast Configuration Wizard – System Flags

Field	Description / Default	
<i>Conference ID Length (MCU)</i>	The number of digits of the Conference ID to be assigned by the MCU. Range: 2-16 (Default: 5)	Note: Selecting 2 digits limits the number of simultaneous ongoing conferences to 99.
<i>Minimum Conference ID Length (User)</i>	The minimum number of digits that the user must enter when manually assigning a numeric ID to a conference. Range: 2-16 (Default: 4)	
<i>Maximum Conference ID Length (User)</i>	The maximum number of digits that the user can enter when manually assigning a Numeric ID to a conference. Range: 2-16 (Default: 8)	
<i>MCU Display Name</i>	The MCU name is displayed on the endpoint's screen. Default name: <i>Polycom RMX 2000</i>	

Table 2-12 Fast Configuration Wizard – System Flags (Continued)

Field	Description / Default
<i>Terminate Conference when Chairperson Exits</i>	When Yes is selected (default), the conference ends when the chairperson exits even if there are other participants connected. When No is selected, the conference automatically ends at the predefined end time, or when all the participants have disconnected from the conference.
<i>Auto Extend Conferences</i>	When Yes is selected (default), allows conferences running on the RMX 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.

These flags can be modified later, if required, by selecting the *System Configuration* option from the *Setup* menu. For more information, see the RMX 2000 Administrator's Guide, "System Configuration" on page 14-10.

33 Click **Save & Close**.

The RMX confirms successful configuration.

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

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

36 In the *Please wait for system reset* message box, click **OK**.



System restart may take up to five minutes.

37 Refresh the browser periodically until the *Login* screen is displayed.

38 When the *Login* screen is displayed, enter your *Username* and *Password* and click **Login**.

On first entry, the default *Username* and *Password* are both **POLYCOM**.

39 Create a new *User* with *Administrator* permissions and delete the default User (**POLYCOM**).

For system security reasons the system is not fully configured until this step has been performed.

For more information see the *RMX 2000 Administrator's Guide*, "Users, Connections and Notes" on page [10-1](#).

The system is now fully configured and if there are no *System Errors*, the green RDY LED on the CNTL module (on the RMX's front panel) turns ON.

User Definition

The RMX is shipped with a default Administrator user called POLYCOM. Once you have defined other authorized administrator users, it is recommended to remove the default user to prevent unauthorized users from logging into the system.

For more information, see the *RMX 2000 Administrator's Guide* "Deleting a User" on page [10-4](#).

Selecting the RMX Web Client Languages

By default, the RMX 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:

- 1** On the RMX menu, click **Setup > Multilingual Setting**.
- 2** Click the check boxes of the languages to appear in the Login screen of the *RMX Web Client*. For more information see the *RMX 2000 Administrator's Guide* "Multilingual Setting" on page [14-59](#).

If the selected language is not supported by the browser or the workstation's Operating System, the *RMX Web Client* is displayed in English.

- 3** Click **OK**.
- 4** Log out and reconnect to the RMX.
The Login screen will display the flags of the selected languages.

RMX's Default Conferencing Settings

The RMX is shipped with default pre-configured conferencing entities, which allow RMX users and participants to start ongoing conferences without further configuration.

The default conferencing entities are:

Table 2-13 Conferencing Entities

Entity	Description										
Meeting Rooms	<p>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:</p> <table><tr><td>Name</td><td>ID</td></tr><tr><td>Maple_Room</td><td>1001</td></tr><tr><td>Oak_Room</td><td>1002</td></tr><tr><td>Juniper_Room</td><td>1003</td></tr><tr><td>Fig_Room</td><td>1004</td></tr></table> <p>Each Meeting Room uses the default <i>Conference Profile</i> called <i>Factory Video Profile</i> running at 384Kbps and has a default duration of one hour.</p>	Name	ID	Maple_Room	1001	Oak_Room	1002	Juniper_Room	1003	Fig_Room	1004
Name	ID										
Maple_Room	1001										
Oak_Room	1002										
Juniper_Room	1003										
Fig_Room	1004										
Conference Profile	<p>Name: <i>Factory Video Profile</i></p> <p>A Conference Profile is assigned to a Meeting Room to define its conferencing parameters, such as line rate and video resolution.</p> <p><i>Factory Video Profile</i> contains the video conference parameters with a bit rate of 384Kbps, <i>Auto Layout</i> and Polycom <i>Skin</i>. The Profile uses an IVR Service called <i>Conference IVR Service</i>.</p>										
Conference IVR Service	<p>Name: <i>Conference IVR Service</i></p> <p>The <i>Conference IVR Service</i> includes an optional video slide and all the voice messages played during the participant's connection process and during the conference.</p> <p>The <i>Conference IVR Service</i> contains a set of voice prompts in English and an optional video slide. It automates the participant's connection to a conference.</p>										

Table 2-13 Conferencing Entities (Continued)

Entity	Description
<i>Entry Queue</i>	<p>Name ID <i>DefaultEQ</i> 1000</p> <p>Using an Entry Queue enables one dial-in number to be used for all conferences. In the Entry Queue, participants are prompted for information to enable routing to their destination conferences.</p> <p>A default Entry Queue called <i>DefaultEQ</i> is provided. 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 on going conference currently running on the MCU. It uses an Entry Queue IVR Service called <i>Entry Queue IVR Service</i>.</p> <p>The default <i>Welcome Slide</i> 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.</p> <p>If no <i>Transit Entry Queue</i> is defined, <i>DefaultEQ</i> is the default <i>Transit Entry Queue</i>. For more information see the <i>RMX 2000 Administrator's Guide</i>, "Transit Entry Queue" on page 3-8.</p> <p>Note: An ISDN/PSTN dial-in number is not assigned to the <i>Entry Queue</i> as the number depends on the dial-in numbers range defined in the <i>Network Service</i>. It must be manually assigned to enable ISDN or PSTN participant connections to this <i>Entry Queue</i>. For more information see the <i>RMX 2000 Administrator's Guide</i>, "ISDN/PSTN Network Services" on page 11-25.</p>
<i>Entry Queue IVR Service</i>	<p>Name: <i>Entry Queue IVR Service</i></p> <p>Includes all the voice messages and video slides used to guide participants through their connection process to the MCU and route them to their destination conference.</p> <p><i>Entry Queue IVR Service</i> is the default Entry Queue IVR Service provided for the default Entry Queue.</p>

Customizing the RMX's Default Conferencing Settings

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

- **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 *RMX 2000 Administrator's Guide, "IVR Services"* on page **12-1**.
- **To modify the conference properties**, such as the conference bit rate, specific video layout for the conference or the background that is used for the video display (skin), create a new Conference Profile. This Profile can be used for defining new ongoing conferences, Meeting Rooms and Single-dial Entry Queues. For more information, see the *RMX 2000 Administrator's Guide, "Defining Profiles"* on page **1-8**.
- **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 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 *RMX 2000 Administrator's Guide, "Defining a New Entry Queue Service"* on page **12-24**.
- **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 *RMX 2000 Administrator's Guide, "Meeting Rooms"* on page **2-1**.
- 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 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 *RMX 2000 Administrator's Guide, "Address Book"* on page **4-1**.

Basic Operation

The most common operations performed via the *RMX Web Client* 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 RMX Web Client

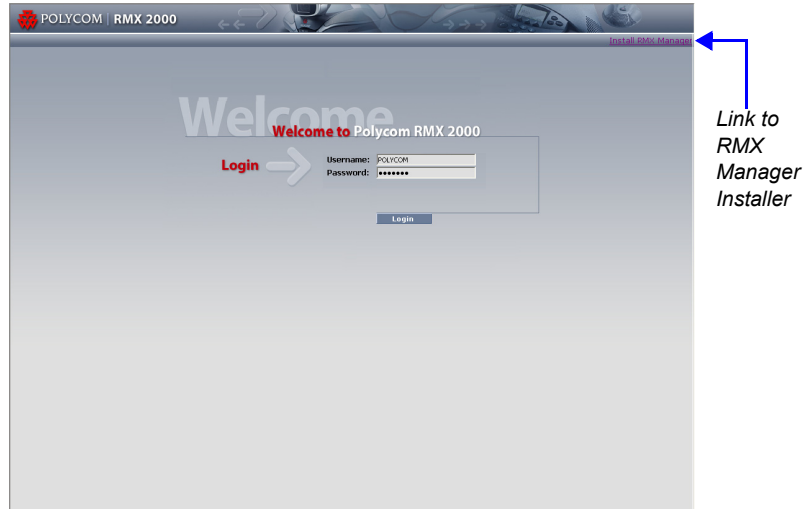
Before you begin you need to get the following information from your system administrator:

- User name
- Password
- MCU Control Unit IP Address

To start the RMX Web Client:

- 1 In your browser address line, enter **http://<Control Unit IP Address>** and press the **Enter** key.

The *Login* screen is displayed.



- 2 Enter your *Username* and *Password* and click the **Login** button.
On first entry, the default *Username* and *Password* are both **POLYCOM**.
The *RMX Web Client* main screen is displayed.



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

The *RMX Manager* is faster than the *RMX Web Client* and can give added efficiency to RMX management tasks, especially when deployed on workstations affected by:

- Lack of performance due to bandwidth constraints within the LAN/WAN environment.
- Slow operation and disconnections that can be caused by the anti-phishing component of various antivirus applications.

For more information see the *RMX 2000 Administrator's Guide*, "RMX Manager" on page **14-1**.

RMX Web Client Screen Components

The *RMX Web Client's* main screen consists of five panes:

- *Conference List*
- *List Pane*
- *RMX Management*
- *Status Bar*
- *Address Book*
- *Conference Templates*

You can Login as a user with Chairperson, *Operator* or *Administrator* authorization. Your authorization level determines your viewing and system functions.

For more information see the *RMX 2000 Administrator's Guide*, "Users, Connections and Notes" on page **10-1**.

The *Administrator's* view is shown below:

The screenshot shows the RMX 2000 Web Client interface. The top navigation bar includes 'View', 'Administration', 'Setup', and 'Help'. The main area is divided into several panes:

- Conferences List:** A table showing conference details. The first row is for a conference named 'SUPP' with ID 43020, starting at 12:23 PM.
- List Pane:** A table showing participant details. The first row is for a participant named 'sdn1' with IP address 124578, role 'Conn', and network 'ISDN/PS'.
- RMX Management:** A sidebar menu with sections 'Frequently Used' (Connections, Users, Signaling Monitor, Hardware Monitor, Meeting Rooms, Reservations) and 'Rarely Used' (IVR Services, IP Network Services, Recording Links, ISDN/PS/TN Network Services, Conference Profiles, Entry Queues, SIP Factories, Log Analysis).
- Address Book:** A table showing contact details. The first row is for a contact named '213' with type 'Dial o'.
- Conference Templates:** A sidebar menu with options like 'p31', 'p41', 'p41sip', 'PartyID', and 'PartyID'.
- Status Bar:** A bottom bar showing system alerts, participant alerts, port usage (Voice: 0 / 40, Video: 12 / 72), and MCU state (MAJOR).

