



Avaya Solution & Interoperability Test Lab

Application Notes for Polycom IP 6000 and Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Polycom SoundStation IP 6000 which was compliance tested with Avaya Communication Manager and Avaya SIP Enablement Services.

The overall objective of the interoperability compliance testing is to verify Polycom SoundStation IP 6000 functionalities in an environment comprised of Avaya Communication Manager, Avaya SIP Enablement Services, various Avaya H.323 and SIP IP Telephones.

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 procedures for configuring Polycom SoundStation IP 6000 which was compliance tested with Avaya Communication Manager and Avaya SIP Enablement Services (SES). The SoundStation IP 6000 is a SIP based IP conference phone that delivers superior performance for small to midsize conference rooms.

These Application Notes assume that Avaya Communication Manager and Avaya SES 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 [3].

1.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of the interoperability compliance testing was primarily on verifying call establishment on the Polycom SoundStation IP 6000. Polycom SoundStation IP 6000 operations such as inbound calls, outbound calls, hold, transfer, forward, conference, Feature Name Extension (FNE), VLAN tag, and Polycom SoundStation IP 6000 interactions with Avaya SIP Enablement Services (SES), Avaya Communication Manager, and Avaya SIP, H.323, and digital telephones were verified. The serviceability testing introduced failure scenarios to see if Polycom SoundStation IP 6000 can recover from failures.

1.2. Support

Technical documentation and software downloads for the SoundStation IP 6000 can be found at: http://www.polycom.com/support/voice/soundstation_ip_series/soundstation_ip6000.html

2. Reference Configuration

Figure 1 illustrates a sample configuration consisting of an Avaya S8300 Server, an Avaya G700 Media Gateway, an Avaya SIP Enablement Services (SES) server, and Polycom SoundStation IP 6000. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. Avaya S8720 Servers with an Avaya G650 Media Gateway were included in the test to provide an inter-switch scenario. For completeness, Avaya 4600 Series SIP IP Telephones, Avaya 4600 Series H.323 IP Telephones, Avaya 9600 Series SIP IP Telephones, Avaya 9600 Series H.323 IP Telephones, and Avaya 6400 Series Digital Telephones, are included in Figure 1 to demonstrate calls between the SIP-based Polycom SoundStation IP 6000 and Avaya SIP, H.323, and digital telephones.

