Avaya Solution & Interoperability Test Lab

Application Notes for configuring Parlance Operator Assistant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Parlance Operator Assistant to interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. Parlance Operator Assistant automates call routing by asking callers to speak the name or dial the extension of a destination.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 for Parlance Operator Assistant (hereafter referred to as Operator Assistant) to interoperate with Avaya Aura® Session Manager 7.0 (hereafter referred to as Session Manager) and Avaya Aura® Communication Manager 7.0 (hereafter referred to as Communication Manager) using SIP trunks. Parlance Operator Assistant automates call routing by asking callers to speak the name or dial the extension of a destination.

In the compliance testing, calls from internal and external callers were routed over SIP trunks to Parlance Operator Assistant. Parlance Operator Assistant played different greeting announcements based on ANI and/or DNIS, used speech recognition and/or DTMF digits to determine the route destination, and used SIP REFER to transfer calls to destinations on Avaya Aura® Communication Manager or on the PSTN.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were placed manually from users on the PSTN and on Communication Manager to Operator Assistant. Speech and DTMF input were used from the callers for requesting transfer to internal user and group destinations on Communication Manager, and to external destinations on the PSTN.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet connection to Operator Assistant.

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.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included G.711MU, ANI, DNIS, speech recognition, DTMF, speaking ahead (barge-in), dialing ahead, call forwarding, invalid number, blind transfer, supervised transfer and incoming simultaneous calls.

The serviceability testing focused on verifying the ability of Operator Assistant to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to Operator Assistant.

2.2. Test Results

All test cases were executed, and the following were observations on Operator Assistant:

- The application only supports the G.711MU codec.

- For Supervised transfer, changes needs to be done in the **PhoneConfig_Overrides.ini** file in the Operator Assistant as shown below, where **10.10.97.228** is the IP address of the Session Manager.

```
[Generic]
;managed_transfer_template = None
basic_transfer_template = sip:%s@10.10.97.228
;sip_2_sip_transfertype = conditional
```

2.3. Support

Technical support on Operator Assistant can be obtained through the following:

- **Phone:** (888) 700-6263
- **Email:** customerservice@parlancecorp.com
- **Web :** www.parlancecorp.com

3. Reference Configuration

As shown in **Figure 1**, SIP trunks were used between Session Manager and Operator Assistant.

A five digit Uniform Dial Plan (UDP) was used to facilitate routing with Operator Assistant. Unique extension ranges were assigned to users on Communication Manager (56xxx), and to Operator Assistant (30xxx).

The configuration of Session Manager is performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Manager, System Manager, and Session Manager is not the focus of these Application Notes and will not be described.

