This Quick Setup guide is intended for technicians installing the following video endpoints:

- Avaya IP Softphone Release 6.0 and Video Integrator
- Polycom VSX video conferencing systems with Release 8.5.3 or later
- Polycom HDX video conferencing systems
- Polycom V500 and V700 video calling systems
- Polycom RMX ideo conferencing bridge platform
- Polycom MGC video conferencing bridge platforms with Release 8.0.0.27
- Avaya Meeting Exchange 5.1 S6800 bridge
- third-party gatekeepers, including Polycom PathNavigator™ and SE200 gatekeepers
- H.320 gateways

This guide also describes how to configure video trunks between two systems running Avaya Communication Manager Release 5.2 or later.
Before performing the procedures in this guide, make sure the following conditions exist:

- Avaya Communication Manager Release 5.2 is installed and running.
- You are familiar with administering Avaya Communication Manager.
- The Avaya-enabled Polycom MGC Manager software is installed.

Network Readiness

You must perform a network readiness or network assessment to ensure your network is capable of supporting the high bandwidth demands of video over IP. You should also consider implementing QoS across your network.

New Features in This Release

Avaya Video Telephony Solutions Release 5.2 introduces the following new features and enhancements:

- Added support for Siren 22 wideband audio codecs when shuffled to direct-IP audio connectivity between the endpoints/MCUs.
- A limitation of two Communication Manager servers apply. Tandeming of more PBXs is not supported and remains single-band audio.
- H.245 signaling support for Polycom's Lost Packet Recovery (LPR) feature.
- Improved video interoperability with third party gatekeepers and PBXs including Cisco's Call Manager (limited to basic video calling).
- A one button Instant Transfer from a deskset to a video executive desktop system, for example a HDX4000, to allow the user to make all call originations from the deskset.
- Support for one-X communicator with H.323 video only, including support for VGA resolution. One-X communicator SIP video will be available in a future AVTS release.
- The display of the status of each video bridge on the list video-bridge command.

Required Administration

Before administering any video endpoints on your system, 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:
     - SIREN14-48K (1 fpp, 20 ms)
       SIREN14-48K 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.
     - G722.1-32K (1 fpp, 20 ms)
       G722.1-32K 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, no 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 2 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 the codec set 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.

Configure a Station Endpoint for Avaya IP Softphone and Video Integrator

Use this procedure to enable video calls for a desktop user. Perform the following steps to configure a station to use Avaya IP Softphone R6.0 and Video Integrator:

3. Use the **display system-parameters customer-options** command to verify the **Maximum Video Capable IP Softphones** (page 2 of form). This number is provided by the RFA license file.
4. Use the **change cos** command to set **Priority Video Calling** (page 2 of form) for the appropriate COS levels.
5. Use the **add station** command to add an Avaya IP Softphone 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 Video Calling** enabled. (See Step 2.)
   - On page 2 of the form, set **Direct IP-IP Audio Connections** to **y**.

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

Configure Polycom VSX/HDX Series Video Conferencing Systems and V500/V700 Video Calling Systems

Use this procedure to configure Polycom VSX/HDX video conferencing systems and V500 and V700 video calling systems. When setting up these systems, you will need to know the following information:

- maximum number of VSX/HDX, V500, and V700 systems on your network
- PIN for each VSX/HDX/V500/V700 system. The PIN can consist of a maximum of eight numeric characters and is defined by the System Administrator.
- the key code that combines the Avaya option with any other Polycom options.
- whether the VSX/HDX system has the multipoint option or IMCU 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:

1. Use the **display system-parameters customer-options** command to verify the **Maximum 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 V500/700 system is considered to be one station.
   - 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 four and seven for multipoint plus eight 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 Video Calling** (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, HDX, V500 or V700 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 Video Calling** enabled. (See Step 2.)
   - On page 2 of the form, set **Direct IP-IP Audio Connections** to **y**.

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

   If the VSX/HDX system has the multipoint option or IMCU option, perform the following steps:

   a. Use the **add station** command to add a second station for the Polycom system.
   b. Set **Type** to **H.323**.
   c. Set **Security Code** to the “pin” you will administer for the VSX/HDX. Make sure the security code is the same as the previous station. All three stations must have the same security code.
   d. Set **IP Video** to **y**.
   e. On page 2 of the form, set **Direct IP-IP Audio Connections** to **y**.
   f. Set **IP Audio Hairpinning** to **y**.
   g. If you want this station to be able to make priority video calls, make sure you select a COS level that has **Priority Video Calling** enabled. (See Step 2.)
   h. Repeat Steps a through g to create the third consecutive station. For VSX systems, you can have up to **six** stations.
   i. 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.
   j. 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.
   k. 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.

   NOTE:
   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. All of the stations you add must be in a circular hunt.

   Repeat Steps 1 through 3 for each Polycom system.

