Avaya Solution & Interoperability Test Lab

Application Notes for Voalte Platform with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Voalte Platform with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Voalte Platform is a healthcare communication solution that allows for the configuration and control of various Voalte branded smartphone communication applications. The Voalte Voice Server within Voalte Platform is a SIP proxy server acting as a back-to-back user agent (B2BUA) allowing Voalte One smartphone extensions to place and receive SIP calls from Avaya Aura® Communication Manager and the PSTN. The Voalte Voice Server integrates with Avaya Aura® Session Manager via a SIP trunk and the Voalte One smartphone extensions register directly with the Voalte Voice Server. The Voalte One smartphone applications communicated with the Avaya IP network over a converged 802.11 wireless network.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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 configuration steps required to integrate the Voalte Platform with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Voalte Platform is a healthcare communication solution that allows for the configuration and control of various Voalte branded smartphone communication applications. The Voalte Voice Server within Voalte Platform is a SIP proxy server acting as a back-to-back user agent (B2BUA) allowing Voalte One smartphone extensions to place and receive SIP calls from Avaya Aura® Communication Manager and the PSTN. The Voalte Voice Server integrates with Avaya Aura® Session Manager via a SIP trunk and the Voalte One smartphone extensions register directly with the Voalte Voice Server. The Voalte One smartphone applications communicated with the Avaya IP network over a converged 802.11 wireless network.

2 General Test Approach and Test Results

Interoperability compliance testing covered feature and serviceability testing. The feature testing focused on establishing calls between Voalte One smartphones, Avaya SIP / H.323 deskphones, and the PSTN. In addition, basic telephony features, such as hold/resume, mute, redirecting calls, and call transfers were verified.

The serviceability testing focused on verifying that the Voalte One smartphones would come back into service after rebooting the Voalte Voice Server, the wireless access point or the smartphone themselves. Also, the Voalte One smartphones were moved outside the WiFi coverage range to verify proper behavior.

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

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and the Voalte Platform did not include use of any specific encryption features as requested by Voalte.

2.1 Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Establishing SIP trunk from the Voalte Voice Server to Session Manager and verifying the exchange of SIP OPTIONS messages.
- Establishing calls between the Voalte One smartphones, the Avaya IP telephony network, and the PSTN.
- Support of G.711 μ-law codec.
- Support of direct IP-to-IP media (also known as "Shuffling" which allows IP endpoints to send audio RTP packets directly to each other without using media resources on the Avaya Media Gateway or Avaya Aura® Media Server). Calls are "shuffled" between Avaya IP Deskphones and the Voalte Voice Server.
- Basic telephony features, including hold/resume/mute, redial, multiple calls, and blind/attended call transfers.
- Call Waiting and handling multiple simultaneous calls on Voalte One smartphone application.
- Redirecting calls to an Avaya IP Deskphone or voicemail when a Voalte One smartphone is busy or doesn't answer the call.
- Playing Music on Hold from the Voalte Voice Server.
- Verification of caller display.
- Calls to the Voalte Auto Attendant to transfer calls to a Voalte One user or the designated operator.
- DTMF support.
- Exchange of SIP Options messages.
- Proper system recovery after a restart of the Voalte Voice Server, Voalte One smartphone applications, or access point.

2.2 Test Results

All test cases passed with the following observations:

- Voalte Platform does not provide conference controller services. However, Voalte One users may be connected to a conference by means of dialing a conference bridge number or being called by a conference, but conference mixing must be hosted outside of the Voalte Platform.
- When the Voalte Voice Server redirects or transfers a call, the server remains bridged to the call until the redirected or transferred call is terminated. This is per design.
- Adjust the **Call Establishment Timeout** field on Voalte One to allow enough time for redirected calls to complete. During the compliance test, this field was set to 80 seconds.

2.3 Support

For Technical support on the Voalte Platform, contact Voalte support via one of these methods:

- **Phone:** (844) 283-7110
- **Web:** https://www.voalte.com/services-support
- **Email:** support@voalte.com

3 Reference Configuration

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

- Avaya Aura® Communication Manager with an Avaya G450 Media Gateway.
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP deskphones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Aura® Messaging serving as the voicemail system. Used by Avaya IP Deskphones and Voalte One users for voicemail coverage.
- Avaya 96x1 Series SIP and H.323 Deskphones.
- Voalte Platform consisting of a Voalte Voice Server and Voalte One smartphone applications.
- A Voalte-approved wireless access point used to provide the Voalte One smartphone access to the converged 802.11 wireless network.

Voalte Voice Server connected to Session Manager via a SIP trunk and the Voalte One smartphone applications registered directly to the Voalte Voice Server.

