Avaya Solution & Interoperability Test Lab

Application Notes for Polycom® SpectraLink® 8400 Series Telephones and Avaya Aura® Communication Manager and Avaya Aura® Session Manager – 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® Session Manager.

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® Session Manager, 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® Session Manager.

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 Session Manager are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult references [1], [2], [3], and [4].

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 Session Manager, 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 Session Manager. 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 S8300D Server, an Avaya G450 Media Gateway, a Session Manager server, and SpectraLink 8400 Series. 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, 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.

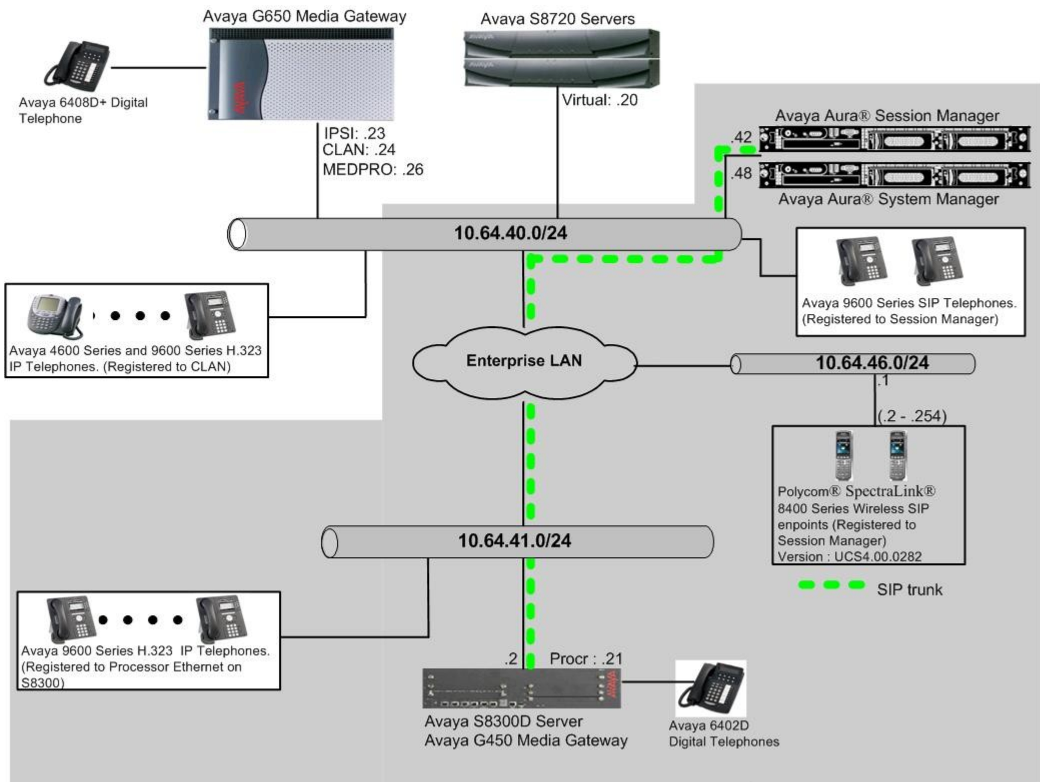


Figure 1: Test Configuration of SpectraLink 8400 Series

4. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya 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 Aura® System Manager	6.1.5.0
Avaya Aura® Session Manager	6.1.1.0.611023
Avaya S8720 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 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 Session Manager. 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: V16                                 Software Package: Enterprise
       Location: 2                                    System ID (SID): 1
       Platform: 28                                   Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 6400  130
                                      Maximum Stations: 2400  24
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 9600  1
                    Maximum Off-PBX Telephones -   OPS: 9600  10
                    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   1
```

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: 4000  30
           Maximum Concurrently Registered IP Stations: 2400  4
             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  1
                   Maximum Video Capable IP Softphones: 2400  1
                       Maximum Administered SIP Trunks: 4000  40
  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    1
           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
```

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 Session Manager. 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
##
##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 Session Manager. 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 Session Manager, 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 Session Manager 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 Session Manager 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 Session Manager in Communication Manager. Enter the **change node-names ip** command, and add a node name for Session Manager 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
SM-2                10.64.21.31
```

5.5. Configure SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for communication between Communication Manager and Session Manager. 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**.
- **Far-end Node Name** - Set to the Session Manager 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 92                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 92             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n                                            SIP Enabled LSP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                  Far-end Node Name: SM-2
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         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? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 3
```

5.6. Configure SIP Trunk

This section describes the steps for administering a trunk group in Communication Manager for communication between Communication Manager and Session Manager. 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.

```
change trunk-group 92                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 92                    Group Type: sip          CDR Reports: y
  Group Name: SM-21.31                     COR: 1        TN: 1          TAC: 1092
   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: 92
                                                     Number of Members: 10
```

5.7. Configure SIP Endpoint

SIP endpoints and off-pbx-telephone stations will be automatically created in Communication Manager when users (SIP endpoints) are created in System manager.
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 Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager as provisioned in the reference configuration. Session Manager is comprised of two functional components: the Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

