Avaya Solution & Interoperability Test Lab

Application Notes of Avaya and Polycom H.323 Video Solution Consisting of Polycom CMA, RMX, RSS, VSX Endpoints, and HDX Endpoints with Avaya Aura™ Communication Manager – Issue 1.0

Abstract

These Application Notes describe a compliance tested solution comprised of Avaya Aura™ Communication Manager, the Polycom CMA 4000, Polycom RMX 2000, Polycom RSS2000, Polycom VSX Endpoints, and Polycom HDX Endpoints. The solution described in these Application Notes pertains only to H.323 interoperability between Avaya Aura™ Communication Manager and the aforementioned Polycom gatekeeper, videoconference endpoints, and MCU. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Interoperability at the Avaya Solution and Interoperability Test Lab.

TJM; Reviewed:
SPOC 11/9/2009

Solution & Interoperability Test Lab Application Notes
©2009 Avaya Inc. All Rights Reserved.

1 of 66
PolycomAppNotes

Table of Contents

1. Introduction

This Application Notes describes a compliance tested solution comprised of the Avaya Video Solution comprising of Avaya Aura™ Communication Manager with Voice and Video End points. Connectivity between Polycom and Avaya was via H.323 signaling. This configuration provides basic point-to-point, multipoint Video/Audio calls and conferencing using Polycom RMX 2000 through Avaya Aura™ Communication Manager and Polycom Gatekeeper.

The configuration in **Figure 1** was used to compliance test Polycom Video Solution interoperability with Avaya Video Solution where some of the Polycom video endpoints were registered to Communication Manager and the others were registered to the Polycom CMA. Vaious types of video and audio calls were tested across the H.323 trunk. The configuration in **Figure 2** was used to test Polycom Interop Test Configuration with Avaya Aura™ Communication Manager Neighbored Gatekeeper where only the Avaya video endpoints (one-X Communicator and IP Softphone) were registed to Communication Manager and all the Polycom endpoints were registered to the Polycom CMA. Various types of video and audio calls were tested across the H.323 trunk.
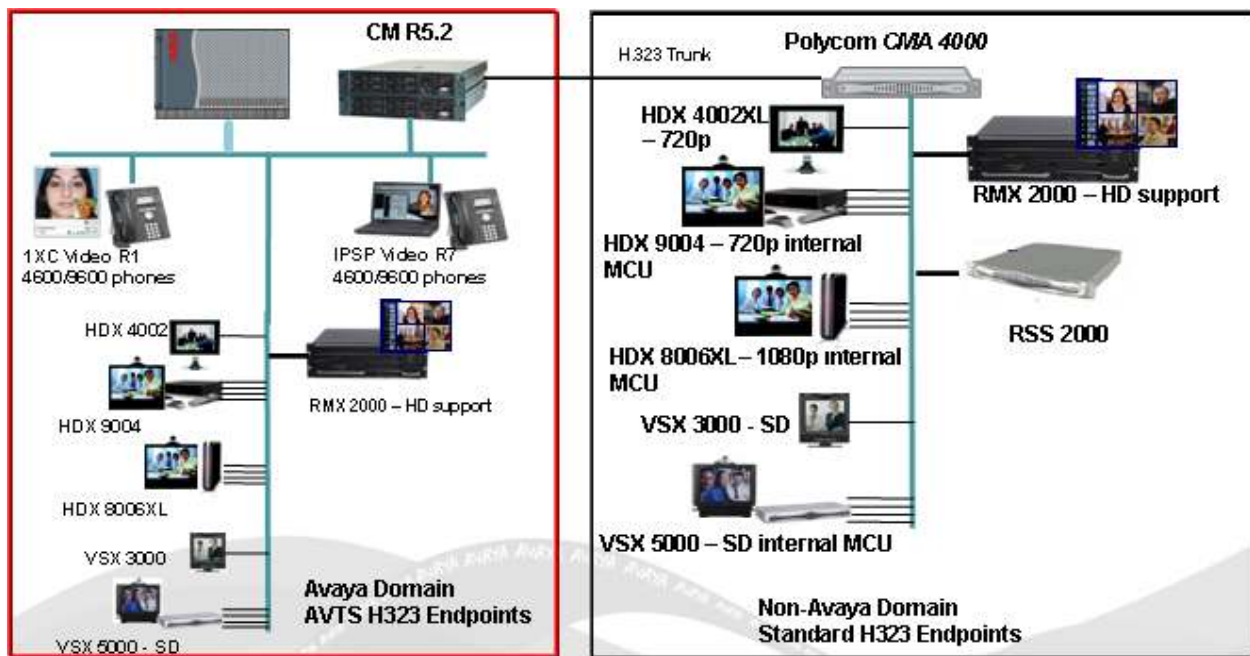
Figure 1: CM Integration & CM Neighbored Gatekeeper

TJM; Reviewed:
SPOC 11/9/2009
Solution & Interoperability Test Lab Application Notes
©2009 Avaya Inc. All Rights Reserved.
4 of 66
PolycomAppNotes
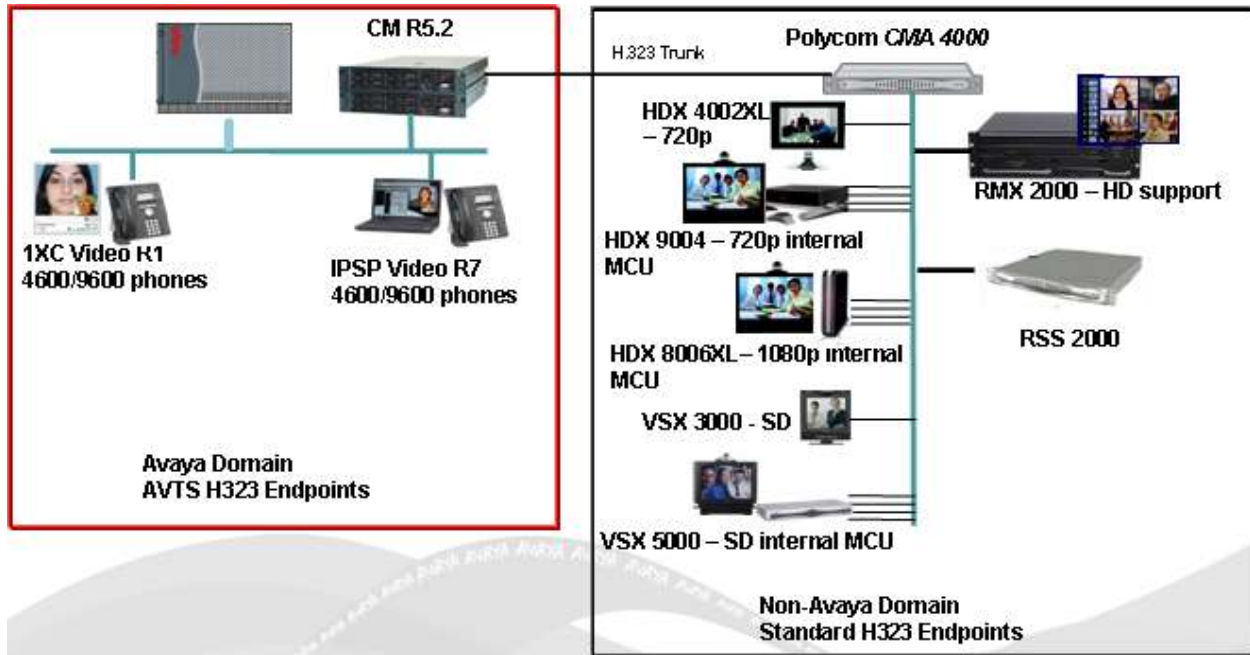
Figure 2 - CM Neighbored Gatekeeper

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya Aura™ Communication Manager	5.02.0.947.3-17436

Avaya Gateway G650
    IPSP
    CONTROL-LAN
    Medpro (TN2602AP)
	
	FW044
	FW032
	FW044

Avaya Phones(96xx,46xx and IPSP)	2.0,2.9,R6 SP6
Avaya one-X Communicator	1.0.0.91
Polycom CMA 4000	4.01.00.ER030
Polycom RMX 2000	4.0.2.7
Polycom RSS 2000	4.0.0.001
Polycom HDX 4002 (h.323)	2.5.0.5
Polycom HDX 8006 (h.323)	2.5.0.5
Polycom HDX 9004 (h.323)	2.5.0.5
Polycom VSX 3000 (h.323)	9.05.1
Polycom VSX 5000 (h.323)	9.05.1

3. Configure the Avaya Aura™ Communication Manager

This section provides the procedures for configuring Avaya Aura™ Communication Manager. The configuration page in this section are accessed using Communication Manager System

Access Terminal (SAT). Log in with the appropriate credentials. The procedures include the following areas:

- Verify Communication Manager License
- Administer IP Node Name for Communication Manager
- Administer Dial Plan
- Administer Signaling Group and Trunk Group
- Administer Routing Pattern
- Administer AAR Analysis Table
- Administer Uniform Dial Plan
- Administer Network Region
- Administer Codec Set

3.1. Verify Communication Manager License

Verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. If not then contact the Avaya sales team or business partner for a proper license file.

For Ad-hoc video conferencing there needs to be some ports available on the system-parameters customer-options form "Maximum Administered Ad-hoc Video Conferencing Ports:".

```
change system-parameters customer-options                     Page   2 of   11
                            IPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                    Maximum Administered H.323 Trunks: 8000   211
          Maximum Concurrently Registered IP Stations: 18000 21
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 12000 0
                       Maximum Video Capable Stations: 12000 12
              Maximum Video Capable IP Softphones: 12000 27
                  Maximum Administered SIP Trunks: 7000  1619
  Maximum Administered Ad-hoc Video Conferencing Ports: 8000  80
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                            Maximum TN2501 VAL Boards: 128   1
                  Maximum Media Gateway VAL Sources: 250   0
          Maximum TN2602 Boards with 80 VoIP Channels: 128   0
         Maximum TN2602 Boards with 320 VoIP Channels: 128   3
  Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

3.2. Administer IP Node Name for Communication Manager

Enter the **change node-names ip** command and add an entry for the Polycom CMA Gatekeeper as shown in the sample configuration screen below. The actual node name and IP address may vary. Submit these changes.

```
change node-names ip                                          Page   1 of   2
                              IP NODE NAMES
    Name              IP Address
video8730clan3B      135.9.88.5
DefaultGW            135.9.88.245
SQA_CMA4000          135.9.88.45
medpro               135.9.88.6
```

3.3. Administer Dial Plan

Enter the **change dialplan analysis** command. Add an entry for local **ext** (extension), **dac** (dial access code), and **fac** (feature access code) as shown in the screen shot below. Submit these changes. 4 is the fac for "Auto Alternate Routing (AAR) Access Code" (see admin on screen capture below), 5xxxx series are the local extensions of the Avaya IP Softphone, One-X Communicator, and Polycom End Points registered to the Communication Manager, and #xxx is the dac for the trunk groups.

```
change dialplan analysis                                      Page   1 of 12
                           DIAL PLAN ANALYSIS TABLE
                              Location:  all          Percent Full:    1

     Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
     String   Length Type    String   Length Type     String   Length Type
    4           1    fac
    5           5    ext
    7           7    ext
    #           4    dac
```

```
change feature-access-codes                                   Page   1 of   9
                            FEATURE ACCESS CODE (FAC)
        Abbreviated Dialing List1 Access Code:
        Abbreviated Dialing List2 Access Code:
        Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                  Announcement Access Code: *10
                  Answer Back Access Code: *11
                     Attendant Access Code:
    Auto Alternate Routing (AAR) Access Code: 4
   Auto Route Selection (ARS) - Access Code 1: 9    Access Code 2:
               Automatic Callback Activation: *12   Deactivation: *13
Call Forwarding Activation Busy/DA: *14    All: *15   Deactivation: *16
   Call Forwarding Enhanced Status:       Act:       Deactivation:
                      Call Park Access Code: *17
                    Call Pickup Access Code: *18
CAS Remote Hold/Answer Hold-Unhold Access Code: *69
                CDR Account Code Access Code:
                    Change COR Access Code:
               Change Coverage Access Code:
```

3.4. Administer Signaling Group and Trunk Group

Prior to configuring a H.323 trunk group for communication with Polycom CMA Gatekeeper, an h.323 signaling group must be configured. Enter the **add signaling-group 33** command and add an entry for Polycom CMA Gatekeeper as shown below. Submit these changes. **NOTE:** The field "Trunk Group for Channel Selection: " can't be populated with any number (33 in this case) until the trunk group on the next step is administered. Don't forget to come back to the signaling-group form and populate this field with the appropriate trunk group.

```
add signaling-group 33                                        Page   1 of   5
                             SIGNALING GROUP

 Group Number: 33               Group Type: h.323
                             Remote Office? n         Max number of NCA TSC: 0
                                    SBS? n            Max number of CA TSC: 0
     IP Video? y          Priority Video? n       Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 33
       TSC Supplementary Service Protocol: a
                        T303 Timer(sec): 10
  H.245 DTMF Signal Tone Duration(msec):
   Near-end Node Name: video8730clan3B      Far-end Node Name: SQA_CMA4000
Near-end Listen Port: 1719                Far-end Listen Port: 1719
                                          Far-end Network Region: 2
        LRQ Required? y                   Calls Share IP Signaling Connection? n
        RRQ Required? n
    Media Encryption? n                       Bypass If IP Threshold Exceeded? n
                                                  H.235 Annex H Required? n
         DTMF over IP: out-of-band        Direct IP-IP Audio Connections? y
 Link Loss Delay Timer(sec): 90                      IP Audio Hairpinning? n
       Enable Layer 3 Test? n              Interworking Message: PROGress
H.323 Station Outgoing Direct Media? n   DCP/Analog Bearer Capability: 3.1kHz
```

Enter the **add trunk-group 33** command and add an entry for the Polycom CMA Gatekeeper as shown in the sample configuration. Submit these changes.

```
add trunk-group 33                                            Page   1 of  21
                             TRUNK GROUP

Group Number: 33                   Group Type: isdn        CDR Reports: y
  Group Name: H.323 TG to CMA 4000        COR: 1      TN: 1        TAC: #033
   Direction: two-way        Outgoing Display? y       Carrier Medium: H.323
 Dial Access? y              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                            Member Assignment Method: auto
                                                    Signaling Group: 33
                                                    Number of Members: 10
```

3.5. Administer Routing Pattern

Enter the **change route-pattern 33** command and add an entry for Polycom CMA Gatekeeper as shown in the sample configuration section. Submit these changes.

