Avaya Solution & Interoperability Test Lab

Application Notes for Smart Action Intelligent Virtual Assistant with Avaya Session Border Controller for Enterprise, Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Smart Action Intelligent Virtual Assistant to interoperate with Avaya Session Border Controller for Enterprise, Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks. Intelligent Virtual Assistant is a cloud-based intelligent IVR. In the compliance testing, Intelligent Virtual Assistant used SIP trunks to Avaya Session Border Controller for Enterprise to support inbound and outbound IVR applications.

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 for Smart Action Intelligent Virtual Assistant (Intelligent Virtual Assistant) to interoperate with Avaya Session Border Controller for Enterprise (Avaya SBCE), Avaya Aura® Communication Manager (Communication Manager) and Avaya Aura® Session Manager (Session Manager) using SIP trunks via UDP. Intelligent Virtual Assistant is a cloud-based IVR. In the compliance testing, Intelligent Virtual Assistant used SIP trunks to Avaya Session Border Controller for Enterprise to IVR functionality. Calls to and from Avaya Aura® Environment to Intelligent Virtual Assistant IVR were routed via Avaya SBCE.

The Intelligent Virtual Assistant used during the testing was deployed on a cloud.

2. General Test Approach and Test Results

The feature test cases were performed manually. The Intelligent Virtual IVR was tested by manually placing calls from users on the PSTN and on Communication Manager to the Intelligent Virtual Assistant IVR. The associated Intelligent Virtual IVR played greeting announcements and collected DTMF input from the caller to decide on the feature to provide, such as transfer to internal or external destinations. Intelligent Virtual Assistant outbound calling to PSTN and Communication Manager were also tested.

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

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes 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.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution and the third party solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of the third party solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

Readers should be aware that network behaviors (e.g. jitter, packet loss, delay, speed, etc.) can vary significantly from one location to another, and may affect the reliability or performance of the overall solution. Different network elements (e.g. session border controllers, soft switches, firewalls, NAT appliances, etc.) can also affect how the solution performs.

If a customer is considering implementation of this solution in a cloud environment, the customer should evaluate and discuss the network characteristics with their cloud service provider and network organizations, and evaluate if the solution is viable to be deployed in the cloud.

The network characteristics required to support this solution are outside the scope of these Application Notes. Readers should consult the appropriate Avaya and third party documentation for the product network requirements. Avaya makes no guarantee that this solution will work in all potential deployment configurations

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included G.711MU, codec negotiation, media shuffling, session refresh, hold/reconnect, inbound DTMF, invalid number, busy destination, and outgoing call screening.

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

2.2. Test Results

All test cases were executed and passed.

2.3. Support

Technical support on Intelligent Virtual Assistant can be obtained through the following:

- **Phone:** 310-776-9200 option 2
- **Email:** support@smartaction.ai
- **Web:** https://www.smartaction.ai/support/

3. Reference Configuration

As shown in **Figure 1**, SIP trunks were used between Intelligent Virtual Assistant and Session Manager (via Avaya SBCE), and the applicable domain name used was "avaya.com". 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, Session Manager and Avaya SBCE is not the focus of these Application Notes and will not be described.

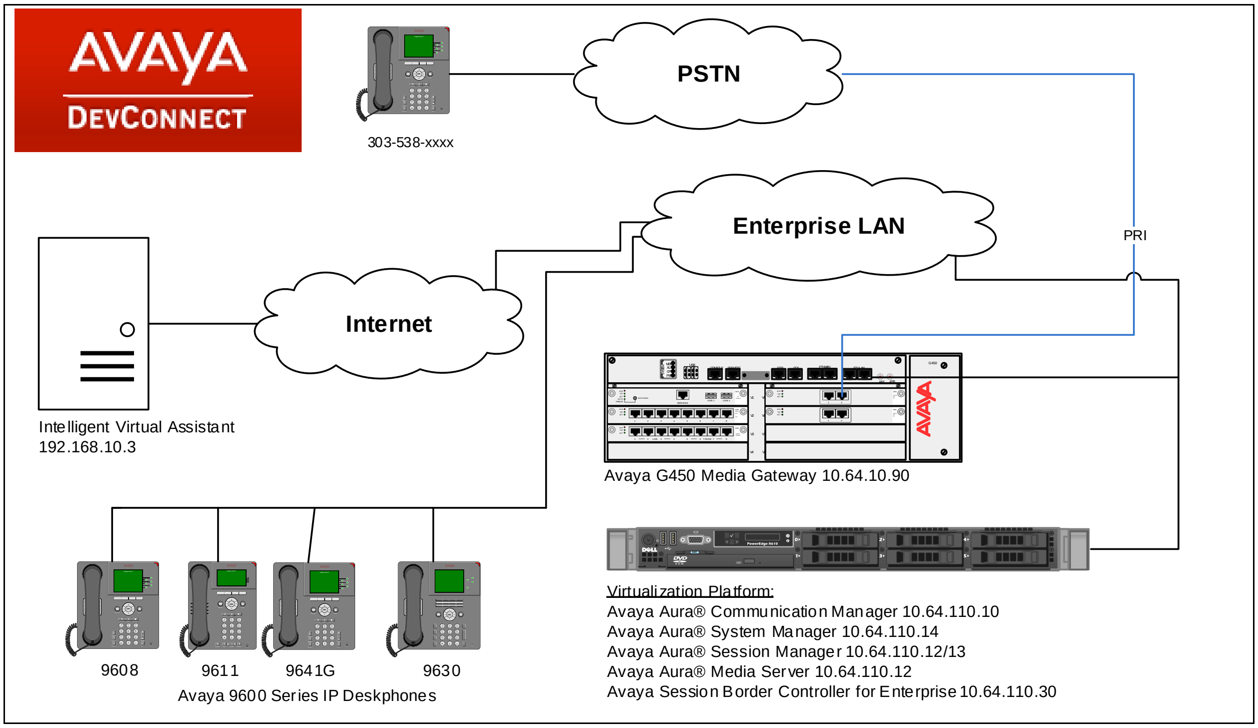


Figure 1: Intelligent Virtual Assistant with Avaya Aura® Environment

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 with Avaya G450 Media Gateway	7.1.2 37.19.0
Avaya Aura® Session Manager	7.1.2
Avaya Aura® System Manager	7.1.2
Avaya Session Border Controller for Enterprise	7.2.2.0
Avaya 96x0 IP Deskphone (H.323)	3.2.8
Avaya 96x1 IP Deskphone (H.323)	6.6.6
Avaya 96x0 IP Deskphone (SIP)	2.6.17
Avaya 96x1G IP Deskphone (SIP)	7.1.1.0
Smart Action Intelligent Virtual Assistant	10.3

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 uniform dial plan
- Administer AAR analysis
- Administer PSTN trunk group
- Administer tandem calling party number

In the compliance testing, a separate set of codec set, network region, trunk group, and signaling group were used for integration with Intelligent Virtual 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: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 36000 0
                   Maximum Video Capable IP Softphones: 18000 3
                       Maximum Administered SIP Trunks: 12000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   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 [1] 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? all
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n




             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? 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". This trunk group is used between Communication Manager and Session Manager. 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: asm                         COR: 1        TN: 1        TAC: 101
   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:
                                                     Number of Members:
```

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

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



   Suppress # Outpulsing? n  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
               Send UCID? y
```

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:** "tls"
- **Near-end Node Name:** An existing C-LAN node name or "procr" in this case.
- **Far-end Node Name:** The existing node name for Session Manager.
- **Near-end Listen Port:** An available port for integration with Communication Manager.
- **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 Intelligent Virtual Assistant.
- **Far-end Domain:** The applicable domain name for the network. The empty Far-end Domain indicates "any" domain.

```
add signaling-group 1                                         Page   1 of   2
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        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: asm
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain:
                                             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): 65                    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 "10".

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

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: asm                        COR: 1        TN: 1        TAC: 101
   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: 1
                                                    Number of Members: 10
```

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**, if desired. 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 Intelligent Virtual Assistant.

```
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: 3329
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
```

5.7. 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 Intelligent Virtual 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:
 6:
 7:
```

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 Intelligent Virtual Assistant via Session Manager, 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.
- **Numbering Format:** "lev0-pvt"

```
change route-pattern 1                                                Page   1 of   3
                    Pattern Number: 1      Pattern Name:
    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                                                                    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
 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: y y y y y n  n            rest                                         none
```

5.9. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to Intelligent Virtual 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 5 and routed to any trunk group 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  5                                              5     Total Administered: 1
                                                            Maximum Entries: 540
```

5.10. Administer AAR Analysis

Use the "change aar analysis 48600" command, and add an entry to specify how to route calls to 48600. In the example shown below, calls with digits 48600 will be routed as an aar call type using route pattern "1" from **Section 0**.

```
change aar analysis 511                                             Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 0

          Dialed             Total     Route     Call    Node  ANI
          String           Min  Max  Pattern     Type    Num   Reqd
    48600                  5    5      1         aar           n
```

