Avaya Solution & Interoperability Test Lab

Application Notes for Polycom® SpectraLink® 8400 Series Telephones and Avaya Aura® Communication Manager and Avaya Aura® SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Polycom® SpectraLink® 8400 Series Telephones which were compliance tested with Avaya Aura® Communication Manager and Avaya Aura® SIP Enablement Services.

The overall objective of the interoperability compliance testing is to verify Polycom® SpectraLink® 8400 Series Telephones functionalities in an environment comprised of Avaya Aura® Communication Manager, Avaya Aura® SIP Enablement Services, various Avaya H.323, SIP IP Telephones, and DCP 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® SpectraLink® 8400 Series Telephones (8440 and 8450) which were compliance tested with Avaya Aura® Communication Manager and Avaya Aura® SIP Enablement Services.

Polycom® SpectraLink® 8400 series Telephones (herein referred to as SpectraLink 8400 Series) improve productivity and responsiveness for on-site mobile professionals across a wide range of industries, including healthcare, retail, manufacturing and hospitality. Built on open standards, SpectraLink 8400 Series transforms the delivery of mobile enterprise applications by bringing the power of thin client and browser technology to front-line professionals in an easy-to-use and easy-to-manage interface. Additionally, SpectraLink 8400 Series supports a broad range of interfaces to enterprise-grade PBX, wireless LAN, and infrastructures to deliver maximum interoperability with the low cost of ownership.

These Application Notes assume that Communication Manager and SIP Enablement Services 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 reference [1], [2], and [3].

2. General Test Approach and Test Results

The general test approach was to place calls to and from SpectraLink 8400 Series and exercise basic telephone operations. The main objectives were to verify the following:

- Registration
- Codecs (G.711MU and G.729A)
- Inbound calls
- Outbound calls
- Hold/Resume
- Call termination (origination/destination)
- Transfer with Shuffling enabled (origination/destination/ attended/unattended)
- Transfer with Shuffling disabled (origination/destination/ attended/unattended)
- Three party conference (origination/destination)
- Avaya Feature Name Extension (FNE)
    - Call Park
    - Call Pickup
    - Call Forward (Unconditional, Busy/no answer)
- MWI
- Voicemail
- Serviceability

2.1. 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 SpectraLink 8400 Series. SpectraLink 8400 Series operations such as inbound calls, outbound calls, hold/resume, transfer, conference, Feature Name Extension (FNE), and SpectraLink 8400 Series interactions with SIP Enablement Services, Communication Manager, and Avaya SIP, H.323, and digital telephones were verified. The serviceability testing introduced failure scenarios to see if SpectraLink 8400 Series can recover from failures.

2.2. Test Results

The test objectives were verified. For serviceability testing, SpectraLink 8400 Series operated properly after recovering from failures such as cable disconnects, and resets of SpectraLink 8400 Series and SIP Enablement Services. SpectraLink 8400 Series successfully negotiated the codec that was used. The features tested worked as expected.

2.3. Support

Technical support on SpectraLink 8400 Series can be obtained through the following:

- **Phone:** (978) 292-5000, and select Option 3.
- **Web:** http://www.polycom.com/support/index.html

3. Reference Configuration

Figure 1 illustrates a sample configuration consisting of an Avaya S8720 Servers, an Avaya G650 Media Gateway, a SIP Enablement Services server, and SpectraLink 8400 Series. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. Avaya S8300D Server with an Avaya G450 Media Gateway was included in the test to provide an inter-switch scenario. For completeness, an Avaya 4600 Series H.323 IP Telephone, 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 SpectraLink 8400 Series and Avaya SIP, H.323, and digital telephones.