4. Install the Polycom system and connect it to your network.
5. Upgrade the Polycom system software (if necessary).
6. Using a web browser, access the Polycom home page for the unit, and select **Admin Settings**>**Network**>**IP Network**.
7. Select the **Enable IP H.323** check box.
8. Select the **Display H.323 Extension** check box.
9. In the H.323 Extension (E.164) box, enter the station number you specified for this system on the Avaya Communication Manager system.
10. From the Use Gatekeeper box, select **Specify with PIN**.
11. In the Gatekeeper IP Address box, enter the IP address of the CLAN or PCLAN followed by ***:1719*** (to specify the correct port to use).
12. In the Authentication PIN box, enter the security code you entered in Step 5.
13. In the Number box in the Gateway area, enter the extension you specified in Step 11.
14. Select the **Enabled PVEC** check box.
15. In the Type of Service box in the Quality of Service area, select the appropriate setting. Both **IP Precedence** and **Diffserve** are supported. Contact your Network Administrator for this information.
16. 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.
17. Select the **Dynamic Bandwidth** check box.
18. From the Maximum Transmit Bandwidth box, select the setting that matches the Maximum Call Rate for Direct-IP Multimedia setting you specified for the Avaya Communication Manager system.
19. From the Maximum Receive Bandwidth box, select the setting that matches the Maximum Call Rate for Direct-IP Multimedia setting you specified for the Avaya Communication Manager system.
20. Complete the Firewall and Streaming sections as necessary.
21. When finished, click the **Update** button.
22. Repeat Steps 4 through 21 for each Polycom system.

Configure a Polycom RMX Series Video Conferencing Bridge Platform

To configure a Polycom RMX system, perform the following steps:

1. Use the **change node-names ip** command to add an entry for the RMX system. Be sure to enter the IP address of the IP board for the RMX system.
2. Use the **add trunk-group** command to add a two-way trunk group for the RMX system. Set the following parameters:
   - **Group Type** to **isdn**.
   - **Direction** to **two-way**.
   - **Carrier Medium** to **H.323**.
   - **Service Type** to **tie**.
3. Use the **add signaling-group** command to add a signaling group for the RMX system. Set the following parameters:
   - **Group Type** to **h.323**.
   - **IP Video** to **y**.
   - **Priority Video**. If you want all incoming calls to receive priority video transmissions, select **y**.
   - **Trunk Group for Channel Selection** to the two-way trunk group you added.
   - **Near-end Node Name**. For example, for an S8300 system, you would enter ***procr***. For an S8500 or S8700 system, you would enter the name of the CLAN board
   - **Near-end Listen Port** to **1720**.
   - **LRQ Required** to **n**.
   - **RRQ Required** to **y**.
   - **Enable Layer 3 Test** to **n**.
   - **Far-end Node Name** to the name you entered for the RMX system.
   - **Far-end Listen Port** to **1720**.
   - **Far-end Network Region**
   - **Calls Share IP Signaling Connection** to **n**.
   - **Direct IP-IP Audio Connections** to **y**.
   - **IP Audio Hairpinning** to **n**.
4. Use the **change trunk-group *xx*** command (where ***xx*** is the trunk group you added in Step 2) to add members to the trunk group. The number of members depends on the maximum simultaneous calls an RMX supports.

5. Use the **change route-pattern *xx*** command (where xx is the route pattern you want to use) to create a route pattern that points to the two-way trunk group.
6. Install the Polycom system and connect it to your network.
7. Upgrade the Polycom system software (if necessary).
8. Access the Polycom home page for the unit.
9. From the Setup menu, select the **System Configuration** tab.
10. 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
11. Under CS_MODULE_PARAMETERS, add: H245_TUNNELING = YES
12. Create an H.323 service, and enter the CLAN IP address of the Avaya Communication Manager system as the primary gatekeeper. Confirm via a status signaling group that the RMX has registered.
13. Create a meeting room to use a test direct dial conference ID.

If you want to configure Ad-hoc conferencing for the RMX system, go to Configure Ad-hoc Video Conferencing for a Polycom RMX Video Conferencing Bridge Platform on page 6.

Configure Ad-hoc Video Conferencing for a Polycom RMX Video Conferencing Bridge Platform

To configure Ad-hoc conferencing for a Polycom RMX system, perform the following steps:

1. Perform the procedures in the section Configure a Polycom RMX Series Video Conferencing Bridge Platform on page 5 to configure the Polycom RMX system.
2. 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 RMX20 system and an RMX80 system, the maximum number of ports is 100.
3. Use the **change cos** command to set **Ad-hoc Video Conferencing** (page 2 of form) for the appropriate COS levels.
4. Use the **add video-bridge** command to configure a video bridge for the RMX system. Set the following parameters:
    - **Name** to the name for this video bridge (for example, *Ad Hoc Video Bridge - RMX*).
    - **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 ports. 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 (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:
  You must specify either a **Priority Factory Number** or a **Standard Factory Number.** You cannot leave both fields blank.

5. Create conference profiles for Ad-hoc (and Meeting Room) style conferences.

   NOTE:
   Be sure to choose **Auto Layout** under the video settings.

6. Create the two entry queues (one for Ad-hoc conferences, and one for priority conferences).
7. For the “Conference IVR Service,” perform the following steps:
   - Under the Welcome tab, disable the **Enable Welcome Messages** check box.
   - Under the Conference Chairperson tab, disable the **Chairperson messages** check box.
   - Under the Conference Password tab, disable the **Enable Password Messages** check box.
8. Under the General tab, perform the following steps:
   - Deselect any .wav file for the First to Join announcement.
   - Disable roll call.
   - Disable click and view.
9. Under the IVR Service tab, click the music icon, and set a silence .wav file as the IVR message. A silence .wav file will disable music from being played to the first party who joins the conference.

