



Avaya Solution & Interoperability Test Lab

Application Notes for Polycom SoundPoint® IP 670 SIP Phone with Avaya Aura™ Session Manager 6.0 and Avaya Aura™ Communication Manager 6.0 - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate a Polycom SoundPoint® IP 670 SIP Phone with a SIP infrastructure consisting of Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager configured as an Evolution Server. During compliance testing, the SoundPoint® IP 670 SIP Phone successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

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 steps required to integrate a Polycom SoundPoint® IP 670 SIP Phone with a SIP infrastructure consisting of Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager configured as an Evolution Server. During compliance testing, the SoundPoint® IP 670 SIP Phone successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

These Application Notes assume that Communication Manager and Session Manager are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult the appropriate document in the reference section at the end of this document.

1.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the SoundPoint IP 670 SIP Phone with Session Manager.
- Calls between SoundPoint IP 670 SIP phones and Avaya SIP, H.323, and digital stations.
- G.711, G.729A, and G.722 codec support.
- Proper recognition of DTMF tones by navigating voicemail menus.
- Proper operation of voicemail with Message Waiting Indication (MWI).
- Basic telephony features including Hold, Transfer, and Conference.
- Extended telephony features using Communication Manager Feature Name Extensions (FNEs) such as Call Forwarding, Call Pickup, and Send All Calls.
- Proper system recovery after a SoundPoint IP 670 SIP phone restart and loss of IP connectivity.
- PC/laptop connectivity to Ethernet jack on phone.

1.2. Support

For technical support on the SoundPoint IP 670 SIP Phone contact Polycom Support through their website at <http://www.polycom.com/support/>.

In addition, additional support information may be obtained through the knowledge base available at

http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html.

2. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following Avaya products:

- Avaya Aura™ Communication Manager running on an Avaya S8800 Server with a G650 Media Gateway. Communication Manager was configured as an Evolution Server.
- Avaya Aura™ Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura™ System Manager used to configure Session Manager.
- Avaya Modular Messaging providing voice mail service for the SIP endpoints.

In addition, two Polycom SoundPoint IP 670 SIP Phones registered with Session Manager and were configured as Off-PBX Stations (OPS) on the Communication Manager.