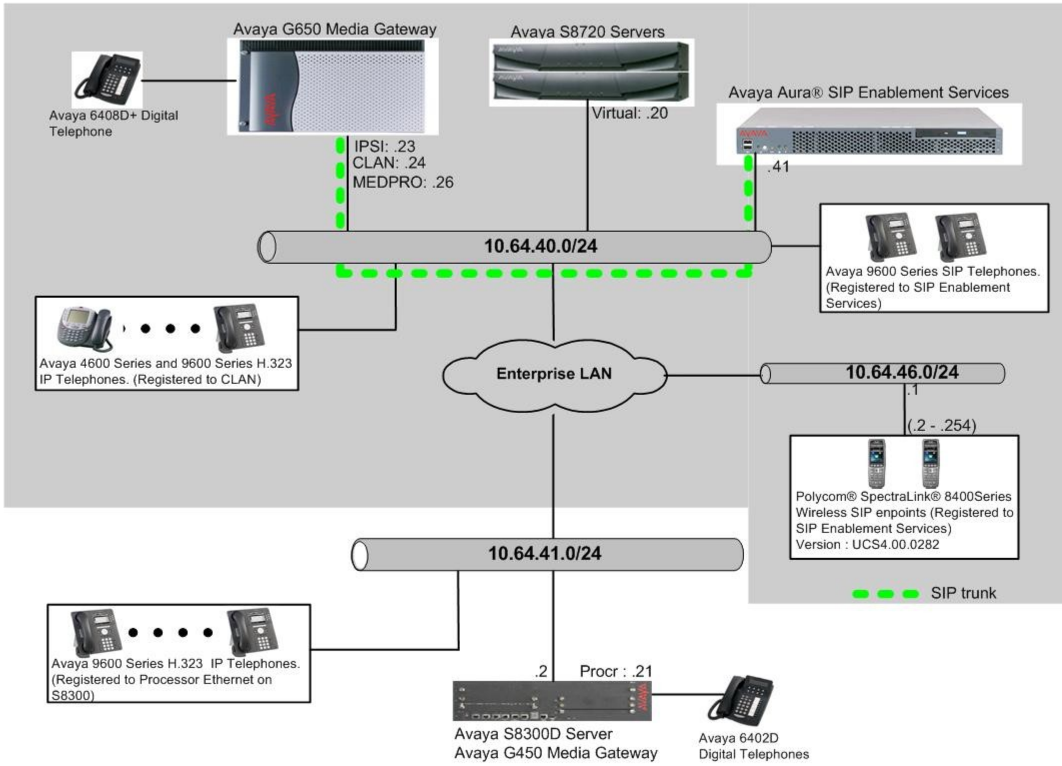


Figure 1: Test Configuration of SpectraLink 8400 Series

4. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S87020 Servers	Avaya Aura® Communication Manager 5.2.1 (R015x.02.1.016.4)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW11 FW044
TN799DP C-LAN Interface	HW01 FW028
TN2302AP IP Media Processor	HW20 FW118
Avaya Aura® SIP Enablement Services	SES-5.2.1.0-016.4
Avaya S8300D Media Server with Avaya G450 Media Gateway	Avaya Aura® Communication Manager 6.0.1 (R016x.00.1.510.1) with SP2 (00.1.510.1-18860)
Avaya 4600 and 9600 Series SIP Telephones	
9620 (SIP)	2.6.4
9630 (SIP)	2.6.4
9650 (SIP)	2.6.4
Avaya 4600 and 9600 Series H.323 Telephones	
4625 (H.323)	2.9
9620 (H.323)	3.1
9630 (H.323)	3.1
9650 (H.323)	3.1
Avaya 6408D+ Digital Telephone	-
SpectraLink 8400 Series	UCS 4.0.0.10555

5. Configure the Avaya Aura® Communication Manager

This section describes the procedure for setting up a SIP trunk between Communication Manager and SIP Enablement Services. 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 Communication Manager System Access Terminal (SAT) interface. SpectraLink 8400 Series and other SIP telephones are configured as off-PBX telephones in Communication Manager.

5.1. 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: 6                              RFA Module ID (MID): 1

                                                             USED
                             Platform Maximum Ports: 44000 10273
                                   Maximum Stations: 36000 10122
                            Maximum XMOBILE Stations: 0     0
                  Maximum Off-PBX Telephones - EC500: 50    0
                  Maximum Off-PBX Telephones -   OPS: 100   4
                  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   44
           Maximum Concurrently Registered IP Stations: 18000 4
             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: 5     0
                  Maximum Video Capable H.323 Stations: 5     0
                   Maximum Video Capable IP Softphones: 5     0
                       Maximum Administered SIP Trunks: 100   50
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   1
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

5.2. IP Codec Set

This section describes the steps for administering a codec set in Communication Manager. This codec set is used in the IP network region for communications between Communication Manager and SIP Enablement Services. 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 5.3** 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.729A 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:
 3:
 4:
```

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
##
## 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
##
##
## 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
##
```

```
## 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
##
## 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
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
##
.
.
.
```

5.3. Configure IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and SIP Enablement Services. 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 **avaya.com**. This should match the SIP Domain value on SIP Enablement Services, in **Section 6.1**.
- **Intra-region IP-IP Direct Audio** – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SIP Enablement Services 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 5.2**.
- **Inter-region IP-IP Direct Audio** – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SIP Enablement Services 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: 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? n
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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
```

5.4. Configure IP Node Name

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

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
ASM                 10.64.40.42
CLAN                10.64.40.24
CLAN-AES            10.64.40.25
G450                10.64.41.21
MEDPRO              10.64.40.26
MM-MAS              10.64.20.63
S8300               10.64.42.21
SES                 10.64.40.41
```

5.5. Configure SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for communication between Communication Manager and 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 **CLAN** as displayed in **Section 5.4**.
- **Far-end Node Name** - Set to the SIP Enablement Services name configured in **Section 5.4**.
- **Far-end Network Region** - Set to the region configured in **Section 5.3**.
- **Far-end Domain** - Set to **avaya.com**. This should match the SIP Domain value in **Section 6.1**.