The main screen can be customized. For more information, see "Customizing the Main Screen" on page **3-11**.

Viewing and System Functionality Permissions

The RMX Web Client user's viewing and system functionality depend on the authorization level assigned to each user as summarized in Table 3-1:

Table 3-1 Viewing and System Permissions

	Authorization Level		
	Chairperson	Operator	Administrator
Viewing Permissions			
Conference List	✓	✓	✓
List Pane	✓	✓	✓
Address Book	✓	✓	✓
Conference Templates		✓	✓
Status Bar		✓	✓
RMX Management		✓	✓
Conference Alarms		✓	✓
Conference Status		✓	✓
Configurations		✓	✓
System Functionality			
Start Conferences	✓	✓	✓
Monitor Conferences	✓	✓	✓
Monitor Participants	✓	✓	✓
Solve Basic Problems		✓	✓
Modify MCU Configuration			✓

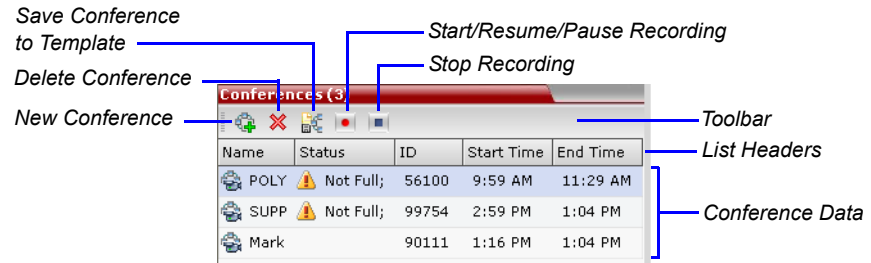


In addition to Chairpersons, Operators and Administrators, an Auditor is a user type that can view Auditor Files and audit the system. For more information see the *RMX 2000 Administrator's Guide "Auditor"* on page [14-67](#).

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).

If *Conference Recording* is enabled the following are displayed in color:

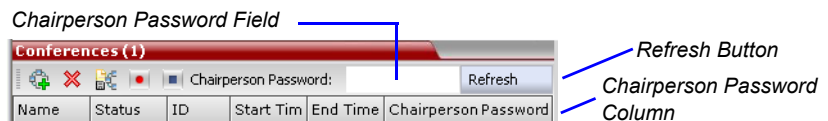
- **Start/Resume Recording** – start/resume recording.
- **Stop Recording** – stop recording.
- **Pause** – toggles with the *Start/Resume* button.

If you are logged in as a user with *Chairperson* permissions:

- You can search and monitor a list of 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 see “Using the *Chairperson Password* for Filtering” on page 3-38.

- 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.



RMX Management

Viewing Permissions		
Chairperson	Operator	Administrator
	✓	✓

The *RMX Management* pane lists the entities that need to be configured to enable the RMX 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.

Status Bar

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



Viewing Permissions		
Chairperson	Operator	Administrator
	✓	✓

System Alerts

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

System Alerts (6)						
Time	Category	Level	Code	Process Name	Description	
9/25/2006	general	major	IP_SERVICE_CHANGED	CSMgr	ip service was changed, reset the RMX (Task statu	
9/13/2006	card	major	NO_CONNECTION_WITH_CARD	Cards	Board ID:0, Card Type:switch, Description: No co	
9/13/2006	general	major	INTERNAL_MCU_RESET	McmsDaemo	No connection with Switch (Task status: Normal)	
9/13/2006	general	major	INSUFFICIENT_RESOURCES	Resource	Insufficient resources (Task status: Normal)	
9/13/2006	card	major	CARD_STARTUP_FAILURE	Cards	Board ID:0, Card Type:illegal, Description: MFA s	
9/13/2006	general	major	CFG_CHANGED	McuMgr	SYSTEM CFG was changed, reset the RMX (Task s	

For more information about **Active Alarms** and **Faults List**, see the *RMX 2000 Administrator's Guide*, "System and Participant Alerts" on page 14-6.

Viewing Permissions		
Chairperson	Operator	Administrator
✓	✓	✓

Participant Alerts

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*.

Participant Alerts (2)											
Conference	Name	Status	Disconnection Time	Role	IP Address	Alias	Network	Dialing Direction	Audio	Video	Encryptio
Marketing	V69	discon	9/21/2006 2:18 PM		172.22.189		H.323	Dial out			
Marketing	V96	discon	9/21/2006 2:18 PM		172.22.186		H.323	Dial out			

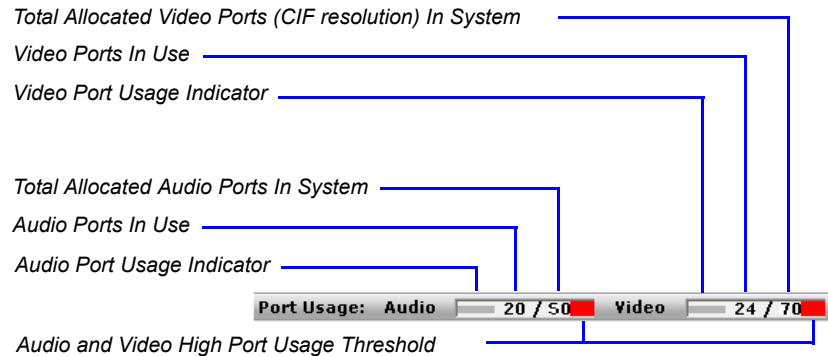
Viewing Permissions		
Chairperson	Operator	Administrator
	✓	✓

Port Usage Gauges

The *Port Usage* gauges indicate:

- 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.



The High Port Usage threshold represents a percentage of the total number of video or voice ports available. It is set to indicate when resource usage is approaching its maximum, resulting in no free resources to run additional conferences. When port usage reaches or exceeds the threshold, the red area of the gauge flashes and a *System Alert* is generated. The default port usage threshold is 80% and it can be modified by the system administrator. For more information, see the *RMX 2000 Administrator's Guide*, "Port Usage Gauges" on page [14-43](#).

Viewing Permissions		
Chairperson	Operator	Administrator
✓	✓	✓

MCU State

The *MCU State* indicator displays one of the following:

- **MCU State: STARTUP** – The MCU is starting up.
- **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.

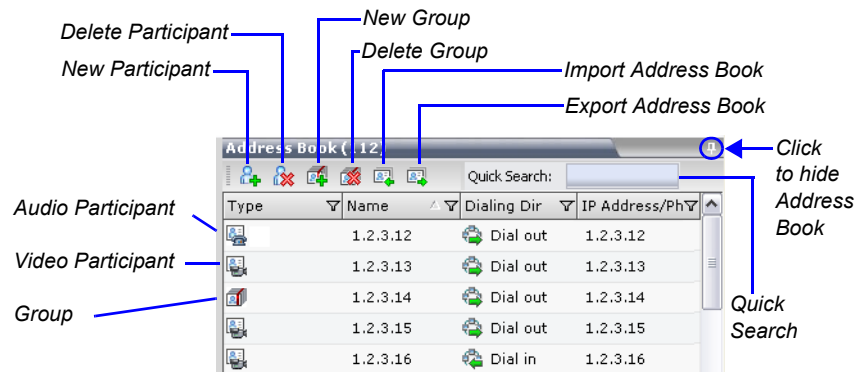
Address Book

Viewing Permissions		
Chairperson	Operator	Administrator
✓	✓	✓

The *Address Book* is a list of *Participants* and *Groups* that have been defined on the RMX. The information in the *Address Book* can be modified only by an administrator. All RMX 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*
- *Delete Participant*
- *Import Address Book*
- *New Group*
- *Delete Group*
- *Export Address Book*



Address Book entries are listed according to:

- **Type** - whether an individual *Participant* or a *Group* of participants
- **Name** - of the participant or group
- **Dialing Direction** - Dial-in or Dial-out
- **IP Address/Phone** - of the participant

Displaying and Hiding the Address Book

The first time you access the *RMX 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*.

Click tab to open Address Book



Conference Templates

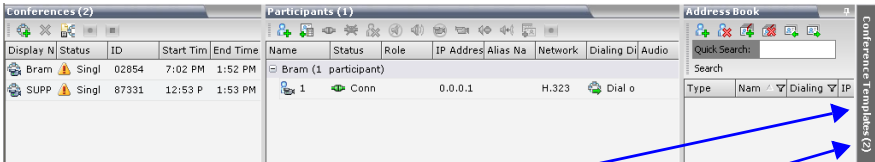
Conference Templates enable administrators and operators to create, save, schedule and activate identical conferences.

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.

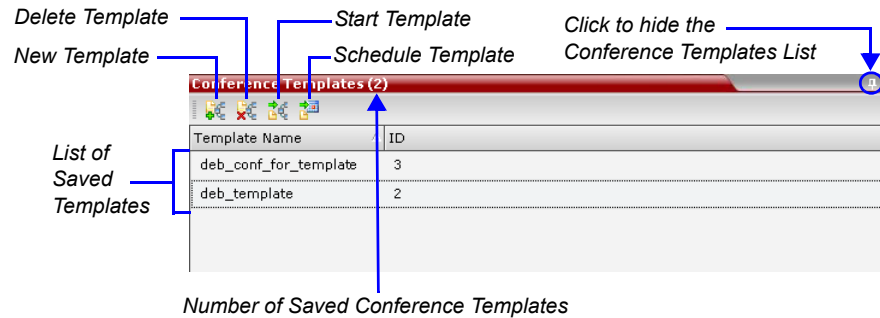
Displaying and Hiding Conference Templates

The *Conference Templates* list pane is initially displayed as a closed tab in the *RMX Web Client* main window. The number of saved *Conference Templates* is indicated on the tab.



Conference Templates Tab
Number of Saved Conference Templates

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 *RMX 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):

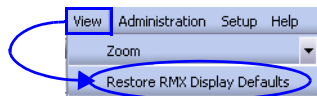
- 1 In the *Conference* list or *List* view pane, click on the column heading of the field to be used for sorting.
A ▾ or ▲ symbol appears in the column heading indicating that the list is sorted by this field, as well as the sort order.
- 2 Click on the column heading to toggle the column's sort order.

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 RMX display window to its default configuration:

- ➔ On the *RMX* menu, click **View > Restore RMX Display Defaults**.



