



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Polycom® SoundStation IP running UC Software release 4.0.2 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager Release 6.2 - Issue 1.1

Abstract

These Application Notes describe a solution for supporting interoperability between Polycom SoundStation IP conference telephones running UC software release 4.0.2 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager release 6.2. Emphasis of the testing was to verify voice calls of SoundStation IP as SIP endpoints registered to Avaya Aura® Session Manager.

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 provide detail configurations of the Polycom SoundStation IP (including 5000, 6000, 7000 and Duo models) with a SIP infrastructure consisting of Avaya Aura® Session Manager (SM) and Avaya Aura® Communication Manager (CM). During compliance testing, SoundStation IP SIP Conference Phones successfully registered with Session Manager, established calls with other Avaya telephones, and executed telephony features such as Hold, Transfer, and Conference.

2. General Test Approach and Test Results

The general test approach was to have the SoundStation IP to register to Session Manager. Calls were then placed from Avaya telephone clients/users to and from the SoundStation IP. Other telephony features such as busy, hold, DTMF, transfer, conference and codec negotiation were also verified.

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

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Registration of the SoundStation IP to SM.
- Call establishment of SoundStation IP with Avaya telephones.
- Telephony features: Basic calls, conference, blind and consultative transfer, DTMF (dual tone multi frequency), leaving and retrieving voicemail message, busy, hold, call forward busy, call forward unconditional, call forward no answer, MWI (Message Waiting Indicator), Reject and Do not Disturb (DND).
- Codec negotiation – G.711, G.729 and G722.
- SoundStation IP calls PSTN telephone via SIP trunk.

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. The SoundStation IP was registered to SM successfully. Calls have been made between Avaya telephones and SoundStation IP with clear voice path.

2.3. Support

Technical support for the Polycom SoundStation IP conference phone can be obtained through Polycom global technical support:

- Phone: 1-888-248-4143 or 1-408-474-2067
- Web: <http://support.polycom.com>

3. Reference Configuration

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

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

In this test configuration, the 2 Polycom SoundStation Duo (hereafter referred to as Duo) were used as representative of the Polycom SoundStation IP phones and they were registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