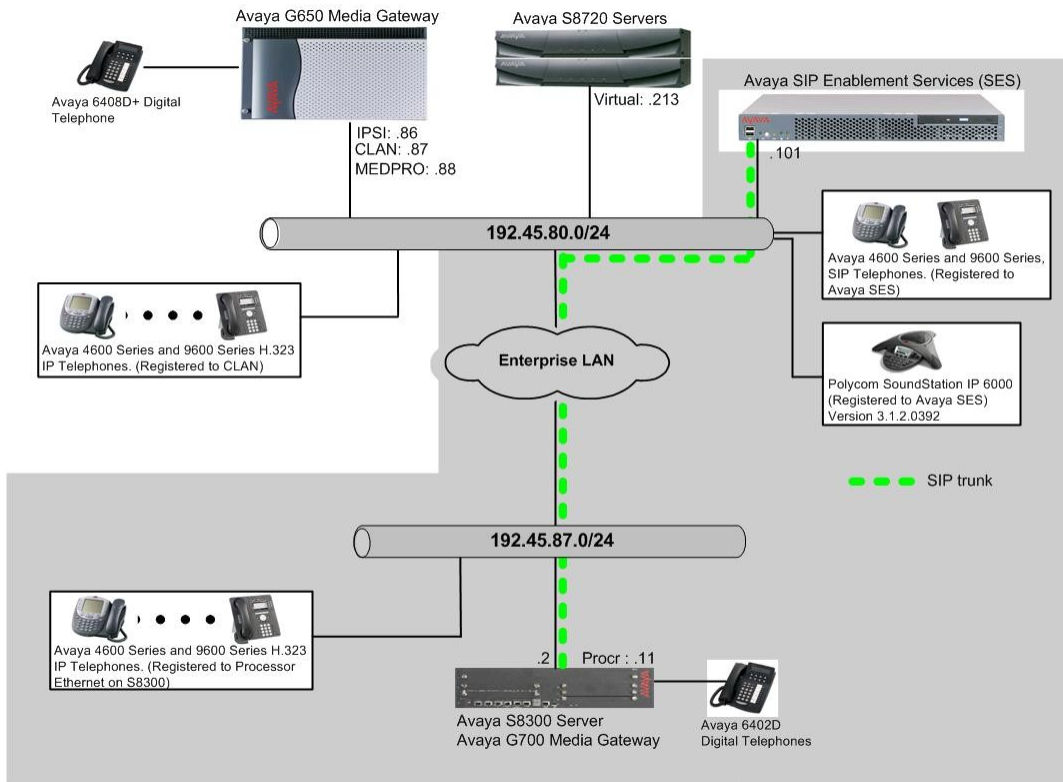


Figure 1: Test Configuration of Polycom SoundStation IP 6000

3. Equipment and Software Validated

The following equipment and software were used for the test configuration.

Equipment		Software/Firmware
Avaya S87020 Servers		Avaya Communication Manager 5.1.2 (R015x.01.2.416.4)
Avaya G650 Media Gateway		-
	TN2312BP IP Server Interface	HW11 FW044
	TN799DP C-LAN Interface	HW01 FW028
	TN2302AP IP Media Processor	HW20 FW118
Avaya S8300 Media Server with Avaya G700 Media Gateway		Avaya Communication Manager 5.1.2 (R015x.01.2.416.4)
Avaya SIP Enablement Services		SES05.1.2-01.2.416.4
Avaya 4600 and 9600 Series SIP Telephones		
	4610 (SIP)	2.2.2
	9620 (SIP)	2.0.5
	9630 (SIP)	2.0.5
Avaya 4600 and 9600 Series IP Telephones		
	4620 (H.323)	2.9
	4625 (H.323)	2.9
	9630 (H.323)	2.0
	9650 (H.323)	2.0
Avaya 6408D+ Digital Telephone		-
Polycom SoundStation IP 6000		3.1.2.0392

4. Configure the Avaya Communication Manager

This section describes the procedure for setting up a SIP trunk between Avaya Communication Manager and Avaya SES. The steps include setting up an IP codec set, an IP network region, IP node name, a signaling group, a trunk group, and a SIP station. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. Polycom SoundStation IP 6000 and other SIP telephones are configured as off-PBX telephones in Avaya Communication Manager.

4.1. Capacity Verification

Enter the **display system-parameters customer-options** command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses

```
display system-parameters customer-options                               Page 1 of 11
                                OPTIONAL FEATURES

G3 Version: V15                                                         Software Package: Standard
Location: 1                                                             RFA System ID (SID): 1
Platform: 22                                                            RFA Module ID (MID): 1

                                USED
Platform Maximum Ports: 900 52
Maximum Stations: 450 9
Maximum XMOBILE Stations: 0 0
Maximum Off-PBX Telephones - EC500: 50 0
Maximum Off-PBX Telephones - OPS: 100 5
Maximum Off-PBX Telephones - PBFMC: 0 0
Maximum Off-PBX Telephones - PVFMC: 0 0
Maximum Off-PBX Telephones - SCCAN: 0 0
```

On **Page 2** of the form, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

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

IP PORT CAPACITIES                                                    USED
Maximum Administered H.323 Trunks: 100 33
Maximum Concurrently Registered IP Stations: 450 3
Maximum Administered Remote Office Trunks: 0 0
Maximum Concurrently Registered Remote Office Stations: 0 0
Maximum Concurrently Registered IP eCons: 0 0
Max Concur Registered Unauthenticated H.323 Stations: 0 0
Maximum Video Capable H.323 Stations: 5 0
Maximum Video Capable IP Softphones: 5 0
Maximum Administered SIP Trunks: 100 10
Maximum Administered Ad-hoc Video Conferencing Ports: 0 0
Maximum Number of DS1 Boards with Echo Cancellation: 0 0
Maximum TN2501 VAL Boards: 0 0
Maximum Media Gateway VAL Sources: 0 0
Maximum TN2602 Boards with 80 VoIP Channels: 0 0
Maximum TN2602 Boards with 320 VoIP Channels: 0 0
Maximum Number of Expanded Meet-me Conference Ports: 0 0
```

4.2. IP Codec Set

This section describes the steps for administering a codec set in Avaya Communication Manager. This codec set is used in the IP network region for communications between Avaya Communication Manager and Avaya SES. Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 4.3** for configuring IP

network region to specify which codec sets may be used within and between network regions. For the compliance testing, G.711MU, G.729, G.722 and G.722.1 were tested for verification.

