Avaya Solution & Interoperability Test Lab

Application Notes for Polycom SoundPoint® IP 550 SIP Phone with Avaya Aura™ Session Manager 6.0 and Avaya Aura™ Communication Manager 6.0 - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate a Polycom SoundPoint® IP 550 SIP Phone with a SIP infrastructure consisting of Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager configured as an Evolution Server. During compliance testing, the SoundPoint® IP 550 SIP Phone successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps required to integrate a Polycom SoundPoint® IP 550 SIP Phone with a SIP infrastructure consisting of Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager configured as an Evolution Server. During compliance testing, the SoundPoint® IP 550 SIP Phone successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

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

1.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the SoundPoint IP 550 SIP Phone with Session Manager.
- Calls between SoundPoint IP 550 SIP phones and Avaya SIP, H.323, and digital stations.
- G.711, G.729A, and G.722 codec support.
- Proper recognition of DTMF tones by navigating voicemail menus.
- Proper operation of voicemail with Message Waiting Indication (MWI).
- Basic telephony features including Hold, Transfer, and Conference.
- Extended telephony features using Communication Manager Feature Name Extensions (FNEs) such as Call Forwarding, Call Pickup, and Send All Calls.
- Proper system recovery after a SoundPoint IP 550 SIP phone restart and loss of IP connectivity.
- PC/laptop connectivity to Ethernet jack on phone.

1.2. Support

For technical support on the SoundPoint IP 550 SIP Phone contact Polycom Support through their website at http://www.polycom.com/support/.

In addition, additional support information may be obtained through the knowledge base available at
http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html.

2. Reference Configuration

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

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

In addition, two Polycom SoundPoint IP 550 SIP Phones registered with Session Manager and were configured as Off-PBX Stations (OPS) on the Communication Manager.

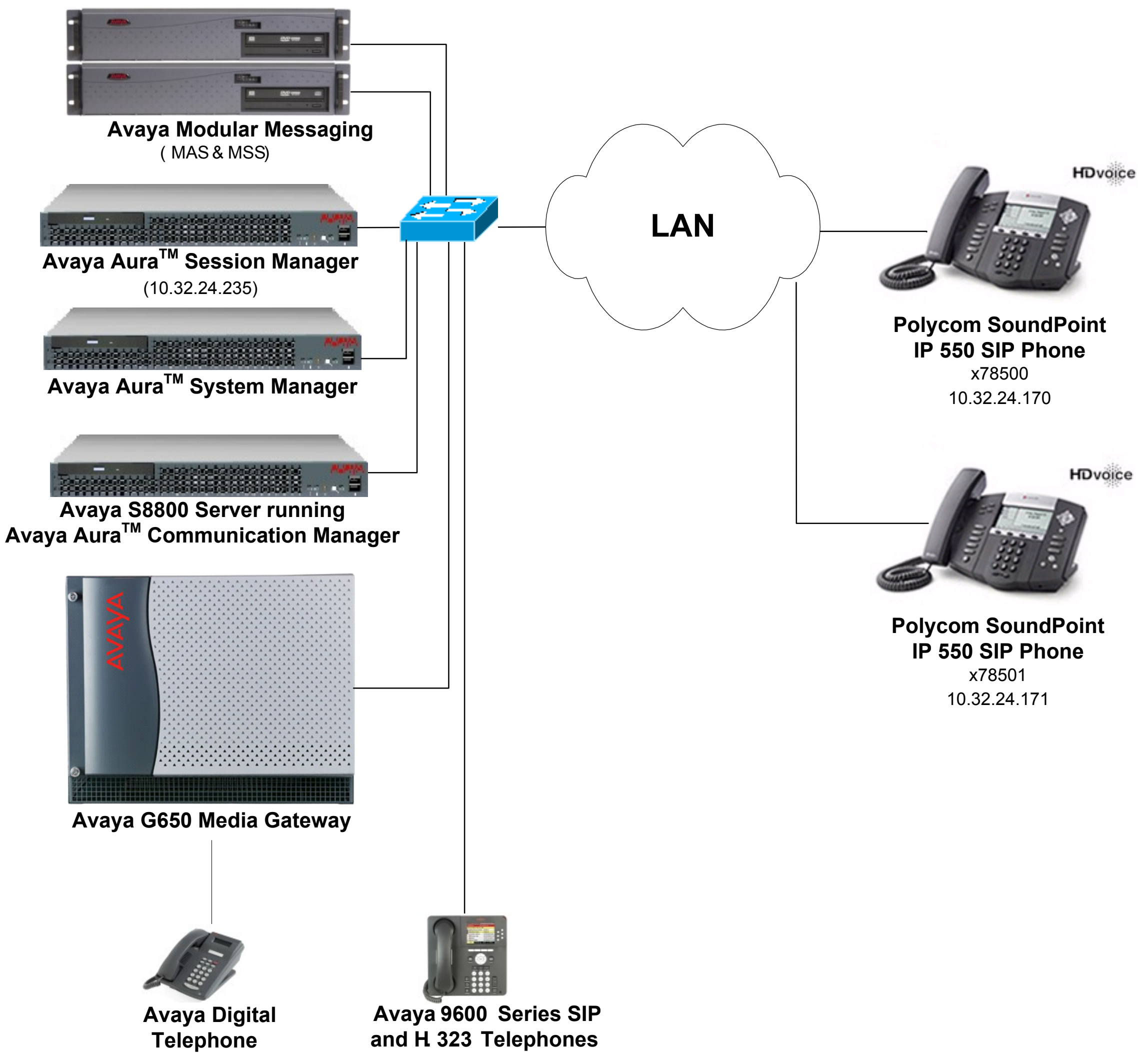


Figure 1: Avaya SIP Network with Polycom SoundPoint IP 550 SIP Phones

2.1. SIP Call Flows

The Polycom SoundPoint IP 550 SIP Phone originates a call by sending a call request (SIP INVITE message) to Session Manager, which then routes the call over a SIP trunk to the Communication Manager for origination services. If the call is destined for another local SIP phone, Communication Manager routes the call back over the SIP trunk to Session Manager for delivery to the destination SIP phone. If the call is destined for an H.323 or digital telephone, Communication Manager routes the call to the H.323 or digital endpoint.

For a call arriving at Communication Manager that is destined for one of the SoundPoint IP 550 SIP Phones, Communication Manager routes the call over the SIP trunk to Session Manager for delivery to the SoundPoint IP 550 SIP Phones.

3. Equipment and Software Validated

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

Hardware/Software Component	Version
Avaya Aura™ Communication Manager running on Avaya S8800 Servers and G650 Media Gateway	6.0 Service Pack 1
Avaya Aura™ Session Manager	6.0 (6.0.0.0.600020)
Avaya Aura™ System Manager	6.0 (6.0.0.0.556-3.0.6.1)
Avaya Modular Messaging	5.2
Avaya 9600 Series IP Telephones	3.110b (H.323) 2.6 (SIP)
Avaya Digital Telephones	--
Polycom SoundPoint IP 550 SIP Phone	3.2.3.1734

4. Configure Avaya Aura^{TM} Communication Manager

This section describes the steps for configuring the SoundPoint IP 550 SIP Phone as an Off-PBX Station (OPS) and configuring a SIP trunk between the Communication Manager and Session Manager. **Section 4.3** covers the station configuration for the SoundPoint IP 550 SIP Phones. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

4.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative. On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

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

     G3 Version: V16                            Software Package: Enterprise
       Location: 2                               System ID (SID): 1
       Platform: 28                              Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 65000 350
                                      Maximum Stations: 41000 197
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 36000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 36
                    Maximum Off-PBX Telephones - PBFMC: 36000 0
                    Maximum Off-PBX Telephones - PVFMC: 36000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2** of the **system-parameters customer-options** form, verify that the number of SIP trunks supported by the system is sufficient.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 60
           Maximum Concurrently Registered IP Stations: 18000 13
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 70
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

4.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the S8800 Server processor, the C-LAN board in the G650 Media Gateway, and signaling interface for Session Manager. The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
Gateway001          10.32.24.1
ModMsg              192.50.10.45
clancrm             10.32.24.20
default             0.0.0.0
devcon-asm          10.32.24.235
medprocrm           10.32.24.21
procr               10.32.24.10
procr6              ::
 ( 8 of 8   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 34
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the SoundPoint IP 550 SIP Phones. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. However, the **IP Codec Set** form may specify multiple codecs, including G.711, G.729A, and G.722, which are supported by the SoundPoint IP 550 SIP Phones.