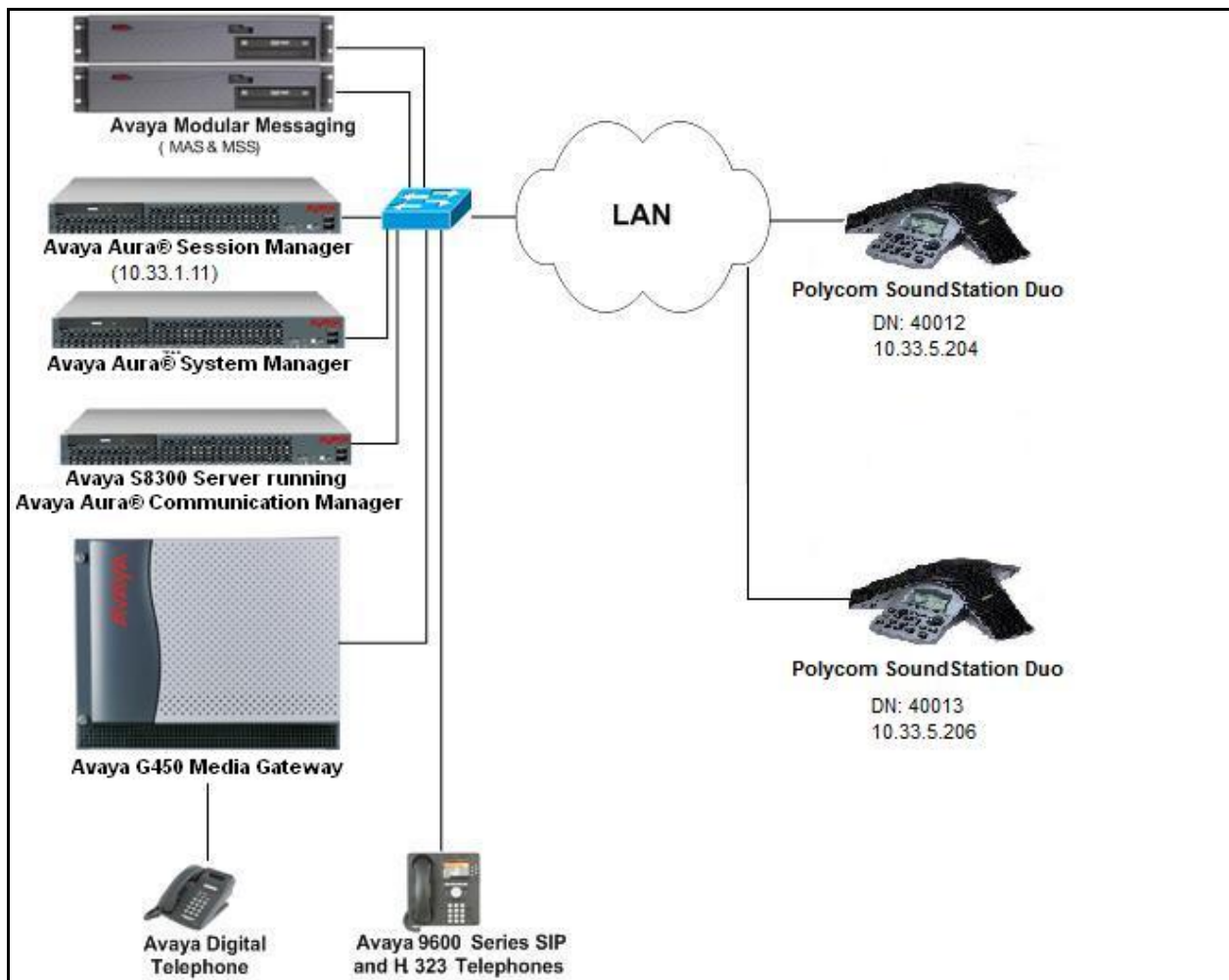


Figure 1: Network Configuration Diagram

4. Equipment and Software Validated

The following equipment and software/firmware were used for the reference configuration:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Avaya S8300 Servers and G450 Media Gateway	6.2 (Avaya CM/R016x.02.0.823.0) (System Platform 6.2.1.0.9) 31.22.0
Avaya Aura® System Manager running on an Avaya S8800 Server	6.2.12.0 (Patch 6.2.12.202 Build Number 6.2.14.1.1925)
Avaya Aura® Session Manager running on S8800 Server.	6.2 (6.2.2.0.622005)
Avaya Modular Messaging	5.2
Avaya 9641G SIP Telephone	6.2.0.69 (SIP)
Avaya 9611G H.323 Telephone	S6.2209
Avaya Digital Telephones	N/A
Polycom SoundStation Duo (SIP)	UC software 4.0.2.8017

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the Duo as an Off-PBX Station (OPS) and configuring a SIP trunk between Communication Manager and Session Manager. **Section Error!** Reference source not found. covers the station configuration for the Duo. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

5.1. 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	15
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.)		

5.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the S8300 Server processor, the C-LAN board in the G450 Media Gateway, and 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	
Name	IP Address	
default	0.0.0.0	
interopsm	10.33.1.11	
procr	10.33.1.22	
procr6	::	
(4 of 4 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 *bvwdev.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 G450 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: bvwdev.com
Name: Main Network Region
MEDIA PARAMETERS
  Codec Set: 1      Intra-region IP-IP Direct Audio: yes
                   Inter-region IP-IP Direct Audio: yes
                   IP Audio Hairpinning? n
  UDP Port Min: 2048
  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 the Duo. 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 Duo conference 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 recommended 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 *bvwdev.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 10		Page 1 of 1
SIGNALING GROUP		
<div style="display: flex; justify-content: space-between;"> Group Number: 10 Group Type: sip </div> <div style="display: flex; justify-content: space-between;"> IMS Enabled? n Transport Method: tcp </div> <div style="display: flex; justify-content: space-between;"> Q-SIP? n </div> <div style="display: flex; justify-content: space-between;"> IP Video? n Enforce SIPS URI for SRTP? y </div> <div style="display: flex; justify-content: space-between;"> Peer Detection Enabled? y Peer Server: SM </div>		
<div style="display: flex; justify-content: space-between;"> Near-end Node Name: procr Far-end Node Name: interopsm </div> <div style="display: flex; justify-content: space-between;"> Near-end Listen Port: 5060 Far-end Listen Port: 5060 </div> <div style="display: flex; justify-content: space-between;"> Far-end Network Region: 1 </div> <div style="display: flex; justify-content: space-between;"> Far-end Secondary Node Name: </div>		
<div style="display: flex; justify-content: space-between;"> Far-end Domain: bvwdev.com Bypass If IP Threshold Exceeded? n </div> <div style="display: flex; justify-content: space-between;"> Incoming Dialog Loopbacks: eliminate RFC 3389 Comfort Noise? n </div> <div style="display: flex; justify-content: space-between;"> DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y </div> <div style="display: flex; justify-content: space-between;"> Session Establishment Timer(min): 3 IP Audio Hairpinning? n </div> <div style="display: flex; justify-content: space-between;"> Enable Layer 3 Test? y Initial IP-IP Direct Media? n </div> <div style="display: flex; justify-content: space-between;"> H.323 Station Outgoing Direct Media? n Alternate Route Timer(sec): 6 </div>		

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 10		Page 1 of 21	
TRUNK GROUP			
Group Number: 10	Group Type: sip	CDR Reports: y	
Group Name: SIP Trunk to Interop SM	COR: 1	TN: 1	TAC: #10
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: 10		
	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 10		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			
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 '400' and whose calls are routed over any trunk group, including SIP trunk group "10", 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	33	10	5
5	58	10	5
5	400	10	5
5	600	10	5
		Total Administered: 4	
		Maximum Entries: 540	