```
change ip-codec-set 1 Page 1 of 2

                                IP Codec Set

Codec Set: 1

Audio          Silence      Frames   Packet
Codec          Suppression Per Pkt  Size(ms)
1: G.711MU     n           2       20
2: G.729       n           2       20
3:
```

To configure a specific codec for Avaya 9600 Series SIP phones, the **46xxsettings.txt** file must be configured. The following shows the **CODEC SETTINGS** section in the 46xxsettings.txt file, that needs to be modified.

```
.
.
.
##### CODEC SETTINGS #####
##
## G.711a Codec Enabled
##   Determines whether G.711 a-law codec is available on
##   the phone.
##   0 for No
##   1 for Yes
## SET ENABLE_G711A 1 (This shows the default)
##
##Added the following statement:
SET ENABLE_G711A 0 (Modified G.711A codec to disable)
##
## G.711u Codec Enabled
##   Determines whether G.711 mu-law codec is available on
##   the phone.
##   0 for No
##   1 for Yes
## SET ENABLE_G711U 1 (This shows the default)
##
##Added the following statement:
SET ENABLE_G711u 0 (Modified G.711U codec to disable)
##
## G.729 Codec Enabled
##   Determines whether G.729 codec is available on the
##   phone.
##   0 for G.729(A) disabled
##   1 for G.729(A) enabled without Annex B support
##   2 for G.729(A) enabled with Annex B support
## SET ENABLE_G729 1 (This shows the default)
##e
##
## G.726 Codec Enabled
##   Determines whether G.726 codec is available on the
##   phone. This parameter is not supported on 16cc phones.
##   0 for No
##   1 for Yes
```

```

## SET ENABLE_G726 1 (This shows the default)
##
## G.726 Payload Type
## Specifies the RTP payload type to be used with the
## G.726 codec. (96-127). This parameter is not supported
## on 16cc phones.
## SET G726_PAYLOAD_TYPE 110
##
## G.722 Codec Enabled
## Determines whether G.722 codec is available on the
## phone. This parameter is not supported on 16cc phones.
## 0 for No
## 1 for Yes
## SET ENABLE_G722 0 (This shows the default)
SET ENABLE_G722 1
##
## DTMF Payload Type
## Specifies the RTP payload type to be used for RFC
## 2833 signaling. (96-127).
## SET DTMF_PAYLOAD_TYPE 120
##
## DTMF Transmission Method
## Specifies whether DTMF tones are sent in-band, as
## regular audio, or out-of-band, using RFC 2833
## procedures.
## 1 for in-band
## 2 for out-of-band using RFC 2833
## SET SEND_DTMF_TYPE 2
##
.
.
.

```

4.3. Configure IP Network Region

This section describes the steps for administering an IP network region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- Authoritative Domain – Enter the appropriate name for the Authoritative Domain. Set to the appropriate domain. During the compliance test, the authoritative domain is set to **testroom.com**. This should match the SIP Domain value on Avaya SES, in **Section 5.1**.
- Intra-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Communication Manager or Avaya SES in the same IP network region. The default value for this field is **yes**.
- Codec Set – Set the codec set number as provisioned in **Section 4.2**.
- Inter-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Communication Manager or Avaya SES in different IP network regions. The default value for this field is **yes**.

```

change ip-network-region 1                                     Page 1 of 19
                                                           IP NETWORK REGION
Region: 1
Location: Authoritative Domain: testroom.com
Name:
MEDIA PARAMETERS
Codec Set: 1
Intra-region IP-IP Direct Audio: yes
Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048
UDP Port Max: 3329
IP Audio Hairpinning? n
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
RTCP Reporting Enabled? y
RTCP MONITOR SERVER PARAMETERS
Use Default Server Parameters? y
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
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5

```

4.4. Configure IP Node Name

This section describes the steps for setting IP node name for Avaya SES in Avaya Communication Manager. Enter the **change node-names ip** command, and add a node name for Avaya SES along with its IP address.

