



Application Notes for Polycom® CX5500 Unified Conference Station with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Polycom® CX5500 Unified Conference Station to interoperate with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0. The Polycom CX5500 Unified Conference Station is a SIP conferencing phone that can register with Avaya Aura® Session Manager as a SIP endpoint in support of voice communications and conferencing requirements.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for Polycom CX5500 Unified Conference Station to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Polycom CX5500 Unified Conference Station (CX5500) is a SIP conference phone that registers with Avaya Aura® Session Manager (Session Manager) as a SIP endpoint combining the functionality of an IP phone and a conferencing station in support of voice communications and conferencing requirements.

Note: In this configuration, only SIP telephony and audio capabilities of the CX5500 were tested. CX5500 also supports USB audio and 360 degree panoramic video when connected to a Microsoft Lync or Skype for Business clients.

2. General Test Approach and Test Results

The general test approach was to place calls to and from the CX5500 and exercise basic telephone operations. The main objectives were to verify the following:

- Registration
- Codecs (G.711, G.722, iLBC and G.729)
- Inbound calls
- Outbound calls
- Hold/Resume
- Call Transfer and Conferencing (Blind and Attended)
- Call termination (origination/destination)
- Avaya Features using Feature Access Codes (FAC)
 - Call Park/Unpark
 - Call Pickup
 - Call Forward (Unconditional, Busy/no answer)
 - Find Me
- Voicemail
- Message Waiting Indicator (MWI)
- Serviceability

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of interoperability compliance testing was primarily on verifying call establishment on the CX5500. The CX5500 operations such as inbound calls, outbound calls, hold/resume, transfer, conference, Feature Access Codes, and its interactions with Session Manager, Communication Manager, and other Avaya SIP, and H.323 phones were verified. The serviceability testing introduced failure scenarios to see if CX5500 can recover from failures.

2.2. Test Results

The test objectives were verified. For serviceability testing, CX5500 operated properly after recovering from failures such as network disconnects, and resets of CX5500.

The features mentioned in **Section 2** worked successfully during compliance testing with the following exceptions, as these features are currently not supported by CX5500:

- Blind Conference calls
- Long Hold Recall Timer
- Find Me
- iLBC Codec is supported only between CX5500 endpoints
- At least one hardware-supported codec needs to be listed on CX5500 for iLBC or G.722 to work. Additionally, these codecs need to be configured at the top of the list in **Section 6.2**

2.3. Support

For technical support on Polycom® CX5500 Unified Conference Station, please contact via the following:

- Web: <http://support.polycom.com>

3. Reference Configuration

Once CX5500 registers as a SIP endpoint with Session Manager, it can place and receive voice calls with various supported features as listed above in **Section 2.1**. The reference configuration used for the compliance test is shown in **Figure 1** below.

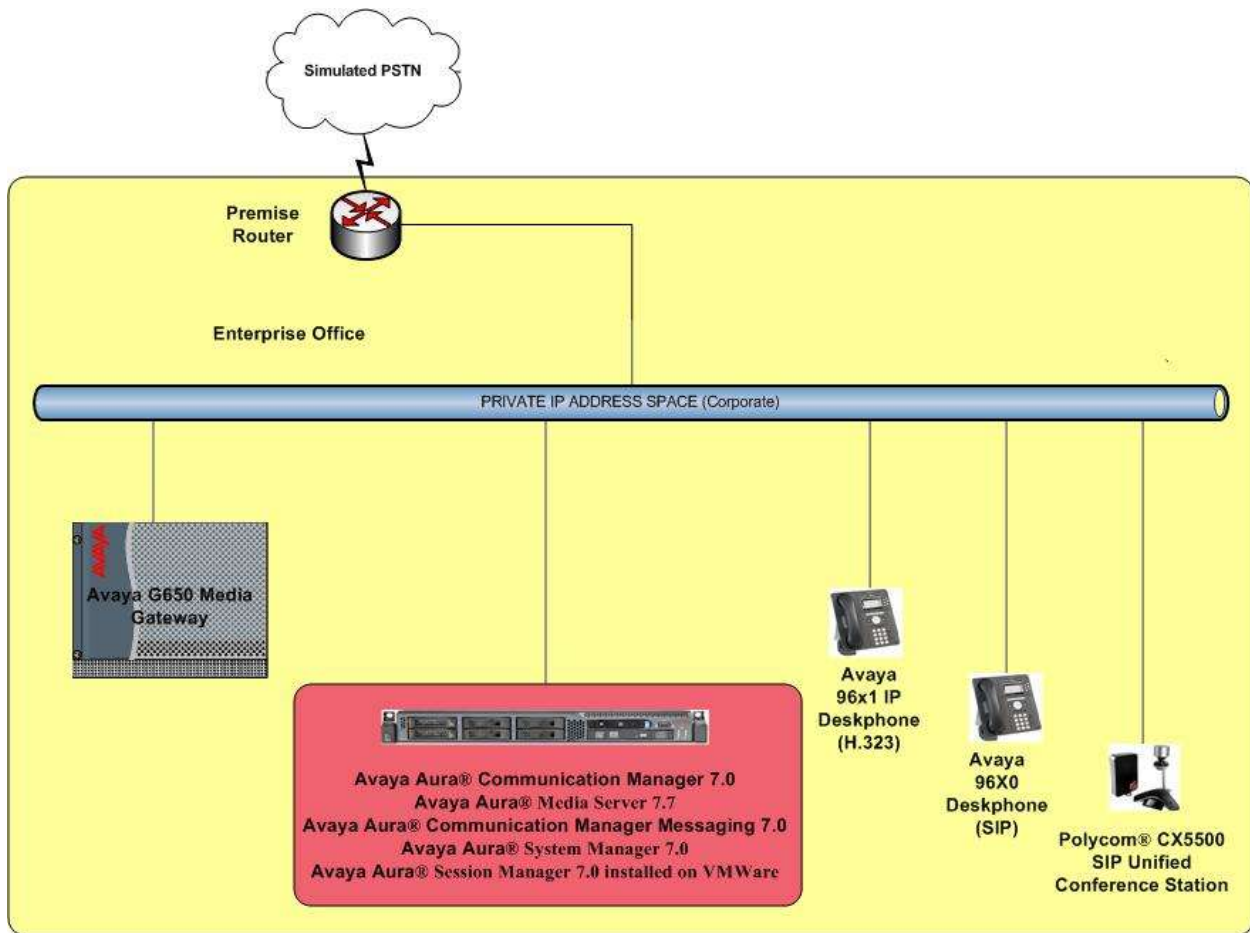


Figure 1: Polycom® CX5500 Unified Conference Station with Avaya Aura® Session Manager and Avaya Aura® Communication Manager

4. Equipment and Software Validated

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

Equipment	Software
Avaya Aura® System Manager installed on VMWare	R7.0 (7.0.0.0.16266)
Avaya Aura® Session Manager installed on VMWare	R7.0 (7.0.0.0.700007)
Avaya Aura® Communication Manager installed on VMWare	R7.0 (vcm-07.00.0.441.0)
Avaya Aura® Media Server installed on VMWare	R7.7 (v.7.7.0.226)
Avaya Aura® Communication Manager Messaging installed on VMWare	R7.0 (vcmm-07.00.0.441.0)
Avaya 96x1 IP Deskphone (H323)	R6.2.2313
Avaya 96x0 IP Deskphone (SIP)	R2.6.9.1
Polycom® CX5500 Unified Conference Station	UCS 5.4.0

5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities and corresponding Entity Links between Session Manager and Communication Manager/Communication Manager Messaging
- Define Communication Manager as administrable Entity (i.e., Managed Element).
- Application Sequence
- Add SIP Users

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL “**https://<ip-address>/SMGR**”, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials

Note that the fields modified in this section are for this reference configuration only; defaults are used for all other fields.