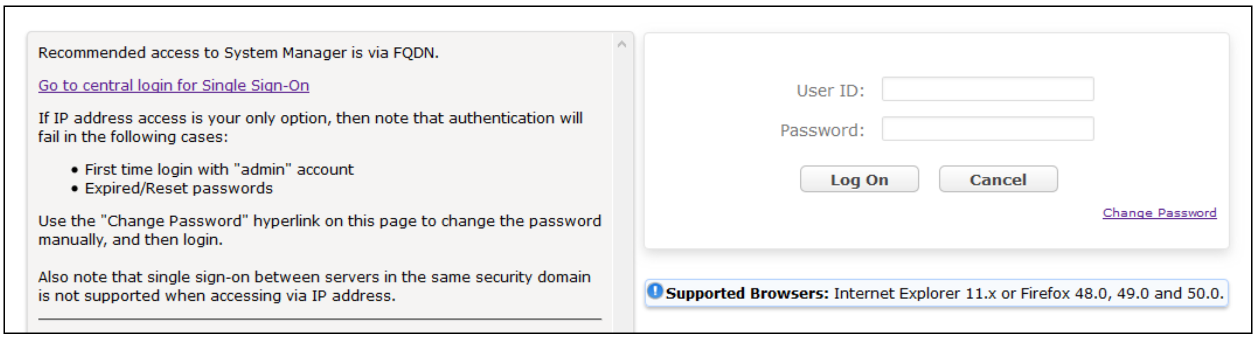
6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer 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 SIP Entities

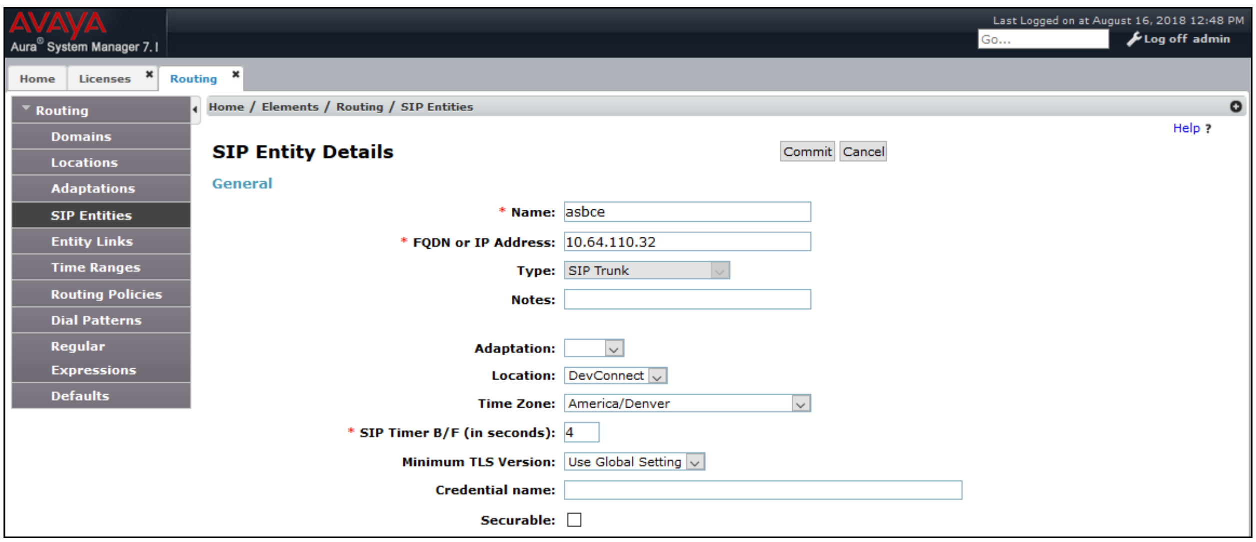
Add two new SIP entities, one for Avaya SBCE and another one for SIP trunks with Communication Manager.

6.2.1. SIP Entity for Avaya Session Border Controller for Enterprise

Select **Routing → SIP Entities** (not shown) from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Intelligent Virtual 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 Avaya SBCE.
- **Type:** "SIP Trunk"
- **Notes:** Any desired notes.
- **Location:** Select the location name.
- **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.

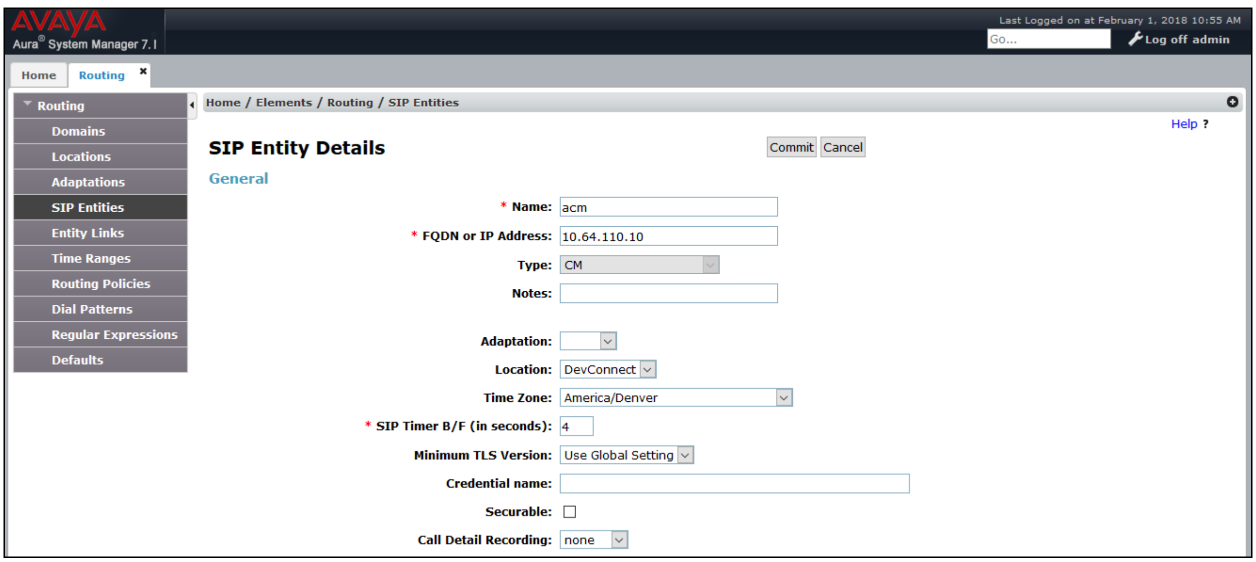
- **SIP Entity 1:** The Session Manager entity name, in this case "asm".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The Avaya SBCE entity name from this section.
- **Port:** "5060"

6.2.2. 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 Intelligent Virtual Assistant.

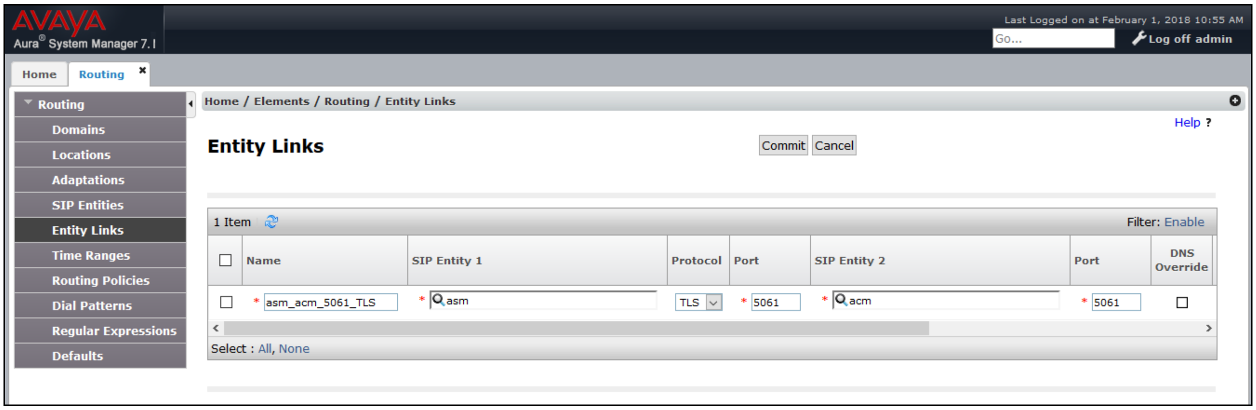
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **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.

- **SIP Entity 1:** The Session Manager entity name, in this case "asm".
- **Protocol:** The signaling group transport method from **Section 5.4**.
- **Port:** The signaling group far-end listen port number from **Section 5.4**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group near-end listen port number from **Section 5.4**.



6.3. Administer Routing Policies

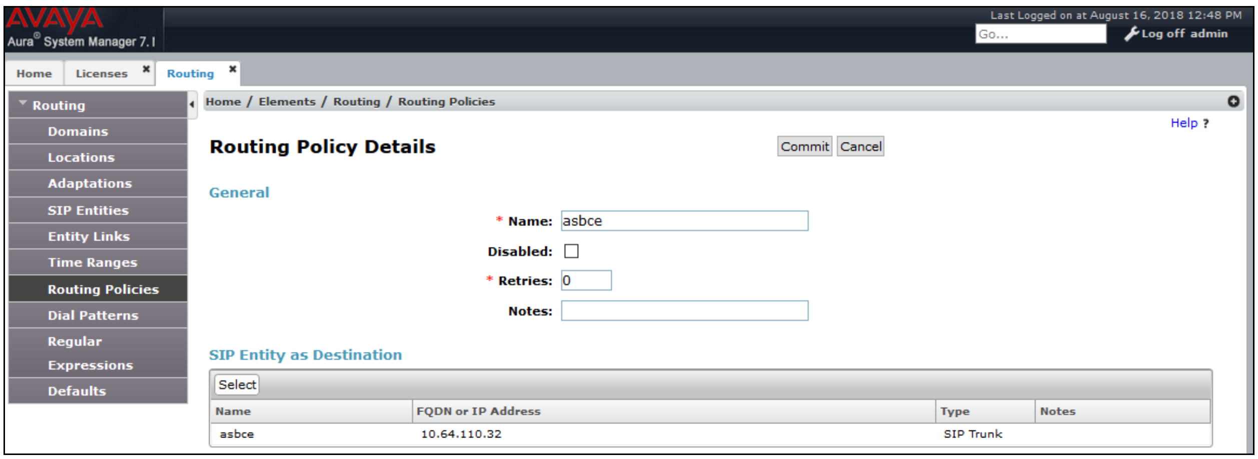
Add two new routing policies, one for Avaya SBCE and another one for Communication Manager.

