SOLUTION GUIDE

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# Polycom® Unified Communications Deployment Guide for Avaya Aura® Environments

## Polycom® Unified Communications Deployment Guide for Avaya Aura® Environments



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# Contents

# About This Guide

This partner solution guide uses a number of conventions that help you to understand information and perform tasks.

## Conventions Used in this Guide

This user guide contains terms, graphical elements, and a few typographic conventions. Familiarizing yourself with these terms, elements, and conventions will help you perform phone tasks.

### Information Elements

The following icons are used to alert you to various types of important information in this guide:

**Icons Used in this Guide**

| *Name* | *Icon* | *Description* |
|---|---|---|
| Note |  | The Note icon highlights information of interest or important information needed to be successful in accomplishing a procedure or to understand a concept. |
| Administrator Tip |  | The Administrator Tip icon highlights techniques, shortcuts, or productivity related tips. |
| Caution |  | The Caution icon highlights information you need to know to avoid a hazard that could potentially impact device performance, application functionality, or successful feature configuration. |
| Warning |  | The Warning icon highlights an action you must perform (or avoid) to prevent issues that may cause you to lose information or your configuration setup, and/or affect phone or network performance. |
| Web Info |  | The Web Info icon highlights supplementary information available online such as documents or downloads on support.polycom.com or other locations. |
| Timesaver |  | The Timesaver icon highlights a faster or alternative method for accomplishing a method or operation. |

| *Name* | *Icon* | *Description* |
|---|---|---|
| Power Tip | | The Power Tip icon highlights faster, alternative procedures for advanced administrators already familiar with the techniques being discussed. |
| Troubleshooting | | The Troubleshooting icon highlights information that may help you solve a relevant problem or to refer you to other relevant troubleshooting resources. |
| Settings | | The Settings icon highlights settings you may need to choose for a specific behavior, to enable a specific feature, or to access customization options. |

## Typographic Conventions

A few typographic conventions, listed next, are used in this guide to distinguish types of in-text information.

**Typographic Conventions**

| *Convention* | *Description* |
|---|---|
| **Bold** | Highlights interface items such as menus, soft keys, file names, and directories. Also used to represent menu selections and text entry to the phone. |
| *Italics* | Used to emphasize text, to show example values or inputs, and to show titles of reference documents available from the Polycom Support Web site and other reference sites. |
| Underlined Blue | Used for URL links to external Web pages or documents. If you click on text in this style, you will be linked to an external document or Web page. |
| Blue Text | Used for cross references to other sections within this document. If you click on text in this style, you will be taken to another part of this document. |
| `Fixed-width-font` | Used for code fragments and parameter names. |

# What's in This Guide?

This partner solution guide is organized into 9 chapters. The first chapter Getting Started lets you know what knowledge, hardware, and software you require before you begin and provides frequently asked questions (FAQs) and resources for further help. Chapters two and three introduce Polycom in an Avaya Aura environment, and Avaya Aura itself. The subsequent chapters show you how to configure and deploy specific Polycom products and systems with Avaya Aura components. The Troubleshooting section shows you solutions to common troubleshooting problems with this solution and the Getting Help section shows you where to obtain further information and how to access the Polycom community.

- **Chapter 1: Getting Started** This chapter contains introductory information about the flexibility of Polycom and Avaya's Unified Communication solutions.
- **Chapter 2: Getting Started with Avaya Aura** This chapter describes the four basic components of Avaya Aura: Avaya Session Manager Instance, Avaya Communication Manager Feature Server, Avaya Communication Manager Evolution Server, and Avaya System Manager.
- **Chapter 3: Testing and Scenarios** This chapter describes the products that have been tested for interoperability with Avaya Communications Manager and Avaya Aura, and provides information about testing scenarios.
- **Chapter 4: Deployment Models and Use Cases** Here, you will find information about testing, certification, and supported versions for various solutions and use cases.
- **Chapter 5: Configuring Avaya Aura Session Manager** This chapter explains how to configure the Avaya Aura Session manager, log on to the System Manager Web Interface, and manage Communications Manager Objects.
- **Chapter 6: Configuring Polycom Systems to Interoperate with Avaya Communication Manager** This chapter provides systems configuration information for HDX, RMX, and Polycom Telepresence m100.
- **Chapter 7: Troubleshooting** This chapter lists some of the common problems you may encounter, and possible solutions to them.
- **Chapter 8: Getting Help** In this chapter, you will find links to support documents and websites from Polycom, Polycom partners, and others. You will also find links to the Polycom Community, which contains a number of discussion forums you can use to share ideas with your colleagues.

# Chapter 1: Getting Started

Avaya Aura® is a suite of SIP-based central communications platform that supports voice, video, messaging, and presence solutions for midsize to large enterprises. You can use Avaya Aura to simplify complex networks and enable members of an organization at any location to interconnect instantly in real time and on any type of infrastructure the endpoints reside on.

This solution guide shows you how to integrate Polycom products in an Avaya Aura environment.

## Before You Begin

Before you set up and install the Polycom and Avaya components in the Avaya Aura environment, you should have these prerequisites:

- Previous knowledge of, and experience with, the Avaya components
- Access to Avaya product documentation and relevant software
- Previous knowledge of, and experience with, Polycom products
- Access to the Polycom product documentation and relevant software

## Frequently Asked Questions

Refer to these frequently asked questions (FAQs) to help answer questions you may have about the solution before you begin.

Q: **What network protocols will the solution work with?**
A: You can use the solution shown in this guide with the SIP or H.323 protocol.

Q: **What network configurations will the solution work with?**
A: You can set up two network scenarios to interoperate Avaya and Polycom devices: a direct registration to Avaya or a neighbored/trunked scenario. All of the Polycom video and audio components used in this guide can register directly to the Avaya gatekeeper or SIP registrar. Alternatively, you can neighbor or trunk Avaya Aura to the Polycom Distributed Media Application (DMA), which you can use to manage the Polycom network.

Q: **What Polycom hardware components work with this solution?**
A: This solution enables you to use the Polycom HDX family, the Polycom RMX family, soft clients, and mobility components. We also have interoperability with the Immersive Telepresence Platforms and the Video Border Proxy E-series. Not all of the products in

the Polycom portfolio interoperate directly with the Avaya call control platforms, and may not be referenced in this document.

Q: **What Avaya hardware components will the solution work with?**

A: The hardware components you can use with this solution depend on the version of Avaya you use. If you want to integrate Polycom products with Avaya's Communication Manager (CM) 5.2.1, the CM is the minimum requirement as it incorporates an H.323 gatekeeper. If you want to integrate Polycom products with Avaya Aura 6, you require a Communication Manager, a Session Manager, and a System Manager. You can use each of these as separate servers or you can incorporate them all into a single midsize enterprise server. There is also a movement towards virtualization of these platforms, although this is not yet fully supported.

# What's New?

For an updated list of Polycom products that have been tested for use with Avaya Communications Manager and Avaya Aura as part of this solution, see Table 3-1: Polycom Interoperability with Avaya.

# Required Solution Hardware

The solutions described in this guide cover a range of products. As such, there is no single list of required hardware; however, these solutions do apply to specific products within each family.

### Polycom

The Polycom systems covered in this book include the following products:

- Polycom® Converged Management Application™ (CMA®)
- Polycom® Converged Management Application™ (CMA®) Desktop for Windows®
- Polycom® Distributed Media Application™ (DMA™) 7000
- Polycom® HDX® systems
- Polycom® Immersive Telepresence (ITP)
- Polycom® RMX® 1500/2000/4000
- Polycom® RSS™ 4000
- Polycom® Telepresence m100
- Polycom® Video Border Proxy™ (VBP®) E-Series
- Polycom® VVX® 1500

### Avaya

The following Avaya Aura® products interoperate with Polycom products for this solution:

- Avaya Aura® Session Manager
- Avaya Aura® Communications Manager, as a Feature Server
- Avaya Aura® Communications Manager, as an Evolution Server
- Avaya Aura® System Manager
- Avaya Aura® one-x® Communicator
- Avaya 1000 Series Video Endpoints
- Avaya Desktop Video Device (ADVD)

## Hardware and Software Dependencies

The following table lists the supported Polycom and Avaya versions or releases that have been tested and verified in a lab environment for this solution.

**Table 1-1: Tested Versions of Polycom and Avaya Products with Communication Manager 5.2.1 – H.323 Only**

| *Server/Endpoint Type* | *Wave 5 October 2011* | *Wave 6 August 2012* | *Wave 7 November 2012* |
|---|---|---|---|
| Avaya Communication Manager | 5.2.1 | 5.02.1.016.4 Patch 18111 | 5.02.1.016.4 Patch 18111 |
| Avaya one-X Communicator | 6.0.01 | 6.1.4.02-SP4-36064 | 6.1.5.07- SP5-37495 |
| DMA 7000 | 3.0 & 4.0 | 5 | 5.1 |
| RMX1500/2000/4000 | 7.03 & 7.2 & 7.6 | 7.7 | 7.8.0 |
| HDX 6/7/8/9000 | 2.6.1 & 3.0.3 | 3.0.5 | 3.1.0 |
| ITP (RPX, APX, OTX) | 3.0.3 | 3.0.5 | 3.1.0 |
| VBP-e | 9.1.5.3 & 11.2.3 | 11.2.10 | 11.2.13RC2 |
| Avaya 96xx Phone | NA | S3.171b | S3.171b |

**Table 1-2: Tested Versions of Polycom and Avaya products with Avaya Aura - SIP / H.323**

**Communication Manager / Session Manager v 6.x**

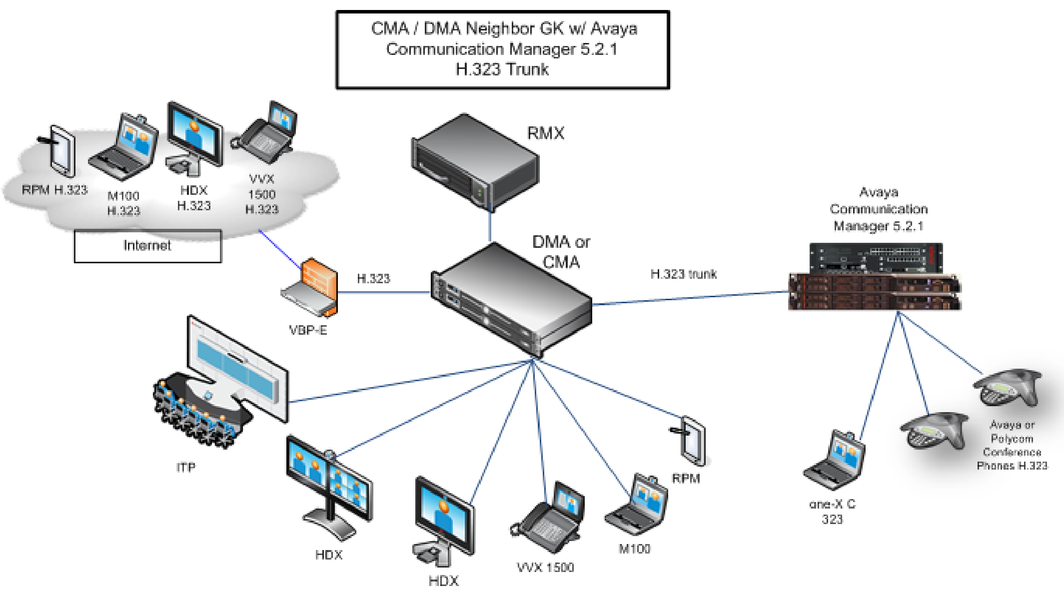
| *Server/Endpoint Type* | *Wave 5*<br>*October 2011* | *Wave 6*<br>*August 2012* | *Wave 7*<br>*November 2012* |
|---|---|---|---|
| Avaya Communication Manager | 6.0.1 | R016x.02.0.823.0 Patch 19761 | R016x.02.0.823.0 Patch 19926 |
| Avaya Session Manager | 6.1.1 | 6.2.0.0.620120 | 6.2.0.0.620110 |
| Avaya one-X Communicator | NA | 6.1.5.04-SP5-37080 | 6.1.5.07-SP5-37495 |
| Avaya 10XX | NA | 4.8.3 -23 | 4.8.3(24) |
| Avaya ADVD Flare | NA | 1.1.1- 019004 | 1.1.1- 019004 |
| DMA 7000 | 3.0 and 4.0 | 5.0.0.ER33 | 5.1 |
| RMX Family | 7.03, 7.2, 7.6 | 7.7.0.222 | 7.8 |
| HDX Family | 2.6.1, 3.0.3 | 3.0.5-22546 | 3.1.0 |
| ITP (RPX, OTX) | 3.0.3 | 3.0.5-22546 | 3.1.0 |
| VBP-e | 9.1.5.3 and 11.2.3 | 11.2.10 | 11.2.13RC2 |
| XMA | NA | NA | 7.1 |
| VVX | NA | 4.0.1 | 4.0.2.11307 |

# Overview of the Solution

There are several ways to interoperate with Avaya systems. Here we have five representative examples: Direct Registration using H.323 and SIP, Neighbored Gatekeeper (with CM 5.2.1 and AURA), and using a SIP proxy.