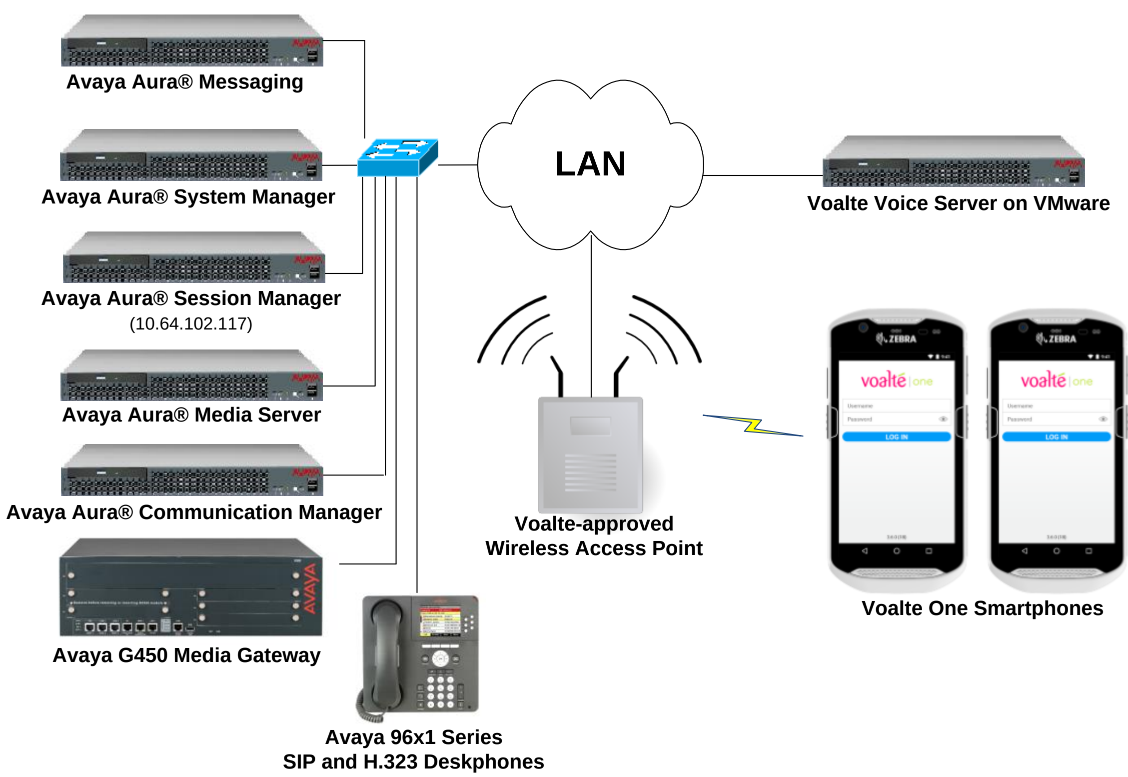


Figure 1: Voalte Platform with Avaya SIP Telephony Network

4 Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	8.0 SP1 (R018x.00.0.822.0 with Patch 24796)
Avaya G450 Media Gateway	FW 38.21.1
Avaya Aura® Media Server	v.7.8.0.393
Avaya Aura® Session Manager	8.0.0.0.800035
Avaya Aura® System Manager	8.0.0 Build No. – 8.0.0.0.931077
Avaya Aura® Messaging	7.1.3.1.0-FP3SP1
Avaya 96x1 Series IP Deskphones	6.6506 (H.323) 7.1.1.0.9 (SIP)
Voalte Platform including: ▪ Voalte Voice Server ▪ Voalte One running on Zebra TC51 Touch Computer	 3.6.1.308 3.6.1 (40)

5 Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Administer IP Node Names
- Administer IP Codec Set
- Administer IP Network Region
- Administer SIP Trunk Group to Session Manager
- Administer AAR Call Routing

Use the System Access Terminal (SAT) to configure Communication Manager and log in with appropriate credentials.

5.1 Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager when configuring the SIP trunk to Session Manager.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon-aes          10.64.102.119
devcon-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
procr6              ::

( 6  of 6    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
```

5.2 Administer IP Codec Set

In the **IP Codec Set** form, accessed via the **change ip-codec-set 1** command, specify the G.711 μ-law codec, which is supported by the Voalte Platform. On the Voalte One smartphone application, the codec was set to SILK. Note the codec set number since it will be used in the IP Network Region covered in the next section.

```
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:
```

5.3 Administer IP Network Region

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 G450 Media Gateway or Avaya Aura® Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: avaya.com
    Name:                       Stub Network Region: n
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: 50999
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.4 Administer SIP Trunk to Session Manager

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 *tls*.
- Specify the Communication Manager (*procr*) 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 TLS port value of *5061* 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 10                                      Page   1 of   2
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                       Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: devcon-sm
 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? y                     Initial IP-IP Direct Media? y
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 SIP calls to Voalte Voice Server. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 10                                                Page   1 of  5
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: To devcon-sm               COR: 1        TN: 1        TAC: 1010
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 10
                                                     Number of Members: 10
```

To display the Voalte One user CPN on a called party's display, set the **Identity for Calling Party Display** to the appropriate value. If the field is set to *P-Asserted-Identity* (PAI), then the Voalte Voice Server must send the PAI header in the SIP INVITE message. If the field is set to *From*, Communication Manager will use the CPN in the From header of the SIP INVITE message.

```
add trunk-group 10                                                Page   5 of   5
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? n

                                      Send Diversion Header? n
                                    Support Request History? y
                               Telephone Event Payload Type:


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n

          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
```

5.5 AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and add an entry that routes 5-digit extensions starting with “78” to route pattern 10 as shown below. Extensions starting with “789” will be routed to Voalte Voice Server by Session Manager. 5-digit numbers beginning with “78” are part of the uniform dial plan in Communication Manager.

```
change aar analysis 78                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 1

          Dialed           Total      Route     Call   Node   ANI
          String          Min  Max  Pattern     Type   Num    Reqd
    78                    5    5      10        lev0          n
```

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below.

```
change route-pattern 10                                         Page   1 of   3
                    Pattern Number: 10     Pattern Name: To devcon-sm
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                           DCS/ IXC
    No          Mrk Lmt List Del  Digits                             QSIG
                             Dgts                                    Intw
 1: 10   0                                                            n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               unk-unk   none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
```

6 Configure Avaya Aura® Session Manager

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

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager and Communication Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies
- Dial Patterns
- Session Manager, corresponding to the Avaya Aura® Session Manager Server to be managed by Avaya Aura® System Manager

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.

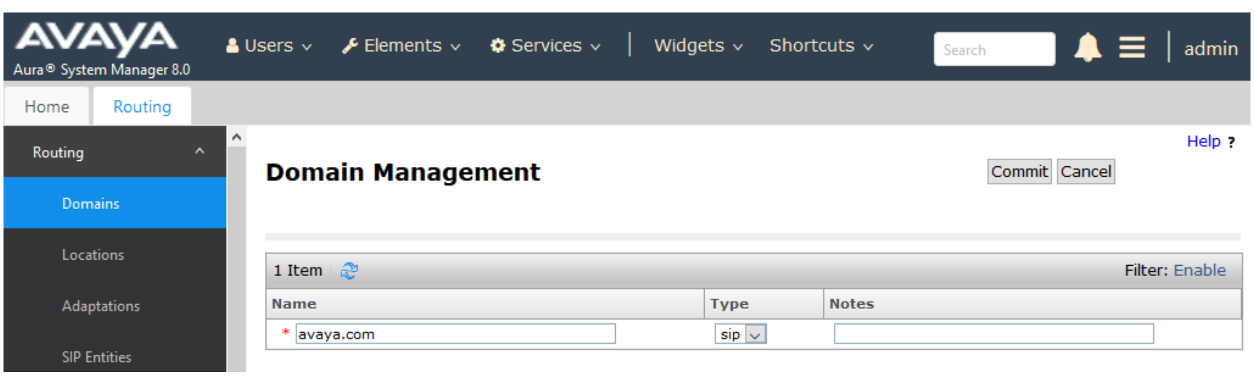
6.1 Specify SIP Domain

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 (not shown). The following screen will then be shown. Fill in the following:

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

Click **Commit**.

