

RMX 2000

Getting Started Guide

Version 2.0



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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 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 2 – First Time Installation and Configuration*

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.

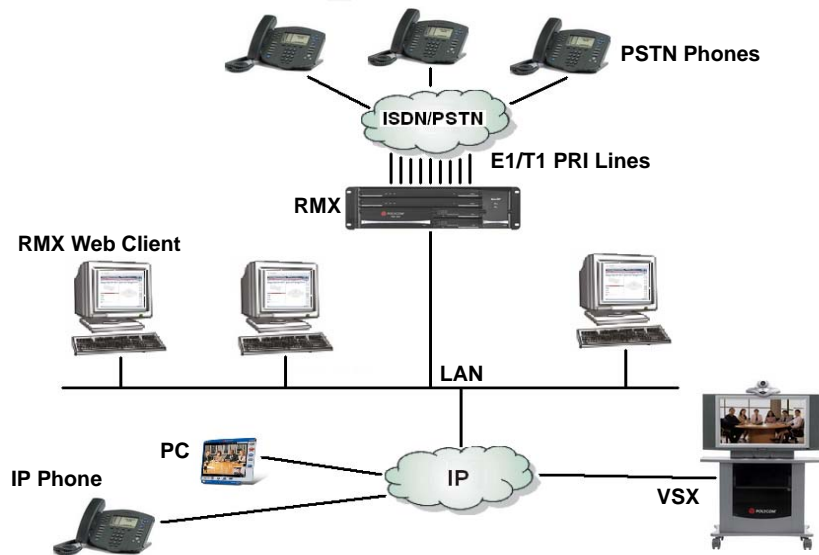


Figure 1-1 Multipoint Video Conferencing using a Polycom RMX 2000

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 ISDN/PSTN cards are supported, each providing connection for up to 7 E1 or 9 T1 PRI lines.



RMX Main Features

Video Display

Dynamic Continuous Presence

The dynamic Continuous Presence capability of the RMX system enables viewing flexibility by offering multiple viewing options and window layouts for video conferencing. The Continuous Presence mode offers 24 layouts to accommodate different numbers of participants and conference settings including support of the VUI annex to the H.264 protocol for endpoints that transmit wide video instead of 4CIF resolution.

Table 1-1 Continuous Presence – Video Layouts

Standard Definition (SD)

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 for PAL systems and 720X480 for NTSC systems. Bit rates for SD range from 256Kbps to 2Mbps.

High Definition (HD)

HD is an ultra-high quality video resolution enabling compliant endpoints to connect to conferences at resolutions of 1280x720 (720p) and at bit rates ranging from 384kbp to 4Mb.

HD is supported in two modes:

- **Continuous Presence** – By default every conference, Entry Queue and Meeting Room has the ability to declare the maximum CP resolution. This includes conferences launched by the *RMX Web Client* and conferences started via the API
- **High Definition Video Switching** – Offers better video quality than HD CP and uses less system resources. *Video Layout* is 1x1 only and the conference requires:
 - a dedicated profile
 - a dedicated Entry Queue
 - all participants to have HD compliant endpoints
 - all participants to connect using the same conference line rateEndpoints that are unable to meet these requirements connect as Secondary (audio only).

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

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.

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

Media Encryption

Encryption is available at the conference and participant levels, based on AES 128 Media Encryption and DH 1024 Key Exchange standards.

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 during a conferences 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

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 (PSTN), voice prompts from an IVR service are used to connect the participants to the appropriate conference.

This service also enables the system to verify the participant's right to start an Ad Hoc conference or to join an on going conference.

Conferencing Capabilities and Options

On Demand Conferencing

The following options are available when setting up conferences:

- New Conference – setup once, use once
The conference is deleted from the MCU after it ends
- Meeting Rooms – setup 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 is created when a user dials in

Connection Methods

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

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
- 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

First Time Installation and Configuration

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

1 Hardware Installation and Setup

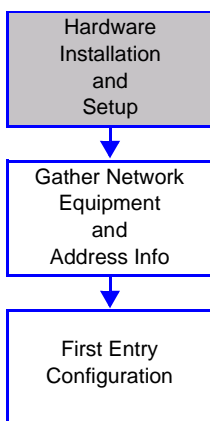
- Mount the RMX in a rack.
- Connect the necessary cables.

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 Service*.



Hardware Installation and Setup

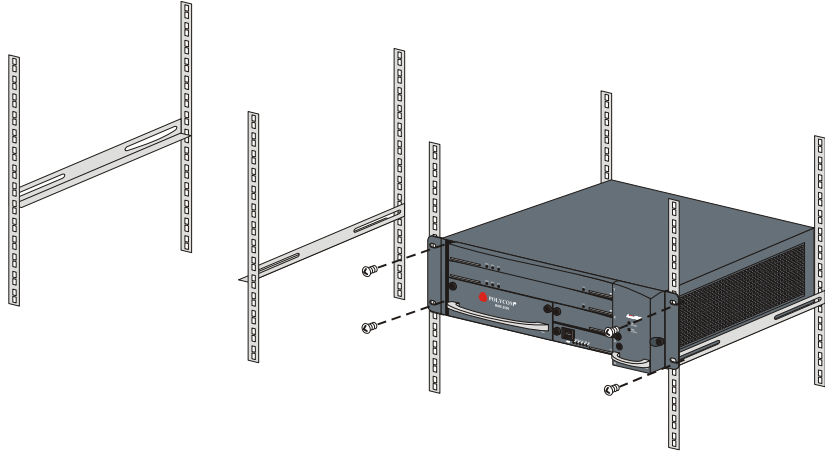
The RMX unit should be mounted in a 19" rack in a well ventilated area. It is important to adhere to the *Site Requirements* as described in the *RMX 2000 Hardware Guide*, "*Site Requirements*" on page 1-3.

Mounting the RMX in a Rack

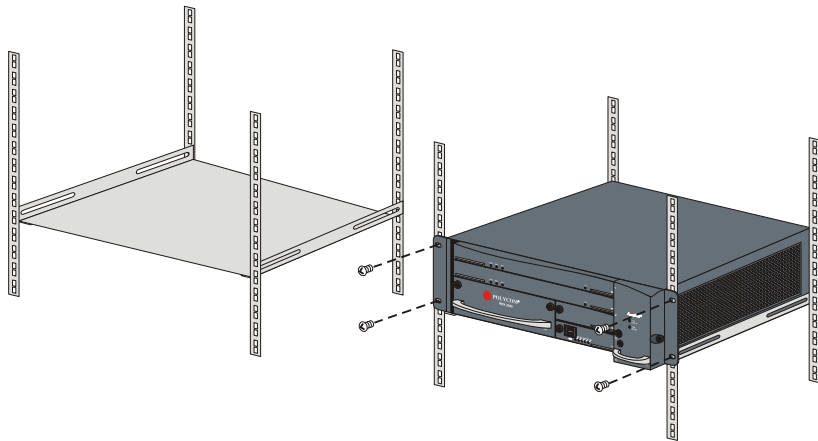
There are two methods for installing the RMX in a rack:

- **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.



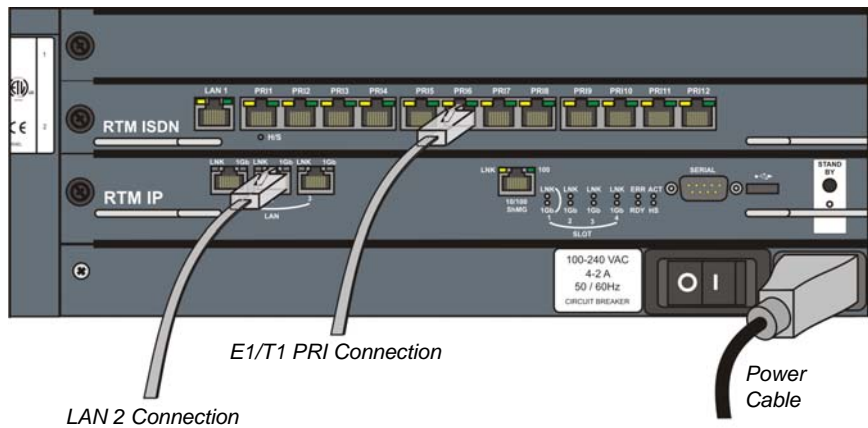
Connecting Cables



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

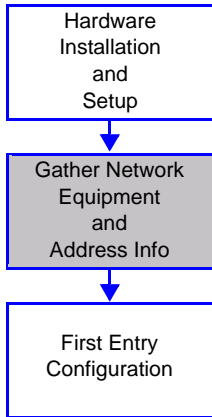
Connect the following cables to the back panel:

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



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 *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
<i>Control Unit IP Address</i>	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 Network Service is used to define the properties of the ISDN/PSTN switch and the ISDN lines running from the ISDN switch to the ISDN card installed in the RMX.

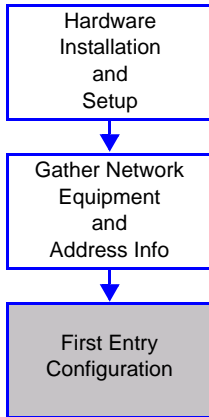
Before configuring the ISDN/PSTN Network Service, obtain the following information from your ISDN/PSTN Service Provider:

- 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: www.eservice.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*.

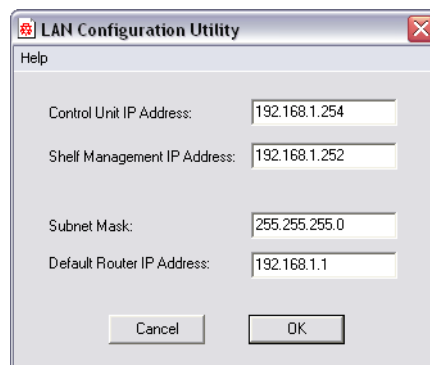
For more information, see the RMX 2000 Administrator's Guide, Appendix F: "Configuring Direct Connections to RMX" on page F-1.

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

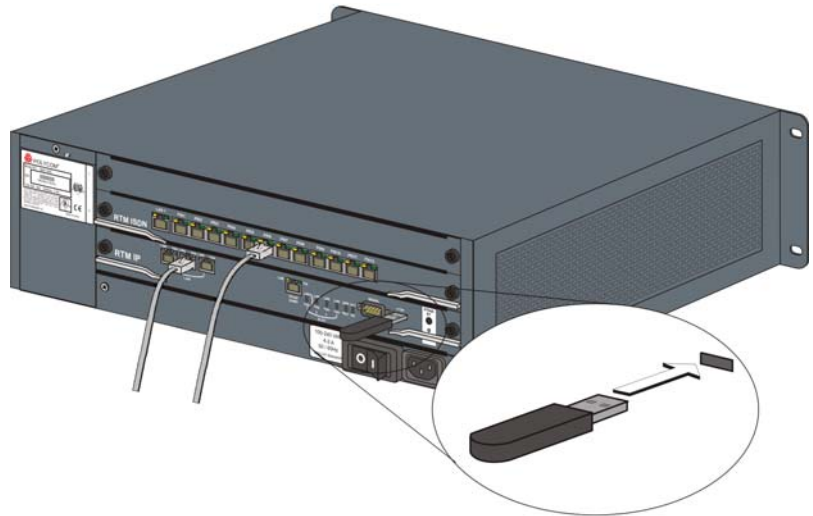


- 2 Modify the following parameters in the utility's dialog box using the information supplied by your network administrator.
 - Control Unit IP Address
 - Shelf Management IP Address
 - *Subnet Mask*
 - *Default Router IP Address*
- 3 Click **OK**.