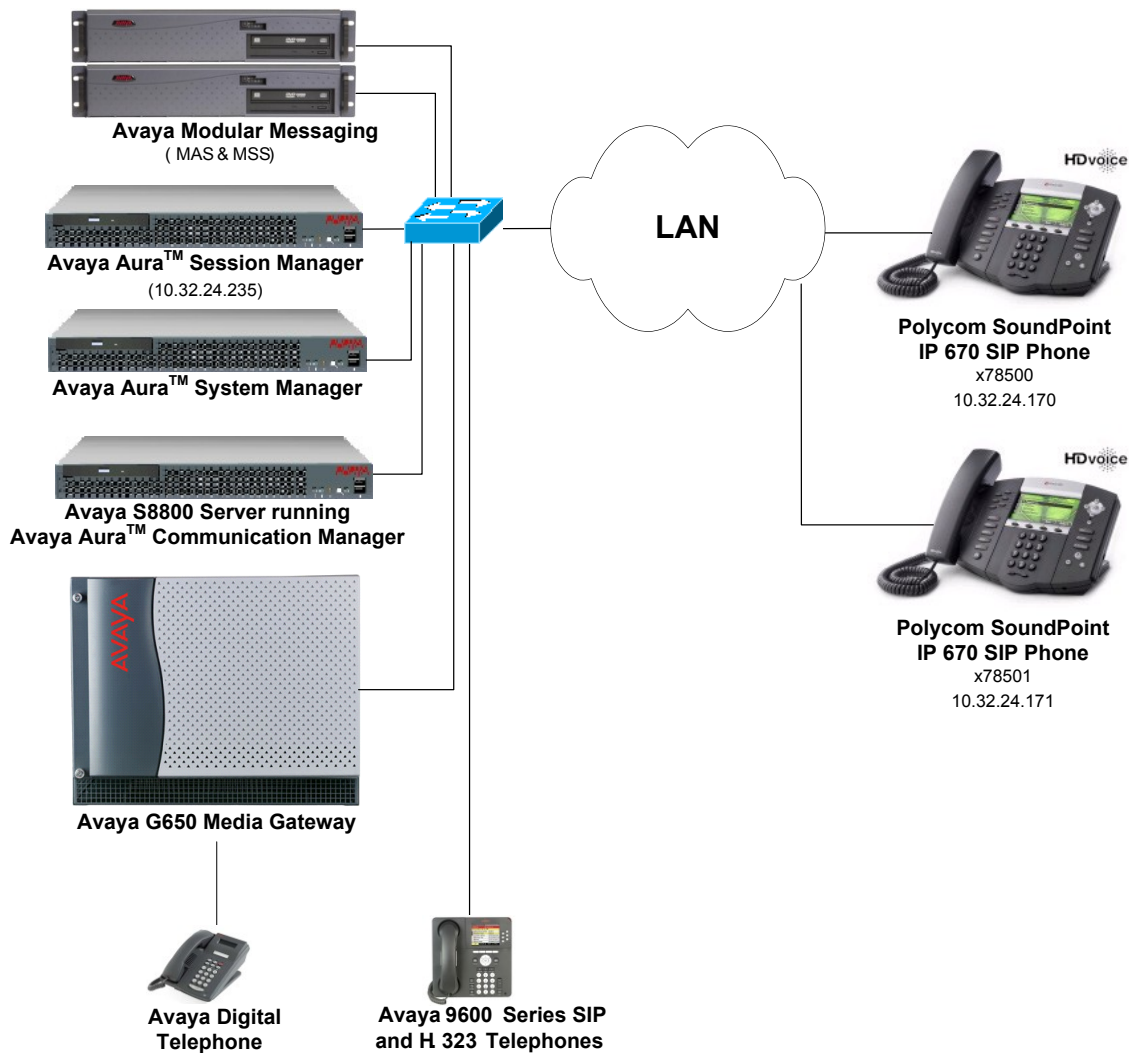


Figure 1: Avaya SIP Network with Polycom SoundPoint IP 670 SIP Phones

2.1. SIP Call Flows

The Polycom SoundPoint IP 670 SIP Phone originates a call by sending a call request (SIP INVITE message) to Session Manager, which then routes the call over a SIP trunk to the Communication Manager for origination services. If the call is destined for another local SIP phone, Communication Manager routes the call back over the SIP trunk to Session Manager for delivery to the destination SIP phone. If the call is destined for an H.323 or digital telephone, Communication Manager routes the call to the H.323 or digital endpoint.

For a call arriving at Communication Manager that is destined for one of the SoundPoint IP 670 SIP Phones, Communication Manager routes the call over the SIP trunk to Session Manager for delivery to the SoundPoint IP 670 SIP Phones.

3. Equipment and Software Validated

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

Hardware Component	Version
Avaya Aura™ Communication Manager on Avaya S8800 Servers and G650 Media Gateway	6.0 Service Pack 1
Avaya Aura™ Session Manager	6.0 (6.0.0.0.600020)
Avaya Aura™ System Manager	6.0 (6.0.0.0.556-3.0.6.1)
Avaya Modular Messaging	5.2
Avaya 9600 Series IP Telephones	3.110b (H.323) 2.6 (SIP)
Avaya Digital Telephones	--
Polycom SoundPoint IP 670 SIP Phone	3.2.3.1734

4. Configure Avaya Aura™ Communication Manager

This section describes the steps for configuring the SoundPoint IP 670 SIP Phone as an Off-PBX Station (OPS) and configuring a SIP trunk between the Communication Manager and Session Manager. **Section 4.3** covers the station configuration for the SoundPoint IP 670 SIP Phones. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

4.1. 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: 67000 350
                                Maximum Stations: 41000 197
                                Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 36000 0
Maximum Off-PBX Telephones - OPS: 41000 36
Maximum Off-PBX Telephones - PBFMC: 36000 0
Maximum Off-PBX Telephones - PVFMC: 36000 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:	12000	60
	Maximum Concurrently Registered IP Stations:	18000	13
	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	70
	Maximum Administered Ad-hoc Video Conferencing Ports:	24000	0
	Maximum Number of DS1 Boards with Echo Cancellation:	522	0
	Maximum TN2501 VAL Boards:	128	1
	Maximum Media Gateway VAL Sources:	250	0
	Maximum TN2602 Boards with 80 VoIP Channels:	128	0
	Maximum TN2602 Boards with 320 VoIP Channels:	128	0
	Maximum Number of Expanded Meet-me Conference Ports:	300	0
(NOTE: You must logoff & login to effect the permission changes.)			

4.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the S8800 Server processor, the C-LAN board in the G650 Media Gateway, and signaling interface for Session Manager. 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			
Gateway001	10.32.24.1			
ModMsg	192.50.10.45			
clancrm	10.32.24.20			
default	0.0.0.0			
devcon-asm	10.32.24.235			
medprocrm	10.32.24.21			
procr	10.32.24.10			
procr6	::			
(8 of 8 administered node-names were displayed)				
Use 'list node-names' command to see all the administered node-names				
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name				

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 G650 Media Gateway. The **IP Network Region** form also specifies the **IP 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 '1') is specified in the SIP signaling group.

```

change ip-network-region 1                                     Page 1 of 20

                                IP NETWORK REGION

Region: 1
Location: 1          Authoritative Domain: avaya.com
Name:
MEDIA PARAMETERS                      Intra-region IP-IP Direct Audio: yes
      Codec Set: 1                      Inter-region IP-IP Direct Audio: yes
      UDP Port Min: 2048                  IP Audio Hairpinning? y
      UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS
      Call Control PHB Value: 34
      Audio PHB Value: 46
      Video PHB Value: 26
802.1P/Q PARAMETERS
      Call Control 802.1p Priority: 7
      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 the SoundPoint IP 670 SIP Phones. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. However, the **IP Codec Set** form may specify multiple codecs, including G.711, G.729A, and G.722, which are supported by the SoundPoint IP 670 SIP Phones.