5.3. Configure Stations

Use the **add station** command to add a station for each Duo phone to be supported. Use *9620SIP* 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 Session Manager as described in **Section 6.7**.

add station 40012		Page 1 of 6
STATION		
Extension: 40012	Lock Messages? n	BCC: 0
Type: 9620SIP	Security Code:	TN: 1
Port: S00003	Coverage Path 1: 1	COR: 1
Name: SIP, 40012	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 19		
	Message Lamp Ext: 40012	
Display Language: english		
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 Duo.

add station 40012		Page 2 of 6
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Coverage Msg Retrieval? y	
LWC Activation? y	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Per Button Ring Control? n	Idle Appearance Preference? n	
Bridged Call Alerting? n	Bridged Idle Line Preference? n	
Active Station Ringing: single	Restrict Last Appearance? y	
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
	EC500 State: enabled	
MWI Served User Type: qsig-mwi		
AUDIX Name:	Coverage After Forwarding? s	
Emergency Location Ext: 40012	Direct IP-IP Audio Connections? y	
Precedence Call Waiting? y	Always Use? n IP Audio Hairpinning? n	

Use the **change off-pbx-telephone station-mapping** command to map Communication Manager extensions (e.g., 40012) to the same extension configured in Session 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 show in these Application Notes. The **Configuration Set** value can reference a set that has the default settings.

change off-pbx-telephone station-mapping 40012							Page 1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
40012	OPS	-		40012	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	
40012	OPS	3	both	all	none		

6. 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, to be managed by System Manager.
- 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 and accept the Copyright Notice.

6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. It can be done by selecting **SIP 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., *bvwdev.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.

AVAYA Avaya Aura® System Manager 6.2

Last Logged on at September 25, 2012 2:55 PM
Help | About | Change Password | Log off admin

Routing * Home

Routing Domains Locations Adaptations SIP Entities Entity Links Time Ranges Routing Policies Dial Patterns Regular Expressions

Home / Elements / Routing / Domains

Domain Management

Help ?

Commit Cancel

1 Item Refresh Filter: Enable

Name	Type	Default	Notes
* bvwdev.com	sip	<input type="checkbox"/>	Polycom testing

* Input Required

Commit Cancel

6.2. 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 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 Communication Manager and Session Manager. Click **Commit** to save the Location definition.



Routing x

Home

- Routing
- Domains
- Locations
- Adaptations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns
- Regular Expressions
- Defaults

Home / Elements / Routing / Locations

[Help ?](#)

Commit Cancel

Location Details

General

* Name: Belleville

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth: 100000

Multimedia Bandwidth: 100000

Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): 1000 Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): 1000 Kbit/Sec

* Minimum Multimedia Bandwidth: 64 Kbit/Sec

* Default Audio Bandwidth: 80 Kbit/sec

Alarm Threshold

Overall Alarm Threshold: 80 %

Multimedia Alarm Threshold: 80 %

* Latency before Overall Alarm Trigger: 5 Minutes

* Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

Add Remove

4 Items Refresh

Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.33.5.*	
<input type="checkbox"/>	* 10.33.*	

Select : All, None

6.3. Add SIP Entities

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

6.3.1. 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 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:** Specify *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 6.2 web interface. The top header includes the Avaya logo, the product name 'Avaya Aura® System Manager 6.2', and a 'Last Logged on' timestamp of September 25, 2012 2:55 PM. Navigation links for 'Help', 'About', 'Change Password', and 'Log off admin' are present. A breadcrumb trail shows 'Home / Elements / Routing / SIP Entities'. The left sidebar contains a menu with 'Routing' selected, and sub-items like 'Domains', 'Locations', 'Adaptations', 'SIP Entities' (highlighted), 'Entity Links', 'Time Ranges', 'Routing Policies', 'Dial Patterns', 'Regular Expressions', and 'Defaults'. The main content area is titled 'SIP Entity Details' and features a 'General' tab. The form fields are as follows: 'Name' (InteropSM), 'FQDN or IP Address' (10.33.1.11), 'Type' (Session Manager), 'Notes' (Interop Session Manager), 'Location' (Belleville), 'Outbound Proxy' (empty), 'Time Zone' (America/New_York), and 'Credential name' (empty). At the bottom, the 'SIP Link Monitoring' section shows 'SIP Link Monitoring' set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located at the top right 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. *bvwdev.com*).

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

Port
TCP Failover port:
TLS Failover port:

5 Items [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP ▼	bvwdev.com ▼	<input type="text"/>

Select : All, None

SIP Responses to an OPTIONS Request

0 Items [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--------------------------	-------------------------------	---------------------	-------

* Input Required

6.3.2. Avaya Aura® Communication Manager

A SIP Entity must be added for 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:** Specify *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.