   To create a silence .wav file:

   a. Open the Windows Sound Recorder application.
   b. From the File menu, select **Save As**.
   c. In the Save As dialog box, click the **Change** button.
   d. In the Name box, enter ***silence.wav***.
   e. From the Format box, select **PCM**.
   f. From the Attributes box, select **16.00 kHz, 16 Bit, Mono**.
   g. Click the **OK** button.

Display Capacity for Ad-hoc Video Conferencing

To display the capacity for Ad-hoc video conferencing on the Avaya Communication Manager system:

1. Use the **display capacity** command to access the System Capacity form.
2. Go to page 7. Ad-hoc Video Conferencing Ports displays the following Ad-hoc video conferencing information:
   - **Used**: the number of video conferencing ports currently in use.
   - **Available**: the number of video conferencing ports currently available.
   - **System Limit**: the total number of video conference ports in your system. (This is the sum of Used ports and Available ports.)

View Video Conferencing Bridges

Use the **list video-bridge** command to view the video conferencing bridges administered on your system.

Configure a Polycom MGC-25 Video Conferencing Bridge Platform for an Avaya S8300 Server

To configure a Polycom MGC-25 system with an Avaya S8300 server, perform the following steps:

1. Use the **change ip-codec-set *x*** command (where ***x*** is the chosen IP codec set) to:
   - Define the wideband codecs (page 1 of form). The following wideband codecs are recommended:
     - SIREN14-48K (1 fpp, 20 ms)
       SIREN14-48K 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.
     - G722.1-32K (1 fpp, 20 ms)
       G722.1-32K 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, no 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.
   - Set **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 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 the codec set you defined in Step 2.
   - **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 for each link between IP network regions.

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

3. Use the **change node-names ip** command to add an entry for the MGC system. Be sure to enter the IP address of the IP board for the MGC system.

4. Use the **add trunk-group** command to add an outgoing trunk group for the MGC system. Set the following parameters:
   - **Group Type** to **isdn**.
   - **Direction** to **outgoing**.
   - **Carrier Medium** to **IP**.
   - **Service Type** to **tie**.
   - **Send Calling Number** to **y** (page 2 of form). This assists with resource allocation on the MGC when predefined participants are configured on the MGC system.
   - **Format** to **private**.

5. Use the **add signaling-group** command to add an outgoing signaling group for the MGC system. Set the following parameters:
   - **Group Type** to **h.323**.
   - **IP Video** to **y**.
   - **Trunk Group for Channel Selection** to ***blank***.
     NOTE:
     You must set this parameter to ***blank***.
   - **Near-end Node Name**
   - **Near-end Listen Port** to **1719**.
   - **LRQ Required** to **y**.
   - **Enable Layer 3 Test** to **n**.
   - **Far-end Node Name** to the name you entered for the MGC system in Step 3.
   - **Far-end Listen Port** to **1719**.
   - **Far-end Network Region**
   - **Calls Share IP Signaling Connection** to **n**.
   - **Direct IP-IP Audio Connections** to **y**.
   - **IP Audio Hairpinning** to **y**.
6. Use the **change trunk-group *xx*** command (where ***xx*** is the trunk group you added in Step 4) to add members to the trunk group.
7. Use the **add trunk-group** command to add an incoming trunk group for the MGC system. Set the following parameters:
   - **Group Type** to **isdn**.
   - **Direction** to **incoming**.
   - **Carrier Medium** to **IP**.
   - **Service Type** to **tie**.
8. Use the **add signaling-group** command to add an incoming signaling group for the MGC system. Set the following parameters:
   - **Group Type** to **h.323**.
   - **IP Video** to **y**.
   - **Priority Video.** If you want all incoming calls to receive priority video transmissions, select **y**.
   - **Trunk Group for Channel Selection** to the incoming trunk group you added in Step 6.
   - **Near-end Node Nam**e. For example, for an S8300 system, you would enter ***procr***. For an S8500 or S8700 system, you would enter the name of the CLAN board.
   - **Near-end Listen Port** to **1720**.
   - **ARQ Required** to **y**.
   - **RRQ Required** to **y**.
   - **Enable Layer 3 Test** to **n**.
   - **Far-end Node Name** to the name you entered for the MGC system in Step 3.
   - **Far-end Listen Port** to **1720**.
   - **Far-end Network Region**
   - **Calls Share IP Signaling Connection** to **n**.
   - **Direct IP-IP Audio Connections** to **y**.
   - **IP Audio Hairpinning** to **y**.
9. Use the **change trunk-group *xx*** command (where ***xx*** is the trunk group you added in Step 7) to add members to the trunk group.
10. Create a route pattern that points to the outgoing MGC trunk.
11. Install the Polycom system and connect it to your network.
12. Upgrade the Polycom system software.
13. Access the Polycom home page for the unit, and select **MCU Configuration>Network Services>IP> IP Default>Network Service Properties**.
14. Create a new H.323 service.
15. On the Settings tab, perform the following steps:
    - Set **Protocol** to **H323**.
    - Specify the **Subnet Mask**.
    - Specify the **Default Router**.
16. On the DNS Settings tab, specify the appropriate DNS servers in the **Use DNS Servers** box.
17. On the H323 tab, perform the following steps:
    - Make sure the **Forwarding** check box is not enabled.
    - In the **Use Gatekeeper** box, specify the gatekeeper.
    - In the **Preferred Gatekeeper IP Address** box, enter the CLAN IP address.
    - In the **Service Mode** box, enter **Pseudo Gatekeeper AVF**.
      NOTE:
      If **Pseudo Gatekeeper AVF** does not appear, you do not have the latest software. You must install the Avaya-enabled Polycom MGC Manager software.