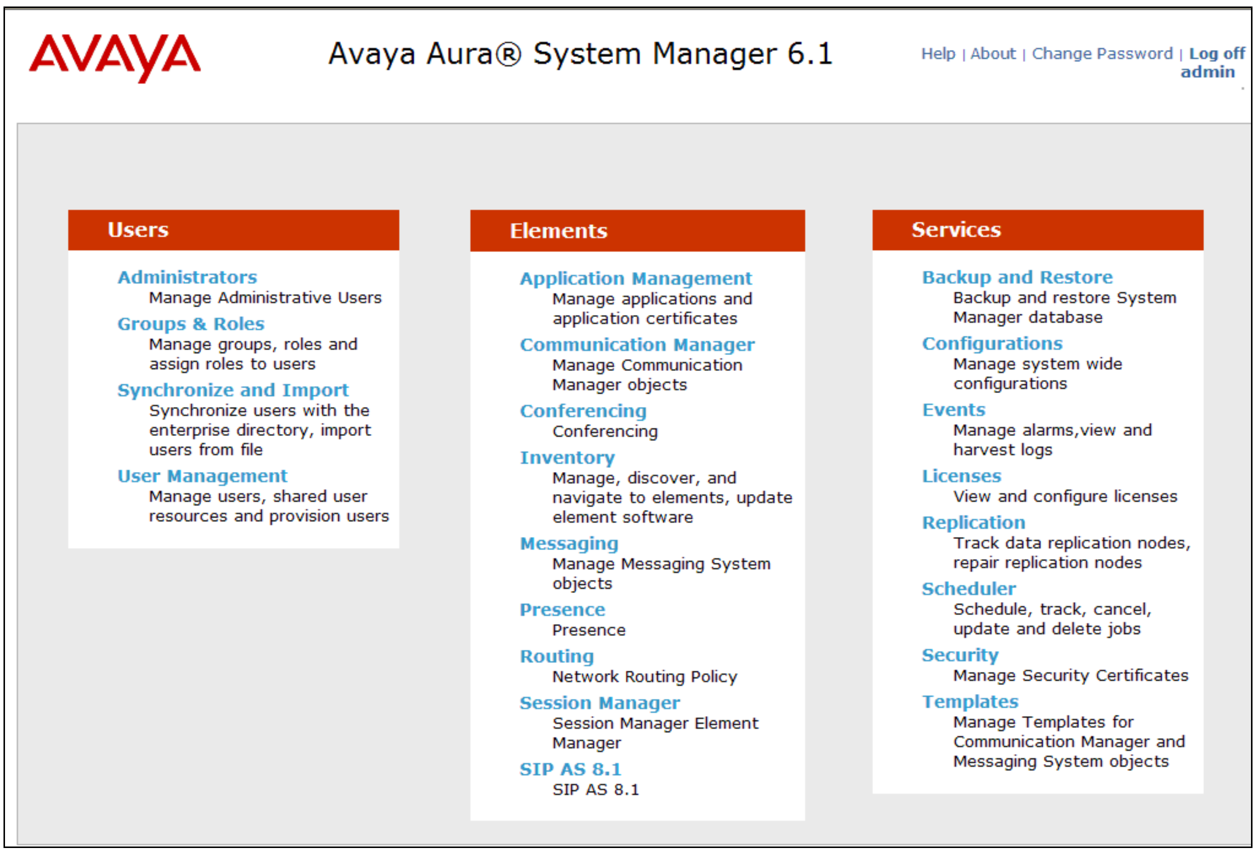
The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

In this section, the following topics are discussed:

- SIP Domains
- Locations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policy
- Dial Patterns
- User Management

6.1. Configure SIP Domain

Launch a web browser, enter **http://<IP address of System Manager>** in the URL, and log in with the appropriate credentials.

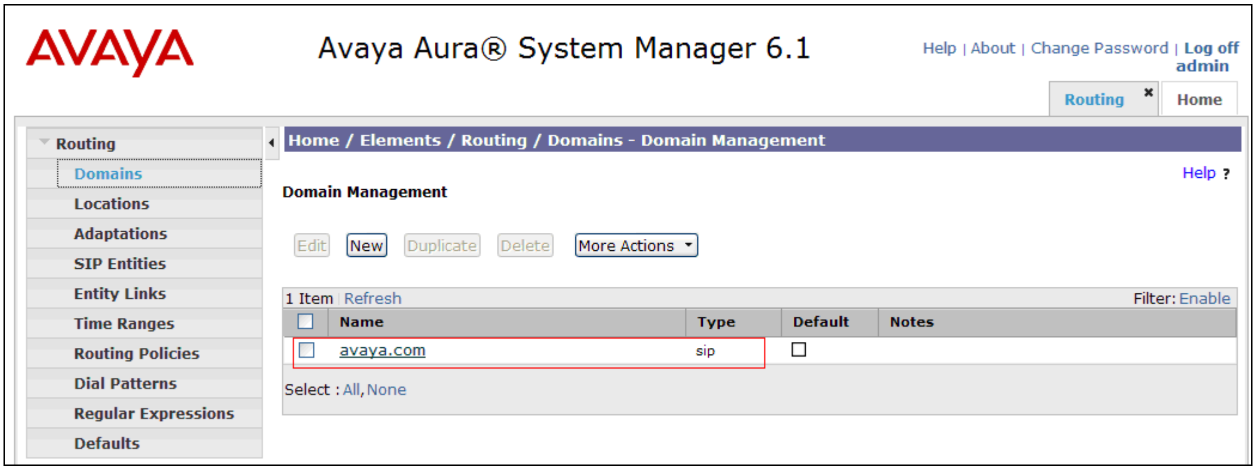


In the main menu, navigate to **Elements → Routing → Domains**, and click on the **New** button (not shown) to create a new SIP Domain. Enter the following values and use default values for remaining fields:

- **Name –** Enter the Authoritative Domain Name specified in **Section 5.3**, which is **avaya.com**.
- **Type** – Select **SIP**

Click **Commit** to save.

The following screen shows the Domains page used during the compliance test.



6.2. Configure Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing.

From the main menu, navigate to **Elements** → **Routing** → **Locations**, and click on the **New** button (not shown) to create a new SIP endpoint location.

General section

Enter the following values and use default values for remaining fields.

- Enter a descriptive Location name in the **Name** field (e.g. **D4H26**).
- Enter a description in the **Notes** field if desired.