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.

The First-time Power-up is complete when the red *ERR* LED on the RMX's front panel remains ON.

On subsequent power-ups, with a defined Management Network Service, the red *ERR* LED remains OFF if there is no System Error.

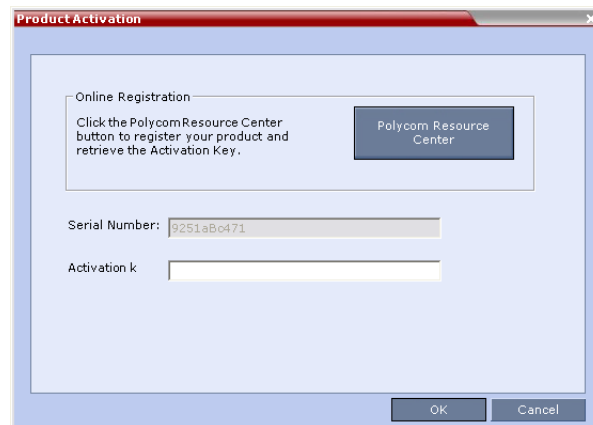
Power -up is complete when the green *RDY* LED of the CNTL module (on the RMX's front panel) turns ON.

- 3 Remove the *USB key*.
- 4 Start the *RMX Web Client* application on the workstation, by entering **http://<Control Unit IP Address>** as defined in the *USB key* in the browser's address line and pressing **Enter**.
- 5 **Optional:** Select the language for the *RMX Web Client*. The language is selected by clicking on the corresponding country flag.


If a language is selected the *RMX Web Client* that is not supported by the browser or the workstation's operating system, the *RMX Web Client* is displayed in English.

- 6 In the *RMX Web Client Welcome 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:

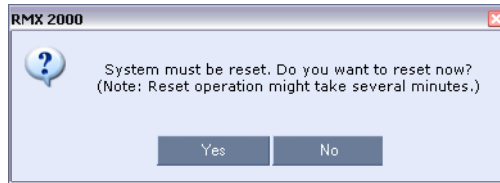


- 7 In the *Activation Key* field, enter or **paste** the *Product Activation Key* obtained earlier and click **OK**.

If you do not have an *Activation Key*, click the  button 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 reset 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 and ISDN/PSTN 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 reset.

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

Fast Configuration Wizard

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

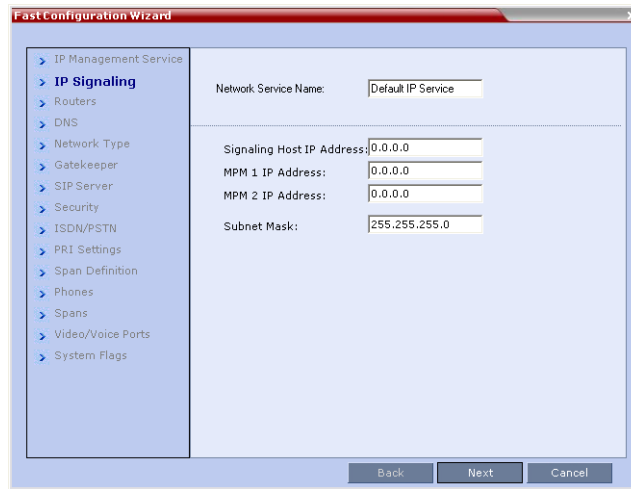


Table 2-2 Fast Configuration Wizard – IP

Field	Description
<i>Network Service Name</i>	The name of the Default IP Service, assigned by the Fast Configuration Wizard. Note: This field is displayed in all dialog boxes.
<i>Signaling Host IP Address</i>	The address used by endpoints when dialing in to the MCU. It is used by the MCU for dialing out as well as for Gatekeeper and SIP Proxy registration.

Table 2-2 Fast Configuration Wizard – IP

Field	Description
MPM 1 IP Address	Endpoints connect to conferences and transmit call media (video, voice and content) via these addresses.
MPM 2 IP Address	
Subnet Mask	Enter the subnet mask of the MCU. Default value: 255.255.255.0.

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

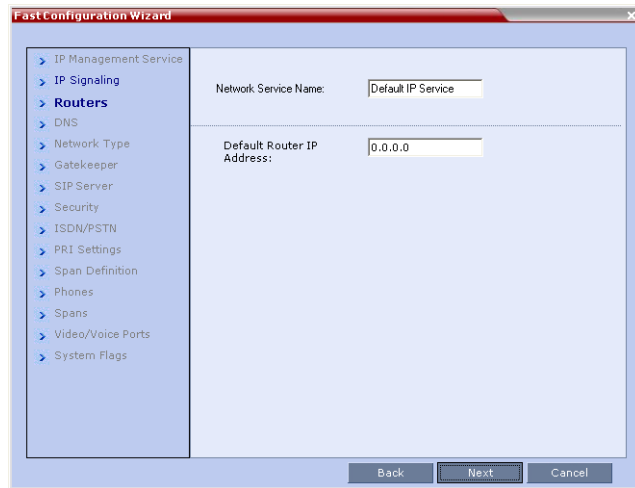


Table 2-3 Fast Configuration Wizard – Routers

Field	Description
Default Router IP Address	Enter the IP address of the default router.

- 4 Click the **Next** button.

5 Enter the required DNS information in the dialog box.

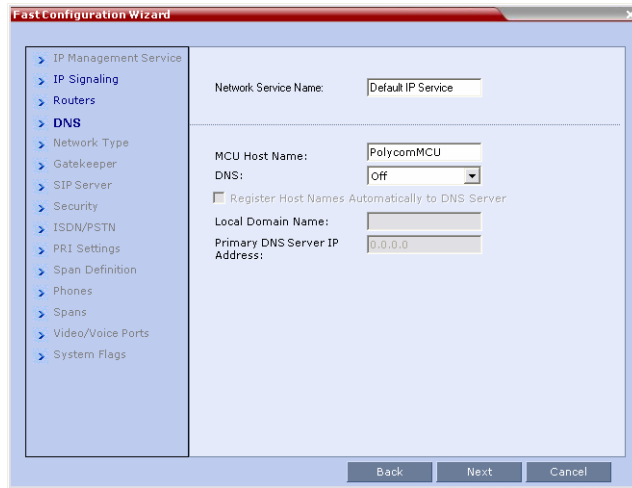
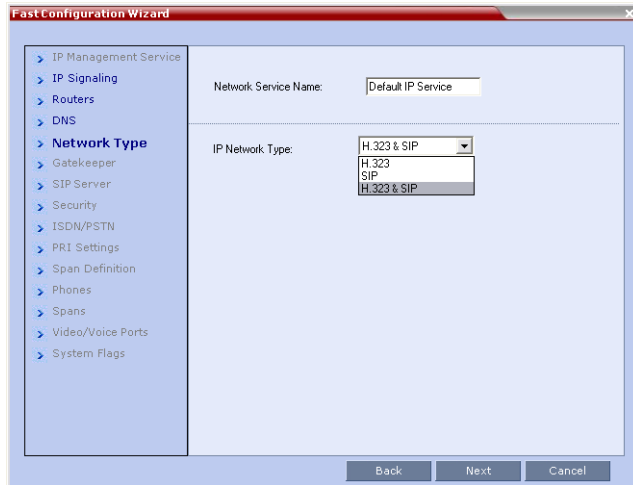


Table 2-4 Fast Configuration Wizard – DNS

Field	Description
<i>MCU Host Name</i>	Enter the name of the MCU on the network. Default name is RMX
<i>DNS</i>	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.
<i>Register Host Names Automatically to DNS Server</i>	Select this option to automatically register the MCU Signaling Host and Shelf Management with the DNS server.
<i>Local Domain Name</i>	Enter the name of the domain where the MCU is installed.
<i>Primary DNS Server IP Address</i>	The static IP addresses of the primary DNS server.

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



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

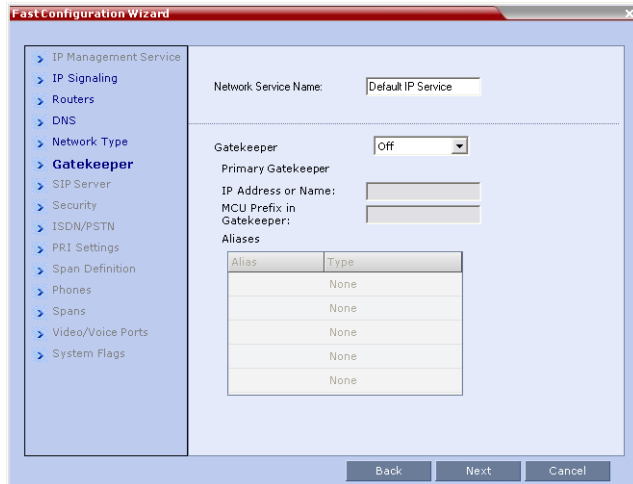


Table 2-5 Fast Configuration Wizard – Gatekeeper

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 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, * #) • URL ID (URL style address) • Transport ID (IP address: port number) • Email ID (email address format) • Party Number (identical to the E.164 format) Note: Although all types are supported, the type of alias to be used depends on your gatekeeper's capabilities.

- 11 Click the **Next** button.
- 12 If you selected **H.323** only, go to **Step 15**.
- 13 Enter the required **SIP Server** information in the dialog box.

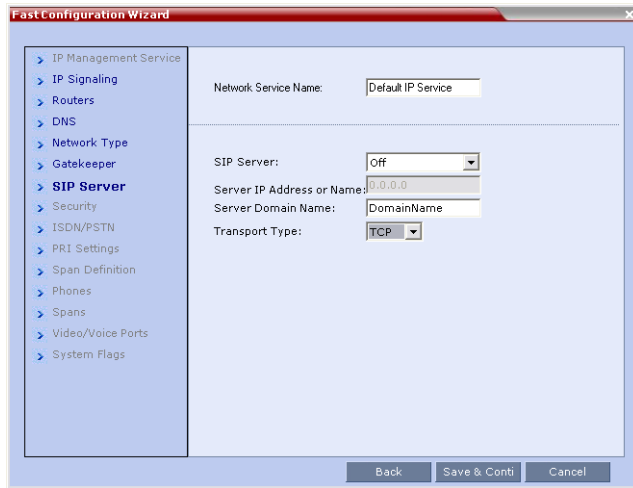


Table 2-6 Fast Configuration Wizard – SIP Server

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:

Table 2-6 Fast Configuration Wizard – SIP Server

Field	Description
<i>Transport Type (cont.)</i>	<p>UDP – Select this option to use UDP for signaling.</p> <p>TCP – Select this option to use TCP for signaling.</p> <p>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. The following protocols are supported:</p> <ul style="list-style-type: none"> • TLS 1.0 • SSL 2.0 • SSL 3.0.