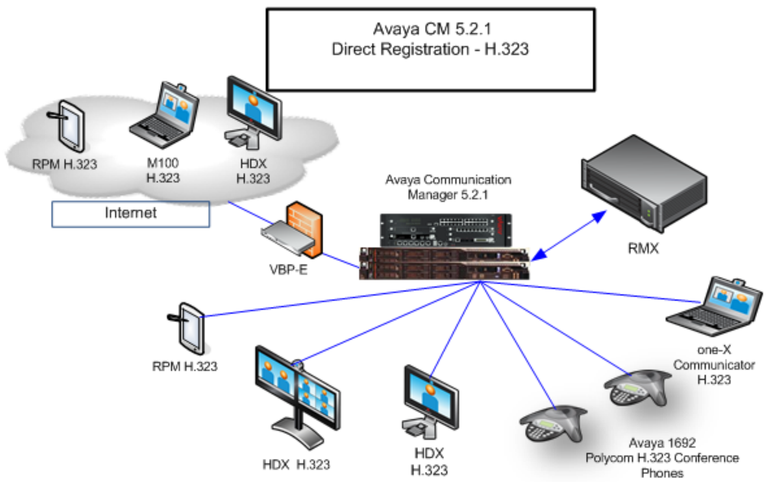
The following diagram shows Polycom operating in an Avaya Communication Manager H.323 environment, in a neighbored gatekeeper scenario. Devices registered to Avaya CM can connect to any devices registered to Polycom (CMA or DMA GK), into a multipoint conference, or dial out to an external system. Likewise, any Polycom-registered device can connect to any Avaya-registered device.

**Figure 1-1: H.323 Neighbored Gatekeeper Scenario**



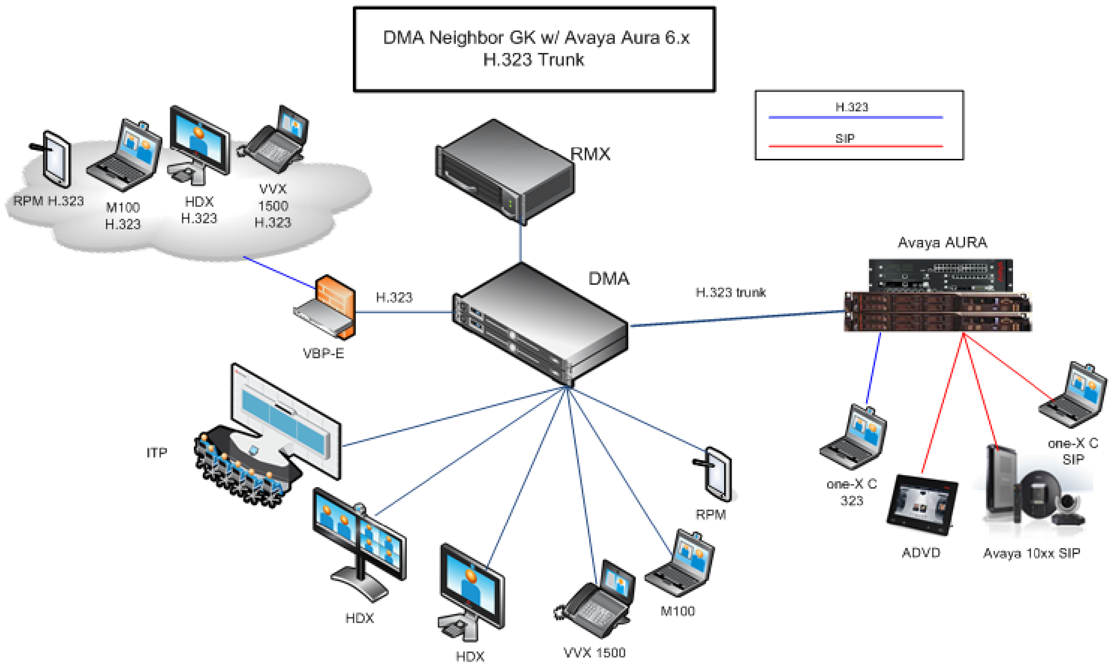
The following diagram shows Polycom operating in an Avaya Communication Manager H.323 environment, in a direct registration scenario. Devices registered to Avaya CM can connect to any devices and into a multipoint conference, or dial out to an external system.

**Figure 1-2: H.323 Direct Registration Scenario**



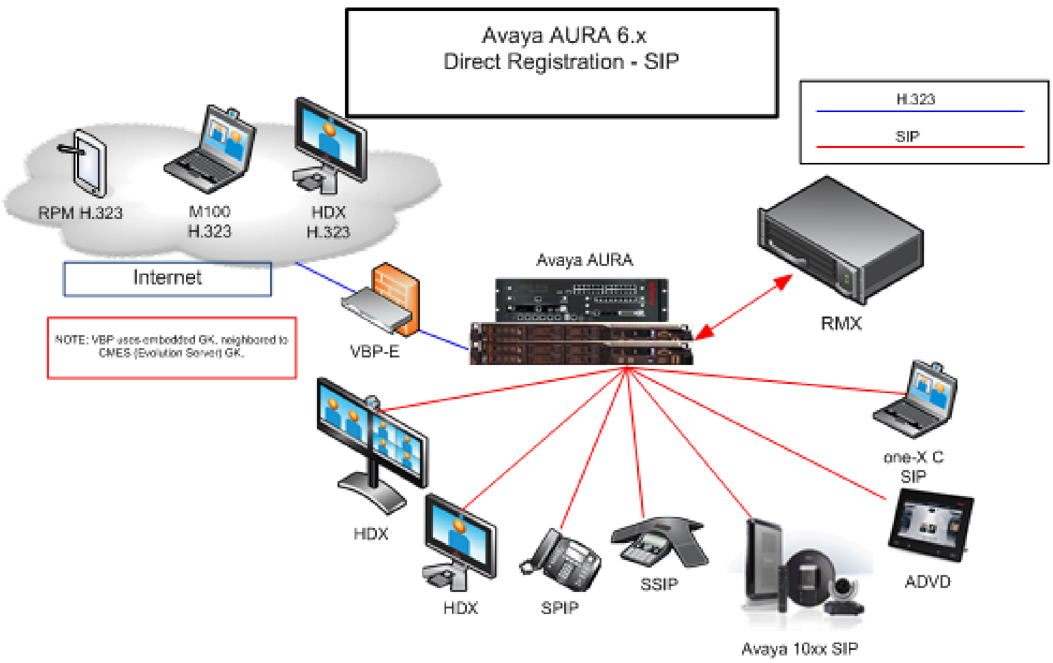
The following diagram shows Polycom operating in an Avaya AURA environment, in a Neighbor scenario. Devices registered to Avaya CM can connect to any devices and into a multipoint conference, or dial out to an external system.

**Figure 1-3: Avaya Aura Neighbor Scenario**



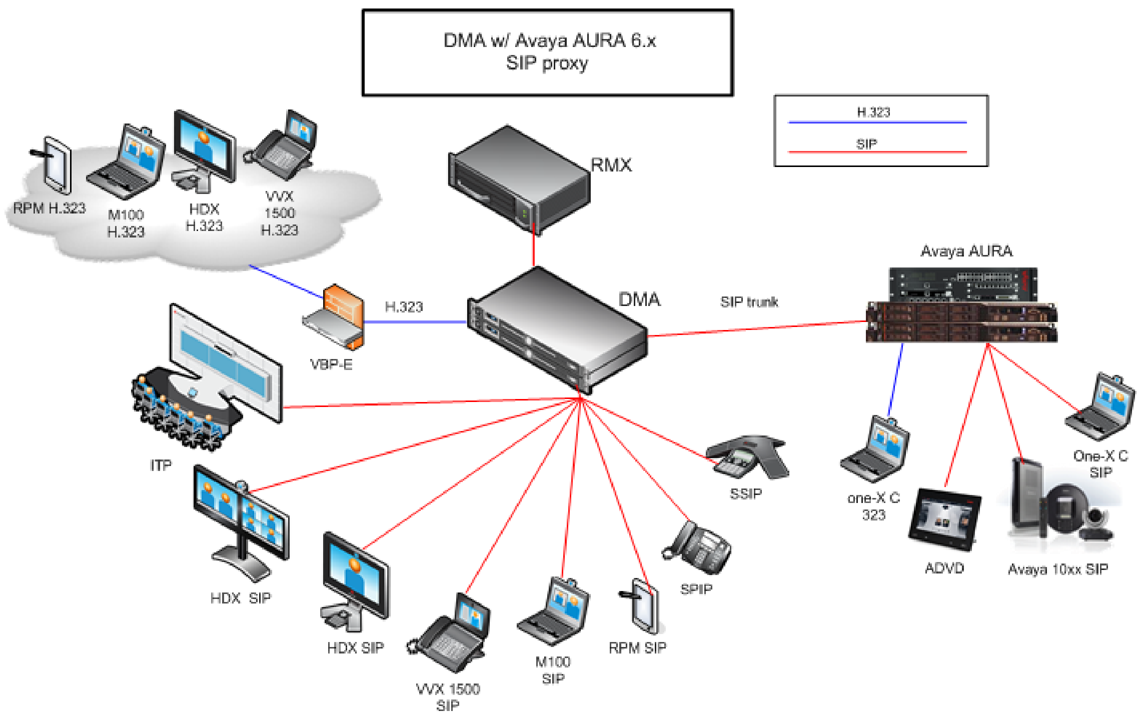
The following diagram shows Polycom and Avaya devices registering to the SIP registrar in Session Manager. If the Communication Manager is operating as a SIP/H.323 gateway, it can also interoperate with the Video Border Proxy, which can neighbor its embedded gatekeeper to the gatekeeper in the Communication Manager (Evolution Server).

**Figure 1-4: Avaya Aura Direct Scenario**



In the drawing below, you see a representation of the Polycom DMA interoperating with the Avaya Session Manager, working as a SIP proxy. Assuming the dial plans are built correctly, any-to-any dialing is possible. In this environment, the DMA is operating as a SIP/ H.323 gateway, able to pass H.323 calls from outside of the Video Border Proxy.

**Figure 1-5: Avaya AURA Proxy Scenario**




# Getting Help and Support Resources

This partner solution guide includes a Getting Help section you can use to find links to Polycom product sites, support sites, and partner resources. You can also find information about The Polycom Community, which provides access to a blog site you can use to discuss hardware, software, and partner solution topics with your colleagues. To register with the Polycom Community, you will need to create a Polycom online account.

The Polycom Community includes access to Polycom support personnel, as well as user-generated hardware, software, and partner solutions topics. You can view top blog posts and participate in threads on any number of recent topics.

# Chapter 2: Getting Started with Avaya Aura®

This chapter describes the Avaya Aura® hardware components for use with this solution and their respective operating environments. To interoperate successfully, diverse communications products require a standard signaling protocol such as Session Initiation Protocol (SIP) or H.323. Avaya Aura Session Manager is a SIP-only routing and integration tool. The Avaya Communication Manager up to v 5.2.1 was H.323 only, and in Aura (above version 6) it could also operate as a SIP/H.323 gateway.

The following Avaya Aura® products interoperate with Polycom products for this solution:

- Understanding Avaya Aura Session Manager
- Using the Avaya System Manager
- Using Avaya Communication Manager as a Feature Server
- Using Avaya Communication Manager as an Evolution Server
- Avaya Aura one-x® Communicator
- Avaya 1000 Series Video Endpoints

## Understanding Avaya Aura Session Manager

Avaya Session Manager integrates all of your organization's SIP devices across an enterprise network, enabling you to manage individual locations, branches, and applications as a single enterprise. The Session Manager provides the following benefits:

- A simplified, network-wide feature deployment
- Centralized routing, SIP trunking, and user profiles
- Cost-effective scalability from small to very large deployments
- High availability with geographic redundancy
- A secure environment that conforms to specific SIP standards and practices

The Avaya Session Manager provides a core communication service that you can use to build on existing equipment and add a SIP-based architecture. Use the Session Manager to connect to any of the following network components:

- Avaya Aura Communication Manager as a SIP-only Feature Server or as a SIP and non-SIP Evolution Server
- Avaya enterprise PBX and small-key PBX systems within branch offices
- Third-party PBXs

- Gateways
- Session Border Controllers (SBCs)
- SIP-enabled adjuncts
- SIP and non-SIP phones

# Using the Avaya System Manager

The Avaya System Manager Common Console is the management interface for Session Manager. To perform any administration or configuration tasks, you must log on to the System Manager Common Console.

Avaya Aura System Manager manages Avaya Aura Session Manager. Polycom HDX systems video endpoints configured as SIP endpoints use the Avaya Aura Session Manager User Registration feature, and require that Avaya Aura Communication Manager operate as a Feature Server.

The Avaya Communication Manager Feature Server only supports IP Multimedia Subsystem (IMS)-SIP users that are registered to Avaya Aura Session Manager. The Communication Manager Feature Server is connected to Session Manager through an IMS enabled SIP signaling group and associated SIP trunk group.

. System Manager includes the following shared management services:

**Table 2-1: Shared Management Services in System Manager**

| ***Service*** | ***Description*** |
|---|---|
| Elements | Provides features for managing individual components of System Manager including Session Manager element administration. |
| Events | Provides features for administering alarms and logs that System Manager and some of its components generate. You can view and change the status of alarms. For logs, you can view logs, harvest logs for System Manager and its components, and manage loggers and appenders. |
| Licenses | Provides features for administering licenses for individual components of the Avaya Aura solution. |
| Routing | Provides features to manage routing applications. You can create and manage routing applications that include domains, adaptations, SIP entities, entity links, time ranges, policies, dial patterns, and regular expressions to configure your network configuration. |
| Security | Provides features for configuring certificates. |

| Service | Description |
| --- | --- |
| System Data | Provides features for these tasks:<br>• Backing up and restoring System Manager configuration data.<br>• Monitoring and scheduling jobs.<br>• Replicating data from remote nodes.<br>• Configuring data retention settings and profiles for various services provided by System Manager. |
| User Management | Provides a central user administration of all user properties. |

# Understanding Avaya Session Manager

A Session Manager instance consists of one server that supports up to 50,000 SIP entities on a SIP enterprise network. An enterprise network can support up to six Session Manager instances. The benefit of using a Session Manager instance is that you can run the Session Manager on a separate server – rather than requiring a separate server for each instance, you need only one server for the entire network.