6.3.1. Routing Policy for Avaya Session Border Controller for Enterprise

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

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 Avaya SBCE entity name from **Section 6.2.1**. The screen below shows the result of the selection.

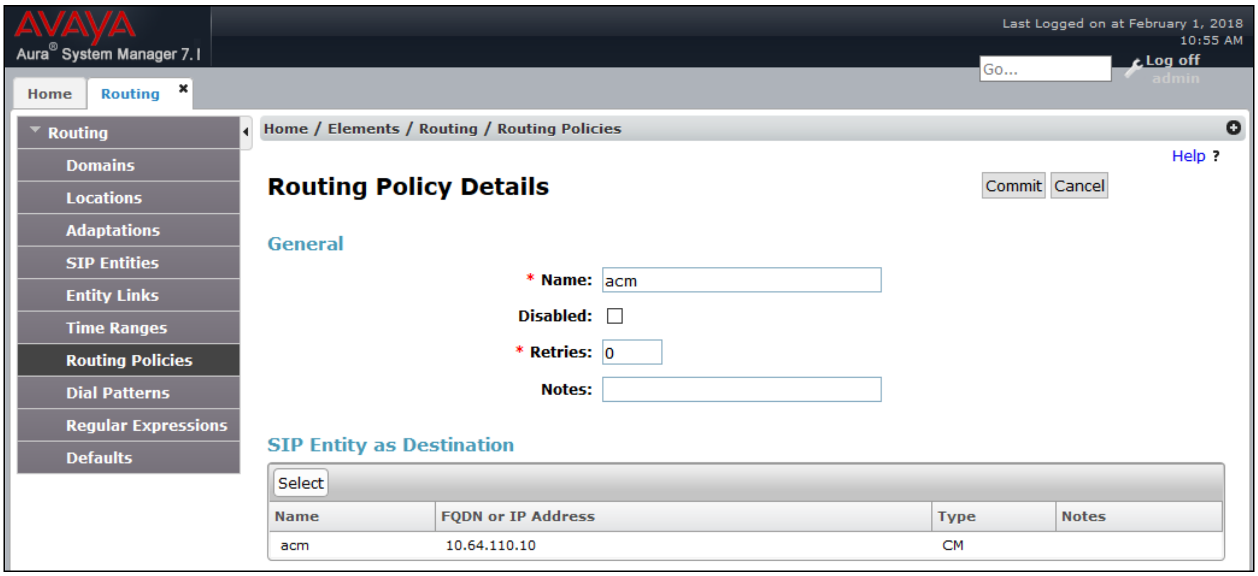


6.3.2. Routing Policy for Communication Manager

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 6.2.2**. The screen below shows the result of the selection.



6.4. Administer Dial Patterns

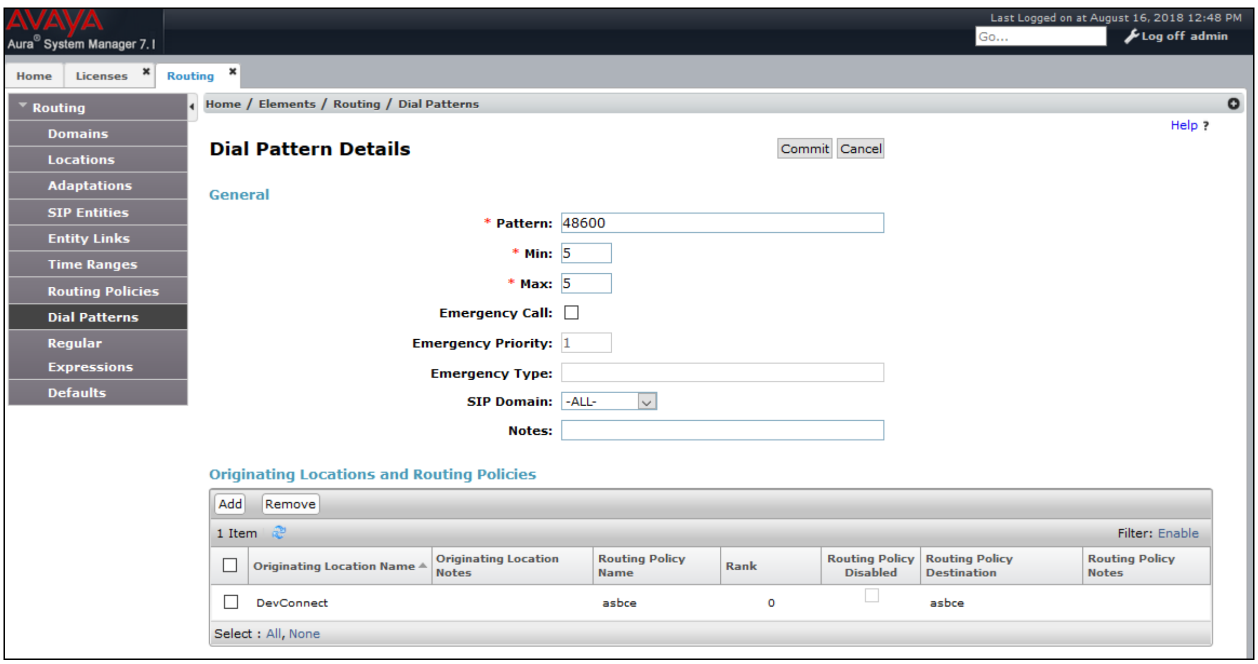
Add a new dial pattern for Intelligent Virtual Assistant, and update existing dial patterns for Communication Manager.

6.4.1. Dial Pattern for Intelligent Virtual Assistant

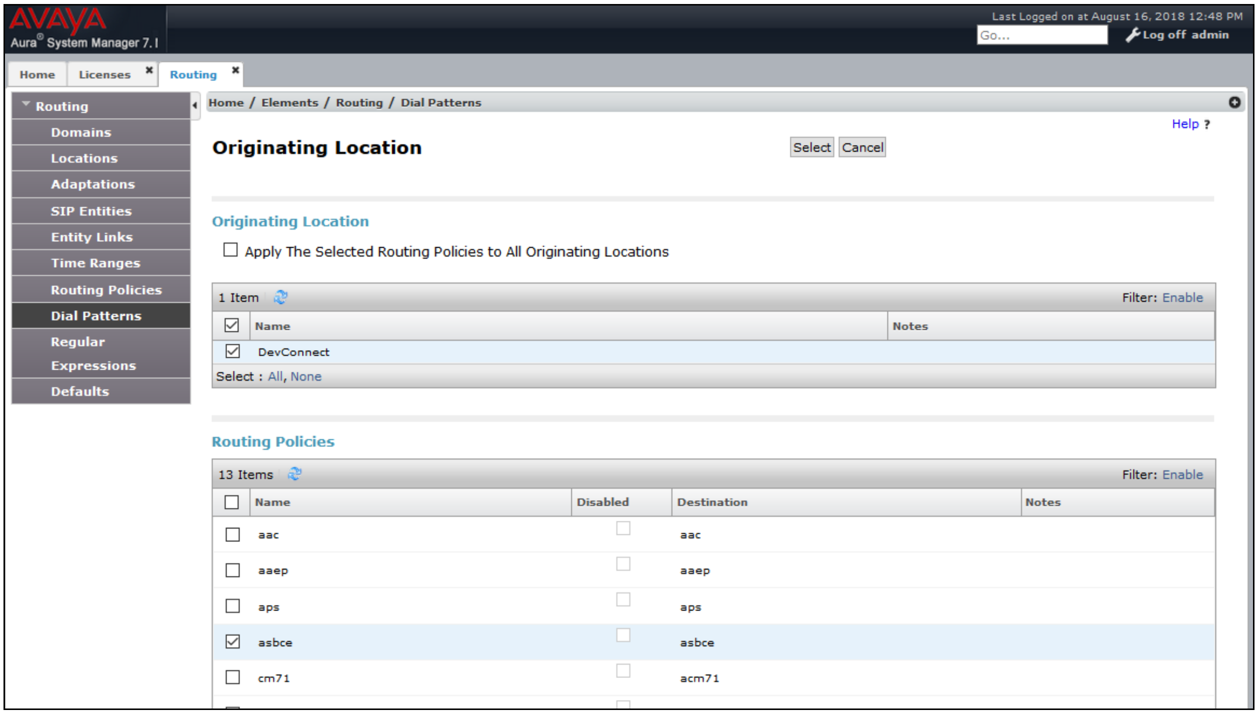
Select **Routing → Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Intelligent Virtual Assistant via Avaya SBCE. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for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

- **Pattern:** A dial pattern to match, in this case "48600".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and select the routing policy for reaching Intelligent Virtual Assistant.