14 Click the **Next** button.

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

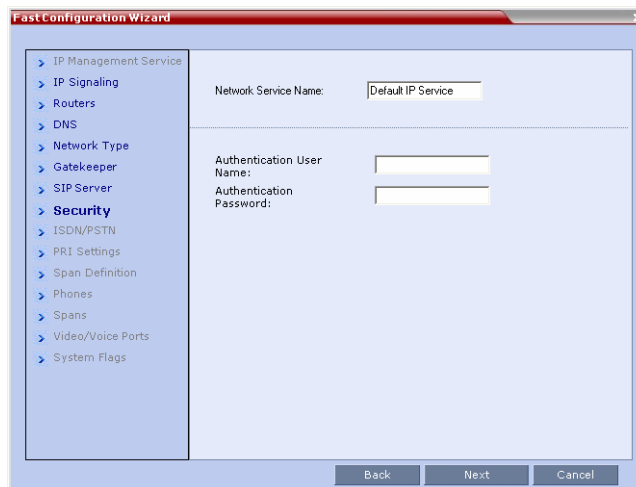


Table 2-7 Fast Configuration Wizard – Security

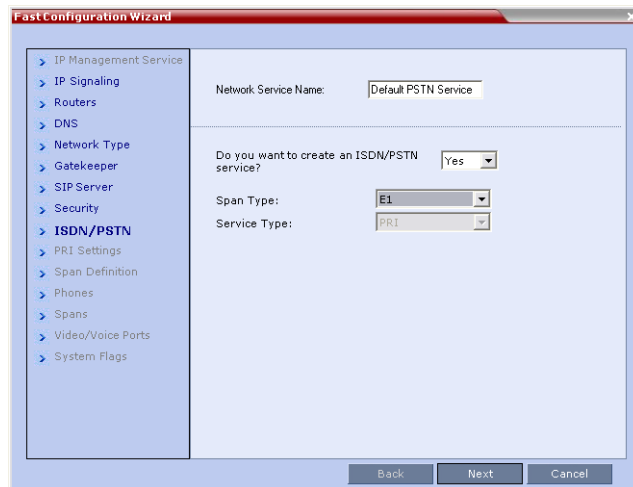
Field	Description
<i>Authentication User Name</i>	Enter the conference, Entry Queue or Meeting Room name as registered with the proxy. This field can contain up to 20 ASCII characters.
<i>Authentication Password</i>	Enter the conference, Entry Queue or Meeting Room password as defined in the proxy. This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

15 Click the **Save &Continue** button.

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 34**.

The *Fast Configuration Wizard's* ISDN/PSTN configuration sequence begins with the *ISDN/PSTN* dialog box:



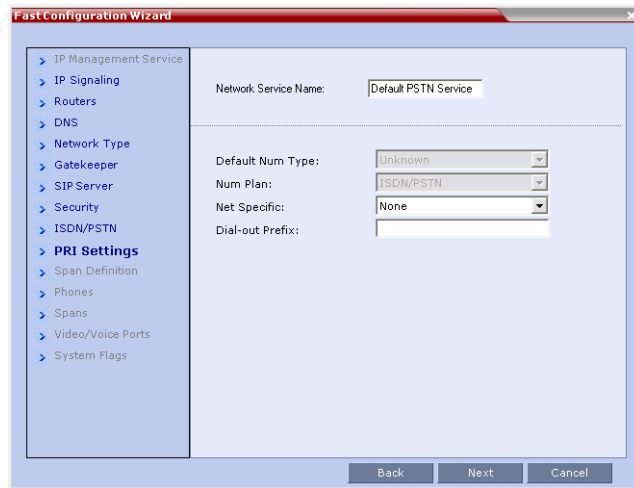
16 Define the following parameters:

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. Default name: PSTN Service 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>Spans are 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. Select the Span Type from the drop-down menu:</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 Note: Only one <i>Span Type</i>, E1 or T1, is supported on the RMX. The <i>Span Type</i> of the first ISDN/PSTN Service defined determines the <i>Span Type</i> of subsequently defined ISDN/PSTN Services.</p>
<i>Service Type</i>	<p>PRI is the only supported service type. It is automatically selected.</p>

17 Click Next.

The *PRI Settings* dialog box opens.



18 Define the following parameters:

Table 2-9 *Fast Configuration Wizard – PRI Settings*

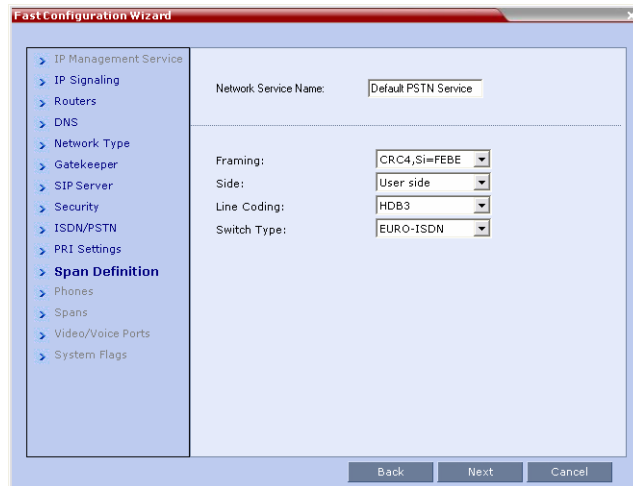
Field	Description
<i>Default Num Type</i>	<p>The Num Type defines how the system handles the dialing digits. For example, if you type eight dialing digits, the Num Type defines whether this number is national or international.</p> <p>Select the Default Num Type from the list.</p> <p>If the PRI lines 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

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>	Some service providers (carriers) 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

19 Click **Next**.

The *Span Definition* dialog box opens.



20 Define the following parameters:

Table 2-10 *Fast Configuration Wizard – Spans Definition*

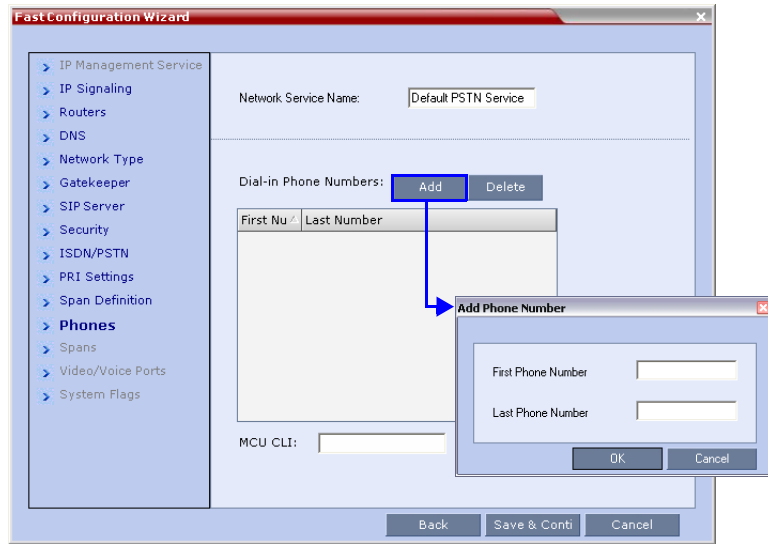
Field	Description
<i>Framing</i>	The frame format used by the carrier for the network interface. Select the Framing format from the list. <ul style="list-style-type: none"> For T1 spans, default is SFBSF. For E1 spans, default is FEBSF.
<i>Side</i>	Select one of the following options: <ul style="list-style-type: 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.

21 Click **Next**.

22 The *Phones* dialog box opens:

23 To define dial-in number ranges click the **Add** button.

The *Add Phone Number* dialog box opens.



24 Define the following parameters:

Table 2-11 *Fast Configuration Wizard – Phones Settings*

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 cannot exceed 1000 numbers.

25 Click **OK**.

The new range is added to the table.

26 **Optional.** Repeat steps **23** to **25** to define additional dial-in ranges.

27 Enter the *MCU CLI* (Calling Line Identification).

In a 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.

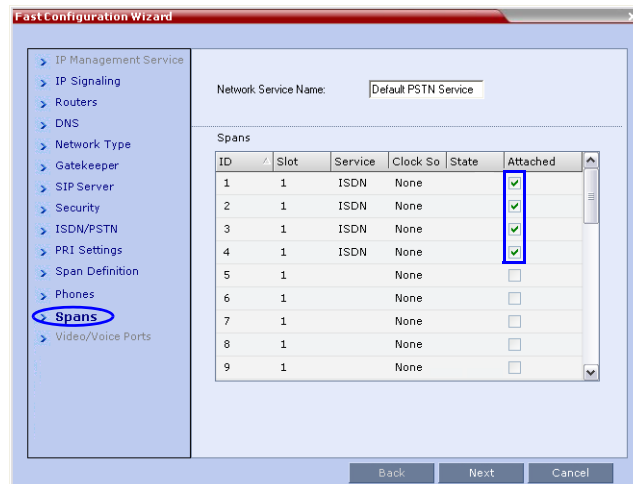
The ISDN/PSTN Network Service is created and confirmed.



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

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

- **ID** - the connector on the ISDN/PSTN card (PRI1 - PRI12).
- **Slot** - the MPM board that the ISDN/PSTN card is connected to (1 or 2)
- **Service** - the network service assigned (ISDN)
- **Clock Source** - the RTM board providing the clock source. The first span to synchronize becomes the primary clock source.
- **State** - the ISDN/PSTN card's clock source (Primary, Backup or None)

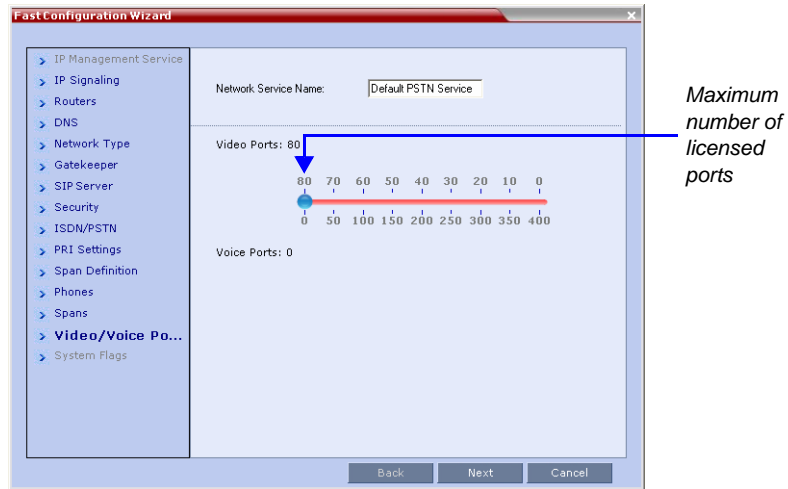


30 Attach spans to existing network services, by marking the appropriate checkboxes in the *Attached* field.

Each ISDN/PSTN card can support 7 E1 or 9 T1 PRI lines.

31 Click **Next**.

The Video/Voice Ports dialog box opens.



The maximum number of video ports displayed in the dialog box is taken from the license key. Only this number can be converted into voice ports.

32 Move the slider to the required setting.

The slider moves only in multiples of 10, converting video ports to voice ports in groups of ten, with each video port converting to five voice ports.

All available ports are initially allocated as video ports.

33 Click **Save & Close**.

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

If the system cannot create the 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.

The *System Flags* dialog box is displayed.