- Direct IP-IP Audio Connections – Set to **y**, since Media Shuffling is enabled during the compliance test

```
add signaling-group 201                                         Page   1 of   1
                                SIGNALING GROUP

 Group Number: 201             Group Type: sip
                        Transport Method: tls
  IMS Enabled? n



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

                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                       Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

5.6. Configure SIP Trunk

This section describes the steps for administering a trunk group in Communication Manager for communication between Communication Manager and SIP Enablement Services. 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 5.5**.
- **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 201                                             Page   1 of  21
                                TRUNK GROUP

Group Number: 201                   Group Type: sip          CDR Reports: y
  Group Name: To SES                       COR: 1        TN: 1        TAC: 116
   Direction: two-way         Outgoing Display? y
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                      Auth Code? n

                                                       Signaling Group: 201
                                                     Number of Members: 10
```

5.7. Configure SIP Endpoint

This section describes the steps for administering OPS stations in Communication Manager and associating the OPS station extensions with the telephone numbers of SpectraLink 8400 Series. 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 **9630SIP**.
- **Name** – Enter a descriptive name

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

```
Add station 28002                                               Page   1 of   6
                                STATION

Extension: 28002                    Lock Messages? n                BCC: 0
     Type: 9630SIP                   Security Code:                  TN: 1
     Port: S30168                 Coverage Path 1: 99               COR: 1
     Name: WiFi-1                 Coverage Path 2:                  COS: 1
                                  Hunt-to Station:
STATION OPTIONS
                                        Time of Day Lock Table:
              Loss Group: 19
                                              Message Lamp Ext: 28002

        Display Language: english               Button Modules: 0

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

On **Page 6** of the STATION form, enter the trunk group number that was assigned in **Section 5.6** for the SIP Trunk field.

```
add station 28002                                               Page   6 of   6
                                STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                    SIP Trunk: 201
```

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 SpectraLink 8400 Series 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 5.6**.
- **Config Set** – Set to 1

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

```
add off-pbx-telephone station-mapping 28003                     Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number     Trunk       Config  Dual
 Extension                 Prefix                      Selection   Set     Mode
 28002            OPS              -      28002          201         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.

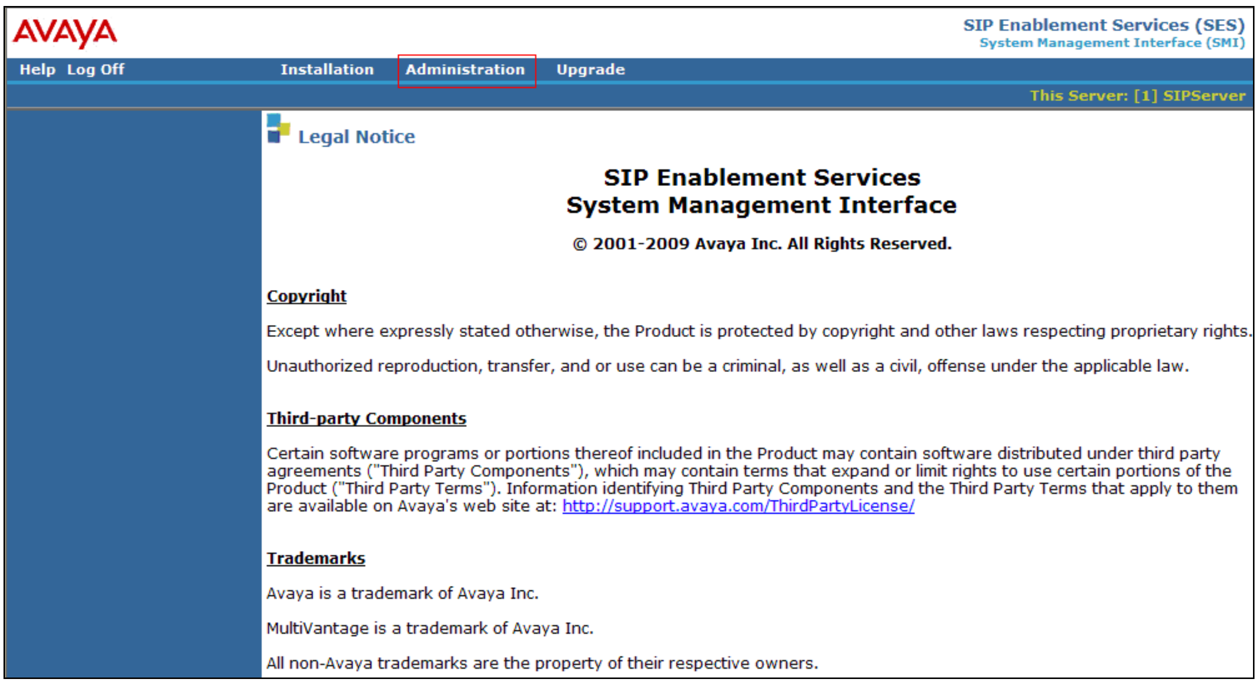
```
change off-pbx-telephone feature-name-extensions set 1          Page   1 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                     Set Name:

     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
```