```

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:
3:
4:
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:

- Set the **Group Type** field to *sip*.
 - Set the **IMS Enabled** field to *n*.
 - The **Transport Method** field was set to *tcp*.
 - Specify the C-LAN board and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
 - Ensure that the TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
 - The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
 - Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
 - The **Direct IP-IP Audio Connections** field was enabled on this form.
 - The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 50		Page 1 of 1
SIGNALING GROUP		
Group Number: 50 IMS Enabled? n Q-SIP? n IP Video? n Peer Detection Enabled? y Group Type: sip Transport Method: tcp SIP Enabled LSP? n Enforce SIPS URI for SRTP? y Peer Server: SM		
Near-end Node Name: clancrm Near-end Listen Port: 5060 Far-end Node Name: devcon-asm Far-end Listen Port: 5060 Far-end Network Region: 1		
Far-end Domain: avaya.com		
Incoming Dialog Loopbacks: eliminate DTMF over IP: rtp-payload Session Establishment Timer(min): 3 Enable Layer 3 Test? n H.323 Station Outgoing Direct Media? n Bypass If IP Threshold Exceeded? n RFC 3389 Comfort Noise? n Direct IP-IP Audio Connections? y IP Audio Hairpinning? n Initial IP-IP 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 50		Page 1 of 21	
TRUNK GROUP			
Group Number: 50	Group Type: sip	CDR Reports: y	
Group Name: To devcon-asm	COR: 1	TN: 1	TAC: 1050
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
	Member Assignment Method: auto		
	Signaling Group: 50		
	Number of Members: 10		

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

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 '7' and whose calls are routed over any trunk group, including SIP trunk group "50", have the number sent to the far-end for display purposes.

change private-numbering 0		Page 1 of 2	
NUMBERING - PRIVATE FORMAT			
Ext	Ext	Trk	Private
Len	Code	Grp (s)	Prefix
5	7		
			Total
			Len
			5
Total Administered: 1			
Maximum Entries: 540			

4.3. Configure Stations

Use the **add station** command to add a station for each SoundPoint IP 670 SIP Phone to be supported. Use *9630SIP* for the **Station Type** and include the **Coverage Path** for voice mail, if applicable. The **Name** field is optional. Use the default values for the other fields on **Page 1**. The SIP station can also be configured automatically by System Manager as described in **Section 5.7**.

add station 78500		Page 1 of 6
STATION		
Extension: 78500	Lock Messages? n	BCC: 0
Type: 9630SIP	Security Code:	TN: 1
Port: IP	Coverage Path 1: 20	COR: 1
Name: Polycom 78500	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 19		
	Message Lamp Ext: 78500	
Display Language: english	Button Modules: 0	
Survivable COR: internal		
Survivable Trunk Dest? y	IP SoftPhone? n	
	IP Video? n	

On **Page 2**, set the **MWI Served User Type** field to the appropriate value to allow MWI notifications to be sent to the SoundPoint IP 670 SIP Phone.

add station 78500		Page 2 of 6
STATION		
FEATURE OPTIONS		
LWC Reception: spe		
LWC Activation? y	Coverage Msg Retrieval? y	
	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? n		
Active Station Ringing: single		
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
	EC500 State: enabled	
MWI Served User Type: qsig-mwi		
	Coverage After Forwarding? s	
	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 78500	Always Use? n IP Audio Hairpinning? n	

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extensions (e.g., 78500) to the same extension configured in System Manager. Enter the field values shown. For the sample configuration, the **Trunk Selection** field is set to *aar* so that AAR call routing is used to route calls to Session Manager. AAR call routing configuration is not shown in these Application Notes. The **Configuration Set** value can reference a set that has the default settings. This form is configured automatically when the station type ends with “SIP”, for example, 9630SIP.

change off-pbx-telephone station-mapping 78500							Page 1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
78500	OPS	-		78500	aar	1	

On **Page 2**, change the **Call Limit** to match the number of *call-appr* entries in the station form. Also, verify that **Mapping Mode** is set to *both* (the default value for a newly added station).

change off-pbx-telephone station-mapping 78500							Page 2 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Appl Name	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location	
78500	OPS	3	both	all	none		

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 corresponding to Session Manager and Communication Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Application Sequence
- Define Communication Manager as Administrable Entity (i.e., Managed Element)
- Session Manager, corresponding to the Avaya Aura™ Session Manager Server to be managed by Avaya Aura™ System Manager
- SIP Users

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

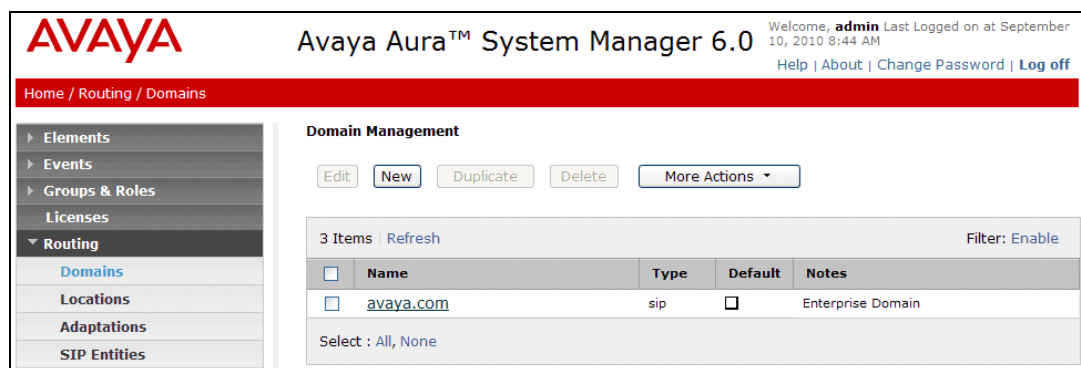
5.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button on the right. The following screen will then be shown. Fill in the following:

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

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.