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The top header includes the Avaya logo, the title "Avaya Aura® System Manager 6.2", and a "Last Logged on at September 25, 2012 4:02 PM" timestamp. Navigation links for "Help", "About", "Change Password", and "Log off admin" are present. A breadcrumb trail shows "Home / Elements / Routing / SIP Entities". The left sidebar contains a menu with "Routing" expanded, showing sub-items like "Domains", "Locations", "Adaptations", "SIP Entities" (highlighted), "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns", "Regular Expressions", and "Defaults". The main content area is titled "SIP Entity Details" and has a "General" tab selected. It contains several input fields: "Name" (Interop CM), "FQDN or IP Address" (10.33.1.22), "Type" (CM), "Notes" (Interop CM6.2), "Adaptation" (empty), "Location" (Belleville), "Time Zone" (America/New_York), "Override Port & Transport with DNS SRV" (checkbox), "SIP Timer B/F (in seconds)" (4), "Credential name" (empty), "Call Detail Recording" (none), and "SIP Link Monitoring" (Use Session Manager Configuration). "Commit" and "Cancel" buttons are at the top right of the form area.

6.4. 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 on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g. *Interop CM to SM*).
- **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 Error! Reference source not found.** will be denied.*

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

Avaya Aura® System Manager 6.2

Last Logged on at September 25, 2012 4:02 PM
Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Entity Links

Entity Links

Help ?
Commit Cancel

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* Interop CM to SM	* InteropSM	TCP	* 5060	* Interop CM	* 5060	Trusted	

* Input Required
Commit Cancel

6.5. 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 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.

Click **Commit** to save the settings.

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The top header includes the Avaya logo, the product name 'Avaya Aura® System Manager 6.2', and a 'Last Logged on' timestamp. A navigation menu on the left lists various system management functions, with 'Inventory' expanded to show 'Manage Elements'. The main content area is titled 'New Communication Manager' and contains two tabs: 'General' (active) and 'Attributes'. The 'General' tab has several input fields: 'Name' (containing 'Interop CM6.2'), 'Type' (a dropdown menu set to 'Communication Manager'), 'Description' (containing 'Polycom Testing CM6.2'), and 'Node' (containing '10.33.1.22'). A 'Reset' button is located next to the 'Type' dropdown. At the top right of the form area are 'Commit' and 'Cancel' buttons. The breadcrumb trail at the top of the form area reads 'Home / Elements / Inventory / Manage Elements'.

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).

Click **Commit** to save the settings.

The screenshot shows the Avaya Aura System Manager 6.2 web interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.2', and a 'Last Logged on at September 25, 2012 4:02 PM' status. There are links for 'Help | About | Change Password | Log off admin'. The main navigation menu on the left includes 'Inventory', 'Manage Elements', 'Upgrade Management', 'Collected Inventory', 'Manage Serviceability Agents', 'Inventory Management', 'Synchronization', and 'CS 1000 and CallPilot Synchronization'. The 'Manage Elements' menu item is highlighted. The main content area is titled 'New Communication Manager' and has 'Commit' and 'Cancel' buttons. The 'Attributes' tab is selected, showing the following fields:

- SNMP Attributes** (dropdown menu)
- * Version** (radio buttons: ☒ None, ☐ V1, ☐ V3)
- * Login** (text field: Interop)
- Password** (password field: masked with dots)
- Confirm Password** (password field: masked with dots)
- Is SSH Connection** (checkbox: checked)
- * Port** (text field: 5022)

6.6. Add Application Sequence

Define an application for Communication Manager. Fill in the following fields:

- **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 6.2 web interface. The top header includes the Avaya logo, the product name "Avaya Aura® System Manager 6.2", and a user status bar indicating "Last Logged on at September 25, 2012 4:02 PM" with links for "Help | About | Change Password | Log off admin". A navigation bar shows "Session Manager" as the active tab, with a "Home" button. The left sidebar contains a tree view with categories: "Session Manager" (containing Dashboard, Session Manager Administration, and Communication Profile Editor), "Network Configuration", "Device and Location Configuration", "Application Configuration", and "Applications" (highlighted in blue). The main content area is titled "Application Editor" and contains a sub-section "Application" with the following fields: "*Name" (text input with "Interop CM"), "*SIP Entity" (dropdown menu with "Interop CM"), "*CM System for SIP Entity" (dropdown menu with "Interop CM6.2" and a "Refresh" button), and "Description" (text input). There are "Commit" and "Cancel" buttons at the top right of the form area. A "View/Add CM Systems" link is also visible next to the CM System dropdown.

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

AVAYA

Avaya Aura® System Manager 6.2

Last Logged on at September 25, 2012 4:02 PM
Help | About | Change Password | **Log off admin**

Session Manager × Home

Home / Elements / Session Manager / Application Configuration / Application Sequences

Help ?

Application Sequence Editor

Commit Cancel

Application Sequence

***Name**

Description

Applications in this Sequence

Move First Move Last Remove

1 Item

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

Select : All, None

Available Applications

3 Items Refresh Filter: Enable

	Name	SIP Entity	Description
+	Interop CM	Interop CM	

6.7. Add SIP Users

Add SIP users corresponding to the SoundStation Duo defined in **Section** Error! Reference source not found.. Alternatively, use the option to automatically generate the SIP stations on Communication Manager Feature Server when adding a new SIP user.