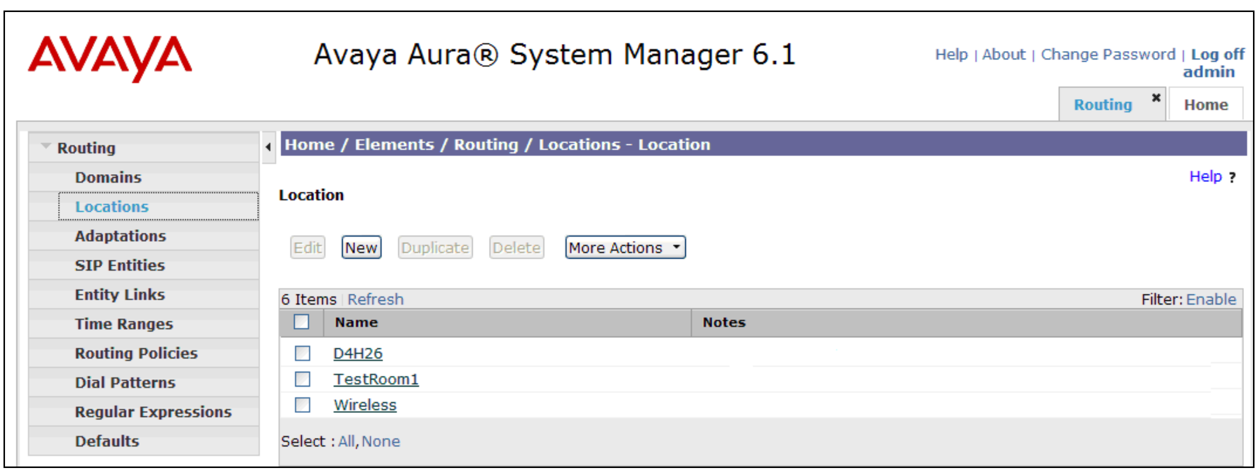
Location Pattern section

Click **Add** and enter the following values:

- Enter the IP address information for the **IP address Pattern** field (e.g. **10.64.40.***).
- Enter a description in the **Notes** field if desired.

Repeat steps in the Location Pattern section if the Location has multiple IP segments. Modify the remaining values on the form, if necessary; otherwise, use all the default values. Click on the **Commit** button.

The following screen shows the Locations list used during the compliance test.



6.3. Configure SIP Entities

A SIP Entity must be added for Session Manager and for each network component that has a SIP trunk provisioned to Session Manager. During the compliance test, the following SIP Entities were configured:

- Session Manager itself. This entity was created prior to the compliance test.
- Communication Manager. This entity was created prior to the compliance test.

Navigate to **Routing → SIP Entities**, and click on the **New** button (not shown) to create a new SIP entity. Provide the following information:

<u>General section</u>

Enter the following values and use default values for remaining fields.

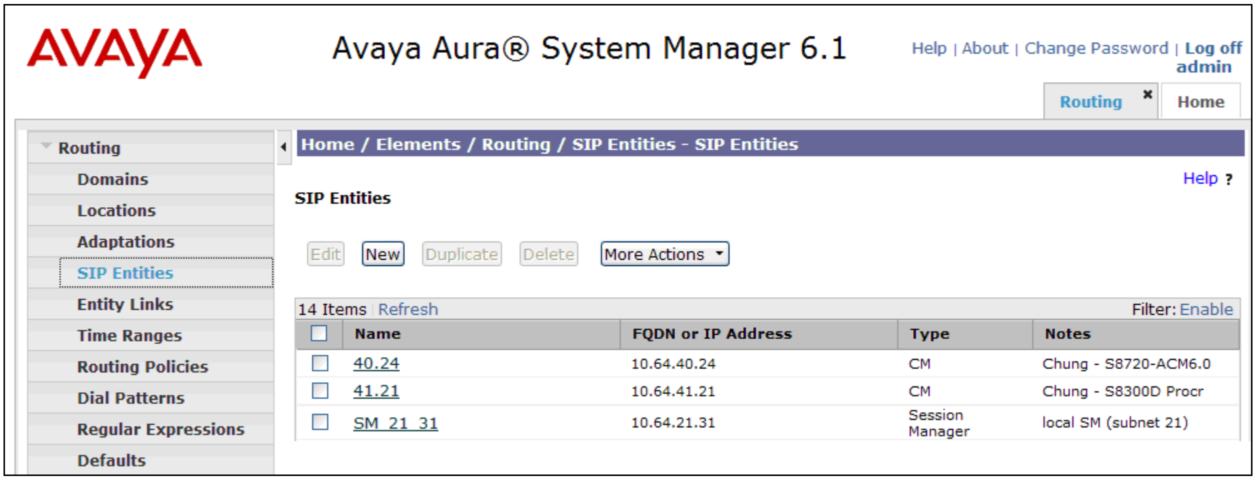
- Enter a descriptive Entity name in the **Name** field.
- Enter IP address for signaling interface on each Communication Manager, virtual SM-100 interface on Session Manager, or 3rd party device in the **FQDN or IP Address** field
- From the **Type** drop down menu select a type that best matches the SIP Entity.
  - For Communication Manager, select CM
  - For Session Manager, select Session Manager
- Enter a description in the **Notes** field if desired.
- Select the appropriate time zone.
- Accept the other default values.

<u>SIP Link Monitoring section</u>

- Accept the other default values.

Click on the **Commit** button to save each SIP entity.
The following screen shows the SIP Entities page used during the compliance test.

Repeat all the steps for each new entity.



6.4. Configure Entity Links

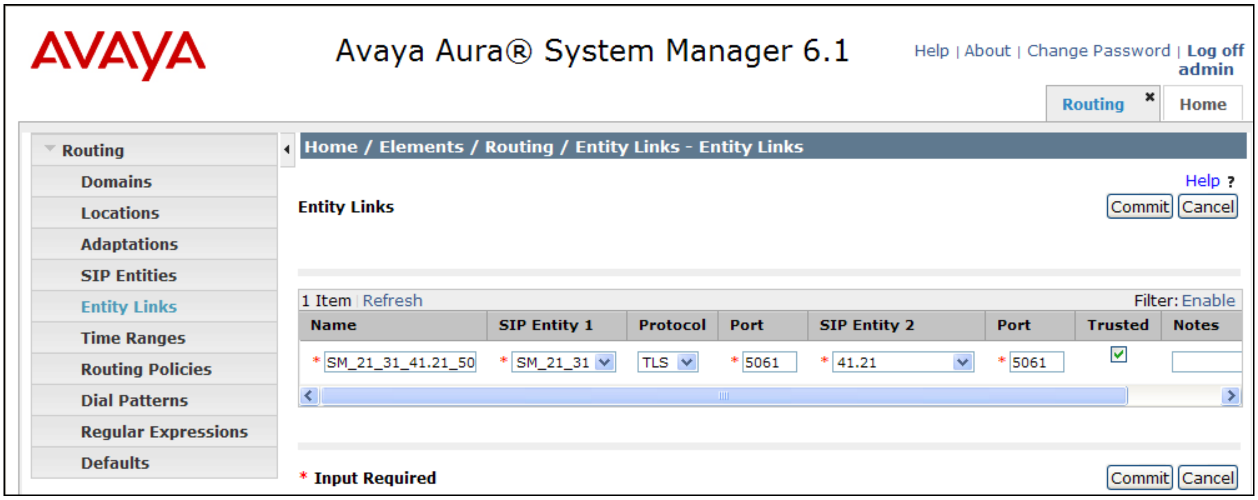
Entity Links define the connections between the SIP Entities and Session Manager. In the compliance test, the following entity links are defined from Session Manager.