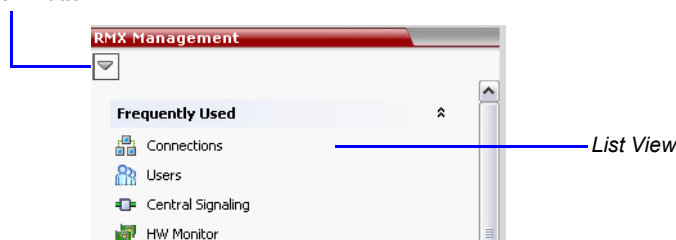
Customizing the RMX Management Pane

The *RMX Management* pane can be viewed either as a list or as a toolbar.

To switch between Toolbar and List Views:

- ➔ In the *RMX Management* pane, click the *Toolbar View* button to switch to Toolbar view.
- ➔ In Toolbar view, click the *List View* button to switch back to List view.

Toolbar View Button



List View Button



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 *RXM Web Client*.

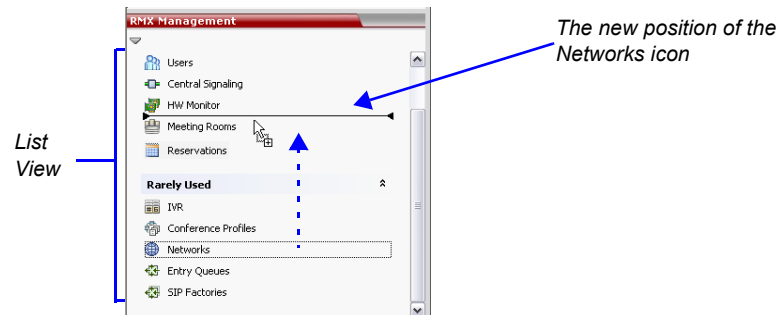
This only works in *List* view because in *Toolbar* view, all items are represented by icons.



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.



The *Frequently Used* and *Rarely Used* sections can be expanded or collapsed by clicking the  and  buttons.

Starting a Conference

There are several ways to start a conference:

- Clicking the *New Conference* button in the *Conferences* pane. For more information, see "Starting a Conference from the Conferences Pane" on page [3-15](#).
- 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 *RMX 2000 Administrator's Guide*, "Meeting Rooms" on page [2-1](#).

- Dialing in to an Ad Hoc Entry Queue which is used as the access point to the MCU.

For a detailed description of Ad Hoc Entry Queues, see the *RMX 2000 Administrator's Guide*, "Entry Queues" on page [3-1](#).

- Start a *Reservation*:
 - If the *Start Time* of the *Reservation* is past due the conference becomes ongoing immediately.
 - If the *Start Time* of the *Reservation* is in the future the conference becomes ongoing, at the specified time on the specified date.

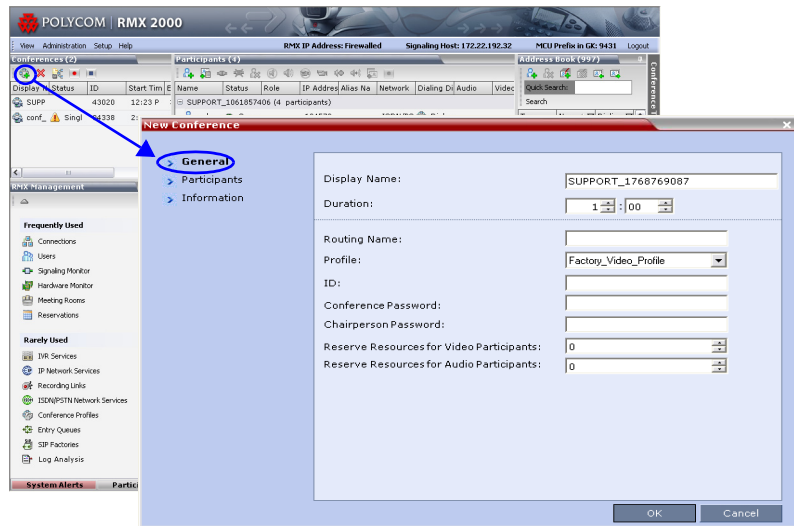
For more information, see "Starting a Reservation" on page [3-25](#).

Starting a Conference from the Conferences Pane

To start a conference from the Conference pane:

- 1 In the *Conferences* pane, click the **New Conference** (🔧) button.

The *New Conference – General* dialog box opens.



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

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

In most cases, the default conference *ID* can be used and you can just click **OK** to launch the conference. If required, you can enter a conference *ID* before clicking **OK** to launch the conference.

If you are the meeting chairperson or organizer using the *RMX Web Client* to start your own meeting, you need to communicate the default conference ID (or the one you created) to the other conference participants so they can dial in.

You can use the *New Conference - General* dialog box to modify the conference parameters. If no defined participants are to be added to the conference, or you do not want to add additional information, click **OK**.

General Tab

2 Define the following parameters:

Table 3-2 New Conference – General Options

Field	Description
<i>Display Name</i>	<p>The Display Name is the conferencing entity name in native language character sets to be displayed in the RMX Web Client.</p> <p>In conferences, Meeting Rooms, Entry Queues and SIP factories the system automatically generates an ASCII name for the <i>Display Name</i> field that can be modified using Unicode encoding.</p> <ul style="list-style-type: none">• 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. <p>The maximum length of text fields also varies according to the mixture of character sets (Unicode and ASCII).</p> <p>Maximum field length in ASCII is 80 characters. If the same name is already used by another conference, Meeting Room or Entry Queue, the RMX displays an error message requesting you to enter a different name.</p> <p>Note: This field is displayed in all tabs.</p>
<i>Duration</i>	<p>Define the duration of the conference in hours using the format HH:MM (default 01:00).</p> <p>Note: This field is displayed in all tabs.</p>

Table 3-2 New Conference – General Options (Continued)

Field	Description
<i>Routing Name</i>	<p><i>Routing Name</i> 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 server. This name must be defined using ASCII characters.</p> <p>Comma, colon and semicolon characters cannot be used in the <i>Routing Name</i>.</p> <p>The <i>Routing Name</i> can be defined by the user or automatically generated by the system if no <i>Routing Name</i> is entered as follows:</p> <ul style="list-style-type: none"> • If ASCII characters are entered as the <i>Display Name</i>, it is used also as the <i>Routing Name</i> • If a combination of Unicode and ASCII characters (or full Unicode text) is entered as the <i>Display Name</i>, the <i>ID</i> (such as Conference ID) is used as the <i>Routing Name</i>. <p>If the same name is already used by another conference, Meeting Room or Entry Queue, the RMX displays an error message and requests that you to enter a different name.</p>
<i>Profile</i>	<p>The system displays the name of the default Conference Profile. Select the required Profile from the list.</p> <p>The Conference Profile includes the Conference line rate, media settings and general settings.</p> <p>For a detailed description of Conference Profiles, see the <i>RMX 2000 Administrator's Guide</i>, "Conference Profiles" on page 1-1.</p>
<i>ID</i>	<p>Enter the unique-per-MCU conference ID. If left blank, the MCU automatically assigns a number once the conference is launched.</p> <p>This ID must be communicated to conference participants to enable them to dial in to the conference.</p>

Table 3-2 New Conference – General Options (Continued)

Field	Description
<i>Conference Password</i>	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.
<i>Chairperson Password</i>	Enter a password to be used by the RMX to identify the <i>Chairperson</i> 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.
<i>Reserve Resources for Video Participants</i>	Enter the number of video participants for which the system must reserve resources. Default: 0 participants. Maximum: <ul style="list-style-type: none"> • MPM Mode: 80 participants. • MPM+ Mode: 80 participants.
<i>Reserve Resources for Audio Participants</i>	Enter the number of audio participants for which the system must reserve resources. Default: 0 participants. Maximum: <ul style="list-style-type: none"> • MPM Mode: 80 participants. • MPM+ Mode: 120 participants.

- 3** If all participants are undefined, dial-in and no additional information is required for the new conference, click **OK**.

- 4 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*.

It is also used to add defined dial-out participants to the conference. Defined dial-out participants are connected to the conference automatically when the conference is launched

- 5 Click the **Participants** tab.

The *Participants* tab opens.

The screenshot shows the 'New Conference' dialog box with the 'Participants' tab selected. The left sidebar has 'General', 'Participants' (highlighted), and 'Information'. The main area contains a 'Participants List' table with columns: Name, IP Address, Alias Name, Network, Dialing ID, and Encryption. Below the table are buttons for 'New', 'Remove', and 'Add from Address Book'. A 'Lecturer' dropdown menu is at the bottom left. 'OK' and 'Cancel' buttons are at the bottom right.

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.

Table 3-3 *New Conference – Participants Tab*

Column / Button	Description
Participants List	
<i>Name</i>	A Unicode field that displays the participant's name and an icon representing the endpoint type: <i>Audio Only</i> or <i>Video</i> .
<i>IP Address/Phone</i>	Indicates the IP address or phone number of the participant's endpoint. <ul style="list-style-type: none"> For dial-out connection, displays the IP address or phone number of the endpoint called by the Polycom RMX 2000. For dial-in connection, displays the participant's IP address or phone number used to identify and route the participant to the appropriate conference.
<i>Alias Name/SIP Address (IP Only)</i>	Displays the alias name of an H.323 endpoint or the SIP URL.
<i>Network</i>	The network communication protocol used by the endpoint to connect to the conference: <i>H.323</i> , <i>SIP</i> or <i>ISDN/PSTN</i> .
<i>Dialing Direction</i>	Dial-in – The participant dials in to the conference Dial-out – The RMX dials out to the participant
<i>Encryption</i>	Displays whether the endpoint uses encryption for its media. The default setting is <i>Auto</i> , indicating that the endpoint must connect according to the conference's encryption setting. Note: The H.320 protocol (ISDN/PSTN) does not support encryption.