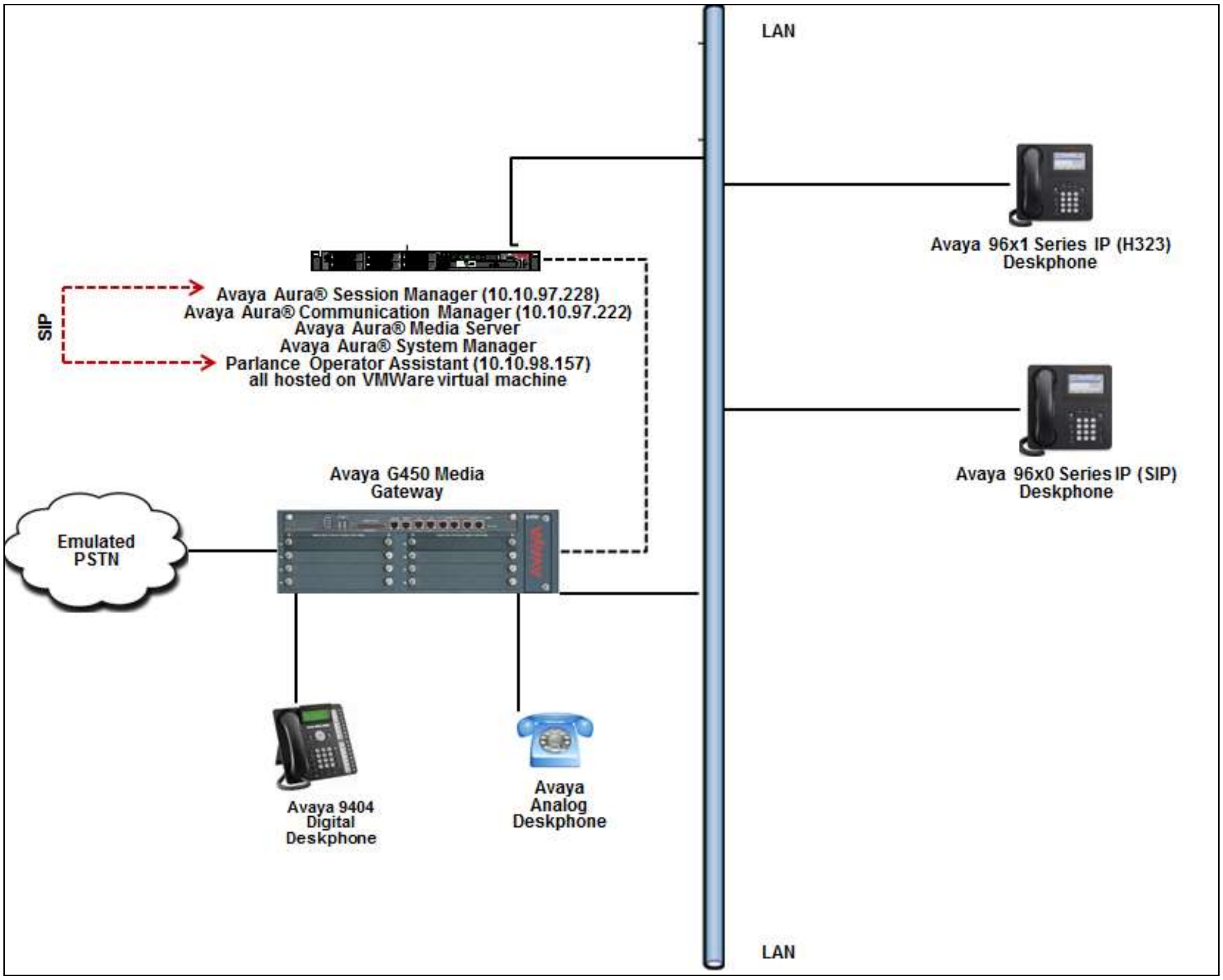


Figure 1: Compliance Testing Configuration

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 in Virtual Environment	7.0.0.3.0-SP3 (R017x.00.0.441.0)
Avaya G450 Media Gateway	37 .21 .0 /1
Avaya Aura® Media Server in Virtual Environment	7.7.0.292
Avaya Aura® Session Manager in Virtual Environment	7.0.0.2.700201
Avaya Aura® System Manager in Virtual Environment	7.0.0.2
Avaya IP Deskphones: • 9641 (H.323) • 9621 (SIP)	 6.6115 7.0.0.39
Avaya Digital Deskphone (9404)	R 0.15 V21
Avaya Analog Deskphone	N/A
Parlance Operator Assistant running on Microsoft Windows Server 2012 R2	N/A

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify license
- Administer system parameters features
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer dial plan
- Administer uniform dial plan
- Administer AAR analysis

In the compliance testing, a separate set of codec set, network region, trunk group, and signaling group were used for integration with Operator Assistant.

5.1. Verify License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  10
           Maximum Concurrently Registered IP Stations: 2400  2
             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  3
                       Maximum Administered SIP Trunks: 4000  24
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
```

5.2. Administer System Parameters Features

Use the "change system-parameters features" command to allow for trunk-to-trunk transfers.

For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to "all" to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class Of Restriction or Class Of Service levels. Refer to **Section 10** for more details.

```
change system-parameters features                               Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y

              Music (or Silence) on Transferred Trunk Calls? no
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
```

5.3. Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "1". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"

```
Add trunk-group 1                                                Page   1 of  22
                                TRUNK GROUP

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: Trunk to SM on VM            COR: 1        TN: 1        TAC: #001
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group:
                                                     Number of Members:
```

Navigate to **Page 3**, and enter "private" for **Numbering Format**.

```
change trunk-group 1                                                Page   3 of  22
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y




                     Numbering Format: private
                                                UUI Treatment: shared
                                              Maximum Size of UUI Contents: 128
                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
```

5.4. Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "1". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Transport Method:** "tcp"
- **Near-end Node Name:** An existing C-LAN node name or "procr".
- **Far-end Node Name:** The existing node name for Session Manager.
- **Near-end Listen Port:** An available port for integration with Parlance.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with Parlance.
- **Far-end Domain:** The applicable domain name for the network.
- **Direct IP-IP Audio Connections:** "y"

```
add signaling-group 1                                           Page   1 of   3
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                    Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 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: SM-VM
 Near-end Listen Port: 5060                   Far-end Listen Port: 5060
                                           Far-end Network Region: 1

Far-end Domain: bvwdev.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): 6
```

5.5. Administer SIP Trunk Group Members

Use the "change trunk-group n" command, where "n" is the trunk group number from **Section 5.3**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Signaling Group:** The signaling group number from **Section 5.4**.
- **Number of Members:** The desired number of members, in this case "24".

```
change trunk-group 1                                            Page   1 of  22
                                TRUNK GROUP

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: Trunk to SM on VM          COR: 1        TN: 1        TAC: #001
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                            Member Assignment Method: auto
                                                     Signaling Group: 1
                                                   Number of Members: 24
```

5.6. Administer IP Network Region

Use the "change ip-network-region n" command, where "n" is the existing far-end network region number used by the SIP signaling group from **Section 5.4**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter "yes" for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. For **Codec Set**, enter an available codec set number for integration with Operator Assistant.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:           Authoritative Domain: bvwdev.com
    Name: Region1                  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: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

Navigate to **Page 4**, and specify this codec set to be used for calls with network regions used by Avaya endpoints and by the trunk to the PSTN. In the compliance testing, network region "1" was used by the Avaya endpoints and by the trunk to the PSTN.

```
change ip-network-region 1                                      Page   4 of  20

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units     Total Norm  Prio Shr Regions          CAC  R  L    e
 1    1                                                              all
 2
```

5.7. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number from **Section 5.6**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that Operator Assistant only supports the G.711 codec variant. The codec shown below was used in the compliance testing.

```
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:
```

5.8. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach Operator Assistant, in this case "1". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.3**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 1                                              Page   1 of   3
                    Pattern Number: 1       Pattern Name: To SM on VM
    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: 1    0                     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                               lev0-pvt  none
```

5.9. Administer Private Numbering

Use the “change private-numbering 0” command, to define the calling party number to send to Operator Assistant. Add an entry for the trunk group defined in **Section 5.3**. In the example shown below, all calls originating from a 5-digit extension beginning with 56 and routed to trunk group 1 will result in a 5-digit calling number. The calling party number will be in the SIP “From” header.