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

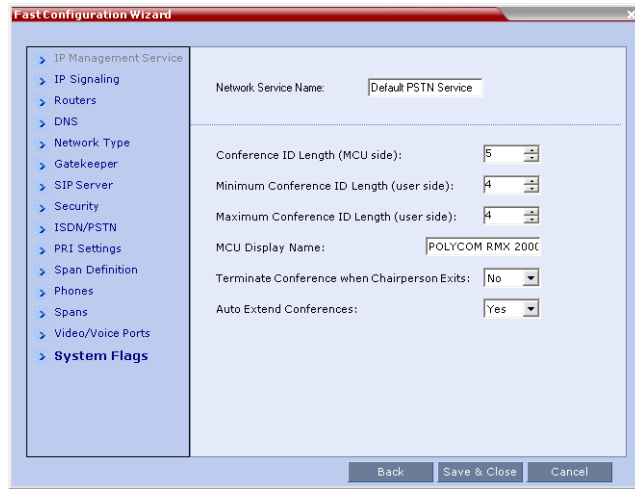


Table 2-12 Fast Configuration Wizard – System Flags

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

Field	Description / Default
<i>Terminate Conference when Chairperson Exits</i>	When Yes is selected (default), the conference end when the chairperson exits even if there are other participants connected.
<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, via the *Setup* menu's *System Configuration* option. For more information, see the *RMX 2000 Administrator's Guide*, "System Configuration" on page 11-5.

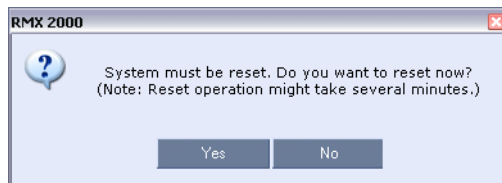
- 35** Click **Save & Close**.

The RMX confirms successful configuration.

- 36** In the *Success Message* box, click **OK**.

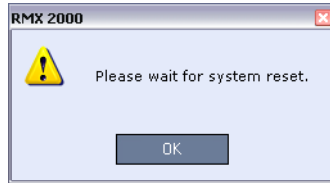


The RMX requests confirmation of reset.



- 37** Click **Yes**.

38 The *Please wait for system reset* request is displayed.



39 Click **OK**.

The language selection options of the *RMX Web Client* can be modified via the *Multilingual Setting* of the system.

For more information see the *RMX 2000 Administrator's Guide "Logger Diagnostic Files"* on page [11-35](#).

RMX's Default Conferencing Settings

The RMX is shipped with default 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										
<i>Meeting Rooms</i>	<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 border="0"> <thead> <tr> <th>Name</th> <th>ID</th> </tr> </thead> <tbody> <tr> <td><i>Maple_Room</i></td> <td><i>1001</i></td> </tr> <tr> <td><i>Oak_Room</i></td> <td><i>1002</i></td> </tr> <tr> <td><i>Juniper_Room</i></td> <td><i>1003</i></td> </tr> <tr> <td><i>Fig_Room</i></td> <td><i>1004</i></td> </tr> </tbody> </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	<i>Maple_Room</i>	<i>1001</i>	<i>Oak_Room</i>	<i>1002</i>	<i>Juniper_Room</i>	<i>1003</i>	<i>Fig_Room</i>	<i>1004</i>
Name	ID										
<i>Maple_Room</i>	<i>1001</i>										
<i>Oak_Room</i>	<i>1002</i>										
<i>Juniper_Room</i>	<i>1003</i>										
<i>Fig_Room</i>	<i>1004</i>										
<i>Conference Profile</i>	<p>Name: <i>Factory Video Profile</i></p> <p>A Conference Profile is assigned to a Meeting Room to define its conferencing properties. <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>										
<i>IVR Service</i>	<p>Name: <i>Conference IVR Service</i></p> <p>The Conference IVR Service contains a set of voice prompts in English that automates the participant's connection to a conference. The IVR Service includes all the voice messages played during the participant connection process and during the conference.</p>										

Table 2-13 Conferencing Entities (Continued)

Entity	Description
<i>Entry Queue</i>	<p>Name: ID <i>DefaultEQ</i> 1000</p> <p>A single dial Routing Lobby for all conferences (optional). A default Entry Queue called <i>DefaultEQ</i> is provided for routing calls from undefined participants to conferences. It uses Entry Queue IVR Service called <i>RMX EQ Service</i>.</p>
<i>Entry Queue 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 and route them to their destination conference.</p> <p><i>Entry Queue IVR Service</i> is the default IVR Service provided for the default Entry Queue.</p> <p>The Entry Queue is also set to Ad Hoc conferencing which allows participants to start new conferences without prior definition by entering a Conference ID that is not used by any on going conference currently running on the MCU, or Meeting Room.</p> <p>The default <i>Welcome Slide</i> displayed at the participants endpoint lists the default Meeting Rooms. The participant can select one of these Meeting Rooms or enter another ID to start a new conference.</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 9-1.

- **To modify the conference bit rate** – select a 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 **1-6**.

- **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 **9-23**.

- **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 **Optional:** Click a flag to select a language other than English.



If a language is selected the *RMX Web Client* that is not supported by the browser or the workstation's operating system, the *RMX Web Client* is displayed in English.

- 3 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.

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*

You can **Login** as a *Chairperson, Operator* or *Administrator*. Your **Login** level determines your viewing and system permissions.

The *Administrator's* view is shown below:

The screenshot displays the RMX 2000 web client interface. The interface is divided into several panes, each labeled on the left side with a blue line pointing to the corresponding area in the screenshot:

- Conferences List:** Points to the top-left pane showing a list of conferences with columns for Name, Status, ID, Start Time, and End Time.
- List:** Points to the top-right pane showing a list of participants with columns for Name, Status, Role, IP Address, Alias Name, Network, Dialing Device, Audio, Video, Encryption, and FECC Token.
- RMX Management:** Points to the middle-left pane showing a tree view of management options, including Frequently Used (Connections, Users, Signaling Monitor, Network Services, Hardware Monitor, Meeting Rooms, Address Book) and Rarely Used (IVR Services, Conference Profiles, Entry Queues, SSP Factories).
- Address Book:** Points to the rightmost pane showing a list of contacts with columns for Name and Dialing Device.
- Status Bar:** Points to the bottom-most pane showing system alerts and participant alerts, along with port usage and MCU status.

The main screen can be customized. For more information, see "*Customizing the Main Screen*" on page 3-9.

Viewing and System Permissions

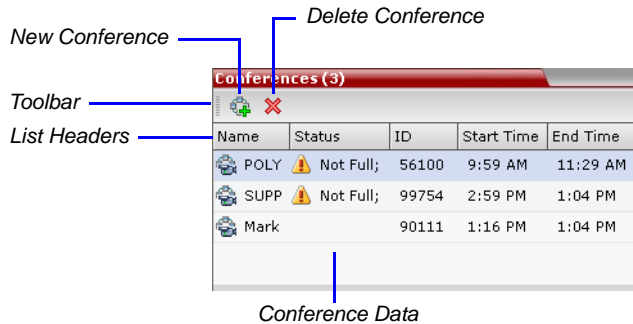
The *RMX Web Client* user's viewing and system permissions depend on the **Login** level as summarized in the table below:

Table 3-1 Viewing and System Permissions

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

Conferences List

The *Conference List* 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.

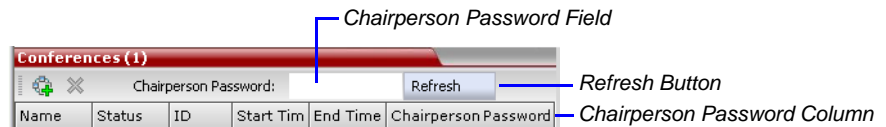


The *Conference List* toolbar contains two buttons:

- **New Conference** – to start a new ongoing conference.
- **Delete Conference** – delete the selected conference(s).

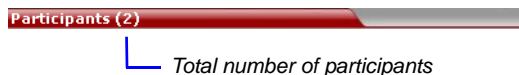
If you are logged in as a *Chairperson*:

- You can monitor a list of conferences for which you have entered the password or that don't have a *Chairperson Password* assigned.
- A *Chairperson Password* entry field and a list **Refresh** button are displayed.
- A *Chairperson Password* column is included in the conference data.



List

The *List* pane displays details of the item selected in the *Conference List* 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 parameters that need to be configured to set up and run conferences.

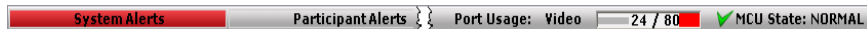
The *Configuration Pane* is divided into two sections:

- **Frequently Used** – parameters often configured monitored or modified.
- **Rarely Used** – parameters configured during initial system set-up and rarely modified afterward.

Only administrators can modify these parameters.

Status Bar

The Status Bar at the bottom of the RMX screen 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 problem exists. The flashing continues until an operator or administrator 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	CSMngr	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 st	
9/13/2006	general	major	CFG_CHANGED	McuMngr	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 11-1.

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*.

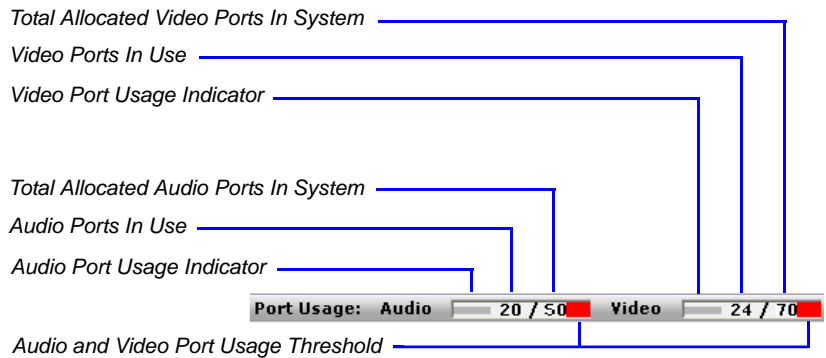
Participant Alerts (2)												
Conference	Name	Status	Disconnection Time	Role	IP Address	Alias	Network	Dialing Direction	Audio	Video	Encryptio	FECC To
Marketing	V69	discon	9/21/2006 2:18 PM		172.22.189	H.323		Dial out				
Marketing	V96	discon	9/21/2006 2:18 PM		172.22.186	H.323		Dial out				

Viewing Permissions		
Chairperson	Operator	Administrator
	✓	✓

Port Usage Gauges

The *Port Usage Gauges* indicate:

- The number of ports in the system
- The number of ports in use
- The Video/Voice Port Configuration (allocation)
- The *Capacity Usage* threshold.






The 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 can be set by the system administrator.

For more information, see the *RMX 2000 Administrator's Guide*, "Port Usage Gauges" on page 11-15.