5.1. Add SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. To add a location, navigate to **Home→Elements→Routing→Domains** and click the **New** (not shown) button on the right.

The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *avaya.com*)
- **Type:** Set to *sip* (default)
- **Notes:** Descriptive text (optional)

Click **Commit**.

Name	Type	Notes
*avaya.com	sip	Used for Devconnect Testing

5.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, navigate to **Home→Elements→Routing→Locations** and click on the **New** (not shown) button on the right. The following screen will then be shown. Fill in the following:

Under **General**:

- **Name:** A descriptive name
- **Notes:** Descriptive text (optional)

Under **Location Pattern**:

- **IP Address Pattern:** A pattern used to logically identify the location
- **Notes:** Descriptive text (optional)

The screen below shows addition of the **Location_102** location used for Communication Manager and other entities. Similarly a location was defined for Session Manager. Click **Commit** to save the Location definition.

The screenshot displays the 'Location Details' configuration page in the Avaya Aura System Manager 7.0 interface. The page is divided into several sections:

- General:** Includes fields for 'Name' (set to 'Location_102') and 'Notes' (set to 'Entities in Subnet 102').
- Dial Plan Transparency in Survivable Mode:** Includes an 'Enabled' checkbox (unchecked), 'Listed Directory Number' field, and 'Associated CM SIP Entity' field.
- Overall Managed Bandwidth:** Includes 'Managed Bandwidth Units' (set to 'Kbit/sec'), 'Total Bandwidth' field, 'Multimedia Bandwidth' field, and a checked checkbox for 'Audio Calls Can Take Multimedia Bandwidth'.
- Per-Call Bandwidth Parameters:** Includes 'Maximum Multimedia Bandwidth (Intra-Location)' (2000 Kbit/Sec), 'Maximum Multimedia Bandwidth (Inter-Location)' (2000 Kbit/Sec), '* Minimum Multimedia Bandwidth' (64 Kbit/Sec), and '* Default Audio Bandwidth' (80 Kbit/Sec).
- Alarm Threshold:** Includes 'Overall Alarm Threshold' (80 %), 'Multimedia Alarm Threshold' (80 %), '* Latency before Overall Alarm Trigger' (5 Minutes), and '* Latency before Multimedia Alarm Trigger' (5 Minutes).
- Location Pattern:** A table with one item: 'IP Address Pattern' with the value '*10.64.102.*'.

The 'Commit' button is visible in the top right corner of the configuration area.

5.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager and Communication Manager. The screens below also show the corresponding Entity Links.

5.3.1. Session Manager Entity

To add a SIP Entity, navigate to **Home→Elements→Routing→SIP Entities**, and click on **New** (not shown) and configure as follows:

Under **General**:

- **Name:** Any descriptive name
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager
- **Type:** Select **Session Manager**
- **Location:** Select one of the locations defined previously
- **Time Zone:** Time zone for this location

Under **Port**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Listen Ports:** Port number on which the system listens for SIP requests
- **Protocol:** Transport protocol to be used to send SIP requests
- **Default Domain:** The domain used for the enterprise (e.g. **avaya.com**)

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The screenshot displays the Avaya System Manager 7.1.1 interface for configuring a SIP Entity. The left sidebar shows the navigation menu with options like Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, View Ranges, Routing Policies, Mail Patterns, Registrar Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and includes a 'Commit' button. The 'General' tab is active, showing fields for Name (SIPN), FQDN or IP Address (10.54.102.157), Type (Session Manager), Location (Session Manager), Outbound Proxy, Time Zone (America/Denver), and Credential name. Below this is the 'SIP Link Monitoring' section with a dropdown menu set to 'Use Session Manager Configuration'. The 'Entity Links' section features a table with columns: Name, SIP Entity 1, Protocol, SIP Entity 2, Port, Connection Policy, and Ready New Service. It lists two entries for SIPN, both using TCP and port 5060. The 'Listen Ports' section at the bottom has a table with columns: Listen Ports, Protocol, Default Domain, and Notes, showing a single entry for port 5060 using TCP and domain avaya.com.

5.3.2. Communication Manager Entity

The following screen displays the Communication Manager entity configured for this reference configuration.

AVAYA
Aura System Manager 7.0

Home / Elements / Routing / SIP Entities

SIP Entity Details

Commit Cancel

General

* Name: CH70Proc

* FQDN or IP Address: 10.64.102.150

Type: CH

Notes: CH 7.0 Proc, Ethernet

Adaptation: [v]

Location: location_102

Time Zone: America/Denver

* SIP Timer B/F (in seconds): 4

Credential name: [v]

Securable: [v]

Call Detail Recording: none

Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Supports Call Admission Control: [v]

Shared Bandwidth Manager: [v]

Primary Session Manager Bandwidth Association: [v]

Backup Session Manager Bandwidth Association: [v]

Entity Links

Override Port & Transport with DNS SRV: [v]

Add Remove

1 Item

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
* SM70_CH70Proc_50	SM70	TCP	* 5060	CH70Proc	* 5060	trusted	[v]

5.3.3. Communication Manager Messaging Entity

The following screen displays the Communication Manager Messaging entity configured for this reference configuration.

The screenshot displays the Avaya System Manager 7.2 interface. The left sidebar shows a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Mail Patterns, Regular Expressions, and Defaults. The main content area is titled "SIP Entity Details" and includes a "General" tab. The configuration fields are as follows:

- Name:** CM Messaging
- FQDN or IP Address:** 10.64.102.151
- Type:** Protocol Messaging
- Notes:** CM Messaging
- Adaptation:** (empty dropdown)
- Location:** Location_002
- Time Zone:** America/Denver
- SIP Timer R/T (in seconds):** 5
- Credential name:** (empty text field)
- Securable:** ☐
- Call Detail Recording:** agree
- Loop Detection Mode:** On
- Loop Count Threshold:** 5
- Loop Detection Interval (in msec):** 200
- SIP Link Monitoring:** Use Session Manager Configuration
- Supports Call Admission Control:** ☐
- Shared Bandwidth Manager:** ☐
- Primary Session Manager Bandwidth Association:** (empty dropdown)
- Backup Session Manager Bandwidth Association:** (empty dropdown)
- Override Port & Transport with DNS SRV:** ☐