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

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  56               1                           5     Total Administered: 4
                                                        Maximum Entries: 540
```

5.10. Administer Dial Plan

This section provides a sample dial plan used for routing calls with dialed digits 30xxx to Operator Assistant. Use the “change dialplan analysis 0” command, and add an entry to specify the use of digits pattern 30, as shown below

```
display dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 2

    Dialed     Total  Call      Dialed    Total  Call      Dialed    Total  Call
    String     Length Type      String    Length Type      String    Length Type
   1              4   ext
   30             5   ext
```

5.11. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 30xxx to Operator Assistant. Note that other routing methods may be used. Use the “change uniform-dialplan 0” command, and add an entry to specify the use of AAR for routing of digits 30xxx, as shown below.

```
change uniform-dialplan 0                                       Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                            Percent Full: 0

  Matching                   Insert              Node
  Pattern          Len Del   Digits        Net Conv Num
 30                 5   0                  aar  n
```

5.12. Administer AAR Analysis

Use the "change aar analysis 0" command, and add an entry to specify how to route calls to 30xxx. In the example shown below, calls with digits 30xxx will be routed as an AAR call using route pattern "1" from **Section 5.8**.

```
change aar analysis 0                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 2

          Dialed           Total      Route    Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
        30                    5    5      1        aar           n
```

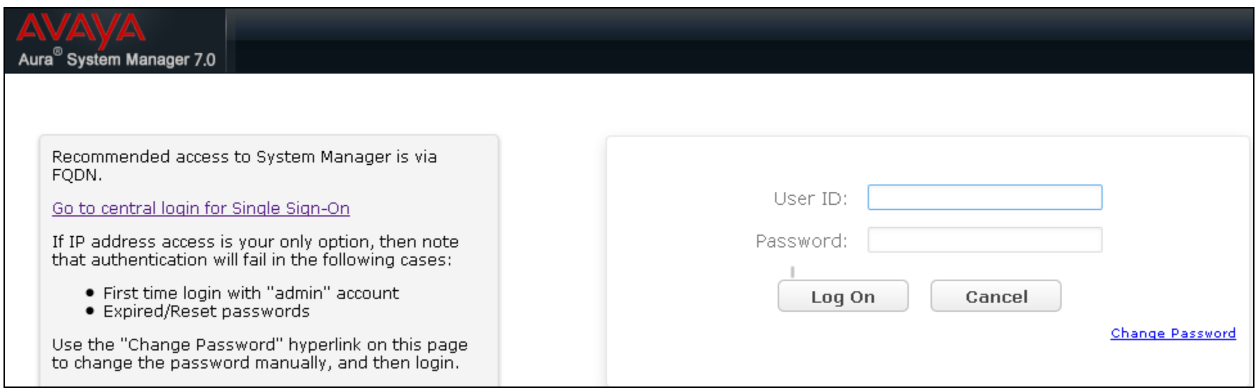
6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer domains
- Administer locations
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