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, select **Locations** on the left and click on the **New** button (not shown) 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 *BR-DevConnect* location, which includes the Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager. Click **Commit** to save the Location definition.

The screenshot displays the Avaya Aura™ System Manager 6.0 web interface. The top navigation bar includes the Avaya logo, the product name, and user information: 'Welcome, admin Last Logged on at September 10, 2010 8:44 AM'. A red breadcrumb trail shows the path: 'Home / Routing / Locations / Location Details'. On the left, a sidebar menu lists various system components, with 'Routing' expanded and 'Locations' selected. The main content area is titled 'Location Details' and contains two sections: 'General' and 'Location Pattern'. In the 'General' section, the 'Name' field is populated with 'BR-DevConnect', and the 'Managed Bandwidth' is set to 80 Kbit/sec. The 'Location Pattern' section shows a table with one item, '10.32.24.*', under the 'IP Address Pattern' column. At the bottom, a red asterisk indicates 'Input Required' for the 'Name' field. The interface includes 'Commit' and 'Cancel' buttons at the top right and bottom right.

AVAYA Avaya Aura™ System Manager 6.0 Welcome, **admin** Last Logged on at September 10, 2010 8:44 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Locations / Location Details

Location Details [Commit](#) [Cancel](#)

General

* **Name:**

Notes:

Managed Bandwidth: [Kbit/sec](#)

* **Average Bandwidth per Call:** [Kbit/sec](#)

Location Pattern

[Add](#) [Remove](#)

1 Item [Refresh](#) Filter: [Enable](#)

☐	IP Address Pattern	Notes
☐	* 10.32.24.*	

Select : [All](#), [None](#)

* **Input Required** [Commit](#) [Cancel](#)

Help

[Help for Locations Details fields](#)

[Help for Committing configuration changes](#)

5.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager and the C-LAN in the G650 Media Gateway.

5.3.1. Avaya Aura™ Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A 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.

The screenshot displays the Avaya Aura System Manager 6.0 web interface. The top header includes the Avaya logo, the product name 'Avaya Aura™ System Manager 6.0', and a user status bar indicating 'Welcome, admin' and 'Last Logged on at September 10, 2010 8:44 AM'. Navigation links for 'Help', 'About', 'Change Password', and 'Log off' are present. A red breadcrumb trail shows the path: 'Home / Routing / SIP Entities / SIP Entity Details'. On the left, a sidebar menu lists various configuration categories, with 'Routing' expanded to show 'SIP Entities' as the active selection. The main content area is titled 'SIP Entity Details' and features a 'General' tab. The form contains several fields: 'Name' (set to 'devcon-asm'), 'FQDN or IP Address' (set to '10.32.24.235'), 'Type' (set to 'Session Manager'), 'Notes' (empty), 'Location' (set to 'BR-DevConnect'), 'Outbound Proxy' (empty), 'Time Zone' (set to 'America/New_York'), and 'Credential name' (empty). At the bottom, the 'SIP Link Monitoring' section has a dropdown set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located in the top right corner of the form area.

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

- **Port:** 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 shows a web interface for configuring SIP ports. At the top, there's a 'Port' header with 'Add' and 'Remove' buttons. Below it, a table lists 4 items. The first row is highlighted with a red box. The table has columns: Port, Protocol, Default Domain, and Notes. The first row shows Port 5060, Protocol TCP, and Default Domain avaya.com. Below the table, there's a 'Select' dropdown showing 'All, None (0 of 4 Selected)'. At the bottom, there's a '* Input Required' message and 'Commit' and 'Cancel' buttons.

Port	Protocol	Default Domain	Notes
5060	TCP	avaya.com	
5060	UDP	avaya.com	
5061	TLS	avaya.com	
5070	TCP	avocs.contoso.com	

5.3.2. Avaya Aura™ Communication Manager

A SIP Entity must be added for the Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., C-LAN board) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

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



- ▶ Elements
- ▶ Events
- ▶ Groups & Roles
- Licenses
- ▼ **Routing**
 - Domains
 - Locations
 - Adaptations
 - SIP Entities**
 - Entity Links
 - Time Ranges
 - Routing Policies
 - Dial Patterns
 - Regular Expressions
 - Defaults
- ▶ Security
- ▶ System Manager Data
- ▶ Users

Help

[Help for SIP Entity Details fields](#)
[Help for Committing configuration changes](#)

SIP Entity Details

[Commit](#) [Cancel](#)

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

SIP Link Monitoring

SIP Link Monitoring:

Entity Links

[Add](#) [Remove](#)

0 Items [Refresh](#)

Filter: [Enable](#)

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
--	--------------	----------	------	--------------	------	---------

* Input Required

[Commit](#) [Cancel](#)

5.4. Add Entity Link

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *devcon13 Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in Section 5.3.2 will be denied.*

Click **Commit** to save the Entity Link definition.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at August 31, 2010 1:45 PM

Help | About | Change Password | Log off

Home / Routing / Entity Links

Entity Links

Commit Cancel

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* devcon13 Link	* devcon-asm	TCP	* 5060	* devcon13	* 5060	☒	

* Input Required

Commit Cancel

5.5. Define Communication Manager as 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, select **Elements→Inventory→Manage Elements** on the left and click on the **New** button (not shown) on the right. In the **Application Type** field that is displayed, select **CM**.