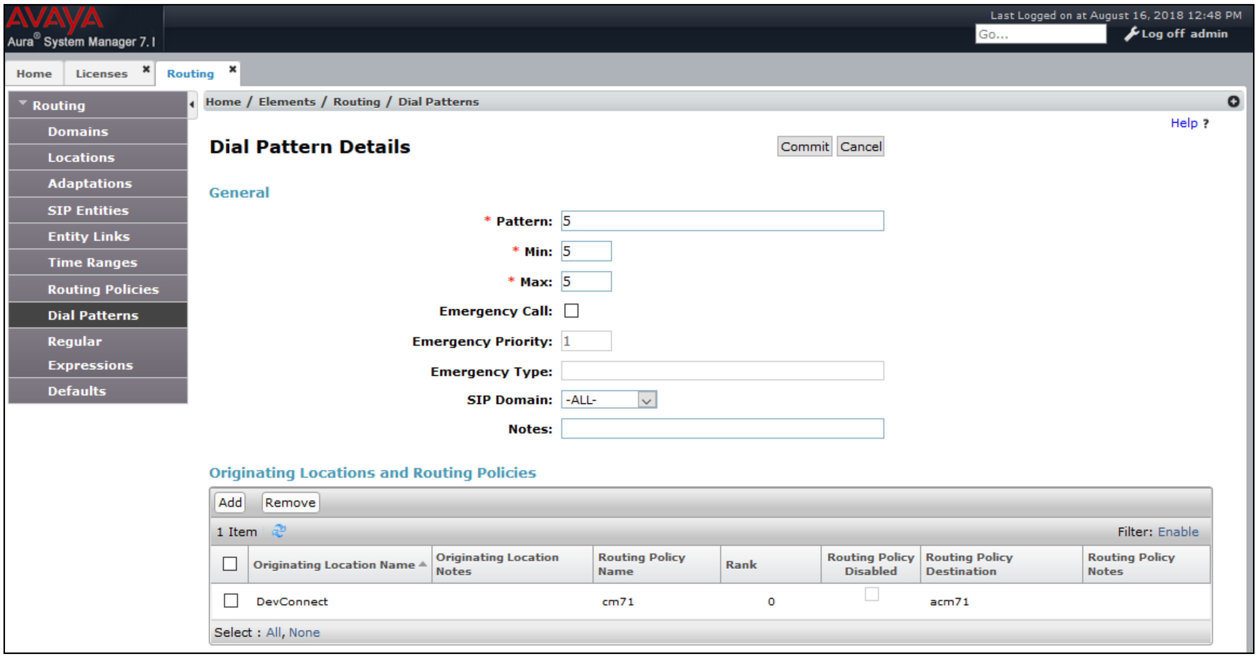


In the compliance testing, the policy allowed for call origination from “DevConnect”, and the Avaya SBCE routing policy from **Section 6.3.1** was selected as shown below.



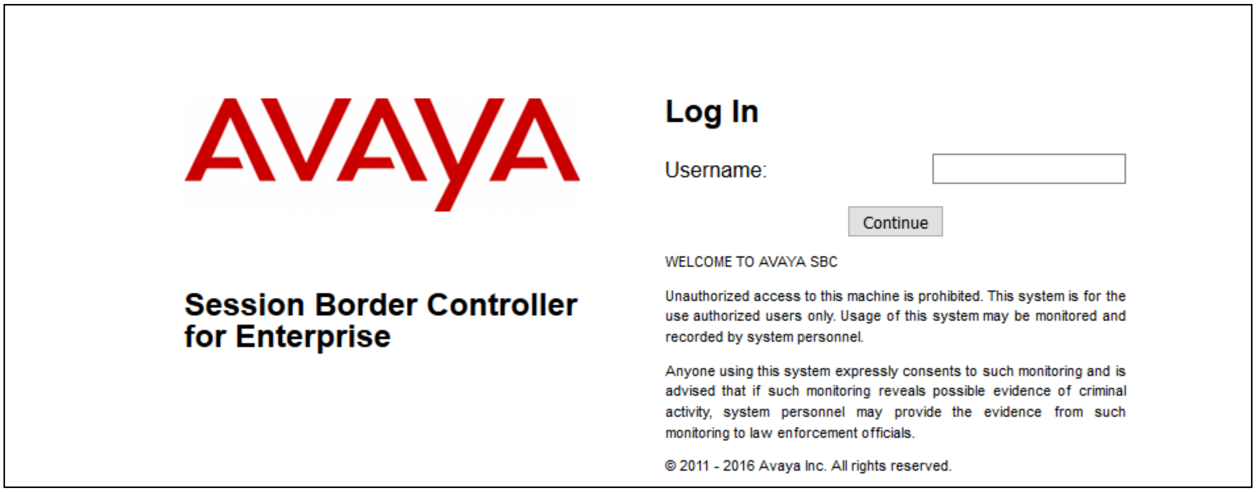
6.4.2. Dial Pattern for Communication Manager

Similar steps were followed as previous section to add dial patterns for Communication Manager.



7. Configure Avaya Session Border Controller for Enterprise

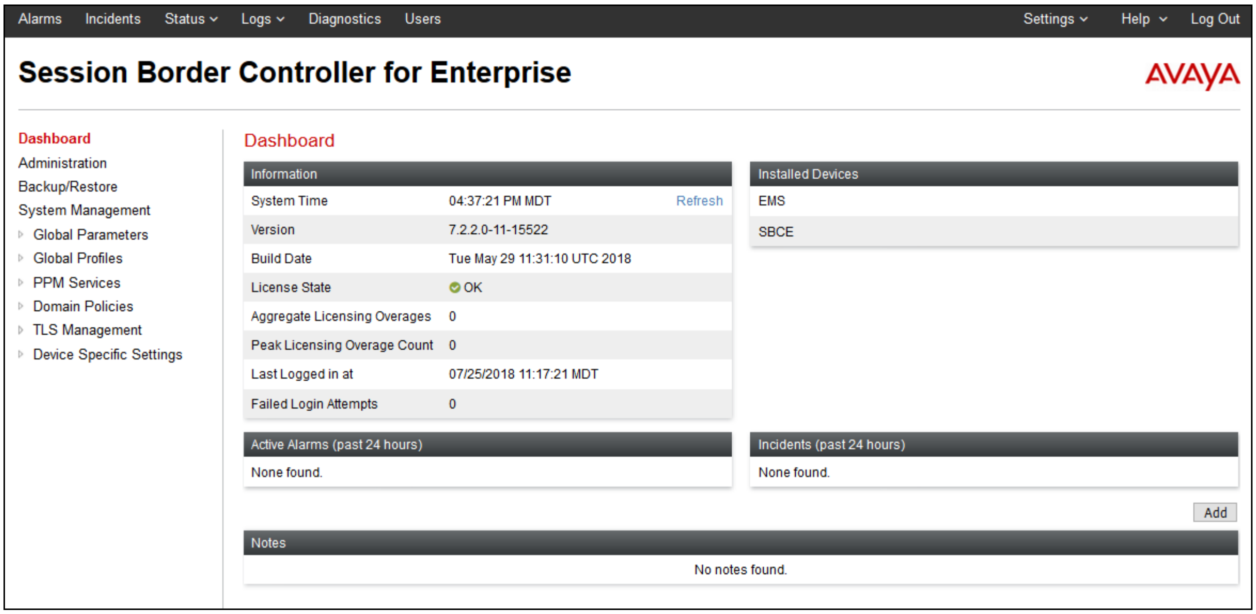
This section describes the configuration of the Avaya Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signaling to provide an interface to Intelligent Virtual Assistant.



7.1. Access Avaya Session Border Controller for Enterprise

Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. A log in screen is presented. Log in using the appropriate username and password.

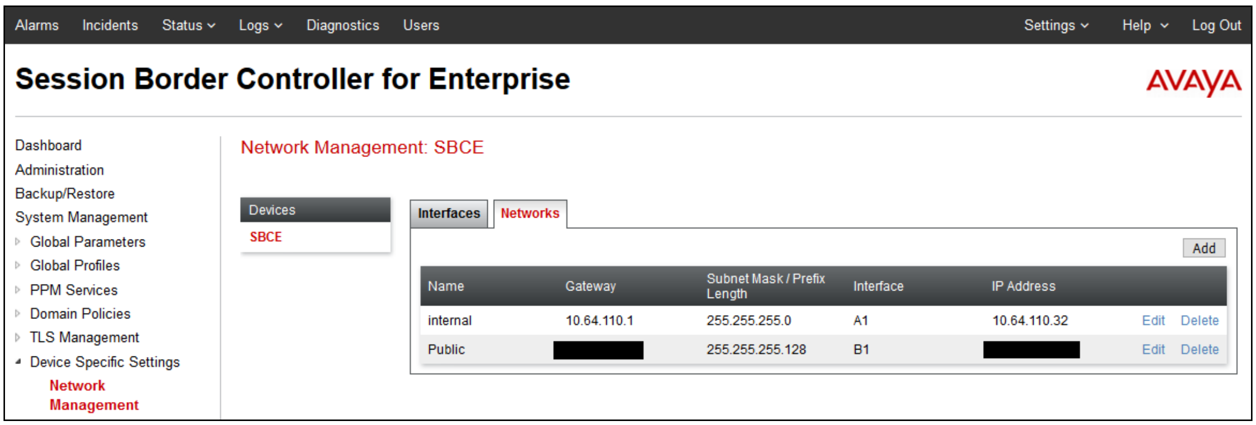
Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