Table 3-3 New Conference – Participants Tab (Continued)

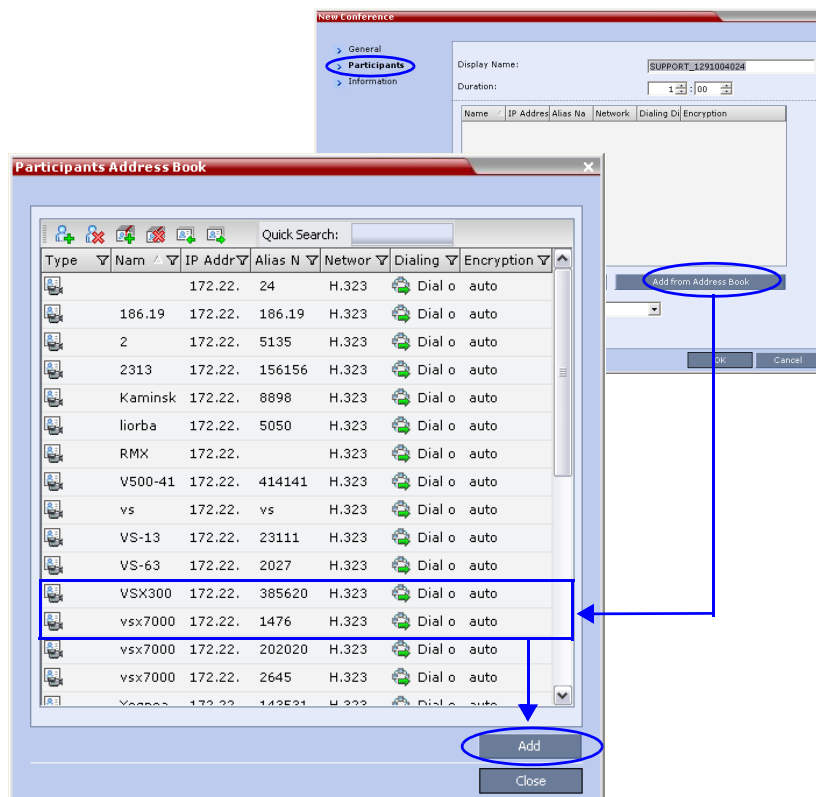
Column / Button	Description
Buttons	
New	Click to define a new participant. For more information, see the <i>RMX 2000 Administrator's Guide</i> , "Adding a new participant to the Address Book Directly" on page 4-4.
Remove	Click to remove the selected participant from the conference.
Add from Address Book	Click to add a participant from the <i>Address Book</i> to the conference.
Lecturer	
<i>Lecturer</i>	This option is used to activate the <i>Lecture Mode</i> . Select the participant you want to designate as <i>Lecturer</i> from the drop-down menu list of conference participants.

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" on page 3-47.

To add participants from the Address Book:

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



- 7 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.
- 8 The selected participants are assigned to the conference and appear in the *Participant List*.
- 9 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:

10 Click the **Information** tab.

The *Information* tab opens.

The screenshot shows the 'New Conference' dialog box with the 'Information' tab selected. The 'Display Name' field contains 'SUPPORT_1291004024'. The 'Duration' field is set to '1 : 00'. There are four empty text input fields labeled 'Info1:', 'Info2:', 'Info3:', and 'Billing Info:'. The 'OK' and 'Cancel' buttons are at the bottom right.

11 Enter the following information:

Table 3-4 *New Conference – Info Options*

Field	Description
<i>Info1, 2, 3</i>	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.
<i>Billing</i>	Enter the conference billing code if applicable.

12 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.

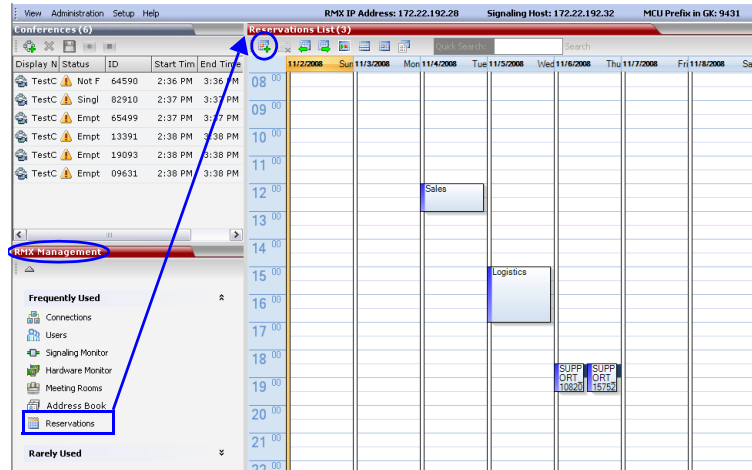
If no participant connects within the time specified in the *Conference Profiles > Auto Terminate > Before First Joins* field, the conference is automatically terminated by the system.

Starting a Reservation

To start a conference from the Reservation Calendar:

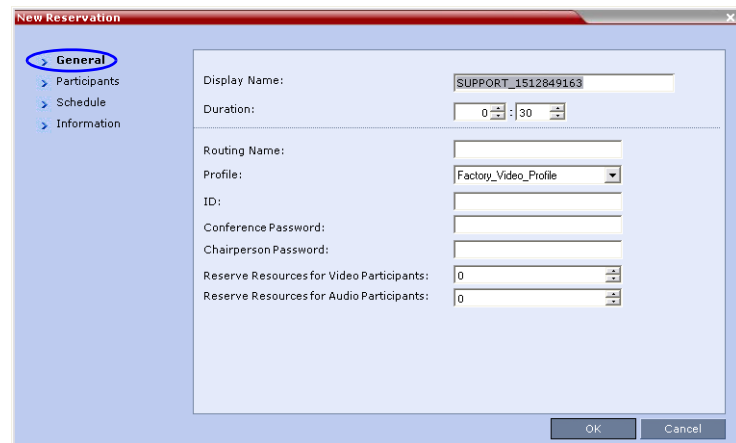
- 1 In the *RMX Management* pane, click the *Reservation Calendar* button ()

The *Reservation Calendar* is displayed.



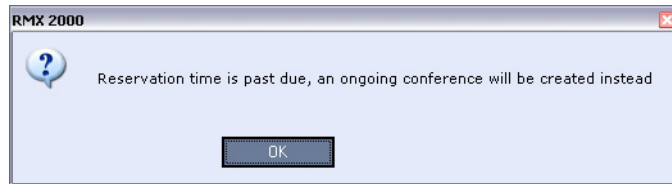
- 2 Click the **New Reservation** () button.

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



- 3 Click the **OK** button.

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



4 Click the **OK** button.

The conference is started.

For more information about *Reservations*, see the *RMX 2000 Administrator's Guide*, "Reservations" on page [5-1](#).

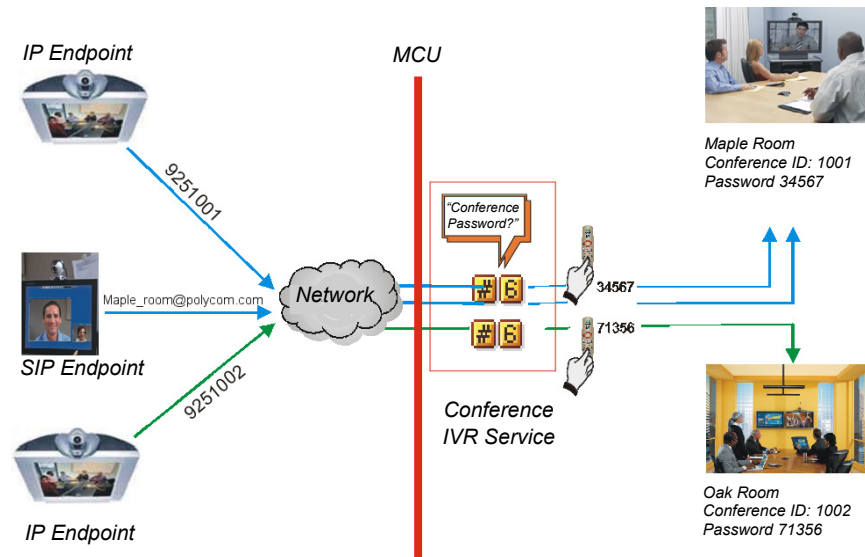
Connecting to a Conference

Direct Dial-in

Direct dial-in connection to conferences and Meeting Rooms is available to IP endpoints only.

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

The chairperson can use the chairperson password as the conference password and does not need to enter the conference password.



Participants connecting to HD Video Switching conferences must have HD capable endpoints and must connect using the same line rate as defined for the conference. If not, they are connected as Secondary (audio only participants).

H.323 Participants

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

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

Example:

Conference routing name	1001
-------------------------	------

➡ The participant dials 1001@polycom.com

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.