- Session Manager ⇔ Communication Manager (Avaya S8300D Server). This entity link was created prior to the compliance test.
- Session Manager ⇔ Communication Manager (Avaya S8720 Servers). This entity link was created prior to the compliance test.

Navigate to **Routing → Entity Links**, and click on the **New** button (not shown) to create a new entity link. Provide the following information:

- Enter a descriptive name in the **Name** field.
- In the **SIP Entity 1** drop down menu, select the Session Manager SIP Entity shown in **Section 6.3** (e.g. **SM_21_31**).
- In the **Protocol** drop down menu, select the protocol to be used.
- In the **Port** field, enter the port to be used (e.g. **5060** or **5061**).
    - TLS – 5061
    - UDP or TCP – 5060
- In the **SIP Entity 2** drop down menu, select one of the two entities in the bullet list above (which were shown in **Section 6.3**).
- In the **Port** field, enter the port to be used (e.g. **5060** or **5061**).
- Enter a description in the **Notes** field if desired.
- Accept the other default values.

Click on the **Commit** button to save each Entity Link definition. The following screen shows an Entity Links page (between Session Manager and Avaya S8300D Server) used during the compliance test.



Repeat the steps to define Entity Link using a different protocol.

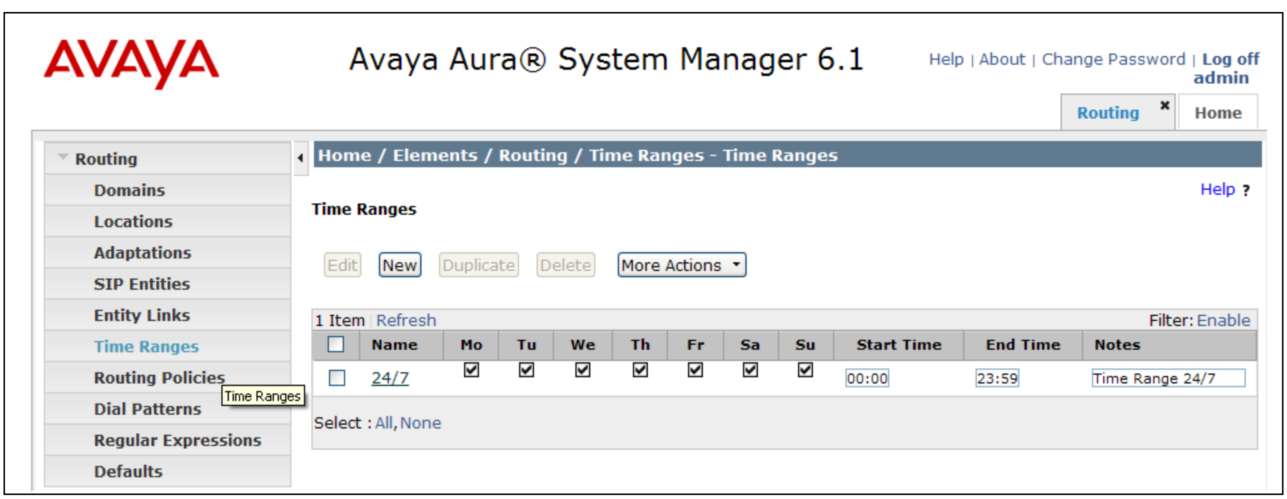
6.5. Time Ranges

The Time Ranges form allows admission control criteria to be specified for Routing Policies (**Section 6.6**). In the reference configuration, no restrictions were used.

To add a Time Range, navigate to **Routing → Time Ranges**, and click on the **New** button (not shown). Provide the following information:

- Enter a descriptive Time Range name in the **Name** field (e.g. **24/7**).
- Check each day of the week.
- In the **Start Time** field, enter **00:00**.
- In the **End Time** field, enter **23:59**.
- Enter a description in the **Notes** field if desired.

Click the **Commit** button. The following screen shows the Time Range page used during the compliance test.



6.6. Configure Routing Policy

Routing Policies associate destination SIP Entities (**Section 6.3**) with Time of Day admission control parameters (**Section 6.5**) and Dial Patterns (**Section 6.7**). In the reference configuration, Routing Policies are defined for:

- Calls to/from Communication Manager.

To add a Routing Policy, navigate to **Routing → Routing Policy**, and click on the **New** button (not shown) on the right. Provide the following information:

General section

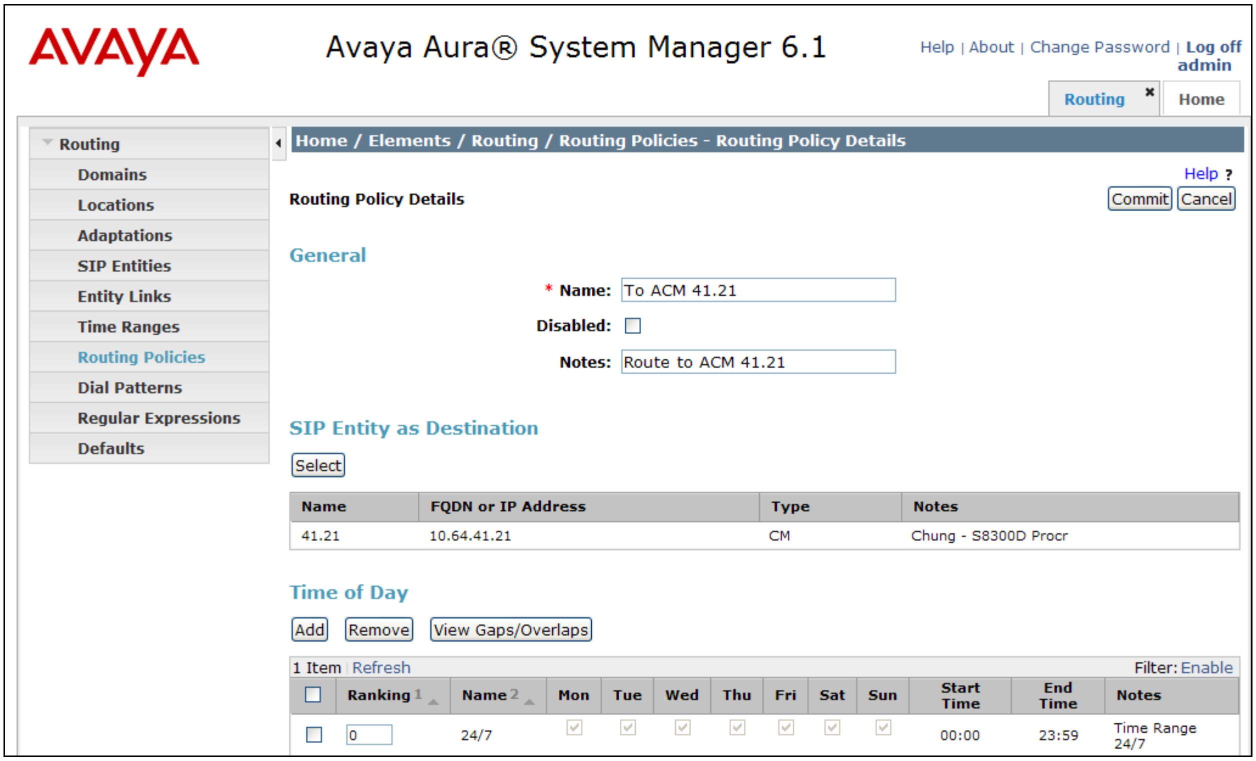
- Enter a descriptive name in the **Name** field.
- Enter a description in the **Notes** field if desired.

SIP Entity as Destination section

- Click the **Select** button.
- Select the SIP Entity that will be the destination for this call (not shown).
- Click the **Select** button and return to the Routing Policy Details form.

Time of Day section – Leave default values.

Click **Commit** to save Routing Policy definition. The following screen shows the Routing Policy used for the entity, **41.21**, during the compliance test.



6.7. Dial Patterns

Dial Patterns define digit strings to be matched for inbound and outbound calls. In addition, the domain in the request URI is also examined. In the compliance test, the following dial patterns are defined from Session Manager.

- 720xx – SIP and H323 endpoints in Avaya S8300D Server
- 2200x and 2800x – SIP and H323 endpoints in Avaya S8720 Servers

To add a Dial Pattern, select **Routing → Dial Patterns,** and click on the **New** button (not shown) on the right. During the compliance test, 5 digit dial plan was utilized. Provide the following information:

General section

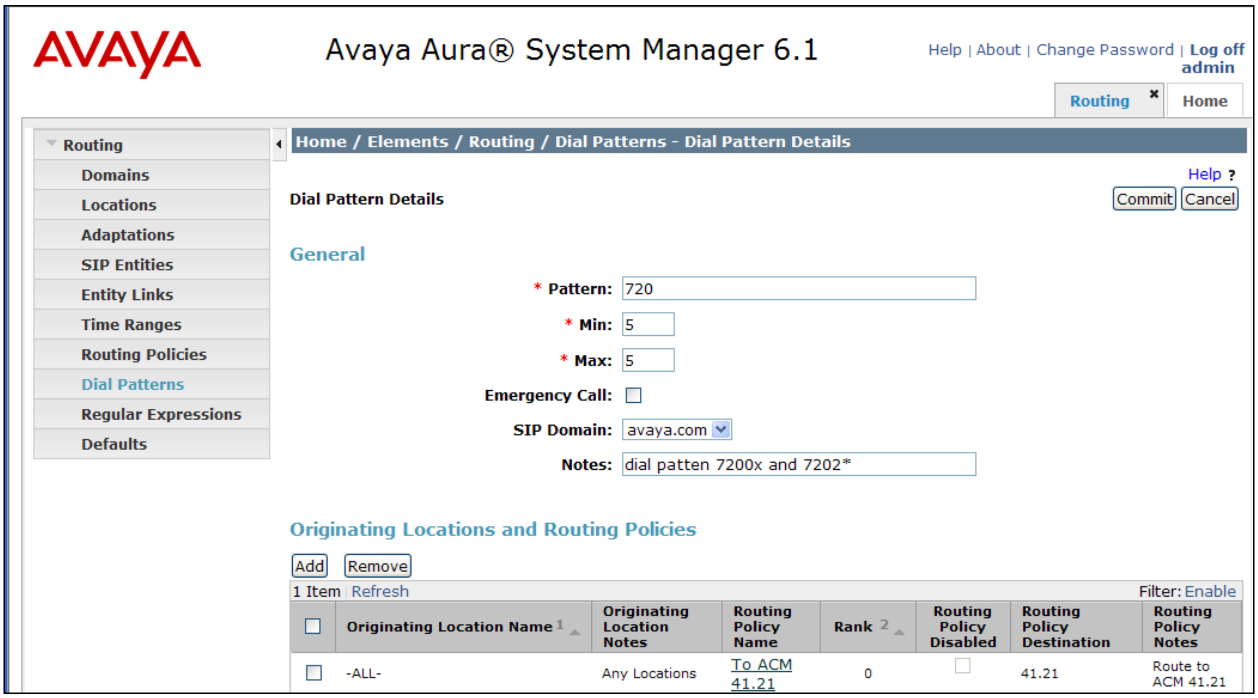
- Enter a unique pattern in the **Pattern** field (e.g. **720**).
- In the **Min** field enter the minimum number of digits (e.g. **5**).
- In the **Max** field enter the maximum number of digits (e.g. **5**).
- In the **SIP Domain** field drop down menu select the domain that will be contained in the Request URI *received* by Session Manager from Communication Manager.
- Enter a description in the **Notes** field if desired.

Originating Locations and Routing Policies section

- Click on the **Add** button and a window will open (not shown).