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



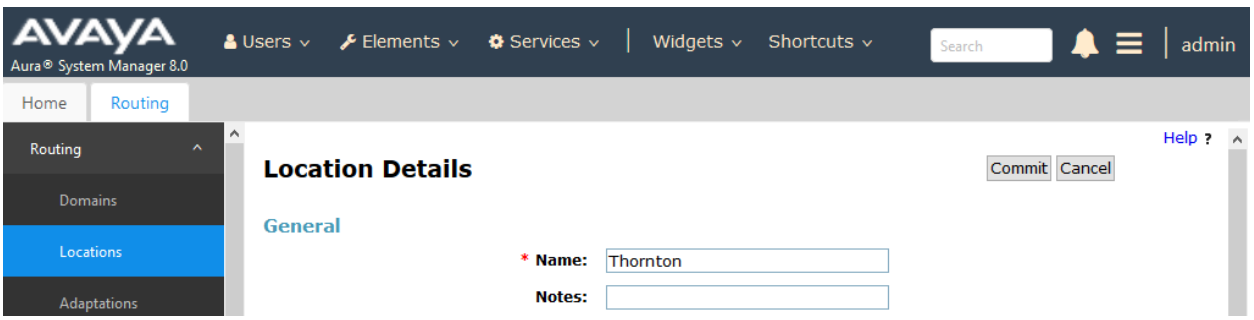
6.2 Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (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).

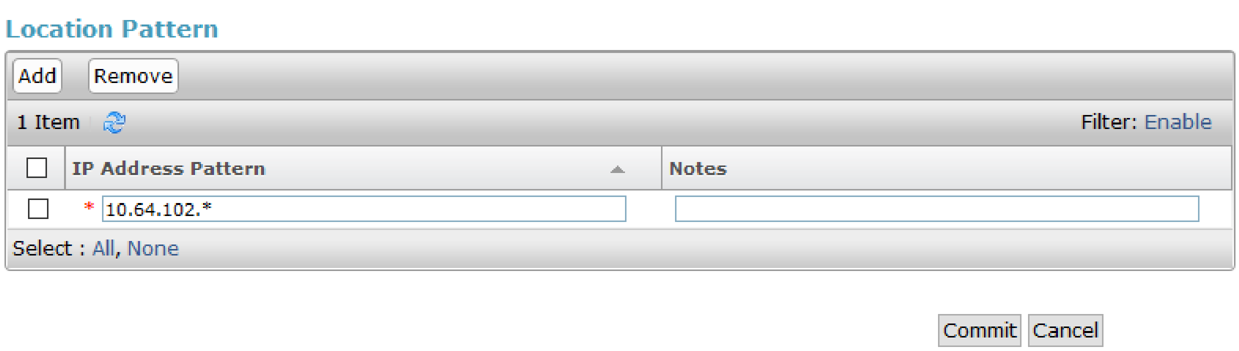
The screen below shows addition of the *Thorton* location, which includes Avaya Aura® Communication Manager and Avaya Aura® Session Manager.



Under *Location Pattern*:

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

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



6.3 Add SIP Entities

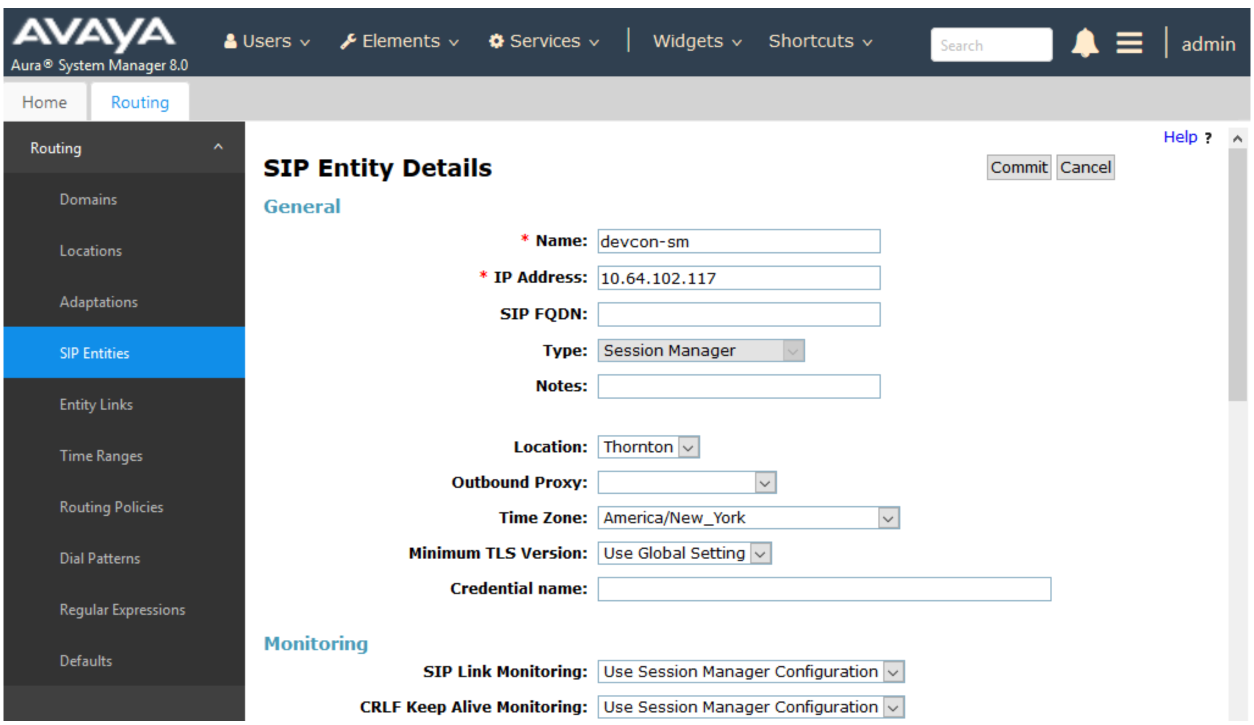
In the sample configuration, a SIP Entity is added for Session Manager, Communication Manager, and Voalte Voice Server.

6.3.1 Avaya Aura® Session Manager

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

Under *General*:

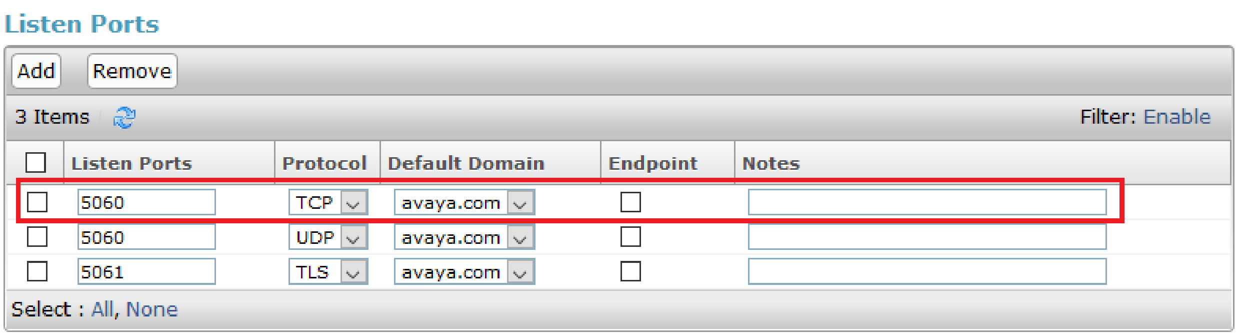
- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.