```

change node-names ip                                         Page 1 of 2
                                                           IP NODE NAMES
Name                IP Address
CLAN                 192.45.80.87
IA770                192.45.87.12
SES                  192.45.80.101
default              0.0.0.0
procr                192.45.87.11

```

4.5. Configure SIP Signaling

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services. Enter the **add signaling-group <s>** command, where **s** is an available signaling group and configure the following:

- Group Type – Set to **sip**.
- Near-end Node Name - Set to **procr** as displayed in **Section 4.4**.
- Far-end Node Name - Set to the Avaya SES name configured in **Section 4.4**.
- Far-end Network Region - Set to the region configured in **Section 4.3**.
- Far-end Domain - Set to **testroom.com**. This should match the SIP Domain value in **Section 5.1**.


```

add signaling-group 3                                     Page 1 of 1

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

Near-end Node Name: procr                               Far-end Node Name: SES
Near-end Listen Port: 5061                             Far-end Listen Port: 5061
                                                         Far-end Network Region: 1
Far-end Domain: testroom.com

                                                         Bypass If IP Threshold Exceeded? n

DTMF over IP: rtp-payload                             Direct IP-IP Audio Connections? y
                                                         IP Audio Hairpinning? n

Enable Layer 3 Test? n
Session Establishment Timer(min): 3                    Alternate Route Timer(sec): 6

```

4.6. Configure SIP Trunk

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. Enter the **add trunk-group <t>** command, where **t** is an unallocated trunk group and configure the following:

- Group Type – Set the Group Type field to **sip**.
- Group Name – Enter a descriptive name.
- TAC (Trunk Access Code) – Set to any available trunk access code.
- Signaling Group – Set to the Group Number field value configured in **Section 4.5**.
- Number of Members – Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

Note: Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted

```

add trunk-group 3                                     Page 1 of 21

TRUNK GROUP

Group Number: 3                                         Group Type: sip
                                                         CDR Reports: y
Group Name: ToSES                                       COR: 1          TN: 1          TAC: 1003
Direction: two-way                                     Outgoing Display? n
Dial Access? n                                         Night Service:
Queue Length: 0
Service Type: tie                                     Auth Code? n

                                                         Signaling Group: 3
                                                         Number of Members: 10

```

4.7. Configure SIP Endpoint

This section describes the steps for administering OPS stations in Avaya Communication Manager and associating the OPS station extensions with the telephone numbers of Polycom SoundStation IP 6000. Enter **add station s**, where **s** is an extension valid in the provisioned dial plan. The following fields were configured for the compliance test.

- Type – Set to **9600SIP**.
- Name – Enter a descriptive name

Repeat this step as necessary to configure additional SIP endpoint extensions.

```
add station 27003                                     Page 1 of 5
                                                    STATION
Extension: 27003                                     Lock Messages? n          BCC: 0
Type: 9600SIP                                       Security Code:             TN: 1
Port: IP                                             Coverage Path 1:          COR: 1
Name: SIP 27003                                       Coverage Path 2:          COS: 1
                                                    Hunt-to Station:
STATION OPTIONS
Loss Group: 2                                         Time of Day Lock Table:
Data Module? n                                       Personalized Ringing Pattern: 1
Speakerphone: 2-way                                   Message Lamp Ext: 27003
Display Language: english                             Mute Button Enabled? y
Survivable COR: internal                               Media Complex Ext:
Survivable Trunk Dest? y                             IP SoftPhone? n
```

Enter the **add off-pbx-telephone station-mapping** command and configure the following:

- Station Extension – Set the extension of the OPS station as configured above.
- Application – Set to **OPS**.
- Phone Number – Enter the number that Polycom SoundStation IP 6000 will use for registration and call termination. In the example below, the Phone Number is the same as the Station Extension, but is not required to be the same.
- Trunk Selection – Set to the trunk group number configured in **Section 4.6**.
- Config Set – Set to 1

Repeat this step as necessary to configure additional off-pbx-telephone station-mapping.