System Manager and Session Manager work together as follows:

- Session Manager is the software component for all enterprise SIP sessions.
- System Manager is the single, centralized management point of control.

Administrators can install Session Manager instances in the same data center or in multiple data centers, and in geographically redundant locations with virtually unlimited distance restrictions.

Session Manager supports the following SIP entities:

- Avaya Aura Session Manager
- Avaya Aura System Manager private branch exchanges
- Avaya Aura Communication Manager Feature Server
- Public Switched Telephone Network (PSTN) service providers
- SIP-enabled adjuncts that work with PBXs to provide services
- Desksets

Session Manager supports the following audio and video endpoints:

- Polycom HDX series video endpoints
- Avaya (OEM) Model 1692 conference phones

The following table describes SIP entities supported by the Avaya Session Manager.

**Table 2-2: Supported SIP Entities**

| *SIP Entity* | *Description* |
|---|---|
| SIP Gateways | SIP gateways work with the non-SIP service provider network. These include non-Avaya SIP systems, SIP-enabled PBXs, and Avaya Communication Manager. Such SIP gateways are supported both for trunking services or line-side services. SIP gateways include trunk gateways, such as the Avaya G860 Media Gateway. |
| SIP PSTN Service Providers | Service providers operate as SIP peer network elements with which Session Manager maintains a trunking relationship. Foreign domain PBXs or SIP switching equipment operate in essentially the same way as the SIP service providers, that is, as a SIP peer network element over a SIP trunk. Supported service providers include AT&T and Verizon. |
| SIP-Enabled Adjuncts | SIP-enabled adjuncts provide supplemental services to PBXs, such as voice mail and conferencing capabilities. The supported Avaya products include Avaya Voice Portal 5.2 and 6.X, Avaya Modular Messaging 5.X, Avaya Aura Messaging 6.X, Avaya Meeting Exchange 6.X, 5.X. The supported Polycom products include the RMX 1500, 2000, and 4000 series. |
| SIP Devices | SIP devices, particularly the Avaya 96XX handsets R2.6, can register to the Session Manager core. Session Manager can support up to 50,000 Avaya SIP devices. Session Manager provides SIP proxy, registrar, location services, and more to this initial set of devices. |

# Using Avaya Communication Manager as a Feature Server

When configured as a Feature Server, Avaya Communication Manager 5.2 provides features available in the Communication Manager to SIP endpoints using the IP Multimedia Subsystem (IMS) half-call model, which enables full application sequencing. You can connect the Communication Manager server to Session Manager with a SIP-IMS Service Control (ISC) interface that uses an IMS-enabled SIP signaling group and associated SIP trunk group. The feature server supports only those SIP endpoints that are registered to the Avaya Aura Session Manager.

The following limitations apply when using Communication Manager as a Feature Server:

- The dial plan for IMS users must route all PSTN calls back to the Session Manager over the IMS trunk group. Routing of such calls directly to ISDN trunks is not supported.

- IPSI port networks are not supported.
- Traditional phones such as DCP, H.323, ISDN, and analog are not supported.

# Using Avaya Communication Manager as an Evolution Server

When you use the Communication Manager as an Evolution Server you can use features available in the Communication Manager with SIP and non-SIP endpoints using the full call model with Communication Manager as the only supported application.

When you configure Communication Manager as an Evolution Server:

- H.323, digital, and analog endpoints register with Communication Manager
- SIP endpoints register with Session Manager
- All endpoints receive service from Communication Manager

The Session Manager routes calls from and to SIP endpoints. The connection from the Evolution Server to the Session Manager server is a non-IMS signaling group, a SIP-ISC interface. You administer Communication Manager as an Evolution Server by disabling IMS on the signaling group to Session Manager. Session Manager can then handle call routing for SIP endpoints and allow endpoints to communicate with other endpoints that are connected to the Evolution Server.

The following table describes the half-call and full-call models.

**Table 2-3: Session Manager Call Models**

| *Model* | *Description* |
|---|---|
| Half-Call | The half-call model separates the processing of a call request into two phases:<br>• Origination — services are applied to the originator of the call<br>• Termination — services are applied to the call recipient<br>The origination and termination phases of the call are separate operations and might be performed by different feature servers.<br>Application sequencing works only when all the servers in a sequence support the half-call model. The number of originating sequenced applications may be different from the number of terminating sequenced applications. |

| *Model* | *Description* |
|---|---|
| Full-Call | In the full-call model, a call request is processed in one step. The origination and termination parts of the call are processed without a break. Traditional Communication Manager adheres to the full-call model.<br>Application sequencing has significant limitations when at least one of the servers in a sequence adheres to the full-call model. When Communication Manager is administered as an Evolution Server, Communication Manager is the only supported application.<br>For the full-call model, the **IMS-enabled?** field on the SIP signaling group form must be disabled. |

# Chapter 3: Testing and Scenarios

This chapter explains interoperability testing performed between Polycom® HDX systems, DMA, RMX, and ITP systems and Avaya Communication Manager (CM) 5.2.1 and Avaya Aura 6.2.

## Products Tested for Interoperability with Avaya Communication Manager 5.2.1

The purpose of the testing documented in this section is to verify interoperability of Polycom HDX, DMA, RMX, and ITP systems, and Avaya endpoints with Avaya Communication Manager (CM) 5.2.1. Interoperability testing was performed with the Polycom and Avaya endpoints and versions listed in the following table.

**Table 3-1: Polycom Interoperability with Avaya**

| ***Polycom*** | ***Avaya CM 5.2.1*** | **Avaya Aura H.323** | **Avaya Aura H.323 v.6.2 SP2** | **Avaya Aura SIP v.6.1** | **Avaya Aura SIP v.6.2 SP2** |
|---|---|---|---|---|---|
| **CMA 4000 \| 5000** | | | | | |
| v. 5.0 | Supported | Supported | Not Tested | Not Applicable | Not Applicable |
| v. 6.0 | Supported | Supported | Not Tested | | |
| **DMA 7000** | | | | | |
| v. 3 | Supported | Not Tested | Not Tested | Not Tested | Not Tested |
| v.4 | Supported | Not Tested | Not Tested | Not Tested | Not Tested |
| v.5 | Supported | Supported | Supported | Supported | Supported |
| v.6 | Supported | Not Tested | Supported | Supported | Supported |
| **RMX Family** | | | | | |
| v. 7.0.3 | Supported | Supported | Not Tested | Not Tested | Not Tested |
| v. 7.2 | Supported | Supported | Not Tested | Not Tested | Not Tested |
| v. 7.6 | Supported | Supported | Not Tested | Not Tested | Not Tested |
| v. 7.7 | Supported | Supported | Supported | Supported | Supported |
| v. 7.8 | Supported | Not Tested | Supported | Not Tested | Supported |

| ***Polycom*** | ***Avaya CM 5.2.1*** | **Avaya Aura H.323** | **Avaya Aura H.323 v.6.2 SP2** | **Avaya Aura SIP v.6.1** | **Avaya Aura SIP v.6.2 SP2** |
|---|---|---|---|---|---|
| **HDX Family** | | | | | |
| v. 2.6.1 | Supported | Supported | Not Tested | Supported | Not Tested |
| v. 3.0.3 | Supported | Supported | Not Tested | Supported | Not Tested |
| v. 3.0.5 | Supported | Not Tested | Supported | Not Tested | Supported |
| v. 3.1.0 | Supported | Not Tested | Supported | Not Tested | Supported |
| **ITP (RMX \| OTX)** | | | | | |
| v. 3.0.3 | Supported | Not Tested | Supported | Not Tested | Supported |
| v. 3.0.5 | Supported | Supported | Supported | Supported | Supported |
| v. 3.1.0 | Supported | Supported | Supported | Supported | Supported |
| **VBP-E** | | | | | |
| v. 9.1.5.3 | Supported | Supported | Not Tested | Not Applicable | Not Applicable |
| v. 11.2.3 | Supported | Supported | Not Tested | Not Applicable | Not Applicable |
| v. 11.2.10 | Supported | Supported | Not Tested | Not Applicable | Not Applicable |
| v. 11.2.13 (RC2) | Supported | Not Tested | Supported | Not Tested | Supported |
| **Group Series** | | | | | |
| v. 1.0.1 | Not Tested | Not Applicable | Supported | Not Applicable | Supported |
| **VVX 1500** | | | | | |
| .v. 4.0.2 | Not Applicable | Not Applicable | Not Applicable | Not Applicable | Supported |

Polycom and Avaya engineering tested the following features:

- The H.323 protocol
- Point to Point, Multi-Point capabilities
- Call speeds were tested from minimum to maximum bandwidth in increments of 56kbps and 64kpbs
- A limited number of telephony features were tested, including call forwarding, call transfer, conferencing, call park.

- Source Input Format (SIF) and Common Intermediate Format (CIF) video resolutions up to 1080p
- People + Content/H.239
- Far End Camera Control (FECC)
- H.264 High Profile (HP)
- Lost Packet Recovery (LPR)

Interoperability testing was performed in the following Avaya Communication Manager (CM) 5.2.1 environments. For an illustration of each environment, see Understanding the Testing Scenarios.

- Avaya CM 5.2 Direct Connect
- Avaya CM 5.2 Direct Connect with VBP-E in LAN/Subscriber-side gatekeeper mode
- Avaya CM 5.2 Direct Connect with VBP-E in embedded gatekeeper mode
- Avaya CM 5.2 Neighbored gatekeeper DMA
- Avaya CM 5.2 Neighbored gatekeeper DMA with VBP-E in LAN/Subscriber-side gatekeeper mode

## Notes on Interoperability

Use the following notes to help you understand interoperability testing.

- You need to disable shuffling to make calls using the internal multipoint capability of the HDX when calls are traversing the VBP-E.
- Calls to the Polycom Telepresence m100 registered to Polycom DMA from endpoints registered with Avaya 5.2 work only when audio shuffling is disabled on Avaya CM5.2.
- After disabling audio shuffling on Avaya CM5.2, the following issues are not reproducible:
    - Video-96044    No video displays from the first connected HDX when a second HDX joined an H.323 call using the internal multipoint capability of the HDX as a multipoint control unit (MCU) in Avaya environment.
    - Video-96049    The HDX in a wide area network (WAN) outside of the VBP crashes when in a call with two Avaya one-X Communicators.
    - CMAD-7490    The m100 automatically goes to a held state when called from an HDX registered to Avaya CM 5.2.

# Products Tested for Interoperability with Avaya Aura 6.2

This section summarizes the Polycom and Avaya products tested for interoperability in an Avaya Aura 6.2 environment.

**Table 3-2: Products Tested for Interoperability with Avaya Aura 6.2**

| *Component / Product* | *Final Versions* |
|---|---|
| Polycom HDX Series | v.3.1.0 |
| Polycom RMX 4000 | v.7.8 |
| Polycom DMA 7000 | v.5.1 |
| Polycom VBP E 5300 | v.11.2.13 RC2 |
| Polycom ITP | v.3.1.0 |
| Polycom VVX | v.4.0.2.11307 |
| Avaya System Manager | v.6.2.0<br>Build: 6.2.0.0.15669-6.2.12.9<br>Update Revision: 6.2.12.1.1822 |
| Avaya Communication Manager | v.R016x.02.0.823.0<br>Patch 19926 |
| Avaya Session Manager | v.6.2.0.0.620110 |
| Avaya one-X Communicator | v.6.1.5.07-SP5-37495 |
| Avaya Desktop Video Device (ADVD) | v1_1_1-019004 |
| Avaya 10xx Series Video Endpoint | v4_8_3_24 |

Polycom and Avaya engineering tested the following features:

- H.323, SIP, and mixed protocol scenarios
- Audio and video protocol validation
- Point-to-point and multi-point capabilities
- Low, medium, and high call speeds
- Avaya Direct Connect without DMA
- Neighboring gatekeeper environment with DMA
- RMX Dial In / Dial Out, and ad-hoc scenarios
- ITP validation – RPX, OTX
- VBP with direct and DMA neighboring gatekeeper
- People + Content H.239

Interoperability testing was performed in the following scenarios:

- While devices are registered directly to the Avaya gatekeeper or SIP registrar

- While devices are registered to the gatekeeper/SIP proxy in DMA, and neighbored to Avaya CM gatekeeper

### Notes on Interoperability

Use the following notes to help you understand interoperability testing.