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

- **Listen Ports:** 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** (not shown) to save the SIP Entity definition.



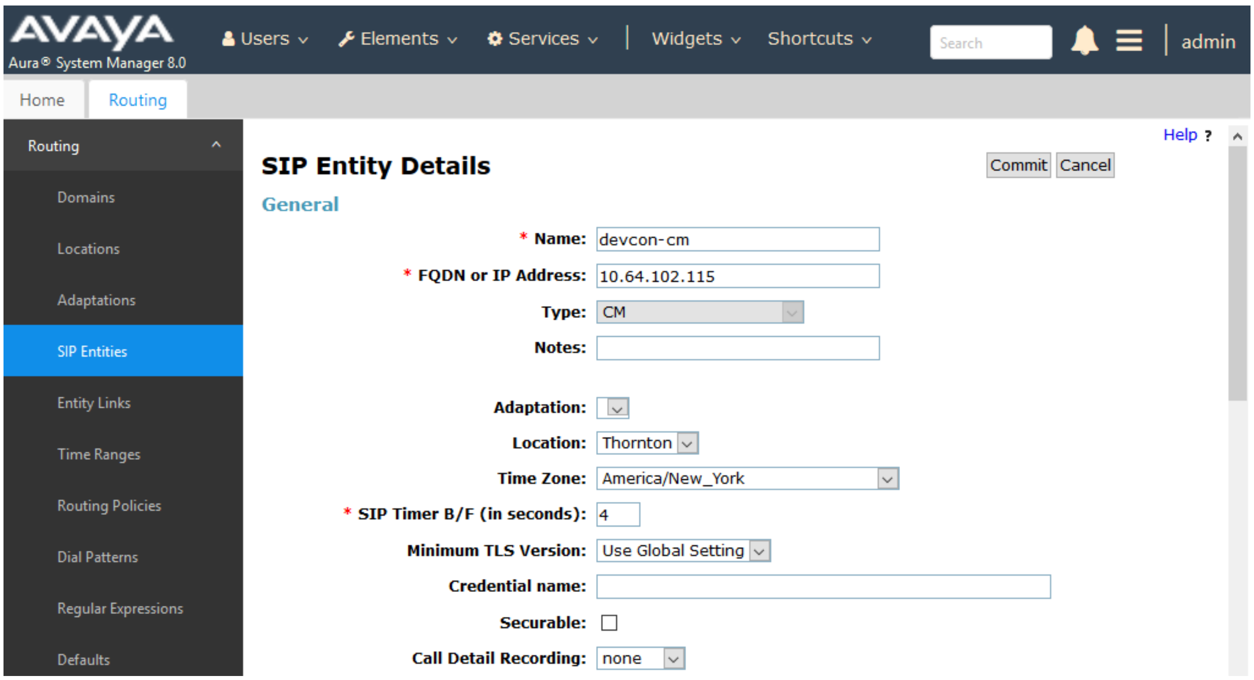
6.3.2 Avaya Aura® Communication Manager

A SIP Entity must be added for Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (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 (e.g., Communication Manager (*procr*)) 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.

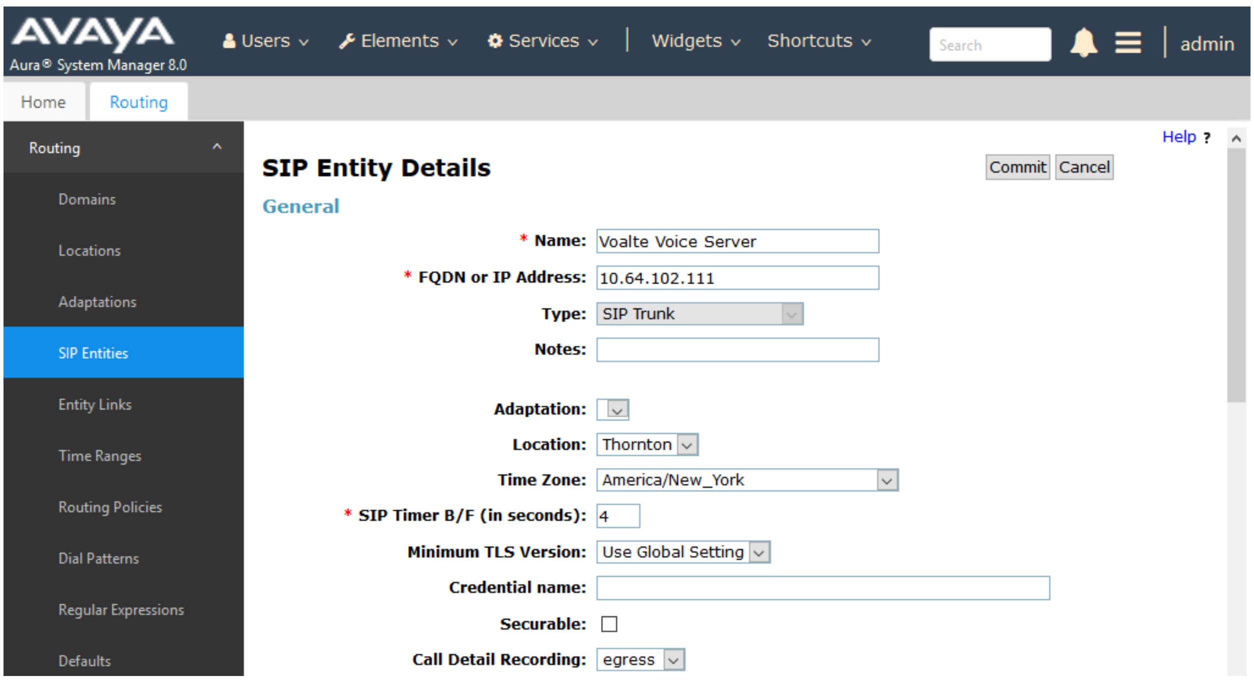


6.3.3 Voalte Voice Server

A SIP Entity must be added for Voalte Voice Server. 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 Voalte Voice Server.
- **Type:** Select *SIP Trunk*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.



6.4 Add Entity Links

This section covers the configuration of Entity Links for Communication Manager and Voalte Voice Server.