```
add off-pbx-telephone station-mapping                Page 1 of 2
                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
Station      Application Dial  CC  Phone Number  Trunk  Config
Extension    Prefix                               Selection Set
27003       OPS      -    27003         3      1
```

The following Avaya feature name extension (FNE) set was utilized during the compliance test. Enter **change off-pbx-telephone feature-name-extensions set 1** to view the feature name extensions. The highlighted fields are tested during the compliance test.

EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

Set Name:

Active Appearance Select: 27051
Automatic Call Back: 27052
Automatic Call-Back Cancel: 27053
Call Forward All: 27054
Call Forward Busy/No Answer: 27055
Call Forward Cancel: 27056
Call Park: 27057
Call Park Answer Back: 27058
Call Pick-Up: 27059
Calling Number Block: 27060
Calling Number Unblock: 27061
Conference on Answer: 27062
Directed Call Pick-Up: 27063
Drop Last Added Party: 27064
Exclusion (Toggle On/Off): 27065
Extended Group Call Pickup:
Held Appearance Select: 27067

EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

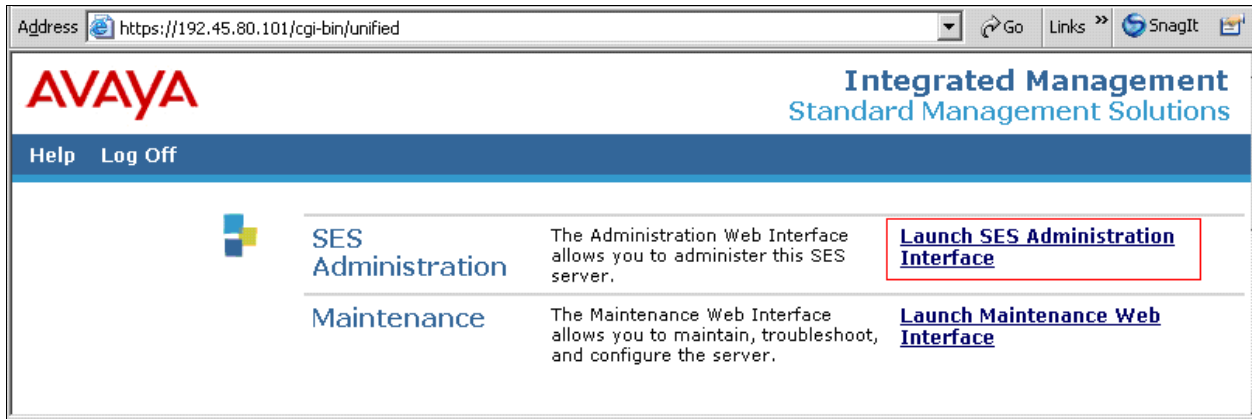
Idle Appearance Select: 27068
Last Number Dialed: 27069
Malicious Call Trace:
Malicious Call Trace Cancel:
Off-Pbx Call Enable: 27072
Off-Pbx Call Disable: 27073
Priority Call: 27074
Send All Calls: 27075
Send All Calls Cancel: 27076
Transfer On Hang-Up: 27077
Transfer to Voice Mail: 27078
Whisper Page Activation: 28079

5. Configure Avaya SIP Enablement Services

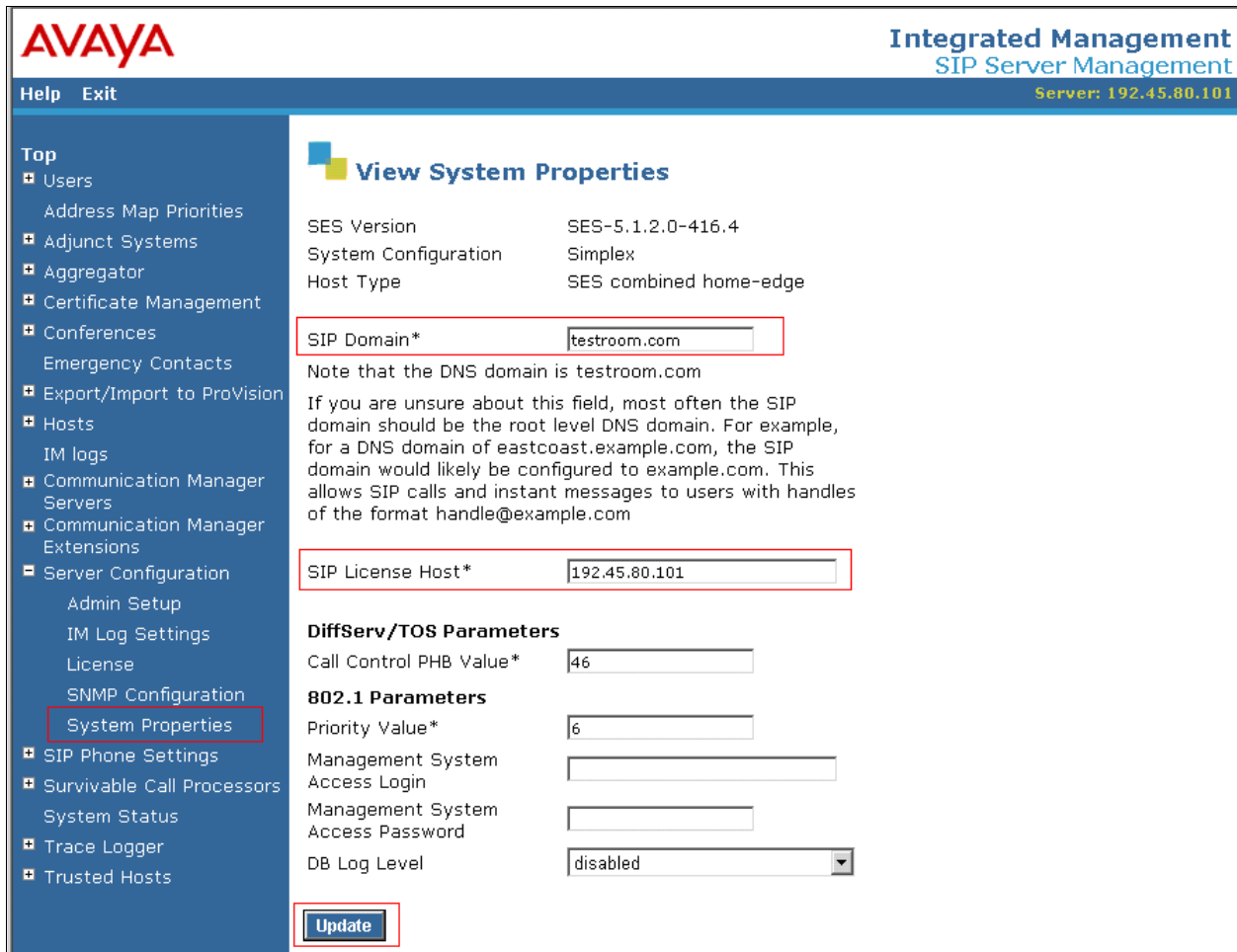
This section describes the steps for creating a SIP trunk between Avaya SES and Avaya Communication Manager. SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. Polycom SoundStation IP 6000 will register with Avaya SES using the SIP user accounts. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

5.1. Configure SES Server Properties

Launch a web browser, enter <https://<IP address of SES server>/admin> in the URL, and log in with the appropriate credentials. Click on the **Launch SES Administration Interface** link upon successful login.



In the Integrated Management SIP Server Management page, select the **Server Configuration** → **System properties** link from the left pane of the screen. Verify the SIP Domain matches the Far-end Domain field value configured for the signaling group on Avaya Communication Manager in **Section 4.5**. Click on the **Update** button, after the completion.



5.2. Configure Communication Manager Server Interface

This section provides steps to add SIP-enabled media servers to the SIP domain. In the Integrated Management SIP Server Management page, select the **Communication Manager Servers** → **Add** link from the left pane of the screen. The following screen shows the Add Media Server Interface page. The highlighted fields were configured for the compliance test:

- Communication Manager Server Interface Name – Enter a descriptive name for the communication manager server interface.
- SIP Trunk IP Address – Enter the IP address for the media server's procr (or CLAN) IP interface that terminates the SIP link from SES.

Click **Add** when finished.

AVAYA Integrated Management SIP Server Management Server: 192.45.80.101

Help Exit

Add Communication Manager Server Interface

Communication Manager Server Interface Name* S8300-G700