In the **New CM Instance** screen, fill in the following fields as follows:

Under *Application*:

- **Name:** Enter an identifier for Communication Manager.
- **Type:** Select *CM* from the drop-down field.
- **Node:** Enter the IP address of the administration interface for Communication Manager.

Under *Attributes*:

- **Login / Password:** Enter the login and password used for administration access.
- **Is SSH Connection:** Enable SSH access.
- **Port:** Enter the port number for SSH administration access (5022).

Defaults can be used for the remaining fields. Click **Commit** to save the settings.

The screenshot shows the Avaya Aura System Manager 6.0 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura™ System Manager 6.0', and a user status bar indicating 'Welcome, admin' and 'Last Logged on at August 31, 2010 1:45 PM'. Below the navigation bar is a red breadcrumb trail: 'Home / Elements / Application Management / Applications / Applications Details'. A left-hand sidebar contains a tree view of system elements, with 'Application Management' expanded. The main content area is titled 'New CM Instance' and features 'Commit' and 'Cancel' buttons. The configuration is organized into sections: 'Application' (with fields for Name: 'devcon13-CM-ES', Type: 'CM', Description: 'devcon13 CM ES', and Node: '10.32.24.10'), 'Port', 'Access Point', 'SNMP Attributes' (with Version set to 'None'), and 'Attributes' (with fields for Login: 'init', Password, Confirm Password, Is SSH Connection checked, and Port: '5022').

5.6. Add Application Sequence

To define an application for Communication Manager, navigate to **Elements**→**Session Manager**→**Application Configuration**→**Applications** on the left and select **New** button (not shown) on the right. Fill in the following fields:

- **Name:** Enter name for application.
- **SIP Entity:** Select the Communication Manager SIP entity.
- **CM System for SIP Entity** Select the Communication Manager managed element.

Click **Commit** to save the Application definition.

The screenshot shows the Avaya Aura System Manager 6.0 interface. The top header includes the Avaya logo, the product name "Avaya Aura™ System Manager 6.0", and a user status bar indicating "Welcome, admin" and "Last Logged on at August 31, 2010 1:45 PM". A navigation breadcrumb trail reads "Home / Elements / Session Manager / Application Configuration / Application Editor".

On the left is a sidebar menu with a tree structure. The "Session Manager" section is expanded, showing sub-items like "Dashboard", "Session Manager Administration", "Communication Profile Editor", "Network Configuration", "Device and Location Configuration", and "Application Configuration". The "Application Configuration" section is further expanded to show "Applications".

The main content area is titled "Application Editor" and contains the following fields:

- Name:** A text input field containing "DEVCON-APP".
- SIP Entity:** A dropdown menu showing "devcon13".
- CM System for SIP Entity:** A dropdown menu showing "devcon13-CM-ES", with a "Refresh" button and a link "View/Add CM Systems".
- Description:** A text input field.

Below these fields is a section titled "Application Attributes (optional)" which contains a table with two columns: "Name" and "Value".

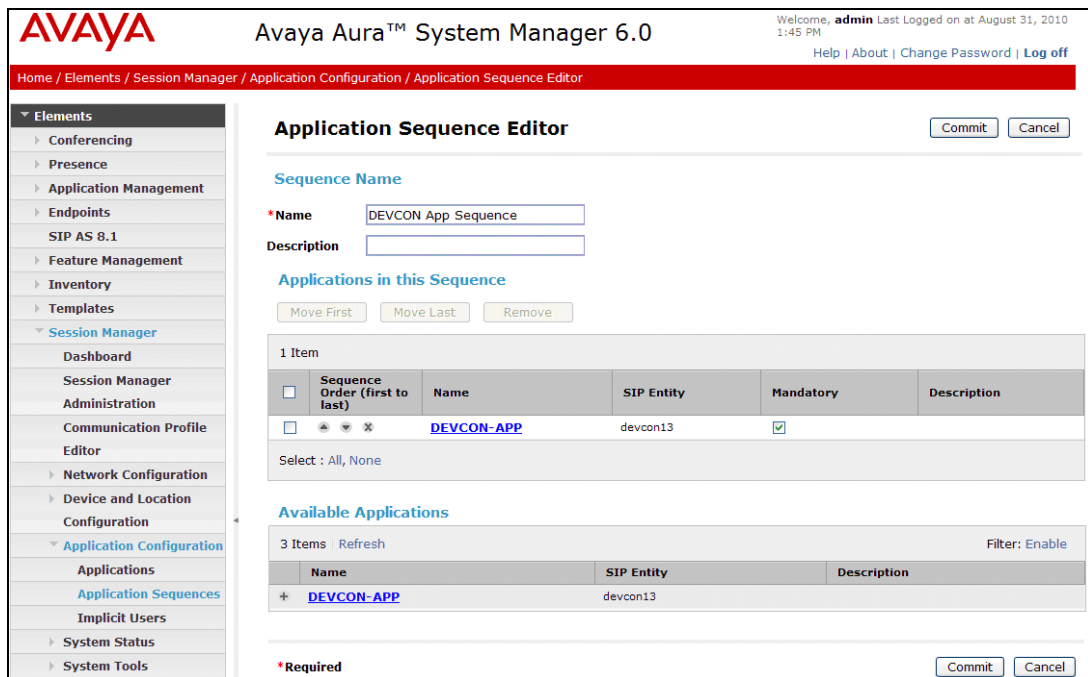
Name	Value
Application Handle	
URI Parameters	

At the bottom of the form, there is a "Required" section with a "Commit" button and a "Cancel" button.