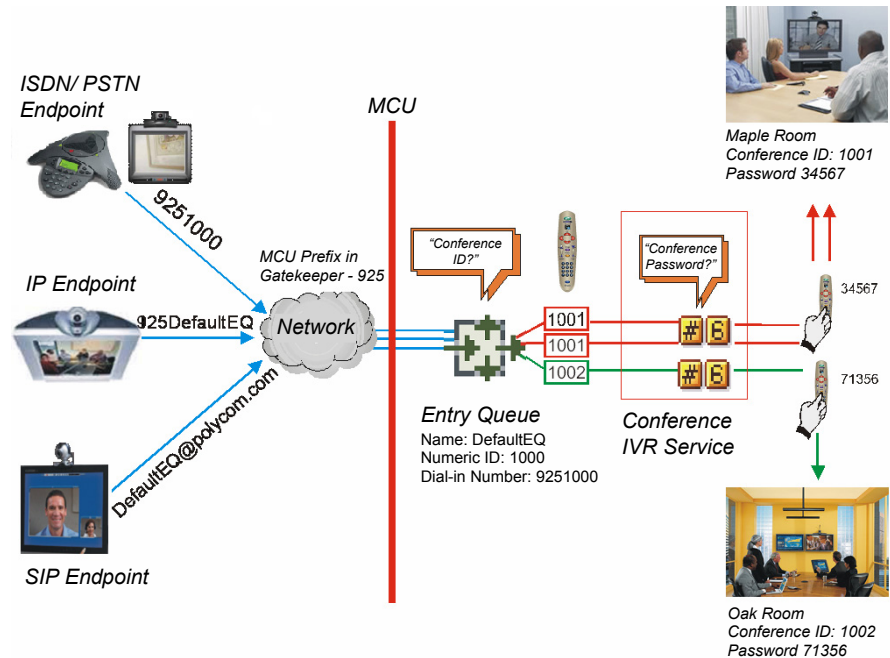


Figure 3-1: Dial-in Connection via Entry Queue

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

H.323 Participants

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

Using an Entry Queue 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>
```

Example:

Entry Queue Routing Name	DefaultEQ
--------------------------	-----------

Domain Name	polycom.com
-------------	-------------

➔ SIP participants dial DefaultEQ@polycom.com

ISDN and PSTN Participants

ISDN and PSTN participants can connect to conferences and Meeting Rooms only via an Entry Queue.

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 9251000

➡ ISDN/PSTN participants dial 9251000

Once connected to the Entry Queue, they enter the conference Numeric ID or password to be routed to the appropriate conference.

Dial-out Participants

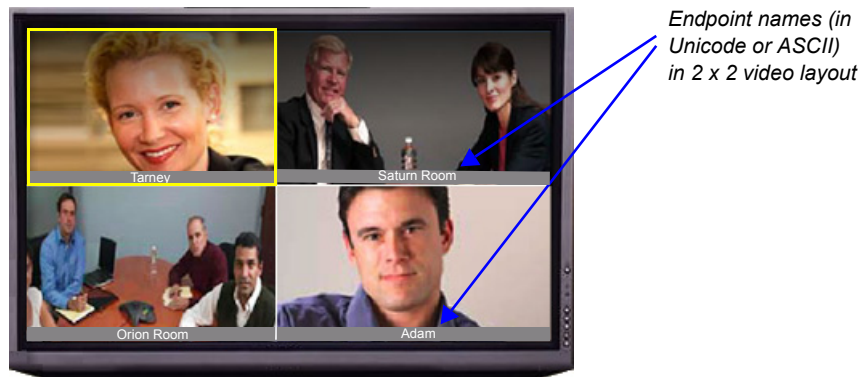
Dial-out participants are defined with their dial-out number. Once they are added to the ongoing conference, the MCU automatically calls them, using the default H.323, SIP or ISDN/PSTN Network Service defined for them.

Text Indication in the Video Layout

Endpoint 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:



The displayed name is determined as follows:

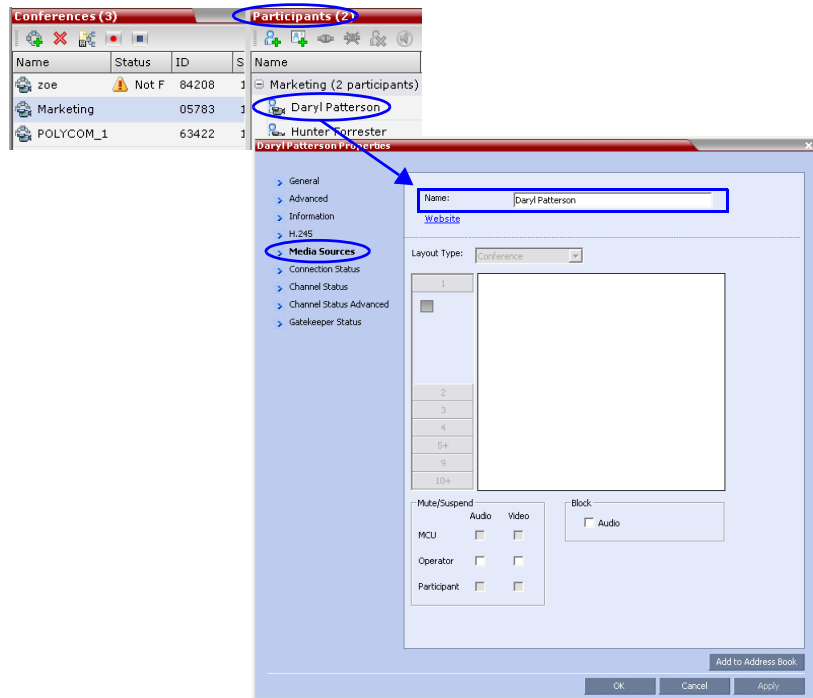
- The system displays the name that is defined at the endpoint.
- If the endpoint does not send its name:
 - For a defined H.323 or SIP participant:
 - The system displays the name from the participant definition.
 - For an undefined H.323 participant:
 - Display the *H.323 ID* alias.
 - or
 - Display 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:
 - 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 *RMX 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**.

The *Participant Properties – Media Sources* dialog box opens:



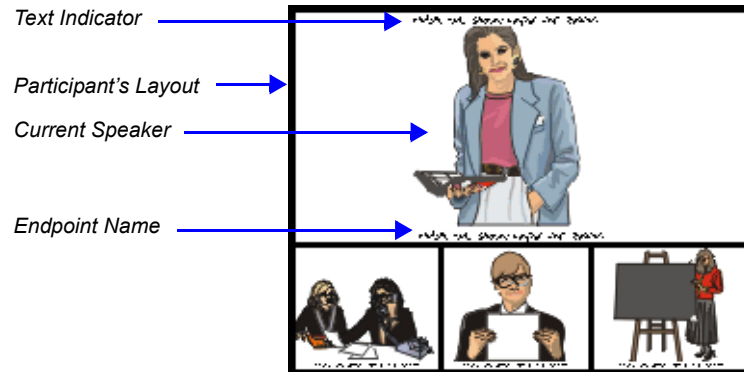
- 2 Enter the new *Display Name* in the *Name* field.
- 3 Click **OK**.

Text Indication

The *Text Indication* appears in the window of the current speaker in the participant's layout in addition to the endpoint name. It displays the conference Secure mode (on or off), total number of connected participants, number of video participants and number of audio participants.

The text indication is automatically displayed when there is a change in the conference Secure state (when Secure is implemented or cancelled) and it appears only for a few seconds (the same duration as the endpoint names).

The conference chairperson or participants can request the display of a *Textual Indication* of the conference's statistics by entering the DTMF code *88 on the endpoint's DTMF input device, such as remote control.



The Text Indication is displayed according to the permission set in the Conference IVR Service:

- Chairperson permission: Only the chairperson sees the indication
- Everyone permission: All participants see the indication.



Participants connected as Secondary (no video) will be considered as audio participants; defined participant which are not currently connected to the conference (disconnected, redial, disconnecting, etc.) are not counted.

Text Indication can be disabled by adding a new flag to the *System Configuration* and setting its value to NO as follows:
ENABLE_TEXTUAL_CONFERENCE_STATUS=NO.

This setting is recommended for MCUs running *Telepresence* conferences. For more information, see the *RMX 2000 Administrator's Guide, "System Configuration"* on page [14-10](#).

Transparent Endpoint Names

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

The *Endpoint Name Transparency* feature can be disabled by adding a new flag to the *System Configuration* and setting its value to NO as follows:
SITE_NAME_TRANSPARENCY=NO.

For more information, see the *RMX 2000 Administrator's Guide, "System Configuration"* on page [14-10](#).

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.

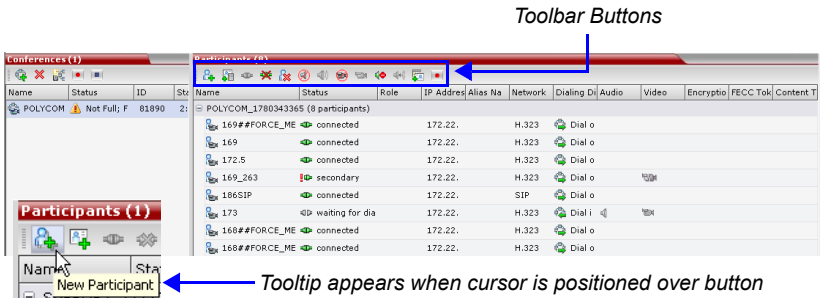
In *MPM mode*, the maximum number of participants (voice and video) that can connect to a conference is 80.

In *MPM+ mode*, the maximum number of participants that can connect to a conference is 200. Of these, 80 can be video participants.

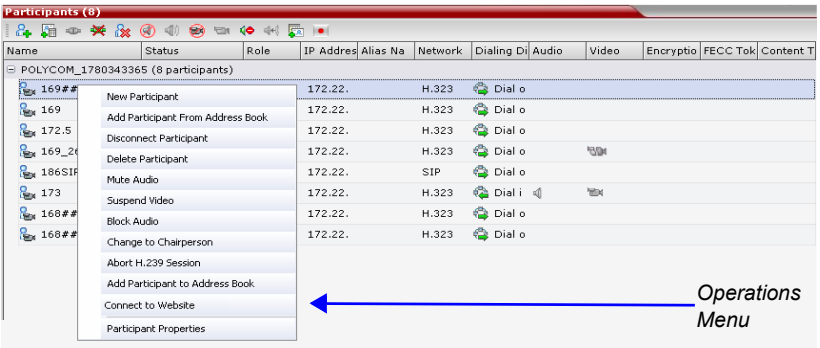
Operation Selection

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 anywhere in the *Conferences* or *Participants* pane and selecting an operation from the menu.

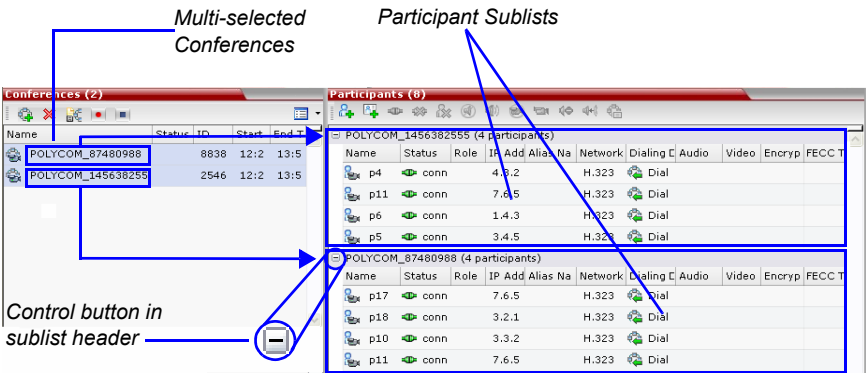


Multi Selection

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

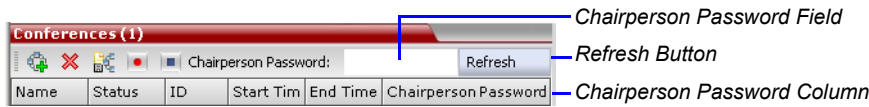
The selected conferences are displayed as sub-lists in the *Participants* list pane.

The sub-lists can be expanded and collapsed by clicking the **+** and **-** sublist control buttons that appear next to the conference name in the sublist headings.



Using the Chairperson Password for Filtering

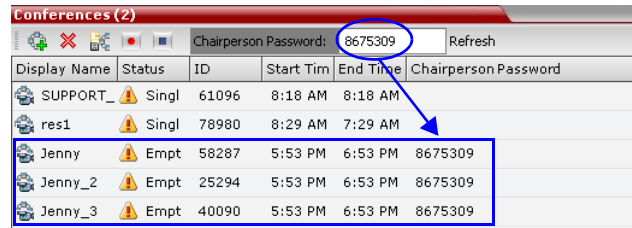
If you are logged in as a chairperson, the *Chairperson Password* field is displayed. It enables you to search for, and display a list of, ongoing conferences for which you have the password.



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.

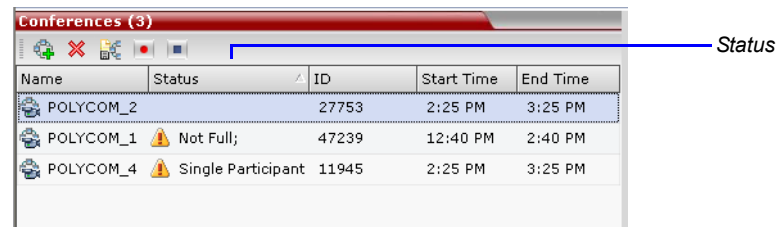


Display Name	Status	ID	Start Time	End Time	Chairperson Password
SUPPORT_	Singl	61096	8:18 AM	8:18 AM	
res1	Singl	78980	8:29 AM	7:29 AM	
Jenny	Empt	58287	5:53 PM	6:53 PM	8675309
Jenny_2	Empt	25294	5:53 PM	6:53 PM	8675309
Jenny_3	Empt	40090	5:53 PM	6:53 PM	8675309

Conference Level Monitoring

Conference level monitoring is available to the administrator, operator and chairperson.

The *Conference List* pane displays information about ongoing conferences.






Name	Status	ID	Start Time	End Time
POLYCOM_2		27753	2:25 PM	3:25 PM
POLYCOM_1	Not Full;	47239	12:40 PM	2:40 PM
POLYCOM_4	Single Participant	11945	2:25 PM	3:25 PM

One or more of the status indicators listed in Table 3-5 may appear in the *Status* column.

No status indicator display in the *Status* column means that the conference is running without problems.

When *Conference Recording* is enabled in a conference's *Profile*, the *Conference Recording* buttons are enabled.

Table 3-5 Conferences – Monitoring Information

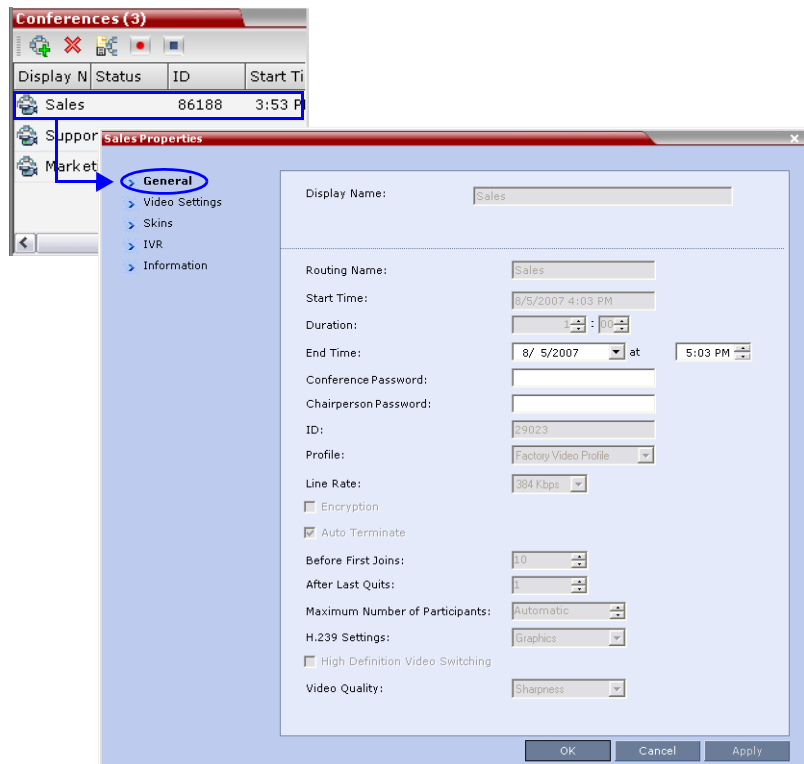
Field	Description
<i>Name</i>	<p>Displays conference name and type of conference:</p> <ul style="list-style-type: none"> •  – Video Conference (including HD CP conferences). •  – High Definition Video Conference running in Video Switching mode. •  – The conference has been secured using the *71 DTMF code.
<i>Status</i>	<p>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, preceded by a warning icon (⚠).</p> <ul style="list-style-type: none"> • Audio – There is a problem with the participant's audio. • Empty – No participants are connected. • Faulty Connection – Participants are connected, but the connection is problematic. • Not Full – Not all the defined participants are connected. • Partially Connected – The connection process is not yet complete; the video channel has not been connected. • Single Participant – Only one participant is connected. • Video – There is a problem with the participant's video.
<i>ID</i>	The Conference ID assigned to the conference.
<i>Start Time</i>	Conference start time.