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
 3:
 4:
 5:
 6:
 7:
```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as follows:

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

```
add signaling-group 50                                       Page   1 of   1
                                SIGNALING GROUP

 Group Number: 50             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        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: clancrm                Far-end Node Name: devcon-asm
 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? n                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the SIP Phones. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 50                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 50                   Group Type: sip          CDR Reports: y
  Group Name: To devcon-asm              COR: 1        TN: 1        TAC: 1050
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 50
                                                    Number of Members: 10
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 50                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y
```

Configure the **Private Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' and whose calls are routed over any trunk group, including SIP trunk group "50", have the number sent to the far-end for display purposes.

```
change private-numbering 0                                     Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  7                                            5     Total Administered: 1
                                                          Maximum Entries: 540
```

4.3. Configure Stations

Use the **add station** command to add a station for each SoundPoint IP 550 SIP Phone to be supported. Use *9630SIP* for the **Station Type** and include the **Coverage Path** for voice mail, if applicable. The **Name** field is optional. Use the default values for the other fields on **Page 1**. The SIP station can also be configured automatically by System Manager as described in **Section 5.7**.

```
add station 78500                                           Page   1 of   6
                                     STATION

Extension: 78500                        Lock Messages? n              BCC: 0
     Type: 9630SIP                      Security Code:                 TN: 1
     Port: IP                         Coverage Path 1: 20             COR: 1
     Name: Polycom 78500              Coverage Path 2:                COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 78500

        Display Language: english                 Button Modules: 0

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

                                                        IP Video? n
```

On **Page 2**, set the **MWI Served User Type** field to the appropriate value to allow MWI notifications to be sent to the SoundPoint IP 550 SIP Phone.

```
add station 78500                                           Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                     Coverage Msg Retrieval? y
                                                           Auto Answer: none
             CDR Privacy? n                           Data Restriction? n
                                              Idle Appearance Preference? n
 Per Button Ring Control? n                 Bridged Idle Line Preference? n
   Bridged Call Alerting? n
  Active Station Ringing: single

        H.320 Conversion? n         Per Station CPN - Send Calling Number?
                                                     EC500 State: enabled

    MWI Served User Type: qsig-mwi

                                                Coverage After Forwarding? s

                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 78500        Always Use? n IP Audio Hairpinning? n
```

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

```
change off-pbx-telephone station-mapping 78500                 Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 78500            OPS           -     78500            aar        1
```

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