- Not all Polycom HDX and RMX models were used in every test
- Not all test scenarios included People + Content, H.239
- Not all call speeds were used in every test
- DMA does not yet support the Shuffling feature, which forced the use of G.711 in most scenarios
- Support for the m100 and RSS was discontinued shortly after testing began; these products were not fully tested

## Understanding the Testing Scenarios

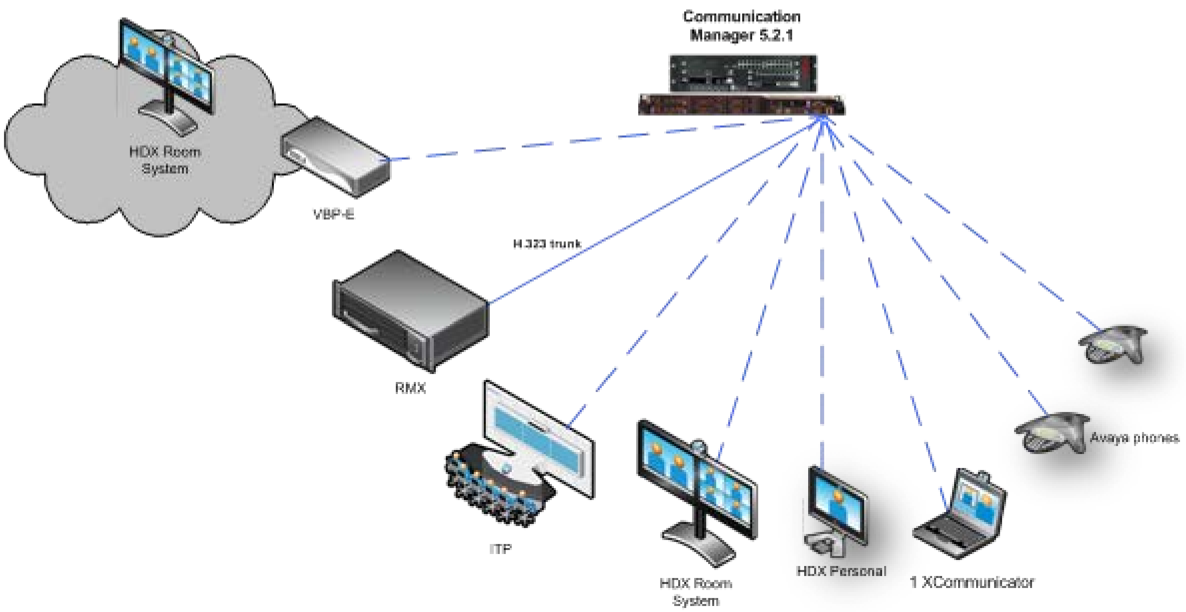
This section provides illustrations and descriptions of the test deployment scenarios used to test Avaya CM 5.2.1 and Avaya Aura 6.2 testing.

## Testing with Avaya CM 5.2.1

This section illustrates and describes two testing scenarios. In the first testing scenario, all endpoints are registered to the gatekeeper within the Avaya CM and used the Polycom VBP pointed to the Avaya CM gatekeeper (LAN / subscriber-side gatekeeper for dial string resolution. The second testing scenario used the VBP and embedded gatekeeper, with the Avaya CM gatekeeper identified as a neighbored gatekeeper. You can perform all internal dialing using extensions and you need to ensure that outbound calls made through the VBP have the target IP addresses mapped to aliases within the VBP.

The following illustration shows the first testing scenario, a direct connection using Avaya CM 5.2.1.

**Figure 3-1: Avaya CM 5.2.1 Direct Connect Scenario**

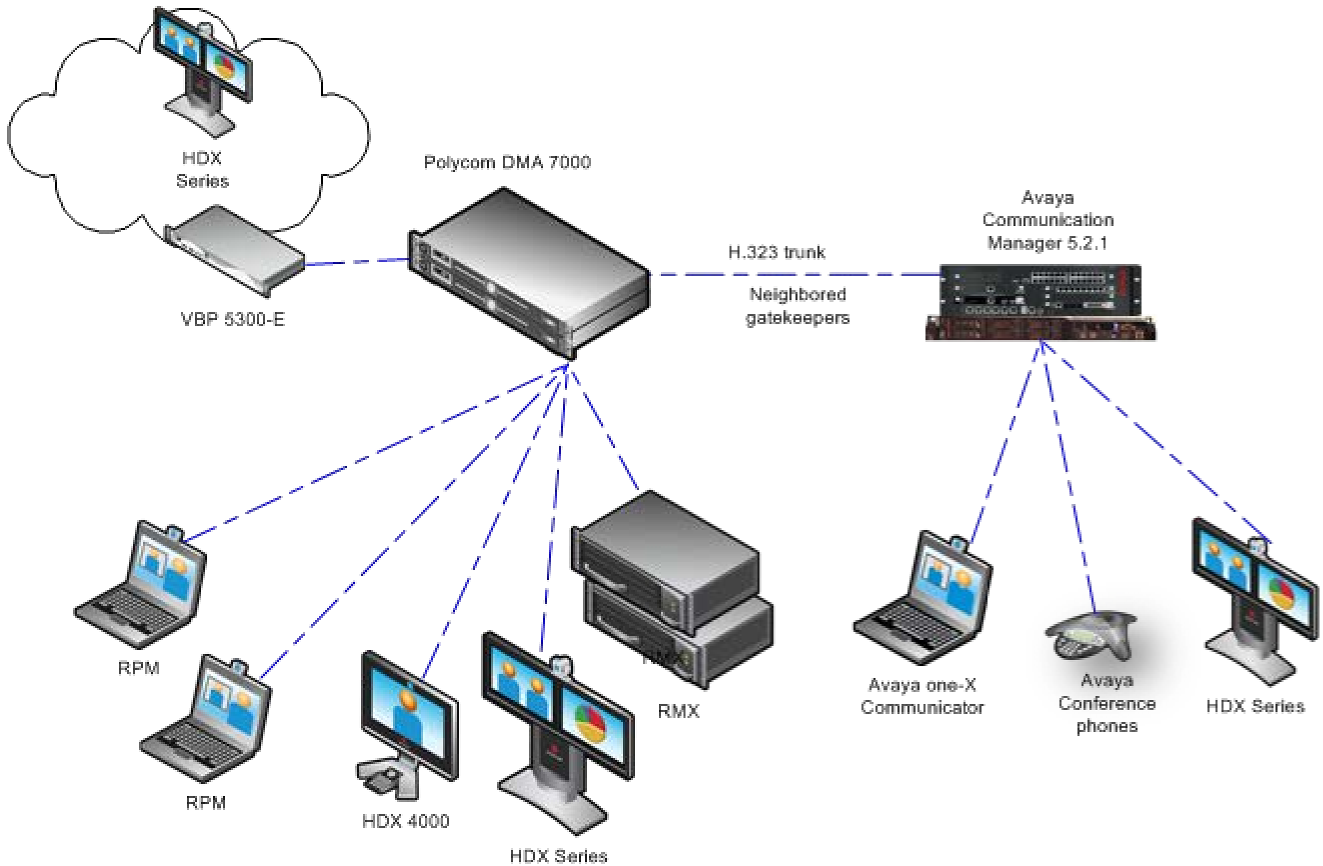



The second Avaya CM 5.2.1 testing scenario involved a neighbored gatekeeper that enables you to dial in point-to-point using devices registered to the gatekeeper in the video border proxy (VBP) or the Avaya CM gatekeeper. When using a Polycom RMX system, you can also create meetings. The Polycom VBP was tested in the LAN-side (subscriber) gatekeeper mode and pointed to the Polycom® Distributed Media Application™ (DMA™). A common use case is to link the predominantly audio-only devices on the Avaya CM to the video devices registered to the Polycom DMA system so you can call directly to a room or multipoint conference.

You can deploy the neighbored gatekeeper scenario within the RealPresence Virtualization Manager using the Polycom® Distributed Media Application™ (DMA™) 7000 or the RealPresence Resource Management using the Polycom® Converged Management Application™ (CMA™) 5000/4000.

The following illustration shows a neighbored gatekeeper scenario.

**Figure 3-2: Avaya Aura 5.2.1 Neighbored Gatekeeper Scenario**

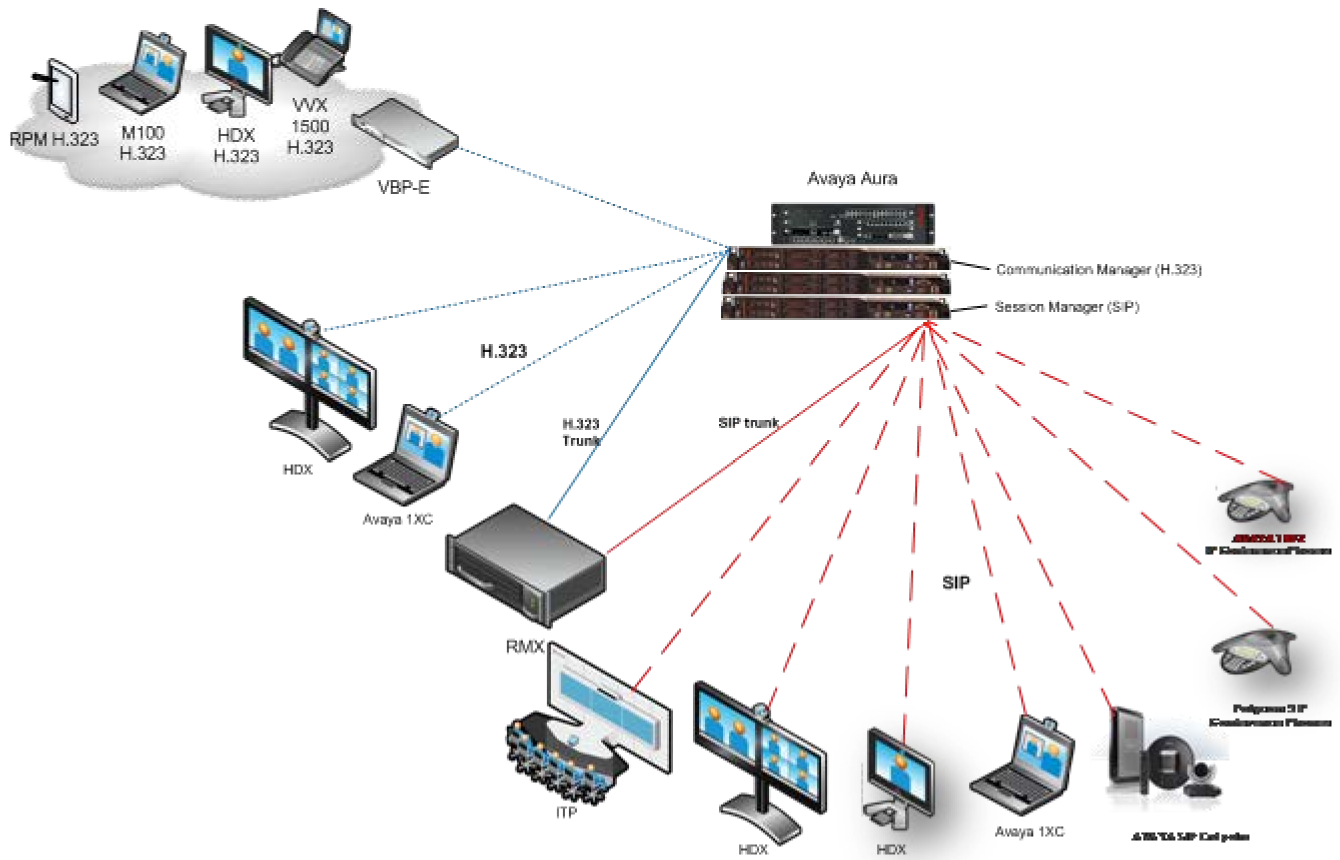



# Testing with Avaya Aura 6.2

This section illustrates and describes two Avaya Aura 6.2 testing scenarios. Using this scenario, you can register endpoints directly to the Avaya platform using SIP with the Session Manager or using H.323 and the Communication Manager Evolution Server. The first testing scenario, a direct connect scenario shown next in Figure 3, is not representative of a real deployment model. Typical environments consist predominantly of SIP-only or H.323-only devices. In some use cases you may have the entire telephony environment SIP-based, while the video devices may be running H.323. In that case, run the VBP in LAN using subscriber side gatekeeper mode and point it to the Communication Manager Evolution Server (CMES).

The following illustration shows a direct connect scenario using Avaya Aura 6.2.

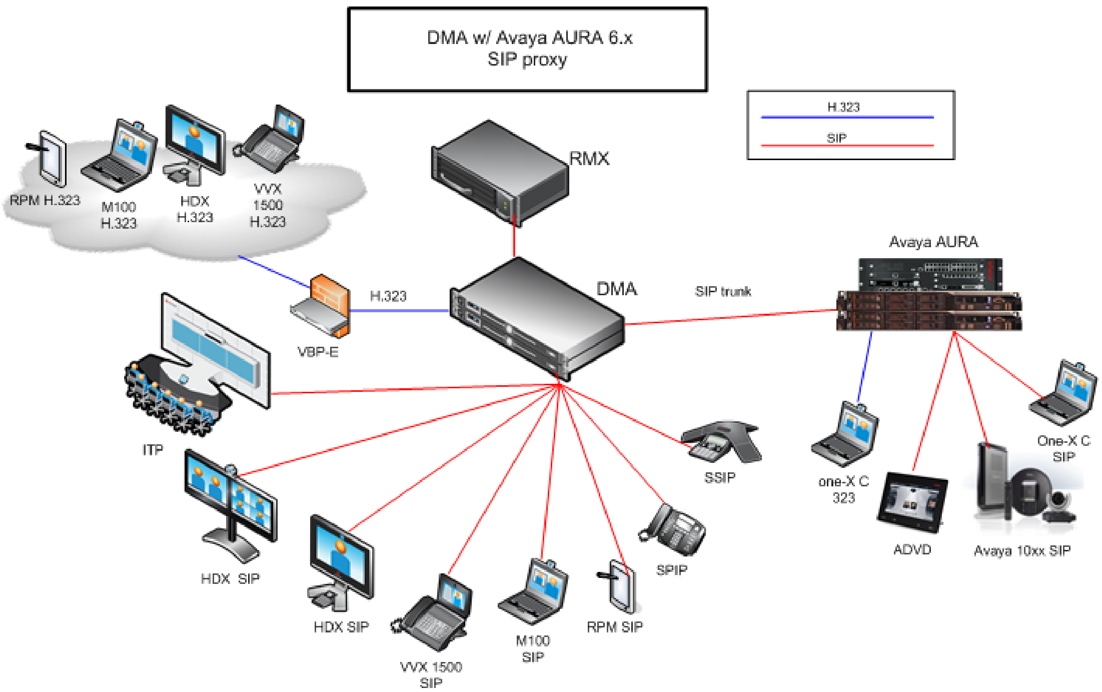
**Figure 3-3: Avaya 6.2 Direct Connect Scenario**




In the diagram above, the RMX shows connections to both H.323 and SIP- Note this is not a recommended configuration.