- In the Prefix box, enter the first digit of the extension numbers range in which the MCU resides.
- Select the **Refresh H.323 Registrations** check box, and select **120 seconds**.

18. On the Spans tab, add a new IP Span.
    - In the **Circuit ID** box, enter a unique identifier for this new circuit.
    - In the **IP Address** box, enter the IP address of the IP card that will use H.323 signaling.
    - In the **H323 Alias 1** box, enter an H.323 ID of an alias for the board that Avaya Communication Manager will route calls to this MGC.
19. Define the IP1 configuration. In the **Circuit ID** box on the IP-Network Parameters tab, enter the unique identifier you specified in Step 16.
20. Create a meeting room in **Meeting Rooms and Entry Queues**. In the **Numeric ID** box on the General tab, enter the dialed digits excluding the prefix defined in the Network Service.

Configure Ad-hoc Video Conferencing for a Polycom MGC Video Conferencing Bridge Platform

To configure Ad-hoc video conferencing for a Polycom MGC video conferencing bridge platform, perform the following steps:

1. Configure the Polycom MGC information on Avaya Communication Manager. Depending on your Avaya server, see the appropriate section to configure your Polycom MGC system:
   - If you are using a Polycom MGC-25 with an Avaya S8300 server, go to Configure a Polycom MGC-25 Video Conferencing Bridge Platform for an Avaya S8300 Server on page 11.
   - If you are using a Polycom MGC with an Avaya S8500 or S87xx server, go to Configure Polycom MGC Video Conferencing Bridge Platforms with Avaya S8500 and S87xx Servers on page 16.

   NOTE:
   Keep in mind the following information when configuring the Polycom MGC:

   - Place the Polycom MGC in a dedicated network region with a dedicated codec set to infer the conference bit rates.
   - Ensure that you specify the correct network region (Far-end Network Region field) when you add the signaling groups for the MGC (Signaling Group form).
   - Ensure that you specify a codec set that reflects the correct conference bit rates for the MGC's network region (IP Codec Set form). Set the conference rate to 384 Kbps on page 2 of the IP Codec Set form.
   - Ensure that all other network regions have direct connectivity to the MGC's network region (**change ip-network-region** command).
2. Use the **display system-parameters customer-options** command to determine the **Maximum Administered Ad-hoc Video Conferencing Ports** (page 2 of form).
3. Use the **change cos** command to set **Ad-hoc Video Conferencing** (page 2 of form) for the appropriate COS levels.

4. Use the **add video-bridge** command to configure a video bridge for the MGC system. Set the following parameters:
   - **Name.**
   - **Max Ports** to the maximum number of Ad-hoc conferencing ports you want to assign to this bridge.
   - **ID Range** to the range of ports.
   - **Priority Factory Number**. If this field is left blank, priority video calls can use this video bridge. However, priority calls prefer to use a video bridge that has a priority factory number.
   - **Standard Factory Number**. If this field is left blank, non-priority endpoints cannot use this video bridge.
     NOTE:
     You must specify either a **Priority Factory Number** or a **Standard Factory Number.** You cannot leave both fields blank. The **Priority Factory Number** and the **Standard Factory Number** can be the same.
   - **Trunk Groups** to the administered incoming or outgoing ISDN H.323 or SIP trunk groups you added for the MGC. All entries must be of the same carrier type (that is, all H.323 trunks or all SIP trunks).
5. Configure the MGC system:
   a. **For MGC-25**: Create conference IDs 900, 901, 902, 903, and 904, with each being video switching 384 Kbps auto conferences.
      For MGC-50: Create conference IDs 900, 901, 902, 903, and 904, with each being video switching 384 Kbps auto conferences or continuous presence.
   b. Set 1*1 transcoding so cascaded conferences display properly.

Configure a Polycom MGC-25 Video Conferencing Bridge Platform for an Avaya S8300 Server

To configure a Polycom MGC-25 system with an Avaya S8300 server, perform the following steps:

1. Use the **change ip-codec-set *x*** command (where ***x*** is the chosen IP codec set) to:
   - Define the wideband codecs (page 1 of form). The following wideband codecs are recommended:
     - SIREN14-48K (1 fpp, 20 ms)
       SIREN14-48K 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.
     - G722.1-32K (1 fpp, 20 ms)
       G722.1-32K 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, no 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.
   - Set **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 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 the codec set you defined in Step 2.
   - **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 for each link between IP network regions.