```
change off-pbx-telephone station-mapping 78500                 Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 78500         OPS      3        both         all         none
```

5. Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

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

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

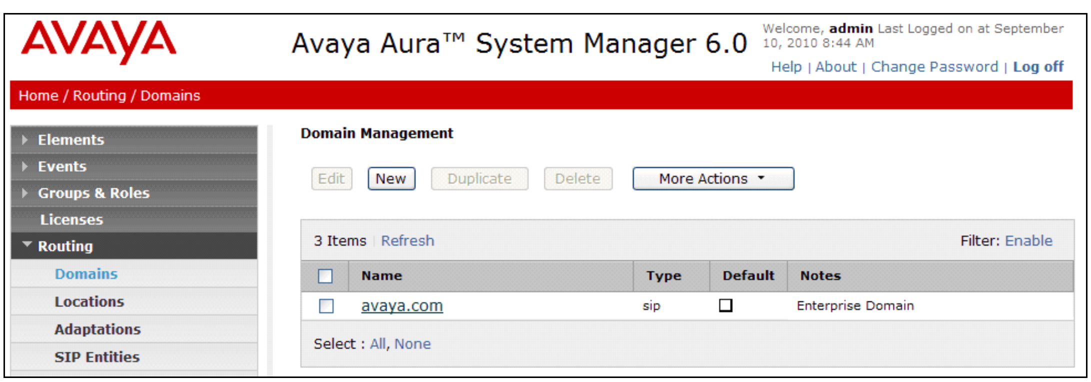
5.1. Specify SIP Domain

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

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

Click **Commit**.

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



5.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

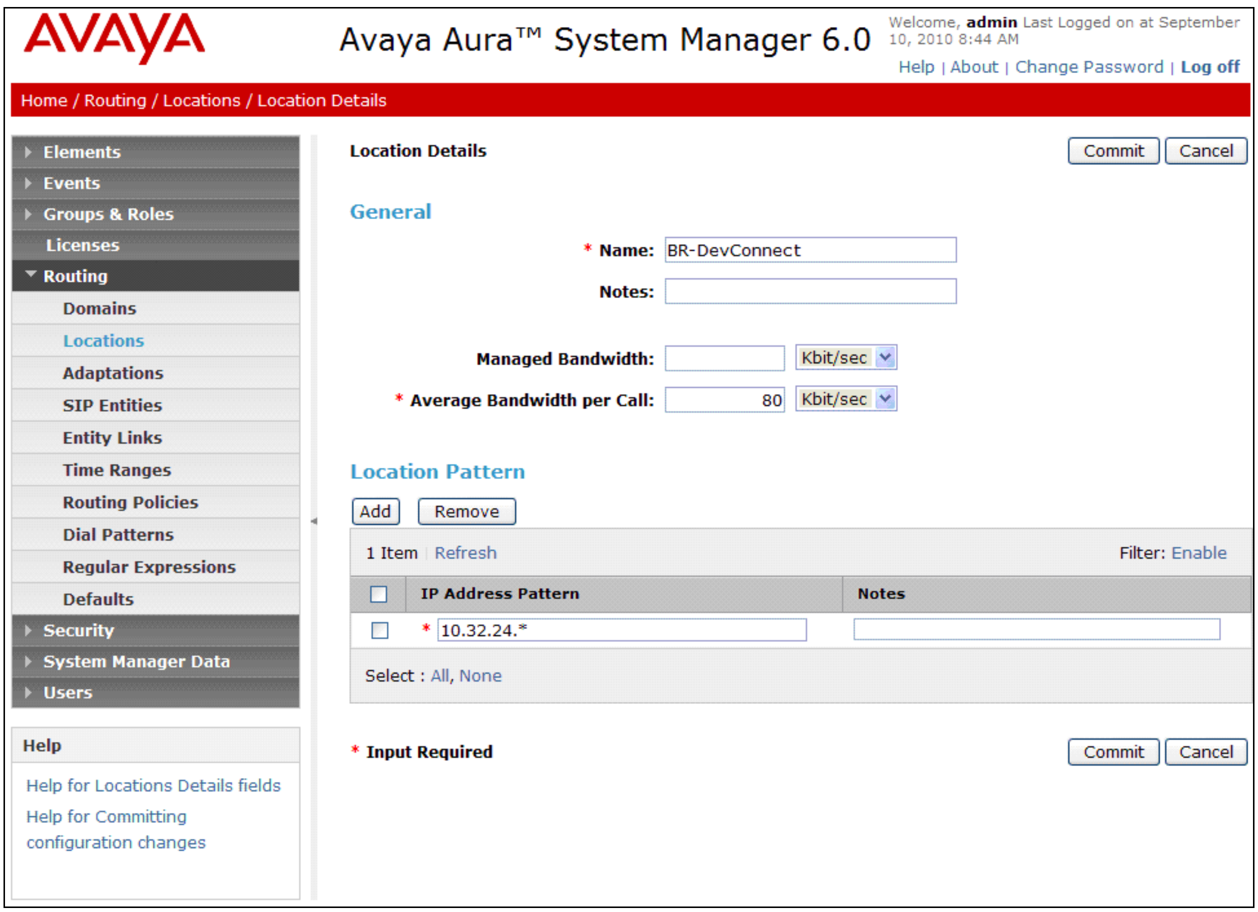
Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the *BR-DevConnect* location, which includes the Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager. Click **Commit** to save the Location definition.



5.3. Add SIP Entities

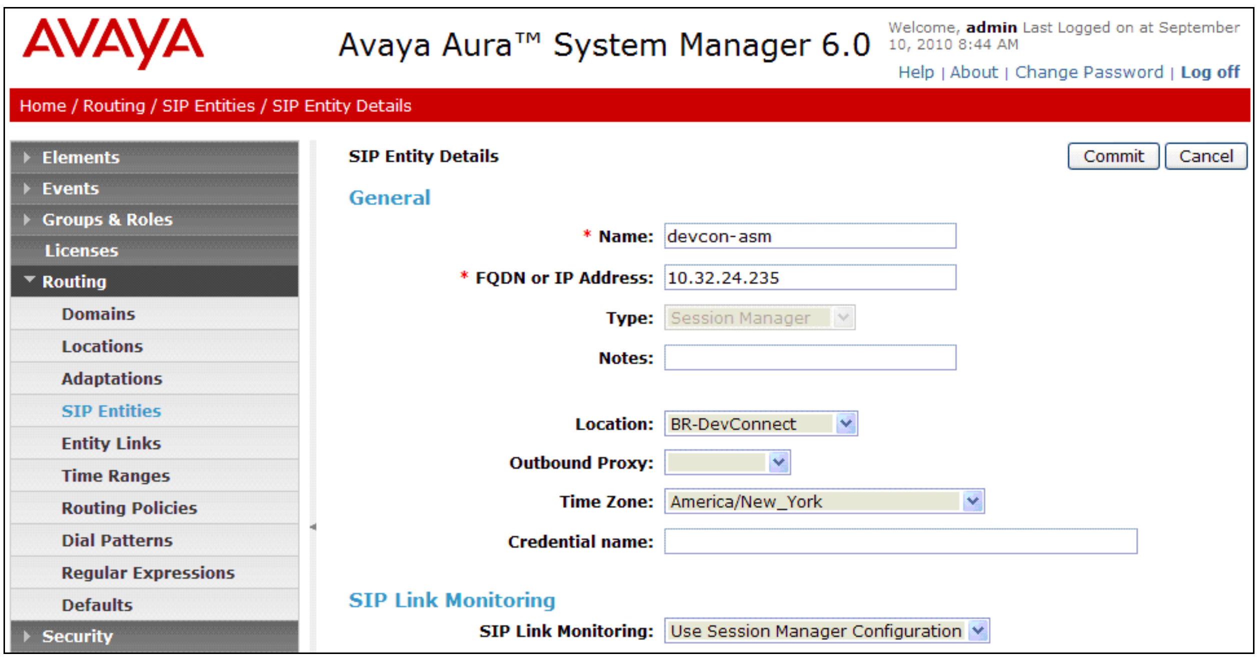
In the sample configuration, a SIP Entity is added for Session Manager and the C-LAN in the G650 Media Gateway.

5.3.1. Avaya Aura^{TM} Session Manager

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

Under *General*:

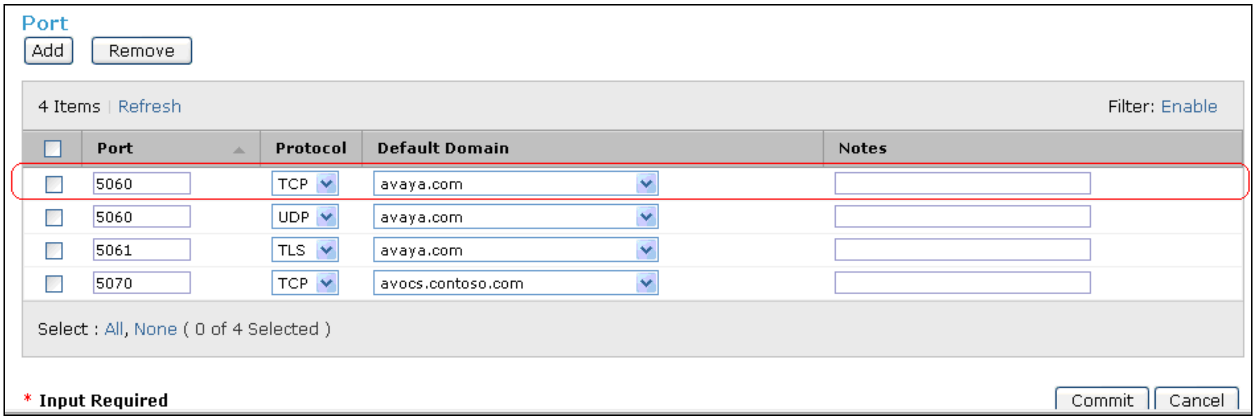
- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.



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

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *avaya.com*).

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



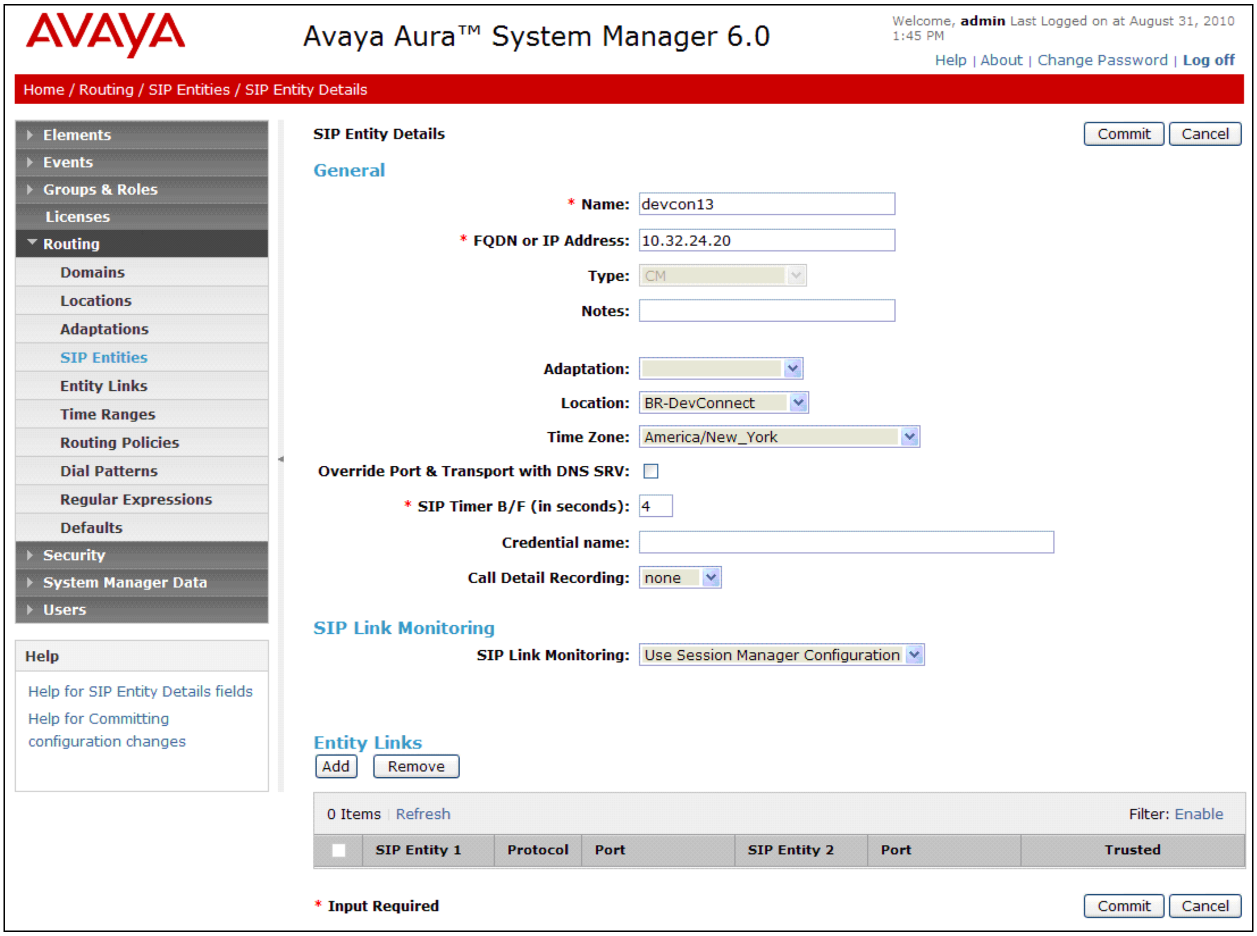
5.3.2. Avaya Aura^{TM} Communication Manager

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

Under *General*:

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

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

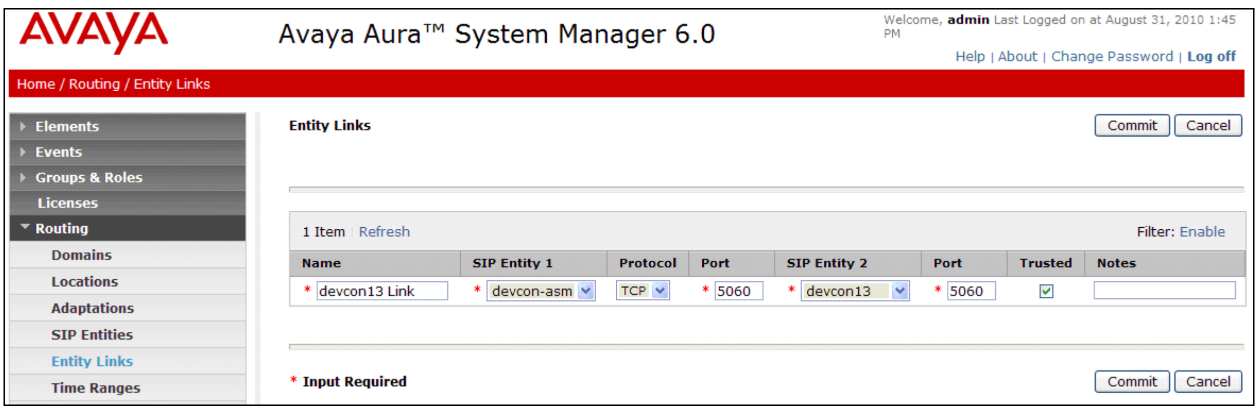

AVAYA
Avaya Aura™ System Manager 6.0
Welcome, admin Last Logged on at August 31, 2010 1:45 PM
Help | About | Change Password | Log off
Home / Routing / SIP Entities / SIP Entity Details
Elements
Events
Groups & Roles
Licenses
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Security
System Manager Data
Users
Help
Help for SIP Entity Details fields
Help for Committing configuration changes
SIP Entity Details
Commit
Cancel
General
* Name: devcon13
* FQDN or IP Address: 10.32.24.20
Type: CM
Notes:
Adaptation:
Location: BR-DevConnect
Time Zone: America/New_York
Override Port & Transport with DNS SRV:
* SIP Timer B/F (in seconds): 4
Credential name:
Call Detail Recording: none
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Entity Links
Add
Remove
0 Items Refresh
Filter: Enable
SIP Entity 1
Protocol
Port
SIP Entity 2
Port
Trusted
* Input Required
Commit
Cancel

5.4. Add Entity Link

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

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

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



5.5. Define Communication Manager as Managed Element

Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP users are added.

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

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

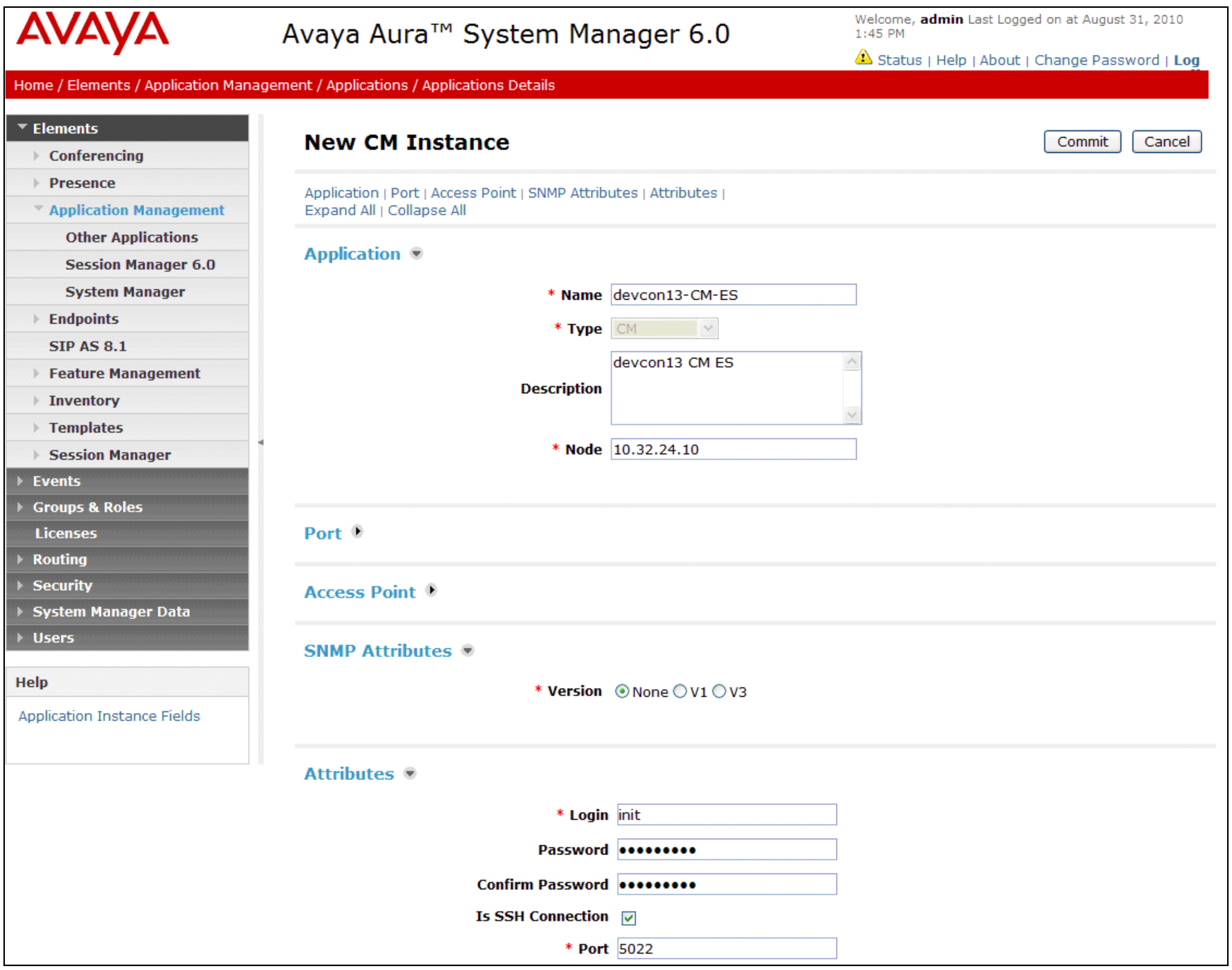
Under *Application*:

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

Under *Attributes*:

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

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