The second scenario below is showing the DMA as a SIP proxy to the Avaya Aura system. In this way you can have an entire video communications deployment managed by the DMA, and still maintain complete interoperability with any devices registered to the Avaya registrar.

**Figure 3-4: Avaya Aura 6.2 Neighbored Proxy Scenario**



# Chapter 4: Deployment Models and Use Cases

The deployment models supported by this solution include the following:

- Using the Joint H.323 Solution
- Using DMA as a Neighbor Gatekeeper with Avaya Communication Manager 5.2
- Understanding the Direct Connect Solution
- Understanding the Joint SIP Solution
- Polycom RMX Dual Connections

## Using the Joint H.323 Solution

This deployment solution supports point-to-point and multipoint conferencing with Avaya Communication Manager version 5.2. The deployment configuration has H.323 endpoints registering to either the Polycom CMA or the Avaya CM. There are no special Avaya telephony features for endpoints that are neighbored through Polycom CMA.

H.323 video encryption is not supported. This solution does not include the Avaya Session Manager.

### Tested, Certified and Supported Versions:

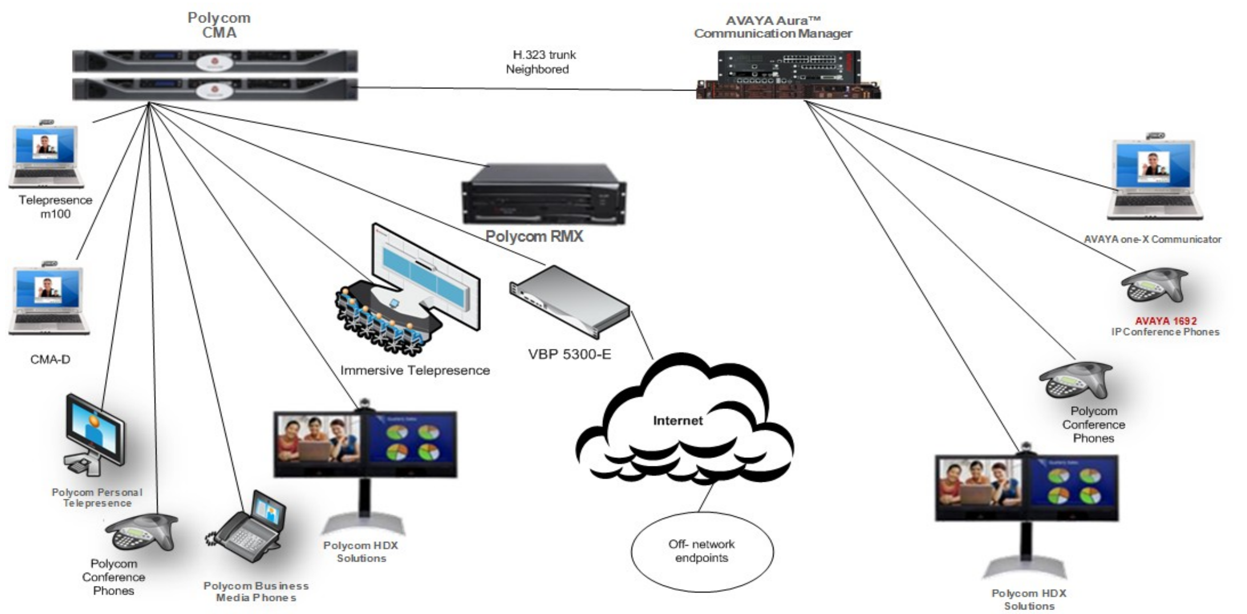
The product versions that are tested, certified, and supported are listed below.

- Avaya one-X Communicator desktop client version 5.2 and 6.0
- CMAD client version 5.0, 5.1 and 5.2
- Polycom CMA 5000 version 6.0
- Polycom HDX Series version 3.0.3
- Polycom RMX 4000 version 7.6
- Polycom Immersive Telepresence version 3.0.3
- Polycom Telepresence m100 version 1.0
- Avaya Communication Manager version 5.02
- Avaya 3641/45 wireless phones
- Avaya 1692 conference phones

**Note: Video Border Proxy Supported Scenario**

The Video Border Proxy (VBP-E series v. 9.1.5.3) is tested, certified and supported in the joint H.323 scenario below, when it is associated with the CMA gatekeeper, and only in conjunction with the Communication Manager version 5.02.

**Figure 4-1: Supported Video Border Proxy Joint H.323 Scenario**




# Using DMA as a Neighbor Gatekeeper with Avaya Communication Manager 5.2

The Polycom DMA system's Call Server capabilities provide H.323 gatekeeper functionality, and can also be configured to associate with an external gatekeeper, such as the Avaya Communication Manager. This scenario has been tested, enabling Avaya endpoints registered to the ACM to dial an endpoint registered to the DMA, or for endpoints on either side to dial into a Meeting Room on an RMX multipoint bridge.

The DMA 7000 gatekeeper must be configured in routed mode. In all versions, this can be accomplished by setting the **Admin > Call Server > Call Server Settings: Gatekeeper call mode to Routed call mode**.

In version 4.0 and later, the DMA system can be configured to use direct mode for most calls, but use routed mode for calls to the Avaya gatekeeper. This can be accomplished by setting the **Network > External Gatekeeper > Edit: Prefer** routed option for the Avaya neighbor.

**Note: Not Tested With Full Suite of Polycom Products**

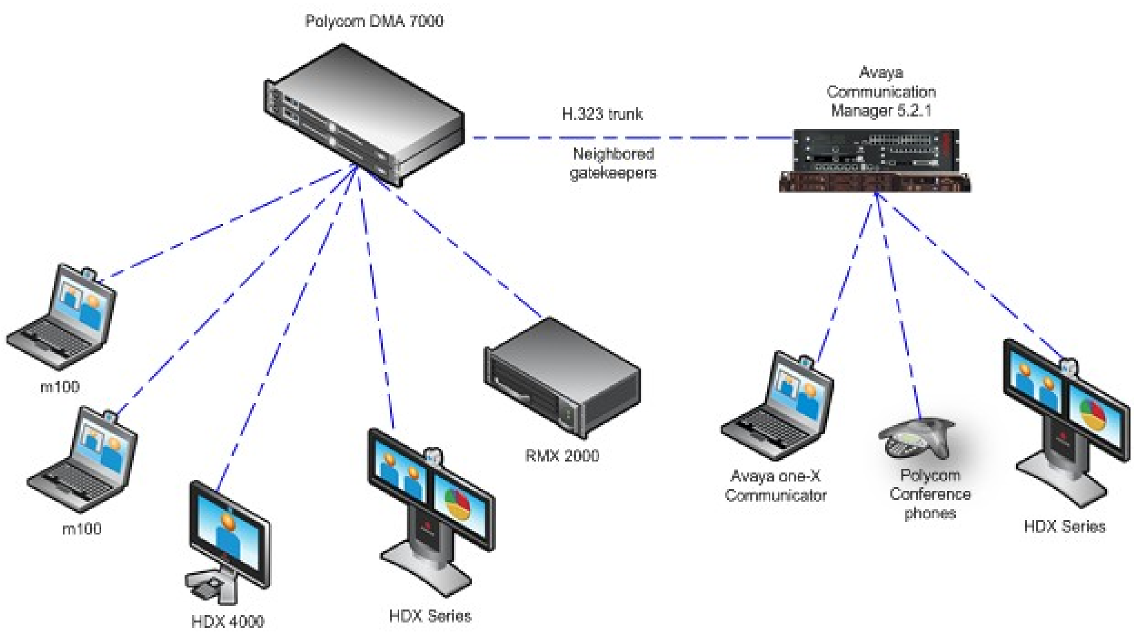
Please be aware this has not yet been tested with the full suite of Polycom products, such as the RSS 4000 and the Video Border Proxy family.

## Tested, Certified, and Supported Versions:

The product versions that are tested, certified, and supported are listed below.

- DMA versions 4.0 , 5.0 and 5.1
- HDX series version 3.0.2 , 3.0.3 , 3.0.5 and 3.1.0
- RMX series version 7.2.1 , 7.6 , 7.7 and 7.8
- Telepresence m100 version 1.0
- Avaya one-X Communicator version 6.1.2 , 6.1.4 and 6.1.5
- Avaya Communication Manager version 5.2.1

**Figure 4-2: Using DMA as a Neighbor Gatekeeper with Avaya Communication Manager 5.2**




## Use Cases

Possible use cases for this solution include:

- HDX systems registered to the DMA can call HDX systems registered to the Avaya Communication Manager, into one-X Communicators registered to the ACM, or to Telepresence m100 clients.

- HDX systems registered to either of the neighbored gatekeepers can call into the RMX multipoint bridge, utilizing either the Entry Queue or dialing directly into a Meeting Room. A Meeting Room conference on the RMX can also dial out to any of those systems.
- The m100 client can call into the RMX bridge, or receive a call from it.
- The Avaya one-X Communicator can call into the RMX bridge, or receive a call from it.

## Getting Help with DMA

For more information about installing, configuring, and administering Polycom products, refer to the Documents and Downloads link at Polycom Support.

For more information about installing and configuring Avaya Aura products, refer to the Documentation link at Avaya Support.

For information on configuring the Avaya Communication Manager for neighboring to the DMA, follow the configuration steps outlined in Application Notes of Avaya and Polycom H.323 Video Solution.

For information on configuring the DMA 7000, refer to the Polycom DMA 7000 Operations Guide at Polycom Support.

**Note: Configure Network Topology to Allow Remote Location**

If the Communication Manager is in a different Site, or in the Internet / VPN site, you must make sure the network topology is configured properly to allow access to the neighbor gatekeeper.

# Understanding the Direct Connect Solution

This deployment solution supports point-to-point and multipoint conferencing with Avaya Communication Manager version 5.2. The deployment configuration has H.323 endpoints registering directly to the Avaya Communication Manager.

## Tested, Certified and Supported Versions:

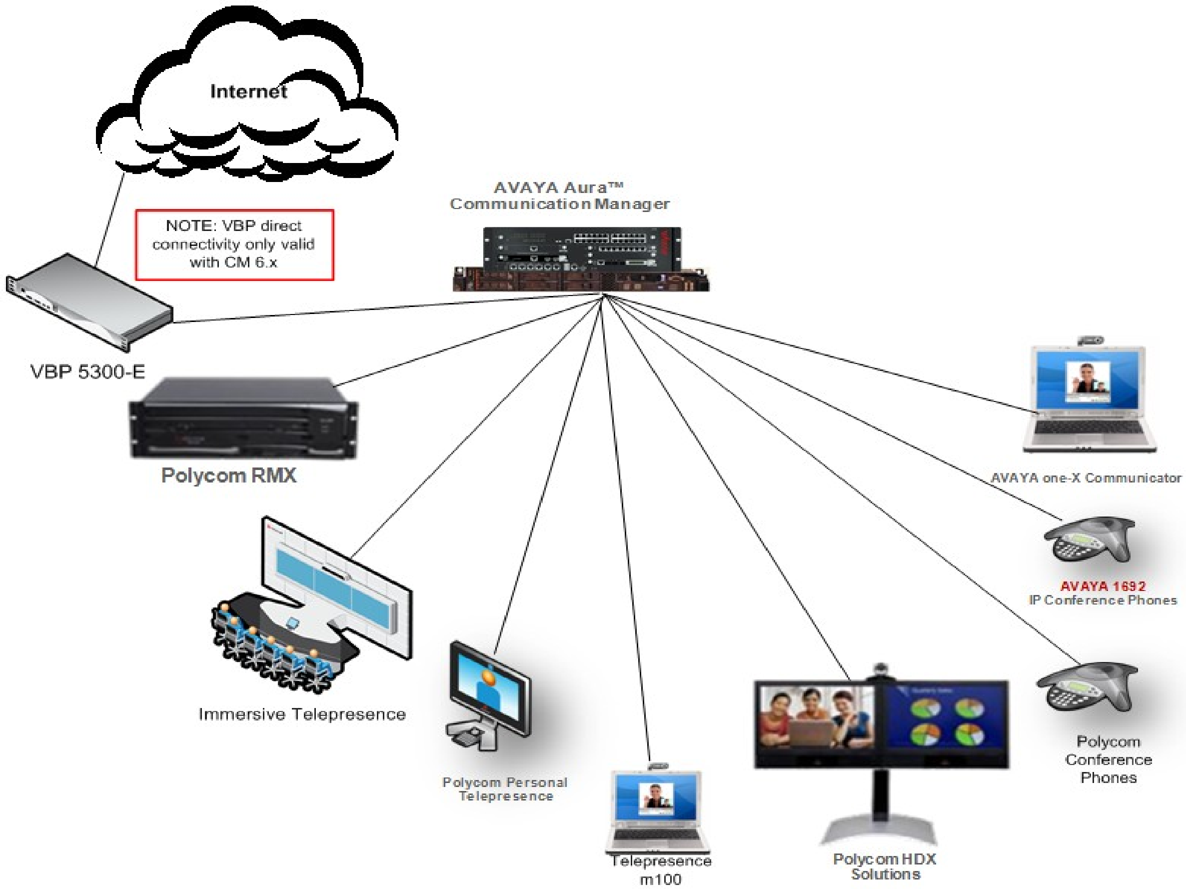
The product versions that are tested, certified, and supported are listed below.