```
change route-pattern 33                                        Page   1 of   3
                    Pattern Number: 4    Pattern Name: Polycom CMA4000
                            SCCAN? n      Secure SIP? n
   Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
   No          Mrk Lmt List Del  Digits                                QSIG
                            Dgts                                        Intw
 1: 33   0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

    BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                                Subaddress
 1: y y y y y n  n            rest                                         none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

3.6. Administer AAR Analysis Table

Enter the **change aar analysis 0** command and add an entry for Polycom CMA Gatekeeper as shown in the sample configuration below. Submit these changes.

```
change aar analysis 0                                          Page   1 of   2
                         AAR DIGIT ANALYSIS TABLE
                            Location:  all          Percent Full:    1

        Dialed          Total      Route    Call   Node  ANI
        String          Min  Max   Pattern  Type   Num   Reqd
   2                    7    7     999      aar          n
   3                    7    7     999      aar          n
   4                    5    5     33       aar          n
   5                    7    7     999      aar          n
   6                    7    7     999      aar          n
   7                    7    7     999      aar          n
   81                   5    5     33       lev0         n
```

3.7. Administer Uniform Dial Plan

Enter the **change uniform-dialplan 0** command and add an entry for Polycom CMA Gatekeeper as shown in the sample configuration below. Submit these changes.

```
change uniform-dialplan 0                                  Page   1 of   2
                        UNIFORM DIAL PLAN TABLE
                                                         Percent Full: 0

  Matching                   Insert              Node
  Pattern        Len Del     Digits      Net Conv Num
  81              5   0                   aar  n
```

3.8. Administer Network Region

Enter the **change ip-network-region 2** command and add entries as shown in sample configuration below. Submit these changes.

```
change ip-network-region 2                                  Page   1 of  19
                           IP NETWORK REGION
  Region: 2
Location: 1      Authoritative Domain: dr.avaya.com
   Name: video_endpoints
MEDIA PARAMETERS                  Intra-region IP-IP Direct Audio: yes
     Codec Set: 1                 Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 2048                       IP Audio Hairpinning? y
  UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS                   RTCP Reporting Enabled? y
 Call Control PHB Value: 46       RTCP MONITOR SERVER PARAMETERS
       Audio PHB Value: 46         Use Default Server Parameters? y
       Video PHB Value: 36
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
       Audio 802.1p Priority: 0
       Video 802.1p Priority: 5     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                 RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
          Keep-Alive Count: 5
```

```
change ip-network-region 2                                      Page   2 of  19
                              IP NETWORK REGION


INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
 Incoming LDN Extension:
 Conversion To Full Public Number - Delete:    Insert:
 Maximum Number of Trunks to Use for IGAR:
 Dial Plan Transparency in Survivable Mode? n

BACKUP SERVERS(IN PRIORITY ORDER)    H.323 SECURITY PROFILES
 1                                    1   challenge
 2                                    2
 3                                    3   any-auth
 4                                    4
 5
 6                                    Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? y
                       Near End TCP Port Min: 61440
                       Near End TCP Port Max: 61444
```

3.9. Administer Codec Set

Enter the **change ip-codec-set 1** command and add entries as shown in sample configuration
below in bold. Submit these changes.

```
change ip-codec-set 1                                           Page   1 of   2

                         IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: SIREN14-32K               1        20
 2: G.722-64K                 2        20
 3: G.711MU       n           2        20
 4: G.729A        n           2        20
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

```
change ip-codec-set 1                                          Page   1 of   2

                          IP Codec Set

                          Allow Direct-IP Multimedia? y
             Maximum Call Rate for Direct-IP Multimedia:  1920:Kbits
      Maximum Call Rate for Priority Direct-IP Multimedia:  1920:Kbits