<i>End Time</i>	The time the conference is expected to end.

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* dialog box appears with the *General* tab open.



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

For more information, see the *RMX 2000 Administrator's Guide*, "Conference Level Monitoring" on page 7-3.

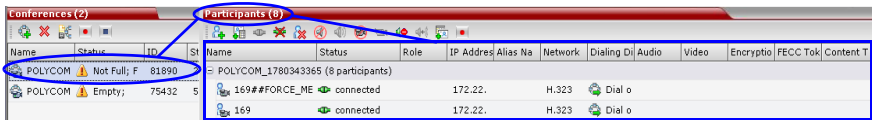
Secured Conference Monitoring

When *Secure Conference Mode* is enabled on the RMX, conference chairpersons can use DTMF codes to secure and unsecure conferences. When a conference has been secured, dial-in and dial-out connections are prohibited, administrators cannot monitor the participants or control the conference. Administrators can manually terminate a secured conference but cannot view the participant list or any of the conference's properties.

Participant Level Monitoring

Participant Connection Monitoring

When a conference is selected in the *Conference List*, details of its participants appear in the *List* pane.



The following participant indicators and properties are displayed:

Table 3-6 Participant Monitoring – Indicators and Properties





Column	Icon/Description
Name	Displays the name and type (icon) of the participant:
	 Audio Participant – Connected via IP phone or ISDN/PSTN.
	 Video Participant – Connected with audio and video channels.
Status	Displays the connection status (text and icon) of the participant. If there is no problem with the participant's connection no indication is displayed.
	 Connected – The participant is successfully connected to the conference.
	 Disconnected – The participant is disconnected from the conference. This status applies only to defined participants.

Table 3-6 Participant Monitoring – Indicators and Properties (Continued)










Column	Icon/Description	
<i>Status (cont.)</i>		Waiting for Dial-in – The system is waiting for the defined participant to dial into the conference.
		Partially Connected – The connection process is not yet complete; the video channel has not been connected.
		Faulty Connection – The participant is connected, but problems occurred in the connection, such as synchronization loss.
		Secondary Connection – The endpoint's video channel cannot be connected to the conference and the participant is connected only via audio.
<i>Role</i>	Displays the participant's role or function in the conference:	
		Chairperson – The participant is defined as the conference chairperson. The chairperson can manage the conference using touch-tone signals (DTMF codes).
		Lecturer – The participant is defined as the conference Lecturer.
		Lecturer and Chairperson – The participant is defined as both the conference Lecturer and Chairperson.
		Cascade Enabled Dial-out Participant – A special participant functioning as a link in a cascaded conference.
		Recording – A special participant functioning as a Recording Link.
<i>IP Address/ Phone</i>	The IP participant's IP address or the ISDN/PSTN participant's phone number.	
<i>Alias Name/ SIP Address</i>	The participant's Alias Name or SIP URI. The alias of an <i>RSS 2000 Recording System</i> if the participant is functioning as a recording link.	

Table 3-6 Participant Monitoring – Indicators and Properties (Continued)











Column	Icon/Description	
<i>Network</i>	The participant's network connection type – H.323, SIP or ISDN/PSTN.	
<i>Dialing Direction</i>		Dial-in – The participant dialed the conference.
		Dial-out – The MCU dialed the participant.
<i>Audio</i>	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.	
		Muted – Participant's audio channel is muted. The participant can still hear the conference.
		Blocked – Transmission of audio from the conference to the participant is blocked.
		Muted and Blocked - Audio channel is muted and blocked.
<i>Video</i>	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.	
		Suspended – Video transmission from the endpoint to the conference is suspended.
		Secondary – Participant is connected only through the audio channel due to problems with the video channel.
<i>Encryption</i>		Indicates that the endpoint is using encryption for its connection to the conference.
<i>FECC Token</i>		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.

Table 3-6 Participant Monitoring – Indicators and Properties (Continued)

Column	Icon/Description	
<i>Content Token</i>		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. For more information, see the <i>RMX 2000 Administrator's Guide</i> , "H.239" on page 8-13 .

For more information, see the *RMX 2000 Administrator's Guide*, "Participant Level Monitoring" on page [7-10](#).

Operations Performed During On Going Conferences

Conference Level operations

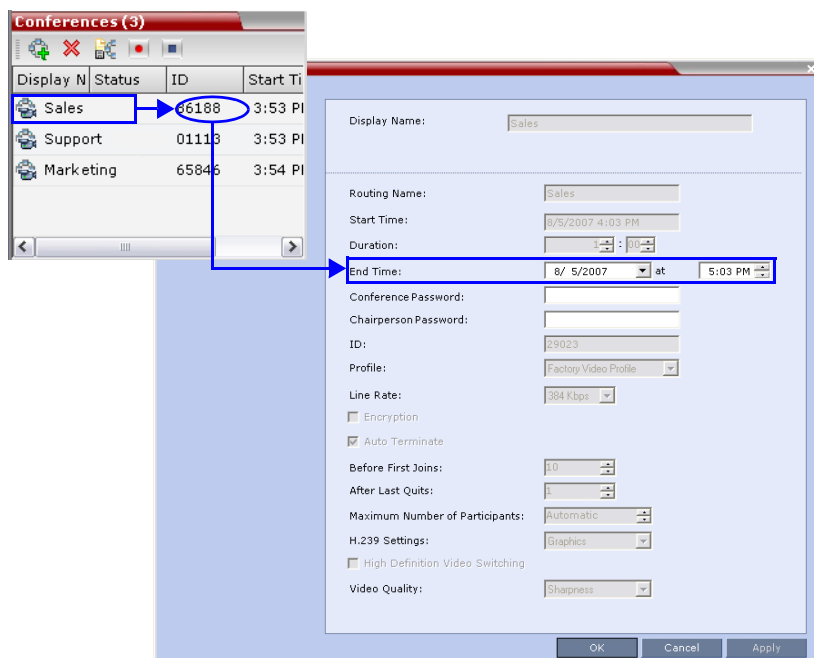
Changing the Duration of a Conference

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 RMX 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**.

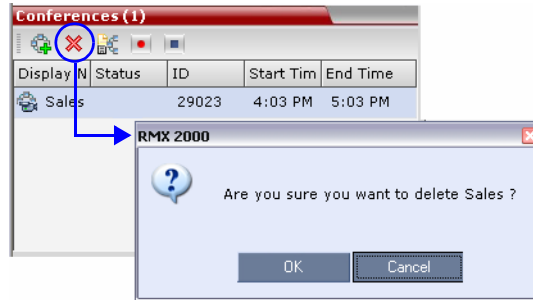


The *End Time* is changed and the *Duration* field is updated.

To terminate a conference manually:

- 1 In the *Conferences* list, select the conference you wish to delete and click the **Delete Conference** (X) button.

You are prompted for confirmation.



- 2 Click **OK** to terminate the conference.

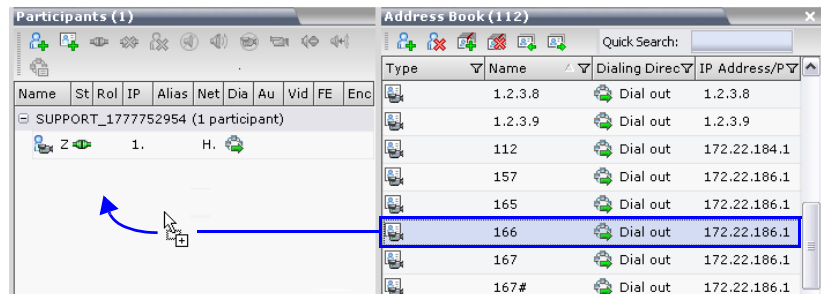
Adding Participants from the Address Book

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.

To drag & drop participants into the Participants List:

- 3 Open the *Address Book*.
- 4 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.

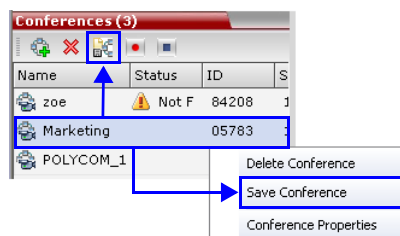


Saving an Ongoing Conference as a Template

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**.



The conference is saved to a template whose name is taken from the ongoing conference *Display Name*.

Changing the Video Layout of a Conference

While the conference is running, you can change the video layout and select one of 24 video layouts supported by the RMX.

Video Layout selection can be done in two levels:

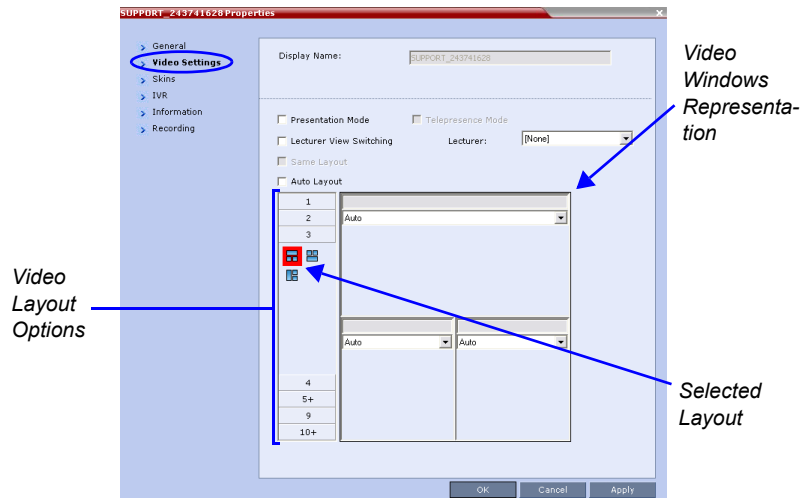
- **Conference Level** – Applies to all conference participants. All participants have the same video layout.
- **Participant Level** – The participant's video layout is changed. All other conference participants' video layouts are not affected.

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

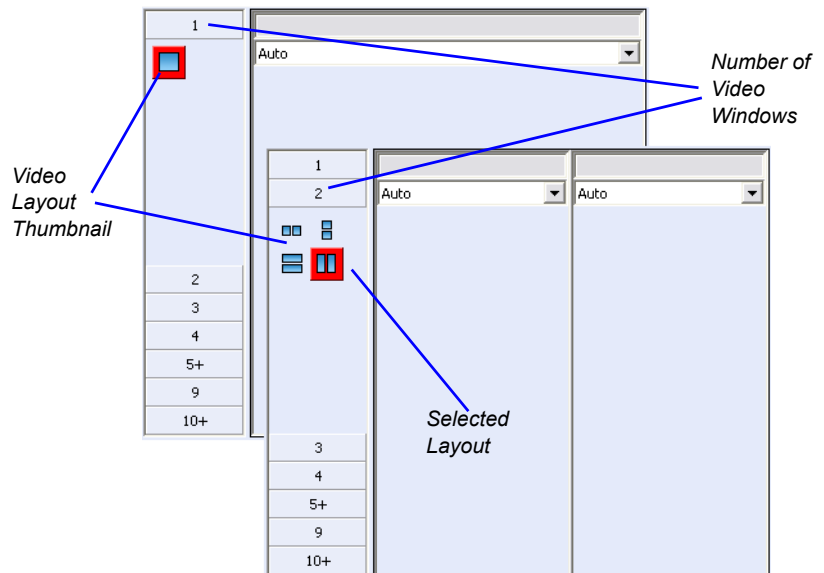
Participant level video layout selection overrides conference level video layout settings.

To change the video layout of a conference:

- 1 In the *Conference Properties* dialog box, select **Video Settings**.



- 2 If **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

Users with Chairperson or Operator permission 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 participants 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.

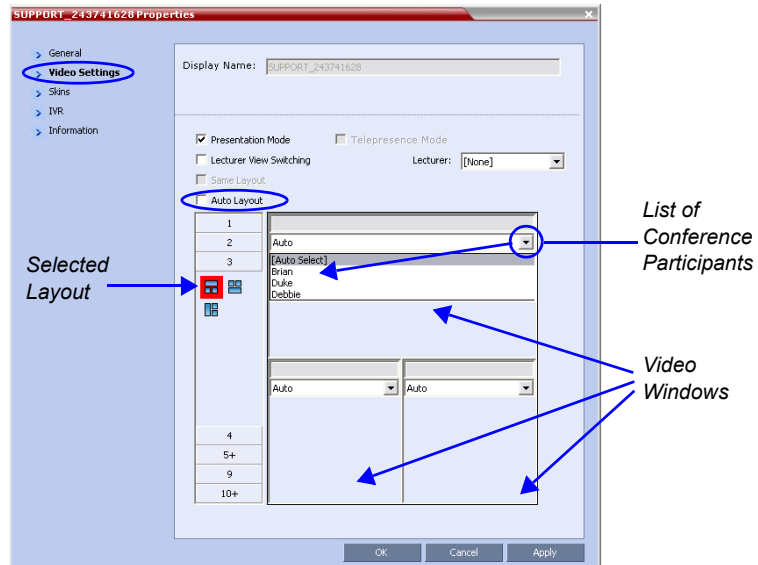
Video Forcing Guidelines:

- A participant cannot appear in two or more windows at the same time.
- Participant level video forcing overrides conference level video forcing.
- A participant can view him/herself in a layout window, by selecting the *Same Layout* option.
- 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 are saved and applied the next time the layout is selected.