6. Configure SIP Enablement Services

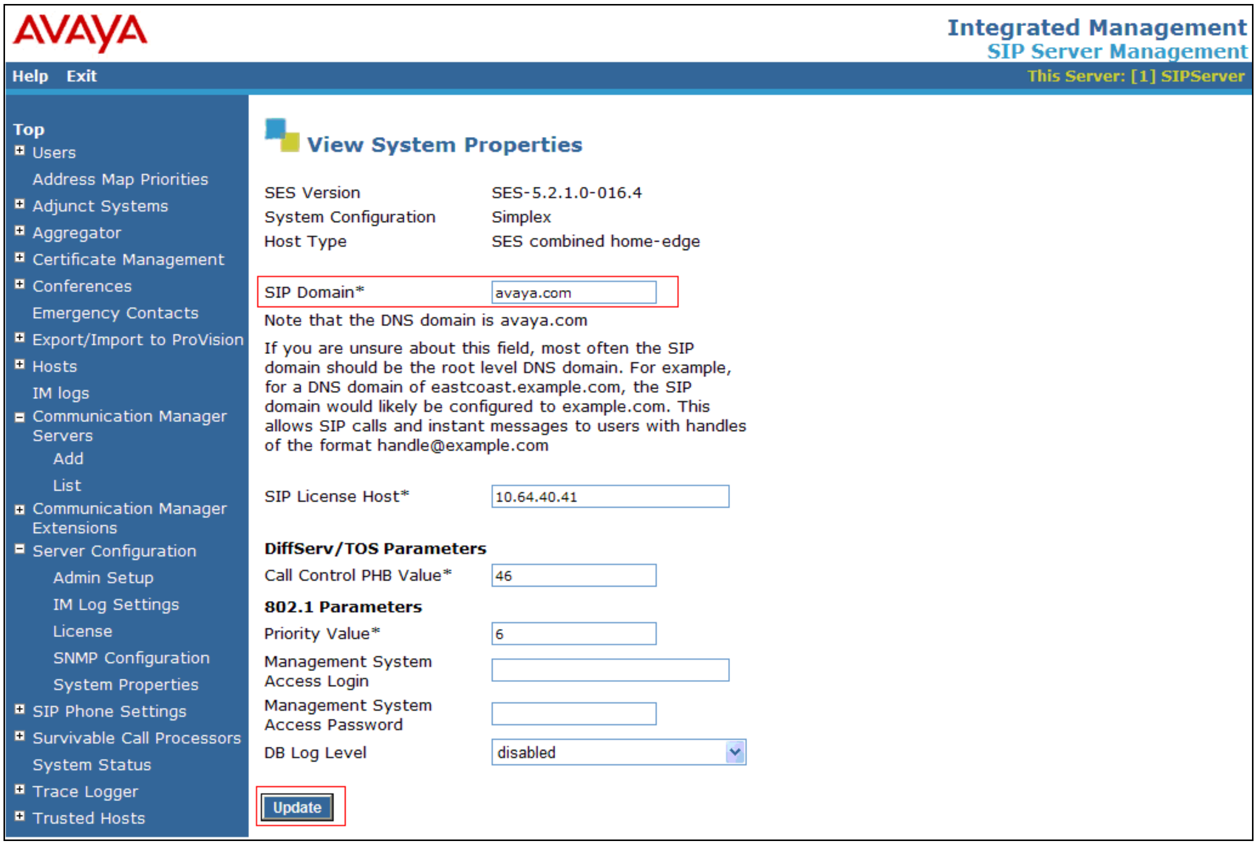
This section describes the steps for creating a SIP trunk between SIP Enablement Services and Communication Manager. SIP user accounts are configured in SIP Enablement Services and associated with a Communication Manager OPS station extension. SpectraLink 8400 Series will register with SIP Enablement Services 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.