                   Mode              Redundancy
      FAX          relay                 0
      Modem        off                   0
      TDD/TTY      US                    3
      Clear-channel  n                   0
```

4. Configure the Polycom CMA 4000 Gatekeeper

This section discusses the configuration of Polycom CMA Gatekeeper when it is integrated with Communication Manager.

Refer to the **Polycom® CMA™ System Getting Started Guide** that comes with the unit for configuring the Polycom CMA for the first time and run the First-time Setup Wizard.

- Neighboring Gatekeeper
- Dial Rules
- Site Configuration
- Site-Links
- Endpoints

4.1. Neighboring Gatekeepers

Open a web browser, enter http://x.x.x.x for the URL, where x.x.x.x is the IP address of the Polycom CMA Gatekeeper. Login with appropriate credentials. Click on the **Admin** tab, **Gatekeeper Settings** and then **Neighboring Gatekeepers**. Select the **Add** button to add a new entry to the table. Enter the **Name, Description, Gatekeeper IP Address, Port, and Gatekeeper Identifier** as shown in the sample configuration and click on **Save**.

4.2. Dial Rules

Click on the **Admin** tab, **Dial Plan and Sites,** and then **Dial Rules**. Select the **Add** button to add a new entry to the table. Under the **General Information** option, enter the **Name**, **Description**, and **Priority**. Check the **Enabled** box. Select **Prefix** from the **Pattern Type** drop-down field and **All** from the **Applicable Site** drop-down field.

Note: This example below is just one way of how to setup a Dialing Rule, configure as appropriate.

Under the **Routing Action** option, enter the appropriate **IP Address Pattern Data** (e.g., 5) and select **Route to a trusted neighbor** from the **Action** drop-down field.

TJM; Reviewed:
SPOC 11/9/2009
Solution & Interoperability Test Lab Application Notes
©2009 Avaya Inc. All Rights Reserved.
15 of 66
PolycomAppNotes
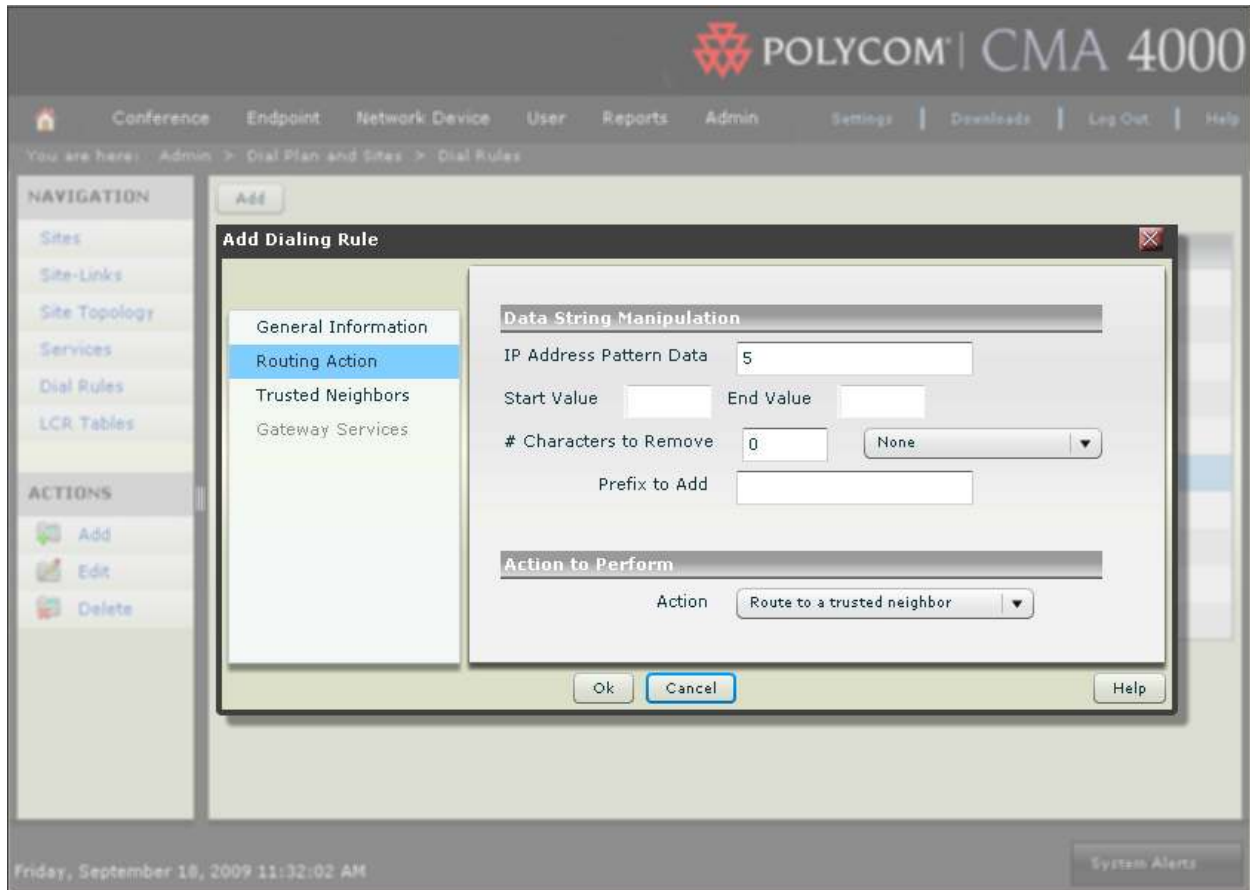
Under the **Trusted Neighbors** option, select the Gatekeeper that was administered earlier and move it over to the **Selected Neighbors** column. Click on **Ok**.

Solution & Interoperability Test Lab Application Notes
©2009 Avaya Inc. All Rights Reserved.
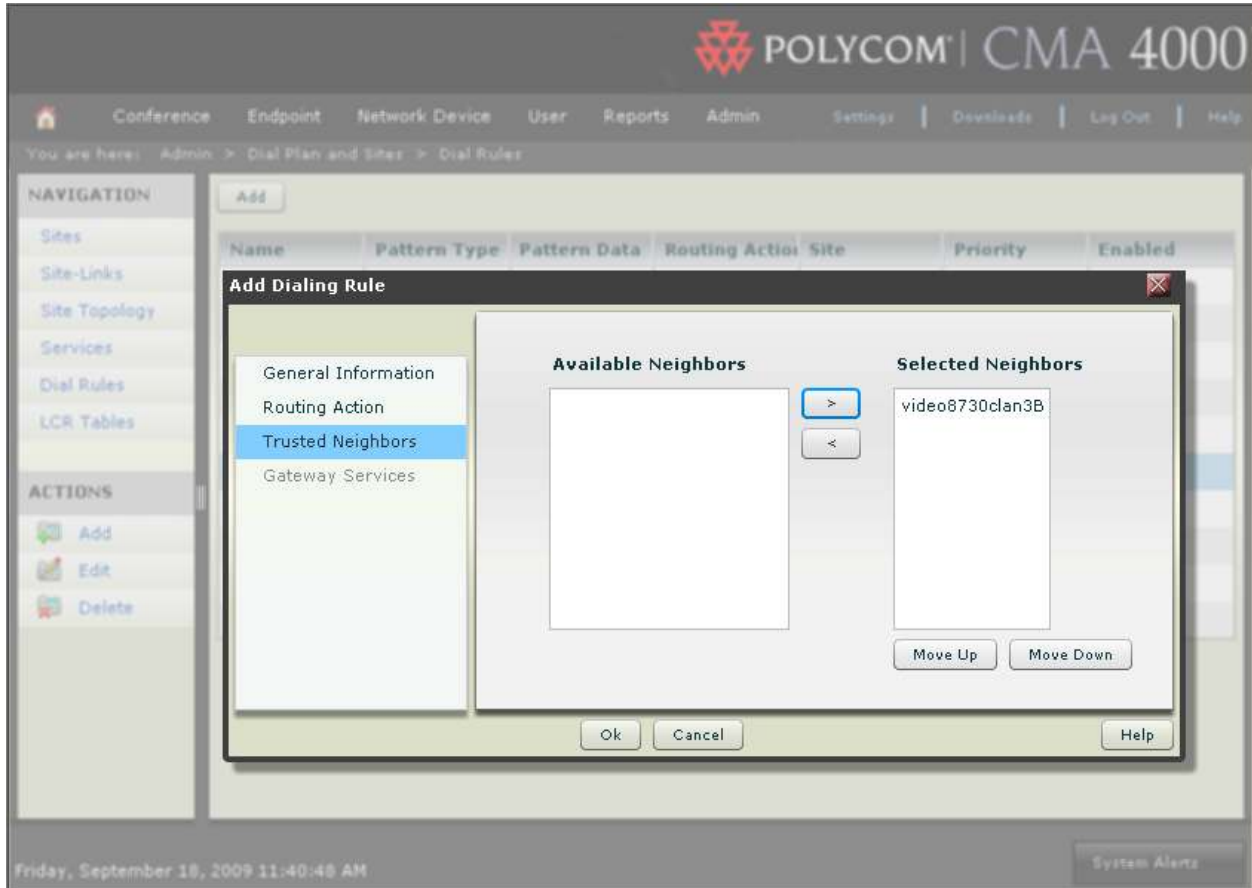
4.3. Site Configuration

Click on the **Admin** tab, **Dial Plan and Sites,** and then **Sites**. Select the **Add** button to add a new entry to the table. Under the **General Info** option enter the **Site Name** and **Description**. Select appropriate values for the **Default LCR Table**, **Assignment Method**, and **Label Color** drop-down field.

Note: This example below is just one way of how to setup a **Site**, configure as appropriate.

Under the **Routing / Bandwidth** option leave all of the default values.

Under the **Subnets** option, type in the **Subnet IP Address/Mask** and click **Add**. This information will be displayed in the Subnet table as seen below. Click on **Ok**.

TJM; Reviewed:
SPOC 11/9/2009
Solution & Interoperability Test Lab Application Notes
©2009 Avaya Inc. All Rights Reserved.
19 of 66
PolycomAppNotes
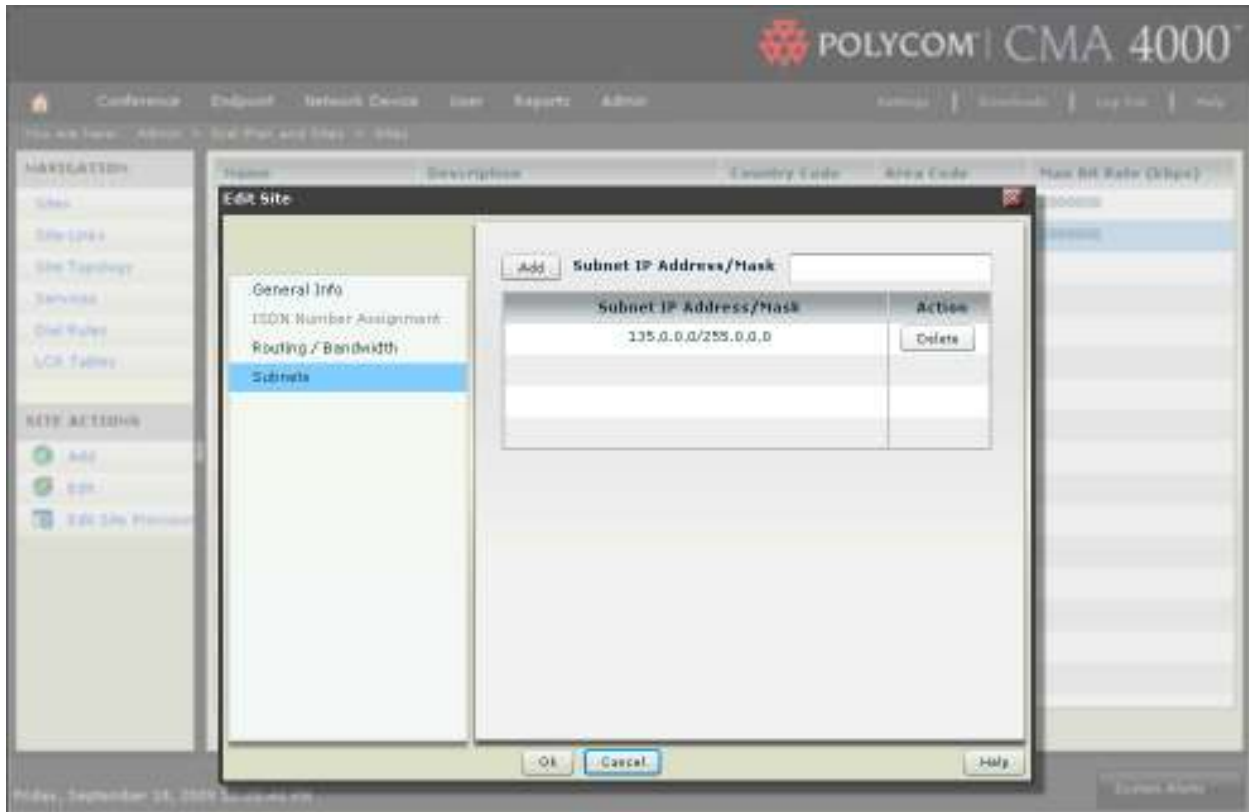
4.4. Site-Links

Click on the **Admin** tab, **Dial Plan and Sites,** and then **Site-Links**. Select **Add** to add a new entry to the table. Enter a **Name** and **Description**. For the **From Site** drop-down list, select the site that was just defined in the previous step. Select a value from the **To Site** drop-down field. Select **Direct** for the **Link Type**. Enter the **Total Bandwidth**, and **Call Max Bit Rate**. Click on **Save**.

Note: This example below is just one way of how to setup a **Site-Link**, configure as appropriate.

4.5. Endpoints

These are endpoints registered to the Polycom CMA. As such they will be recognized only by the Polycom CMA and not by the Communication Manager as valid endpoints.
.
Click on the **Endpoint** tab and then **Monitor View**. Select **Add** to add a new endpoint to the table. Enter a **Device Type** from the drop down menu, **IP Address** of the device, the **Admin ID,** and the **Password**. Click on **Find Device**.
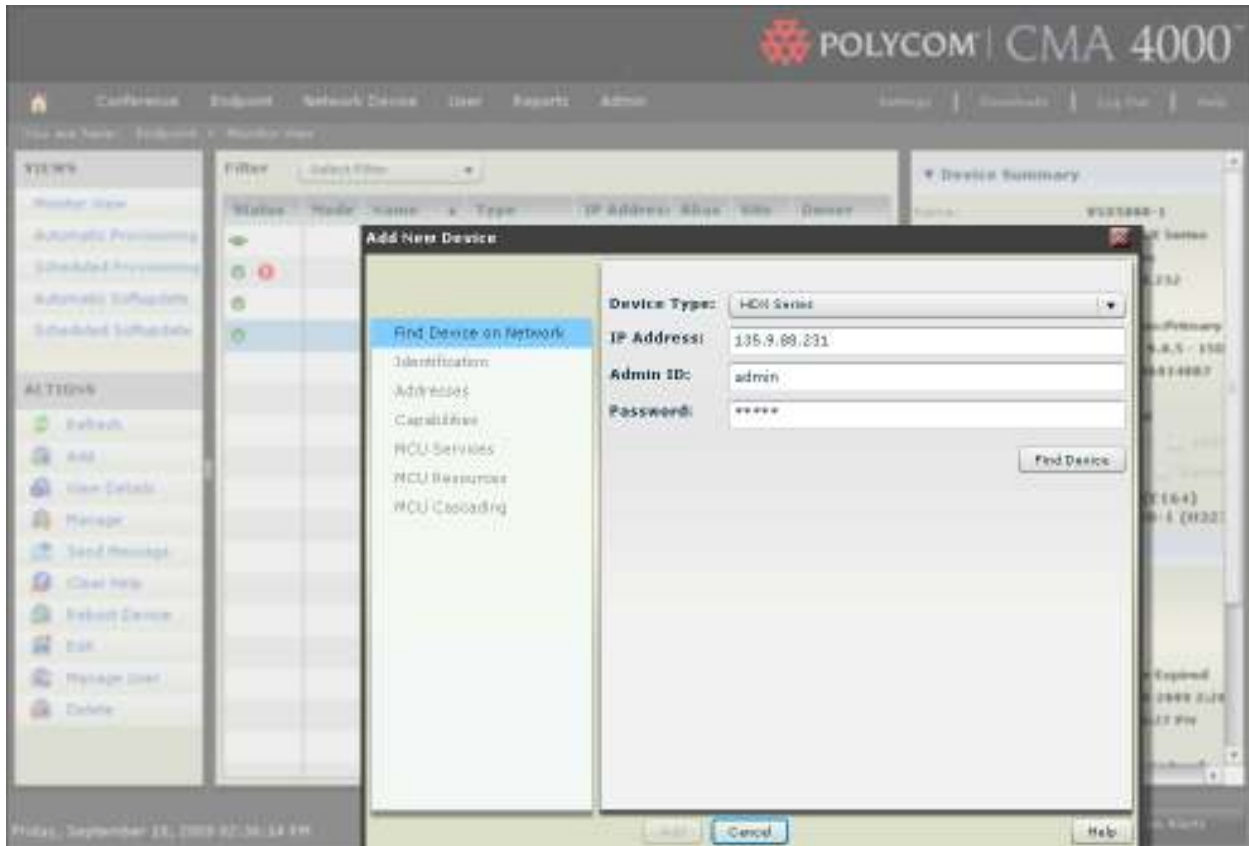
If the device was successfully found, you will see the following message displayed:
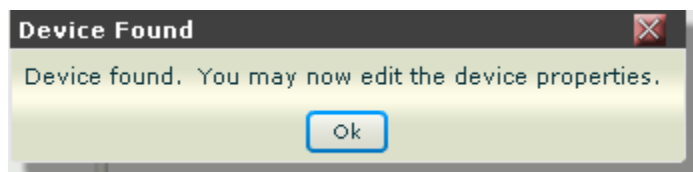
Select the **Identification** option from the Add New Device list. This is the data that was sent by the Polycom Endpoint that was found. Fill in a **Description** if necessary (see below).

Select the **Address** option from the Add New Device list. The values displayed below are all values that were automatically setup when the Device (Polycom Endpoint) was found with the exception of the E164 Type and Value. It's necessary to enter an Alias Value for the Alias Type of E164, which is the extension that the endpoint will be assigned. In the example below the extension (E164) of this endpoint is 50081. Add additional details to this form as necessary.

Select the **Capabilities** option from the Add New Device list. The values displayed below are all default values that were automatically setup when the Device (Polycom Endpoint) was found. Select the additional items to this form as necessary. Click on **Add**. Repeat this **Section 4.5 Endpoints** to add additional Polycom Endpoints.

5. Configure and Register the Polycom RMX 2000 MCU

This section describes the configuration of Polycom RMX™ MCU and its registration to the Communication Manager.

For specific details refer to the **Polycom RMX 2000 Installation & Configuration Guide** that comes with the unit. Be certain to have retrieved the Product Registration and Activation Key from the Polycom Resource Center website http://portal.polycom.com before continuing.

5.1. Initial RMX Configuration

When the RMX is installed for the first time, you must change the default IP addresses to your local network settings. This can be done by modifying the default settings in the USB key shipped with the unit using the LAN Configuration Utility and uploading them to the RMX.