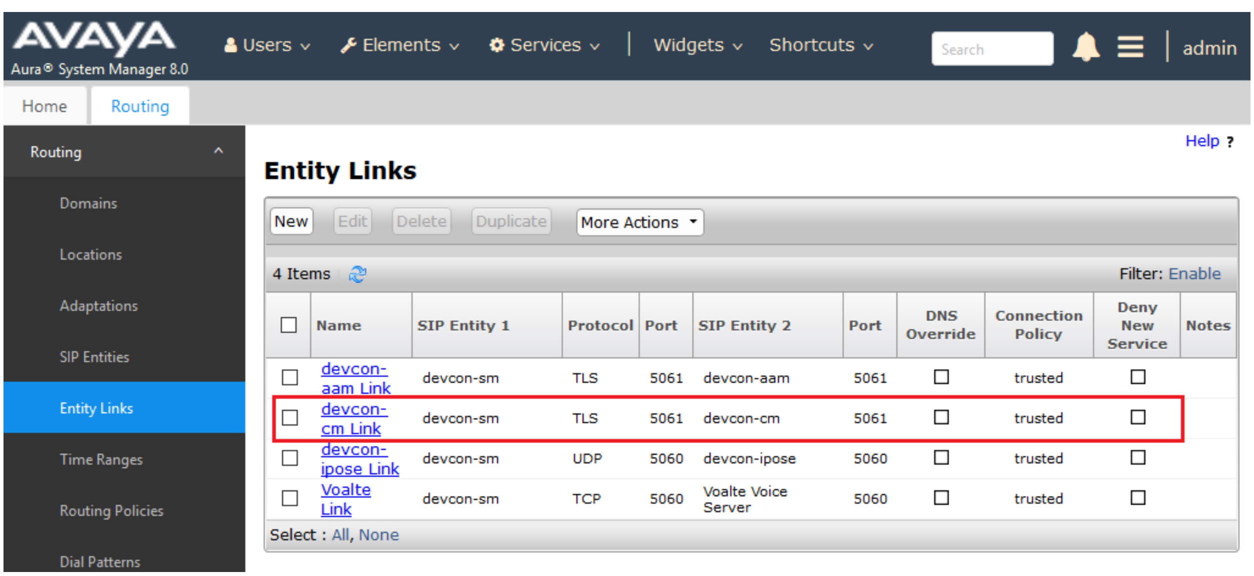
6.4.1 Communication Manager Entity Link

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., *devcon-cm 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.
- **Connection Policy:** Select *trusted*. *Note: If trusted is not selected, calls from the associated SIP Entity specified in* ***Section 6.3.2*** *will be denied.*

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

The following Entity Link is between Session Manager and Communication Manager.

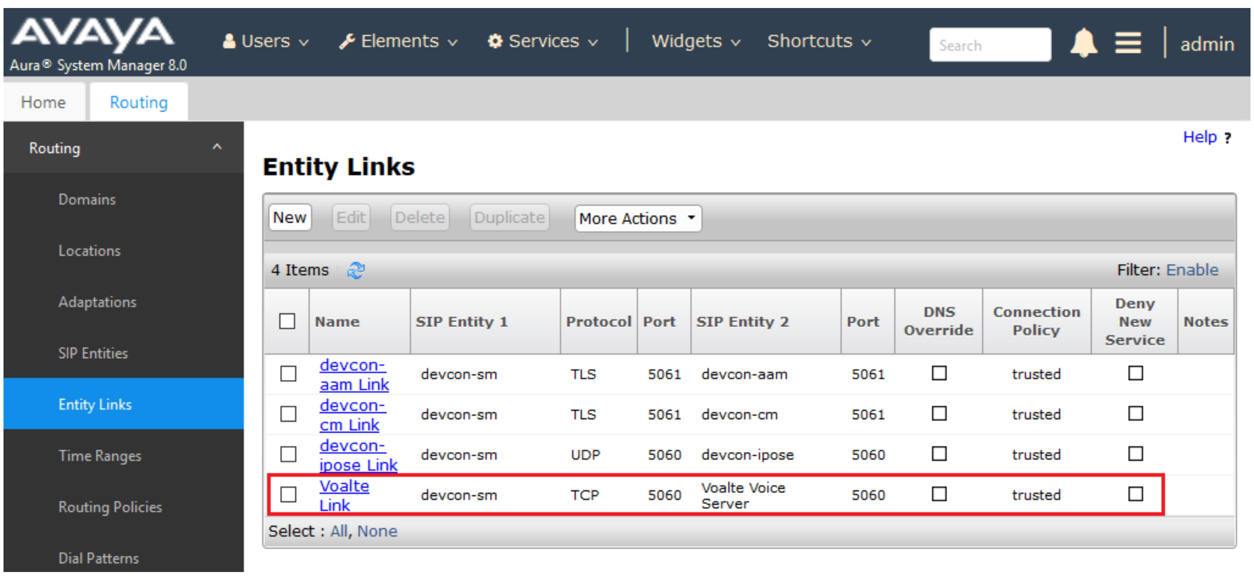


6.4.2 Voalte Voice Server Entity Link

The SIP trunk from Session Manager to Voalte Voice Server 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., *Voalte 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 *Voalte Voice Server* SIP entity.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Selected *trusted*. *Note: If the link is not trusted, calls from the associated SIP Entity specified in* ***Section 6.3.3****will be denied.*

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



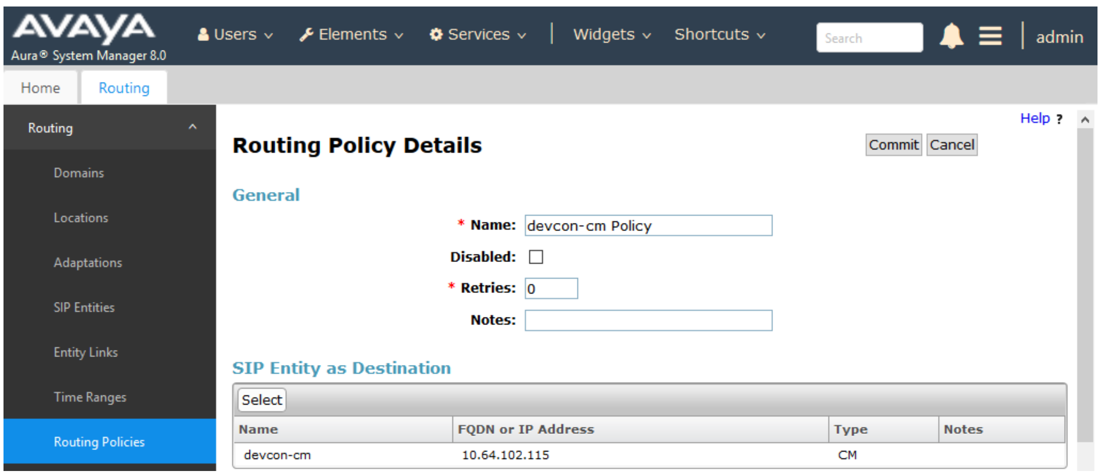
6.5 Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.3.** Two routing policies were added – one for Communication Manager and one for Voalte Voice Server. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