Host 192.45.80.101

SIP Trunk

SIP Trunk Link Type TCP TLS

SIP Trunk IP Address* 192.45.87.11

Communication Manager Server

Communication Manager Server Admin Address* (see Help)

Communication Manager Server Admin Port* 5022

Communication Manager Server Admin Login*

Communication Manager Server Admin Password*

Communication Manager Server Admin Password Confirm*

SMS Connection Type SSH Telnet Not Available

Note: If the Communication Manager Server connection type is changed and the admin port value is not also changed, changing connection type to SSH will change the admin port to 5022 when Add or Update is clicked and changing connection type to Telnet will change admin port to 5023 when Add or Update is clicked.

Fields marked * are required.

Add

5.3. Configure Users

This section provides steps to add users to be administered in the SIP Enablement Services (SES) database. In the Integrated Management SIP Server Management page, select the **Users** → **Add** link from the left pane of the screen. The highlighted fields were configured for the compliance test:

- Primary Handle – Enter the phone number of Polycom SoundStation IP 6000. This number was configured in **Section 4.7**.
- Password / Confirm Password – Enter a password; both field entries must match exactly.
- First Name – Enter the first name of the user in alphanumeric characters.
- Last Name – Enter the last name of the user in alphanumeric characters.
- Add Media Server Extension - Select this field if you want to associate a new extension number with this user in the database now. If so, the Add MS Extension screen will be displayed next, after this user profile has been added. If not, in the future you may choose to associate extensions with the user.

Click **Add** when finished.

The screenshot shows the Avaya Integrated Management SIP Server Management web interface. The page title is "Integrated Management SIP Server Management" with a server IP of "192.45.80.101". The main content area is titled "Add User" and contains a form with the following fields:

- Primary Handle*: 27003
- User ID: 27003
- Password*: *****
- Confirm Password*: *****
- Host*: 192.45.80.101
- First Name*: SIP
- Last Name*: 27003
- Address 1: [Empty]
- Address 2: [Empty]
- Office: [Empty]
- City: [Empty]
- State: [Empty]
- Country: [Empty]
- Zip: [Empty]
- Survivable Call Processor: none
- Add Communication Manager Extension:

Fields marked * are required. The "Add" button is located at the bottom left of the form area.

From the next screen, enter the numeric telephone extension you want to create in the database. Select the extension's Communication Manager server from the drop-down list. Click on the **Add** button.

6. Configure Polycom SoundStation IP 6000

This section provides steps to configure Polycom SoundStation IP 6000. The latest firmware was provided by Polycom. The following files are needed to be configured, as the phone boots up to register with Avaya SES:

- 0004F2E1C57D.cfg – The first file that the phone searches while booting up is <MAC>.cfg file. The header, **0004F2E1C57D**, indicates the MAC address of Polycom SoundStation IP 6000. In this configuration file, there are sub-configuration files that are listed under CONFIG_FILES field; 27003.cfg, phone1.cfg, sip.cfg. During the compliance test, the first file, which is 27003.cfg, was modified. The other two files were not modified.