Viewing Permissions		
Chairperson	Operator	Administrator
✓	✓	✓

MCU State

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

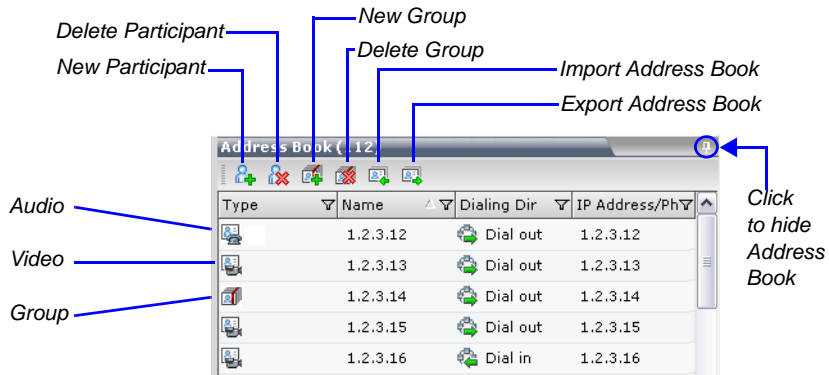
-  **MCU State: NORMAL** – The MCU is functioning normally.
-  **MCU State: MINOR** – The MCU has a minor problem but users can keep working.
-  **MCU State: MAJOR** – The MCU has a major problem. MCU behavior could be effected and attention is required.

Address Book

The *Address Book* is a list of *Participants* and *Groups* that have been defined on the RMX. The information in the *Address Book* enables RMX users to easily assign participants to conferences.

The *Address Book* toolbar contains 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 in the top right corner of the screen.

Click the tab to re-open the *Address Book*.

Click tab to open Address Book



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 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 *Main 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.

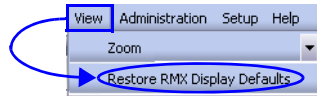
- 2 Click on the column heading to toggle the column's sort order.

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 window to its default configuration:

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



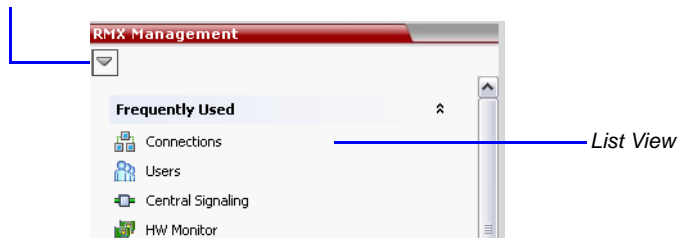
Customizing the RMX Management Pane

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*.

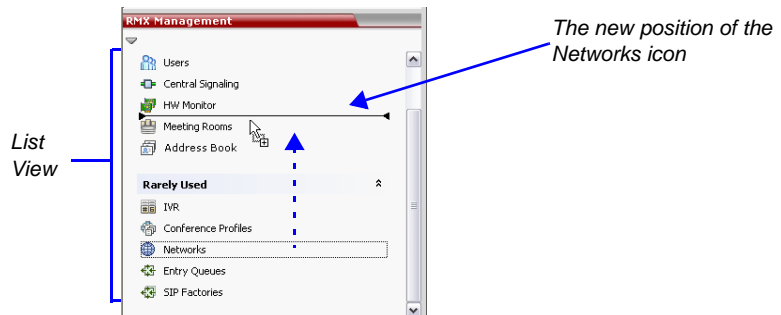
This only works in *List View* because in *Toolbar* view, all items are represented by icons.



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.
- Dialing in to a Meeting Room.
 - A Meeting Room is a conference that is saved in the MCU. It remains in passive mode until it is activated by the first participant, or the meeting organizer, dialing in.

For more information about Meeting Rooms, see the *Polycom RMX 2000 Administrator's Guide*, "Meeting Rooms" on page 2-1.

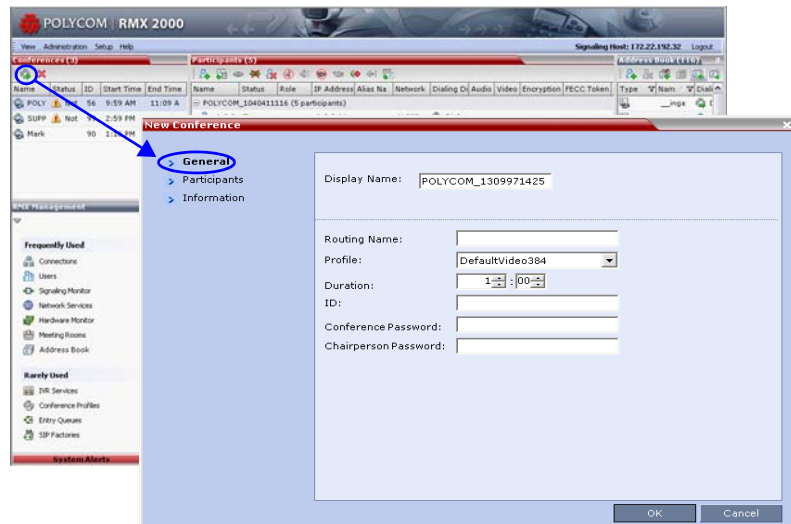
- Dialing in to an Ad Hoc Entry Queue which is used as the access point to the MCU.

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

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, you can accept the default conference *ID* and click **OK** to launch the conference. If required, you can change the 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 *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.

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

Field	Description
<p><i>Display Name (cont.)</i></p>	<p>The maximum length of text fields also varies according to the mixture of character sets (Unicode and ASCII). Maximum field length in ASCII is 80 characters. If the same name is already used by another conference, Meeting Room or Entry Queue, the RMX displays an error message requesting you to enter a different name. Note: This field is displayed in all tabs.</p>
<p><i>Routing Name</i></p>	<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 defined using ASCII characters. Comma, colon and semicolon characters cannot be used in the <i>Routing Name</i>. 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>
<p><i>Profile</i></p>	<p>The system displays the name of the default Conference Profile. Select the required Profile from the list. The Conference Profile includes the Conference line rate, media settings and general settings. For a detailed description of Conference Profiles, see the <i>RMX 2000 Administrator's Guide</i>, "Conference Profiles" on page 1-1.</p>

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

Field	Description	
<i>Duration</i>	Define the duration of the conference in hours using the format HH:MM (default 01:00).	
<i>ID</i>	Enter the unique-per-MCU conference ID. If left blank, the MCU automatically assigns a number once the conference is launched. This ID must be communicated to conference participants to enable them to dial in.	
<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.	These fields are numeric and have a default length of 4 characters. The administrator can modify them in the Setup - System Configuration settings. For more information, see <i>RMX Administrator's Guide</i> , "System Configuration" on page 11-5.
<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.	

- 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.

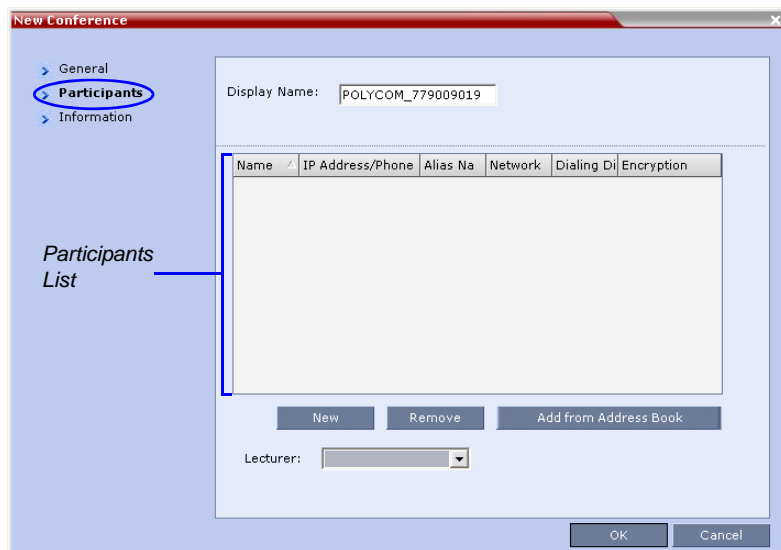
Participants Tab



This procedure is optional.

5 Click **Participants**.

The *Participants* tab opens.






6 Define the following parameters:

Table 3-3 *New Conference – Participants Options*

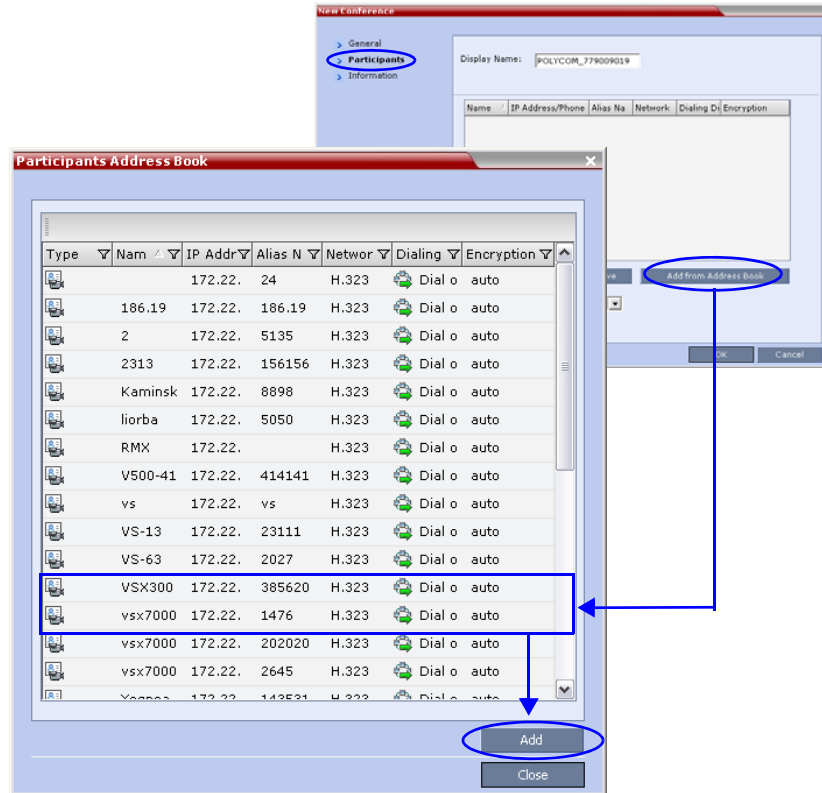
Field / Button	Description
Participants List	
<i>Name</i>	A Unicode field that displays the participant's name and an icon representing the endpoint type: <i>Audio Only</i> or <i>Video</i> .

Table 3-3 *New Conference – Participants Options (Continued)*

Field / Button	Description
<i>IP Address/Phone</i>	<p>Indicates the IP address or phone number of the participant's endpoint.</p> <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, SIP or ISDN/PSTN</i> .
<i>Connection</i>	<p>Dial-in – The participant dials in to the conference</p> <p>Dial-out – The RMX dials out to the participant</p>
<i>Encryption</i>	<p>Displays whether the endpoint uses encryption for its media.</p> <p>The default setting is <i>Auto</i>, indicating that the endpoint must connect according to the conference's encryption setting.</p> <p>Note: The H.320 protocol (ISDN/PSTN) does not support encryption.</p>
	<p>Define a new participant.</p> <p>For more information, see the <i>RMX 2000 Administrator's Guide</i>, "Adding a new participant to the Address Book" on page 4-4.</p>
	Click to remove the selected participant from the conference.
	Click to add a participant from the <i>Address Book</i> to the conference.
<i>Lecturer</i>	This option is used to activate the <i>Lecture</i> mode. Select the participant you want to designate as <i>Lecturer</i> from the drop-down menu list of conference participants.

To add participants from the Address Book:

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



- 8 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.

- 9 The selected participants are assigned to the conference and appear in the *Participant List*.
- 10 Select additional Participants or click **Close** to return to the *Participants* tab.