6.1. Configure SIP Enablement Services Server Properties

Launch a web browser, enter **https://<IP address of SIP Enablement Services server>/admin** in the URL, and log in with the appropriate credentials. Navigate to **Administration → SIP Enablement Services** upon successful login.



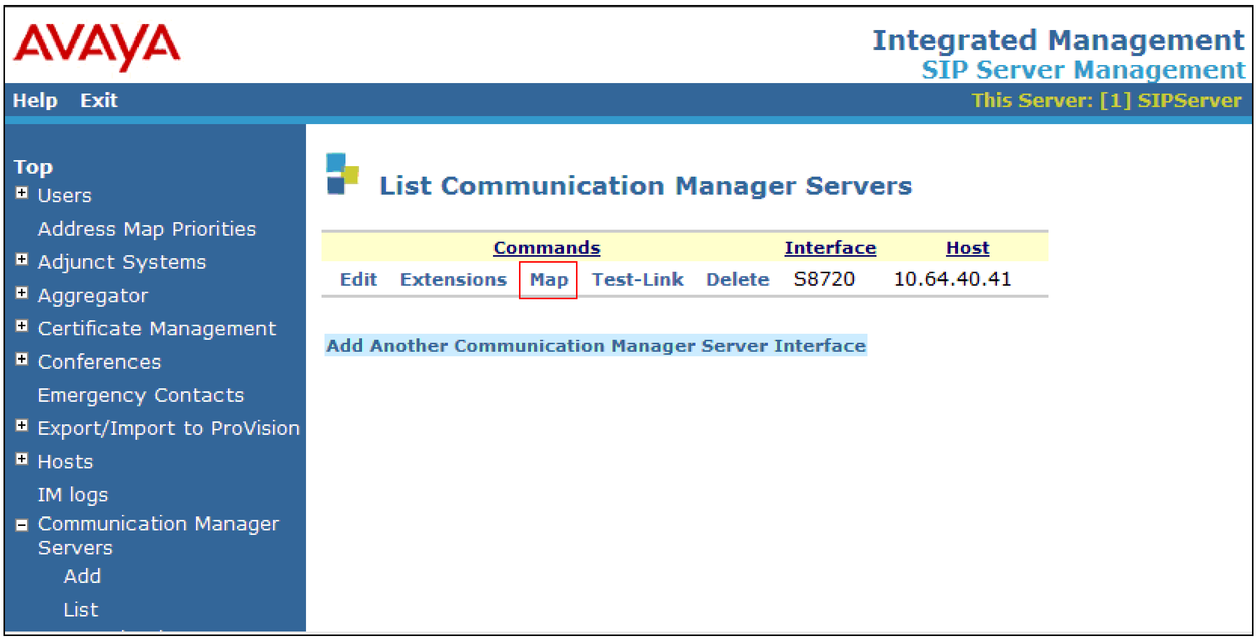
In the **Integrated Management SIP Server Management** page, select the **Server Configuration → System properties** link from the left pane of the screen. Verify the SIP Domain matches the Far-end Domain field value configured for the signaling group on Communication Manager in **Section 5.5**. Click on the **Update** button, after the completion.



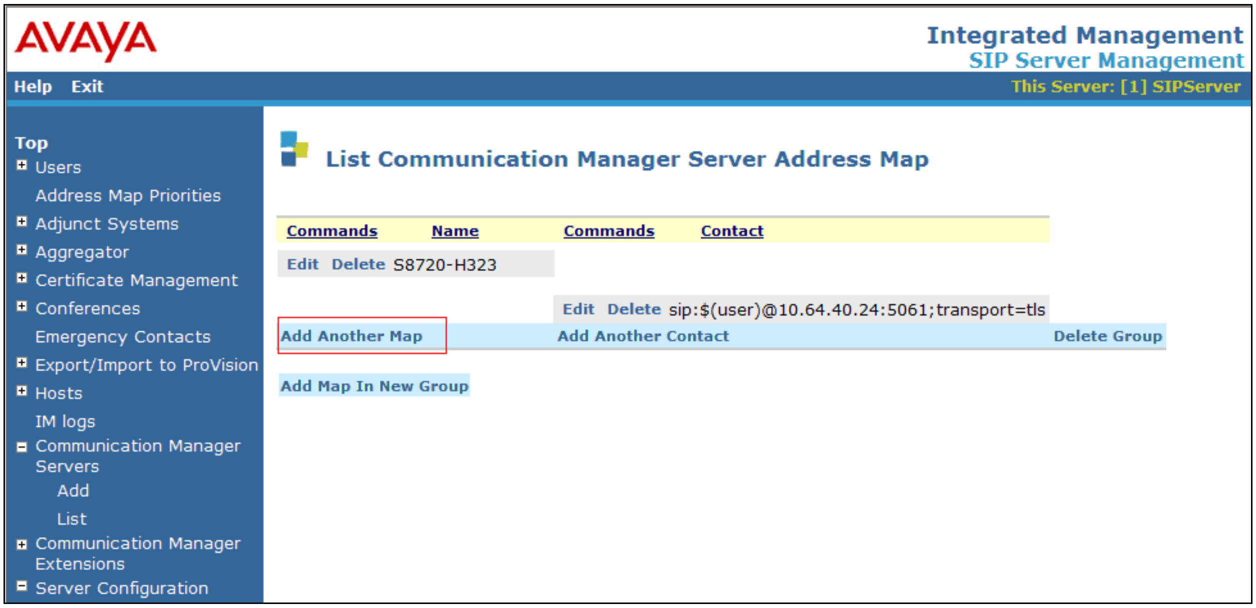
6.2. Configure Communication Manager Server Interface