Next, define the Application Sequence for Communication Manager as shown below. Navigate to **Session Manager → Application Configuration → Application Sequence**, and then click **New** in the resulting page (not shown). Provide a descriptive name and add an available application under the **Available Applications** section (not shown).

Verify a new entry is added to the **Applications in this Sequence** table and the **Mandatory** column is  as shown below. Click **Commit**.

Note: The Application Sequence defined for Communication Manager Evolution Server can only contain a single Application.



AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at August 31, 2010 1:45 PM

Help | About | Change Password | **Log off**

Home / Elements / Session Manager / Application Configuration / Application Sequence Editor

Application Sequence Editor Commit Cancel

Sequence Name

*Name

Description

Applications in this Sequence

Move First Move Last Remove

1 Item

☐	Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
☐	1	DEVCON-APP	devcon13	☒	

Select : All, None

Available Applications

3 Items Refresh Filter: Enable

☐	Name	SIP Entity	Description
☒	DEVCON-APP	devcon13	

*Required Commit Cancel

5.7. Add SIP Users

Add SIP users corresponding to the SoundPoint IP 670 SIP Phone defined in **Section 4.3**. Alternatively, use the option to automatically generate the SIP stations on Communication Manager Evolution Server when adding a new SIP user.

To add new SIP users, expand **Users** and select **Manage Users** from left and select **New** button (not shown) on the right.

Enter values for the following required attributes for a new SIP user in the **General** section of the new user form.

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.

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

The screenshot displays the Avaya Aura™ System Manager 6.0 interface. At the top, the Avaya logo is on the left, the title 'Avaya Aura™ System Manager 6.0' is in the center, and the user status 'Welcome, admin Last Logged on at August 31, 2010 1:45 PM' is on the right. Below the title bar is a red navigation bar with the breadcrumb 'Home / Users / Manage Users / New User'. A sidebar on the left contains a tree view with categories: Elements, Events, Groups & Roles, Licenses, Routing, Security, System Manager Data, and Users. The 'Users' category is expanded, and 'Manage Users' is highlighted. The main content area is titled 'New User Profile' and includes 'Commit' and 'Cancel' buttons. Below the title are tabs for 'General', 'Identity', 'Communication Profile', 'Roles', 'Group Membership', 'Default Contact List', and 'Private Contacts'. The 'General' tab is active, showing a form with the following fields: 'Last Name' (required, value: 78500), 'First Name' (required, value: Polycom), 'Middle Name', and 'Description'. The 'Expand All' and 'Collapse All' links are visible below the tabs.

Enter values for the following required attributes for a new SIP user in the **Identity** section of the new user form.

- **Login Name:** Enter <extension>@<sip domain> of the user (e.g., 78500@avaya.com).
- **Authentication Type:** Select *Basic*.
- **SMGR Login Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.
- **Shared Communication Profile Password:** Enter the password which will be used by the SIP phone to log into Session Manager.
- **Confirm Password:** Re-enter the password from above.

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

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at August 31, 2010 1:45 PM

[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

[Home](#) / [Users](#) / [Manage Users](#) / [New User](#)

New User Profile Commit Cancel

[General](#) | [Identity](#) | [Communication Profile](#) | [Roles](#) | [Group Membership](#) | [Default Contact List](#) | [Private Contacts](#) | [Expand All](#) | [Collapse All](#)

General

Identity

* **Login Name:** 78500@avaya.com

* **Authentication Type:** Basic

SMGR Login Password:

* **Password:**

* **Confirm Password:**

Shared Communication Profile Password:

Confirm Password:

Localized Display Name: Polycom 78500

Endpoint Display Name: Polycom 78500

Honorific:

Language Preference: English

Time Zone: Eastern Time (US & Canada)

Scroll down to the **Communication Profile** section and select **New** to define a **Communication Profile** for the new SIP user. Enter values for the following required fields:

- **Name:** Enter name of communication profile.
- **Default:** Select field to indicate that this is the default profile.

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

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and select SIP domain.

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

The screenshot displays the 'Users' management interface. On the left is a sidebar with 'Users' expanded, showing options like 'Manage Users', 'Public Contact Lists', 'Shared Addresses', and 'System Presence ACLs'. Below this is a 'Help' section with links for creating, editing, and deleting users and contacts. The main area is titled 'Identity' and contains a 'Communication Profile' section with 'New', 'Delete', 'Done', and 'Cancel' buttons. Below this is a table with one row: 'Primary' (selected) and 'None'. The 'Name' field is set to 'Primary' and the 'Default' checkbox is checked. The 'Communication Address' section has 'New', 'Edit', and 'Delete' buttons. Below it is a table with columns 'Type', 'Handle', and 'Domain', showing 'No Records found'. The 'Type' dropdown is set to 'Avaya SIP'. The 'Fully Qualified Address' field contains '78500' and the domain dropdown is set to 'avaya.com'. 'Add' and 'Cancel' buttons are at the bottom right.