   Repeat Step 2 for each IP network region that will be used for video in this system.
3. Use the **change node-names ip** command to add an entry for the MGC system. Be sure to enter the IP address of the IP board for the MGC system.
4. Use the **add trunk-group** command to add an outgoing trunk group for the MGC system. Set the following parameters:
   - **Group Type** to **isdn**.
   - **Direction** to **outgoing**.
   - **Carrier Medium** to **IP**.
   - **Service Type** to **tie**.
   - **Send Calling Number** to **y** (page 2 of form). This assists with resource allocation on the MGC when predefined participants are configured on the MGC system.
   - **Format** to **private**.
5. Use the **add signaling-group** command to add an outgoing signaling group for the MGC system. Set the following parameters:
   - **Group Type** to **h.323**.
   - **IP Video** to **y**.
   - **Trunk Group for Channel Selection** to ***blank***.
     NOTE:
     You must set this parameter to ***blank***.
   - **Near-end Node Name**
   - **Near-end Listen Port** to **1719**.
   - **LRQ Required** to **y**.
   - **Enable Layer 3 Test** to **n**.
   - **Far-end Node Name** to the name you entered for the MGC system in Step 3.
   - **Far-end Listen Port** to **1719**.
   - **Far-end Network Region**
   - **Calls Share IP Signaling Connection** to **n**.
   - **Direct IP-IP Audio Connections** to **y**.
   - **IP Audio Hairpinning** to **y**.
6. Use the **change trunk-group *xx*** command (where ***xx*** is the trunk group you added in Step 4) to add members to the trunk group.
7. Use the **add trunk-group** command to add an incoming trunk group for the MGC system. Set the following parameters:
   - **Group Type** to **isdn**.
   - **Direction** to **incoming**.
   - **Carrier Medium** to **IP**.
   - **Service Type** to **tie**.
8. Use the **add signaling-group** command to add an incoming signaling group for the MGC system. Set the following parameters:
   - **Group Type** to **h.323**.
   - **IP Video** to **y**.
   - **Priority Video.** If you want all incoming calls to receive priority video transmissions, select **y.**
   - **Trunk Group for Channel Selection** to the incoming trunk group you added in Step 6.
   - **Near-end Node Nam**e. For example, for an S8300 system, you would enter ***procr***. For an S8500 or S8700 system, you would enter the name of the CLAN board.

- **Near-end Listen Port** to **1720**.
- **ARQ Required** to **y**.
- **RRQ Required** to **y**.
- **Enable Layer 3 Test** to **n**.
- **Far-end Node Name** to the name you entered for the MGC system in Step 3.
- **Far-end Listen Port** to **1720**.
- **Far-end Network Region**
- **Calls Share IP Signaling Connection** to **n**.
- **Direct IP-IP Audio Connections** to **y**.
- **IP Audio Hairpinning** to **y**.

9. Use the **change trunk-group *xx*** command (where ***xx*** is the trunk group you added in Step 7) to add members to the trunk group.
10. Create a route pattern that points to the outgoing MGC trunk.
11. Install the Polycom system and connect it to your network.
12. Upgrade the Polycom system software.
13. Access the Polycom home page for the unit, and select **MCU Configuration>Network Services>IP> IP Default>Network Service Properties**.
14. Create a new H.323 service.
15. On the Settings tab, perform the following steps:
    - Set **Protocol** to **H323**.
    - Specify the **Subnet Mask**.
    - Specify the **Default Router**.
16. On the DNS Settings tab, specify the appropriate DNS servers in the **Use DNS Servers** box.
17. On the H323 tab, perform the following steps:
    - Make sure the **Forwarding** check box is not enabled.
    - In the **Use Gatekeeper** box, specify the gatekeeper.
    - In the **Preferred Gatekeeper IP Address** box, enter the CLAN IP address.
    - In the **Service Mode** box, enter **Pseudo Gatekeeper AVF**.
      NOTE:
      If **Pseudo Gatekeeper AVF** does not appear, you do not have the latest software. You must install the Avaya-enabled Polycom MGC Manager software.
    - In the Prefix box, enter the first digit of the extension numbers range in which the MCU resides.
    - Select the **Refresh H.323 Registrations** check box, and select **120 seconds**.
18. On the Spans tab, add a new IP Span.
    - In the **Circuit ID** box, enter a unique identifier for this new circuit.
    - In the **IP Address** box, enter the IP address of the IP card that will use H.323 signaling.
    - In the **H323 Alias 1** box, enter an H.323 ID of an alias for the board that Avaya Communication Manager will route calls to this MGC.
19. Define the IP1 configuration. In the **Circuit ID** box on the IP-Network Parameters tab, enter the unique identifier you specified in Step 16.
20. Create a meeting room in **Meeting Rooms and Entry Queues**. In the **Numeric ID** box on the General tab, enter the dialed digits excluding the prefix defined in the Network Service.