Enter values for the following required attributes for a new SIP user in the new user 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., 40012@bvwddev.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 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 Aura® System Manager 6.2

Last Logged on at September 25, 2012 4:02 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[User Management](#) * [Home](#)

Home / Users / User Management / Manage Users

New User Profile [Help ?](#)

[Commit & Continue](#) [Commit](#) [Cancel](#)

Identity * **Communication Profile** * **Membership** **Contacts**

Identity ▾

* **Last Name:** SIP

* **First Name:** 40012

Middle Name:

Description: Polycom Duo endpoint

* **Login Name:** 40012@bvwddev.com

* **Authentication Type:** Basic ▾

* **Password:** ••••

* **Confirm Password:** ••••

Localized Display Name: SIP, 40012

Endpoint Display Name: SIP, 40012

Title:

Language Preference: English (United States) ▾

Time Zone: (-4:0)Eastern Time (US & Canada) ▾

Click the *Communication Profile* tab 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 *sip*.
- **SubType:** Select *username*.
- **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**.

Avaya Aura® System Manager 6.2

Last Logged on at September 25, 2012 4:02 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

User Management

Manage Users
Public Contacts
Shared Addresses
System Presence ACLs

Home / Users / User Management / Manage Users

Help ?

New User Profile

Commit & Continue Commit Cancel

Identity * Communication Profile * Membership Contacts

Communication Profile

Communication Profile Password:
Confirm Password:

New Delete Done Cancel

Name
Primary

Select : None

* Name: Primary

Default : ☒

Communication Address

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

* Fully Qualified Address: 40012 @ bvwddev.com

Add Cancel

In the *Session Manager Profile* section, specify the Session Manager entity and assign the **Application Sequence** defined in **Section Error! Reference source not found.** 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.

☒ Session Manager Profile

* Primary Session Manager InteropSM

Primary	Secondary	Maximum
8	0	8

Secondary Session Manager (None)

Primary	Secondary	Maximum

Origination Application Sequence Interop CM

Termination Application Sequence Interop CM

Conference Factory Set (None)

Survivability Server (None)

* Home Location Belleville

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

- **System:** Select the managed element corresponding to Communication Manager.
- **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.
- **Port:** Enter *S00003*.
- **Override Endpoint Name:** Enable the field.

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

The screenshot shows the 'CM Endpoint Profile' configuration window. It contains the following fields and settings:

- System:** Interop CM6.2 (dropdown)
- Profile Type:** Endpoint (dropdown)
- Use Existing Endpoints:** ☐
- Extension:** 40012 (text field with search icon)
- Template:** DEFAULT_9620SIP_CM_6_2 (dropdown)
- Set Type:** 9620SIP (text field)
- Security Code:** (empty text field)
- Port:** S00003 (text field with search icon)
- Voice Mail Number:** (empty text field)
- Preferred Handle:** (None) (dropdown)
- Delete Endpoint on Unassign of Endpoint from User or on Delete User:** ☐
- Override Endpoint Name:** ☒

6.8. Add Session Manager

To complete the configuration, adding Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add**, 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 Session Manager.
- **Description:** Descriptive comment (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP

- **Default Gateway:** address of Session Manager.
Enter the IP address of the default gateway for Session Manager.

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

Avaya Aura® System Manager 6.2

Last Logged on at September 25, 2012 4:02 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Session Manager

Home

Home / Elements / Session Manager / Session Manager Administration

Session Manager

Dashboard

Session Manager Administration

Communication Profile Editor

Network Configuration

Device and Location Configuration

Application Configuration

System Status

System Tools

Performance

Edit Session Manager

Commit
Cancel

General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server | Expand All | Collapse All

General

SIP Entity Name
InteropSM

Description
Interop SM in Cage

*Management Access Point Host Name/IP
10.33.1.10

*Direct Routing to Endpoints
Enable

Security Module

SIP Entity IP Address
10.33.1.11

*Network Mask
255.255.255.192

*Default Gateway
10.33.1.1

*Call Control PHB
46

*QOS Priority
6

*Speed & Duplex
Auto

VLAN ID

7. Configure Polycom SoundStation Duo SIP interface

This section describes how to set up the Duo network interface, to access the Duo SIP endpoint web interface and to configure the Duo for testing.

7.1. Determine the IP address used by the Duo

This section shows how to determine the network IP address used by the Duo.

On the Duo (not shown), push the '**Menu**' button and navigate to **2. Status** → **2. Network** → **1. TCP/IP Parameters**. In this example configuration, the following parameters are used as bellow. Others are left at default.

- **DHCP:** Enabled
- **IP Address:** 010.033.005.204
- **Subnet Mask:** 255.255.255.000
- **IP Gateway:** 010.033.005.001

7.2. Polycom SoundStation Duo Web Configuration Utility

This section shows how to log in to the home page of Duo Web Configuration Utility to manage and configure the Duo phone.