- Click on the boxes for the appropriate Originating Locations, and Routing Policies (see **Section 6.6**) that pertain to this Dial Pattern.
  - Originating Location –Check the **Apply The Selected Routing Policies to All Originating Locations** box.
  - Routing Policies **To ACM 41.21**.
  - Click on the **Select** button and return to the Dial Pattern window.

Click the **Commit** button to save the new definition. The following screen shows the dial pattern used for the S8300D during the compliance test.



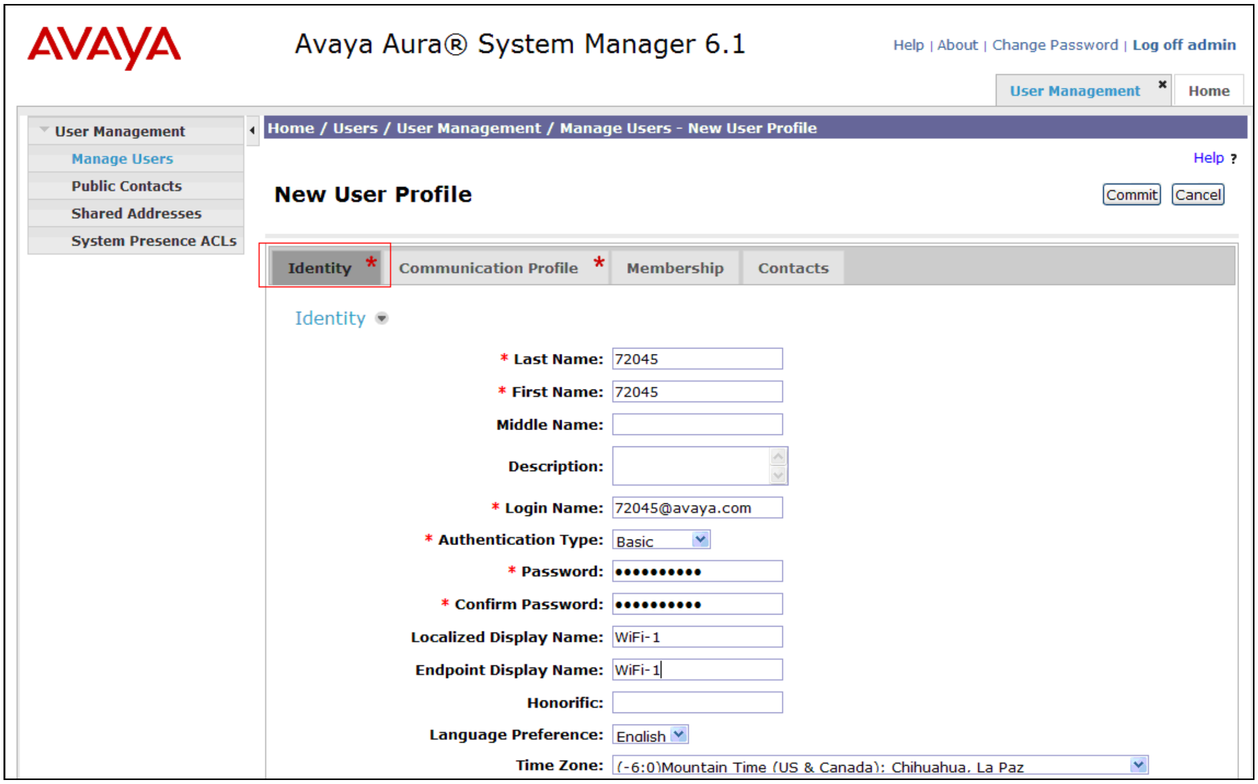
6.8. Configure SIP Users

During the compliance test, no special users were created for this solution. All users were created prior to the compliance test. However, the steps to configure a user are included. Add new SIP users for each Polycom WiFi station. .

To add new SIP users, Navigate to **Home → Users → User Management→ Manage Users**. Click **New (not shown)** and provide the following information:

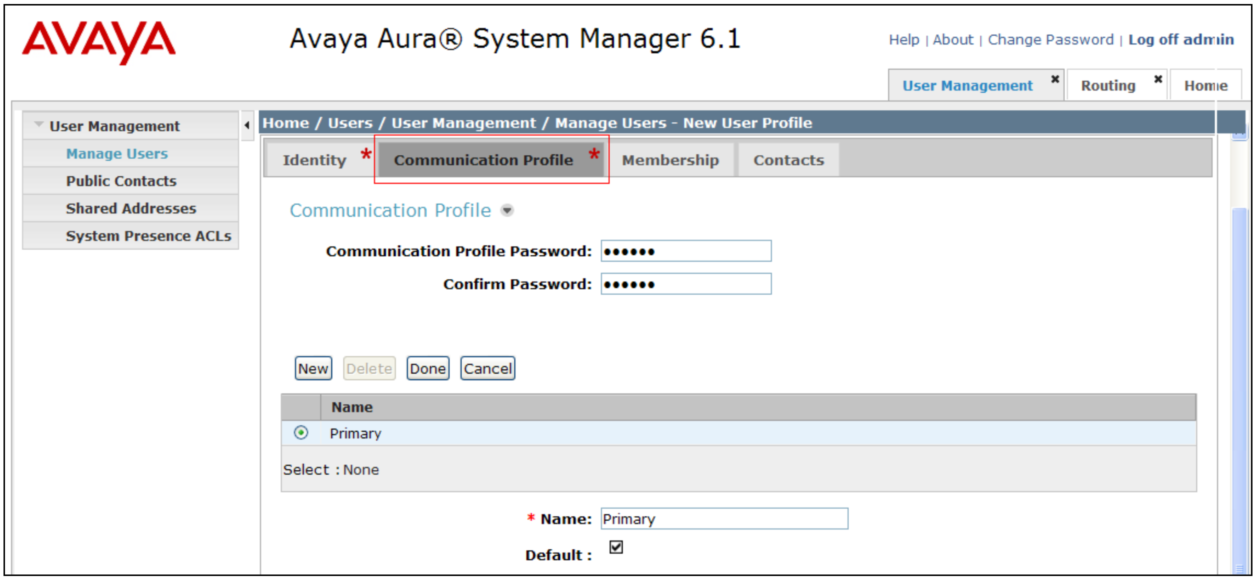
- Identity section
    - **Last Name –** Enter last name of user.
    - **First Name –** Enter first name of user.