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 → List** link from the left pane of the screen. The following screen shows the **List Communication Manager Servers** page. Prior to the compliance test, the Communication Manger Server Interface was already configured.

Click the **Map** link to go to the Add Address Map screen for that Communication Manager server.



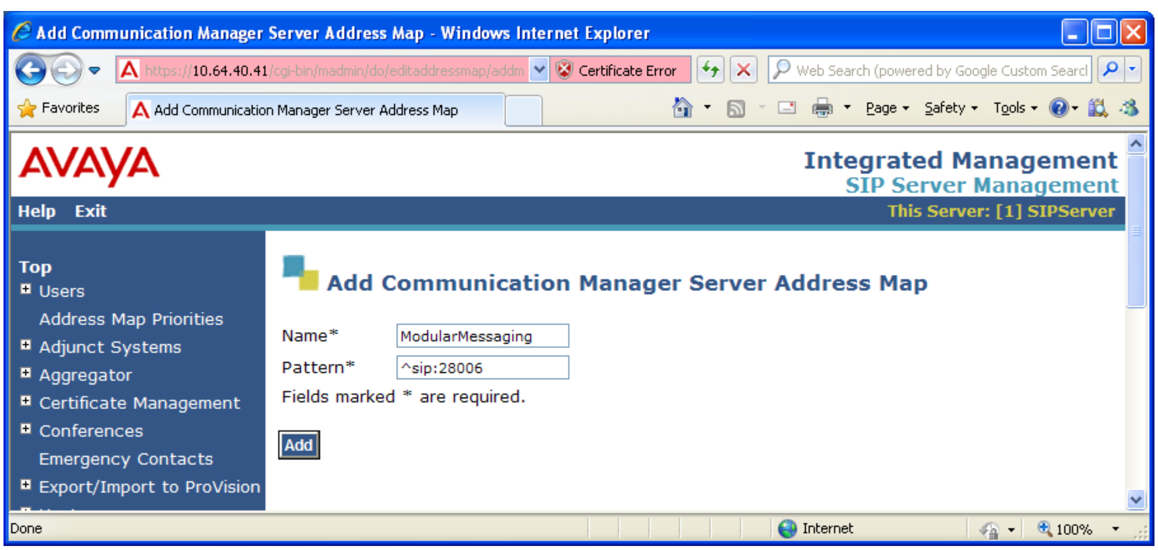
The following screen displays the **Communication Manager Address Map** page. To add new address map to associate with Communication Manager Server, click on **Add Another Map**.



In the **Add Communication Manager Server Address Map** screen, provide the following information:

- **Name** – Enter a descriptive name for this map.
- **Pattern** –

Click on the **Add** button.