Open the web browser, and in the address field enter the Duo IP address as format **http://10.33.5.204** and the Duo login page will appear as shown bellow. Select 'Admin' and enter the default password, **456**.



Click **Submit**, the homepage of Duo appears.



7.3. Configure the Lines for Polycom SoundStation Duo

This section shows how to configure the Duo to register with Session Manager. On the homepage of configuration screen, click on the **Simple Setup** menu, the **Simple Setup** page appears as shown below. Enter values as highlighted in areas marked with red boxes in the screenshot below and leave other areas at default. Click **Save**.

The screenshot shows the 'Simple Setup' configuration page for the Polycom SoundStation Duo. The navigation bar is the same as the homepage. The 'Simple Setup' page is active, displaying 'You are here: Simple Setup'. The main content area is divided into two sections: a 'Simple Setup' section with a phone image and 'VIEWS' (Home, Simple Setup), and a configuration form. The configuration form has several sections: 'Country' (USA (Default)), 'Language' (Phone Language: English (Internal), Web Utility Language: Add), 'Time Synchronization' (SNTP Server, Time Zone: (GMT) Western Europe Time, London, Lisbon, Casablanca), 'SIP Server' (Address: 10.33.1.11, Port: 5060), 'SIP Outbound Proxy' (Address: 10.33.1.11, Port: 5060), and 'SIP Line Identification' (Display Name: Poly1, Address: 40012, Authentication User ID: 40012, Authentication Password: ****, Label). Red boxes highlight the Country, Language, SIP Server Address and Port, SIP Outbound Proxy Address and Port, and SIP Line Identification fields. At the bottom are buttons for Cancel, Reset to Default, View Modifications, and Save.

7.4. SIP Settings

This section shows how to set SIP parameters for Duo.

On the homepage of the Duo Web Configuration Utility, navigate to menu **Settings → SIP**, **SIP** page appears. Enter values as highlighted in the areas marked with red-boxes in the screenshot below and leave other areas at default. Click **Save**.

Note: The default local Digitmap configuration used by Duo may require customization. More detailed information about local Digitmap configuration is available in the Administrator's Guide for Polycom UC Software [3] and Polycom Technical Bulletin 11572 [4] – see **Section 10** for additional reference.

POLYCOM | SoundStation Duo

Home Simple Setup Preferences Settings Diagnostics Utilities

You are here: Settings > SIP

VIEWS

- Microbrowser
- Logging
- Applications
- Audio Codec Priority
- Audio Codec Profiles
- Provisioning Server
- Syslog
- Paging/PTT Configuration
- PSTN Settings
- SIP**
- Lines
- Change Password
- Phone Lock

SIP

Local Settings

- * Local SIP Port: 5060
- Calls Per Line Key: 1
- New SDP Type: ☐ Enable ☒ Disable
- Live Communication Server Support: ☐ Enable ☒ Disable
- * Non Standard Line Seize: ☒ Enable ☐ Disable
- * Digitmap: [2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxx|[2-9]xxxxxxxx|[2-9]xxT
- * Digitmap Timeout: 3|3|3|3|3
- Remove End-of-Dial Marker: ☒ Enable ☐ Disable
- * Digitmap Impossible Match: 0