- Avaya one-X Communicator desktop client version 5.2 , 6.0 , 6.1.4 and 6.1.5
- Polycom HDX Series version 2.6.1 , 3.0.3 , 3.0.5 and 3.1.0
- Polycom RMX Series version 7.03 , 7.2 , 7.6 , 7.7 , 7.8
- Polycom Immersive Telepresence version 3.0.3 , 3.0.5 , 3.1.0
- Polycom Telepresence m100 version 1.0
- Polycom VBP-E Series version 9.1.5.3 , 11.2.3 , 11.2.10 , 11.2.13 RC2

- Avaya Communication Manager version 5.2
- Avaya 3641/45 wireless phones
- Avaya 1692 conference phones

This deployment configuration is shown in the following figure.

**Figure 4-3: Direct Connect Solution**




# Using the Joint SIP Solution

This deployment model provides point-to-point and multipoint conferencing with other Avaya Aura 6.1 SIP video capable products and includes these configurations:

- Polycom HDX system to Avaya Session Manager 6.1
- Polycom RMX to Avaya Session Manager 6.1

## Tested, Certified and Supported Versions:

The product versions that are tested, certified, and supported are listed below.

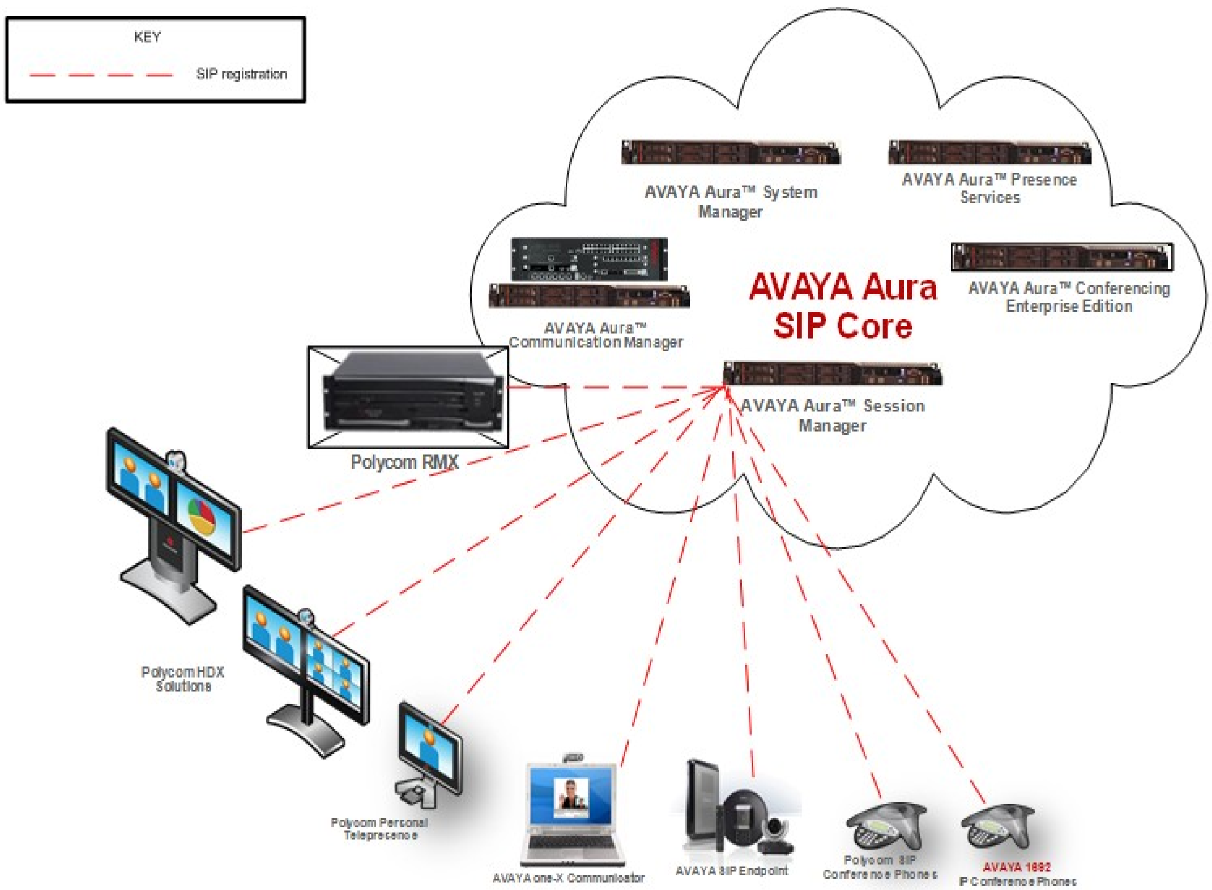
- Avaya one-X Communicator desktop client version 6.0, 6.1.5

- Avaya Communication Manager version 6.0.1, 6.2
- Polycom HDX series version 2.6.1 , 3.0.3 , 3.0.5 , 3.1.0
- Polycom RMX version 7.0.3 , 7.2 , 7.6 , 7.7 , 7.8

## Joint SIP Deployment Model

The joint SIP deployment model configuration is shown in the following figure.

**Figure 4-4: Joint SIP Deployment Model**




### Features

This deployment model has the following features:

- 1080p HD to CIF video resolution
- Wideband audio
- People+Content for Polycom HDX systems only
- Far End Camera Control supported for Polycom HDX systems only
- Avaya SIP Video implementation
- Hold/Unhold: Avaya one-X Communicator 6.0 is the initiator, not Polycom HDX systems

- Audio Mute/Unmute: Avaya one-X Communicator 6.0 is the initiator, not Polycom HDX systems
- Call Transfer: Avaya one-X Communicator 6.0 is the initiator, not Polycom HDX systems
- 6-Party Audio Ad Hoc Conference Call hosted on Avaya Aura Conferencing SE 6.0 SP1: Polycom HDX systems not as initiator

**Note: Bridge Contexts and Presence**

Presence has no context for a bridge.

## Use Cases

Possible use cases for this solution include:

- Polycom HDX systems (SIP) interoperate with Avaya one-X Communicator (SIP). Polycom HDX systems (SIP) interoperate with Avaya 1000 series SIP video endpoints.
- Polycom HDX systems (SIP) interoperate with HDX systems (SIP).
- Meet-me call: Avaya one-X Communicator 6.0 (SIP) calls a Polycom RMX 7.0.2 and 7.1 v7.6.0.103 (SIP).
- Meet-me call: Avaya 1000 Series Video Endpoint (SIP) calls a Polycom RMX 7.0.2 and 7.1 v7.6.0.103 (SIP).
- Meet-me call: Polycom HDX system (SIP) calls a Polycom RMX 7.0.2 and 7.1 v7.6.0.103 (SIP).
- Meeting room call: Polycom HDX system (SIP) initiates call to RMX meeting room, which is configured with a dial-out to Avaya Aura Conferencing Standard Edition 6.0 SP1 to add Avaya conferencing audio conferencing participants to the multipoint call.
- Scheduled call: Polycom RMX 7.0.2 and 7.1 v7.6.0.103 (SIP) calls an Avaya Video Endpoint (SIP).
- Scheduled call: Polycom RMX 7.0.2 and 7.1 v7.6.0.103 (SIP) calls an HDX system (SIP).

# Polycom RMX Dual Connections

In the first deployment model, Polycom RMX H.323 is registered to Polycom CMA that is neighbored to an Avaya Aura Evolution Server. Polycom RMX SIP is directly registered to Avaya Aura Session Manager. The Aura Evolution Server also acts as an H.323/SIP gateway. In this model, users can maintain some or all of their video endpoints on H.323, while migrating to SIP.

## Tested, Certified and Supported Versions:

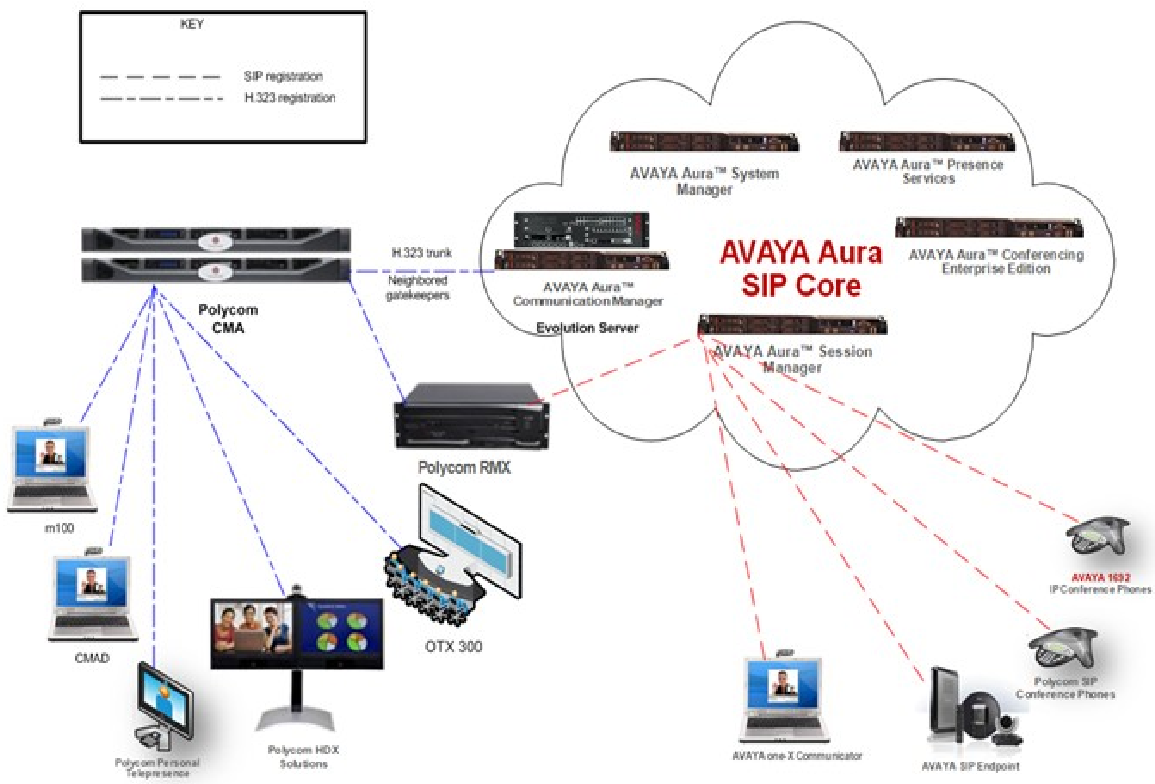
The product versions that are tested, certified, and supported are listed below.

- Avaya one-X Communicator desktop client version 6.0
- Avaya Communication Manager version 6.0.1
- Polycom HDX Series version 2.6.1
- Polycom RMX 4000 version 7.0.3 and 7.2
- Polycom Telepresence m100 version 1.0
- Polycom VSX Series (H.323 only) version 9.0.6.1 and higher

## RMX to CMA Neighbored Deployment Model

In this deployment model, illustrated in Figure 5-5, users can maintain an H.323 infrastructure, and use their Polycom CMA server for software updates, upgrades, provisioning and management, while migrating smoothly to a SIP model. Note that this deployment scenario has not been tested.

**Figure 4-5: RMX to CMA Neighbored Deployment Model**



## Features

This deployment model has the following features:

- 1080p HD to CIF video resolution (HD H.264 interworking)
- Wideband audio

**Note: Limitations in Evolution Server Release 6.0.1**

People+Content, video encryption, and Far End Camera Control are not supported in the Avaya Communication Manager Evolution Server release 6.0.1.

## Use Cases

The following is a list of possible use cases for this solution. Note that Polycom VSX systems have not been tested.

- Polycom HDX systems and VSX systems (H.323) interoperate with Avaya one-X Communicator (SIP).
- Polycom HDX systems and VSX systems (H.323) interoperate with Avaya Video endpoints (SIP), for example, Avaya 1000 series.
- Polycom HDX system (H.323) interoperates with HDX system (SIP).
- Polycom CMA Desktop (H.323) interoperates with Avaya one-X Communicator (SIP).
- Polycom CMA Desktop (H.323) interoperates with Avaya Video endpoints (SIP), for example, Avaya 1000 series.
- Avaya one-X Communicator H.323 (CMES) interoperates with an HDX system (SIP).
- Meet-me call: Polycom HDX system, Polycom CMA Desktop, Polycom VSX, and Avaya one-X Communicator (H.323) able to call into to Polycom RMX 7.0.2 & 7.1 v7.6.0.103.
- Meet-me call: Polycom HDX system, Avaya 10x0s, and Avaya one-X Communicator (SIP) able to call into RMX 7.0.2 & 7.1 v7.6.0.103.
- Scheduled call: Polycom RMX 7.0.2 and 7.1 v7.6.0.103 calls out to Polycom HDX system, Polycom VSX system, and Polycom CMA Desktop (H.323).
- Scheduled call: Polycom RMX 7.0.2 and 7.1 v7.6.0.103 calls out to Polycom HDX system, Avaya one-X Communicator, and Avaya 10x0s (SIP).