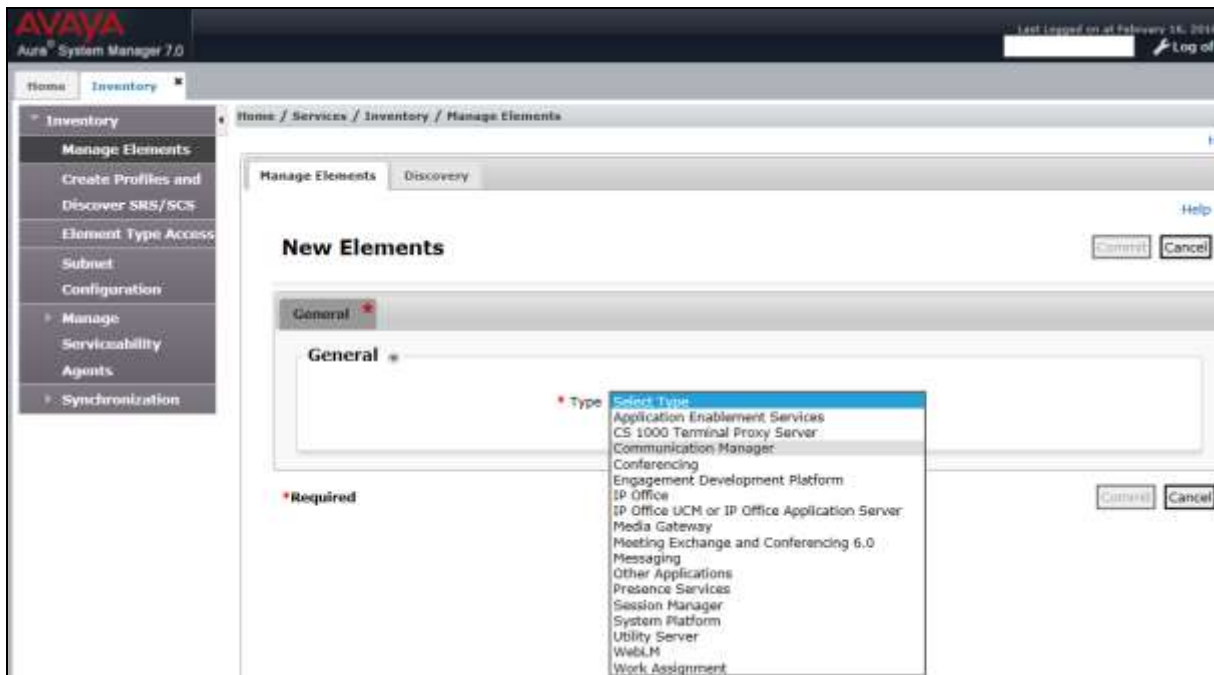
At the bottom, there is a table for "Entity Links" with columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Display Name Service. The table contains one entry:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Display Name Service
* SIP CM Messaging	SIP CM	SIP	* 5060	CM Messaging	* 5060	Trusted	

5.4. Define Communication Manager as a Managed Element

Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP users are added.

To define Communication Manager as a managed element, navigate to **Home→Services→Inventory→Manage Elements** on the left and click on the **New** (not shown) button on the right. In the **Type** field that is displayed, select *Communication Manager*.



In the **Add Communication Manager** screen, fill in the following fields as follows:

Under *General Attributes*:

- **Name:** Enter an identifier for Communication Manager
- **Hostname or IP Address:** Enter the IP address of the administration interface for Communication Manager
- **Login:** Enter the login used for administration access to Communication Manager
- **Authentication Type:** Select the **Password** button
- **Password:** Enter a valid password
- **Confirm Password:** This should match the password entered in the **Password** field above

Click **Commit** to save.

The screenshot shows the 'Add Communication Manager' configuration screen in Avaya Aura System Manager 7.0. The left sidebar contains a navigation menu with categories like 'Inventory', 'Manage Elements', 'Create Profiles and Discover SRS/SCS', 'Element Type Access', 'Subnet', 'Configuration', 'Manage', 'Serviceability', 'Agents', and 'Synchronization'. The main panel is titled 'Add Communication Manager' and has two tabs: 'General Attributes (0)' and 'SNMP Attributes (5)'. The 'General Attributes' tab is selected, showing the following fields:

- Name:** CM70
- Hostname or IP Address:** 10.64.102.150
- Login:** (empty)
- Authentication Type:** Password (selected), ASG Key (unselected)
- Password:** (masked with asterisks)
- Confirm Password:** (masked with asterisks)
- SSH Connection:** (checked)
- RSA SSH Fingerprint (Primary IP):** (empty)
- RSA SSH Fingerprint (Alternate IP):** (empty)

On the right side of the form, there are additional fields:

- Description:** Communication Manager
- Alternate IP Address:** (empty)
- Enable Notifications:** (unchecked)
- Port:** 5022
- Location:** (empty)
- Add to Communication Manager:** (checked)

At the bottom right of the form, there are three buttons: 'Commit', 'Clear', and 'Cancel'.

5.5. Add Application Sequence

Navigate to **Home**→**Elements**→**Session Manager**→**Application Configuration**→**Applications** and configure as follows:

- **Name:** Enter any descriptive name
- **SIP Entity:** Select the Communication Manager SIP Entity configured in **Section 5.3.2**
- **CM System for SIP Entity:** Select the system configured in **Section 5.4**

Click **Commit** to save the application configuration.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options like Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, and Device and Location Configuration. The main content area is titled 'Application Editor' and contains a form for creating a new application. The form fields are: Name (CM70), SIP Entity (CM70Procr), CM System for SIP Entity (CM70), and Description (CM 7.0). There are 'Commit' and 'Cancel' buttons at the top right of the form area.

Next, define the **Application Sequence** for Communication Manager as shown below.

The screenshot shows the Avaya Aura System Manager 7.0 interface for the 'Application Sequence Editor'. The left sidebar is the same as the previous screenshot. The main content area is titled 'Application Sequence Editor' and contains a form for creating a new application sequence. The form fields are: Name (CM70Sequencing) and Description (App. Sequencing with CM 7.0). Below the form is a table titled 'Applications in this Sequence' with columns: Sequence Order (first to last), Name, SIP Entity, Mandatory, and Description. The table contains one item: CM70, CM70Procr, and CM 7.0. Below the table is a section titled 'Available Applications' with a table containing one item: CM70, CM70Procr, and CM 7.0. There are 'Commit' and 'Cancel' buttons at the top right of the form area.

Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
<input type="checkbox"/>	CM70	CM70Procr	<input checked="" type="checkbox"/>	CM 7.0

Name	SIP Entity	Description
CM70	CM70Procr	CM 7.0

5.6. Add SIP Users

CX5500 was entered as a SIP user on Session Manager using the following steps. Navigate to **Home→Users→User Management→Manage Users** and configure as follows. This configuration is automatically synchronized with Communication Manger as verified in **Section 6.3**.