Outbound Proxy

- Address: 10.33.1.11
- Port: 5060
- Transport: TCPonly

Server 1

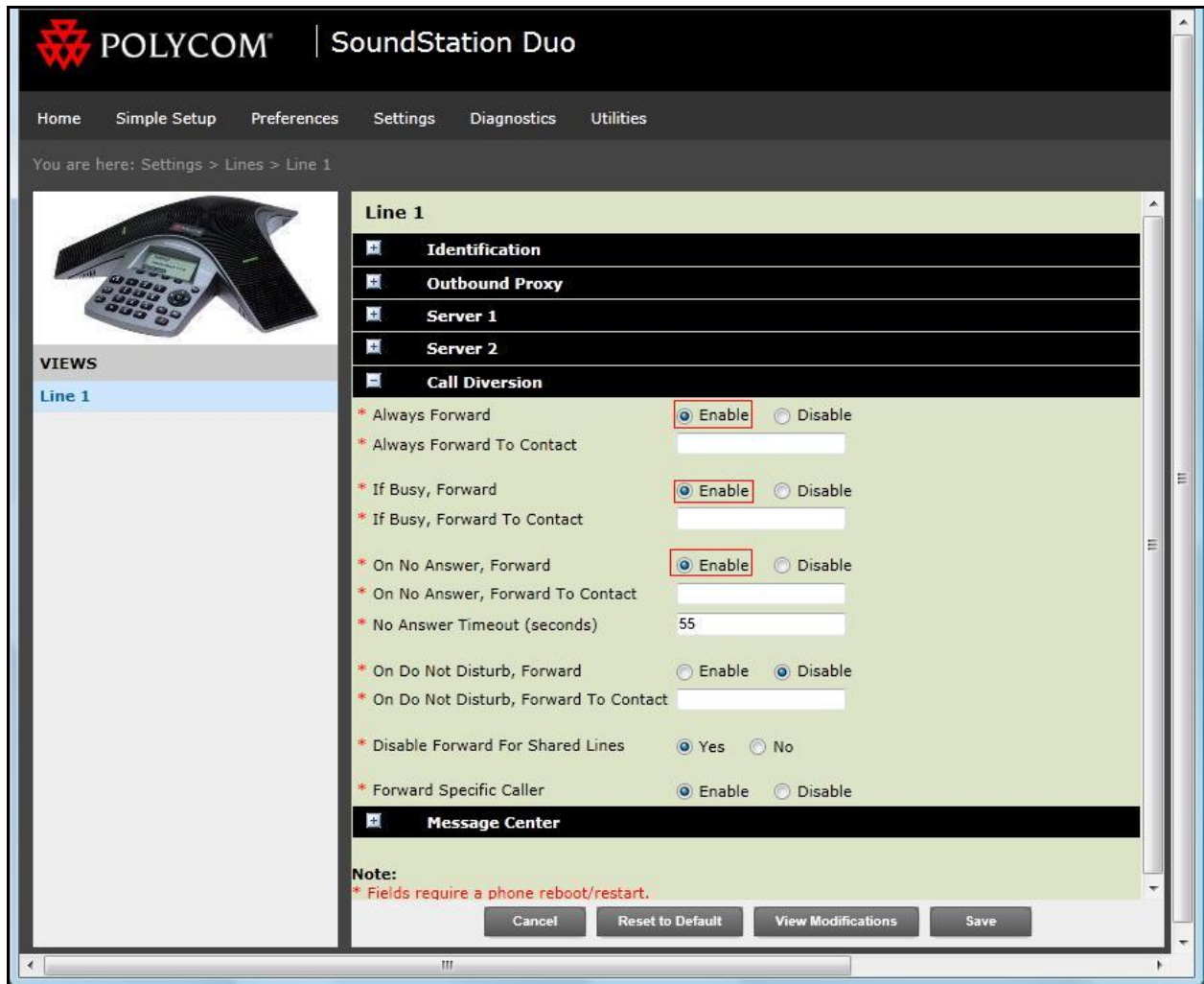
- Address: 10.33.1.11
- Port: 5060
- Transport: TCPonly
- Expires (s): 3600
- Register: ☒ Yes ☐ No
- Retry Timeout (ms): 0
- Retry Maximum Count: 3
- Line Seize Timeout (s): 30

Cancel Reset to Default View Modifications Save

7.5. Local Call Forward Settings

This section shows how to set up call forward settings for Duo.

On the homepage of the Duo Web Configuration Utility, navigate to menu **Settings** → **Lines**, **Line 1** page appears. Click on 'Call Diversion' to expand the 'Call Diversion' section. Enable values as highlighted in the areas marked with red boxes and leave other areas at default. Click **Save**.