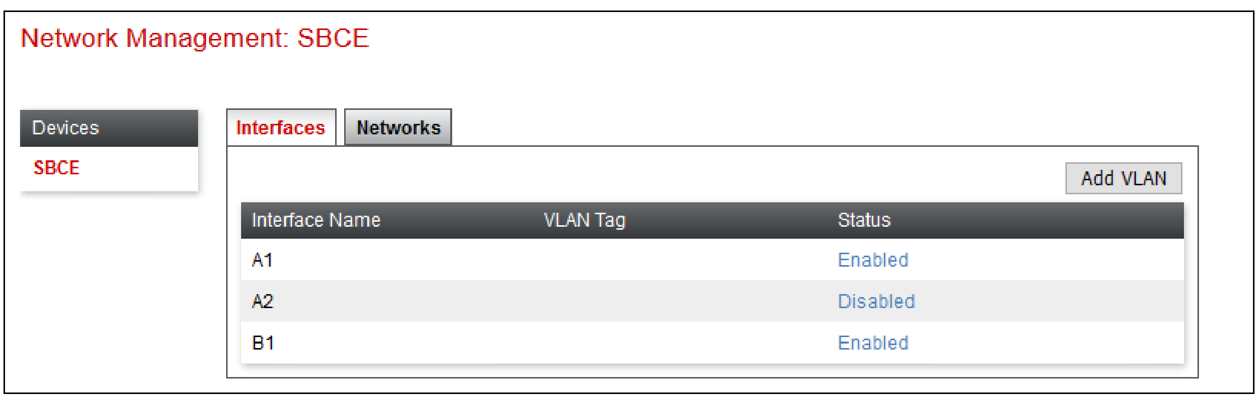
7.2. Define Network Management

Network information is required on the Avaya SBCE to allocate IP addresses and subnet masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one physical interface assigned.

To define the network information, navigate to **Device Specific Settings → Network Management** in the main menu on the left hand side and click on **Add**. The following interfaces were added for Session Manager and Public Internet. For security reasons the IP Address of external interface has been masked.



Select the **Interface Configuration** tab and click on the **Status** of the physical interface to toggle it. A status of **Disabled** will be changed to **Enabled**.



Note: to ensure that the Avaya SBCE uses the interfaces defined, the Application must be restarted.

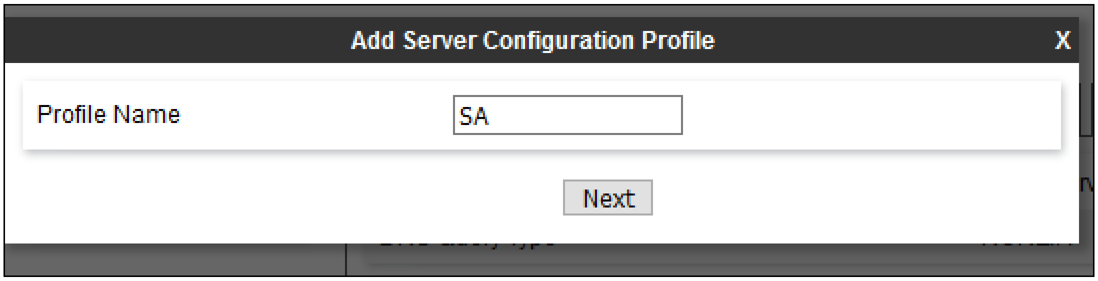
- Click on **System Management** in the main menu (not shown).
- Select **Restart Application** indicated by an icon in the status bar (not shown).

A status box will appear (not shown) that will indicate when the application has restarted.

7.3. Define Servers

A server definition is required for each server connected to the Avaya SBCE. In this case, the Intelligent Virtual Assistant as a Trunk Server. The server here is added with a name of **SA.**

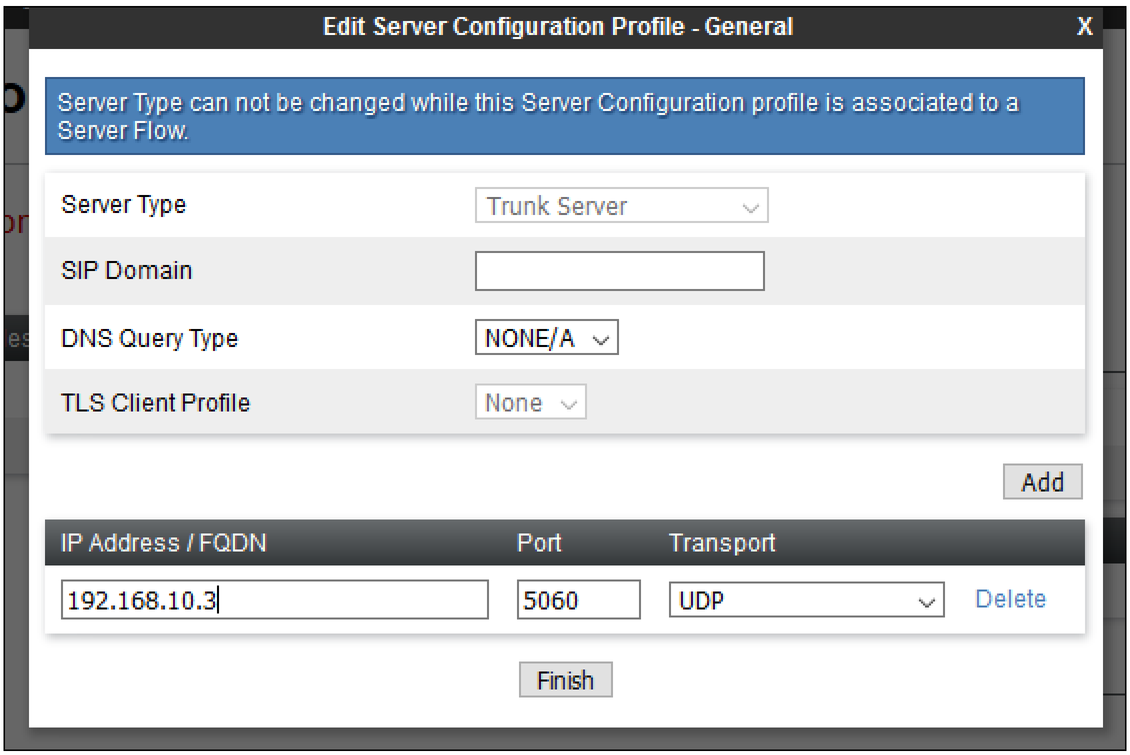
To define the server for Intelligent Virtual Assistant, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the pop-up menu.



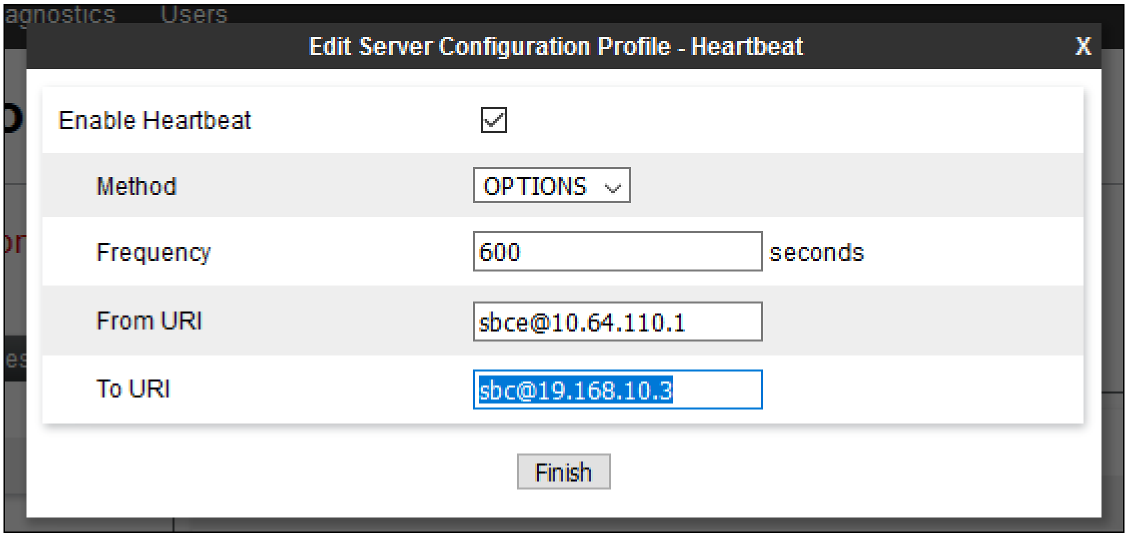
Click on **Next** and enter details in the dialogue box.

- In the **Server Type** drop down menu, select **Trunk Server**.
- Click on **Add** to enter an IP address
- In the **IP Addresses / FQDN** box, type the IP Address of Intelligent Virtual Assistant.
- In the **Port** box, enter the port to be used.
- In the **Transport** drop down menu, select **UDP**.
- Click on **Next**.

Note that the IP Address shown below is not the actual public IP Address that was used. It has been changed with a private IP Address for security reasons.



Click on **Next** and configure as follows.