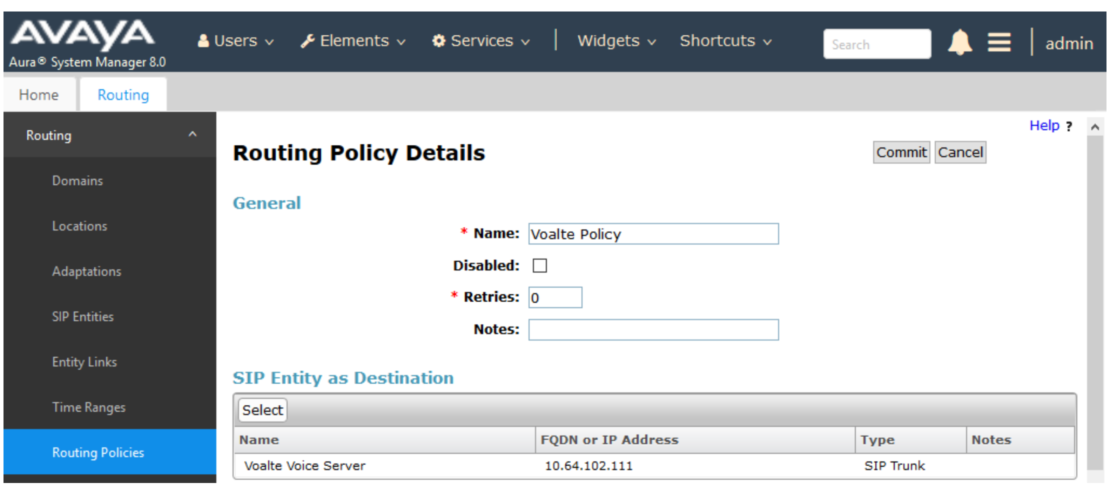
Under *General*:
Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy for Voalte Voice Server.



6.6 Add Dial Patterns

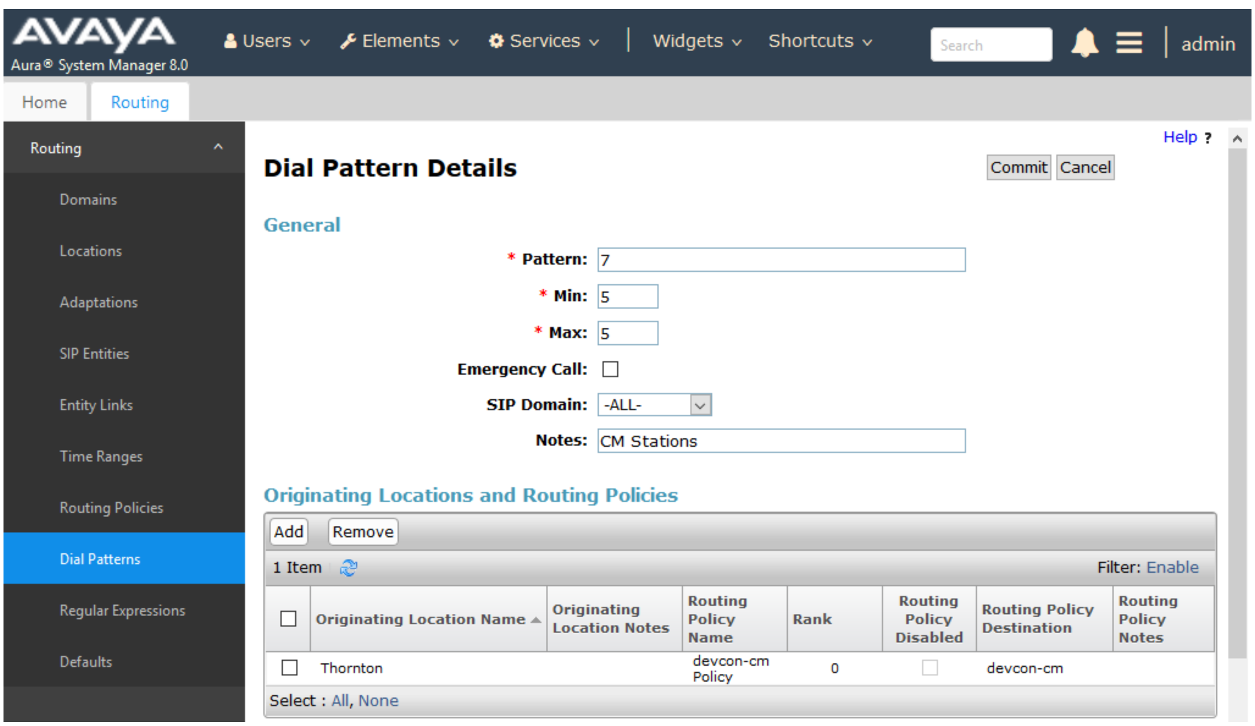
Dial patterns must be defined to direct calls to the appropriate SIP Entity. In the sample configuration, a 5-digit number beginning with '7' will be routed to Communication Manager and a 5-digit number beginning with "789" will be routed to Voalte Voice Server. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