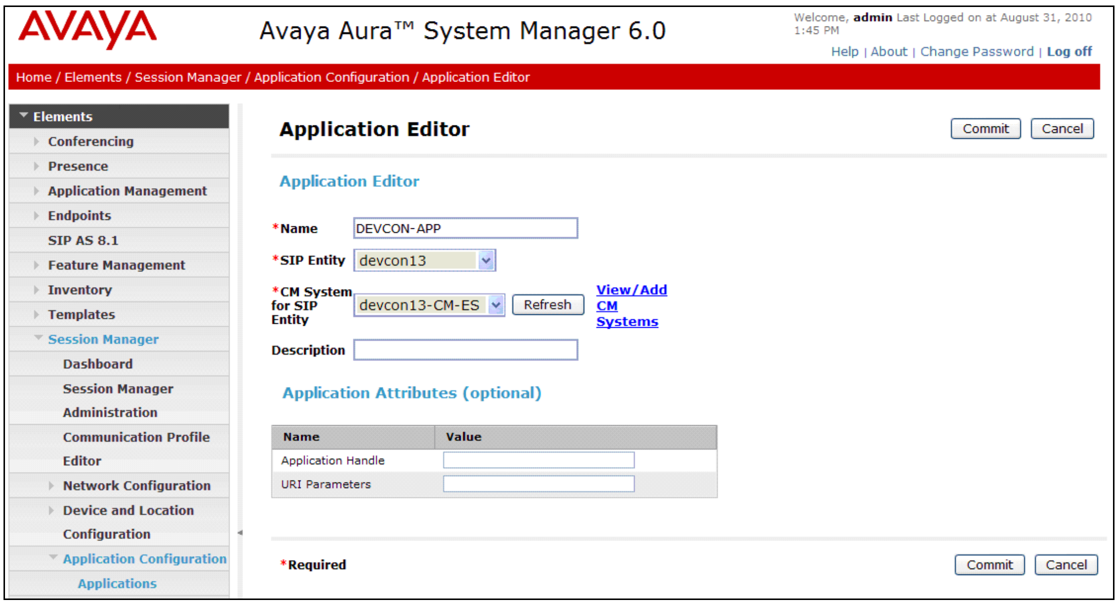


5.6. Add Application Sequence

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

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

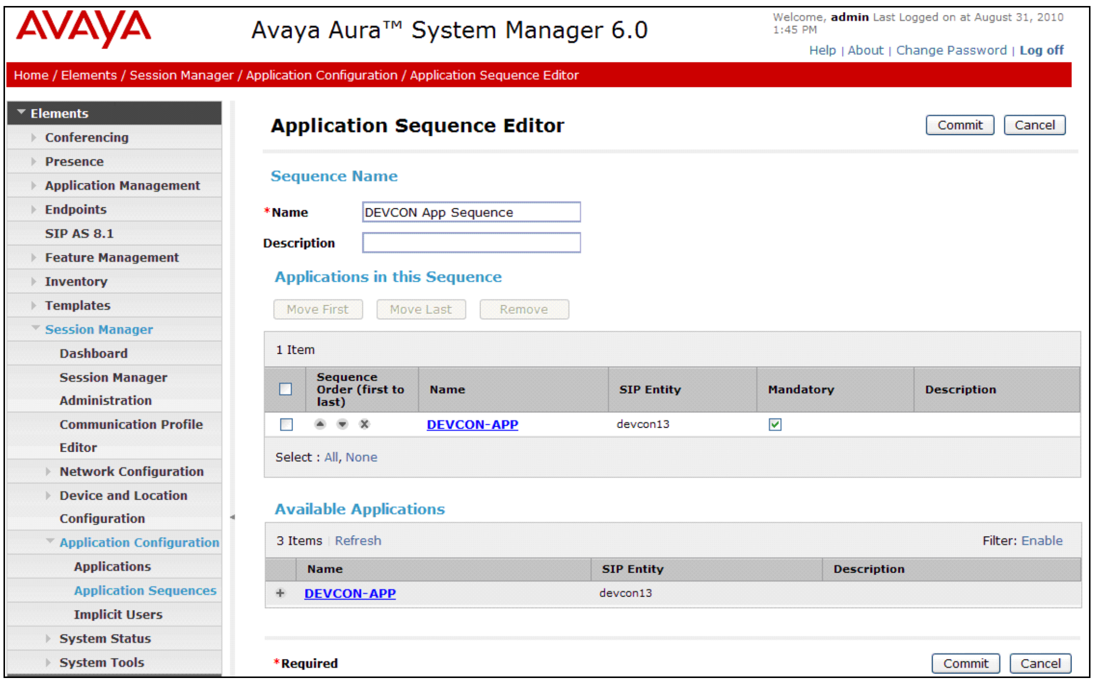
Click **Commit** to save the Application definition.



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

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

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



5.7. Add SIP Users

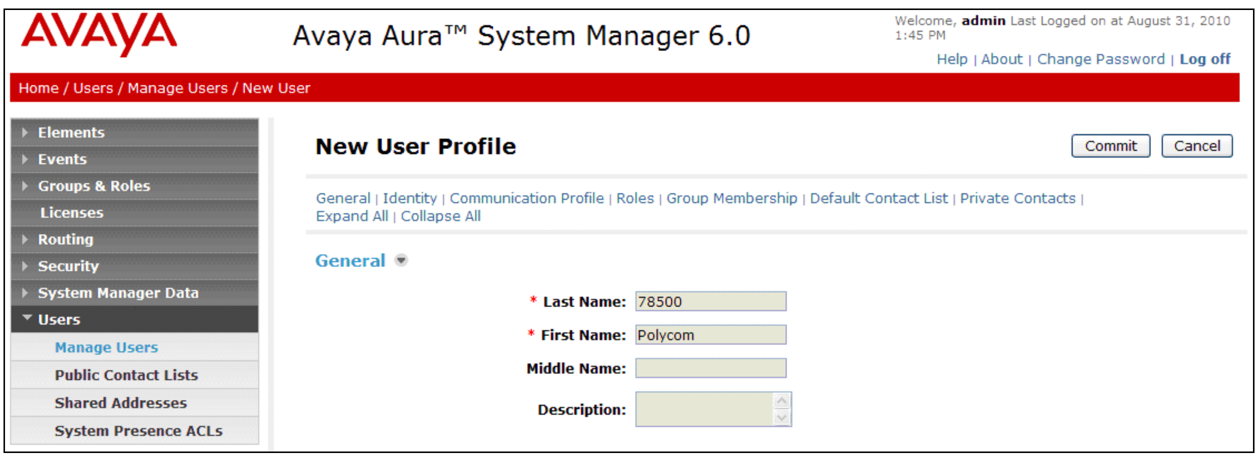
Add SIP users corresponding to the SoundPoint IP 550 SIP Phone defined in **Section 4.3.** Alternatively, use the option to automatically generate the SIP stations on Communication Manager Evolution Server when adding a new SIP user.

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

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

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

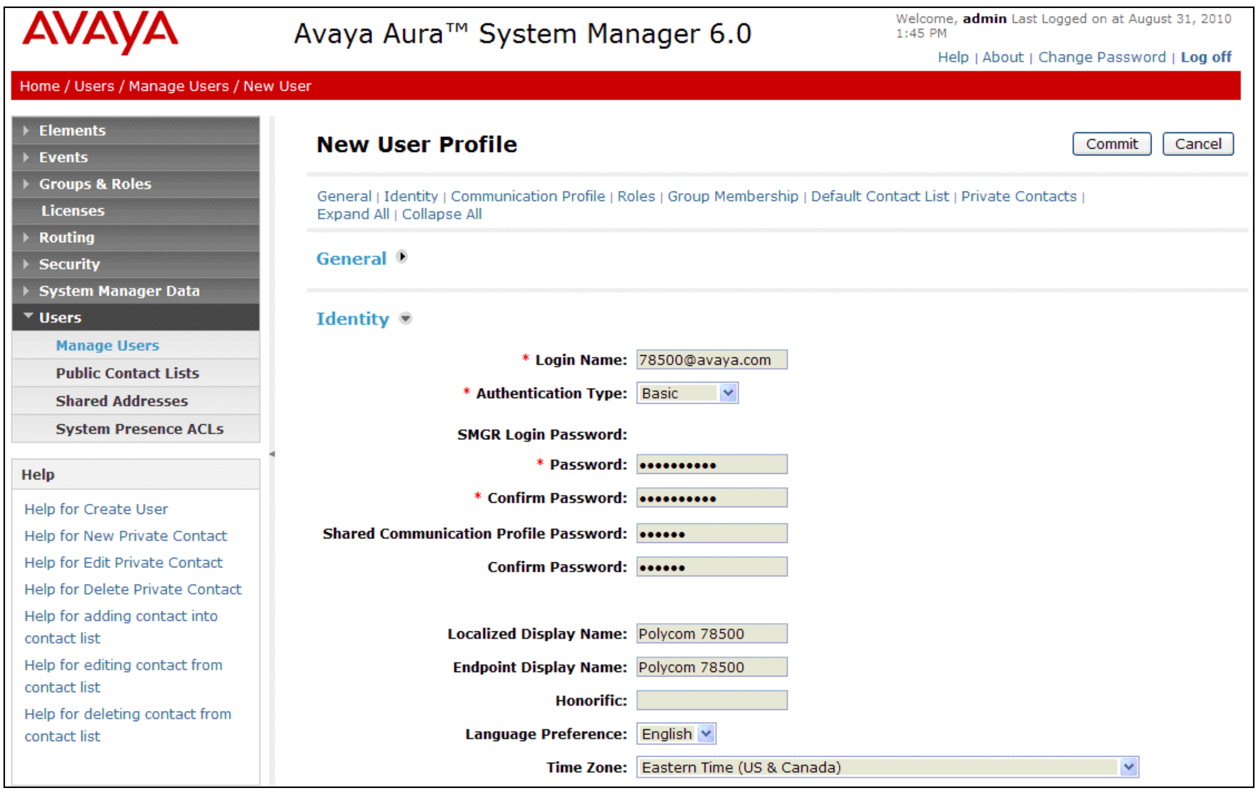
The screen below shows the information when adding a new SIP user to the sample configuration.



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

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

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



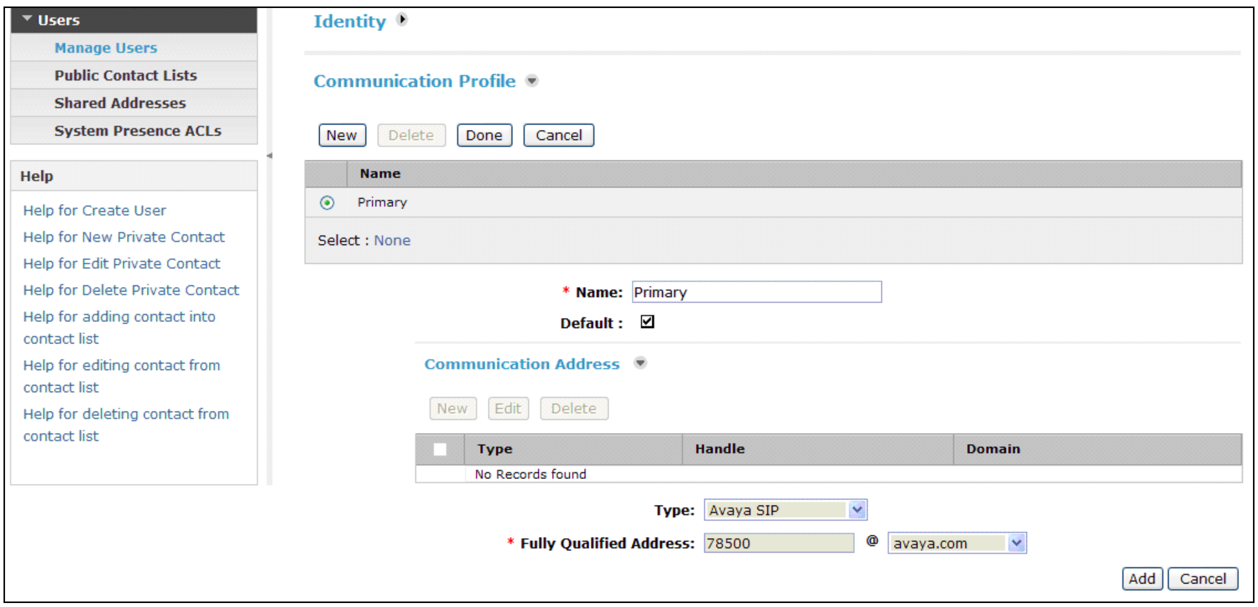
Scroll down to the **Communication Profile** section and select **New** to define a **Communication Profile** for the new SIP user. Enter values for the following required fields:

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

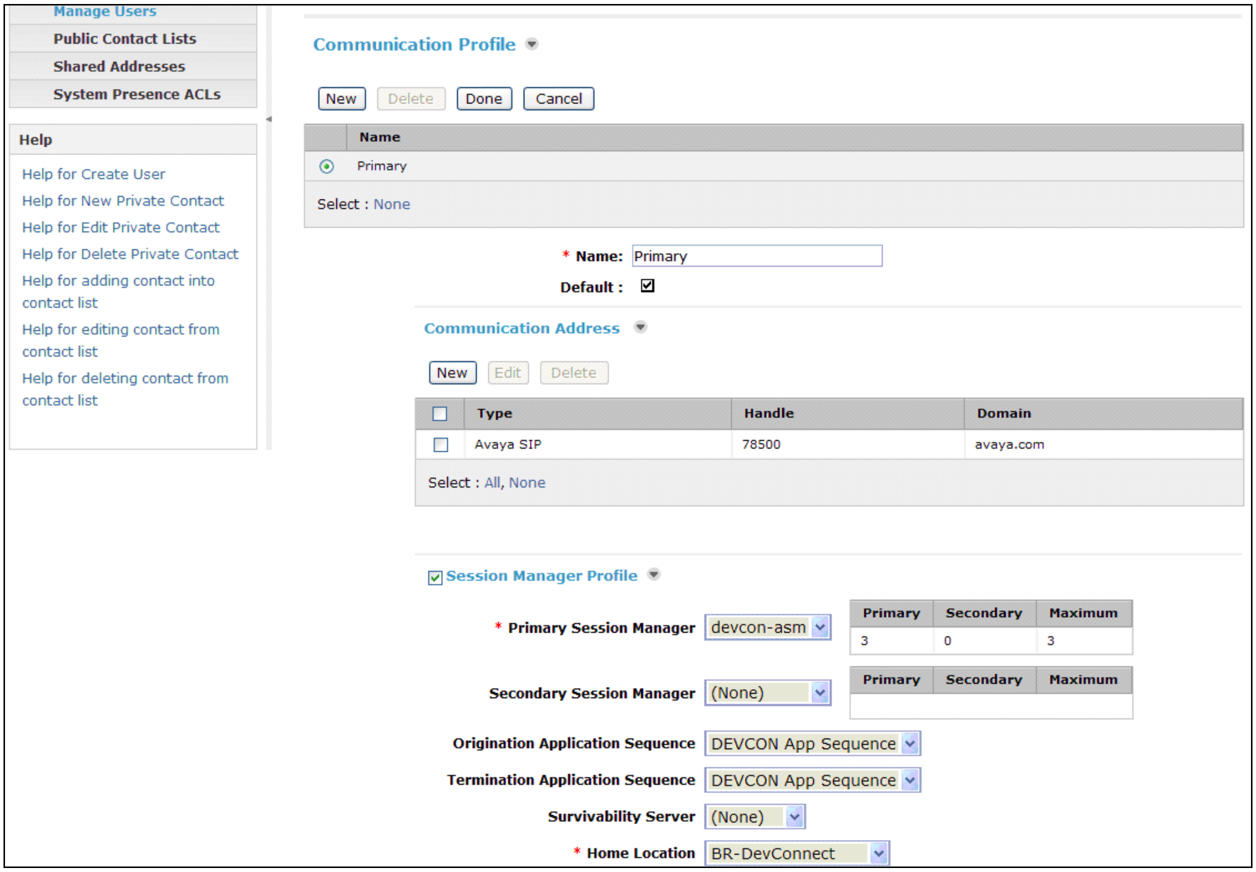
Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

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