Enter values for the following required attributes for a SIP user in the **New User Profile** form:

- **Last Name:** Enter the last name of the user
- **First Name:** Enter the first name of the user
- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., *50071@avaya.com*)
- **Password:** Enter the password used to register with System Manager
- **Confirm Password:** Re-enter the password from above

The screenshot shows the 'New User Profile' form in the Avaya System Manager interface. The form is divided into several tabs: 'Identity', 'Communication Profile', 'Workday', and 'Contacts'. The 'Identity' tab is currently selected. The form contains the following fields and values:

- User Provisioning Rule:** A dropdown menu.
- Identity:**
 - Last Name:** SIP
 - Last Name (Latin Translation):** SIP
 - First Name:** 50071
 - First Name (Latin Translation):** 50071
 - Mobile Name:**
 - Description:** Polycom SIP Endpoint
 - Login Name:** 50071@avaya.com
 - Authentication Type:** Basic
 - Password:** [REDACTED]
 - Confirm Password:** [REDACTED]
 - Localized Display Name:** SIP-50071
 - Endpoint Display Name:** SIP-50071
 - Title:**
 - Language Preference:** English (United States)
 - Time Zone:** [REDACTED]

Click the **Communication Profile** tab and enter values for the following required fields:

- **Communication Profile Password:** Enter a valid password.
- **Confirm Password:** Make sure that it matches the password entered above

Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP* (default)
- **Fully Qualified Address:** Enter extension number and SIP domain

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.

The screenshot displays the 'New User Profile' form in the Avaya System Manager 7.0 interface. The form is divided into several sections: 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' section is active, showing fields for 'Communication Profile Password' and 'Confirm Password'. Below this, there is a 'Name' field with a dropdown menu set to 'Primary'. The 'Communication Address' section is also visible, showing a 'Type' dropdown set to 'Avaya SIP' and a 'Fully Qualified Address' field with the value '5001'. The form includes various buttons such as 'Add', 'Cancel', 'Save', and 'Cancel & Continue'.

In the **Session Manager Profile** section, specify the Session Manager entity configured in **Section 5.3.1** and assign the **Application Sequence** defined in **Section 5.5** to both the **Originating Sequence** and **Termination Sequence** fields. Additionally, set **Home Location** field to *Session Manager* configured in **Section 5.2**.

☒ **Session Manager Profile**

SIP Registration

* Primary Session Manager

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices

Block New Registration When Maximum Registrations Active?
☒

Primary	Secondary	Maximum
12	0	12

Application Sequences

Origination Sequence

Termination Sequence

Call Routing Settings

* Home Location

Conference Factory Set

Call History Settings

Enable Centralized Call History?
☐

In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager in **Section 5.4**
- **Profile Type:** Select ***Endpoint***
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager
- **Extension:** Enter extension number of SIP user
- **Template:** Select template for type of SIP phone which is set to **9621SIP_DEFAULT_CM_7_0** for CX5500

☒ **CM Endpoint Profile**

* System

CM70

* Profile Type

Endpoint

Use Existing Endpoints

☐

* Extension

50071

Endpoint Editor

* Template

9621SIP_DEFAULT_CM_7_0

Set Type

9621SIP

Security Code

Port

IP

Voice Mail Number

Preferred Handle

50071@avaya.com

Calculate Route Pattern

☐

Sip Trunk

aar

Enhanced Callr-Info display for 1-line phones

☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User

☒

Override Endpoint Name and Localized Name

☒

Allow H.323 and SIP Endpoint Dual Registration

☐

6. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the CX5500 as an Off-PBX Station (OPS) and configuring a SIP trunk between Communication Manager and Session Manager. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials. Note that the fields modified in this section are for this reference configuration only; defaults are used for all other fields.

6.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative. On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

display system-parameters customer-options		Page	1 of 11
OPTIONAL FEATURES			
G3 Version: V16	Software Package: Enterprise		
Location: 2	System ID (SID): 1		
Platform: 28	Module ID (MID): 1		
		USED	
Platform Maximum Ports:	6400	25	
Maximum Stations:	2400	10	
Maximum XMOBILE Stations:	2400	0	
Maximum Off-PBX Telephones - EC500:	9600	0	
Maximum Off-PBX Telephones - OPS:	9600	5	
Maximum Off-PBX Telephones - PBFMC:	9600	0	
Maximum Off-PBX Telephones - PVFMC:	9600	0	
Maximum Off-PBX Telephones - SCCAN:	0	0	
Maximum Survivable Processors:	313	0	
(NOTE: You must logoff & login to effect the permission changes.)			

On **Page 2** of the **system-parameters customer-options** form, verify that the number of SIP trunks supported by the system is sufficient.

display system-parameters customer-options	Page	2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	4000	0
Maximum Concurrently Registered IP Stations:	2400	2
Maximum Administered Remote Office Trunks:	4000	0
Maximum Concurrently Registered Remote Office Stations:	2400	0
Maximum Concurrently Registered IP eCons:	68	0
Max Concur Registered Unauthenticated H.323 Stations:	100	0
Maximum Video Capable Stations:	2400	0
Maximum Video Capable IP Softphones:	2400	0
Maximum Administered SIP Trunks:	4000	160
Maximum Administered Ad-hoc Video Conferencing Ports:	4000	0
Maximum Number of DS1 Boards with Echo Cancellation:	80	0
Maximum TN2501 VAL Boards:	10	0
Maximum Media Gateway VAL Sources:	50	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.)		

6.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for Session Manager (**ASM70**) and Media Server (**AMS70**). The host names will be used throughout the other configuration screens of Communication Manager.

change node-names ip	Page	1 of 2
IP NODE NAMES		
Name	IP Address	
default	0.0.0.0	
ASM70	10.64.102.157	
CMM70	10.64.102.151	
AMS70	10.64.102.158	
procr	10.64.102.150	
procr6	::	

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. The **IP Network Region** form also specifies the **Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 2) is specified in the SIP signaling group.

```

change ip-network-region 2                                     Page 1 of 20

                                IP NETWORK REGION

Region: 2
Location: 1      Authoritative Domain: avaya.com
Name: Main Network Region
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
      Codec Set: 2      Inter-region IP-IP Direct Audio: yes
      UDP Port Min: 2048      IP Audio Hairpinning? n
      UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
      Audio PHB Value: 46
      Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
      Audio 802.1p Priority: 6
      Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS      RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
      Keep-Alive Count: 5

```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to CX5500. The form is accessed via the **change ip-codec-set 2** command. Note that IP codec set 2 was specified in IP Network Region 2 shown above. The following form shows the list of codecs tested. The order of these codecs was changed to support the some of the codecs for reasons listed in **Section 2.2**.

```

change ip-codec-set 2                                     Page 1 of 2

                                IP Codec Set

Codec Set: 2

Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt   Size (ms)
1: G.711MU      n      2      20
2: G.711A      n      2      20
3: G.722-64K      2      20
4: iLBC      1      20-30
5:
6:
7:

```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as follows:

- **Group Type:** Set to *sip*
- **Transport Method:** Set to *tcp*
- **Near-end Node Name:** Set to *procr* node configured in this section
- **Far-end Node Name:** Set to *ASM70* node configured in this section
- **Far-end Network Region:** Set to network region configured in this section
- **Far-end Domain:** Set to *avaya.com* to match the Session Manager domain configured in **Section 5.1**
- Verify **Direct IP-IP Audio Connections** field is set to *y* for shuffling
- Verify **DTMF over IP** field is set to the default value of *rtp-payload* indicating DTMF transmission using RFC 2833

add signaling-group 2		Page 1 of 1
SIGNALING GROUP		
Group Number: 2	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: procr	Far-end Node Name: ASM70	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 2	
	Far-end Secondary Node Name:	
Far-end Domain: avaya.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
	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	Initial IP-IP Direct Media? n	
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6	