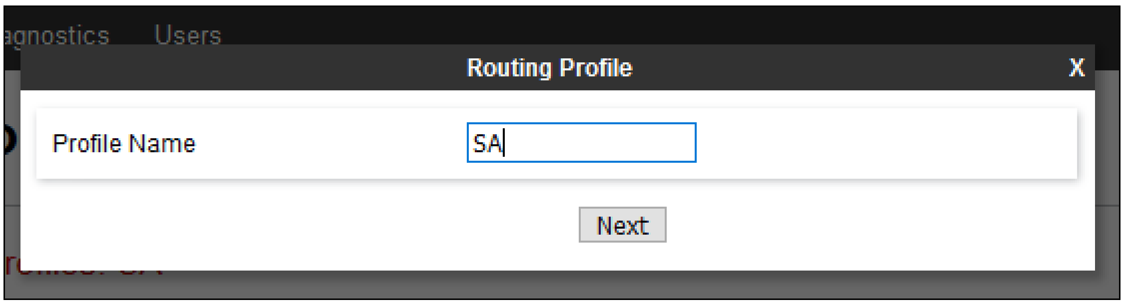


Select **Next** and then **Finish** (not shown).

7.4. Define Routing

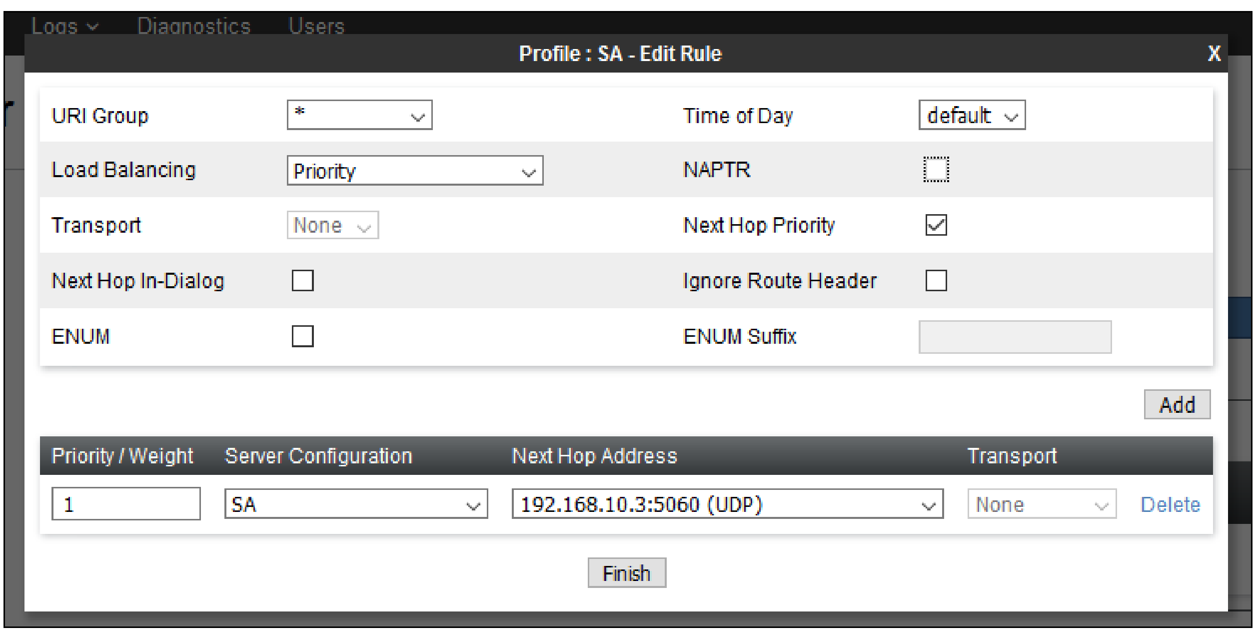
Routing information is required for routing calls to Intelligent Virtual Assistant. The IP addresses and ports defined here will be used as the destination addresses for signaling.

To define routing to the Intelligent Virtual Assistant SIP Trunk, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the dialogue box.



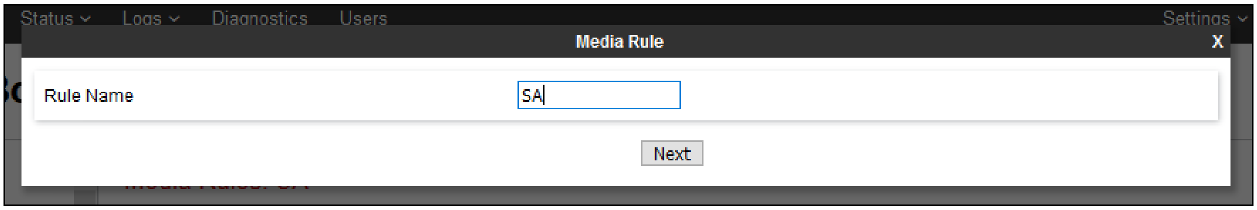
Click on **Next** and enter details for the Routing Profile:

- Click on **Add** to specify the IP address for the Intelligent Virtual Assistant SIP trunk.
- Assign a priority in the **Priority / Weight** field, during testing a value of **1** was used.
- Select the Server Configuration defined in **Section 7.3** in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field
- Click **Finish**.

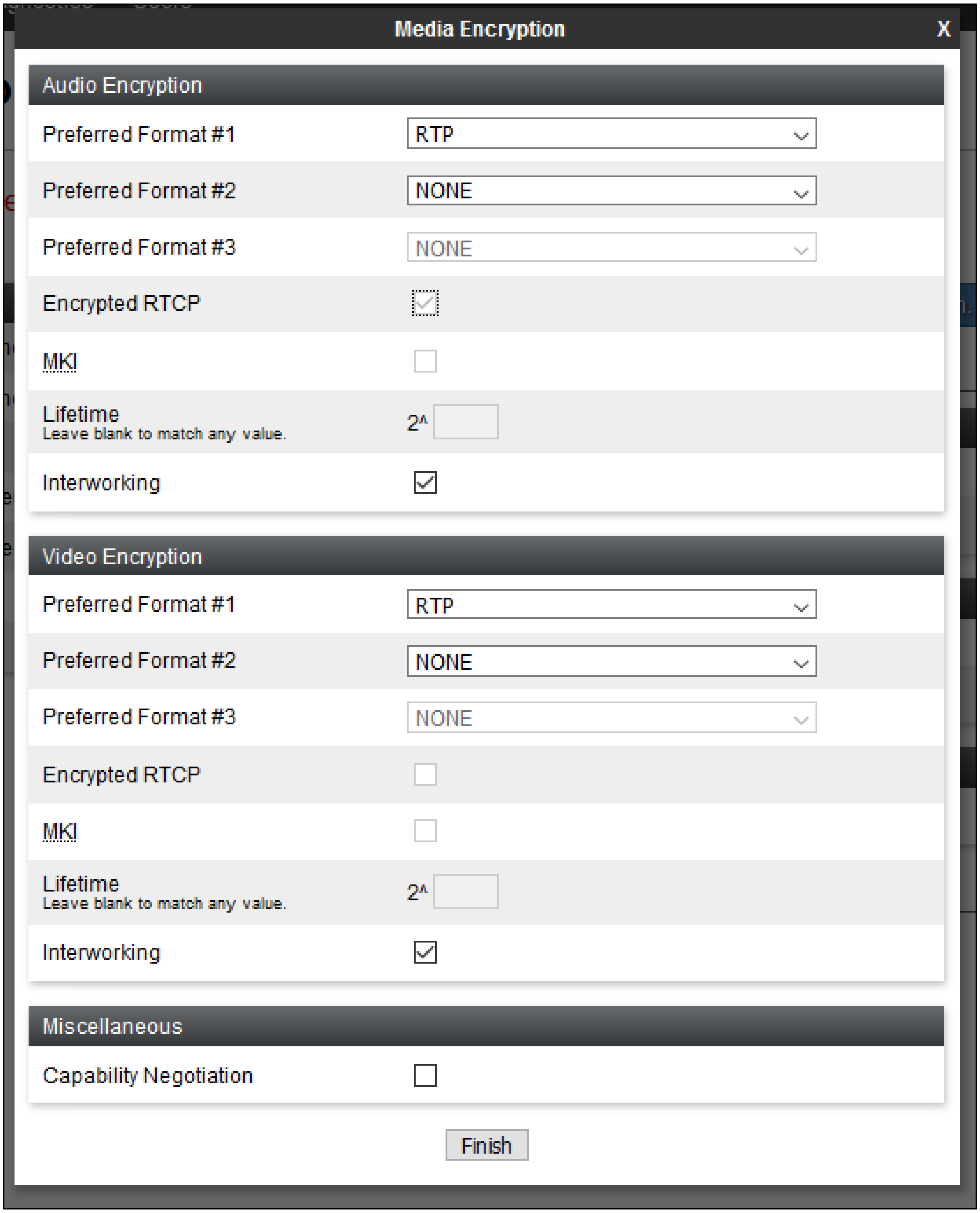


7.5. Define Media Rules

Audio formats need to be specified for Intelligent Virtual Assistant. To create a Media Rule for Intelligent Virtual Assistant, navigate to **Domain Policies → Media Rules.** Click on **Add** and enter an appropriate name in the pop-up menu and select **Next**.