6.1. Launch System Manager

Access the System Manager web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



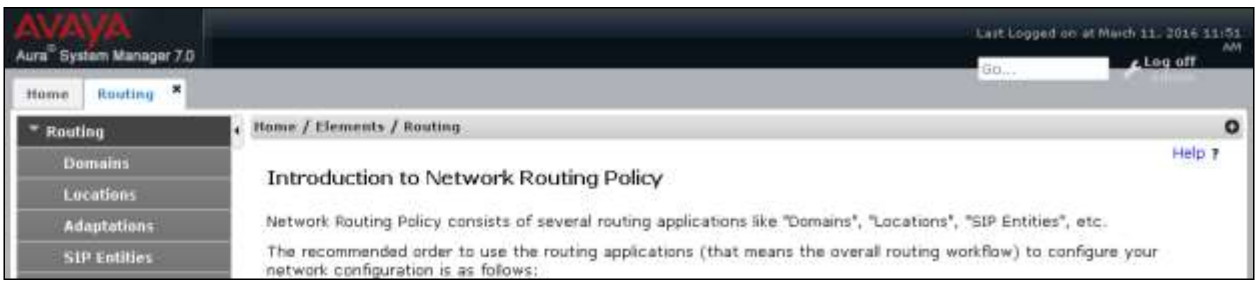
6.2. Administer Domains

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Domains** from the left pane, and click **New** in the subsequent screen (not shown) to add a new domain

The **Domain Management** screen is displayed. In the **Name** field enter the domain name, select *sip* from the **Type** drop down menu and provide any optional **Notes**.

6.3. Administer Locations

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for Operator Assistant.



The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.

Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of all devices involved in the compliance testing in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.

6.4. Administer SIP Entities

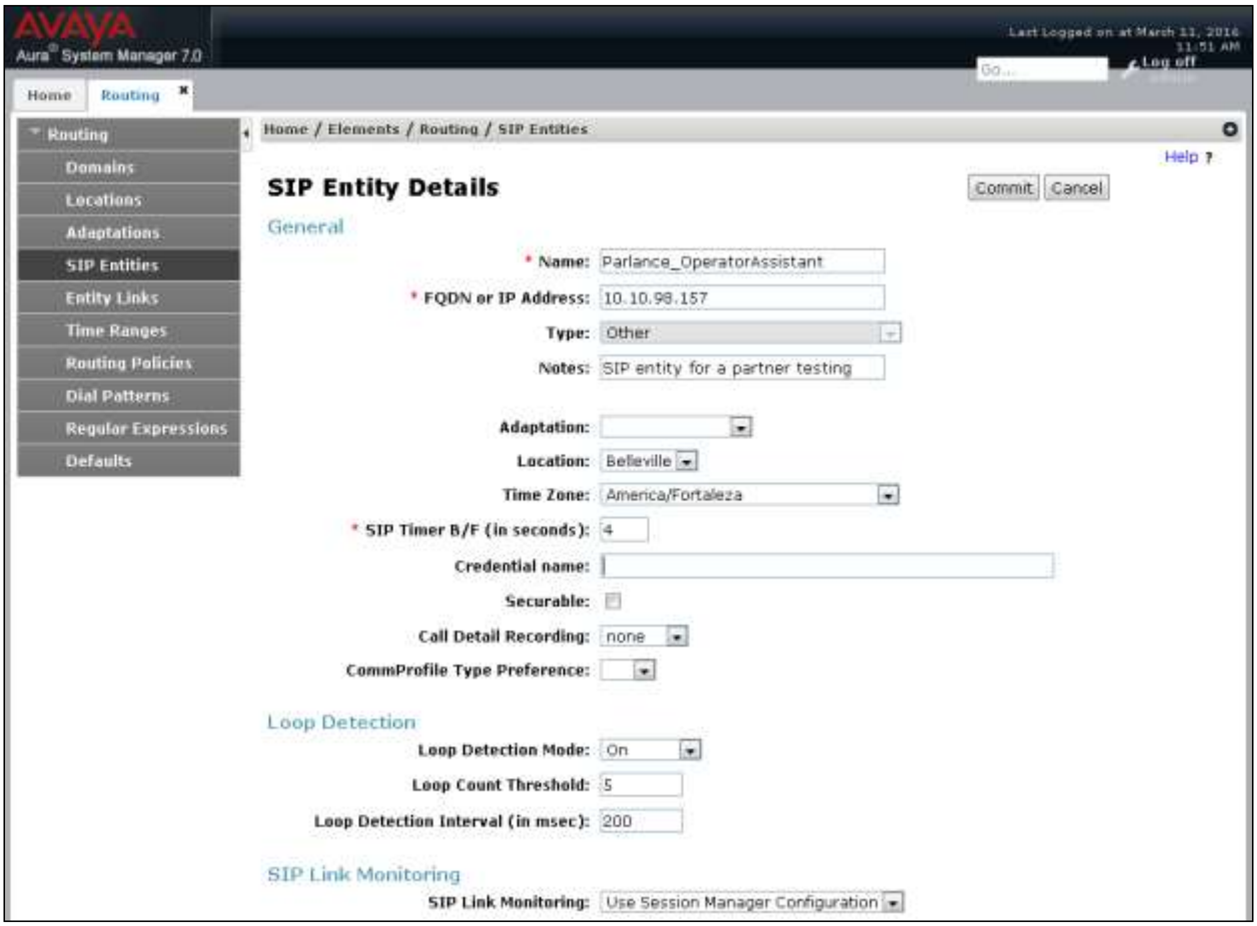
Add two new SIP entities, one for Operator Assistant and one for the new SIP trunks with Communication Manager.

6.4.1. SIP Entity for Operator Assistant

Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Operator Assistant.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

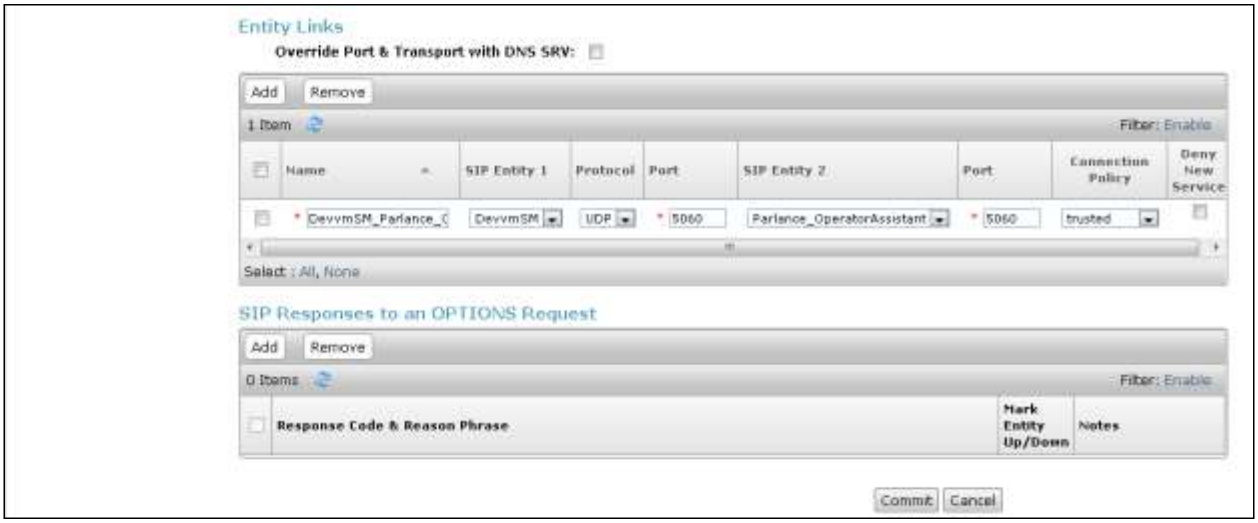
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the Operator Assistant server.
- **Type:** "Other"
- **Notes:** Any desired notes.
- **Location:** Select the Operator Assistant location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.



Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "DevvmSM".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The Operator Assistant entity name from this section.
- **Port:** "5060"
- **Connection Policy:** "trusted"

Note that Operator Assistant can only support UDP protocol.

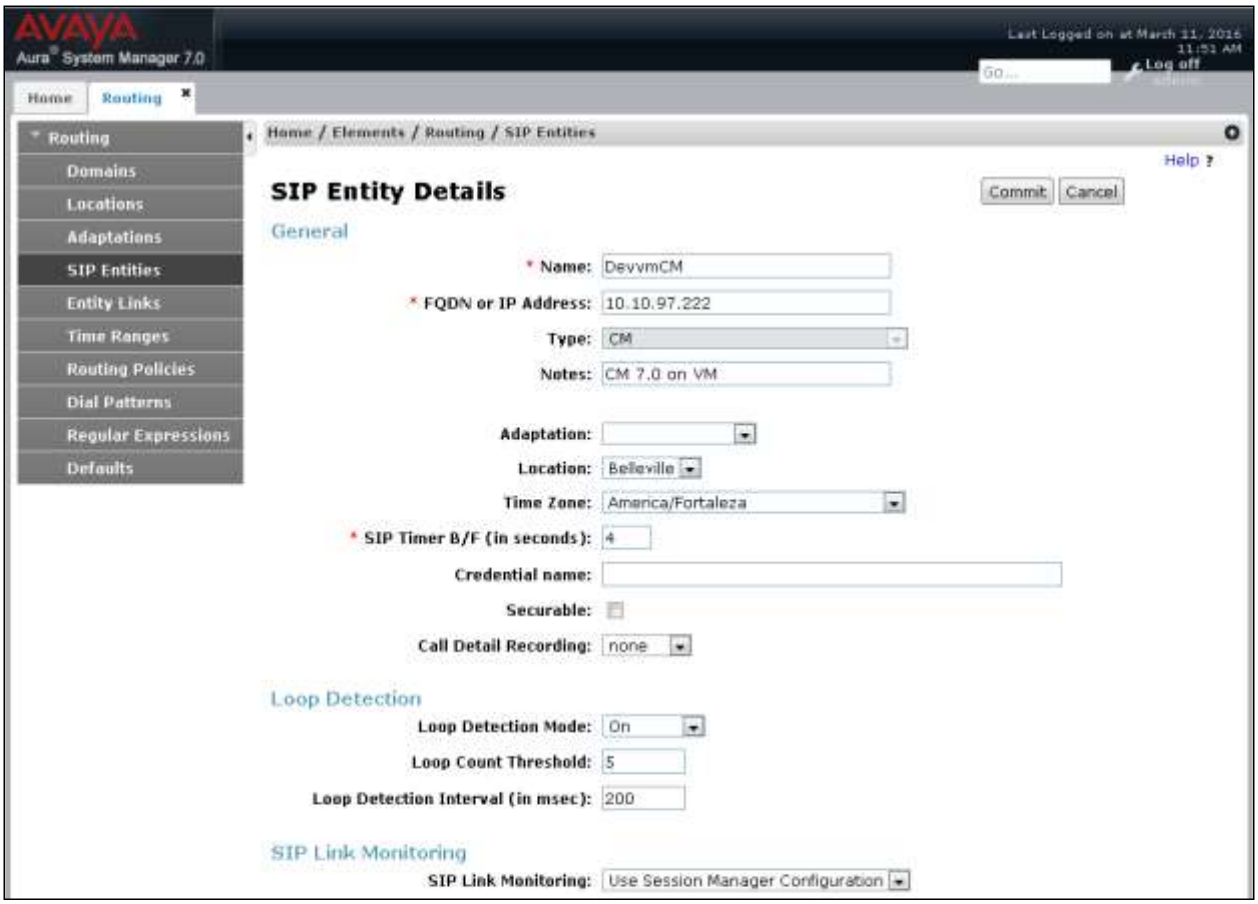


6.4.2. SIP Entity for Communication Manager

Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that this SIP entity is used for integration with Operator Assistant.

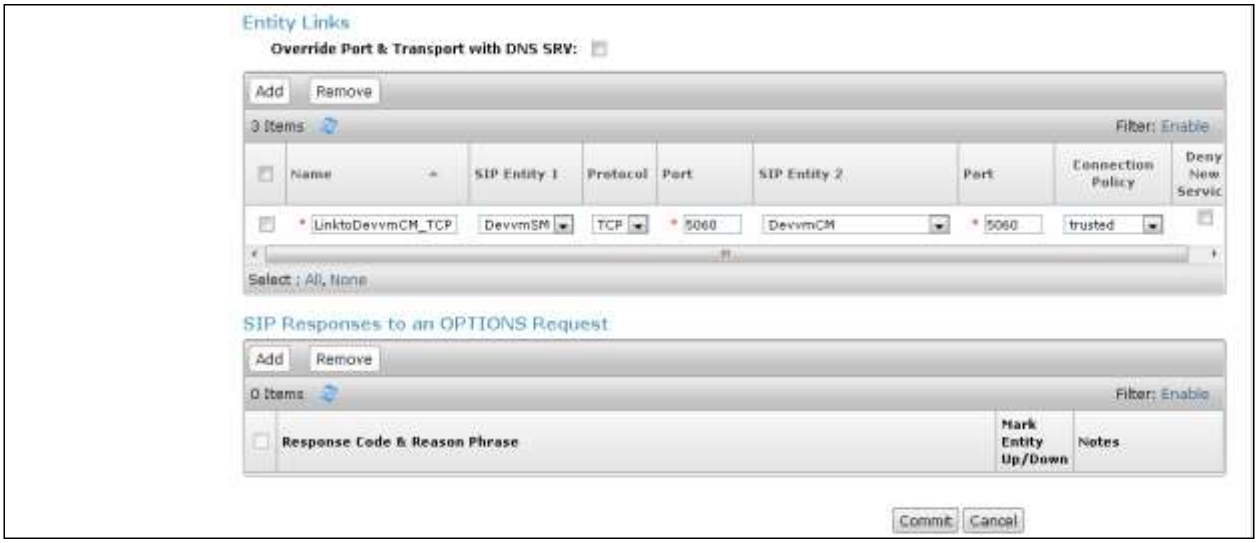
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing CLAN or the processor interface.