1. Insert the USB key into a PC.
2. Double-click **LanConfigUtility.exe** to start the utility.
3. Modify the required network parameters in the utility's dialog box using the information supplied by your network administrator.
4. Click **OK**.
5. Insert the USB key with the modified IP addresses in the USB port on the back panel of the RMX.
6. Power the RMX **On**. System power-up sequence may take up to five minutes.
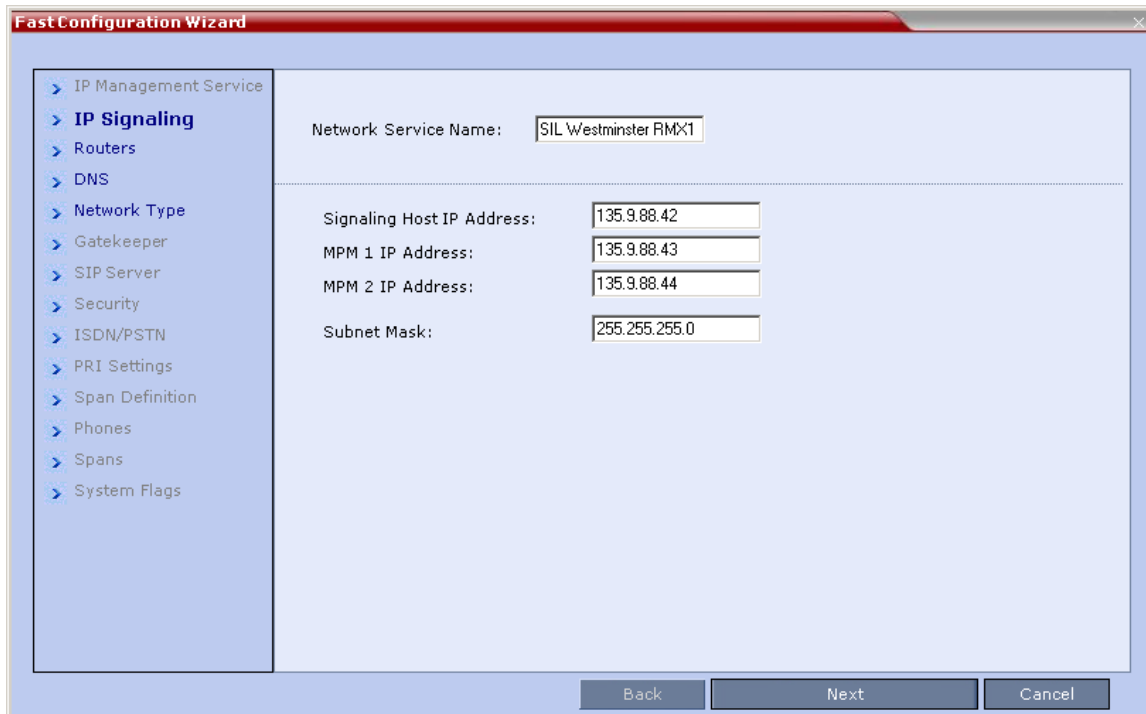7. Once the RMX is powered up and loaded remove the USB key.

5.2. Registering RMX to Communication Manager Gatekeeper

Start the RMX Web Client application by entering in the browser's address line the IP Address of the control unit as defined in the USB key in the format: http://<Control Unit IP Address> and press **Enter**. In the RMX Web Client Welcome Screen, enter the default Username (**POLYCOM**) and Password (**POLYCOM**) and click **Login**.

The Product Activation dialog box appears with the serial number filled in. In the Activation Key field, enter or paste the Product Activation Key retrieved earlier and click **OK**. If you don't have an Activation Key, click the **Polycom Resource Center** button to access the Service & Support page of the Polycom website.

The system prompts with a reset dialog box, click **No.** As no Default IP Network Service is defined, the system automatically starts the Fast Configuration Wizard. You should have collected all of the Network information prior to running this Wizard.

Enter the appropriate information on the following FastConfiguration Wizard screens as shown in this sample configuration:
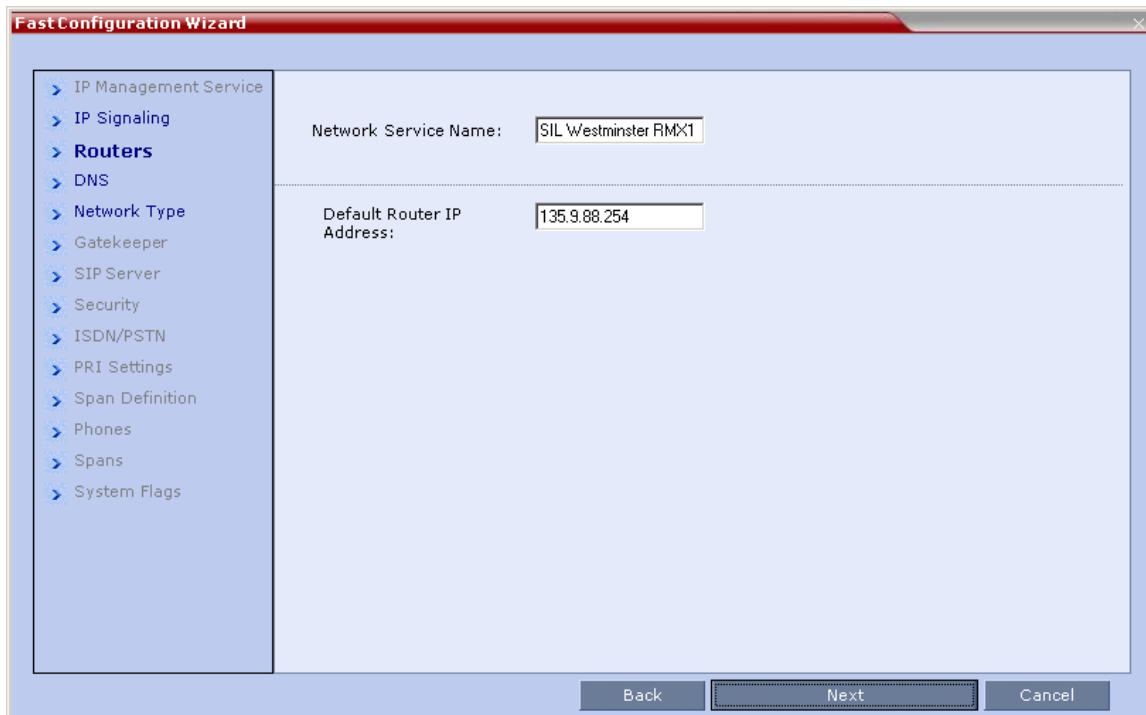

Click **Next**.

Click **Next**.

Click **Next**.

Click **Next**.

In the **IP Address or Name** field enter the address of the Communication Manager Gatekeeper.

Click **Save & Continue**.

Click **Save & Close.** The RMX confirms successful configuration.

TJM; Reviewed:
SPOC 11/9/2009

Solution & Interoperability Test Lab Application Notes
©2009 Avaya Inc. All Rights Reserved.

28 of 66
PolycomAppNotes

Next you will be prompted to reset the MCU, Select **Yes**.
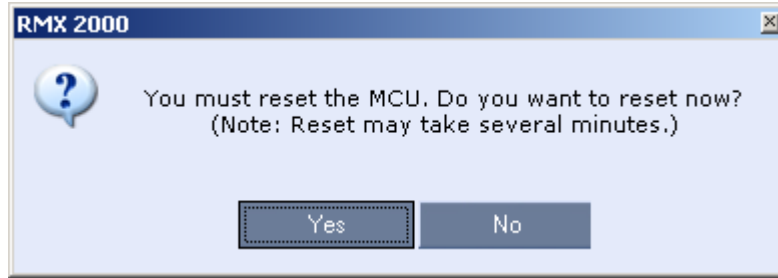
5.3. Configure System Configuration on Polycom RMX

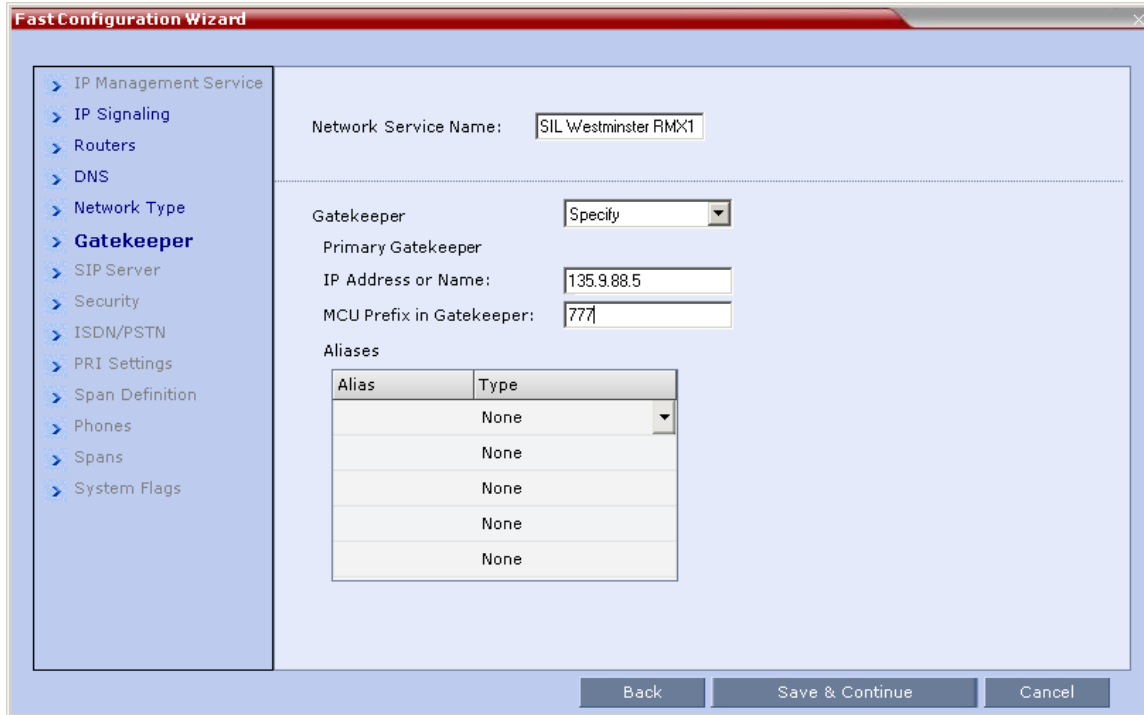Login to the RMX web interface with the appropriate credentials. From the menu bar located at the top, select **Setup** and, then select **System Configuration**. This will open the System Flags window displayed in the screen capture below.

Under the MCMS_PARAMETERS_USER, configure the following settings:
- MCU_DISPLAY_NAME = POLYCOM RMX 2000
- ENABLE_AUTO_EXTENSION = YES
- NUMERIC_CONF_ID_LEN = 5
- CP_REGARD_TO_INCOMING_SETUP_RATE = NO
- NUMERIC_CONF_ID_MAX_LEN = 8
- NUMERIC_CONF_ID_MIN_LEN = 4
- TERMINATE_CONF_AFTER_CHAIR_DROP = NO
- H323_FREE_VIDEO_RESOURCES = NO
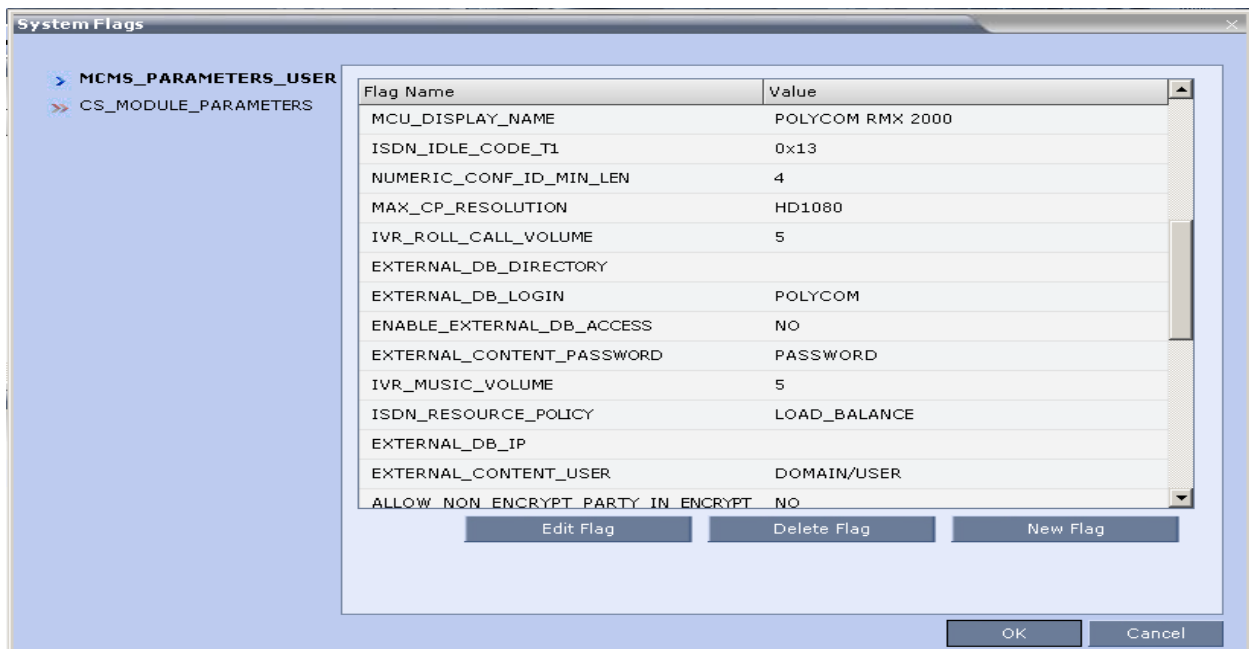
Under the CS_MODULE_PARAMETERS, add a new flag by selecting **New Flag.** Once the
New Flag window opens enter the following: New Flag: **H245_TUNNELING** Value: **YES**
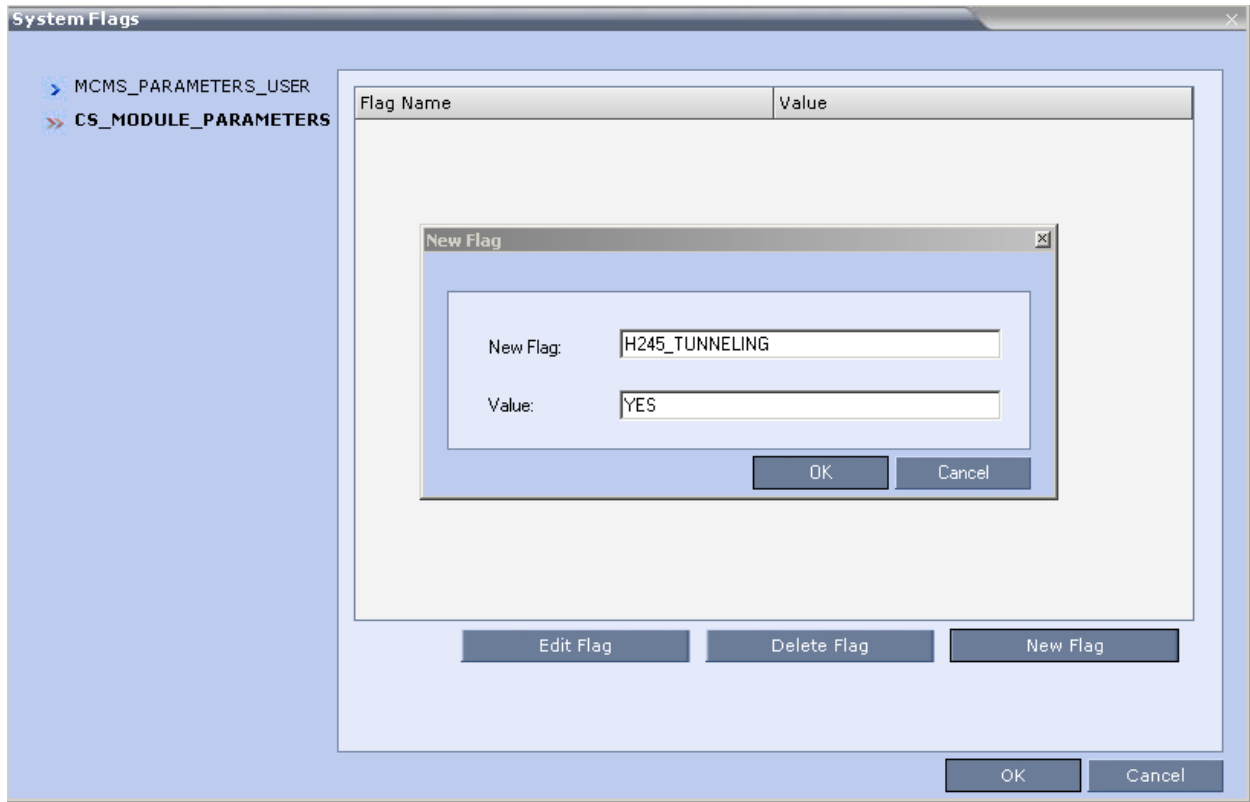
Then select **OK** from the New Flag window and then **OK** again in the System Flags window.
You will then be prompted to reset the MCU. Select **Yes**. After the RMX has reset you can log
back in to confirm the Flag values have been administered.
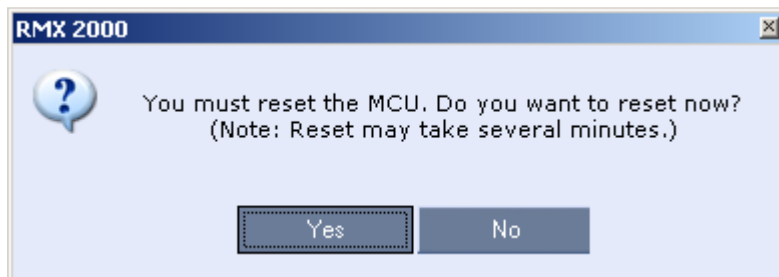
5.4. Create Conference Profile on Polycom RMX

Login to the Polycom RMX web interface. Under the **RMX Management** section of the interface locate **Conference Profiles** beneath the **Rarely Used** section and click on it. Click on the green plus to add a new entry to the table.

The following is a sample showing how a Conference Profile can be setup. The **General** item in the New Profile list below is the main screen that needs to be configured. The default values can be used for the **Video Settings**, **Skins**, **IVR**, and **Recording**. Customize as necessary.

The Conference Profiles are used on the entry queues.
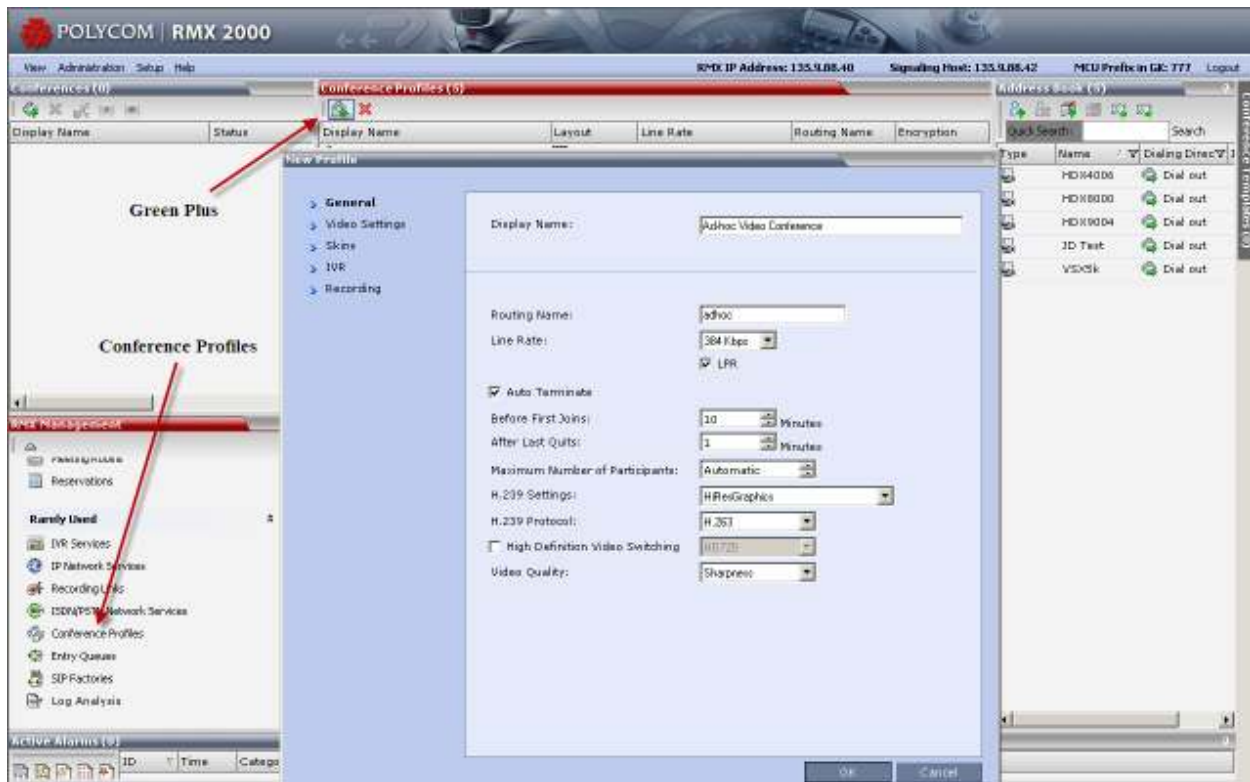
5.5. Create Entry Queues on Polycom RMX

Locate and click on **Entry Queues** under the **RMX Management**, **Rarely Used** section. Click on the green plus to add a new entry.

This is an example of a standard entry queue. Select the **Profile** that was configured in the previous step. Note the **ID: 1000** is the ID used by Communication Manager as the **Standard Factory Number** administered above. Since this is an Ad Hoc entry queue be certain to check the **Ad Hoc** box. Click **OK**.

Click on the green plus again to add a second entry for a priority entry queue. This is an example of a priority entry queue. Select the appropriate **Profile**. In this example a High Def profile was selected. Note the **ID: 1001** is the ID used by Communication Manager as the **Priority Factory Number** administered above. Since this is an Ad Hoc entry queue be certain to check the **Ad Hoc** box. Click **OK**.

5.6. Create Meeting Rooms on Polycom RMX

Add a Meeting Room by selecting **Meeting Rooms** under **RMX Management → Frequently Used**. Then select the green plus to create a **New Meeting Room**. (See screen capture below).
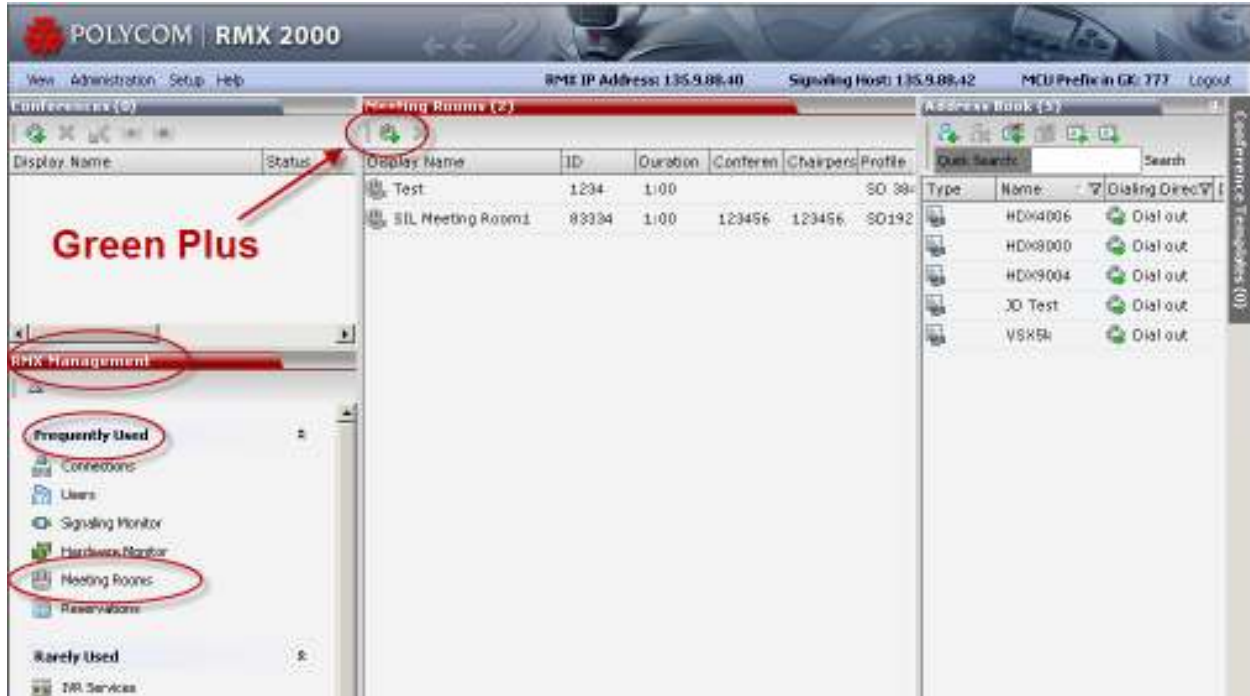
The following is a sample showing how a Meeting Room can be setup. The **General** item in the New Meeting Room list below is the main screen that needs to be configured. The default values can be used for the **Participants** and **Information** options. Customize as necessary. The **ID: 83334** is the number that the endpoints registered with the Communication Manager Gatekeeper and Polycom CMA Gatekeeper will dial to access this RMX bridge.

6. Configuring Communication Manager for Polycom RMX 2000

6.1. Administer IP Node Name for Communication Manager

Enter the **change node-names ip** command and add an entry for the Polycom RMX MCU as shown in the sample configuration screen below. The actual node name and IP address may vary. Submit these changes.