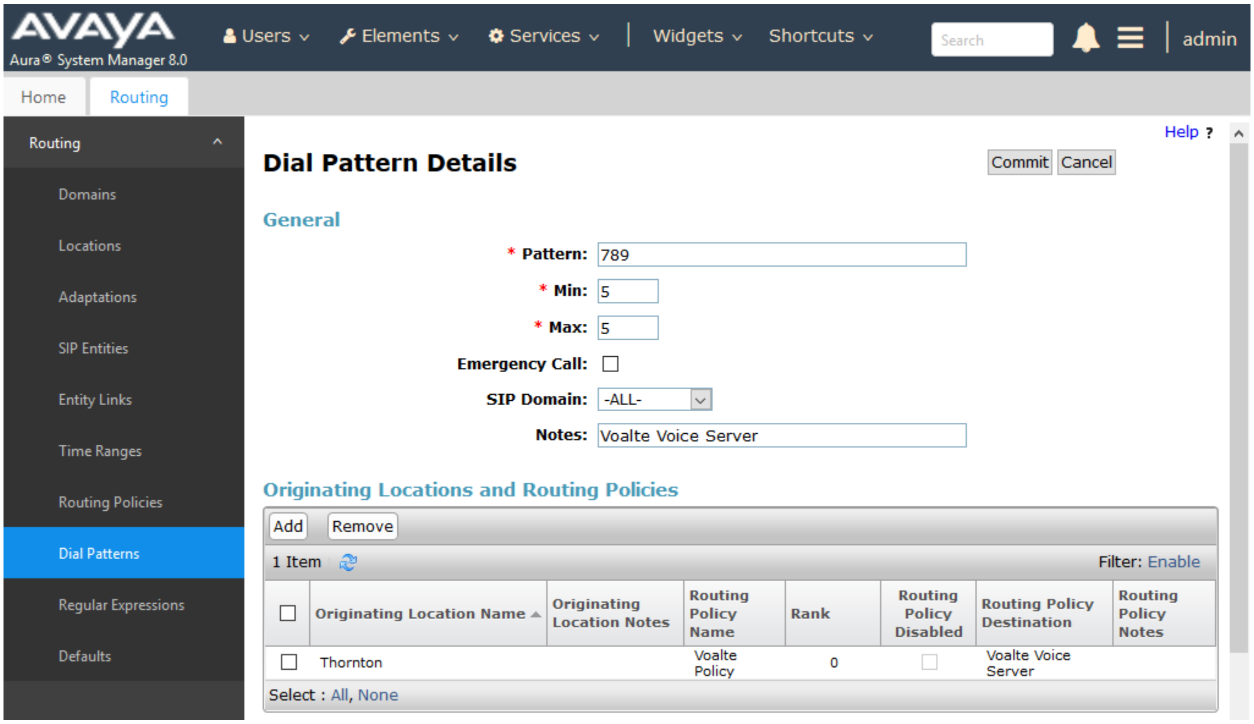
- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern (optional).

Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definition for Communication Manager. Based on these digits, Session Manager will route the call to Communication Manager via a SIP trunk.



The following screen shows the dial pattern definition for Voalte Voice Server. Extensions starting with “789” correspond to Voalte One users and the Auto Attendant. Session Manager will route these calls to Voalte Voice Server via a SIP trunk.



6.7 Add Session Manager

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

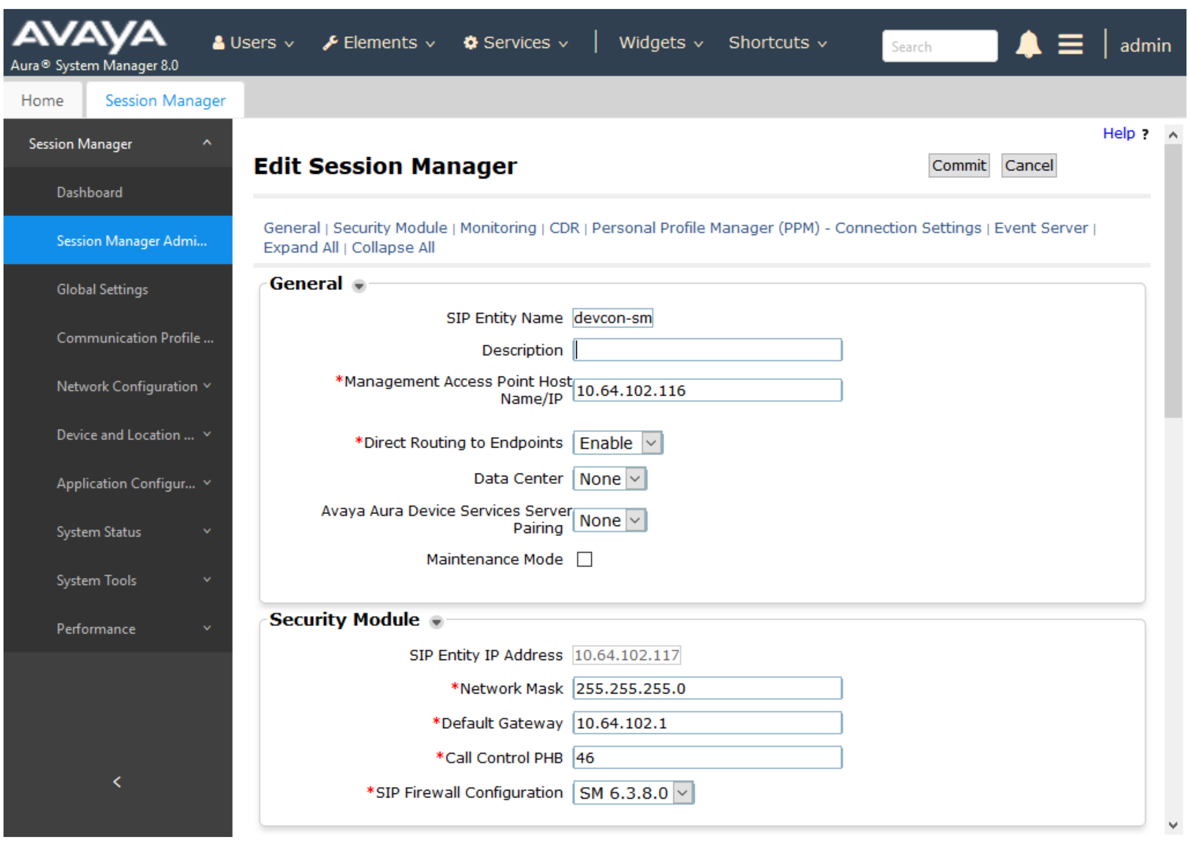
Under *Identity*:

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

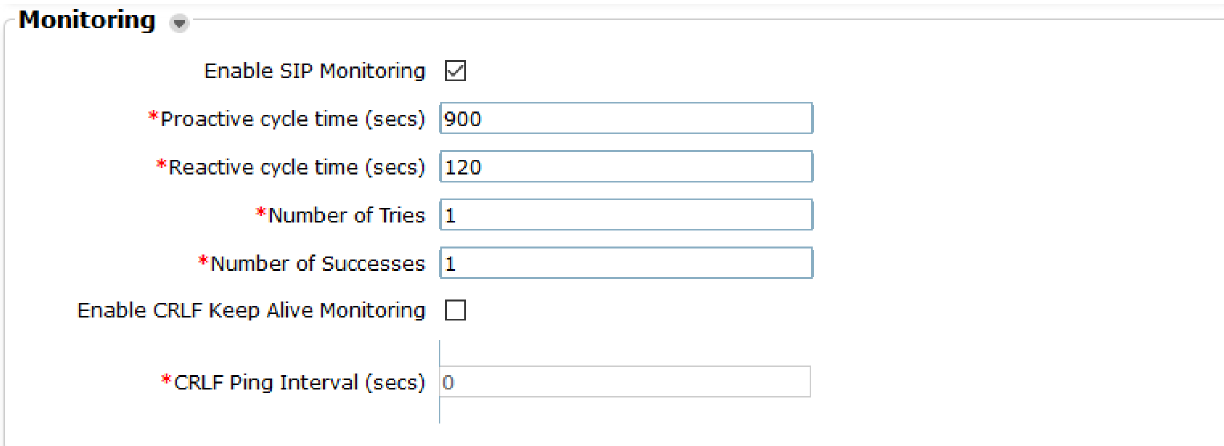
Under *Security Module*:

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

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



The following screen shows the **Monitoring** section, which determines how frequently Session Manager sends SIP Options messages to Voalte Voice Server. Use default values for the remaining fields. Click **Commit** to add this Session Manager. In the following configuration, Session Manager sends a SIP Options message every 900 secs. If there is no response, Session Manager will send a SIP Options message every 120 secs.