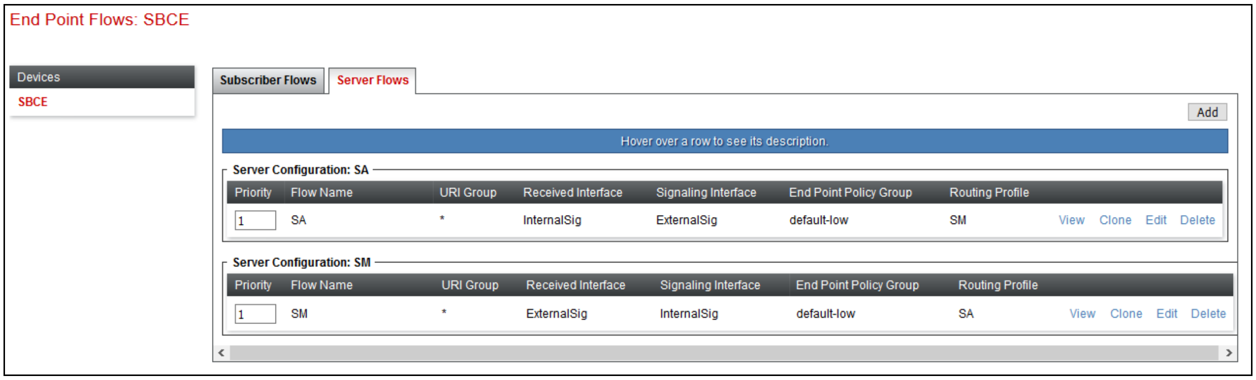


On the **Media Rule** pop-up**,** under **Audio Encryption,** select a **Preferred Format #1** and select **RTP,** select **Next.**

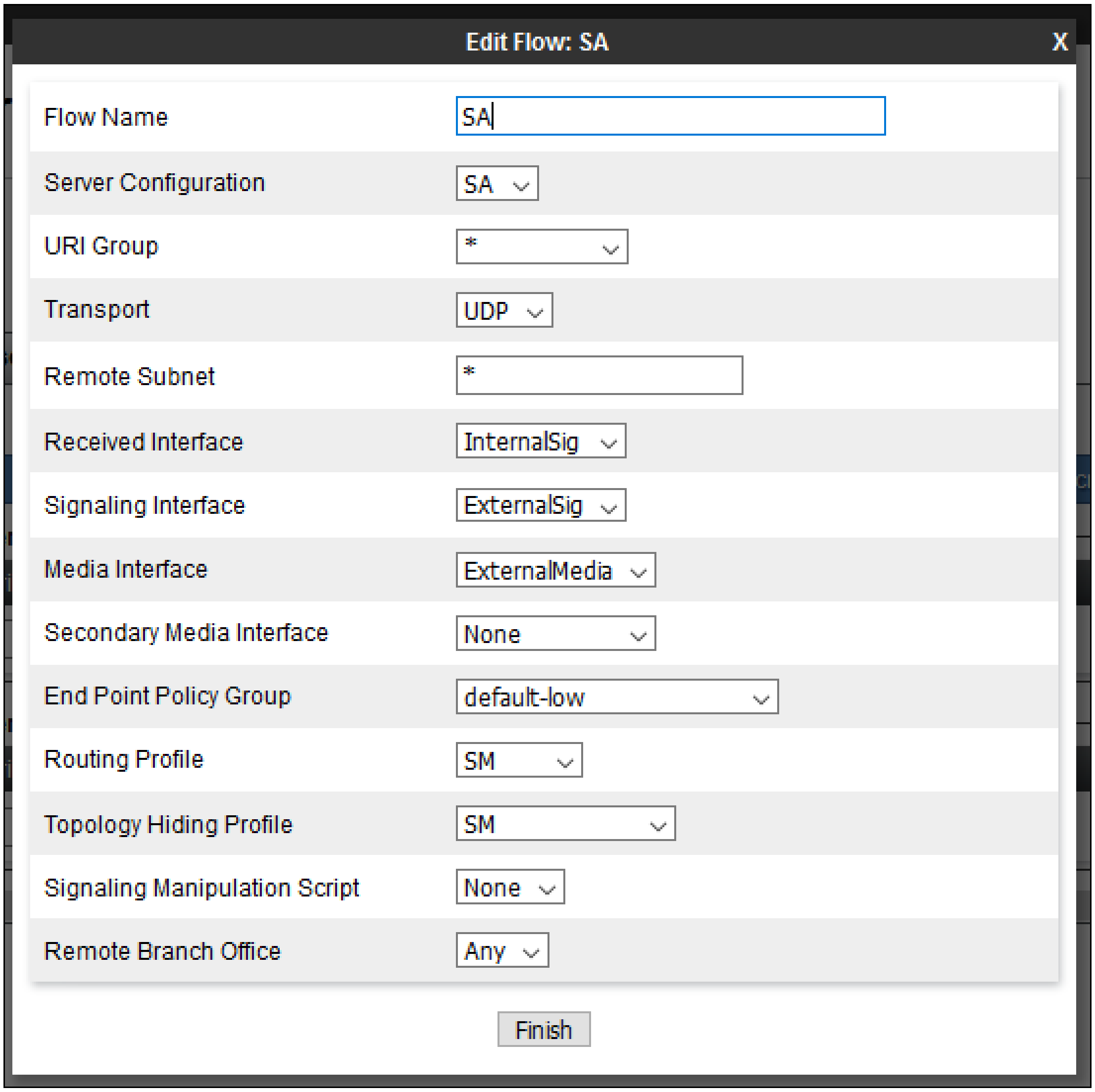


7.6. Server Flows

Server Flows combine the previously defined profiles for Session Manager and Intelligent Virtual Assistant. These End Point Server Flows allow calls to be routed to and from Intelligent Virtual Assistant. Navigate to **Device Specific Setting → End Point Flows → Server Flows.** The screen capture below displays the configured Server Flows. Configure the fields as shown in the screen capture.



Screen captures for configuration of each Server Flow are as shown below:



8. Configure Smart Action Intelligent Virtual Assistant

Configuration is this section is performed by Smart Action engineers. Configuration in this section is for informational purposes only.

8.1. Configure Adjacency

The first configuration object is to establish an adjacency with Avaya SBCE Public IP Address. It is configured to establish a SIP trunk configuration using port 5060, with the default protocol (which is UDP). The adjacency is named as “Customer: Avaya”.

Configuration Objects	Values
Object name	SIP Adjacency "Customer:avaya"
Node	Unknown
Path	Connection to Session Controller "LAX-SBC" / Session Controller "LAX-SBC" / Adjacencies / Adjacencies before SIP Adjacency "S* / SIP Adjacency "**Customer:Avaya**"
SNMP Index OID	3.16.67.117.115.116.111.109.101.114.37.51.65.97.118.97.121.97 (3.Customer:avaya)
CORBA Internal name	BOOPerimetaConnSE.3/BOOPerimetaSE/BOOPeriAdjacencyContainerSE/BOOPeriAdjacencySipSE.Customer:avaya
Identifiers	LAX-SBC/Customer:Avaya

8.2. Adjacency Settings

Table below displays the Adjacency Settings configured during this test.

Parameter	Value
Description	Avaya test adjacency for certification testing
Adjacency Type	preset-peering
Account	None
CollectStatistics	detail
DeactivationMode	normal
NetworkHeading	Network
NetworkServiceAddress	GeneralPeering-01
NetworkServiceNetworkID	2
NetworkLocalAddress	**Public IP Address of Smart Action**
NetworkLocalPort	single-port
NetworkSinglePort	5060

NetworkSignalingPeer	true
NetworkSignalingPeerPort	5060
NetworkTrustSignalingPeer	True
NetworkSignalingPeerGroup	None
NetworkRemoteAddressRangeIP	**Public IP Address of Avaya SBCE**
NetworkRemoteAddressRangePrefixLength	32
NetworkRemoteAddressRangeTrusted	true
RoutingHeading	Routing
RoutingSimpleRouting	disabled
ConnectionHeading	Connections
ConnectionMandateTransport	allow-any-transport
ConnectionNAT	force-on
ConnectionsListenForTransports	default-for-adjacency-type
ConnectionTLS	disabled
IPRoutingHeading	IP Routing
IPRoutingForceSignalingPeer	all-requests
IPRoutingRedirectMode	Use default
IPRoutingRedirectModeDefault	Pass-through
IPRoutingDynamicRoutingDomainMatch	50.207.80.111
PrivacyAndSecurityHeading	Privacy and Security
PrivacyAndSecurityPrivacy	trusted
PrivacyAndSecurityContactHeaderUsernameHandling	rewrite
PrivacyAndSecurityUseUniquePortPerUsername	false
MediaHeading	Media
MediaMSCFallback	none
MediaMSCLocationID	default
MediaRealm	GeneralPeeringMedia1
SIPHeading	SIP
SIPOutboundFloodPolicing	false
SIPDefaultInteropProfile	BOOPerimetaConnSE.3/BOOPerimetaSE/BOOPeriInteropProfilesContainerSE/BOOPeriInterop
ProfileSE_Peer	// Peer
RegistrationHeading	Registration
RegistrationRequired	false
RegistrationOutgoingTimer	0
RegistrationOutgoingNegotiateLocalExpiry	false
TrunkGroupHeading	Trunk group
TrunkGroupTrunkGroup	passthrough
MessageManipulationHeading	Message Manipulation

MessageManipulationErrorProfile	None
MessageManipulationLuaConfigSet	None
CallFlowHeading	Call Flow
CallFlowREFERToINVITETransfer	False
IMSHeading	IMS
IMSPChargingVectorHeader	Use default
IMSPChargingVectorHeaderDefaultValue	passthrough
AlarmsHeading	Alarms
AlarmState	Clear
StatusHeading	Status
RequestedStatus	Active
ActualStatus	Active

8.3. Call Policy Settings

The second configuration object establishes a call policy in which is matched for any calls coming in via the Avaya adjacency (as defined above), and send those calls to a routing table named "DID_Maps". Table below displays the Call Policy Settings configured during this test.