- **Type:** "CM"
- **Notes:** Any desired notes.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.



Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "DevvmSM".
- **Protocol:** The signaling group transport method from **Section 5.4**.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **Connection Policy:** "trusted"



6.5. Administer Routing Policies

Add two new routing policies, one for Operator Assistant and one for the new SIP trunks with Communication Manager.

6.5.1. Routing Policy for Operator Assistant

Select **Routing → Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Operator Assistant.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Operator Assistant entity name from **Section 6.4.1**. The screen below shows the result of the selection.

6.5.2. Routing Policy for Communication Manager

Select **Routing → Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Communication Manager.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 6.4.2**. The screen below shows the result of the selection.

6.6. Administer Dial Patterns

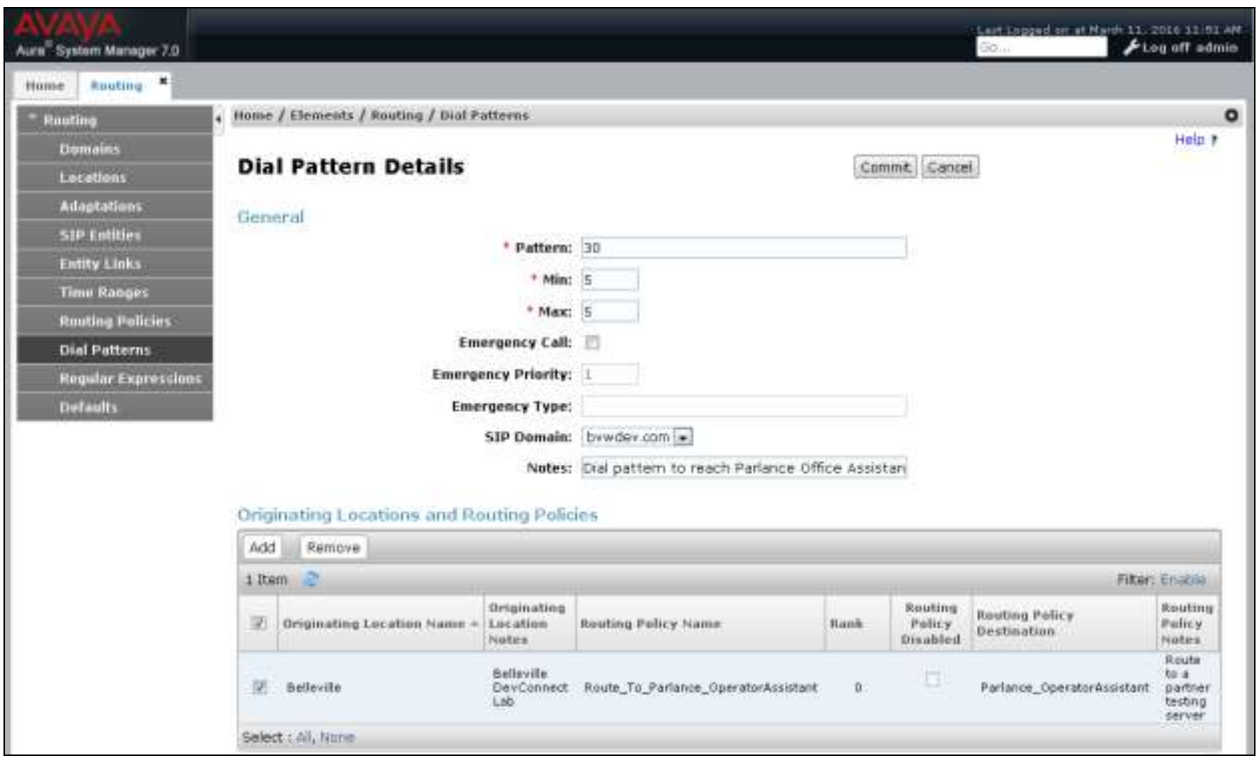
Add a new dial pattern for Operator Assistant, and update existing dial patterns for Communication Manager.

6.6.1. Dial Pattern for Operator Assistant

Select **Routing → Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Operator Assistant. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case "30".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The signaling group domain name from **Section 5.4**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Operator Assistant. In the compliance testing, the entry allowed for call originations from all Communication Manager endpoints in locations "Belleville". The Operator Assistant routing policy from **Section 6.5.1** was selected as shown below.

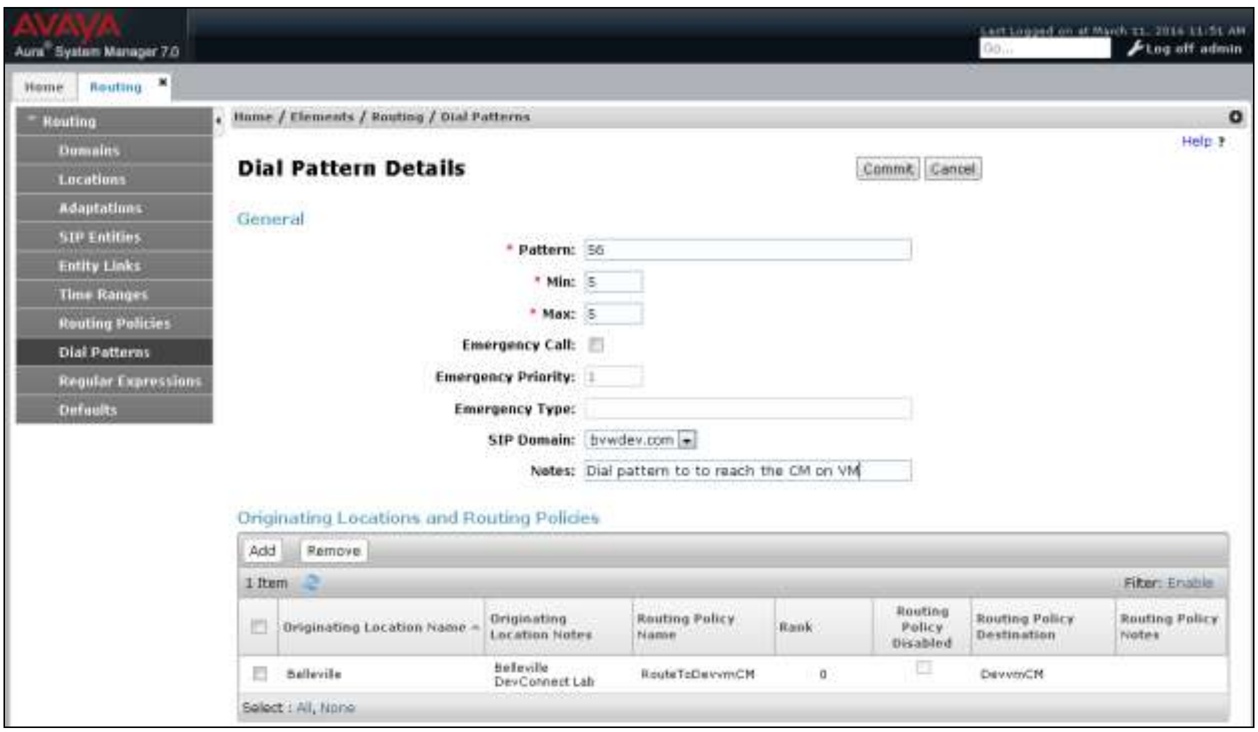


6.6.2. Dial Pattern for Communication Manager