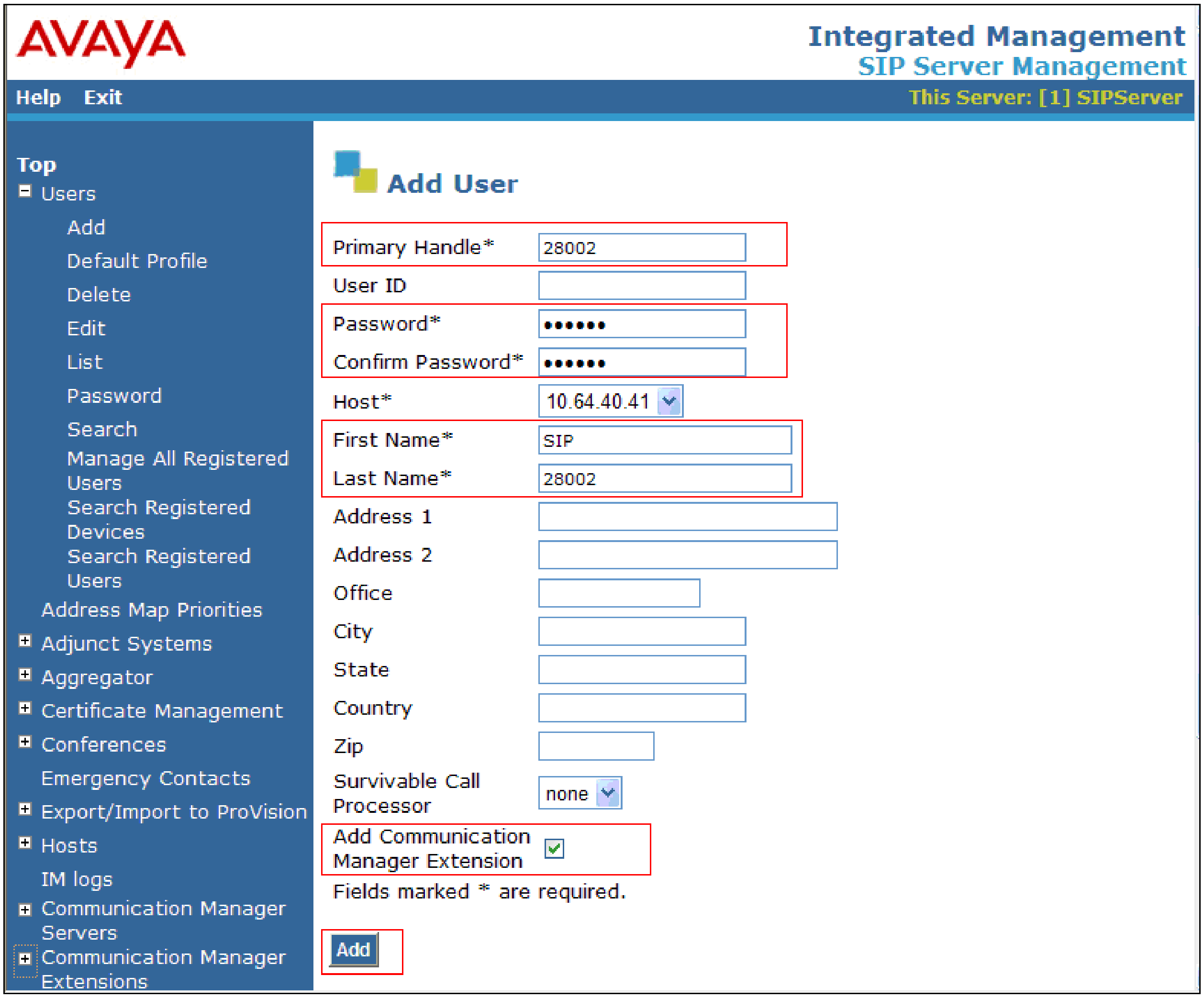


6.3. Configure Users

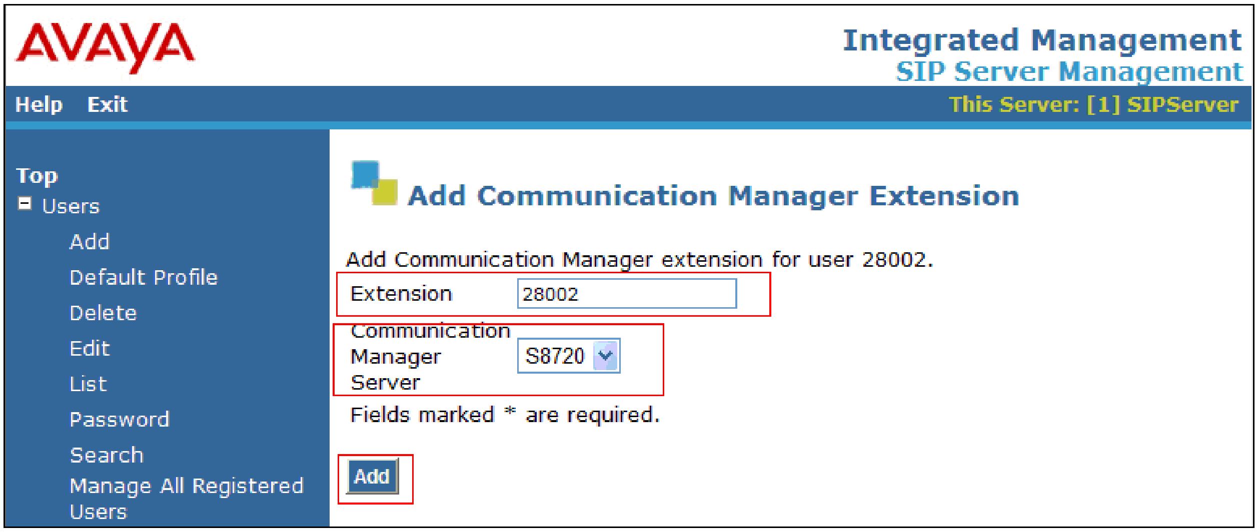
This section provides steps to add users to be administered in the SIP Enablement Services database. In the **Integrated Management SIP Server Management** page, select the **Users → Add** link from the left pane of the screen. The highlighted fields were configured for the compliance test:

- **Primary Handle** – Enter the phone number of SpectraLink 8400 Series. This number was configured in **Section 5.7**.
- **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 Communication Manager Extension** - Select this field if you want to associate a new extension number with this user in the database now. If so, the **Add Communication Manager 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.


AVAYA
Integrated Management
SIP Server Management
Help Exit
This Server: [1] SIPServer
Top
Users
Add
Default Profile
Delete
Edit
List
Password
Search
Manage All Registered Users
Search Registered Devices
Search Registered Users
Address Map Priorities
Adjunct Systems
Aggregator
Certificate Management
Conferences
Emergency Contacts
Export/Import to ProVision
Hosts
IM logs
Communication Manager Servers
Communication Manager Extensions
Add User
Primary Handle* 28002
User ID
Password* ••••••
Confirm Password* ••••••
Host* 10.64.40.41
First Name* SIP
Last Name* 28002
Address 1
Address 2
Office
City
State
Country
Zip
Survivable Call Processor none
Add Communication Manager Extension
Fields marked * are required.
Add

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.



7. Configure SpectraLink 8400 Series

This section provides steps to configure SpectraLink 8400 Series. The latest firmware was provided by Polycom SpectraLink. For additional information regarding configuring the SpectraLink 8400 series handsets please refer to the latest product documentation available at www.polycom.com. The following files need to be configured, as the phone boots up to register with SIP Enablement Services:

- **00907a0cd950.cfg** – The first file that the phone searches while booting up is **<MAC>.cfg** file. The header, **00907a0cd950**, indicates the MAC address of SpectraLink 8400 Series. In this configuration file, there are sub-configuration files that are listed under CONFIG_FILES field; sip_28002.cfg. During the compliance test, sip_28002.cfg was modified.

```
<?xml version="1.0" encoding="utf-8" 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" APP_NET_LOAD_FILE_PATH=""
CONFIG_FILES="sip_28002.cfg" MISC_FILES="" LOG_FILE_DIRECTORY=""
OVERRIDES_DIRECTORY="" CONTACTS_DIRECTORY="" />
```

- **Sip_28002.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 SIP Enablement Services.

```
<?xml version="1.0" encoding="utf-8"?>
<PHONE_CONFIG>
  <reg reg.1.address="28002@avaya.com" reg.1.displayName="28002" reg.1.label="28002"
reg.1.auth.userId="28002" reg.1.auth.password="123456"
reg.1.server.1.address="10.64.40.41" reg.1.server.1.port="5060" />
<msg.mwi msg.mwi.1.subscribe="28002@avaya.com" />
```

```
</PHONE_CONFIG>
```

8. Verification Steps

The following steps may be used to verify the configuration:

- Verify that SpectraLink 8400 Series successfully registers with SIP Enablement Services server by following the **Users → Registered Users** link on the SES Administration Web Interface.
- Place calls to and from SpectraLink 8400 Series 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 5.6**, and **r** is trunk group member. This will verify whether the call is shuffled or not.

9. Conclusion

SpectraLink 8400 Series was compliance tested with Communication Manager (Version 5.2.1) and SIP Enablement Services (Version 5.2.1). SpectraLink 8400 Series (UCS 4.0.0.10555) functioned properly for feature and serviceability. During compliance testing, SpectraLink 8400 Series successfully registered with SIP Enablement Services, 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] *Administering Avaya Aura™ Communication Manager,* Release 5.2, Issue 5.0, May 2009, Document Number 03-300509.
[2] *Administering Avaya Aura™ SIP Enablement Services on the Avaya S8300 Server,* Issue 2.0, May 2009, Document Number 03-602508.

The following document was provided by Polycom SpectraLink.

[3] *Polycom® SpectraLink® 8400 Series Wireless Handset User Guide*, February 2011, 1725-36720-001 Rev A

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.