    - **Login Name –** Enter extension number@sip domain name. The domain name is defined in **Section 5.3**.
    - **Authentication Type –** Verify **Basic** is selected.
    - **SMGR Login Password –** Enter password to be used to log into System Manager.
    - **Confirm Password –** Repeat value entered above.
    - Enter **Localized Display Name**
    - Enter **Endpoint Display Name**
    - Select **English** as **Language Preference**
    - Set the apropriate **Time Zone.**



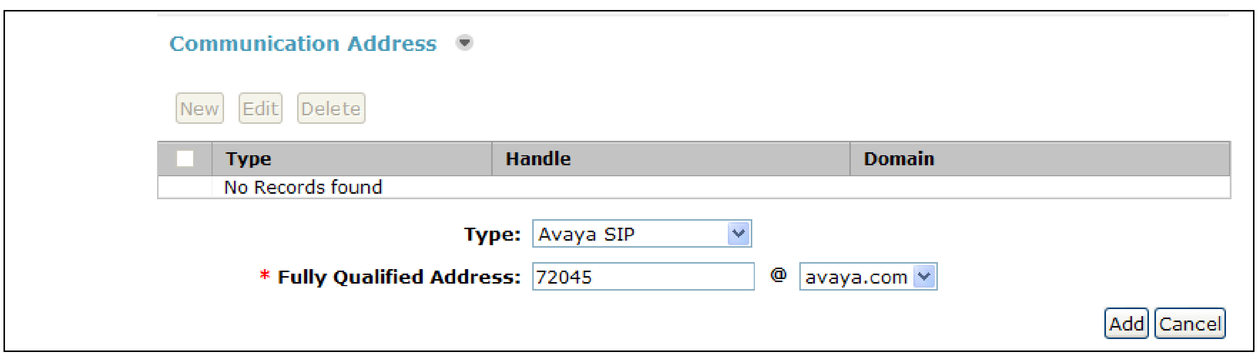
- <u>Communication Profile section</u>
  Provide the following information:
  - **Communication Profile Password** – Enter a numeric value used to logon to SIP telephone.
  - **Confirm Password** – Repeat numeric password

  Verify there is a default entry identified as the **Primary** profile for the new SIP user. If an entry does not exist, select **New** and enter values for the following required attributes:
  - **Name** – Enter **Primary**.
  - **Default** – Enter ☑

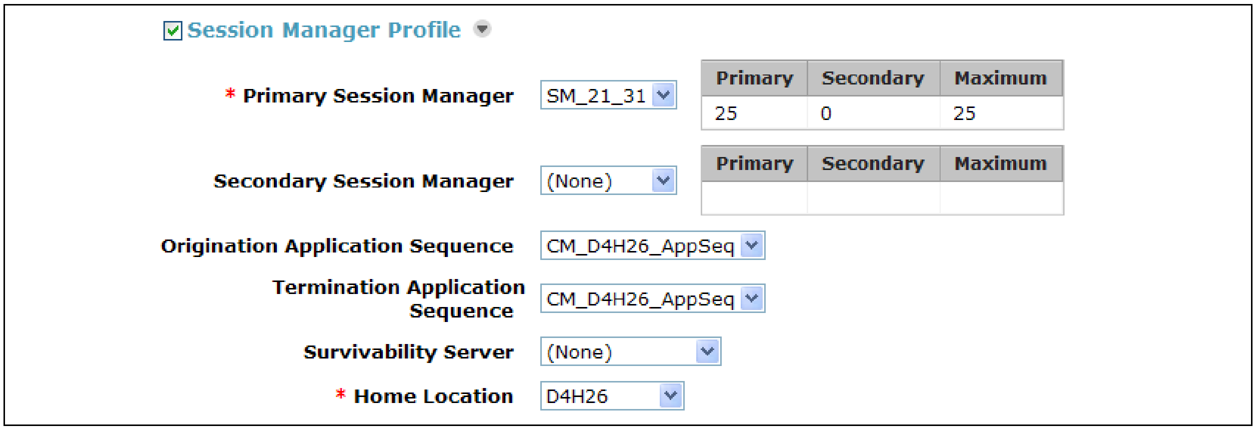


- <u>Communication Address sub-section</u>
  Select **New** to define a **Communication Address** for the new SIP user, and provide the following information.
  - **Type** – Select **Avaya SIP** using drop-down menu.
  - **Fully Qualified Address** – Enter same extension number and domain used for Login Name, created previously.

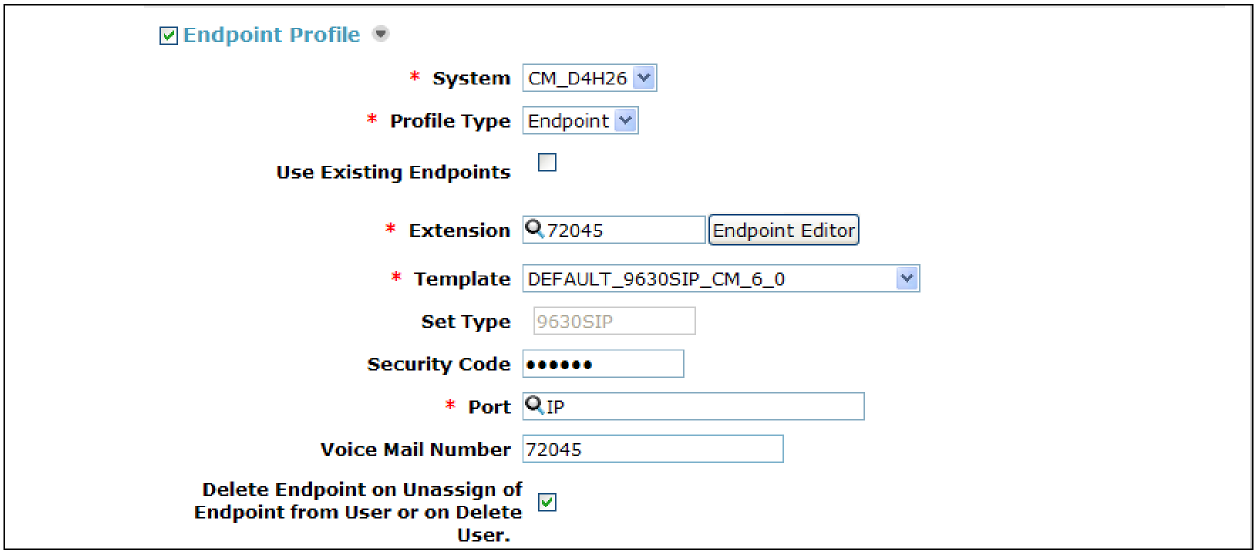
  Click the **Add** button to save the Communication Address for the new SIP user.