Type	Handle	Domain
No Records found		

In the *Session Manager* section, specify the Session Manager entity from **Section 5.3.1** for **Primary Session Manager** and assign the **Application Sequence** defined in **Section 5.6** to the new SIP user as part of defining the **SIP Communication Profile**. The **Application Sequence** can be used for both the originating and terminating sequence. Set the **Home Location** field to the **Location** configured in **Section 5.2**. Click **Commit**.

[Manage Users](#)
[Public Contact Lists](#)
[Shared Addresses](#)
[System Presence ACLs](#)

Help
[Help for Create User](#)
[Help for New Private Contact](#)
[Help for Edit Private Contact](#)
[Help for Delete Private Contact](#)
[Help for adding contact into contact list](#)
[Help for editing contact from contact list](#)
[Help for deleting contact from contact list](#)

Communication Profile ▾

New Delete Done Cancel

Name
Primary

Select : None

* Name: Primary

Default : ☒

Communication Address ▾

New Edit Delete

☐	Type	Handle	Domain
☐	Avaya SIP	78500	avaya.com

Select : All, None

☒ Session Manager Profile ▾

* Primary Session Manager devcon-asm ▾

Primary	Secondary	Maximum
3	0	3

Secondary Session Manager (None) ▾

Primary	Secondary	Maximum

Origination Application Sequence DEVCON App Sequence ▾

Termination Application Sequence DEVCON App Sequence ▾

Survivability Server (None) ▾

* Home Location BR-DevConnect ▾

JAO; Reviewed:
SPOC 12/16/2010

Solution & Interoperability Test Lab Application Notes
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25 of 32
Polycom-SP670

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

- **System:** Select the managed element corresponding to Communication Manager.
- **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.
- **Port:** Enter *IP*.
- **Delete Station on Unassign of Station:** Enable field to automatically delete station when **Station Profile** is un-assigned from user.

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

The screenshot displays the 'Communication Profile' configuration page. On the left is a sidebar with navigation links: 'Manage Users', 'Public Contact Lists', 'Shared Addresses', 'System Presence ACLs', and a 'Help' section with links for creating, editing, and deleting users and contacts. The main content area is titled 'Communication Profile' and includes buttons for 'New', 'Delete', 'Done', and 'Cancel'. Below these is a table with one row labeled 'Primary'. The 'Select : None' dropdown is visible. The 'Name' field is set to 'Primary' and the 'Default' checkbox is checked. The 'Communication Address' section is expanded, showing 'Session Manager Profile' and 'Endpoint Profile' checkboxes, both of which are checked. The 'Endpoint Profile' section contains the following fields: 'System' (devcon13-CM-ES), 'Use Existing Endpoints' (unchecked), 'Extension' (78500) with an 'Endpoint Editor' button, 'Template' (DEFAULT_9630SIP_CM_6_0), 'Set Type' (9630SIP), 'Security Code' (empty), 'Port' (IP), and 'Voice Mail Number' (empty). At the bottom, the 'Delete Endpoint on Unassign of Endpoint from User' checkbox is checked.

5.8. Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between Avaya Aura™ System Manager and Avaya Aura™ Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Avaya Aura™ Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Avaya Aura™ Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Avaya Aura™ Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Avaya Aura™ Session Manager

Use default values for the remaining fields. Click **Commit** to add this Session Manager.

6. Configure Polycom SoundPoint® IP 670 SIP Phone

The configuration of the SoundPoint® IP 670 SIP Phone was performed via the phone's menu-driven LCD user interface and its embedded Web interface. The phone's LAN connection interface was initially configured via the phone's LCD screen. To configure the IP parameters for the phone, click the MENU key on the phone and navigate to **Settings→Advanced→Admin Settings→Network Configuration**. A valid password will be required. The rest of the configuration was performed through the phone's embedded Web interface. Refer to [3] for additional information on configuring the SoundPoint® IP 670 SIP Phone.

Note: To verify that the phone is running the compliance-tested SIP application version, press the **Menu** key on the phone, and then select **Status→Platform→Application**. Refer to [3] for upgrade instructions, if required.

From an internet browser, enter `http://<ip-addr>` in the URL field, where `<ip-addr>` is the phone's IP address. Select **SIP** to configure the **SIP Configuration Parameters** screen shown below. In the **Server 1** section, set the **Address** field to the Session Manager's SIP interface and configure the transport protocol and port used for the SIP messages. In this example, SIP messages were sent using TCP over port 5060.

Note: Although the **Outbound Proxy Address** was configured, it was not required in this test configuration.

The screenshot displays the Polycom SoundPoint IP Configuration web interface. The top navigation bar includes the Polycom logo and the title "SoundPoint IP Configuration". Below this is a menu bar with tabs: Home, General, Network, SIP, H.323, and Lines. The "SIP" tab is selected, leading to the "SIP Configuration Parameters" screen. This screen has two main sections: "Servers" and "Local Settings". The "Servers" section is expanded, showing configuration for "Outbound Proxy" and "Server 1".