```
change node-names ip                                          Page   1 of   2
                                IP NODE NAMES
    Name              IP Address
video8730clan3B     135.9.88.5
DefaultGW           135.9.88.245
SQA_RMX1            135.9.88.42
SQACMA_4000         135.9.88.45
medpro              135.9.88.6
```

6.2. Administer Signaling Group and Trunk Group

Prior to configuring an h.323 trunk group for communication with Polycom RMX MCU, an h.323 signaling group must be configured. Enter the **add signaling-group 32** command and add an entry for Polycom RMX as shown below. Submit these changes. **NOTE:** The field "Trunk Group for Channel Selection: " can't be populated with any number (32 in this case) until the trunk group on the next step is administered. Don't forget to come back to the signaling-group form and populate this field with the appropriate trunk group.

```
add signaling-group 32                                        Page   1 of   5
                            SIGNALING GROUP


 Group Number: 32                 Group Type: h.323
                              Remote Office? n          Max number of NCA TSC: 0
                                      SBS? n            Max number of CA TSC: 0
    IP Video? y           Priority Video? y        Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 32
       TSC Supplementary Service Protocol: a          Network Call Transfer? n
                        T303 Timer(sec): 10
   H.245 DTMF Signal Tone Duration(msec):
    Near-end Node Name: video8730clan3B        Far-end Node Name: SQA_RMX1
Near-end Listen Port: 1720                     Far-end Listen Port: 1720
                                            Far-end Network Region: 2
       ARQ Required? y                 Calls Share IP Signaling Connection? n
       RRQ Required? y
   Media Encryption? n                        Bypass If IP Threshold Exceeded? n
                                                    H.235 Annex H Required? n
         DTMF over IP: out-of-band         Direct IP-IP Audio Connections? y
 Link Loss Delay Timer(sec): 90                     IP Audio Hairpinning? n
       Enable Layer 3 Test? n                   Interworking Message: PROGress
H.323 Station Outgoing Direct Media? n   DCP/Analog Bearer Capability: 3.1kHz
```

Enter the **add trunk-group 32** command and add an entry for the Polycom RMX MCU as shown in the sample configuration. Submit these changes.

```
add trunk-group 32                                        Page   1 of  21
                              TRUNK GROUP

Group Number: 32                    Group Type: isdn        CDR Reports: y
  Group Name: SQA RMX1 TG                  COR: 1      TN: 1      TAC: #032
   Direction: two-way       Outgoing Display? y      Carrier Medium: H.323
 Dial Access? y             Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                      Auth Code? n
                                           Member Assignment Method: auto
                                                     Signaling Group: 32
                                                   Number of Members: 31
```

6.3. Administer Routing Pattern

Enter the **change route-pattern 32** command and add an entry for Polycom RMX MCU as shown in the sample configuration section. Submit these changes.

```
change route-pattern 32                                    Page   1 of   3
                    Pattern Number: 4   Pattern Name: Polycom RMX2000
                          SCCAN? n     Secure SIP? n
   Grp FRL NPA Pfx Hop Toll No.  Inserted                     DCS/ IXC
   No          Mrk Lmt List Del  Digits                       QSIG
                          Dgts                                 Intw
 1: 32   0                                                      n   user
 2:                                                             n   user
 3:                                                             n   user
 4:                                                             n   user
 5:                                                             n   user
 6:                                                             n   user

    BCC VALUE  TSC CA-TSC   ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                               Dgts Format
                                                   Subaddress
 1: y y y y y n  n          rest                                     none
 2: y y y y y n  n          rest                                     none
 3: y y y y y n  n          rest                                     none
 4: y y y y y n  n          rest                                     none
 5: y y y y y n  n          rest                                     none
 6: y y y y y n  n          rest                                     none
```

6.4. Administer AAR Analysis Table

Enter the **change aar analysis 0** command and add an entry for Polycom RMX MCU as shown in the sample configuration below. Submit these changes.

```
change aar analysis 0                                        Page   1 of   2
                         AAR DIGIT ANALYSIS TABLE
                            Location:  all        Percent Full:    1

         Dialed           Total      Route    Call   Node  ANI
         String         Min  Max   Pattern   Type    Num   Reqd
     2                   7    7      999      aar            n
     3                   7    7      999      aar            n
     4                   5    5      33       aar            n
     5                   7    7      999      aar            n
     6                   7    7      999      aar            n
     7                   7    7      999      aar            n
     81                  5    5      33       lev0           n
     83334               5    5      32       lev0           n
```

6.5. Administer Uniform Dial Plan

Enter the **change uniform-dialplan 0** command and add an entry for Polycom RMX MCU as shown in the sample configuration below. Submit these changes.

```
change uniform-dialplan 0                                    Page   1 of   2
                      UNIFORM DIAL PLAN TABLE
                                                    Percent Full: 0

 Matching                    Insert            Node
 Pattern       Len Del       Digits     Net Conv Num
 81             5   0                    aar  n
 83334          5   0                    aar  n
```

6.6. Verify Ad-hoc Video Conferencing Ports

From the Communication Manager SAT interface use the **display system-parameters customer-options** command to verify the **Maximum Administered Ad-hoc Video Conferencing Ports** (page 2 of form). The maximum number of Ad-hoc video conferencing ports allowed is the sum of the ports on your RMX systems. For example, if you have an RMX2000 system the maximum number of ports is 2,000.

```
display system-parameters customer-options                      Page   2 of  11
                          OPTIONAL FEATURES


IP PORT CAPACITIES                                                    USED
                    Maximum Administered H.323 Trunks: 8000  211
         Maximum Concurrently Registered IP Stations: 18000 18
            Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
             Maximum Concurrently Registered IP eCons: 128   0
   Max Concur Registered Unauthenticated H.323 Stations: 12000 0
                      Maximum Video Capable Stations: 12000 12
                Maximum Video Capable IP Softphones: 12000 27
                    Maximum Administered SIP Trunks: 5000  1619
   Maximum Administered Ad-hoc Video Conferencing Ports: 8000  80
    Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                          Maximum TN2501 VAL Boards: 128   1
                 Maximum Media Gateway VAL Sources: 250   0
         Maximum TN2602 Boards with 80 VoIP Channels: 128   0
        Maximum TN2602 Boards with 320 VoIP Channels: 128   3
  Maximum Number of Expanded Meet-me Conference Ports: 300   0


         (NOTE: You must logoff & login to effect the permission changes.)
```

6.7. Administer Class of Service

Use the **change cos** command to set **Ad-hoc Video Conferencing** (page 2 of the form) for the appropriate Class Of Service levels. It's necessary for each of the Avaya and Polycom video endpoint to be in a COS that has Ad-hoc Video Conferencing enabled in order to conduct an ad-hoc conference call. To identify which COS a video endpoint is in run the command **display station xxxxx** (see screen capture below).

```
change cos                                              Page   2 of   2
                        CLASS OF SERVICE


                        0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 VIP Caller             n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n

 Masking CPN/Name Override  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Call Forwarding Enhanced   y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Priority Ip Video          n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Ad-hoc Video Conferencing  y  y  y  n  n  n  n  n  n  n  n  n  n  n  n  n
```

```
display station 50081                                   Page   1 of   4
                          STATION
Extension: 50081                  Lock Messages? n           BCC: M
    Type: H.323                   Security Code: 123456        TN: 1
    Port: S01767                  Coverage Path 1:            COR: 1
    Name: HDX8000-1               Coverage Path 2:            COS: 2
                                  Hunt-to Station: 50082      Tests? y
STATION OPTIONS
                                   Time of Day Lock Table:
           Loss Group: 19     Message Waiting Indicator: none
```

```
                                              Authentication Required? y
         Survivable COR: internal
  Survivable Trunk Dest? y
           DTMF over IP: in-band                 IP Video? y
```

6.8. Administer Video Bridge for Ad-hoc Conferencing

Use the **add video-bridge** command to configure an Ad-hoc video bridge for the RMX system.

Set the following parameters:
- **Name** to the name for this video bridge
- **Max Ports** to the maximum number of Ad-hoc conferencing ports you want to assign to this bridge. (The minimum you can enter is 3.) This is equivalent to the number of ports for Ad-hoc use on the associated RMX. You can use Max Ports to limit the extent of Ad-hoc usage of an RMX and thereby reserve ports for scheduled usage.
- **Trunk Groups** to the administered two-way ISDN H.323 trunk groups you added. All entries must be of the same carrier type (that is, all H.323 trunks).
- **Far End Resource Info?** to **y**.
- **ID Range** to the range of conference IDs. The IDs you specify on this form must *NOT* be configured on the RMX. You must leave these IDs free for the factory to create its own conferences there. Note that AAR and UDP are not used to connect to these meeting room numbers. Conference IDs (and factory numbers) are completely independent of the dial plan.
- **Priority Factory Number**. This number represents the Entry Queue created on the RMX and corresponds to a priority conference service level (for example, 784 Kbps). The Priority Factory Number must *NEVER* be in the conference ID range. If this field is left blank, all conferences can use the bridge. However, priority conferences will try to find a video bridge that has a priority factory number (if there is one).
- **Standard Factory Number**. This number represents the Entry Queue created on the RMX and corresponds to a standard conference service level (for example, 384 Kbps). The Standard Factory Number must *NEVER* be in the conference ID range. If this field is left blank, non-priority conferences cannot use this video bridge. A conference started by a priority user with non-priority users may be moved to a priority bridge, and the non-priority users will connect to it and receive video. Note that you must specify either a **Priority Factory Number** or a **Standard Factory Number.** You cannot leave both fields blank.

```
add video-bridge 32                                         Page   1 of   1
                              VIDEO BRIDGE

 Bridge ID: 32
      Name: Ad-hoc Video Bridge - RMX

 Max Ports: 80

 Trunk Groups: (Must have at least one incoming and one outgoing, or a two-way)
 1: 32
 2:
```

```
   3:

   Far End Resource Info? y

             ID Range: 49000      to 49100
  Priority Factory Number: 1001
  Standard Factory Number: 1000
```

Here are some helpful troubleshooting screens for Ad-hoc video:

```
display capacity                                          Page   7 of  13
                         SYSTEM CAPACITY


                                                       System
                                        Used Available Limit
                                        ----------------------
                        TRUNKS
                          DS1 Circuit Packs:    1    521      522
                  DS1 With Echo Cancellation:    0    522      522
                    ICHT For ISDN/SIP Trunks:    8   9991     9999
          ISDN CBC Service Selection Trunks:    0    200      200
                             Trunk Groups:    25   1975     2000
                             Trunk Ports:  1854  10146    12000
      H.323 Trunks (included in 'Trunk ports'):  211   7789     8000
  Remote Office Trunks (included in 'Trunk ports'):   0     0        0
          SBS Trunks (included in 'Trunk ports'):    0   1000     1000
          SIP Trunks (included in 'Trunk ports'): 1619   3381     5000
          Ad-hoc Video Conferencing Ports:    80   7920     8000
```

```
list video-bridge

                          VIDEO BRIDGES

No.  Name                      Ports  Type   Status    Network Region
32   Ad-hoc Video Bridge - RMX   80    H.323  in-svc         2
```

```
status video-bridge 32
                       VIDEO BRIDGE STATUS

        Name: Ad-hoc Video Bridge - RMX        Ports Used: 0     of 80
 Bridge Status: in-service                    Network Region: 2

   PRIORITY CONFERENCE STATUS
             Call Rate: 1920 Kbits Maximum    Ports Available: 40   of 40
    Conferences Available: 40   of 400        Ports Rsvd per Conf: 1
 Conference Failure Rate: 0   of 0     Participant Failure Rate: 0   of 0