**Note: Limitations of Avaya Video Endpoints**

When the Polycom RMX is dual registered through SIP and H.323, and a Polycom HDX system sends content, other Polycom HDX endpoints see People+Content, but the Avaya video endpoints only show content. When content sharing is stopped, endpoints see People again. Also, if content is pushed from the Avaya 1000 series through the RMX, both Polycom HDX systems and Avaya endpoints can only see content but no people. This is a current limitation of the Avaya video endpoints solution when registered to Session Manager and meeting on the Polycom MCU.

# Chapter 5: Configuring Avaya Aura® Session Manager

This chapter provides an overview of how to set up and configure Avaya Session Manager to interoperate with Polycom products. For more detailed information about configuring Avaya Aura® Session Manager, refer to the Avaya Support site.

## Verifying System Capabilities and Licensing

Before you set up your system, you must verify that the correct system capabilities and licensing have been configured. If there is insufficient capacity or a required feature is not available, contact an authorized Avaya sales representative to make the appropriate changes.

The following screen shots show the configuration screens from the Avaya Communication Manager.

1 To verify that an adequate number of SIP trunk members are licensed for the system, enter the **display system-parameters customer-options** command:

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

                               IP PORT CAPACITIES                            USED
                            Maximum Administered H.323 Trunks: 12000  0
                    Maximum Concurrently Registered IP Stations: 18000  0
                       Maximum Administered Remote Office Trunks: 12000  0
       Maximum Concurrently Registered Remote Office Stations: 18000  0
                    Maximum Concurrently Registered IP eCons: 414   0
          Max Concur Registered Unauthenticated H.323 Stations: 100   0
                              Maximum Video Capable Stations: 18000  0
                          Maximum Video Capable IP Softphones: 18000  0
                              Maximum Administered SIP Trunks: 24000  128
         Maximum Administered Ad-hoc Video Conferencing Ports: 24000  50
             Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                                   Maximum TN2501 VAL Boards: 128   0
                             Maximum Media Gateway VAL Sources: 250   0
                    Maximum TN2602 Boards with 80 VoIP Channels: 128   0
                   Maximum TN2602 Boards with 320 VoIP Channels: 128   0
             Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

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

**2** Verify that ARS is enabled (on page 3 of display system-parameters customer options):

```
Display system-parameters customer-options                       Page 3 of 11
                                OPTIONAL FEATURES

 A/D Grp/Sys list Dialing Start at 01?  n                          CAS Main?  n
Answer Supervision by Call Classifier?  n                  Change COR by FAC?  n
                                  ARS   y    Computer Telephony Adjunct Links?  y
                 ARS/AAR Partitioning?  y     Cvg of Calls Redirected Off-net?  y
          ARS/AAR Dialing without FAC?  y                        DCS (Basic)?  y
          ASAI Link Core Capabilities?  y                   DCS Call Coverage  ?
```

**3** To enable Private Numbering, use the **change system-parameters customer-options** command to verify that Private Networking is enabled:

```
Display system-parameters customer-options            Page 5 of 11
                                OPTIONAL FEATURES

                 Multinational Locations?  y              Station and Trunk MSP?  y
Multiple Level Precedence & Preemption?  n       Station as Virtual Extension?  y
                      Multiple Locations?  y  System Management Data Transfer?  n
          Personal Station Access (PSA)?  Y                 Tenant Partitioning?  n
                        PNC Duplication?  n        Terminal Trans. Init. (TTI)?  Y
                    Port Network Support?  n                Time of Day Routing?  n
                        Posted Messages?  n      TN2501 VAL Maximum Capacity?  y
                    Uniform Dialing Plan?  y                 Private Networking?  y
         Usage Allocation Enhancements?  y          Processor and System MSP?  y
                    Processor Ethernet?  y                 Wideband Switching?  n
                              Wireless?  y
```

4 Enter the **change node-names ip** command to add the node-name and IP address for the Session Manager's software asset, if not previously added.

```
change node-names ip                                   Page 1 of 2
                                   IP NODE NAMES

NAMES Name            IP Address
default               0.0.0.0
procr                 135.9.88.72
procr6                ::
silasm4               135.9.88.62
```

5 Enter the **change ip-codec-set n** command, where *n* is the next available number. Enter the following values:

- Enter **G.711MU** and **G.729** as supported types of Audio Codecs.
- Retain the default value of **n** in the Silence Suppression field.
- Enter **2** in the Frames Per Pkt field.
- Enter **20** in the Packet Size (ms) field.
- If your system uses media encryption, enter the required value in the Media Encryption field. For the sample configuration, none was used.

```
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. G.711MU          n                         2                  20
2. G.729            n                         2                  20
Media Encryption
1: none
```

**6** Enter the **change ip-network-region 1** command to set the Authoritative Domain. For the sample configuration, dr.avaya.com was used. Verify that the **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** fields are set to yes.

```
change ip-network-region 1                                          page 1 of 19
                                 IP NETWORK REGION

Region: 1
Location: 1                      Authoritative Domain: dr.avaya.com
Name: CMFS-Video
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?                N
UDP Port Max: 16585
```

**7** Enter the **add signaling-group n** command, where *n* is an available signaling group number for one of the SIP trunks to the Session Manager. Fill in the requested fields. In the sample configuration, trunk group 1 and signaling group 1 were used to connect to Avaya Aura Session Manager. For the remaining fields, you can use the default values.

- Enter **sip** in the Group Type field.
- Enter **tcp** in the Transport Method field.
- Enter **y** in the IMS Enabled? field.
- Enter **y** in the IP Video? field.
- Enter **y** in the Peer Detection Enabled? field.
- Use the default value in the Peer Server field. The default value is replaced with SM after the SIP trunk to Session Manager is established.
- Enter **proc** in the Near-end Node Name field.
- Enter **Session Manager** node name in the Far-end Node Name field.
- Enter **5060** in the Near-end Listen Port field.
- Enter **5060** in the Far-end Listen Port field.
- Enter Authoritative Domain in the Far-end Domain field.
- Enter **y** in the Enable Layer 3 Test field.
- Enter **y** in the Direct IP-IP Early Media? field.

```
display signaling-group 1                                           page 1 of 1
```

SIGNALING GROUP

| | | | |
|---|---|---|---|
| **Group Number:** | **1** | Group Type: | **sip** |
| | | Transport Method: | **tcp** |
| **IMS Enabled?** | **y** | | |
| **IP Video?** | **y** | Priority Video? | y |
| **Peer Detection Enabled?** | **y** | **Peer Server:** | **SM** |
| **Near-end Node Name:** | **procr** | **Far-end Node Name:** | **silasm4** |
| **Near-end Listen Port:** | **5060** | **Far-end Listen Port** | **5060** |
| Far-end Network Region | 1 | **Far-end Domain:** | **dr.avaya.com** |
| Bypass if IP Threshold Exceeded? | n | Incoming Dialog Loopbacks: | eliminate |
| RFC 3389 Comfort Noise? | n | DTMF over IP: | rtp-payload |
| Direct IP-IP Audio Connections? | y | | |
| Session Establishment Timer (min) | 3 | IP Audio Hairpinning? | n |
| **Enable Layer 3 Test?** | **y** | **Direct IP-IP Early Media?** | **y** |
| H.323 Station Outgoing Direct Media? | n | Alternate Route Timer (sec) | 6 |

# Configuring Avaya Aura Session Manager

This section describes the System Manager Web Interface, which is the main user interface used to perform management tasks in an Avaya Aura system. Once you have access to the web interface, you can configure Communications Manager Objects like endpoints and templates.

## Logging on to System Manager Web Interface

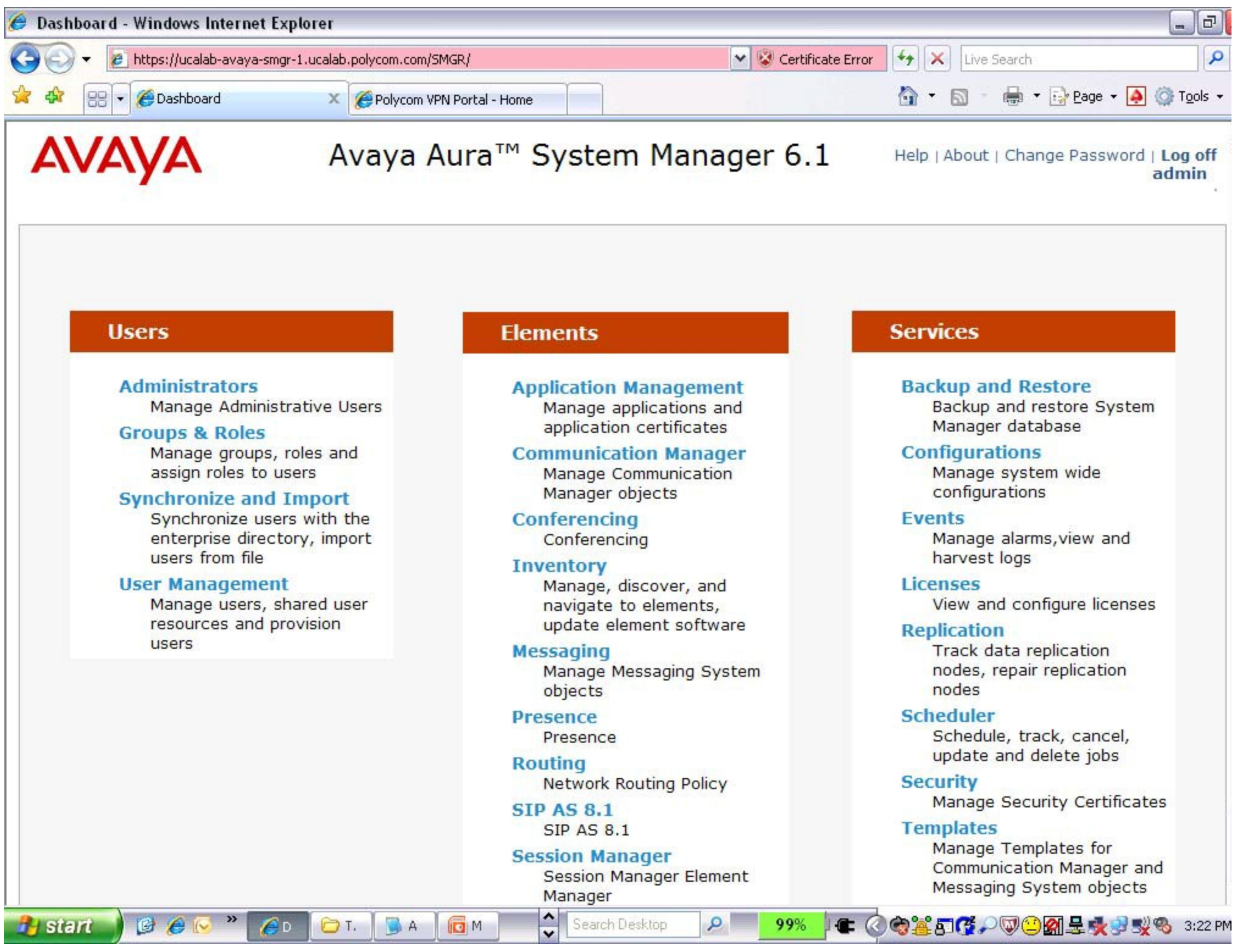
The Avaya Aura System Manager web interface, or web console, is the main user interface of Avaya Aura System Manager. You need to log on to the System Manager web console before you can perform any tasks. You must have a user account to access the System Manager Web Interface; contact your system administrator to create an account if you do not already have one.

**To log on to the System Manager web interface:**

1. Open a web browser and access the System Manager web interface: (https://<SERVER_NAME>/ SMGR).
2. Enter your user name in the User ID field.
3. Enter your password in the Password field.
4. Click **Log On**.

If your user name and password match an authorized Avaya Aura System Manager user account, the System Manager home page appears with the System Manager version number. The System Manager dashboard, shown in the following figure, displays a navigation menu that provides access to shared services. The tasks you can perform depend on your user role.

**Figure 5-1 The Avaya Aura System Manager Dashboard**




## Managing Communications Manager Objects

Avaya Aura System Manager displays a collection of Communication Manager objects under Communication Manager. System Manager allows you to directly add, edit, review, or delete these objects through Communication Manager.

### Endpoint Management

You can create and manage endpoints through the System Manager web console. Endpoint management provides support for Communication Manager endpoint objects, and helps you add, change, delete, and view endpoint data.

## Templates

You can use templates to specify the parameters of an endpoint or a subscriber and then reuse that template for subsequent endpoint or subscriber tasks. The system provides default templates you cannot change, and you can also add your own custom templates that you can modify or remove.

## Adding an Endpoint

**To add an endpoint, complete these steps:**

1. Click **Communication Manager** on the System Manager console under Elements.
2. Click **Endpoints > Manage Endpoints** in the left navigation pane.
3. Select a Communication Manager from the Communication Manager list.
4. Click **Show List**.