Select **Routing → Dial Patterns** from the left pane, and click on the first existing dial pattern for Communication Manager in the subsequent screen, in this case dial pattern "56" (not shown). The **Dial Pattern Details** screen is displayed.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy as necessary for calls from Operator Assistant. In the compliance testing, the new policy allowed for call origination from the Operator Assistant location from **Section 6.2**, and the Communication Manager routing policy from **Section 6.5.2** was selected as shown below. Retain the default values in the remaining fields.

Follow the procedures in this section to make similar changes to the applicable Communication Manager dial pattern to reach the PSTN. In the compliance testing, Operator Assistant will add the prefix "9" for outbound calls to the PSTN, and therefore the existing dial pattern for "9" was also changed (not shown below).



7. Configure Parlance Operator Assistant

The Parlance Operator Assistant will be provisioned completely by Parlance engineers based on site requirements and therefore no configuration details will be provided in these application notes.

To obtain information on Operator Assistant configuration, refer to **Section 2.3**.

8. Verification Steps

This section provides tests that can be performed to verify proper configuration of Communication Manager and Session Manager.

8.1. Verify Avaya Aura® Communication Manager

From the SAT interface, verify the status of the SIP trunk groups by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 5.3**. Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 1                                                         Page   1

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0001/001 T00001   in-service/idle    no
0001/002 T00002   in-service/idle    no
0001/003 T00003   in-service/idle    no
0001/004 T00004   in-service/idle    no
0001/005 T00005   in-service/idle    no
0001/006 T00006   in-service/idle    no
0001/007 T00007   in-service/idle    no
0001/008 T00008   in-service/idle    no
0001/009 T00009   in-service/idle    no
0001/010 T00010   in-service/idle    no
0001/011 T00011   in-service/idle    no
0001/012 T00012   in-service/idle    no
0001/013 T00013   in-service/idle    no
0001/014 T00014   in-service/idle    no
```