```
<?xml version="1.0" standalone="yes"?>
<!-- Default Master SIP Configuration File-->
<!-- Edit and rename this file to <Ethernet-address>.cfg for each phone.-->
<!-- $Revision: 1.14 $ $Date: 2005/07/27 18:43:30 $ -->
<APPLICATION APP_FILE_PATH="sip.ld" CONFIG_FILES="27003.cfg, phone1.cfg, sip.cfg"
MISC_FILES="" LOG_FILE_DIRECTORY="" OVERRIDES_DIRECTORY="" CONTACTS_DIRECTORY=""/>
```

- 27003.cfg – This is an extension configuration file. This file includes UserID, Password, Fully Qualified Domain Name (FQDN) of the phone, and the IP address of Avaya SES.

```
<?xml version="1.0" encoding="UTF-8" standalone="yes"?>
<!-- Example Per-phone Configuration File -->
<!-- $RCSfile: phone1.cfg,v $ $Revision: 1.85.6.3 $ -->
<phone1>
  <reg reg.1.displayName="27003" reg.1.address="27003@testroom.com" reg.1.label=""
reg.1.type="private" reg.1.lcs="" reg.1.csta="" reg.1.thirdPartyName=""
reg.1.auth.userId="27003" reg.1.auth.password="123456"
reg.1.auth.optimizedInFailover="" reg.1.server.1.address="192.45.80.101"
reg.1.server.1.port="" reg.1.server.1.transport="UDPonly"
reg.1.server.2.transport="UDPonly" reg.1.server.1.expires=""
reg.1.server.1.expires.overlap="" reg.1.server.1.register=""
reg.1.server.1.retryTimeOut="" reg.1.server.1.retryMaxCount=""
reg.1.server.1.expires.lineSeize="" reg.1.server.1.lcs=""
reg.1.outboundProxy.address="" reg.1.outboundProxy.port=""
reg.1.outboundProxy.transport="" reg.1.acd-login-logout="0" reg.1.acd-agent-
available="0" reg.1.proxyRequire="" reg.1.ringType="2" reg.1.lineKeys=""
```