POLYCOM | SoundStation Duo

Home Simple Setup Preferences Settings Diagnostics Utilities

You are here: Settings > Lines > Line 1

Line 1

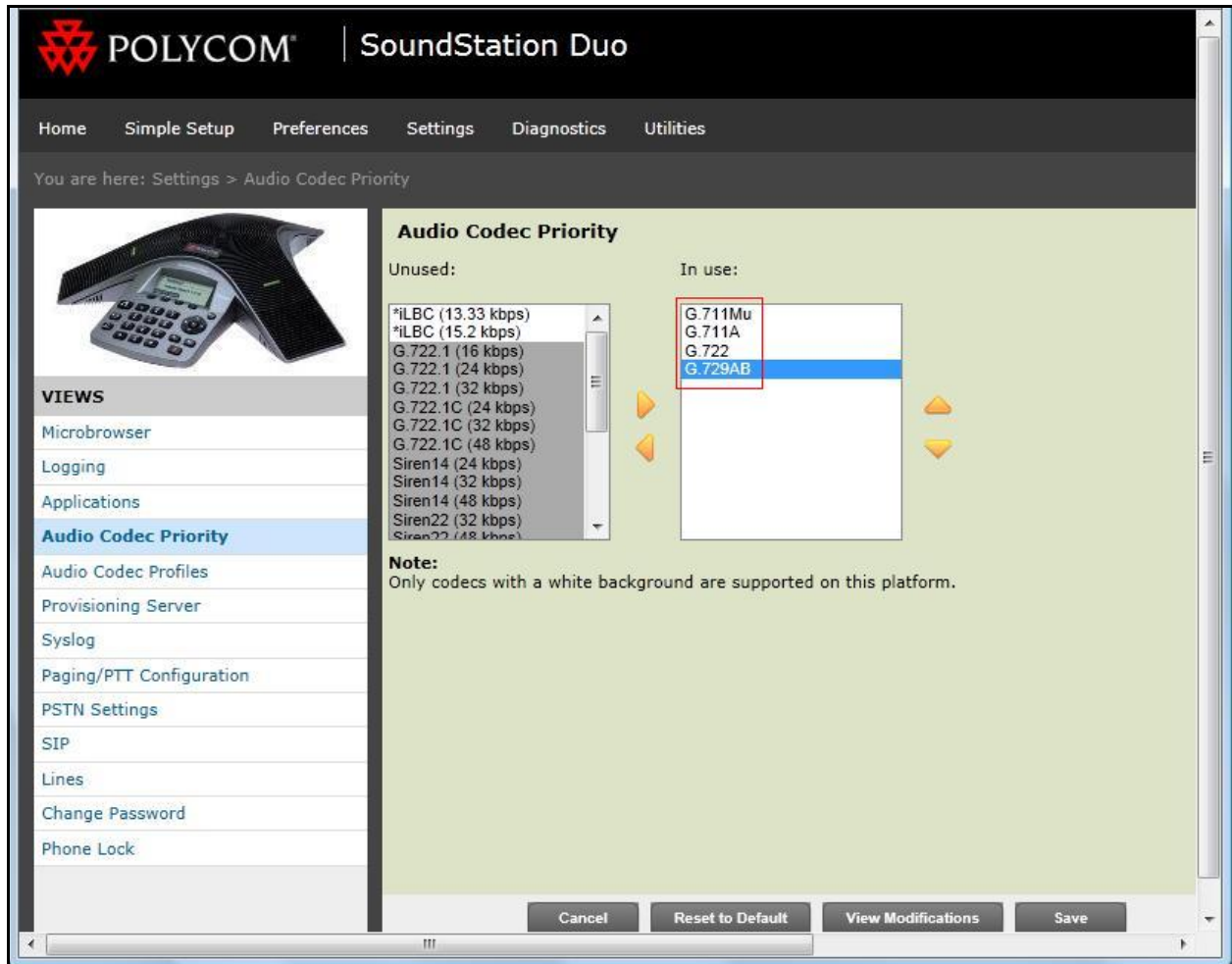
- Identification
- Outbound Proxy
- Server 1
- Server 2
- Call Diversion
 - * Always Forward ☒ Enable ☐ Disable
 - * Always Forward To Contact
 - * If Busy, Forward ☒ Enable ☐ Disable
 - * If Busy, Forward To Contact
 - * On No Answer, Forward ☒ Enable ☐ Disable
 - * On No Answer, Forward To Contact
 - * No Answer Timeout (seconds) 55
 - * On Do Not Disturb, Forward ☐ Enable ☒ Disable
 - * On Do Not Disturb, Forward To Contact
 - * Disable Forward For Shared Lines ☒ Yes ☐ No
 - * Forward Specific Caller ☒ Enable ☐ Disable
- Message Center

Note:
* Fields require a phone reboot/restart.

Cancel Reset to Default View Modifications Save

7.6. Audio Codec Settings

On the homepage of Duo Web Configuration Utility, navigate to menu **Settings** → **Audio Codec Priority**. Select the codec list in the order of priority as shown in the areas marked with a red box in the screenshot below. Click **Save**.



7.7. Voice Mail Setting

On the homepage of the Duo Web Configuration Utility, navigate to menu **Settings** → **Lines**, the **Line 1** page appears. Click on **'Message Center'** to expand the 'Message Center' section. Enter values as highlighted in the areas marked with red-boxes in the screenshot below and leave other areas at default. Click **Save**.

POLYCOM | SoundStation Duo

Home Simple Setup Preferences Settings Diagnostics Utilities

You are here: Settings > Lines > Line 1

Line 1

Identification

Display Name: Poly1

Address: 40012

Authentication User ID: 40012

Authentication Password: ****

Label:

Type: ☒ Private ☐ Shared

Third Party Name:

Number of Line Keys: 1

Calls Per Line: 8

Ring Type: Low Trill

Outbound Proxy

Server 1

Server 2

Call Diversion

Message Center

Subscription Address: 40012

Callback Mode: Contact

Callback Contact: 33000

Cancel Reset to Default View Modifications Save

8. Verification Steps

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

1. Verify that the Duo has successfully registered with Session Manager.
2. Verify basic telephony features by establishing calls between a Duo and another phone.
3. Call a Duo that currently has no voice messages, and leave a message. Verify that the message waiting indicator (i.e., envelop icon) appears on the Duo LCD display. 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 illustrate the procedures necessary for configuring the Polycom SoundStation IP to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All feature functionality test cases described in **Section 2.1** were passed.

10. Additional References

Product documentation for the Avaya products may be found at:

<https://support.avaya.com>

Product documentation for the Polycom Soundstation IP products may be found at:

<http://www.polycom.com>

[1] *Administering Avaya Aura® Communication Manager Server Options*, July 2012, Release 6.2, Issue 3.0, Document Number 03-603479.

[2] *Administering Avaya Aura® Session Manager*, July 2012, Release 6.2, Document Number 03-603324.

[3] Polycom SoundStation IP Series Documents:

Administrator's Guide for the Polycom® UC Software

http://support.polycom.com/PolycomService/support/us/support/voice/soundstation_ip_series

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