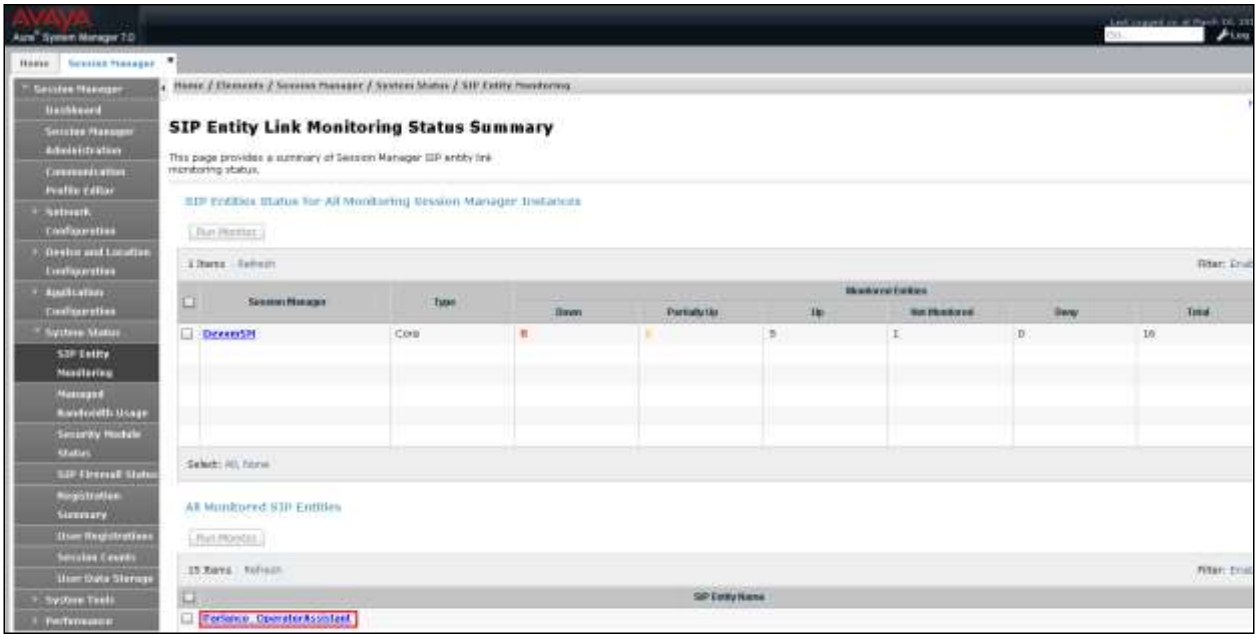
Verify the status of the SIP signaling groups by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 5.4**. Verify that the **Group State** is "in-service", as shown below.

```
status signaling-group 1
                         STATUS SIGNALING GROUP

        Group ID: 1
      Group Type: sip

     Group State: in-service
```