- Windows that are not assigned any participant display the current speaker and last speakers.

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**.

Switching between participants is renewed and is audio activated.

Enabling and Disabling Video Clarity™

The user can enable or disable Video Clarity™ during an ongoing conference.

To enable or disable Video Clarity:

- 1** In the *Conference List* pane, double-click the name of the conference for which you want to enable or disable *Video Clarity* or
right-click the conference name and then click **Conference Properties**.
- 2** Click the **Video Settings** tab.
- 3** Select or clear the **Video Clarity** check box as required.
- 4** Click **OK**.

Participant Level Operations

Participant Level Operations enable you to modify and control the connections and statuses of participants in ongoing conferences, as described in Table 3-7.

Table 3-7 *Participant Level Operations*










Menu Option	Button	Description
<i>New Participant</i>		Define a new participant. For more information about the <i>New Participant</i> dialog box tab, see Table 3-3 on page 3-20.
<i>Add Participant From Address Book</i>		Open the <i>Address Book</i> to select the participant for the conference. For more information about the <i>Address Book</i> , see the <i>RMX 2000 Administrator's Guide</i> , "Address Book" on page 4-1.
<i>Connect Participant</i>		Connect a disconnected defined dial-out participant to the conference.
<i>Disconnect Participant</i>		Disconnect the participant from the conference.
<i>Delete Participant</i>		Delete the selected participants from the conference.
<i>Mute Audio</i>		Mute the audio transmission from the participant to the conference. The <i>Audio Muted</i> indicator appears in the <i>Participants List</i> and the <i>Unmute Audio</i> button () becomes active.
<i>Unmute Audio</i>		Resume the participant's audio transmission to the conference. The <i>Mute Audio</i> button () becomes active.

Table 3-7 Participant Level Operations (Continued)










Menu Option	Button	Description
<i>Suspend Video</i>		Suspend the video transmission from the participant to the conference. The suppressed participant's video is not transmitted to the conference but the participant still receives conference video. The <i>Suspend Video</i> indicator appears in the <i>Participants List</i> and the <i>Resume Video</i> button () becomes active.
<i>Resume Video</i>		Resume the participant's video transmission to the conference. The <i>Suspend Video</i> button becomes active ()
<i>Block Audio</i>		To block the audio transmission from the conference to the participant. When blocked, the participant can still be heard by the conference. The <i>Audio Blocked</i> indicator appears in the <i>Participants List</i> and the <i>Unblock Audio</i> button () becomes active.
<i>Unblock Audio</i>		Resume the audio transmission from the conference to the participant. The <i>Block Audio</i> button () becomes active.
<i>Add Participant to Address Book</i>		Add selected participant's details to the <i>Participant Address Book</i> .
<i>Abort H.239 Session</i>		To withdraw the Content Token from the participant back to the MCU for re-assignment.
<i>Change to Chairperson</i>		Define the selected participant as the conference leader/chairperson.
<i>Change to Regular Participant</i>		Define the chairperson as a regular participant without chairperson privileges.

Table 3-7 *Participant Level Operations (Continued)*

Menu Option	Button	Description
<i>Connect to Website</i>		Connect directly to the internal website of the participant's endpoint to perform administrative, configuration and troubleshooting activities.
<i>AGC (Auto Gain Control)</i>		Enable AGC for the participant with weak audio signal during ongoing conferences. Note: Enabling AGC may result in amplification of background noise.
<i>Participant Properties</i>		To view all <i>Participant Properties</i> . For more information, see the <i>RMX 2000 Administrator's Guide</i> , " <i>Participant Level Monitoring</i> " on page 7-10 .

Personal Layout Control with the RMX Web Client

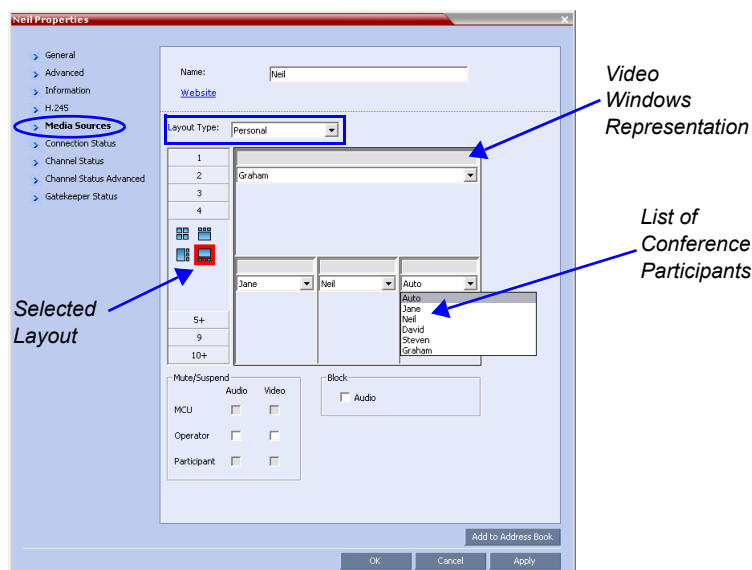
RMX users can use the *RMX Web Client* to change the *Video Layouts* of individual participants and force participants to its windows without affecting the *Video Layouts* of other participants.

To change a participant's Video Layout and Video Forcing:

- 1 In the *Participants* list, double click the participant or right-click the participant and then click **Participant Properties**.

The *Participant Properties – Media Sources* dialog box opens.

- 2 In the *Layout Type* list, select **Personal**.



- 3 Select the number of Video Windows.
- 4 Select the required Video Layout.
- 5 To video force participants to windows in the selected video layout, in the window to which you want to force a participant, select the name of the participant to force from the list of conference participants.
- 6 Repeat step 5 to force participants to other windows.
- 7 Click **OK**.

To cancel the Personal Video Layout selection and return to the conference layout:

- 1 In the *Participant Properties* dialog box, select the **Media Sources** tab.
- 2 In the *Layout Type* list, select **Conference**.
- 3 Click **OK**.

The participant will now see the conference video layout with its forced participants.

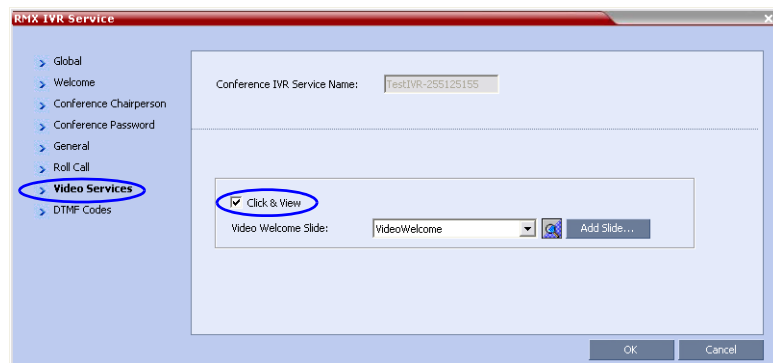
To cancel the Personal Video Forcing for a window without returning to the conference layout:

- 1 In the *Participant Properties – Media Sources* dialog box, in the video layout window, select **Auto** in the *Participants* list.
- 2 Click **OK**.

Switching between participants is renewed and is audio activated.

Personal Layout Selection with Click&View

With the **Click&View** application, participants can change their *Personal Layouts* via DTMF codes entered from their endpoints. This option is available only if the **Click&View** option is selected in the *Conference IVR Service*.



To change Personal Layout with Click&View:

- 1 **Enable Click&View** – on the endpoint's keypad, enter .

The *Click&View* application is displayed on the screen.



When using a *Polycom VSX* endpoint, an additional must be entered to enable the remote DTMF keypad. The full *Click&View* entry sequence is: , .

The Personal Layout keypad options menu is displayed on the video screen.



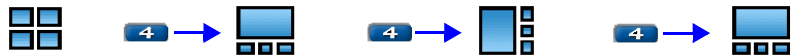
- 2** On the endpoint's remote keypad, press the number corresponding to the number of video squares you wish to select.

For example, if you want a four-square video layout, press **4**.

The video window layout of your screen changes to the first four-window layout as follows:



Repeated presses of the **4** key, within eight seconds, cycles through the following series of four-square layout options:


















In any multi-square layout, pressing **#** forces the current speaker to the top left window.

In full view, pressing **#** forces the next participant to full view.

In any video layout, pressing **0** reverts to the conference layout.

The following table summarizes the Video Layout options available via *Click&View*.

Table 3-8 Video Layout Options

DTMF Code	Layout Options