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



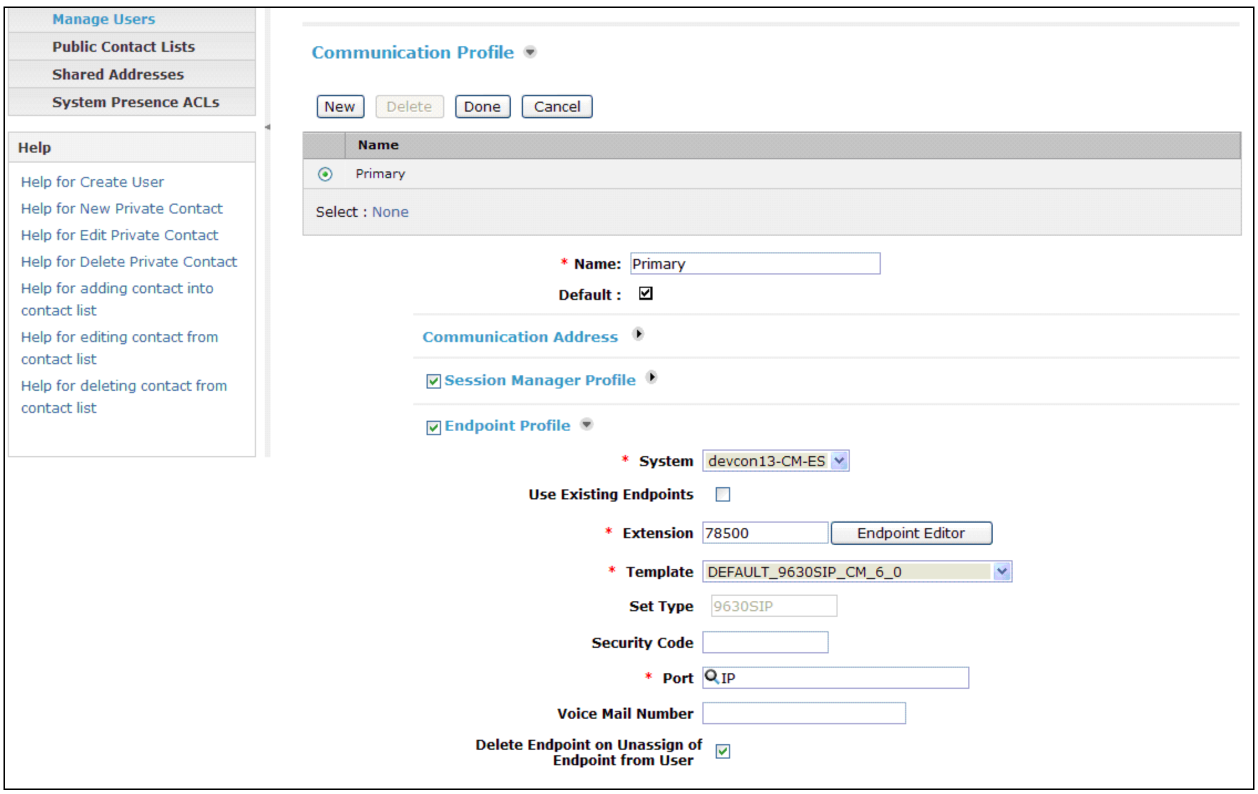
In the *Session Manager* section, specify the Session Manager entity from **Section 5.3.1** for **Primary Session Manager** and assign the **Application Sequence** defined in **Section 5.6** to the new SIP user as part of defining the **SIP Communication Profile**. The **Application Sequence** can be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



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

- **System:** Select the managed element corresponding to Communication Manager.
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone.
- **Port:** Enter *IP*.
- **Delete Station on Unassign of Station:** Enable field to automatically delete station when **Station Profile** is un-assigned from user.

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



5.8. Add Session Manager

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

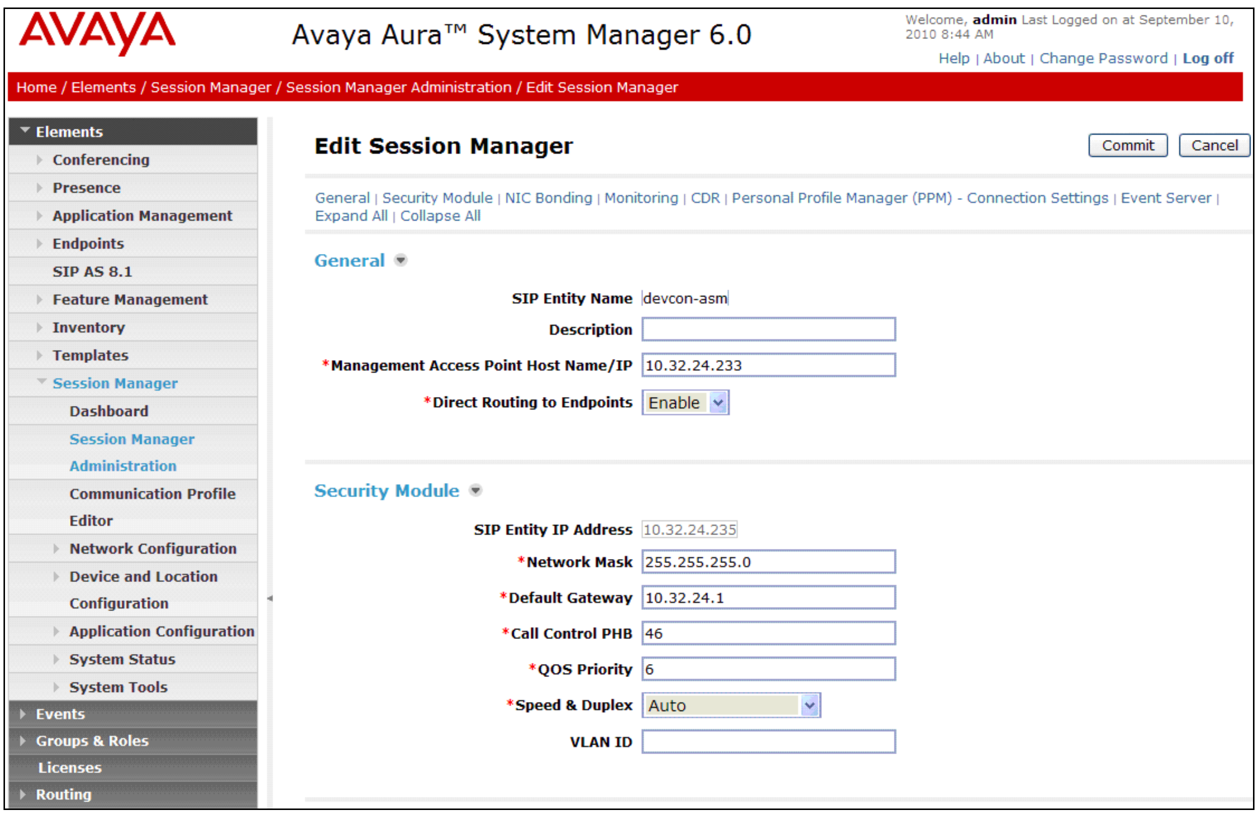
Under *Identity*:

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

Under *Security Module*:

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

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



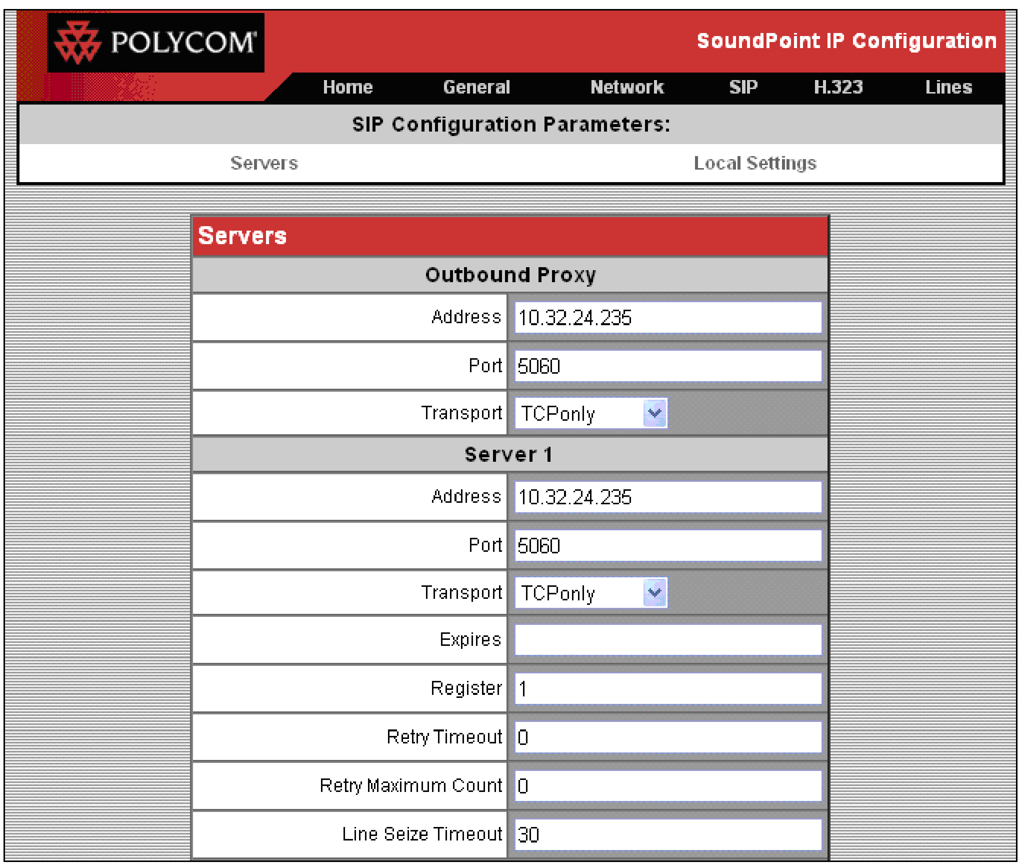
6. Configure Polycom SoundPoint® IP 550 SIP Phone

The configuration of the SoundPoint® IP 550 SIP Phone was performed via the phone's menu-driven LCD user interface and its embedded Web interface. The phone's LAN connection interface was initially configured via the phone's LCD screen. To configure the IP parameters for the phone, click the MENU key on the phone and navigate to **Settings→Advanced→Admin Settings→Network Configuration**. A valid password will be required. The rest of the configuration was performed through the phone's embedded Web interface. Refer to [3] for additional information on configuring the SoundPoint® IP 550 SIP Phone.