 STANDARD CONFERENCE STATUS
             Call Rate: 384  Kbits Maximum    Ports Available: 40   of 40
    Conferences Available: 40   of 400        Ports Rsvd per Conf: 1
 Conference Failure Rate: 0   of 0     Participant Failure Rate: 0   of 0
```

7. Configure the Polycom RSS 2000

The RMX enables audio and video recording of conferences using Polycom RSS 2000 recording system. The recording system can be installed at the same site as the conferencing MCU or at a remote site. Several MCU's can share the same recording system.

The following are the steps to configure the RSS 2000 as a new installation.

1. Plug a crossover LAN cable to LAN1 on the back of the RSS unit. Plug into laptop and set IP settings to:
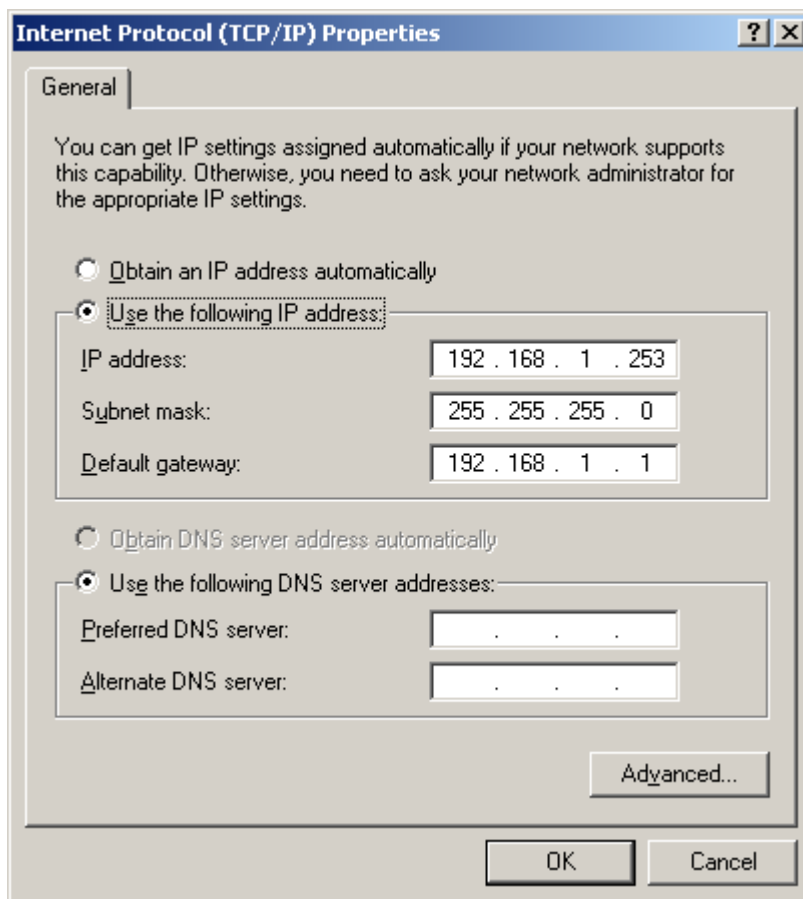
2. Open a browser and go to: http://192.168.1.254
   There are two default accounts:
   administrator/polycom and POLYCOM/POLYCOM

3. Go to System Management → System Configuration → IP settings and modify the IP address, Subnet Mask, and Default Gateway. The user has the option of setting a static IP or choosing DHCP. Click the Save button.
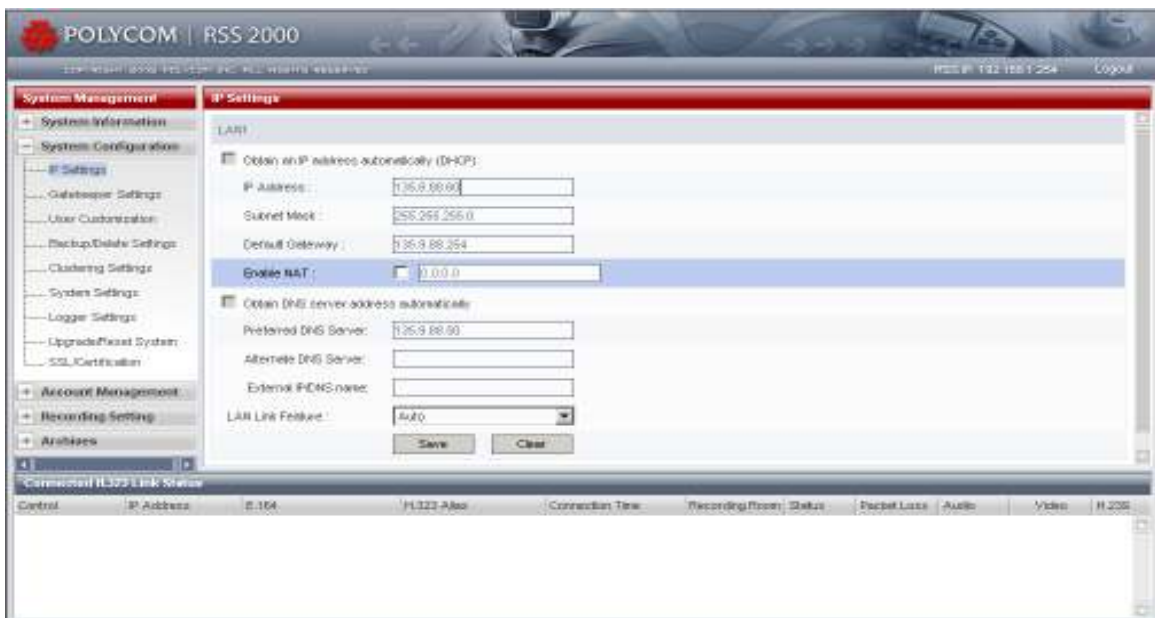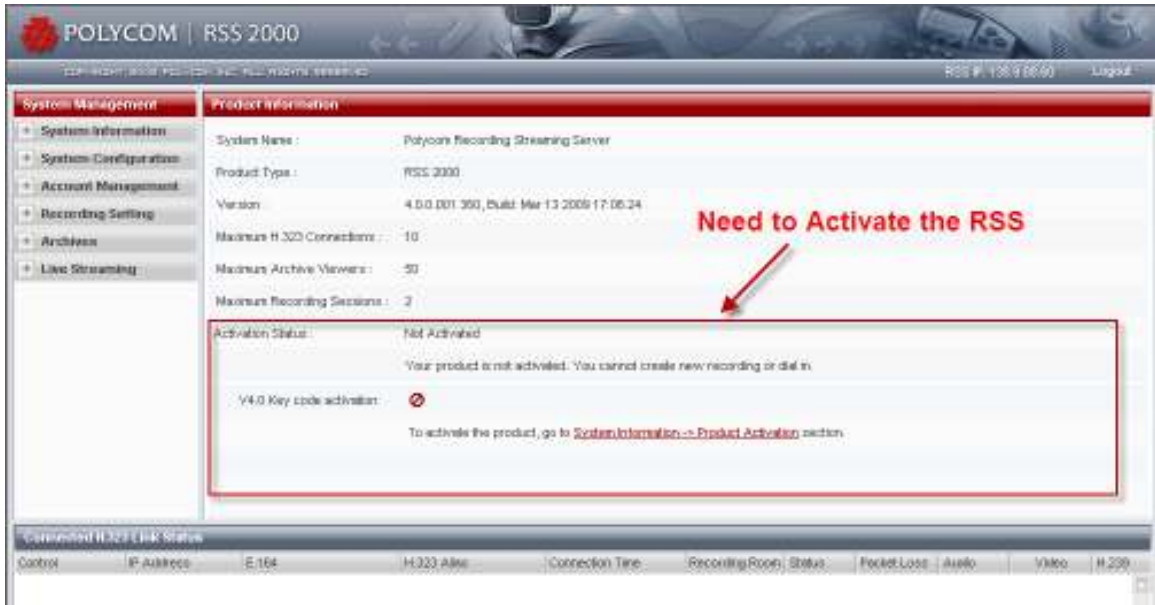
After clicking the **Save** button you will see the following reset message:
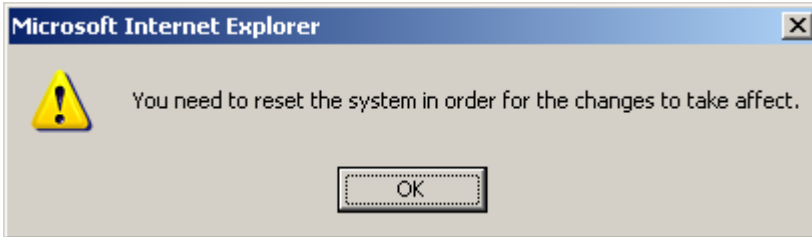
4. In order to reset the system go to System Configuration → Upgrade/Reset System and select **Restart**.
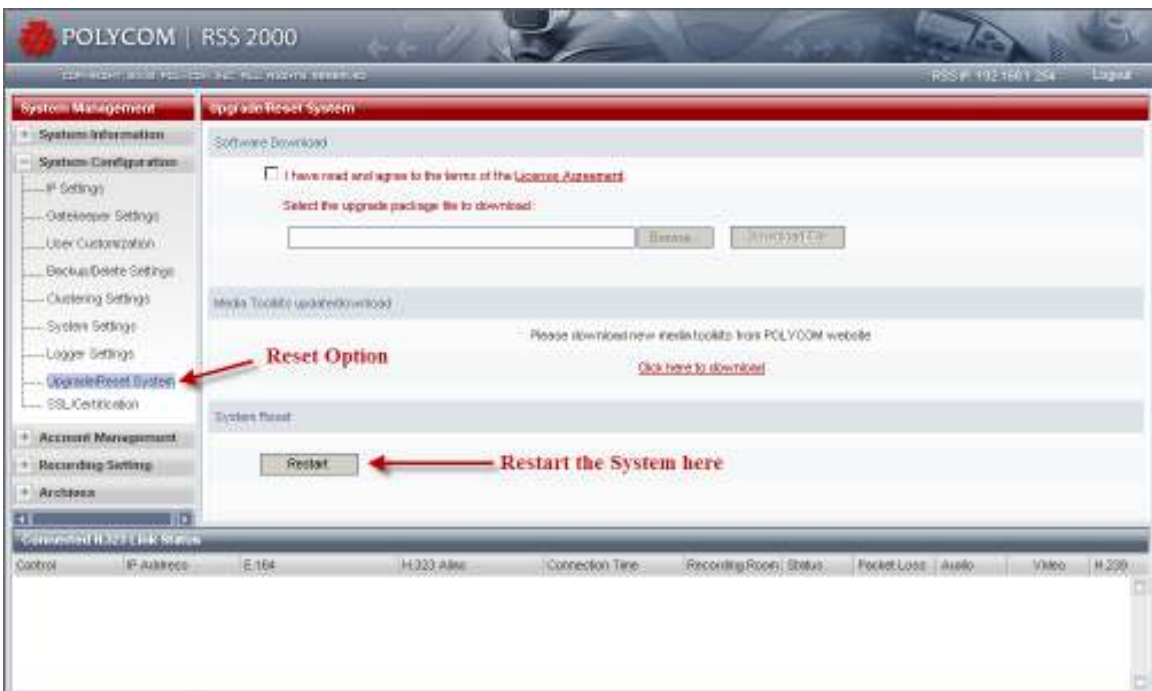
5. Once the RSS 2000 has restarted login and activate the RSS with the correct Key codes. From the **System Management → System Information → Product Information** section you will be able to identify the Activation Status. The screen capture below is an example of an RSS 2000 that is Not Activated.

TJM; Reviewed:
SPOC 11/9/2009

Solution & Interoperability Test Lab Application Notes
©2009 Avaya Inc. All Rights Reserved.

46 of 66
PolycomAppNotes

6. For new installations, an activation keycode must be entered before attempting to use the RSS 2000. The following are two types of keycodes.

Package Name	Description
K-keycode	Option Activation (Basic, Multicast, Clustering , Encryption)
U-keycode	New Version activation (Will not affect the current installed options)

From the **System Information → Product Activation** section click on **Polycom Resource Center**.
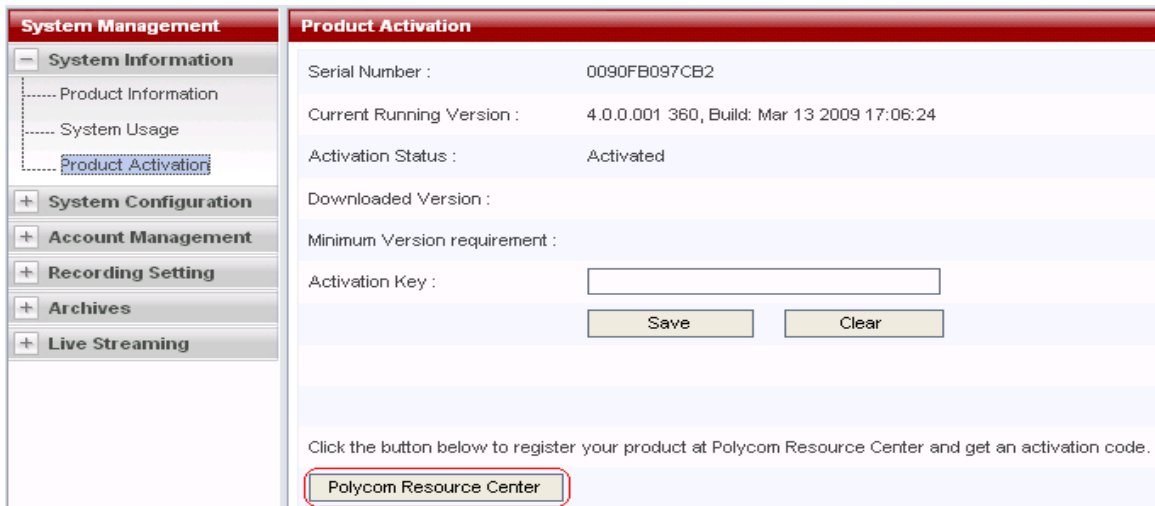
Go to the PRC and under product activation select the following:

7. For a new installation, the system will be shipped with a K License. This K License will be used by the user to go to the PRC and receive a K Keycode from the Single License number screen as shown in the screenshots. Do not click 'Retrieve Software Key Code' in this step.

Single License Number

Please enter the License Number and Serial Number of your product to generate a Key Code. To retrieve a previously enabled Key Code, please enter the product's serial number and leave License Number blank.

License Number:

—This license number is provided on a single piece of paper shipped with the system. Enter the number here.

Serial Number:

Key Code:

Generate

8. The user will receive a K Keycode with the purchased, enabled options (Multicast, Clustering, and Ecryption). Keep this keycode in your records. After entering the option activation K Keycode in the RSS 2000 web UI activation screen, the user will need to go back to the PRC and generate a U Keycode to permit a software upgrade. This is done in the Single Upgrade Key Code page shown below, accessed by clicking the Retrieve Software Key Code button.

To install the U-keycode:

a. Enter 4.0 in the Version field and enter the RSS 2000 Serial Number in the Serial Number field.

The user will receive a U-Key code permitting activation of Version 4.x software. This keycode will also need to be entered in the product activation section on the RSS 2000 Web UI):

Retrieve Your Upgrade Key Code

Single Upgrade Key Code

To retrieve the latest software upgrade Key Code for your system, please enter the complete system Serial Number, the Version Number that you want to install and then click **Retrieve**.

*Serial Number: Model:

*Version: 4.0   Need Help?

Retrieve
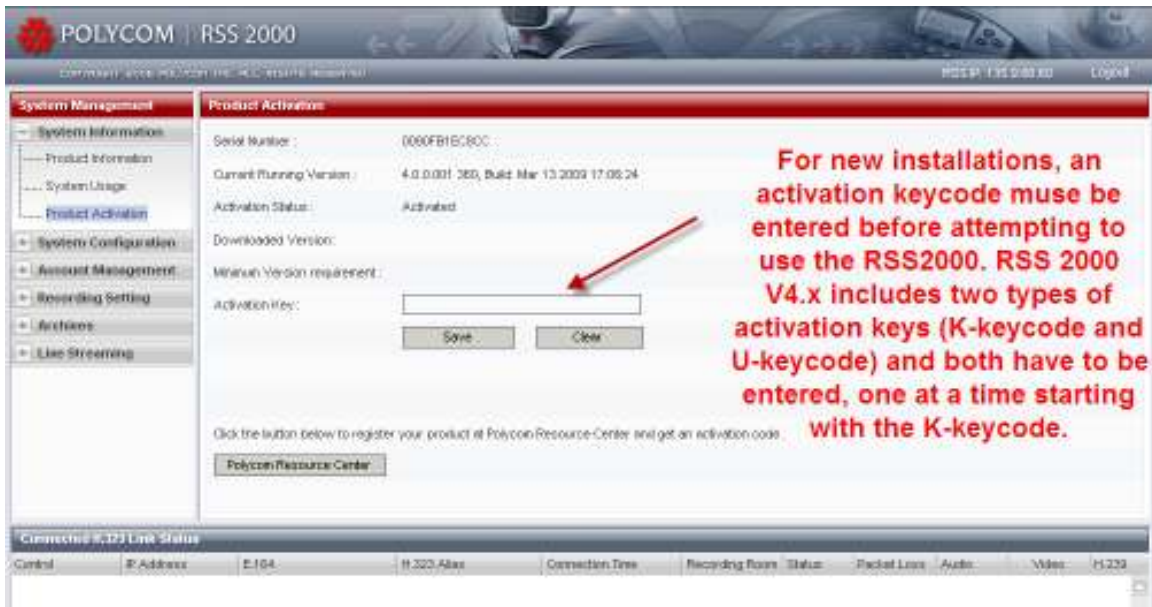
After entering the correct K keycode and U keycode the System Information → Product Information section will display Activation Status: Activated. The Key code activation will also have a green check mark next to it as well as any of the other features (Multicast, Clustering, and Encryption) that were ordered and paid for. The RSS 2000 will be ready for use.
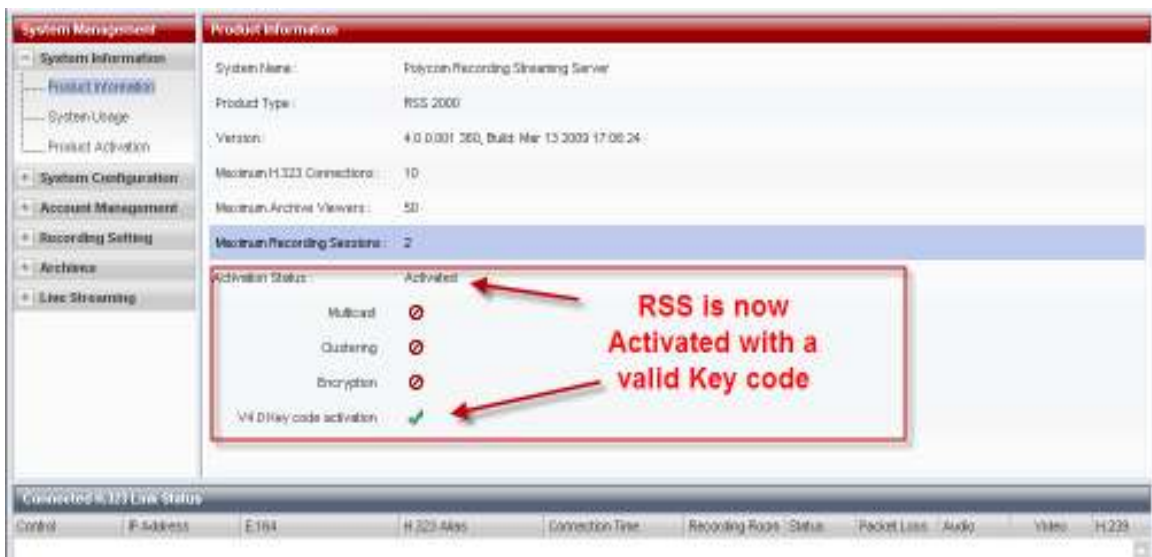
9. Go to **System Management → System Configuration → Gatekeeper Settings** and define the Gatekeeper Settings as necessary. In this case the Gatekeeper is the CMA 4000. In the screen capture below the RSS 2000 is already registered with the CMA.

10. Define the reset of the System Management options in the list as deemed necessary.

8. Configuring the RMX to Enable Recording

11. Recording conferences is enabled via a Recording Link on the RMX, which is a dial-out connection from the conference to the recording system. Recording can start automatically, when the first participant connects to a conference, or on request, when the RMX user or conference chairperson initiates it.

    Use the RMX's Recording Link function to initiate a recording connection with the RSS 2000 and record RMX conferences, or add the RSS as a participant in an RMX conference.

    Before using the RMX to perform recording, you must first set Recording Link parameters and the address of the RSS 2000 server on the RMX. Select **Recording Links** from the **Rarely Used** section and click on the green plus (New Recording Link). Enter an appropriate **Name**, an **Alias** if desired, and the IP Address of the RSS 2000. Also enter an Alias Type if desired.
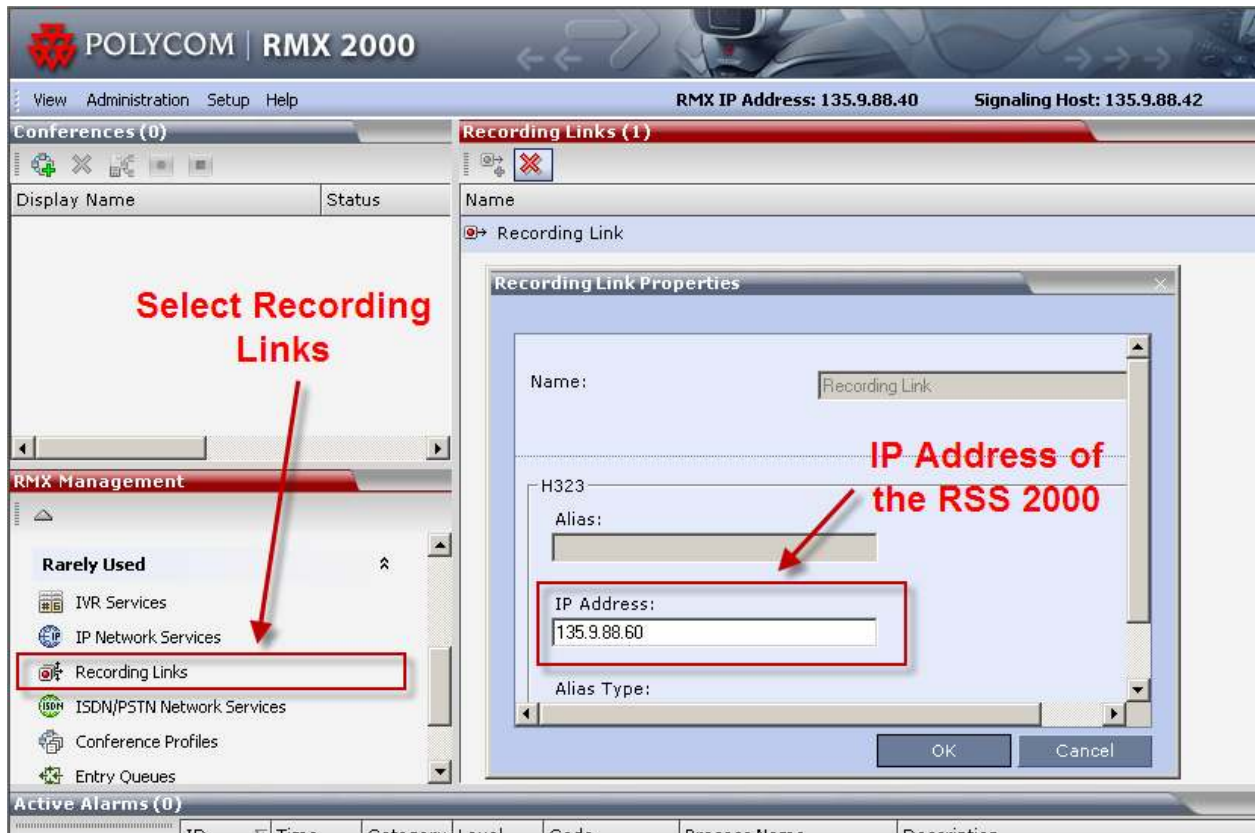
The RMX will then establish a connection to the desired RSS 2000 when a conference is started. Depending on the RMX's settings, the RSS 2000 can support two types of recording methods: Automatically activated conference recordings, or recordings initiated by MCU operators and conference chairperson participants. The RSS 2000 records the audio/video bit streams transmitted by the RMX. It should be noted that RMXs are used to perform H.239 dual-stream recording. By default, all conferences, Entry Queues, and Meeting Rooms launched on the RMX 2000 have H.239 capabilities. The RMX supports only H.239, so this is not a concern for recording RMX conferences. Other devices, like the MGC and many Polycom endpoints, support H.239 and the proprietary People + Content standard. In these cases, ensure that the device has H.239 enabled as the RSS cannot record the proprietary People + Content protocol.

When an RMX is used to perform a recording, the resulting files will be named according to the following rule: the RMX's Recorded Conference Name + Recording time stamp (Date & Time. For instance, if a conference from the RMX is being recorded and the conference name is set to 'Demo' performs a recording at 10:15 on April 03, 2009, the resulting file will have the name "Demo_APR_03_2009_10: 15".

8.1. Enabling the Recording in the Conference Profile

12. In the *RMX Management* pane, click the **Conference Profiles** button. The *Conference Profiles* list is displayed.

TJM; Reviewed:
SPOC 11/9/2009
Solution & Interoperability Test Lab Application Notes
©2009 Avaya Inc. All Rights Reserved.
52 of 66
PolycomAppNotes

13. Create a new profile by clicking the green plus button (**New Profile)** or modify an existing profile by double-clicking or right-clicking an existing profile and then selecting **Profile Properties**.
14. In the *Profile Properties* dialog box, click the **Recording** tab.
15. Select the **Enable Recording** check box and select desired Start Recording option. Click OK.
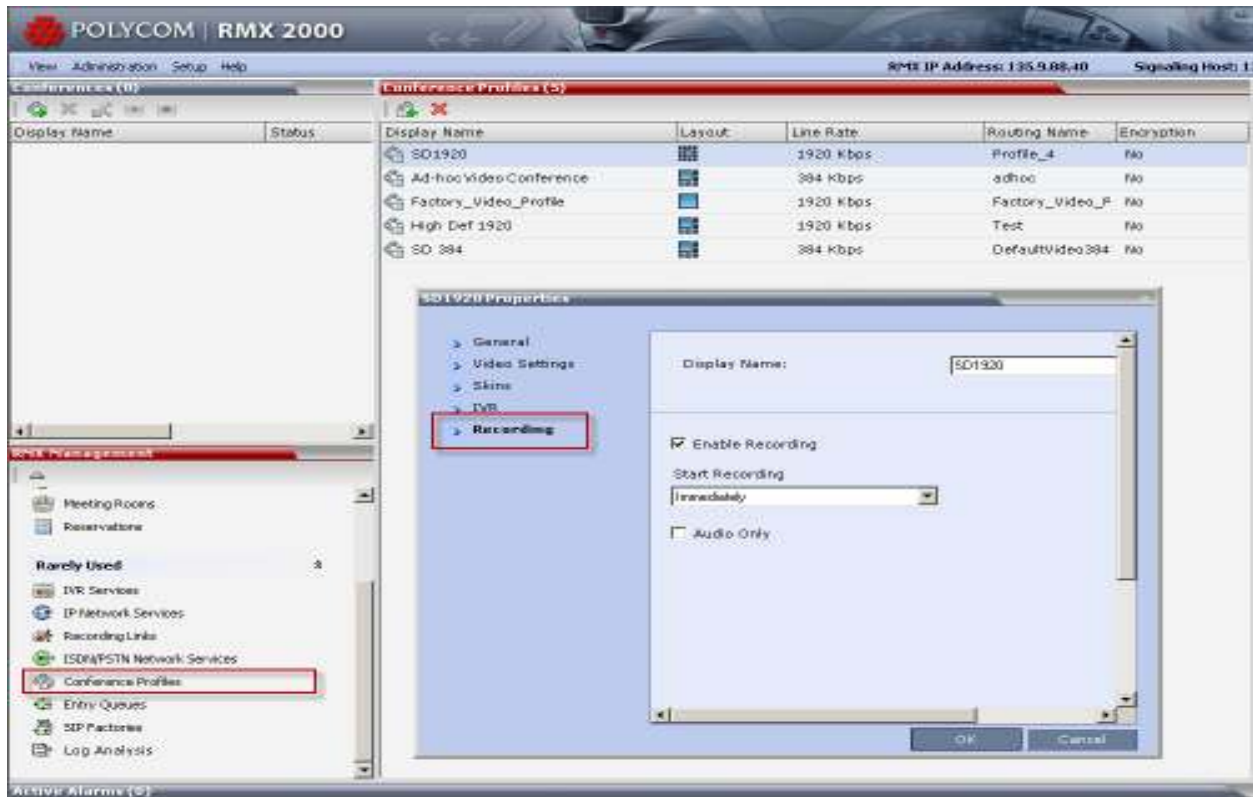
8.2. Managing the Recording Process

When a conference is started and recording is enabled in its Profile, the system will automatically start the recording if the *Start Recording* parameter is set to *immediately*. If it is set to *Upon Request*, the system waits for the chairperson or RMX user's request. Once the recording is initiated for a conference, the RMX connects to the Recording device (RSS 2000) using the default Recording Link. The connection that is created between the conference and the recording device is represented as a special participant (Recording) which represents the Recording Link. Once the recording has started, the recording process can be stopped and restarted from the Chairperson's endpoint (using DTMF codes) or from the RMX Web Client. After the recording process has finished, the recording can be identified in the RSS 2000 by its RMX conference name.

8.2.1. Using the RMX Web Client to Manage the Recording Process

When logged into the RMX and a conference is active you will see under Display Name the name of the conference(s). In this case it's SIL Meeting Room1. Click on it once. This will bring up the participants that are in this conference.

To manage the recording process using the right-click menu:

Right-click the *Recording* participant in the conference and select one of the options.

To manage the recording process using the Conference toolbar:

In the *Conferences* pane, click one of the buttons (explained below) in the Conference toolbar. **Note**: The recording buttons will only be displayed in the conference toolbar for that is recording-enabled.

TJM; Reviewed:
SPOC 11/9/2009

Solution & Interoperability Test Lab Application Notes
©2009 Avaya Inc. All Rights Reserved.

54 of 66
PolycomAppNotes

Button	Description
●	Start/Resume recording. This button toggles with the *Pause* button.
■	Stop recording.
‖	Pause recording. This button toggles with the *Start/ Resume* button.

Conferences List - Recording Toolbar buttons

Using DTMF Codes to Manage the Recording Process

By entering the appropriate DTMF code on the endpoint, the chairperson can **Stop** the recording (*74), **Pause** it (*75), or **Start/Resume** the recording (*73). For more information on managing the recording process via DTMF codes, see the *RSS 2000 User's Guide*.

For additional details please refer to the Polycom product users guides concerning how to set up an RSS 2000 and RMX 2000 Recording Link.

9. Configure the Video Endpoints

Before administering any video endpoints on the Communication Manager, you must perform the following steps:
1. Use the **change ip-codec-set** *x* command (where *x* is the chosen IP codec set) to:
    - Define the codecs (page 1 of form). The following codecs are recommended:
        - SIREN14-32K (1 fpp, 20 ms)
          SIREN14-32K are wideband codecs. Since most Polycom systems are not configured for stereo, it is not recommended to use a stereo SIREN codec as a default.
        - G722-64K (1 fpp, 20 ms)
          G722-64K are wideband codecs. These codecs allow wideband with video endpoints that do not support SIREN codecs.
        - G.729A (no silence suppression, 2 fpp, 20 ms)
          Polycom systems do not support all variants of G.729 codecs. If you want to use G.729, you must specify G.729A. If you specify G.729, audio problems arise. All variants of G.729 codecs are narrowband codecs.
          NOTE:
          Wideband codecs should appear before narrowband codecs in the codec set.
    - Set **Allow Direct-IP Multimedia** to **y** (page 2 of form).
    - Set **Maximum Call Rate for Direct-IP Multimedia**. This setting is the combined audio and video transmit rate or receive rate for non-priority (normal) video calls. You can use this setting to limit the amount of bandwidth used for normal video calls.

For example, if you select 384 Kbits, a maximum of 384 Kbits will be used to transmit *and* to receive audio/video.

- **Maximum Call Rate for Priority Direct-IP Multimedia**. This setting is the combined audio and video transmit rate or receive rate for priority video calls. You can use this setting to limit the amount of bandwidth used for priority video calls. For example, if you select 384 Kbits, a maximum of 384 Kbits will be used to transmit *and* to receive audio/video.

Repeat Step 1 for each IP codec set that will be used for video.

2. Use the **change ip-network-region *x*** command (where *x* is the chosen IP network region) to set the following parameters:
   - **Intra-region IP-IP Direct Audio** to **yes**.
   - **Inter-region IP-IP Direct Audio** to **yes**.
     NOTE: Shuffling is recommended. However, you can set shuffling to **no**, and video calls will work properly.
   - **Security Procedures 1** to **any-auth** (page 2 of form).
   - **Codec set** (page 3 of form) to one of the codec sets you defined in Step 1.
   - **Video Norm** (page 3 of form) to the amount of bandwidth that you want to allocate for the normal video pool to each IP network region.
   - **Video Prio** (page 3 of form) to the amount of bandwidth that you want to allocate for the priority video pool to each IP network region.
   - **Video Shr** (page 3 of form). Specify whether the normal video pool can be shared with the audio pool for each link between IP network regions.

Repeat Step 2 for each IP network region that will be used for video in this system.

9.1. Configure a Station Endpoint for Avaya IP Softphone and One-X Communicator on Communication Manager

Use this procedure to enable video calls for a desktop user. Perform the following steps to configure a station for Avaya IP Softphone and 1XC:

1. Use the **display system-parameters customer-options** command to verify the Maximum Video Capable IP Softphones (page 2 of the form). This number is provided by the RFA license file.