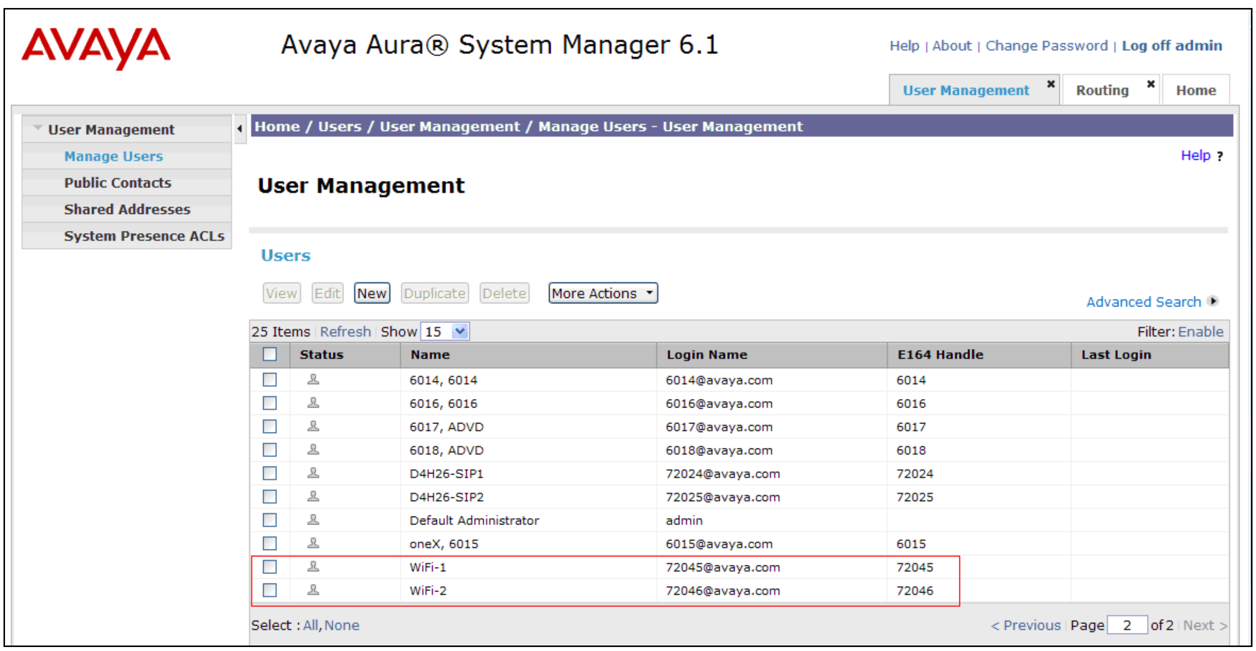
- <u>Session Manager Profile section</u>
  - **Primary Session Manager** – Select one of the Session Managers.
  - **Secondary Session Manager** – Select **(None)** from drop-down menu.
  - **Origination Application Sequence –** Select Application Sequence defined (not shown) for Communication Manager.
  - **Termination Application Sequence** – Select Application Sequence defined (not shown) for Communication Manager.
  - **Survivability Server** – Select **(None)** from drop-down menu.
  - **Home Location** – Select Location defined in **Section 6.2**.



- Endpoint Profile section
  - **System** – Select Managed Element defined in **System Manager** (not shown) for Communication Manager.
  - **Use Existing Endpoints -** Leave unchecked to automatically create a new endpoint on Communication Manager when the new user is created. Or else, check the box if endpoint is already defined in Communication Manager.
  - **Extension** - Enter same extension number used in this section.
  - **Template** – Select template for type of SIP phone. During the compliance test, DEFAULT_9630SIP_CM_6_0 was selected.
  - **Security Code** – Enter numeric value used to logon to SIP telephone. (**Note:** this field must match the value entered for the Shared Communication Profile Password field.)
  - **Port** – Select **IP** from drop down menu
  - **Voice Mail Number –** Enter **Pilot Number** for Avaya Modular Messaging if installed. Or else, leave field blank. This feature is not used during the compliance test.
  - **Delete Station on Unassign of Endpoint** – Check the box to automatically delete station when Endpoint Profile is un-assigned from user.



Click **Commit** to save definition of new user. The following screen shows the created users during the compliance test.



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 Session Manager:

- **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_72045.cfg. During the compliance test, sip_72045.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_72045.cfg" MISC_FILES="" LOG_FILE_DIRECTORY=""
OVERRIDES_DIRECTORY="" CONTACTS_DIRECTORY="" />
```

- **sip_72045.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 Session Manager.

```
<?xml version="1.0" encoding="utf-8"?>
<PHONE_CONFIG>
  <reg reg.1.address="72045@avaya.com" reg.1.displayName="72045" reg.1.label="72045"
reg.1.auth.userId="72045" reg.1.auth.password="123456"
reg.1.server.1.address="10.64.21.31" reg.1.server.1.port="5060" />
<msg.mwi msg.mwi.1.subscribe="72045@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 Session Manager server by following the **Session Manager →System Status → User Registrations** link on the System Manager 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 6.0.1) and Session Manager (Version 6.1). SpectraLink 8400 Series (UCS 4.0.0.10555) functioned properly for feature and serviceability. During compliance testing, SpectraLink 8400 Series successfully registered with Session Manager, 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*, June 2010, Release 6.0, Document Number 03-300509.

[2] *Administering Avaya® Session Manager,* November 2010, Release 6.1, Document Number 03-603324.

[3] *Administering Avaya® System Manager*, November 2010, Release 6.1.

The following document was provided by Polycom.

[4] *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.