7 Configure Voalte Platform

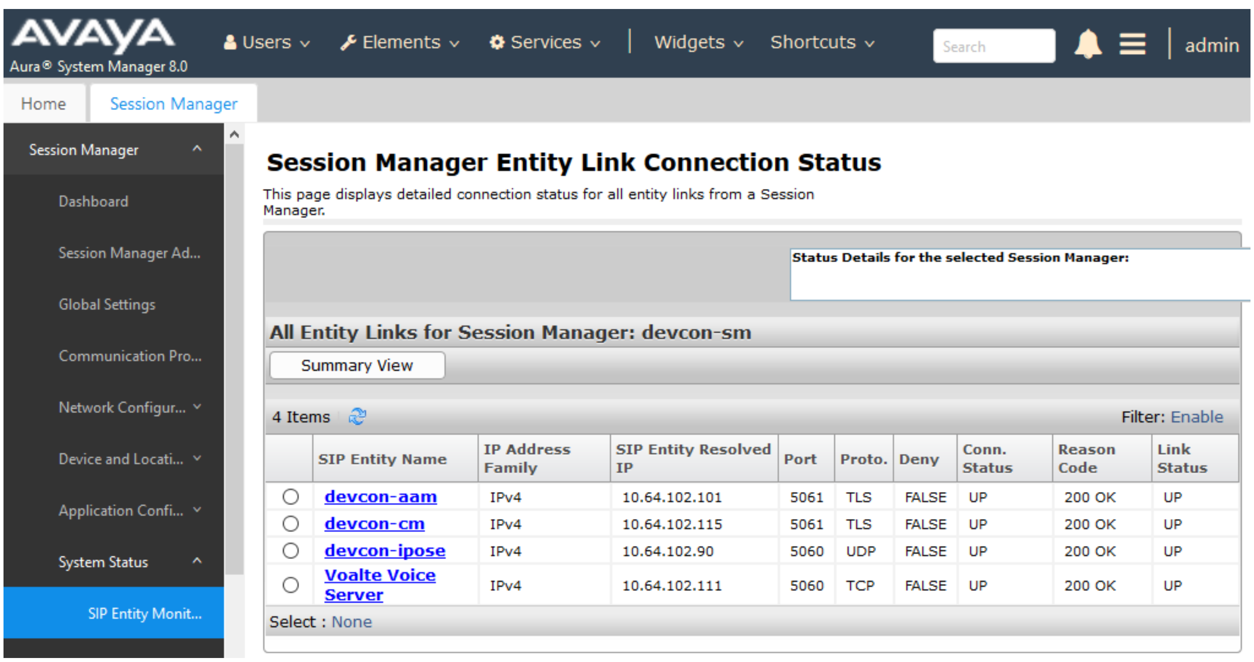
The Voalte Platform solution is a managed service and is configured and managed by Voalte support. Therefore, Voalte has requested not to include configuration details in these Application Notes. If issues or questions arise related to the Voalte Platform, contact to Voalte directly as shown in **Section 2.3**. For an overview of how the Voalte Voice Server integrates with an IP telephony network, refer to **[3]**. For more information on using Voalte One smartphone application, refer to **[4]**.

8 Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager, Avaya Aura® Session Manager and the Voalte Platform.

1. Log into Voalte One. Verify that the Voalte One smartphone application can communicate with the Voalte Voice Server and successfully log in.

2. On System Manager, verify that the SIP trunks to Communication Manager and Voalte Voice Server are up by navigating to **Home→Elements→Session Manager→System Status→SIP Entity Monitoring** and clicking on the Session Manager link to display all the entity links. The **Link Status** should be *UP* for each entity link.



3. Place a call from a Voalte One user to an Avaya IP Deskphone. Verify that the call is successfully established with two-way audio. Repeat the call in the opposite direction.

4. To view the call status on Communication Manager, use the **status trunk** command, with the trunk member option, to display the trunk status while the call is active, including the IP endpoints involved, codec type, and whether the call was shuffled or not. If the call is shuffled, the **Audio Connection Type** field would be set to *ip-direct*; otherwise, the field would be set to *ip-tdm*.

```
status trunk 10/1                                              Page   1 of   3
                              TRUNK STATUS

 Trunk Group/Member: 0010/0001            Service State: in-service/active
              Port: T00001             Maintenance Busy? no
 Signaling Group ID: 10

   IGAR Connection? no

    Connected Ports: S00012
```

```
status trunk 10/1                                              Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling   IP Address                             Port
   Near-end:  10.64.102.115                        : 5061
    Far-end:  10.64.102.117                        : 5061
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.711MU
   Audio      IP Address                             Port
   Near-end:  192.168.100.55                       : 13360
    Far-end:  10.64.102.111                        : 17500

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:              Video Far-end Codec:
```

9 Conclusion

These Application Notes describe the configuration steps required to integrate the Voalte Voice Server with an Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Voalte One users registered with the Voalte Voice Server, which in turn was able to establish calls using the SIP trunk to Avaya Aura® Session Manager. Basic telephony feature were verified. All test cases passed with observations noted in **Section 2.2**.

10 References

This section references the product documentation relevant to these Application Notes.

[1] *Administering Avaya Aura® Communication Manager*, Release 8.0, Issue 1, July 2018, available at http://www.avaya.com..

[2] *Administering Avaya Aura® Session Manager*, Release 8.0, Issue 2, August 2018, available at http://www.avaya.com.

[3] *Voalte Voice Service and PBX Integration Overview*, available from Voalte.

[4] *Voalte One User Manual*, Version 3.6, Rev 01. 1.0, available from Voalte.

[5] *Voalte One for Zebra TC51-HC Getting Started*, available at Voalte.

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.