Configure Polycom PathNavigator Gatekeepers

Perform the following steps to configure a Polycom PathNavigator gatekeeper:

1. Use the **change ip-codec-set 1** command to set the following parameters:
   - **Allow Direct-IP Multimedia** to **y** (page 2 of form).
   - **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.
2. Use the **change ip-network-region** command to put the gatekeeper in its own network region. Set the following parameters:
   - **Intra-region IP-IP Direct Audio** to **no**.
   - **Inter-region IP-IP Direct Audio** to **no**.
   - **Security Procedures 1** to **any-auth** (page 2 of form).
   - **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 for each link between IP network regions.
3. Use the **change node-names ip** command to add an entry for the Polycom PathNavigator gatekeeper. Be sure to enter the IP address of the IP board for the gatekeeper.
4. Use the **add signaling-group** command to add a signaling group for the gatekeeper. Set the following parameters:
   - **Group Type** to **h.323**.
   - **IP Video** to **y**.
   - **Priority Video.** If you want all incoming calls to receive priority video transmissions, select **y.**
   - **Near-end Listen Port** to **1719**.
   - **LRQ Required** to **y**.
   - **Far-end Node Name** to the name you entered for the gatekeeper in Step 3.
   - **Far-end Listen Port** to **1719**.
   - **Far-end Network Region** to the IP network region you specified in Step 2.
   - **Direct IP-IP Audio Connections** to **y**.
   - **IP Audio Hairpinning** to **y**.
5. Use the **add trunk-group** command to add a trunk group for the gatekeeper. Set the following parameters:
   - **Group Type** to **isdn**.
   - **Carrier Medium** to **IP**.
   - Add members to this trunk group.
6. Use the **change signaling-group *xx*** command (where ***xx*** is the signaling group you added in Step 4) to set **Trunk Group for Channel Selection** to the trunk group you added in Step 5.
7. Create a route pattern to the gatekeeper.
8. Configure the gatekeeper.

Configure Video Trunks between two Avaya Communication Manager Systems

Perform the following steps to configure a video trunk between two systems running Avaya Communication Manager:

1. Use the **change ip-codec-set 1** command to set the following parameters:
   - **Allow Direct-IP Multimedia** to **y** (page 2 of form).
   - **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.
2. Use the **change node-names ip** command to add an entry for the trunk. Be sure to enter the IP address of the CLAN or PCLAN of the other Communication Manager system.
3. Use the **change ip-network-region *x*** command (where ***x*** is the chosen IP network region) to set the following parameters:
   - **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 for each link between IP network regions.
4. Use the **add signaling-group** command to add a signaling group for the video trunk. Set the following parameters:
   - **Group Type** to **h.323**.
   - **IP Video** to **y**.
   - **Near-end Listen Port**
   - **LRQ Required** to **n**.
   - **Far-end Node Name**
   - **Far-end Listen Port**
   - **Far-end Network Region.** If you set the maximum bandwidth for video calls in Step 3, assign that IP network region.
   - **Calls Share IP Signaling Connection** to ***n***.
     NOTE:
     You must set this parameter to ***n***.
   - **Direct IP-IP Audio Connections** to **y**.
   - **IP Audio Hairpinning** to **y**.
5. Use the **add trunk-group** command to add a trunk group for the video trunk. Set the following parameters:
   - **Group Type** to **isdn**.
   - For Avaya Communication Manager Release 3.01, set **Carrier Medium** to **IP**.
   - For Avaya Communication Manager Release 3.1 or later, set **Carrier Medium** to **H.323**.
   - Add members to this trunk group.
6. Use the **change signaling-group *xx*** command (where ***xx*** is the signaling group you added in Step 3) to set **Trunk Group for Channel Selection** to the trunk group you added in Step 4.
7. Create a route pattern for the trunk group.

Configure Polycom MGC Video Conferencing Bridge Platforms with Avaya S8500 and S87xx Servers

This section provides a set of guidelines to configure Polycom MGC video conferencing bridge platforms for use with Avaya S8500 servers and Avaya S87xx servers. You must understand these guidelines before you administer these systems.

Managing Bandwidth for Video

You can use IP network regions to manage the amount of bandwidth used by video. Using the **change ip-network-region *x*** command (where ***x*** is the chosen IP network region), you can specify the maximum amount of bandwidth that can be used for video to each IP network region.

On page 3 of the ip-network-region form, set:

- **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 for each link between IP network regions.

NOTE:
After setting the bandwidth pools to each IP network region, you must assign this IP network region to the appropriate signaling group. Then, you will assign this signaling group to the appropriate trunk groups.

You can monitor the status of video bandwidth usage using the **status ip-network-region *x*** command (where ***x*** is the chosen IP network region). The current video bandwidth usage to each network region is displayed.

Trunk Groups

For incoming trunk groups, you must:

- Create one incoming trunk group per Polycom MGC system.
- Set the following parameters:
  - **Direction** to **incoming**.
  - **Service Type** to **tie**.
  - **Disconnect Supervision - In?** to **y** to allow Polycom MGC-Avaya Communication Manager-Polycom MGC calls (that is, trunks calling trunks).

For outgoing trunk groups, you must:

- Create one “primary” outgoing trunk group for the “primary” Polycom MGC board (IP1).
- Set the following parameters:
  - **Direction** to **outgoing**.
  - **Service Type** to **tie**.
  - You may set **Outgoing Display** to **y**.
  - **Send Calling Number** to **y**. This allows participant matching in the Polycom MGC.
  - **Format** to **private**.
- If you want board redundancy, create one “secondary” outgoing trunk group for the other boards (IP2.n) in the Polycom MGC system.