Servers	
Outbound Proxy	
Address	10.32.24.235
Port	5060
Transport	TCPonly
Server 1	
Address	10.32.24.235
Port	5060
Transport	TCPonly
Expires	
Register	1
Retry Timeout	0
Retry Maximum Count	0
Line Seize Timeout	30

Next, scroll down to the **Local Settings** section and configure the **Digitmap** field to cover the dial strings supported by the dial plan. In this configuration, 5-digit numbers starting with '2' and '7' were supported. Click **Submit** and wait until the phone reboots.

Local Settings	
Local SIP Port	
Calls Per Line Key	
New SDP Type	☐ Enabled ☒ Disabled
Live Communication Server Support	☐ Enabled ☒ Disabled
Non Standard Line Seize	☒ Enabled ☐ Disabled
Digitmap	2xxxx 7xxxx [2-9] 11 0T 011xxx.T [0-1]
Digitmap Timeout	3 3 3 3 3
Remove End-Of-Dial Marker	☒ Enabled ☐ Disabled
Digitmap Impossible Match	0
top	Submit

After the phone reboots, access the **Lines** screen from the phone's embedded Web interface. In the **Identification** section, provide a descriptive **Display Name** and specify the phone's extension in the **Address** field. In the **Authentication User ID** and **Authentication Password** fields, configure the extension and password, respectively, used to register with Session Manager. The content of the **Label** field will be used as the phone's call appearance label on the display. The **Number Of Line Keys** field was set to 3.

POLYCOM		SoundPoint IP Configuration									
Home		General		Network		SIP		H.323		Lines	
Line Parameters:											
Line 1	Line 2	Line 3	Line 4	Line 5	Line 6	Line 7	Line 8	Line 9	Line 10	Line 11	Line 12
Line 13	Line 14	Line 15	Line 16	Line 17	Line 18	Line 19	Line 20	Line 21	Line 22	Line 23	Line 24
Line 25	Line 26	Line 27	Line 28	Line 29	Line 30	Line 31	Line 32	Line 33	Line 34		
Line 1											
Identification											
Display Name		SoundPoint 78500									
Address		78500									
Authentication User ID		78500									
Authentication Password		••••									
Label		78500									
Type		☒ Private ☐ Shared									
Third Party Name											
Number Of Line Keys		3									
Calls Per Line											

Scroll down to the **Message Center** section and set the **Subscriber** field to the phone's extension to enable MWI. The **Callback Mode** and **Callback Contact** fields were set to *Contact* and the voicemail pilot number, respectively, so that the voicemail system can be dialed through the **Message Center** menu option on the phone. Click **Submit** to save the settings and reboot the phone.

Message Center	
Subscriber	78500
Callback Mode	Contact ▼
Callback Contact	29000
top	Submit

The following screen simply shows the codecs supported by the endpoint. No additional configuration is required here.

General Configuration Parameters:				
User Preferences	Time	Audio Processing	Video Processing	Background
Sampled Audio	Microbrowser	Logging	Applications	Power Saving
Audio Processing				
Codec Preferences				
G.711Mu	2 ▼			
G.711A	3 ▼			
G.722	1 ▼			
G.729AB	4 ▼			
iLBC 13.3kbps	Not Used ▼			
iLBC 15.2kbps	Not Used ▼			
G.711Mu Codec Profile				
Payload Size	20			
Jitter Buffer Minimum	40			
Jitter Buffer Shrink	500			
Jitter Buffer Maximum	160			
G.711A Codec Profile				
Payload Size	20			
Jitter Buffer Minimum	40			
Jitter Buffer Shrink	500			
Jitter Buffer Maximum	160			
G.722 Codec Profile				
Payload Size	20			
Jitter Buffer Minimum	40			
Jitter Buffer Shrink	1500			
Jitter Buffer Maximum	200			
G.729AB Codec Profile				
Payload Size	20			
Jitter Buffer Minimum	40			

7. General Test Approach and Test Results

To verify interoperability of the SoundPoint IP 670 SIP Phone with Communication Manager and Session Manager, calls were made between Polycom SoundPoint IP 670 SIP Phones and Avaya SIP, H.323, and digital stations using various codec settings and exercising common PBX features. The telephony features listed in **Section 1.1** were activated and deactivated using phone buttons and FNEs. All test cases passed.

8. Verification Steps

The following steps can be used to verify and/or troubleshoot installations in the field.

1. Verify that the SoundPoint IP 670 SIP Phones have successfully registered with Session Manager.
2. Verify basic telephony features by establishing calls between a SoundPoint IP 670 SIP Phone and another phone.
3. Call a SoundPoint IP 670 SIP phone that currently has no voice messages, and leave a message. Verify that the message waiting indicator (i.e., Voicemail button) illuminates. Call the voicemail system and retrieve voice messages. Verify that after hearing all messages, that the message waiting indicator is extinguished.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Polycom SoundPoint IP 670 SIP Phone with Avaya AuraTM Communication Manager and Avaya AuraTM Session Manager. The SoundPoint IP 670 SIP Phone successfully registered with Session Manager and basic telephony features were verified. All test cases passed.

10. References

This section references the Avaya and Polycom documentation relevant to these Application Notes. The following Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administering Avaya AuraTM Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509.
- [2] *Administering Avaya AuraTM Session Manager*, August 2010, Issue 3, Release 6.0, Document Number 03-603324.
- [3] *Administrator's Guide for the Polycom SoundPoint IP / SoundStation IP / VVX Family*, SIP 3.2.2, November 2009, Document Number 1725-11530-322 Rev. A.

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