```
display system-parameters customer-options                    Page   2 of  11
                         OPTIONAL FEATURES

IP PORT CAPACITIES                                               USED
                  Maximum Administered H.323 Trunks: 8000   211
          Maximum Concurrently Registered IP Stations: 18000 18
            Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
             Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 12000 0
                    Maximum Video Capable Stations: 12000 12
                Maximum Video Capable IP Softphones: 12000 27
```

2. Use the **change cos** command to set **Priority Ip Video** (page 2 of form) for the appropriate COS levels.

```
display cos                                                    Page   2 of   2
                              CLASS OF SERVICE


                          0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 VIP Caller               n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n

 Masking CPN/Name Override n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Call Forwarding Enhanced y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Priority Ip Video        n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Ad-hoc Video Conferencing y  y  y  n  n  n  n  n  n  n  n  n  n  n  n  n
```

3. Use the **add station** command to add an Avaya IP Softphone or 1XC station, and set the following parameters for that station:
   - **IP Softphone** to **y**.
   - **IP Video Softphone** to **y**.
   - If you want this station to be able to make priority video calls, make sure you select a COS level that has **Priority Ip Video** enabled.
   - On page 2 of the form, set **Direct IP-IP Audio Connections** to **y**.

   Repeat Step 3 for each video-enabled Avaya IP Softphone or 1XC endpoint you want to configure.

```
add station 50025                                             Page   1 of   5
                                   STATION

Extension: 50025                   Lock Messages? n              BCC: M
     Type: 4620                    Security Code: 123456          TN: 1
     Port: S01586                  Coverage Path 1:               COR: 1
     Name: IP Video Softphone      Coverage Path 2:               COS: 1
                                   Hunt-to Station:
STATION OPTIONS
                                   Time of Day Lock Table:
           Loss Group: 19      Personalized Ringing Pattern: 1
                                     Message Lamp Ext: 50025
          Speakerphone: 2-way       Mute Button Enabled? y
      Display Language: english        Expansion Module? n
 Survivable GK Node Name:
        Survivable COR: internal      Media Complex Ext:
  Survivable Trunk Dest? y             IP SoftPhone? y

                                   IP Video Softphone? y


                                   Customizable Labels? y
```

9.2. Configure Polycom VSX/HDX Series Video Conferencing Systems on Communication Manager

Use this procedure to configure Polycom VSX/HDX video conferencing systems. When setting up these systems, you will need to know the following information:

- Maximum number of VSX/HDX systems on your network
- PIN for each VSX/HDX system. The PIN can consist of a maximum of eight numeric characters and is defined by the System Administrator. The PIN must be the same number as is defined on the station form **Security Code**.
- The key code that combines the Avaya option with any other Polycom options.
- Whether the VSX/HDX system has the multipoint option. If so, you must combine the Polycom Software License for this capability with the "Avaya Option" Polycom Software License to create a single Key Code to input into the unit.
- IP address of the voice system.
- IP codec sets you want to use.
- IP network regions you want to use.

Perform the following steps to configure Polycom systems on Communication Manager:
1. Use the **display system-parameters customer-options** command to verify the M**aximum Video Capable Stations** (page 2 of form). This number is provided by the RFA license file. The **Maximum Video Capable Stations** was determined using the following criteria.
    - Each single-point VSX/HDX system is considered to be one station.
    - Each VSX multpoint system can be **three** to **six** stations.
    - Each HDX system can be three stations for multipoint plus 4 and seven for multipoint plus 8 multipoint licensed options for the HDX9004. The HDX9002 only has multipoint plus 4 as an option.
2. Use the **change cos** command to set **Priority Ip Video** (page 2 of form) for the appropriate COS levels.
3. Use the **add station** command to add a station for the Polycom system. Set the following parameters:
    - **Type** to **H.323**.
    - **Security Code** to the "pin" you will administer for the VSX or HDX system.
    - **IP Video** to **y**.
    - If you want this station to be able to make priority video calls, make sure you select a COS level that has **Priority Ip Video** enabled.
    - On page 2 of the form, set **Direct IP-IP Audio Connections** to **y**.

    NOTE: You can create an alias for VSX/HDX stations.