1	
2	   
3	  
4	   
5	  
6	
8	
9	   

Conference Control Using DTMF Codes

Participants and chairpersons can manage their connection to ongoing conferences from their endpoints, using touch-tone signals (DTMF codes) from their endpoints. Table 3-9 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.

For more information, see the *RMX 2000 Administrator's Guide*, "Defining a New Conference IVR Service" on page [12-9](#).

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

Table 3-9 Conference IVR Service Properties - DTMF Codes

Operation	DTMF String	Permission
Mute My Line	*6	All
Unmute My Line	#6	All
Increase Broadcast Volume	*9	All
Decrease Broadcast Volume	#9	All
Mute All Except Me	*5	Chairperson
Cancel Mute All Except Me	#5	Chairperson
Change Password	*77	Chairperson
Mute Incoming Participants	*86	Chairperson
Unmute Incoming Participants	#86	Chairperson
Play Help Menu	*83	All
Enable Roll Call	*32	Chairperson
Disable Roll Call	#32	Chairperson
Roll Call Review Names	*33	Chairperson
Roll Call Stop Review Names	#33	Chairperson
Terminate Conference	*87	Chairperson
Start Click&View	**	All
Change To Chairperson	*78	All
Increase Listening Volume	*76	All
Decrease Listening Volume	#76	All
Override Mute All	Configurable	All
Secure Conference	*71	Chairperson
Unsecure Conference	#71	Chairperson
Show Participants	*88	All

Appendix A

Glossary

This appendix lists the terms and abbreviations that are related to the Polycom RMX 2000, and are commonly used in the RMX 2000 documentation.

Abbreviation/ Term	Explanation
<i>AGC</i>	Auto Gain Control. A mechanism that regulates noise and audio volume by keeping the received signal of all participants balanced.
<i>Bandwidth</i>	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.
<i>Bonding</i>	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.
<i>Bps, Kbps</i>	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

Abbreviation/ Term	Explanation
<i>BRI</i>	Basic Rate Interface. A type of ISDN connection for transiting data, consisting of 3 channels: two B-channels (each of 64 Kbps) and one D-channel (16 Kbps).
<i>Carrier</i>	A telephone or other company that provides telecommunication transmission services.
<i>CIF, 4CIF, QCIF</i>	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.
<i>Codec</i>	<u>Coder-decoder</u> . 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 videoconferencing.
<i>Conference</i>	Connection between two or more endpoints exchanging video and audio information. If only two endpoints are involved, a conference is called <i>point-to-point</i> and no MCU is required. If more than two endpoints are involved, it is called a <i>multipoint</i> conference, and an MCU (Multipoint Control Unit) is required as the management system. For more information, see MCU.
<i>CSU</i>	Channel Service Unit. Customer-provided equipment that is used as an interface between a communication network and the data terminal.
<i>DBA</i>	Dynamic Bandwidth Allocation. Used to allocate the bandwidth needed to transmit the additional packets. for LPR.

Abbreviation/ Term	Explanation
<i>DTMF</i>	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.
<i>E1 Line</i>	A 2Mb digital switched line used in Europe.
<i>Endpoint</i>	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).
<i>FECC</i>	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.
<i>Frame</i>	A group of bits that make up an elementary block of video data for transmission by certain protocols.
<i>Frame Rate</i>	The number of video frames displayed on-screen during one second, measured in fps (frames per second).
<i>G.711</i>	ITU-T audio algorithm, 64Kbps, 3.4 kHz.
<i>G.722</i>	ITU-T audio algorithm, 64Kbps, 7 kHz.
<i>G.728</i>	ITU-T audio algorithm, 16Kbps, 3.4 kHz.
<i>Gatekeeper</i>	A type of server that performs two main functions: translates LAN alias addresses of terminals and gateways to IP addresses and provides bandwidth management.
<i>H.221</i>	ITU-T standard that defines how to multiplex video, audio, control, and user data into one serial bit stream.

Abbreviation/ Term	Explanation
<i>H.230</i>	ITU-T standard that defines simple multipoint control systems procedures and describes network maintenance functions.
<i>H.231</i>	ITU-T standard that defines a set of MCU functions and operational requirements.
<i>H.242</i>	ITU-T standard that defines initiation of communications between systems and capabilities negotiation procedures.
<i>H.243</i>	ITU-T standard that defines initiation of communications between systems and capabilities negotiation procedures in multipoint conferences.
<i>H.261</i>	ITU-T standard that defines the Px64 video coding algorithm.
<i>H.263</i>	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.
<i>H.264*</i>	A proprietary Polycom Video compression standard.
<i>H.264</i>	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.
<i>H.320</i>	ITU-T standard that defines how the H-series video conferencing recommendations work together.
<i>H.323</i>	ITU-T standard for audio, video and data communications across IP-based (LAN) networks, including the Internet.
<i>IP</i>	Internet Protocol. The working protocol that forms the basis of the internet.
<i>ISDN</i>	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.

Abbreviation/ Term	Explanation
<i>ITU-T Standard</i>	International Telecommunications Union, Telecommunication Standardization Sector (formerly CCITT). An international group that produces official standards for telecommunications.
<i>LAN</i>	Local Area Network. A group of computers and other devices linked via a network's operating system.
<i>LDAP</i>	Lightweight Directory Access Protocol.
<i>Line Rate</i>	The amount of bandwidth used by a communication device, measured in Kbps (kilobits per second).
<i>LPR</i>	Lost Packet Recovery. An algorithm that creates additional packets that contain recovery information necessary to reconstruct lost packets.
<i>MCU</i>	Multipoint Control Unit. Device which allows more than two sites to be connected in a video conference.
<i>Null modem cable</i>	A serial cable designed to eliminate the need for communication equipment when two digital devices are directly connected to each other.
<i>Participant</i>	A person using an endpoint to connect to a conference. When using a Room System, several participants use a single endpoint.
<i>PRI</i>	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.
<i>PSTN</i>	Public Switched Telephone Network.
<i>QCIF</i>	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.
<i>QoS</i>	Quality of Service. QoS defines the performance of a network service, such as the average delay between packets.

Abbreviation/ Term	Explanation
<i>RS-232</i>	A standard for serial interface connection.
<i>SIP</i>	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.
<i>Span</i>	An ISDN line or leased line. A span may be of either T1 (United States) or E1 (Europe) type. Also called a circuit.
<i>T1 Line</i>	A 1.5Mb digital switched line used in the United States.
<i>ToS</i>	Type of Service. ToS defines optimization tagging for routing audio and video packets.
<i>WAN</i>	Wide Area Network. A communications network that services a geographical area larger than the LAN.
<i>Whiteboard</i>	An on-screen shared notebook for placement of shared documents.