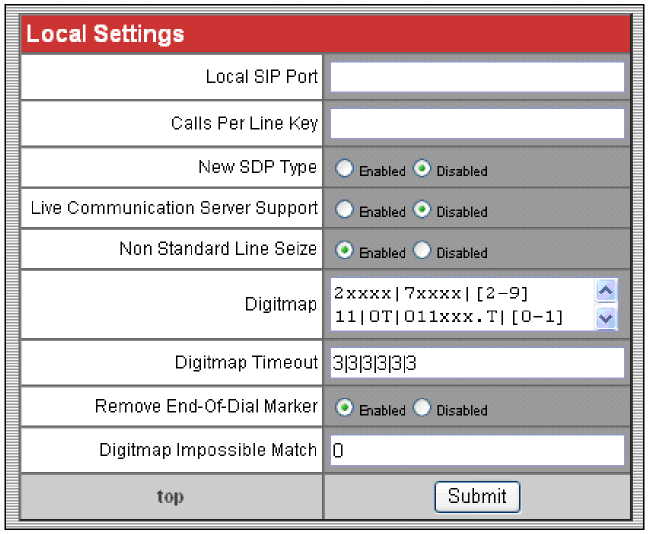
Note: To verify that the phone is running the compliance-tested SIP application version, press the **Menu** key on the phone, and then select **Status→Platform→Application**. Refer to [3] for upgrade instructions, if required.

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

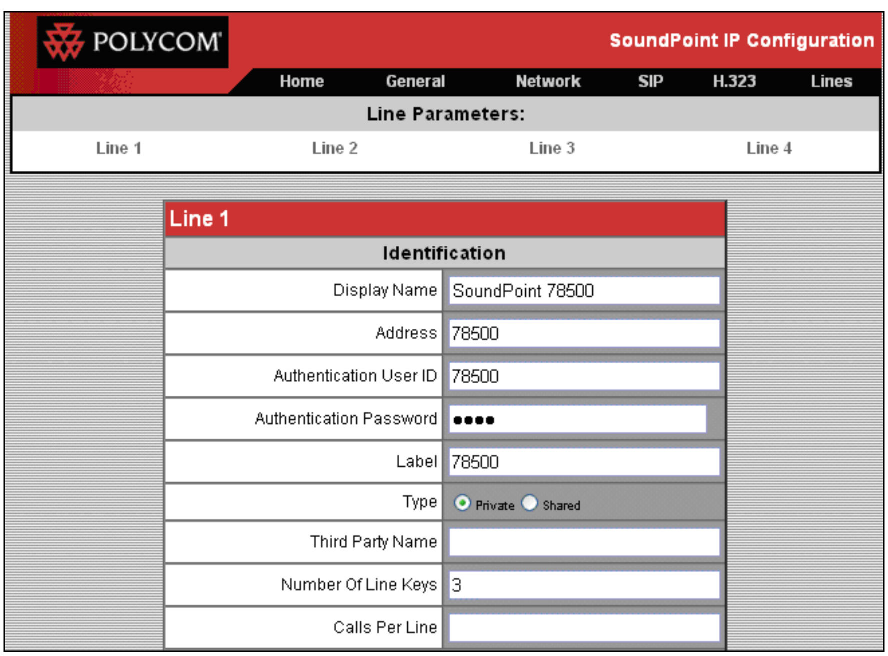
Note: Although the **Outbound Proxy Address** was configured, it was not required in this test configuration.



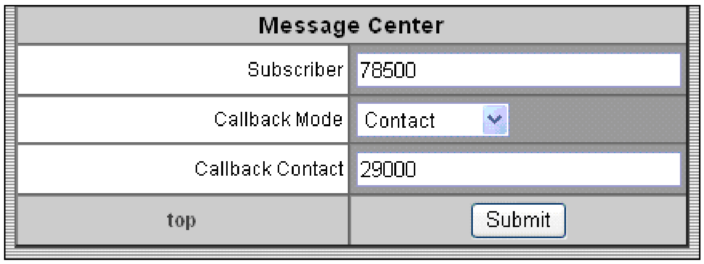
Next, scroll down to the **Local Settings** section and configure the **Digitmap** field to cover the dial strings supported by the dial plan. In this configuration, 5-digit numbers starting with '2' and '7' were supported. Click **Submit** and wait until the phone reboots.



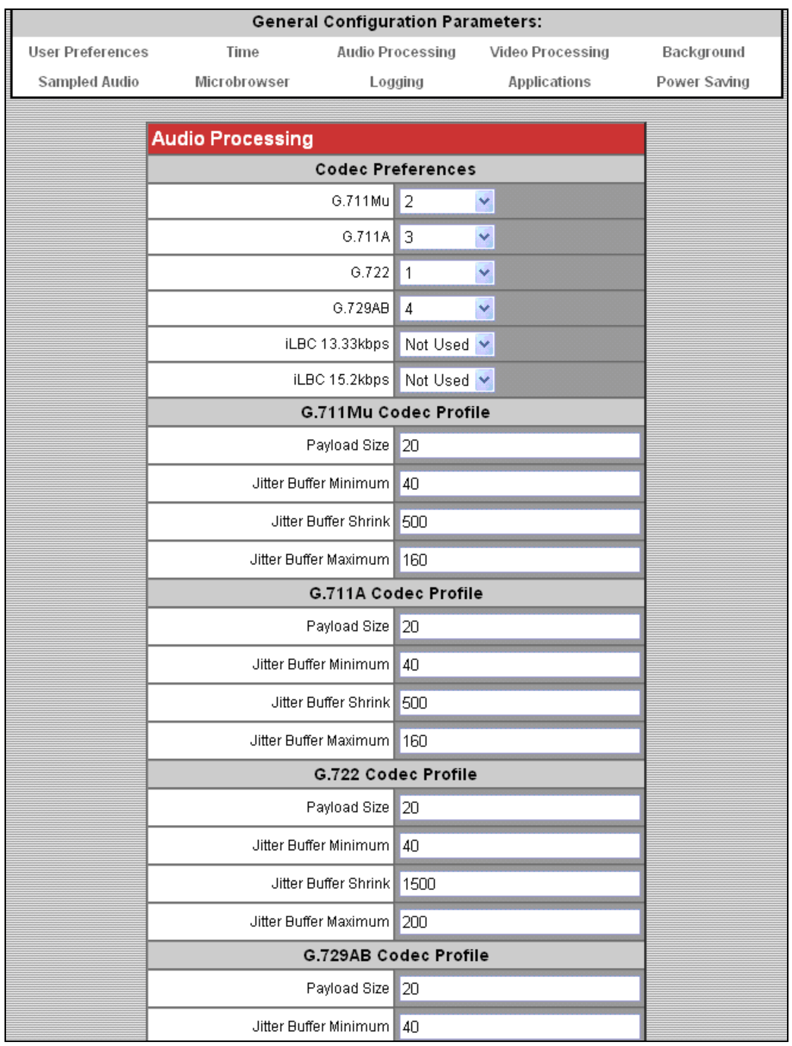
After the phone reboots, access the **Lines** screen from the phone's embedded Web interface. In the **Identification** section, provide a descriptive **Display Name** and specify the phone's extension in the **Address** field. In the **Authentication User ID** and **Authentication Password** fields,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



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



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



7. General Test Approach and Test Results