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Signaling Groups

For incoming signaling groups, you must:

- Create one incoming signaling group (RRQ signaling group) per MGC board in the Polycom MGC system for a primary CLAN. This incoming signaling group allows an Polycom MGC board to register and is used for incoming calls. The “primary” CLAN is the CLAN you want to use with the MGC. The IP address of this CLAN is administered on the MGC via the Polycom MGC Manager software.

  Keep in mind the following information:

  - For Avaya Communication Manager Release 3.0.1, one signaling group can support a maximum of 31 calls.
  - For Avaya Communication Manager Release 3.1 or later, one signaling group can support a maximum of 255 calls.
- Set the following parameters for the incoming signaling groups:
  - **IP Video?** to **y**.
  - **Incoming Priority Video.** If you want all incoming calls to receive priority video transmissions, select **y**.
  - **Trunk Group for Channel Selection**.
  - **Near-end Listen Port** to **1720**.
  - **Far-end Listen Port** to **1720**.
  - **ARQ Required?** to **y**.
  - **RRQ Required?** to **y**.
  - **Enable Layer 3 Test** to **n**.
  - **Direct IP-IP Audio Connections to**? to **y**.
  - **Far-end Network Region.** If you set the maximum bandwidth for video calls in an IP network region, assign the appropriate IP network region.
- Optional: Create one incoming signaling group per MGC board in the Polycom MGC system for a secondary CLAN. If the primary CLAN fails, the MGC will use the alternate gatekeeper list as a list of alternate CLANs with which to register.

For outgoing signaling groups, you must:

- Create one outgoing signaling group (LRQ signaling group) per board in the Polycom MGC system for the primary CLAN. This outgoing signaling group is used for outgoing LRQ calls to the co-resident gatekeeper in the Polycom MGC system.

  Keep in mind the following information:

  - For Avaya Communication Manager Release 3.0.1, one signaling group can support a maximum of 31 calls.
  - For Avaya Communication Manager Release 3.1 or later, one signaling group can support a maximum of 255 calls.
- Set the following parameters for the outgoing signaling groups:
  - **IP Video?** to **y**.
  - **Near-end Listen Port** to **1719**.
  - **Far-end Listen Port** to **1719**.
  - **LRQ Required?** to **y**.
  - **Enable Layer 3 Test** to **n**.
  - **Direct IP-IP Audio Connections to?** to **y**.
  - **Far-end Network Region.** If you set the maximum bandwidth for video calls in an IP network region, assign the appropriate IP network region.

  NOTE:
  Do not set Trunk Group for Channel Selection. If you set this parameter, all incoming video calls may fail or video may close during audio shuffle.
- Optional: Create one outgoing signaling group per board in the Polycom MGC system for the secondary CLAN.
- Optional: Create additional outgoing signaling groups per board in the Polycom MGC system per CLAN for additional CLAN load sharing and additional call capacity.

Group Member Assignments

Keep in mind the following information:

- Group member assignments for the incoming trunk may consist of all incoming signaling groups for the Polycom MGC system.
- Group member assignments for the primary outgoing trunk may consist of all signaling groups defined for multiple CLANs to the one primary board in the Polycom MGC system. **Do not mix Polycom MGC boards**.
- Group member assignments for the secondary outgoing trunk may consist of all signaling groups defined for multiple CLANs to the other boards in the Polycom MGC system.

Outgoing Rules

Keep in mind the following information:

- Change dial plan analysis and uniform dial plan as required to support the Polycom MGC-assigned extension range.
- Add AAR analysis for dial plan extension range for use by the Polycom MGC system. Use **lev0** as the type to force use of the private numbering plan.
- On the route pattern form, add an entry for each outgoing trunk group, and set **LAR** to next. Only one route pattern should be required per Polycom MGC system.
- Regarding digit manipulation, present the digits to the Polycom MGC system in the form of “prefix” plus digits. For example, suppose the Polycom MGC system is administered with the prefix **7**, and there is a five-digit dial plan with **715xx** routing to the Polycom MGC system. You should present the digits **715xx** instead of **15xx**, where **15xx** is the meeting rooms. The Polycom MGC system allows support of multiple prefixes.

Examples

Example 1: S87xx with 2 CLANs/MGC-50 with 2 boards, board redundancy, and no CLAN redundancy

In this example, CLAN1 is chosen as the CLAN for use with the Polycom MGC system. CLAN2 is not used. If CLAN1 fails, the Polycom MGC system will not be allowed to register to CLAN2 and will no longer be able to make incoming or outgoing calls. If the Polycom MGC board IP1 fails, IP2 will continue to be able to make incoming and outgoing calls.

Trunk 1: MGC-A Incoming:

SigGroup 1: CLAN1, IP1; 31 trunk members
SigGroup 2: CLAN1, IP2; 31 trunk members

Trunk 2: MGC-A Outgoing Primary:

SigGroup 3: CLAN1, IP1; 31 trunk members

Trunk 3: MGC-A Outgoing Secondary:

SigGroup 4: CLAN1, IP2; 31 trunk members

Example 2: S87xx with 2 CLANs/MGC-50 with 2 boards, board redundancy, and CLAN redundancy

In this example, CLAN1 is chosen as the primary CLAN for the Polycom MGC system to use, and CLAN2 is a backup. CLAN2 provides backup to CLAN1 for Polycom MGC registration and incoming and outgoing calls. If Polycom MGC board IP1 fails, IP2 will continue to be able to make incoming and outgoing calls.