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the SIP Phones. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

add trunk-group 2		Page 1 of 21	
TRUNK GROUP			
Group Number: 2	Group Type: sip	CDR Reports: y	
Group Name: SIP Endpoints/CM Messaging	COR: 1	TN: 1	TAC: 102
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Member Assignment Method: auto	
		Signaling Group: 2	
		Number of Members: 15	

On **Page 3** of the **Trunk Group** form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

add trunk-group 2		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private			
		UI Treatment: service-provider	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	
		Modify Tandem Calling Number: no	
Show ANSWERED BY on Display? y			
DSN Term? n			

Configure the **Private Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with **5** and whose calls are routed over any trunk group, including SIP trunk group **2**, have the number sent to the far-end for display purposes.

change private-numbering 0		Page 1 of 2	
NUMBERING - PRIVATE FORMAT			
Ext Len	Ext Code	Trk Grp(s)	Private Prefix
5	33	10	5
5	58	10	5
5	5	2	5
5	600	10	5
		Total Administered: 4	
		Maximum Entries: 540	

6.3. Configure Signaling Group for Avaya Aura® Media Server

Another signaling group was created between Communication Manager and Media Server to provide media resources for IP telephony in parallel with Media Gateway G650 resource.

Following signaling group was created for this reference configuration:

- **Group Type:** Set to *sip*
- **Transport Method:** Set to *tcp*
- **Peer Detection Enable:** Set to *n*
- **Peer Server:** Set to *AMS*
- **Near-end Node Name:** Set to *procr* node shown in **Section 6.2**
- **Far-end Node Name:** Set to *AMS70* node configured in **Section 6.2**
- **Far-end Network Region:** Set to network region configured in **Section 6.2**

```
add signaling-group 3                                     Page 1 of 1
                                     SIGNALING GROUP

Group Number: 3                Group Type: sip
                               Transport Method: tcp

Peer Detection Enabled? n Peer Server: AMS

Near-end Node Name: procr      Far-end Node Name: AMS70
Near-end Listen Port: 5060     Far-end Listen Port: 5060
                               Far-end Network Region: 2

Far-end Domain: 10.64.102.158
```

6.4. Verify SIP Stations

Use the **display station** command to view each CX5500 SIP endpoint configured in **Section 5.6**.

```
display station 50071                                     Page 1 of 6
                                     STATION

Extension: 50071          Lock Messages? n          BCC: 0
Type: 9621SIP             Security Code:             TN: 1
Port: S00003              Coverage Path 1: 1          COR: 1
Name: 50071 SIP           Coverage Path 2:             COS: 1
                          Hunt-to Station:

STATION OPTIONS

Loss Group: 19            Time of Day Lock Table:

                          Message Lamp Ext: 40012

Display Language: english

Survivable COR: internal
Survivable Trunk Dest? y   IP SoftPhone? n

                          IP Video? n
```

Use the **display off-pbx-telephone station-mapping** to verify proper entry of CX5500 SIP station in Communication Manager.

display off-pbx-telephone station-mapping 50071							Page	1	of	3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION										
Station	Application	Dial	CC	Phone	Number	Trunk	Config	Dual		
Extension		Prefix				Selection	Set	Mode		
50071	OPS	-		50071		aar	1			

On **Page 2**, verify that the **Call Limit** matches the number of *call-appr* entries in the station form.

display off-pbx-telephone station-mapping 50071							Page	2	of	3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION										
Station	Appl	Call	Mapping	Calls	Bridged	Location				
Extension	Name	Limit	Mode	Allowed	Calls					
50071	OPS	3	both	all	none					

7. Configure Polycom® CX5500 Unified Conference Station

This section describes how to set up the CX5500 network and SIP interface along with authentication information to register with Session Manager. Note that the fields modified in this section are for this reference configuration only; defaults are used for all other fields.

7.1. Set the IP address used by CX5500

This section shows how to set the network IP address CX5500.

On the CX5500, push the **Settings** button and navigate to **Advanced**→**Administration Settings**→**Network Configuration**→**Network Interfaces**→**Ethernet Menu** and configure as follows (not shown):


- **DHCP:** *Disabled*
- **IP Address:** *010.080.130.071*
- **Subnet Mask:** *255.255.255.000*
- **IP Gateway:** *010.080.130.001*

7.2. Launch Web interface for CX5500

Open the web browser, and in the address field enter the CX5500 IP address as format *http://10.80.130.71* and the login page will appear as shown below. Select **Admin**, enter the default password and click **Submit**.




The following home page is displayed.

 **Polycom** | CX5500

HomeSimple SetupPreferencesSettingsDiagnosticsUtilities

You are here: Home



VIEWS
[Home](#)
[Simple Setup](#)

Home
Phone Information
Phone ModelCX5500
Part NumberCX5500 Rev:20
MAC Address00:E0:DB:40:64:29
IP Address10.80.130.71
UC Software Version5.4.0.
Device Software Version1.1.6.50142
Product Serial Number821404406429DA
Power Data Box Hardware Version003.8
Tabletop Hardware Version002.;

7.3. Configure the Lines for CX5500

Navigate to **Settings**→**Lines** and configure as follows:

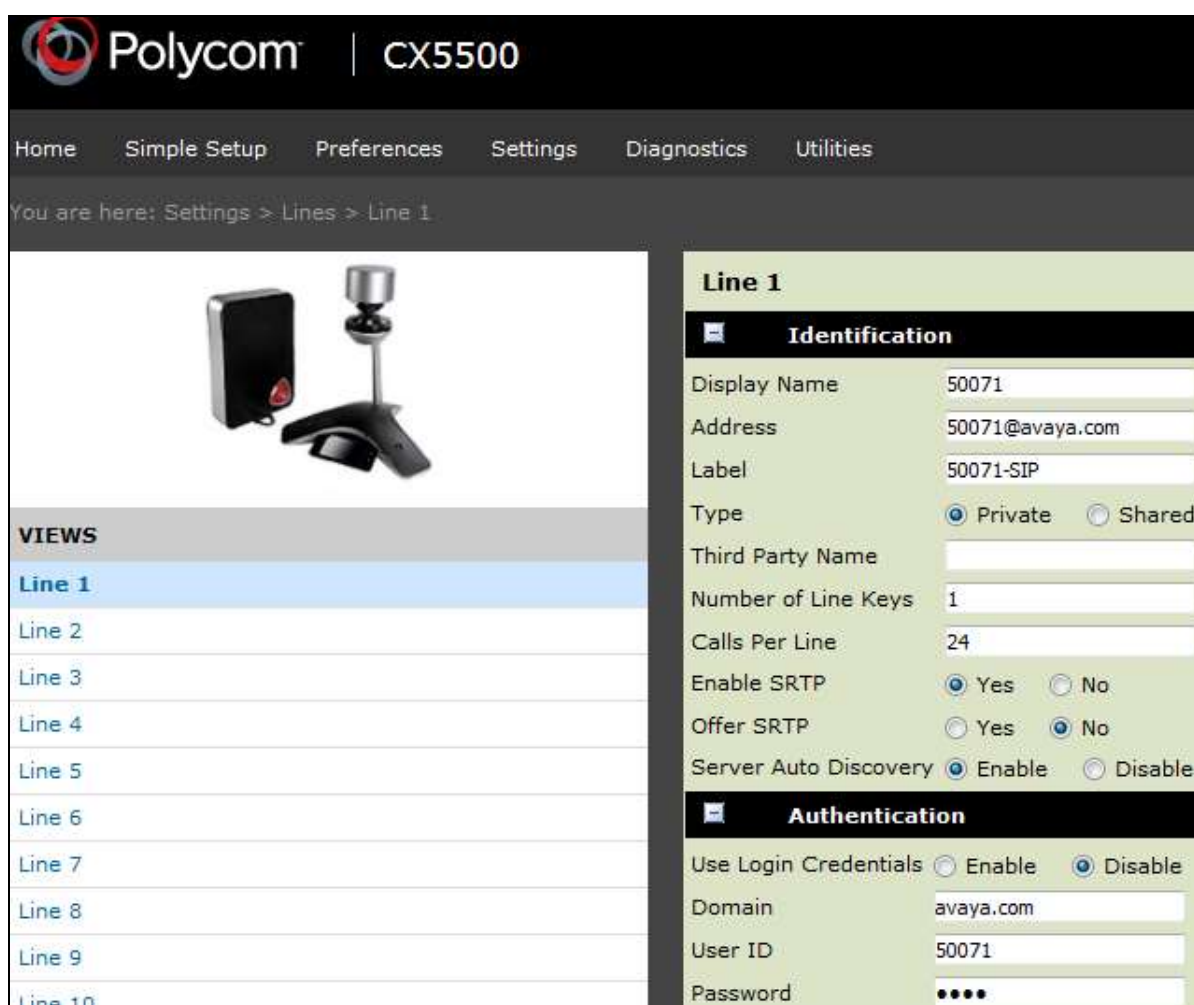
Under **Identification** section:

- **Display Name:** Set to any valid string
- **Address:** Set to the **Login Name** in **Section 5.6**
- **Label:** Set to any valid string

Under **Authentication** section:

- **Domain:** Set to the domain configured in **Section 5.1**
- **User ID:** Set to Extension of **Login Name** in **Section 5.6**
- **Password:** Set to **Communication Profile Password** field value configured in **Section 5.6**

Click **Save** (not shown)



The screenshot displays the Polycom CX5500 web interface. The top navigation bar includes links for Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. Below the navigation bar, a breadcrumb trail indicates the current location: Settings > Lines > Line 1. The main content area is divided into two sections. On the left, under the heading 'VIEWS', there is a list of lines from Line 1 to Line 10, with Line 1 selected and highlighted in blue. On the right, the configuration details for Line 1 are shown. This section is divided into two sub-sections: Identification and Authentication. The Identification section includes fields for Display Name (50071), Address (50071@avaya.com), Label (50071-SIP), Type (radio buttons for Private and Shared, with Private selected), Third Party Name, Number of Line Keys (1), Calls Per Line (24), Enable SRTP (radio buttons for Yes and No, with Yes selected), Offer SRTP (radio buttons for Yes and No, with No selected), and Server Auto Discovery (radio buttons for Enable and Disable, with Enable selected). The Authentication section includes fields for Use Login Credentials (radio buttons for Enable and Disable, with Disable selected), Domain (avaya.com), User ID (50071), and Password (represented by four dots).

7.4. SIP Settings

Navigate to **Settings**→**SIP** and configure as follows:

Under **Local Settings** section,

- Set **Digitmap Impossible Match** field to **2** to disable the automatic dial if the digits match in **Digitmap** field. This was done to enable Feature Access Codes to work properly

Under **Server1** section

- **Address:** Set to the IP address of Session Manager signaling interface
- **Port:** Set to **5060** for TCP
- **Transport:** Set to **TCPpreferred**

Click **Save** (not shown)

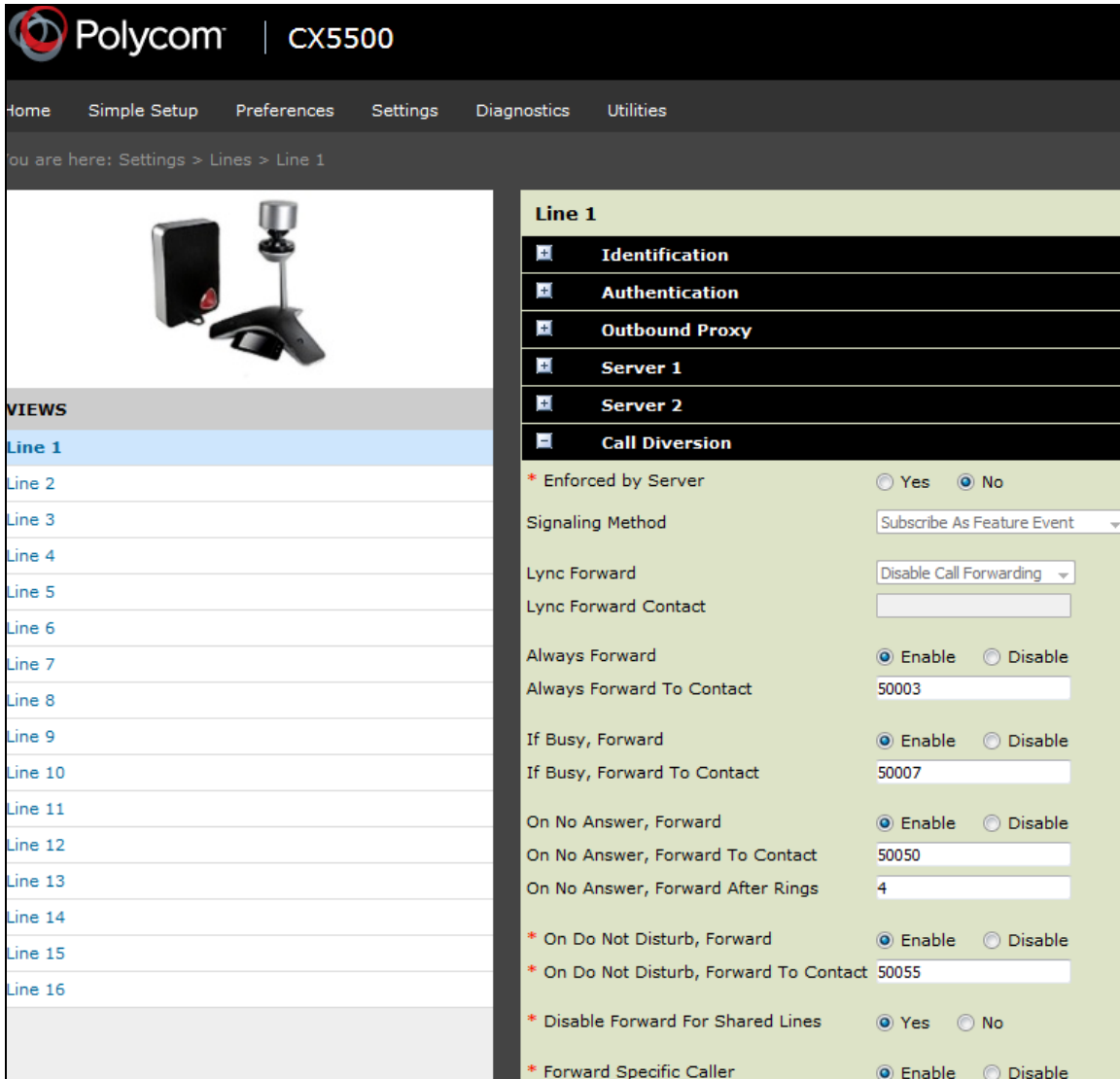
Note: The default local Digitmap configuration may require customization. Refer to **Section 10 [9]** for further details.