To verify interoperability of the SoundPoint IP 550 SIP Phone with Communication Manager and Session Manager, calls were made between Polycom SoundPoint IP 550 SIP Phones and Avaya SIP, H.323, and digital stations using various codec settings and exercising common PBX features. The telephony features listed in **Section 1.1** were activated and deactivated using phone buttons and FNEs. All test cases passed.

8. Verification Steps

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

1. Verify that the SoundPoint IP 550 SIP Phones have successfully registered with Session Manager.
2. Verify basic telephony features by establishing calls between a SoundPoint IP 550 SIP Phone and another phone.
3. Call a SoundPoint IP 550 SIP phone that currently has no voice messages, and leave a message. Verify that the message waiting indicator (i.e., Voicemail button) illuminates. Call the voicemail system and retrieve voice messages. Verify that after hearing all messages, that the message waiting indicator is extinguished.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Polycom SoundPoint IP 550 SIP Phone with Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager. The SoundPoint IP 550 SIP Phone successfully registered with Session Manager and basic telephony features were verified. All test cases passed.

10. References

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

[1] *Administering Avaya Aura™ Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509.

[2] *Administering Avaya Aura™ Session Manager*, August 2010, Issue 3, Release 6.0, Document Number 03-603324.

[3] *Administrator's Guide for the Polycom SoundPoint IP / SoundStation IP / VVX Family*, SIP 3.2.2, November 2009, Document Number 1725-11530-322 Rev. A.