```
add station 50080                                             Page   1 of   4
                                STATION

Extension: 50081                      Lock Messages? n            BCC: M
     Type: H.323                      Security Code: 123456        TN: 1
     Port: S01767                    Coverage Path 1:             COR: 1
     Name: HDX8000                   Coverage Path 2:             COS: 2
                                     Hunt-to Station:            Tests? y
STATION OPTIONS
                                       Time of Day Lock Table:
             Loss Group: 19        Message Waiting Indicator: none

                                     Authentication Required? y


        Survivable COR: internal
  Survivable Trunk Dest? y
           DTMF over IP: in-band
                                                 IP Video? y
```

If the VSX/HDX system has the multipoint option, perform the following steps:
1. Use the **add station** command to add a second station for the Polycom system.
2. Set **Type** to **H.323**.
3. Set **Security Code** to the "pin" you will administer for theVSX/HDX. Make sure the security code is the same as the previous station. All the stations configured for a single VSX or HDX system must have the same security code.
4. Set **IP Video** to **y**.
5. On page 2 of the form, set **Direct IP-IP Audio Connections** to **y**.
6. Set **IP Audio Hairpinning** to **y**.
7. If you want this station to be able to make priority video calls, make sure you select a COS level that has **Priority Ip Video** enabled.
8. Repeat Steps 1 through 7 to create the third consecutive station. For VSX systems, you can have up to **six** stations.
9. Use the **change station** *xx* command (where *xx* is the first station you added for the Polycom system) to set **Hunt-to Station** to the second station you added for the Polycom system.
10. Use the **change station** *xx* command (where *xx* is the second station you added for the Polycom system) to set **Hunt-to Station** to the third station you added for the Polycom system.
11. Use the **change station** *xx* command (where *xx* is the third station you added for the Polycom system) to set **Hunt-to Station** to the first station you added for the Polycom system. All stations must be in a circular hunt. If you added more than three stations for the Polycom system, use the **change station** *xx* command to set **Hunt-to Station** for each station.

```
add station 50081                                            Page   1 of   4
                                STATION

Extension: 50081                     Lock Messages? n              BCC: M
      Type: H.323                    Security Code: 123456          TN: 1
      Port: S01767                  Coverage Path 1:               COR: 1
      Name: HDX8000                 Coverage Path 2:               COS: 2
                                    Hunt-to Station: 50082        Tests? y
STATION OPTIONS
                                       Time of Day Lock Table:
             Loss Group: 19         Message Waiting Indicator: none

                                     Authentication Required? y


          Survivable COR: internal
   Survivable Trunk Dest? y
           DTMF over IP: in-band
                                               IP Video? y
```

```
add station 50082                                            Page   1 of   4
                                STATION

Extension: 50081                     Lock Messages? n              BCC: M
      Type: H.323                    Security Code: 123456          TN: 1
      Port: S01767                  Coverage Path 1:               COR: 1
      Name: HDX8000                 Coverage Path 2:               COS: 2
                                    Hunt-to Station: 50083        Tests? y
STATION OPTIONS
                                       Time of Day Lock Table:
             Loss Group: 19         Message Waiting Indicator: none

                                     Authentication Required? y


          Survivable COR: internal
   Survivable Trunk Dest? y
           DTMF over IP: in-band
                                               IP Video? y
```

```
add station 50083                                            Page   1 of   4
                                STATION

Extension: 50081                     Lock Messages? n              BCC: M
      Type: H.323                    Security Code: 123456          TN: 1
      Port: S01767                  Coverage Path 1:               COR: 1
      Name: HDX8000                 Coverage Path 2:               COS: 2
                                    Hunt-to Station: 50081        Tests? y
STATION OPTIONS
                                       Time of Day Lock Table:
             Loss Group: 19         Message Waiting Indicator: none

                                     Authentication Required? y


          Survivable COR: internal
   Survivable Trunk Dest? y
           DTMF over IP: in-band
                                               IP Video? y
```

9.3. Configure Polycom VSX/HDX Series Video Conferencing Systems

Perform the following steps to configure VSX/HDX Polycom Systems registered to Communication Manager Gatekeeper:

1. Install the Polycom system and connect it to your network.
2. Upgrade the Polycom system software (if necessary).
3. Using a web browser, access the Polycom home page for the unit, and select **Admin Settings**>**Network**>**IP Network**.
4. Select the **Enable IP H.323** check box.
5. Select the **Display H.323 Extension** check box.
6. In the **H.323 Extension (E.164)** box, enter the first station extension number you specified for this system on the Communication Manager system.
7. From the Use Gatekeeper box, select **Specify with PIN**.
8. In the Gatekeeper IP Address box, enter the IP address of the CLAN or Processor Ethernet (procr) followed by *:1719* (to specify the correct port to use).
9. In the Authentication PIN box, enter the security code you entered when administering the station on Communication Manager.
10. In the Number box in the Gateway area, enter the H.323 extension you specified.
11. Select the **Enabled PVEC** check box.
12. In the Type of Service box in the Quality of Service area, select the appropriate setting. Both **IP Precedence** and **DiffServ** are supported. Contact your Network Administrator for this information.
13. In the Type of Service Value boxes (Video, Audio, and Far End Camera Control), enter the QoS values for the IP Network Region settings in which the VSX/HDX station belongs.
14. Select the **Dynamic Bandwidth** check box.
15. From the Maximum Transmit Bandwidth box, select the setting that matches the Maximum Call Rate for Direct-IP Multimedia setting you specified for the Communication Manager system.
16. From the Maximum Receive Bandwidth box, select the setting that matches the Maximum Call Rate for Direct-IP Multimedia setting you specified for the Communication Manager system.
17. Complete the Firewall and Streaming sections as necessary.
18. When finished, click the **Update** button at the top (see **Figure 3** below).
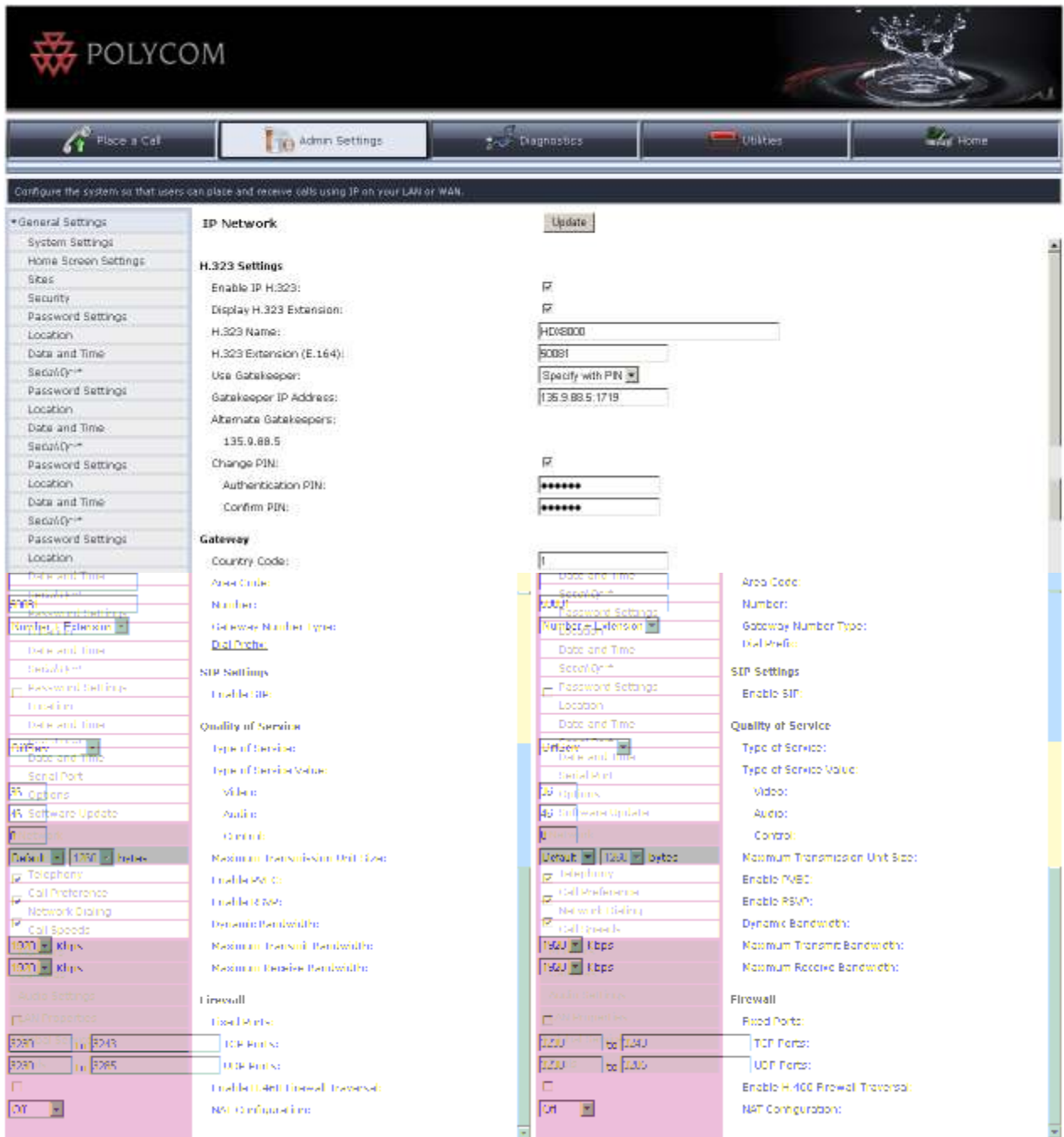
Repeat above Steps for each Polycom system.

Figure 3. Example of a Polycom HDX8000 registered to Communication Manager

Perform the following steps to configure VSX/HDX Polycom Systems registered to Polycom CMA Gatekeeper:

1. Install the Polycom system and connect it to your network.
2. Upgrade the Polycom system software (if necessary).
3. Using a web browser, access the Polycom home page for the unit, and select **Admin Settings**>**Network**>**IP Network**.
4. Select the **Enable IP H.323** check box.
5. Select the **Display H.323 Extension** check box.
6. In the **H.323 Extension (E.164)** box, enter the first station number you specified for this system on the Polycom CMA Gatekeeper.
7. From the Use Gatekeeper box, select **Specify**.
8. In the Gatekeeper IP Address box, enter the IP address of the Polycom CMA Gatekeeper followed by *:1719* (to specify the correct port to use).
9. In the Number box in the Gateway area, enter the H.323 extension you specified.
10. Select the **Enabled PVEC** check box.
11. In the Type of Service box in the Quality of Service area, select the appropriate setting. Both **IP Precedence** and **DiffServ** are supported. Contact your Network Administrator for this information.
12. In the Type of Service Value boxes (Video, Audio, and Control), enter the QoS values necessary. Contact your Network Administrator for this information.
13. Select the **Dynamic Bandwidth** check box.
14. From the Maximum Transmit Bandwidth box, select the appropriate setting. Contact your Network Administrator for this information.
15. From the Maximum Receive Bandwidth box, select the appropriate setting. Contact your Network Administrator for this information.
16. Complete the Firewall and Streaming sections as necessary.
17. When finished, click the **Update** button at the top (see **Figure 4** below).

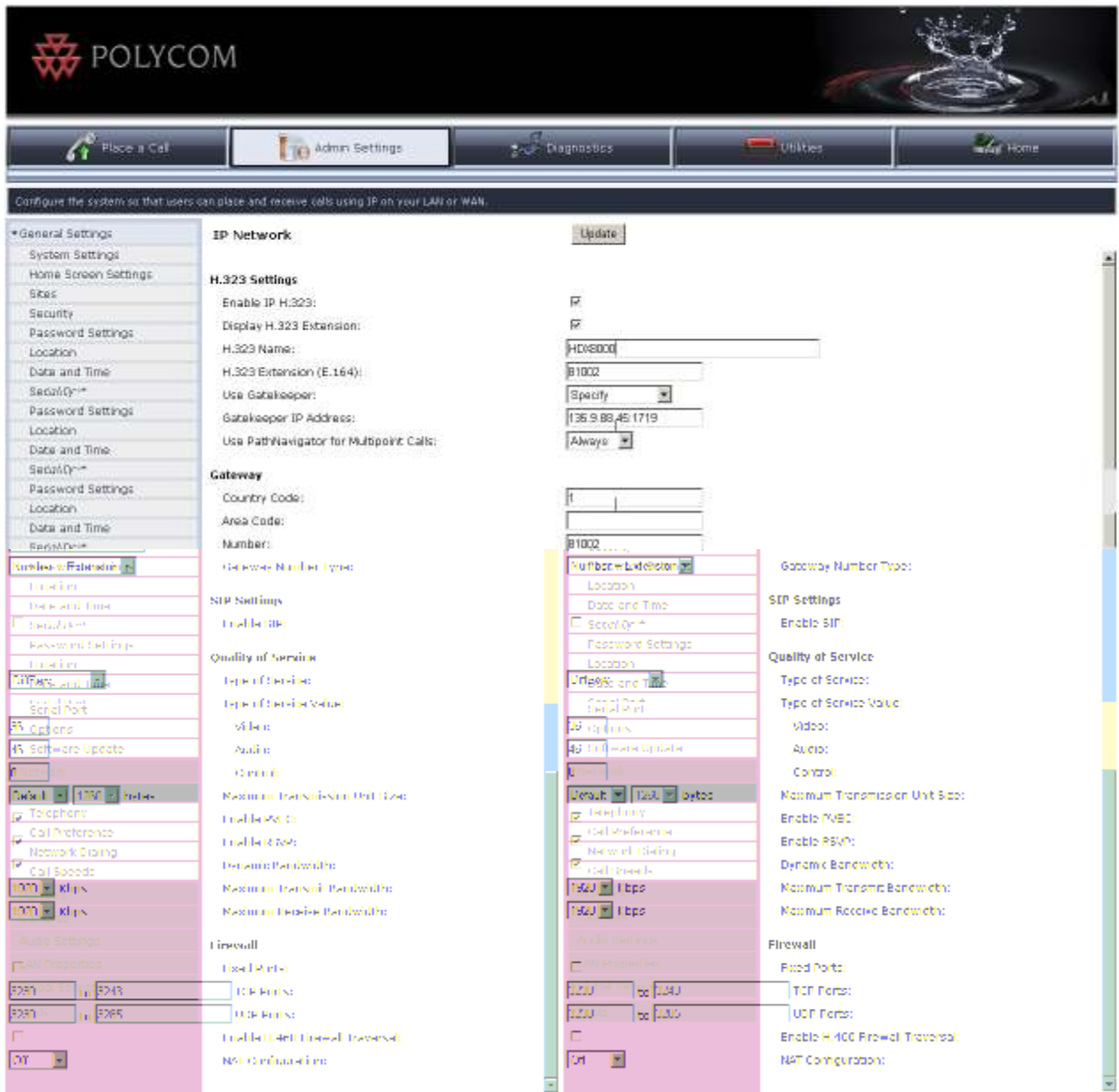Repeat above Steps for each Polycom system.

Figure 4. Example of a Polycom HDX8000 registered to Communication Manager

10. Conclusion

The H.323 Video interoperability between Avaya and Polycom consisting of Communication Manager Gatekeeper, Polycom CMA 4000 Gatekeeper, RMX 2000, RSS 2000, Avaya IP Softphone, Avaya One-X Communicator, VSX 3000, VSX5000, HDX 4002, HDX 8006, and HDX9005 Endpoints has been tested and passed.

11. Additional References

Avaya references, available at http://support.avaya.com

Polycom references are available at http://www.polycom.com/support/