```
reg.1.callsPerLineKey="" reg.1.bargeInEnabled="" reg.1.serverFeatureControl.dnd=""
reg.1.serverFeatureControl.cf="" reg.1.strictLineSeize="" reg.2.displayName=""
.
.
.
```

7. General Test Approach and Test Results

The general test approach was to place calls to and from Polycom SoundStation IP 6000 and exercise basic telephone operations. The main objectives were to verify that:

- Polycom SoundStation IP 6000 successfully registers with Avaya SES.
- Successfully establish calls between Polycom SoundStation IP 6000 and Avaya SIP, H.323, and digital telephones attached to Avaya SES or Avaya Communication Manager.
- Polycom SoundStation IP 6000 successfully negotiates the right codec (G.711MU, G.711A, G.729, and G.722).
- Polycom SoundStation IP 6000 successfully holds a call.
- Polycom SoundStation IP 6000 successfully transfers a call (Origination, destination, and attended and unattended).
- Polycom SoundStation IP 6000 successfully establishes a three party conference call.
- Polycom SoundStation IP 6000 successfully verifies following FNE features:
 - Call Park
 - Call Pickup
 - Auto Redial
 - Last Number Dialed
 - Send All Calls
 - Call Forward (Unconditional, Busy/no answer)
 - Find Me

For serviceability testing, failures such as cable pulls and hardware resets were applied.

The test objectives were verified. For serviceability testing, the Polycom SoundStation IP 6000 operated properly after recovering from failures such as cable disconnects, and resets of the Polycom SoundStation IP 6000 and the Avaya SES server. Polycom SoundStation IP 6000 successfully negotiated the codec that was used. The features tested and worked as expected.

Between Polycom SoundStation IP 6000 and Avaya 9600 Series SIP telephones, the following codecs were verified for operation:

- G.711Mu
- G.711A
- G.729
- G.722

Calls between a Polycom SoundStation IP 6000 and a Polycom SoundStation IP 7000, using Avaya Communication Manger and Avaya SES, with a G.722.1 codec were also successful. Note that the Avaya 9600 Series SIP telephones do not support G.722.1 codec. Refer to **Section 4.2** to see the complete list of codecs that Avaya 9600 Series telephones support.

Note: By member's request, the MWI feature was not tested. Thus, steps for configuring voicemail and MWI are not included in the App Notes.

8. Verification Steps

The following steps may be used to verify the configuration:

The following steps may be used to verify the configuration:

- Verify that Polycom SoundStation IP 6000 successfully registers with Avaya SES server by following the **Users** -> **Registered Users** link on the SES Administration Web Interface.
- Place calls to and from Polycom SoundStation IP 6000 and verify that the calls are successfully established with two-way talk path.
- While calls are established, Enter **status trunk <t:r>** command, where **t** is the SIP trunk group configured in **Section 4.6**, and **r** is trunk group member. This will verify whether the call is shuffled or not.

9. Conclusion

Polycom SoundStation IP 6000 was compliance tested with Avaya Communication Manager (Version 5.1.2) and Avaya SES (Version 5.1.2). Polycom SoundStation IP 6000 (Version 3.1.2.0392) functioned properly for feature and serviceability. During compliance testing, Polycom SoundStation IP 6000 successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like three-way conference, transfers, hold, etc.

10. Additional References

The following Avaya product documentation can be found at <http://support.avaya.com>
[1] *Administrator Guide for Avaya Communication Manager*, Release 5.0, Issue 4, January 2008, Document Number 03-300509.
[2] *Administering SIP Enablement Services on the Avaya S8300 Server*, Issue 1.0, January 2008, Document Number 03-602508.

The following document was provided by Polycom.

[3] *Administrator's Guide for the Polycom SoundPoint IP/SoundStation IP Family*, August 2008, 1725-11530-310 Rev A SIP 3.1.

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