The screenshot displays the Polycom CX5500 web interface. The top navigation bar includes links for Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. The breadcrumb trail indicates the current location: Settings > SIP. On the left, a sidebar menu lists various configuration categories, with 'SIP' currently selected. The main content area is divided into three sections: Local Settings, Outbound Proxy, and Server 1. The Local Settings section contains fields for Local SIP Port (0), Calls Per Line Key (2), Enable Roaming buddies for (None), New SDP Type (Disable), Live Communication Server Support (Disable), Non Standard Line Seize (Enable), Disable Forward For Shared Line (Enable), Digitmap (a complex alphanumeric string), Digitmap Timeout (33333333), Remove End-of-Dial Marker (Enable), and Digitmap Impossible Match (2). The Outbound Proxy section is currently empty. The Server 1 section includes fields for Special Interop (Standard), Address (10.64.102.157), Port (5060), Transport (TCPpreferred), Expires (3600), Subscription Expires (3600), Register (Yes), Retry Timeout (0), Retry Maximum Count (3), and Line Seize Timeout (30).

SIP	
Local Settings	
* Local SIP Port	0
Calls Per Line Key	2
Enable Roaming buddies for	None
New SDP Type	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Live Communication Server Support	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
* Non Standard Line Seize	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Disable Forward For Shared Line	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Digitmap	[2-9]11 0T 011xxx.T [0-1][2-9]xxxxxxxxxx [2-9]xxxxxxxxxx [2-9]xxxT **x.T x.T
* Digitmap Timeout	33333333
Remove End-of-Dial Marker	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
* Digitmap Impossible Match	2
Outbound Proxy	
Server 1	
Special Interop	Standard
Address	10.64.102.157
Port	5060
Transport	TCPpreferred
Expires (s)	3600
Subscription Expires (s)	3600
Register	<input checked="" type="radio"/> Yes <input type="radio"/> No
Retry Timeout (ms)	0
Retry Maximum Count	3
Line Seize Timeout (s)	30

7.5. Local Call Forward Settings

Navigate to **Settings**→**Lines** and configure **Call Diversion** section as shown screen below. These features can also be enabled directly from the phone too.



The screenshot displays the Polycom CX5500 web interface. The top navigation bar includes links for Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. Below this, a breadcrumb trail indicates the current location: Settings > Lines > Line 1. On the left, a 'VIEWS' sidebar lists lines 1 through 16, with 'Line 1' selected. The main content area is titled 'Line 1' and contains several expandable sections: Identification, Authentication, Outbound Proxy, Server 1, Server 2, and Call Diversion. The 'Call Diversion' section is expanded, showing various forwarding options with radio buttons for 'Yes' or 'No' and text input fields for contact numbers. A small image of a Polycom phone is shown in the top left of the main content area.

Line 1	
Identification	
Authentication	
Outbound Proxy	
Server 1	
Server 2	
Call Diversion	
* Enforced by Server	<input type="radio"/> Yes <input checked="" type="radio"/> No
Signaling Method	Subscribe As Feature Event
Lync Forward	Disable Call Forwarding
Lync Forward Contact	
Always Forward	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Always Forward To Contact	50003
If Busy, Forward	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
If Busy, Forward To Contact	50007
On No Answer, Forward	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
On No Answer, Forward To Contact	50050
On No Answer, Forward After Rings	4
* On Do Not Disturb, Forward	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
* On Do Not Disturb, Forward To Contact	50055
* Disable Forward For Shared Lines	<input checked="" type="radio"/> Yes <input type="radio"/> No
* Forward Specific Caller	<input checked="" type="radio"/> Enable <input type="radio"/> Disable

7.6. Audio Codec Settings

Navigate to **Settings**→**Codec Priorities** and configure as shown below. The codecs shown in the **In use** column were tested in this reference configuration. The priority can be changed by moving the codecs up or down the order.

The screenshot displays the Polycom CX5500 web interface. The top navigation bar includes links for Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. Below this, a breadcrumb trail indicates the current location: Settings > Audio Codec Priority. The main content area is titled "Audio Codec Priority" and is divided into two columns: "Unused" and "In use".

Unused:

- LBC (13.33 kbps)
- G.722.1 (16 kbps)
- G.722.1 (24 kbps)
- G.722.1C (24 kbps)
- G.722.1C (32 kbps)
- Siren14 (24 kbps)
- Siren14 (32 kbps)
- Siren22 (32 kbps)
- Siren22 (48 kbps)
- G.719 (32 kbps)
- G.719 (48 kbps)

In use:

- G.711Mu
- G.729AB
- G.711A
- G.722
- LBC (15.2 kbps)

Note:
Only codecs with a white background are supported on this platform.