Parameter	**Value**
Object Name	Avaya call policy
Node	Unknown
Path	Connection to Session Controller "LAX-SBC" / Session Controller "LAX-SBC" / Call Policy Sets / Call Policy Set 1 "Routing based on called number" (active) / Routing Table "Initial_Routing_Table" (Source adjacency) / Avaya call policy
SNMP Index OID	3.21.73.110.105.116.105.97.108.95.82.111.117.116.105.110.103.95.84.97.98.108.101 (3.Initial_Routing_Table)
CORBA Internal Name	BOOPerimetaConnSE.3/BOOPerimetaSE/BOOPeriCallPolicySetsContainerSE/BOOPeriCallPolicySetSE.1/BOOPeriRoutingTableSE.Initial_Routing_Table/BOOPeriRoutingEntrySE.4
Identifiers	LAX-SBC1/Initial_Routing_Table/4
MatchAdjacencyHeading	Match adjacency
MatchAdjacencyMatchAdjacency	specify
MatchAdjacencyAdjacencyToMatch	BOOPerimetaConnSE.3/BOOPerimetaSE/BOOPeriAdjacencyContainerSE/BOOPeriAdjacencySipSE.Customer%3Aavaya //Customer:avaya
NumberManipulationHeading	Number Manipulation

NumberManipulationApplyDestinationNumberManipulation	false
NumberManipulationApplySourceNumberManipulation	false
TrunkGroupIDManipulationHeading	Trunk Group ID Manipulation
TrunkGroupIDManipulationDestinationTrunkGroupIDManipulation	none
TrafficGroupHeading	Traffic Group
TrafficGroupApplyTrafficGroup	false
ActionHeading	Action
ActionAction	Next-table
ActionNextTable	BOOPerimetaConnSE.3/BOOPerimetaSE/BOOPeriCallPolicySetsContainerSE/BOOPeriCallPolicySetSE.1/BOOPeriRoutingTableSE.DID_Maps //"**DID_Maps**" ()
ActionDestinationAdjacency	None
AlarmsHeading	Alarms
AlarmState	Clear
RoutingTableType	source-adjacency

8.4. Routing Policy Settings

The final configuration object sets up a routing policy based on the destination phone number / extension. In the case of the testing, the number we matched on was the extension 48600 (**Section 5.9**). When that condition occurs, the call is forwarded to one of our QA IVR servers, named SA:PBX8. Table below displays the Routing Policy Settings configured during this test.

Parameter	Value
Object Name	Entry 16
Node	Unknown
Path	Connection to Session Controller "LAX-SBC" / Session Controller "LAX-SBC" / Call Policy Sets / Call Policy Set 1 "Routing based on called number" (active) / Routing Table "DID_Maps" (Destination ID) / Entry 16
SNMP Index OID	3.8.68.73.68.95.77.97.112.115 (3.DID_Maps)
CORBA Internal name	BOOPerimetaConnSE.3/BOOPerimetaSE/BOOPeriCallPolicySetsContainerSE/BOOPeriCallPolicySetSE.1/BOOPeriRoutingTableSE.DID_Maps/BOOPeriRoutingEntrySE.16
Identifiers	LAX-SBC1/DID_Maps

Parameter	Value
MatchIDHeading	Match ID
MatchIDMatchType	digits
MatchIDDigitsToMatch	**48600**
MatchIDPerformPrefixMatching	false
NumberManipulationHeading	Trunk Group ID Manipulation
TrunkGroupIDManipulationDestinationTrunkGroupIDManipulation	none
TrunkGroupIDManipulationSourceTrunkGroupIDManipulation	none
TrafficGroupHeading	Traffic Group
TrafficGroupApplyTrafficGroup	false
Action Heading	Action
ActionAction	complete
ActionDestinationAdjacency	BOOPerimetaConnSE.3/BOOPerimetaSE/BOOPeriAdjacencyContainerSE/BOOPeriAdjacencySipSE.SA%3APBX8 //SA:PBX8
AlarmsHeading	Alarms
AlarmState	Clear
RoutingTableType	Destination-id

9. Verification Steps

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

9.1. 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

                             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
```

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.

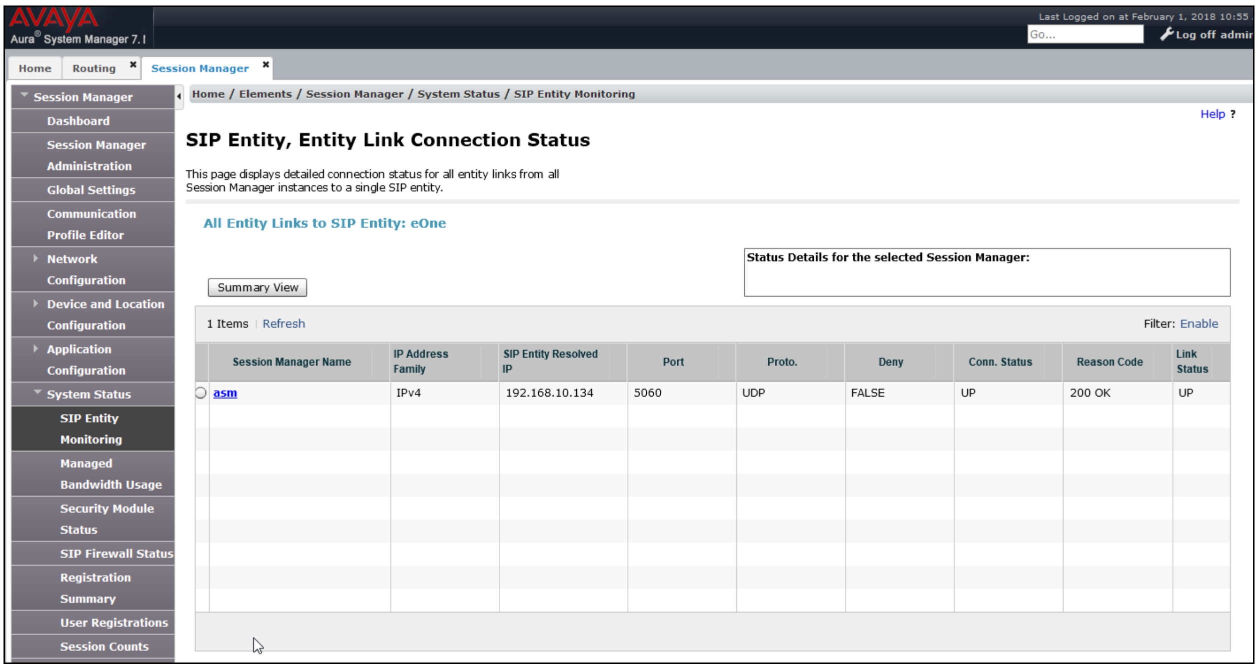
```
status signaling-group 1
                         STATUS SIGNALING GROUP

        Group ID: 1
      Group Type: sip

     Group State: in-service
```

9.2. 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).

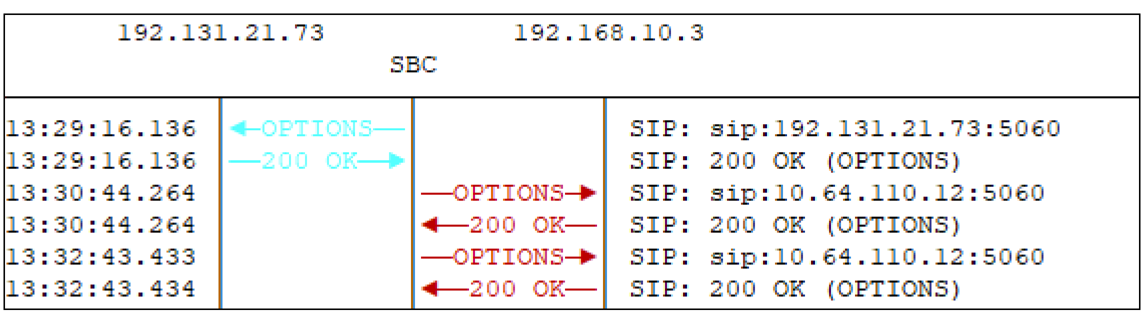
Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the Avaya SBCE entity name from **Section 6.2.1**.

The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are "Up".



9.3. Verify Avaya Session Border Controller for Enterprise

Log onto Avaya SBCE via a shell console and run **tracesbc** command. Verify the OPTIONS from SBC to Session Manager and Intelligent Virtual Assistant are getting responded with 200 OK. This verifies SIP connectivity to and from Avaya SBCE.



10. Conclusion

These Application Notes describe the configuration steps required for Intelligent Virtual Assistant to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks. All feature and serviceability test cases were completed.

11. Additional References

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

1. *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 10, Release 7.1, August 2017, available at http://support.avaya.com.

2. *Administering Avaya Aura® Session Manager*, Release 7.1, Issue 7, September 2017, available at http://support.avaya.com.

3. *Installing Intelligent Virtual Assistant*, available from http://www.instruments.com.

4. *Intelligent Virtual Assistant Application Server*, available from http://www.instruments.com.

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.