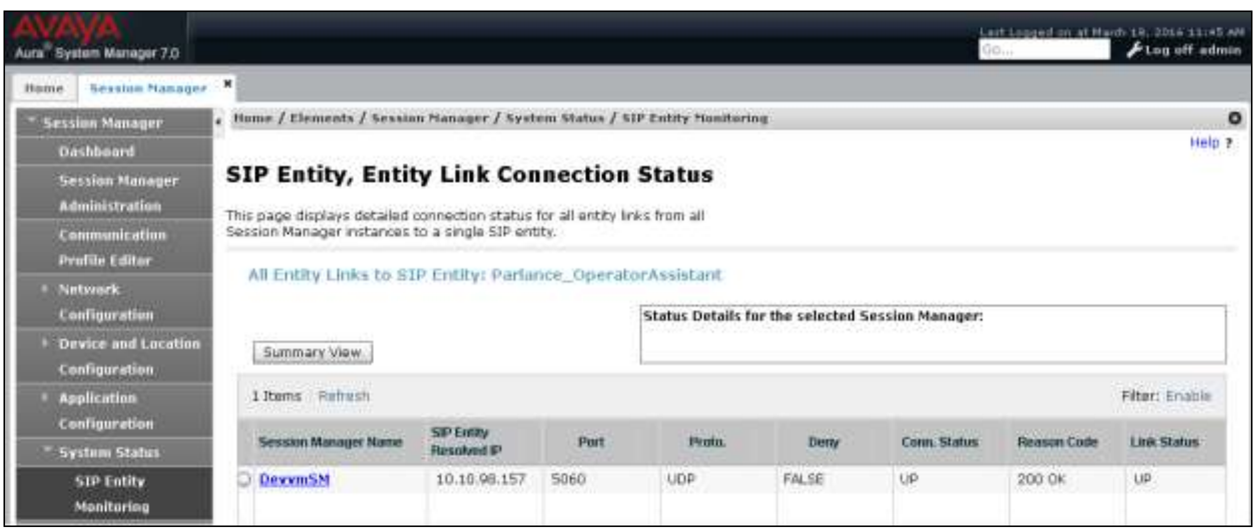
8.2. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements → Session Manager** to display the **Session Manager Dashboard** screen (not shown).

Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the Operator Assistant entity name from **Section 6.4.1**.



The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are "UP", as shown below.



9. Conclusion

These Application Notes describe the configuration steps required for Parlance Operator Assistant to successfully interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

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

1. *Implementing Avaya Aura® Session Manager* Document ID 03-603473.
2. *Administering Avaya Aura® Session Manager*, Doc ID 03-603324.
3. *Deploying Avaya Aura® System Manager*, Release 7.0.
4. *Administering Avaya Aura® System Manager for Release 7.0*, Release 7.0.
5. *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager*.
6. *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 7.7.
7. *Administering Avaya Aura® Communication Manager*, Release 7.0, 03-300509.
8. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 7.0, 555-245-205.

To obtain information on documents related to Parlance Operator Assistant, refer to **Section 2.3**.

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.