Note that the incoming signaling groups for CLAN2 will be out-of-service unless the Polycom MGC boards are forced to register with CLAN2 in the event that CLAN1 is unavailable.

Trunk 1: MGC-A Incoming:

SigGroup 1: CLAN1, IP1; 31 trunk members
SigGroup 2: CLAN1, IP2; 31 trunk members
SigGroup 3: CLAN2, IP1; 31 trunk members
SigGroup 4: CLAN2, IP2; 31 trunk members

Trunk 2: MGC-A Outgoing Primary:

SigGroup 5: CLAN1, IP1; 31 trunk members
SigGroup 6: CLAN2, IP1; 31 trunk members

Trunk 3: MGC-A Outgoing Secondary:

SigGroup 7: CLAN1, IP2; 31 trunk members
SigGroup 8: CLAN2, IP2; 31 trunk members

Example 3: S87xx with 2 CLANs/MGC-50 with 3 boards, board redundancy, and CLAN redundancy

In this example, CLAN1 is chosen as the primary CLAN for the Polycom MGC system to use, and CLAN2 is a backup. CLAN2 provides backup to CLAN1 for Polycom MGC registration and incoming and outgoing calls. If Polycom MGC board IP1 fails, IP2 and IP3 will continue to be able to make incoming and outgoing calls.

Note that the incoming signaling groups for CLAN2 will be out-of-service unless the Polycom MGC boards are forced to register with CLAN2 in the event that CLAN1 is unavailable.

Trunk 1: MGC-A Incoming:

SigGroup 1: CLAN1, IP1; 31 trunk members
SigGroup 2: CLAN1, IP2; 31 trunk members
SigGroup 3: CLAN1, IP3; 31 trunk members
SigGroup 4: CLAN2, IP1; 31 trunk members
SigGroup 5: CLAN2, IP2; 31 trunk members
SigGroup 6: CLAN2, IP3; 31 trunk members

Trunk 2: MGC-A Outgoing Primary:

SigGroup 7: CLAN1, IP1; 31 trunk members
SigGroup 8: CLAN2, IP1; 31 trunk members

Trunk 3: MGC-A Outgoing Secondary:

SigGroup 9: CLAN1, IP2; 31 trunk members
SigGroup 10: CLAN2, IP2; 31 trunk members
SigGroup 11: CLAN1, IP3; 31 trunk members
SigGroup 12: CLAN2, IP3; 31 trunk members

Example 4: S87xx with 4 CLANs/MGC-50 with 3 boards, board redundancy, and outgoing CLAN redundancy

In this example, CLAN1 is chosen as the primary CLAN for the Polycom MGC system to use, and CLAN2 is a backup. CLAN2 provides backup to CLAN1 for Polycom MGC registration and incoming and outgoing calls. If Polycom MGC board IP1 fails, IP2 and IP3 will continue to be able to make incoming and outgoing calls.

Note that the incoming signaling groups for CLAN2 will be out-of-service unless the Polycom MGC boards are forced to register with CLAN2 in the event that CLAN1 is unavailable.

Note that CLAN3 and CLAN4 are not used. Additional outgoing call capacity could be added by defining additional outgoing signaling groups between CLAN3 to IP1, IP2, and IP3, and CLAN4 to IP1, IP2, and IP3. Additional incoming call capacity could be added by defining additional incoming signaling groups between CLAN3 and CLAN4 to boards IP1, IP2, and IP3.

Trunk 1: MGC-A Incoming:

SigGroup 1: CLAN1, IP1; 31 trunk members
SigGroup 2: CLAN1, IP2; 31 trunk members
SigGroup 3: CLAN1, IP3; 31 trunk members
SigGroup 4: CLAN2, IP1; 31 trunk members
SigGroup 5: CLAN2, IP2; 31 trunk members
SigGroup 6: CLAN2, IP3; 31 trunk members

Trunk 2: MGC-A Outgoing Primary:

SigGroup 7: CLAN1, IP1; 31 trunk members
SigGroup 8: CLAN2, IP1; 31 trunk members

Trunk 3: MGC-A Outgoing Secondary:

SigGroup 9: CLAN1, IP2; 31 trunk members
SigGroup 10: CLAN2, IP2; 31 trunk members
SigGroup 11: CLAN1, IP3; 31 trunk members
SigGroup 12: CLAN2, IP3; 31 trunk members

Monitor the Status of Video Bandwidth Usage

This procedure describes how to monitor the status of video bandwidth usage for inter-network regions.

Use the **status ip-network-region *x*** command (where ***x*** is the chosen IP network region) to view the video bandwidth usage. The current video bandwidth usage to each network region is displayed.

Use the **change ip-network-region *x*** command (where ***x*** is the chosen IP network region) to specify the maximum amount of bandwidth that can be used for video to each IP network region.