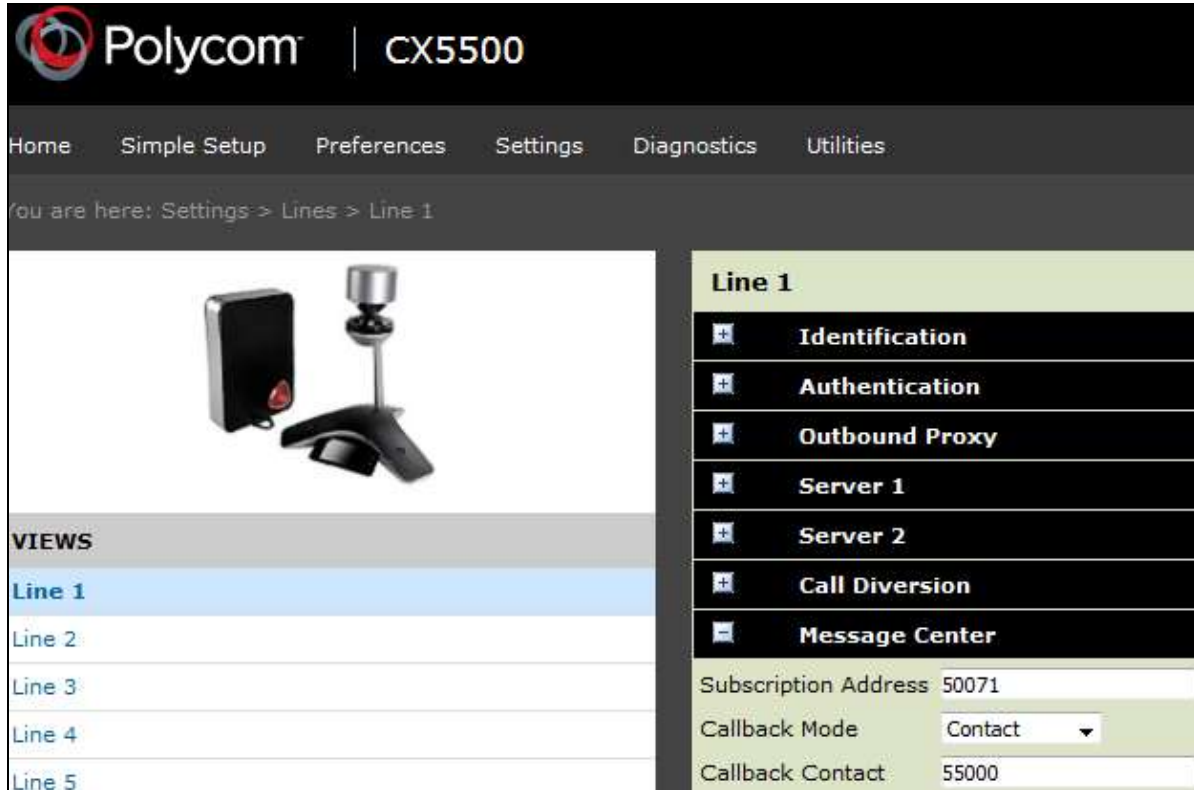
On the left side of the interface, there is a "VIEWS" section with links for Device, Set Date/Time, Logging, and Applications. Above this section is an image of the CX5500 device.

7.7. Voice Mail Setting

Navigate to **Settings**→**Lines** and configure **Message Center** section as follows:

- **Subscription Address:** Set to the Authentication ID field value **Section 7.3**
- **Callback Mode:** Set to the ***Contact***
- **Callback Contact:** Set to voicemail messaging Pilot number

Click **Save** (not shown)

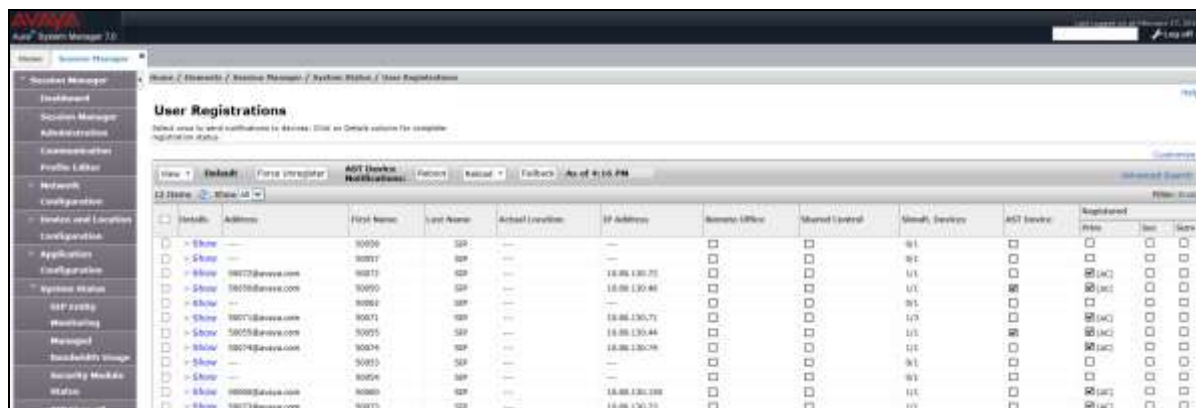


The screenshot displays the Polycom CX5500 web interface. The top navigation bar includes links for Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. Below this, a breadcrumb trail indicates the current location: Settings > Lines > Line 1. The main content area is divided into two sections. On the left, under the heading 'VIEWS', there is a list of lines: Line 1 (highlighted in blue), Line 2, Line 3, Line 4, and Line 5. Above this list is an image of a Polycom CX5500 device. On the right, the 'Line 1' configuration panel is shown. It contains a list of expandable sections: Identification, Authentication, Outbound Proxy, Server 1, Server 2, Call Diversion, and Message Center. The 'Message Center' section is currently expanded, revealing three configuration fields: 'Subscription Address' set to '50071', 'Callback Mode' set to 'Contact' (via a dropdown menu), and 'Callback Contact' set to '55000'.

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and Communication Manager with CX5500.

- Navigate to **Home→Elements→Session Manager→System Status→User Registrations** to verify that CX5500 is registered with Session Manager. The following screen shows the registered SIP users with Session Manager:



Details	Address	First Name	Last Name	Actual Location	IP Address	Business Office	Shared Control	Shared Device	ACD Service	Registered	Status
Show	10000	10000	SIP	---	---	---	---	---	---	---	---
Show	10001	10001	SIP	---	---	---	---	---	---	---	---
Show	10002	10002	SIP	---	---	---	---	---	---	---	---
Show	10003	10003	SIP	---	---	---	---	---	---	---	---
Show	10004	10004	SIP	---	---	---	---	---	---	---	---
Show	10005	10005	SIP	---	---	---	---	---	---	---	---
Show	10006	10006	SIP	---	---	---	---	---	---	---	---
Show	10007	10007	SIP	---	---	---	---	---	---	---	---
Show	10008	10008	SIP	---	---	---	---	---	---	---	---
Show	10009	10009	SIP	---	---	---	---	---	---	---	---
Show	10010	10010	SIP	---	---	---	---	---	---	---	---
Show	10011	10011	SIP	---	---	---	---	---	---	---	---

- Verify that basic calls can be made from and to CX5500 and another telephone registered with Communication Manager.

9. Conclusion

These Application Notes describe the configuration steps required for Polycom® CX5500 Unified Conference Station to successfully interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature and serviceability test cases were completed with the exceptions noted in **Section 2.2**.

10. Additional References

This section references the product documentation available at support.avaya.com relevant to these Application Notes.

- [1] [Deploying Avaya Aura® System Manager, Release 7.0, November 2015](#)
- [2] [Administering Avaya Aura® System Manager, Release 7.0, January 2016](#)
- [3] [Deploying Avaya Aura® Session Manager on VMWare, Release 7.0, August 2015](#)
- [4] [Administering Avaya Aura® Session Manager, Release 7.0, August 2015](#)
- [5] [Deploying Avaya Aura® Communication Manager in Virtualized Environment, Release 7.0, August 2015](#)
- [6] [Deploying and Updating Avaya Aura® Media Server Appliance, Release 7.7, October 2015](#)
- [7] [Implementing Avaya Aura® Media Server, Release 7.7, January 2016](#)
- [8] [Deploying Avaya Aura® Communication Manager Messaging, Release 7.0, September 2015](#)
- [9] Polycom CX5500 Conference Phone technical product documentation is available at <http://support.polycom.com/PolycomService/support/us/support/voice/cx/cx5500.html>

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