   The system displays the available Endpoints list on the Communication Manager that you selected.
5. Click **New**, and select the template based on the set type you want to add.

   The system displays all the sections on the Add Endpoint page.
6. Complete the Add Endpoint page and click **Commit** to add the endpoint.

   Before you add an endpoint, you must complete the mandatory fields that are marked with an asterisk (*) under the General options, Feature Options, Site Data, Data Module/Analog Adjunct, Abbreviated Call Dialing, Enhanced Call Fwd, Button Assignment sections

**Note: Adding Endpoints with Non-Supported Set Types**

To add an endpoint with a non-supported set type, add the endpoint using Element Cut Through. For alias endpoints, you can choose the corresponding Alias set type from the Template field. System Manager automatically creates a template for the Alias set types based on the aliased-to set type. Alias endpoint templates have names beginning with Alias. Before the Alias endpoint type Template appears in the drop-down menu, you need to create an alias set type on the managed Communication Manager. You can then use the template to add an endpoint.

## Avaya Communication Manager Objects

The objects and object types used by Avaya Communication Manager are listed next, in Table 6-1.

**Table 5-1: Avaya Communication Manager Objects**

| *Object Type* | *Objects* |
|---|---|
| Call Center | Agents<br>Announcements<br>Audio Group<br>Best Service Routing<br>Holiday Tables<br>Variables<br>Vector<br>Vector Directory Number<br>Vector Routing Table<br>Service Hours Tables |
| Coverage | Coverage Answer Group<br>Coverage Path<br>Coverage Remote<br>Coverage Time of Day |
| Endpoints | Alias Endpoint<br>Intra Switch CDR<br>Manage Endpoints<br>Off PBX Endpoint Mapping<br>Site Data<br>Xmobile Configuration<br>Groups Group Page<br>Hunt Group<br>Intercom Group<br>Pickup Group<br>Terminating Extension Group<br>Trunk Group |

| *Object Type* | *Objects* |
|---|---|
| Network | Automatic Alternate Routing Analysis<br>Automatic Alternate Routing Digit Conversion<br>Automatic Route Selection Analysis<br>Automatic Route Selection Digit Conversion<br>Automatic Route Selection Toll<br>Data Modules<br>IP Interfaces<br>IP Network Regions<br>Node Names<br>Route Pattern<br>Signaling Groups |
| Parameters | System Parameters - CDR Options<br>System Parameters - Customer Options<br>System Parameters - Features<br>System Parameters - Security<br>System Parameters - Special Applications |
| System | Abbreviated Dialing<br>Enhanced Abbreviated Dialing<br>Group Abbreviated Dialing<br>Personal Authorization Code<br>Class of Restriction<br>Class of Service<br>Class of Service Group<br>Dialplan Analysis<br>Dialplan Parameters<br>Feature Access Codes<br>Locations<br>Uniform Dial Plan |

# Chapter 6: Configuring Polycom® Systems to Interoperate with Avaya Communication Manager

This chapter describes how to configure Polycom systems when interoperating with Avaya products. Configuration instructions are provided for HDX systems, RMX systems, and Polycom Telepresence m100 systems.

## Configuring HDX Systems

For detailed information about setting up and configuring HDX Systems to interoperate with Avaya products, refer to the *Administrator's Guide for Polycom HDX Systems*.

**Note: Registering As An H.323 Gatekeeper**

If you are registering your system to the Avaya Aura Communication Manager as an H.323 gatekeeper, you need to have an Avaya option license key installed. This license key is available at no charge from your Polycom representative. Installing the license key enables your system to register to the gatekeeper using authentication, which results in an extra field displaying on the interface, and allows communication with the Avaya H.323 stack.

### Configuring Integration with Avaya Networks

**To install the Avaya option key:**

1. Obtain an Avaya key code from your Polycom account representative, which will enable the HDX system to register correctly and use the Avaya communications protocols.
2. On the Polycom HDX system go to **System > Admin Settings > General Settings > Options** and enter the key code for the Avaya option.

**To configure the Polycom HDX system to use Avaya network features:**

1. Do one of the following:
   - In the local interface, go to **System** > **Admin Settings** > **Network** > **IP** > **H.323 Settings** (select if necessary).
   - In the web interface, go to **Admin Settings** > **Network** > **IP Network**.

2 Set Use Gatekeeper to Specify with PIN.

   a Enter the **H.323 Extension (E.164)** provided by the Avaya Communication Manager administrator.

   b Enter the Avaya Communication Manager IP address for **Gatekeeper IP Address**.

   c Enter the **Authentication PIN** provided by the Avaya Communication Manager administrator.

3 Do one of the following:

   ○ In the local interface, go to **System** > **Admin Settings** > **Network** > **IP** > **Call Preference** (select if necessary).

   ○ In the web interface, go to **Admin Settings** > **Network** > **IP Network** > **Call Preference**.

4 Set **Enable H.239**.

## System Functionality

Polycom HDX systems running H.323 and registered to the Avaya H.323 Gatekeeper with an Avaya option key can use the following features on an Avaya telephony network:

- Call forwarding (all, busy, no answer) — Configured by the Avaya Communication Manager administrator and implemented by the user
- Call coverage — Configured by the Avaya Communication Manager administrator
- Transfer — Implemented via flash hook and dialing digits
- Audio conference — Implemented via flash hook and dialing digits
- Call park
- Answer back
- DTMF tones for Avaya functions

Refer to the Avaya documentation and User's Guide for Polycom HDX Systems for information about these features.

**Note: Accessing Features**

Some features require multiple buttons to be manipulated, as the Polycom HDX systems do not have dedicated telephony function controls.

**Note: AES Encryption Restriction**

AES Encryption is not supported for systems registered to an Avaya H.323 gatekeeper.

**Note: SIP Limitations**

The SIP protocol has been widely adapted for voice over IP communications and basic video conferencing; however, many of the advanced video conferencing capabilities are not yet standardized. Many capabilities also depend on the SIP server.
Examples of features that are not supported using SIP are:

- Cascaded multipoint is not supported in SIP calls.

For more information about SIP compatibility issues, refer to the *Release Notes for Polycom HDX Systems*.

If you are going to configure the HDX system to use the Avaya Session Manager, please refer to the Polycom HDX Systems Administrator's Guide for detailed information about setting up and registering to the Session Manager and configuring for SIP

# Configuring RMX Systems

For detailed information about setting up and configuring RMX Systems to interoperate with Avaya products, refer to the RMX documentation on the Polycom Support site: support.polycom.com.

# Configuring Polycom Telepresence m100 Systems

This section provides an overview of how to set up and configure Polycom Telepresence m100 systems to interoperate with Avaya products. For more detailed information about configuring Polycom Telepresence m100, refer to the Polycom Telepresence m100 documentation on the Polycom Support site: support.polycom.com.

## Configuring Polycom Telepresence m100 Systems Properties

The instructions in this section will allow you to configure the properties of Telepresence m100 systems for interoperation in an Avaya Communication Manager environment.

**Note: Getting Your Configuration Entries**

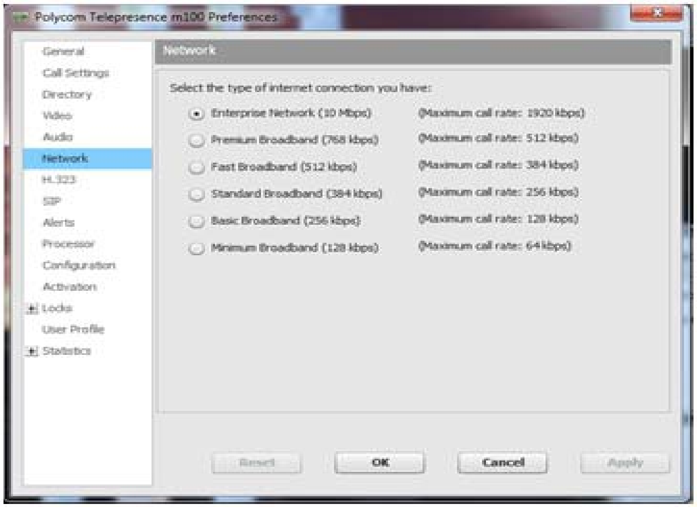
Check with your Network Administrator for the correct entries for your environment.

**To configure Polycom Telepresence m100 properties:**

Once the application is installed and launched on your PC, go to **Menu > Preferences**, as shown next.

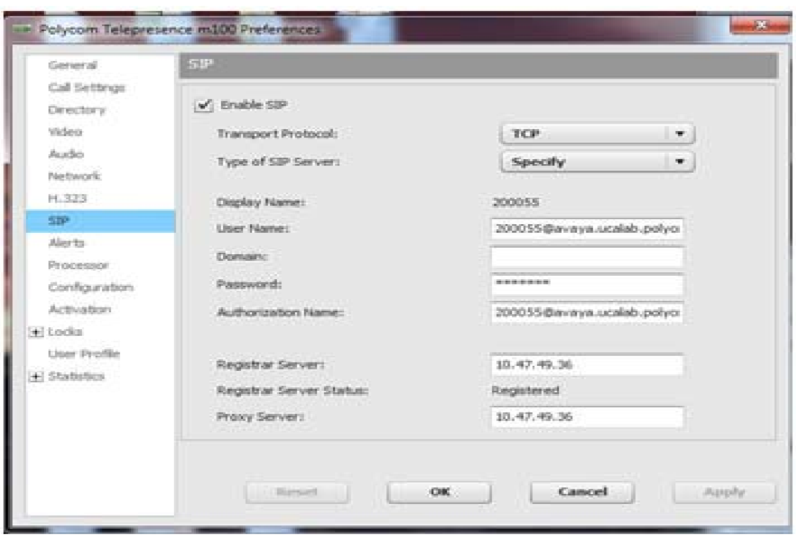
## Network Settings

On the Network page, you can choose the type of network you are connected to, depending on your location and bandwidth availability:




## SIP Settings

The application will utilize either H.323 or SIP, depending on your network and preference. Although the Avaya Communication Manager supports both H.323 and SIP, you can NOT register the m100 application directly to it using H.323, as we currently do not support the registration with authentication within this application. It has been tested in a neighbored scenario, registered to a CMA or DMA via H.323, which is neighbored into the Avaya gatekeeper.




**In the SIP tab, make sure the following fields are set correctly:**

- **Enable SIP** is checked

- **Transport Protocol** Specifies the protocol the system uses for SIP signaling
  - TCP Provides reliable transport via TCP for SIP signaling.
  - UDP Provides best-effort transport via UDP for SIP signaling.
- **Type of SIP Server** Specify
- **Display Name** Specifies the user name for authentication with a Registrar Server
- **User Name** Specifies the user name for authentication with a Registrar Server
- **Domain** Specifies the domain name for authentication with Registrar Server
- **Password** Specifies the password for authentication with Registrar Server
- **Authorization Name** Specifies the name to use for authentication when registering with Registrar Server
- **Registrar Server** Specifies the DNS name or IP address of the Registrar Server
- **Registrar Server Status** Specifies if the system is register with Registrar Server
- **Proxy Server** Specifies the DNS name or IP address of the SIP Proxy Server
-

# Chapter 7: Troubleshooting

Use this section to find solutions to common troubleshooting problems.

**How do I set up an HDX (or other system) with internal multipoint to work directly with the Avaya CM 5.2.1?**

On the Communication Manager, you will need to create multiple stations for this HDX, (N-1). For instance, if you have four-way multipoint capability, you will need to create three stations and configure them in a circular hunt group, for example, 1001 points to 1002, 1002 points to 1003, and 1003 points to 1001.

**Where can I get the Avaya option key for my HDX systems so I can register them directly to the Communication Manager? And why do I need it?**

The Avaya option key used to be a chargeable item that you could obtain from Avaya. The Avaya option key is now a no-cost option issued by Polycom. You can request an Avaya option key from your reseller, partner, Polycom Global Services, or send an email to Avaya@polycom.com.

You will need the serial number of your system to generate a key code.

Avaya uses a slightly modified H.323 stack and Avaya's registration procedure requires endpoint devices to register with authentication to operate properly. The Avaya license key enables both of these capabilities and the gatekeeper registration options on the HDX will display an additional choice 'register with authentication'. This code is created on the Communication Manager, and assigned to you by your administrator.

**Do I need a license key for my RMX to register to the Communication Manager?**

Yes, you will need to acquire the license key. The fastest method to get the license key is to emailAvaya@polycom.com. You will need your system serial number.

# Chapter 8: Getting Help

Use this section to find additional help and guidance with this solution and all Polycom products, and find links to the Polycom Community.

## Polycom and Partner Resources

For more information about installing, configuring, and administering Polycom products, refer to Documents and Downloads at Polycom Support.

To find all Polycom partner solutions, see Polycom Global Strategic Partner Solutions.

For more information on solution with this Polycom partner, see Polycom Global Strategic Partner Solutions.

For more information about installing, configuring, and administering Polycom products, refer to the Documents and Downloads link at Polycom Support.

For more information about installing and configuring Avaya Aura products, refer to the Documentation link at Avaya Support.

For more information about CMA and CMAD configurations, refer to Application Notes of Avaya and Polycom H.323 Video Solution.

For general Avaya-related questions, please send an email to Avaya@polycom.com

# The Polycom Community

The Polycom Community gives you access to the latest developer and support information. Participate in discussion forums to share ideas and solve problems with your colleagues. To register with the Polycom Community, simply create a Polycom online account. When logged in, you can access Polycom support personnel and participate in developer and support forums to find the latest information on hardware, software, and partner solutions topics.