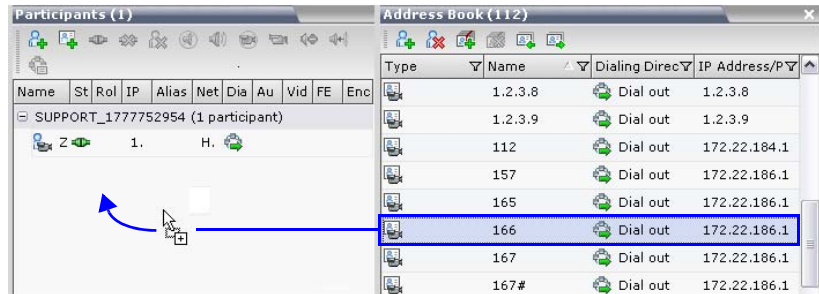
Using Drag & Drop to add Participants from the Address Book:

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:

- 11** Open the *Address Book*.
- 12** 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.



Info Tab



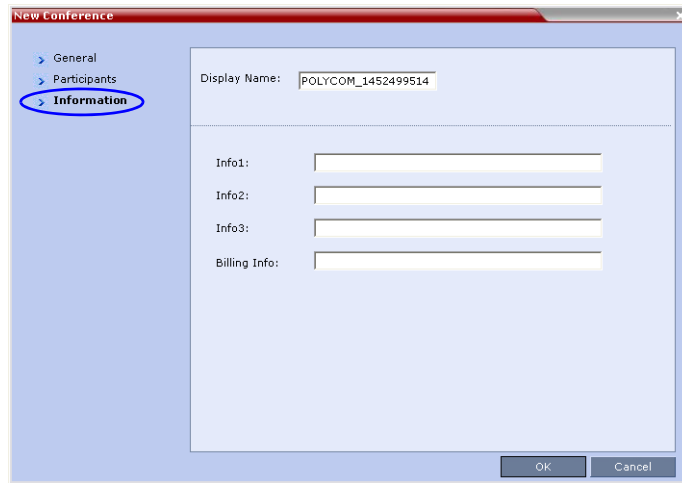
This procedure is optional.

To add information to the conference:

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

- 13** Click **Info**.

The *Information* tab opens.



- 14** 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.
<i>Billing</i>	Enter the conference billing code if applicable.

- 15** Click **OK**.

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

If no participants were defined or as long as no participants are connected the conference's Status is listed as empty and a warning icon (⚠) appears in the *Status* column.

The status changes when participants connect to the conference.

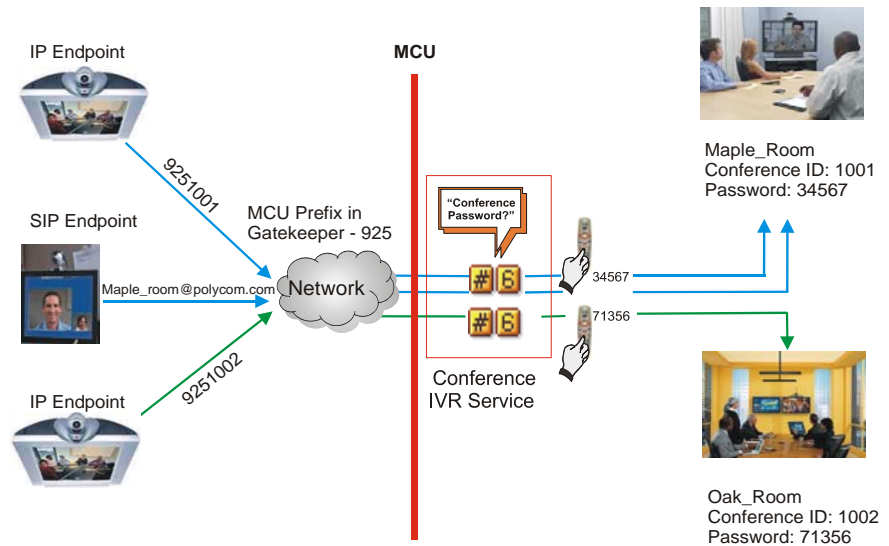
Connecting to a Conference

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 VSW conferences must have HD capable endpoints and must use 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:

Gatekeeper Prefix	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 name and domain name in the following format:

conference_name@domain_name

Example:

➔ The participant dials Maple_room@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.

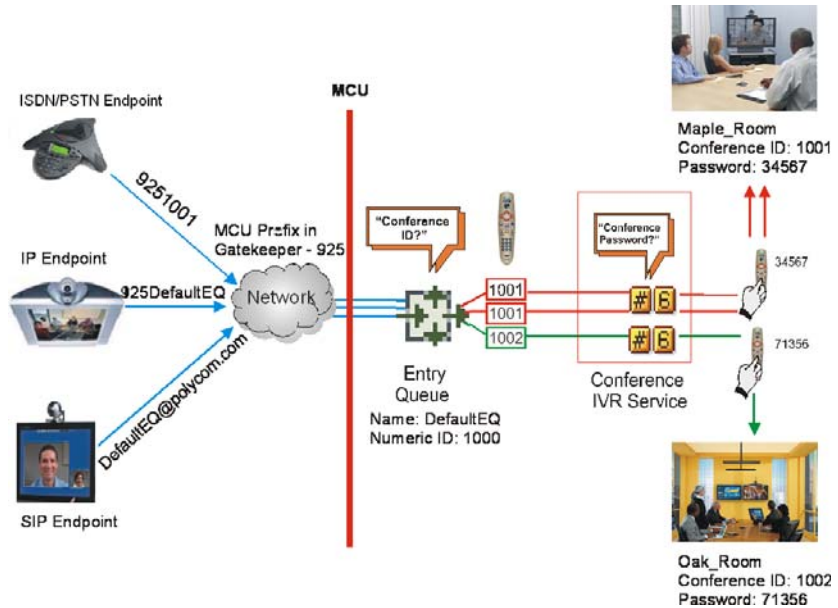


Figure 3-1: Dial-in Connection via Entry Queue

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:

Gatekeeper Prefix	925
Entry Queue ID	1000
☛ The participant dials	9251000

H.323 participants can bypass the Entry Queue 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 add 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 name>@<domain name>
```

Example:

Entry Queue Name DefaultEQ

Domain Name polycom.com

➔ SIP participants dial DefaultEQ@polycom.com

ISDN/PSTN Participants

PSTN participants are *Audio Only* participants. They 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 PSTN participants.

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

Dial-in 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.

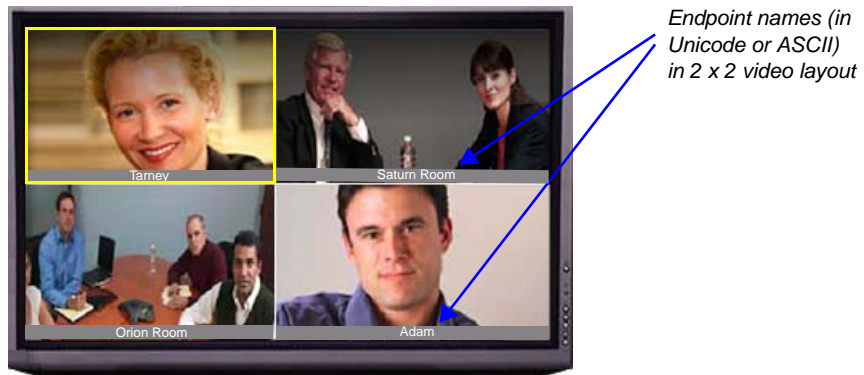
Dial-out Participants

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.

Endpoint Names in the Video Layout

During conferences you can view the endpoint name in the 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 *Polycom ViaVideo* endpoint window:



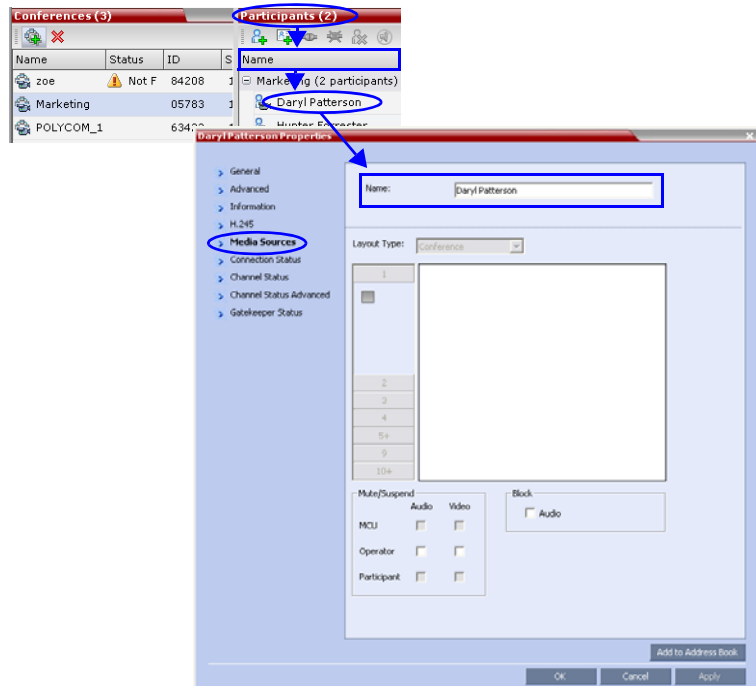
The displayed name is determined as follows:

- 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
- 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 select **Participant Properties** from the drop-down menu.

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



- 2 Enter the new Display Name in the *Name* field and click **OK**.

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*. The *Value* of the new flag must be: `SITE_NAME_TRANSPARENCY=NO`.

For more information, see the *RMX 2000 Administrator's Guide*, "System Configuration" on page 11-5.

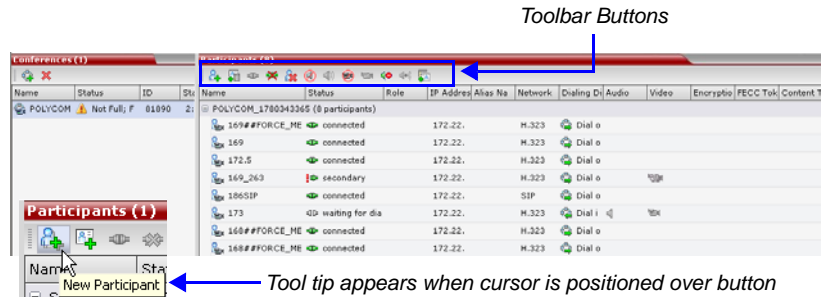
Monitoring On Going Conferences

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.

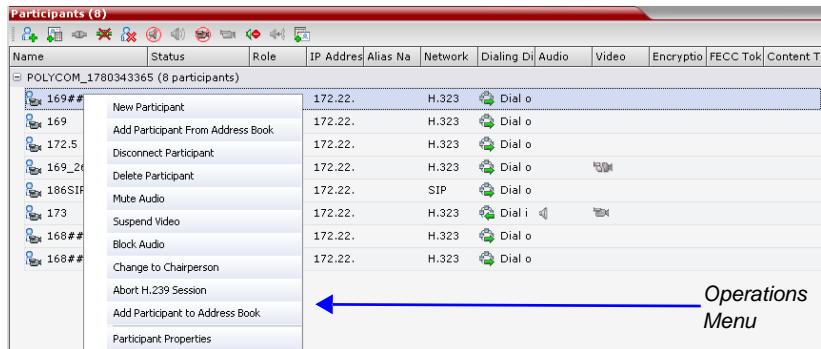
Operation Selection

All Monitoring and Operations procedures performed during on going conferences can be performed by either of two methods:

- Using the buttons in the toolbars.



- Right-clicking anywhere in the relevant pane and selecting an operation from the menu.

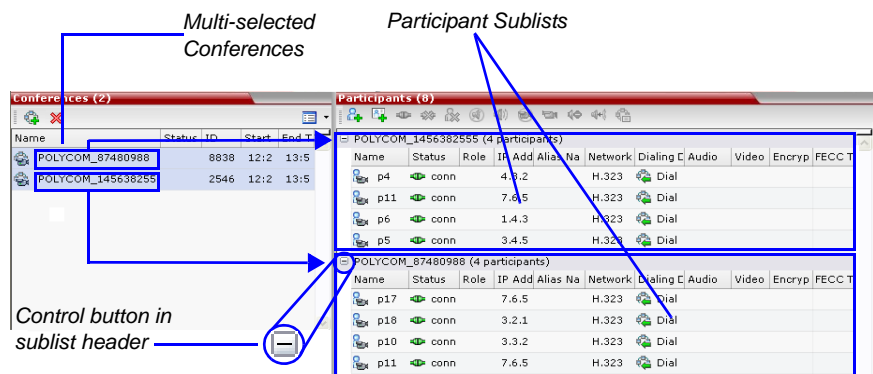


Multi Selection

Using **multi-select**, you can monitor and perform simultaneous operations on multiple participants in multiple conferences.

Multi-selected conferences are displayed as sublists in the *Participants List* pane.

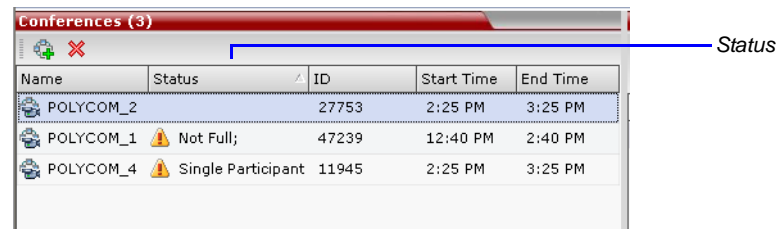
The sublists can be expanded and collapsed by clicking the **+** and **-** sublist control buttons that appear next to the conference name in the sublist headings.



Conference Level Monitoring

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



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



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

No status indicator means that the conference is running without problems.

Table 3-5 Conferences – Monitoring Information

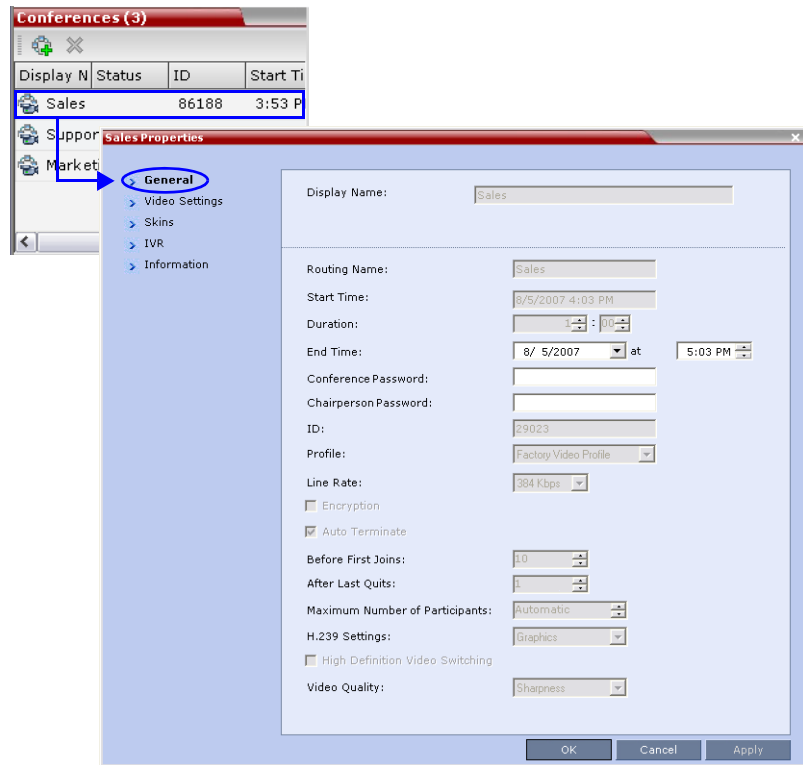
Field	Description
<i>Name</i>	<p>Displays conference name and type of conference:</p> <ul style="list-style-type: none">  – Video Conference (including HD CP conferences).  – High Definition Video Conference running in Video Switching mode.
<i>Status</i>	<p>Displays the status of the ongoing conference. If there is no problem with the participant's connection no indication is displayed. If one of the following statuses occur, the appropriate indication, 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.



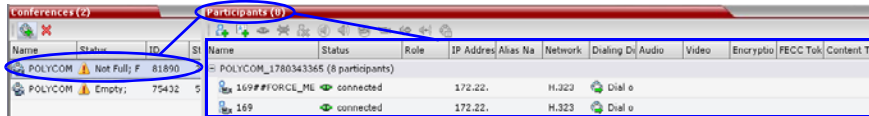
You can view all the conference’s properties but those that appear with a gray background cannot be modified.

For more information, see the *RMX 2000 Administrator’s Guide*, "Conference Level Monitoring" on page 5-3.

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








Field	Description
Name	Displays the name and type 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 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.
	 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.	

Table 3-6 Participant Monitoring – Indicators and Properties














Field	Description	
<i>Status (cont.)</i>		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.
<i>IP Address/ Phone</i>	The participant’s IP address.	
<i>Alias Name/ SIP Address</i>	The participant’s Alias Name or SIP URI.	
<i>Network</i>	The participant’s network connection type – H.323,SIP or ISDN/PSTN.	
<i>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 there is no problem with the participant’s audio connection and the channel is neither muted nor blocked, no indication is displayed.	
		Muted – Audio channel is muted.
		Blocked – Transmission of audio from the conference to the participant is blocked.

Table 3-6 Participant Monitoring – Indicators and Properties

Field	Description	
Video	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.
Encryption		Indicates that the endpoint is using encryption for its connection to the conference.
FECC Token		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.
Content Token		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 6-12 .

For more information, see the *RMX 2000 Administrator's Guide*, "Participant Level Monitoring" on page **5-8**.

Operations Performed During On Going Conferences

Conference Level operations

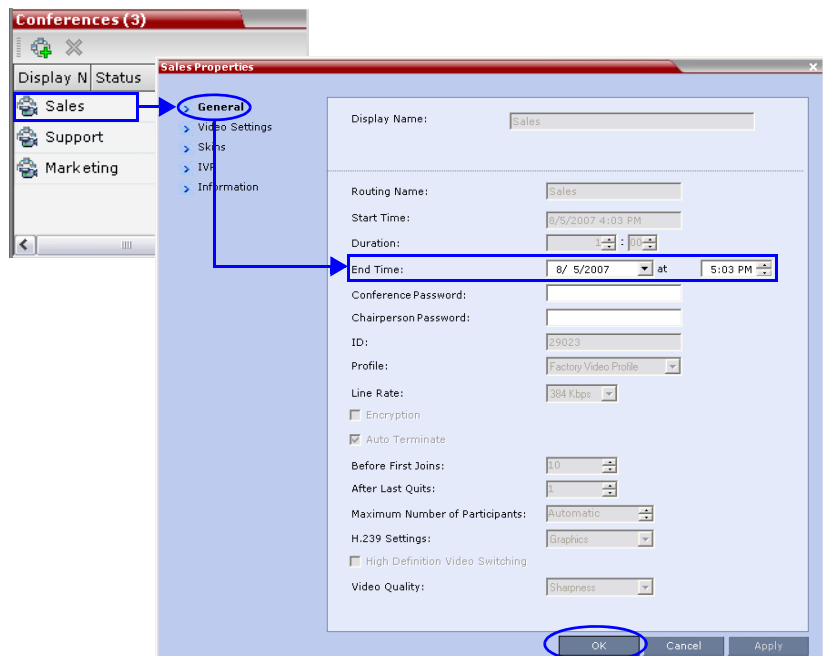
Changing the Duration of a Conference

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**.

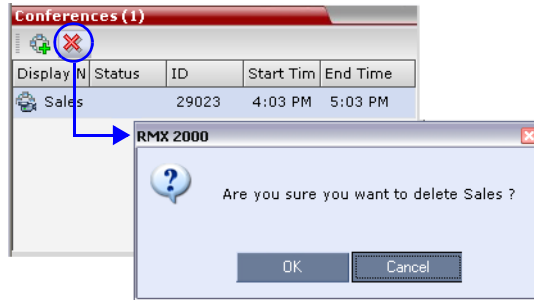


The *End Time* is changed and the *Duration* field is updated.

To terminate a conference manually:

- 1 In the conference 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.

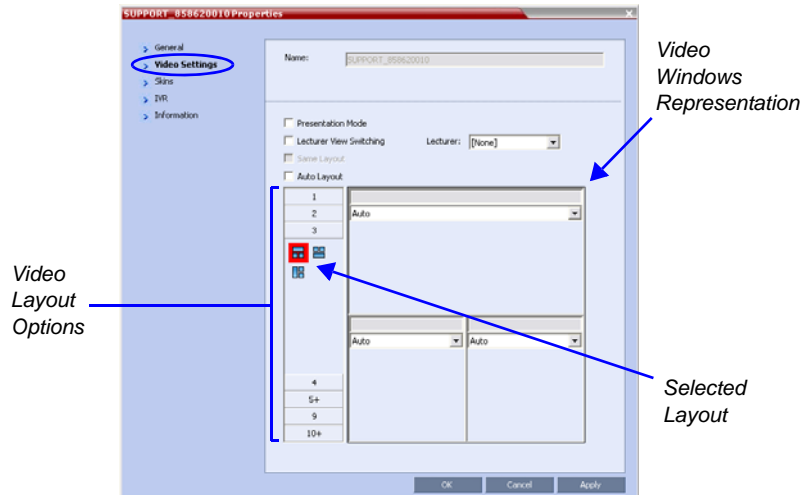
Changing the Video Layout of a Conference

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

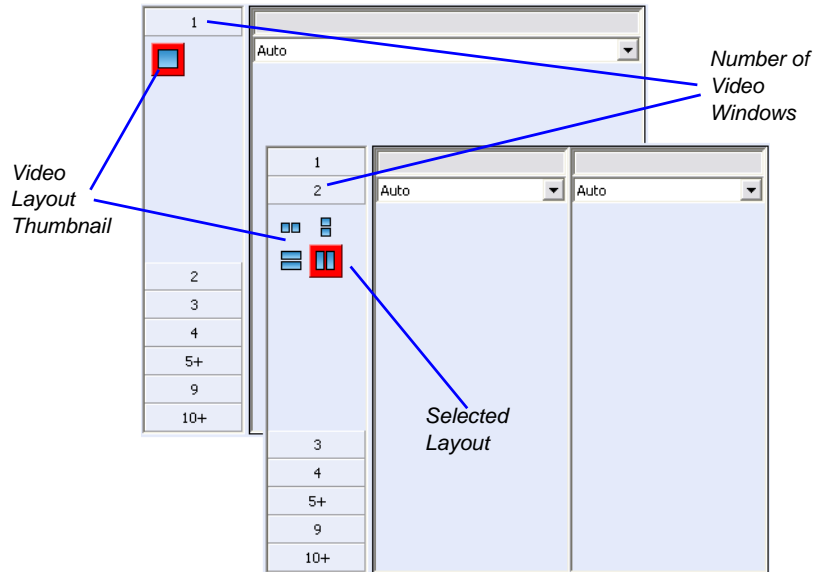
The initial video layout is selected in the conference *Profile*.

To change the video layout of a conference:

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



- 2 From the *Video Layout* options, select the *Number of Windows* to display and the *Video Layout* thumbnail required and click **OK**.



Video Forcing

The chairperson or operator can select which participant appears in each of the video layout windows for any participant by using *Video Forcing*. Conferences start with the layout defined in the *Conference Profile*.

Video Forcing applies to Dynamic Continuous Presence conferences.

Video Forcing works on 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.

Video Forcing can be cancelled by individual participants via *Personal Layout Control* without affecting other participants.

For more information see "*Personal Layout Control*" on page [3-42](#).

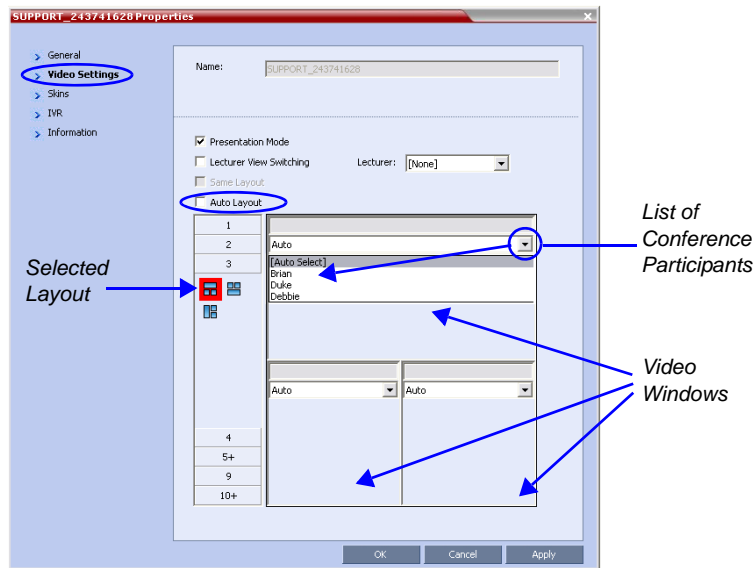
Video Forcing Guidelines:

- 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 video switching is audio-activated. The video forcing settings 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** Un-check the **Auto Layout** box.
- 3** In the window to which you want to force a participant, select the participant's name from the list of conference participants.



- 4** Repeat step 3 to force participants to other windows.

5 Click OK.

To cancel Video Forcing for a window:

➤ In the window's *Participants* list, select **Auto Select**.

After selecting **Auto Select**, switching between participants is audio activated.

Participant Level Operations

Participant Level Operations enable you to modify and control the connections and statuses of participants in on going conferences.

Table 3-7 lists the Participant Level Operations that can be performed.

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 3-16.
<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.

Table 3-7 Participant Level Operations (Continued)












Menu Option	Button	Description
<i>Unmute Audio</i>		Participant's audio transmission to the conference resumes. The <i>Mute Audio</i> button () becomes active.
<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>		Participant's video transmission to the conference resumes. 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>		Conference audio transmission to the participant resumes. 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>		Select to withdraw the Content Token from the participant.
<i>Change to Chairperson</i>		Define the selected participant as the conference leader/chairperson.

Table 3-7 Participant Level Operations (Continued)

Menu Option	Button	Description
<i>Change to Regular Participant</i>		Define the chairperson as a regular participant without chair privileges.
<i>Participant Properties</i>		Select for a detailed view of all <i>Participant Properties</i> . For more information, see the <i>RMX 2000 Administrator's Guide</i> , "Participant Level Monitoring" on page 5-8.

Video Forcing

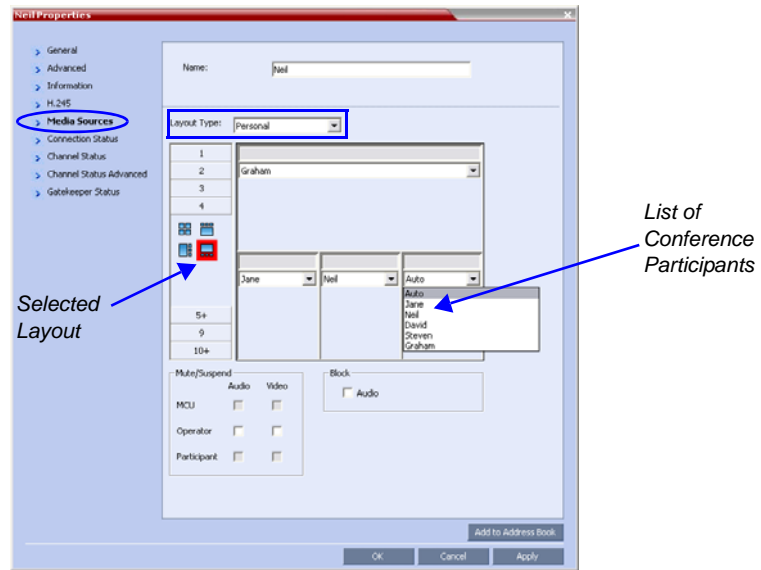
Video Forcing Guidelines:

See "Video Forcing Guidelines:" on page 3-37.

To video force a participant to a window:

- 1 In the *Participants* list, double click the participant or right-click the participant and select **Participant Properties** from the drop-down menu.

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



- 2 In the *Layout Type* menu, select **Personal**
- 3 In the window to which you want to force a participant, select the participant's name from the list of conference participants.
- 4 Repeat step 3 to force participants to other windows.
- 5 Click **OK**.

To cancel **Video Forcing** for a window:

- ➔ In the *Layout Type* menu, select **Conference**.
After selecting **Conference**, switching between participants is audio activated.

Personal Layout Control

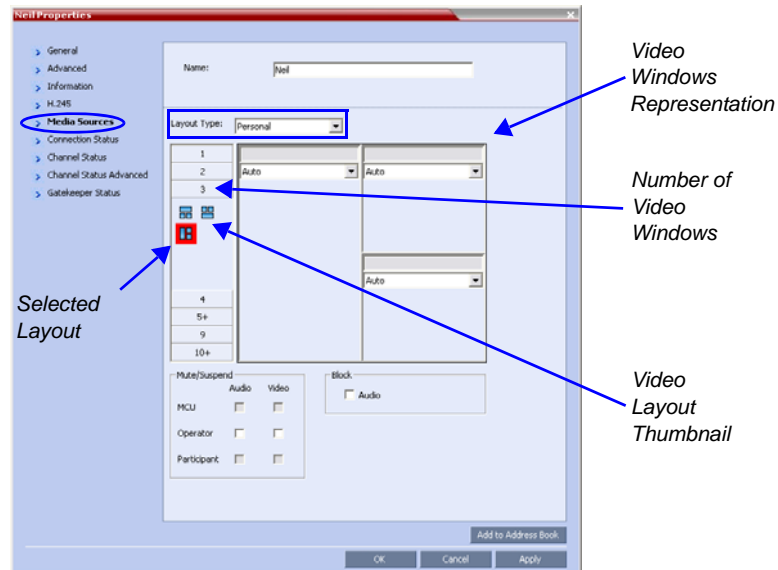
Personal Layout Control With the *RMX Web Client*

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

To change a participant **Video Layout**:

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

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



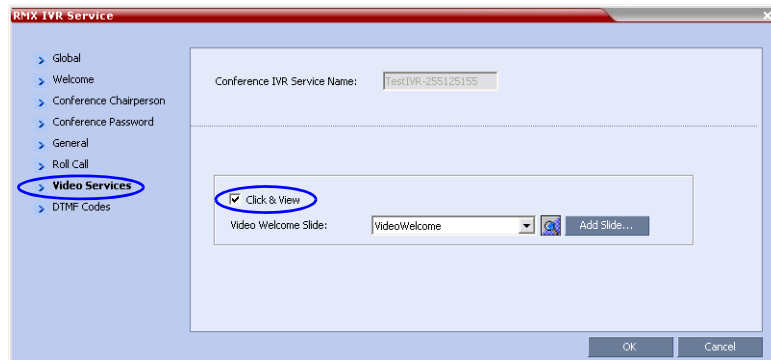
- 2 In the *Layout Type* drop-down menu, select **Personal**
- 3 Select the number of Video Windows.
- 4 Select the Video Layout
- 5 Click OK.

To cancel Video Forcing for a window:

- ➔ In the *Layout Type* menu, select **Conference**.
The participant's *Video Layout* reverts to that of the conference.

Personal Layout Control 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 *IVR Service* used in the conference.



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 a must be entered to enable the keypad. The full *Click&View* entry sequence is: , .

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



- 2 On the endpoint's 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:















Pressing **#** in any multi-square layout forces the current speaker to the top left window.

Pressing **#** in full view forces the next participant to full view.

Pressing **0** returns the video layout 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.

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.

Table 3-9 below contains a list of DTMF Codes.

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

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

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>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>Bps, Kbps</i>	Bits and kilobits per second; a unit of bandwidth, that is the amount of data that can flow during one second over a communications line (using a transmission medium). 1 Kbps=1000 Bps
<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.

Abbreviation/ Term	Explanation
<i>Codec</i>	<u>C</u> oder- <u>d</u> ecoder. 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>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>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.

Abbreviation/ Term	Explanation
G.722	ITU-T audio algorithm, 64Kbps, 7 kHz.
G.728	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.
H.221	ITU-T standard that defines how to multiplex video, audio, control, and user data into one serial bit stream.
H.230	ITU-T standard that defines simple multipoint control systems procedures and describes network maintenance functions.
H.231	ITU-T standard that defines a set of MCU functions and operational requirements.
H.242	ITU-T standard that defines initiation of communications between systems and capabilities negotiation procedures.
H.243	ITU-T standard that defines initiation of communications between systems and capabilities negotiation procedures in multipoint conferences.
H.261	ITU-T standard that defines the Px64 video coding algorithm.
H.263	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.
H.264*	A proprietary Polycom Video compression standard.
H.264	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.
H.320	ITU-T standard that defines how the H-series video conferencing recommendations work together.
H.323	ITU-T standard for audio, video and data communications across IP-based (LAN) networks, including the Internet.

Abbreviation/ Term	Explanation
<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.
<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>Line Rate</i>	The amount of bandwidth used by a communication device, measured in Kbps (kilobits per second).
<i>LDAP</i>	Lightweight Directory Access Protocol.
<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>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.
<i>RS-232</i>	A standard for serial interface connection.

Abbreviation/ Term	Explanation
<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>T1 Line</i>	A 1.5 Mb 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.

