



## **Application Notes for Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 8.0 with Frontier Communications SIP Trunking Service – Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service on an enterprise solution consisting of Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 8.0 to interoperate with Frontier Communications SIP Trunking service. These Application Notes update previously published Application Notes with newer versions of Communication Manager, Session Manager, and Avaya Session Border Controller for Enterprise.

The test was performed to verify SIP trunk features including basic calls, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls were placed to and from the PSTN with various Avaya endpoints.

The Frontier Communications SIP Trunking service provides customers with PSTN access via a SIP trunk between the enterprise and the Frontier Communications network, as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

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.

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# 1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service between the Frontier Communications network and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager 8.0 (Communication Manager), Avaya Aura® Session Manager 8.0 (Session Manager), Avaya Aura® Experience Portal 7.2 (Experience Portal), Avaya Session Border Controller for Enterprise 8.0 (Avaya SBCE) and various Avaya endpoints, listed in **Section 4**.

The Frontier Communications SIP Trunking service referenced within these Application Notes is designed for business customers. Customers using this service with this Avaya enterprise solution are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

The terms “Service Provider”, “Frontier” or “Frontier Communications” will be used interchangeably throughout these Application Notes.

## 2. General Test Approach and Test Results

A simulated CPE site containing all the equipment for the Avaya SIP-enabled enterprise solution was installed at the Avaya Solution and Interoperability Lab. The enterprise site was configured to connect to the network via a broadband connection to the public Internet.

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 only (private network side). 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.

### 2.1. Interoperability Compliance Testing

To verify SIP trunk interoperability, the following features and functionality were covered during the interoperability compliance test:

- Static IP SIP Trunk authentication.

- Response to SIP OPTIONS queries.
- Incoming calls from the PSTN were routed to DID numbers assigned by Frontier Communications. Incoming PSTN calls were terminated to the following endpoints: Avaya 96x1 Series IP Deskphones (H.323 and SIP), Avaya J179 IP Deskphones (H.323), Avaya 2420 Digital Deskphones, Avaya one-X® Communicator softphone (H.323 and SIP), Avaya Equinox softphone (SIP) and analog Deskphones.
- Inbound and outbound PSTN calls to/from Remote Workers using Avaya 96x1 Deskphones (SIP).
- Outgoing calls to the PSTN were routed via Frontier Communications network to various PSTN destinations.
- Proper disconnect when the caller abandons the call before the call is answered.
- Proper disconnect via normal call termination by the caller or the called parties.
- Proper disconnect by the network for calls that are not answered (with voicemail off).
- Proper response to busy endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Proper Codec negotiation and two-way speech-path. Testing was performed with codecs: G.711MU and G.729.
- No matching codecs.
- DTMF tone transmissions as out-of-band RTP events as per RFC2833:
  - Outbound call to PSTN application requiring DTMF (e.g., an IVR or voice mail system).
  - Inbound call from PSTN to Avaya CPE application requiring DTMF (e.g., Aura® Messaging, Experience Portal, Avaya vector digit collection steps).
- Calling number blocking (Privacy).
- Call Hold/Resume (long and short duration).
- Call Forward (unconditional, busy, no answer).
- Blind Call Transfers.
- Consultative Call Transfers.
- Station Conference.
- EC500 (Extension to Cellular) calls.
- Inbound caller interaction with Experience Portal applications, including prompting, caller DTMF input, wait treatment (e.g., announcements and/or music on hold).
- Experience Portal use of SIP REFER to redirect inbound calls, via the Avaya SBCE, to the appropriate Communication Manager agents and extensions.
- Call and two-way talk path establishment between callers and Communication Manager agents and extensions following redirection from Experience Portal.
- Routing inbound vector call to call center agent queues.
- G.711 pass-through fax.
- Simultaneous active calls.
- Long duration calls (over one hour).
- Proper response/error treatment to all trunks busy.
- Proper response/error treatment when disabling SIP connection.

**Note** – Remote Worker was tested as part of this solution. The configuration necessary to support remote workers is beyond the scope of these Application Notes and is not included in these Application Notes. Consult reference [12] in the **References** section for additional information on this topic.

Items not supported or not tested included the following:

- The SIP REFER method for call redirection is not fully supported by Frontier, therefore it was not tested.
- T.38 fax was not tested for reasons mentioned under **Section 2.2**.
- Inbound toll-free calls were not tested.
- 0, 0+10 digits, 411 Directory Assistance, 911 Emergency and international calls are supported by Frontier but were not tested.

## 2.2. Test Results

Interoperability testing of the Frontier Communications SIP Trunking Service with the Avaya SIP-enabled enterprise solution was completed with successful results for all test cases with the observations/limitations noted below:

- **SIP OPTIONS** – SIP OPTIONS messages sent by Frontier to the enterprise contained a non-routable SIP URI, causing Avaya Session Manager to respond with “404 Not Found (No route available)”. Since the SIP OPTIONS messages sent by Frontier to the enterprise were intended for link monitoring any response received by Frontier was acceptable. This observation was reported to Frontier with Frontier confirming that any response was acceptable to keep the SIP trunk link up.
- **T.38 Fax** – With Communication Manager configured as “T.38-G711-fallback” (refer to **Section 5.4**), on incoming fax call attempts from the PSTN to Communication Manager, Frontier responded with "488 Not Acceptable Here" to the re-INVITE message sent by Communication Manager to switch from G.711 audio to T.38 fax, this resulted on the fax call defaulting to G.711 pass-through. Incoming fax calls were successfully tested using the G.711 pass-through method. On outgoing fax calls from Communication Manager to the PSTN, Frontier did not send the re-INVITE message to Communication Manager to switch from G.711 audio to T.38 fax within the 4 seconds time-out interval expected by Communication Manager, this caused Communication Manager to send a re-INVITE message to Frontier for G.711, this resulted on the fax being sent via G.711 pass-through. Outbound fax calls using the G.711 pass-through method was unreliable. It should be noted that due to the unpredictability of G.711 pass-through techniques, which only works well on networks with very few hops and with limited end-to-end delay, G.711 fax pass-through is delivered on a “best effort” basis; its success is not guaranteed, and it should be used at the customer’s discretion. T.38 fax is supported in the Frontier’s production environment, but currently it’s not supported in the Frontier’s lab environment used during the testing.
- **Call display on H.323 Telephones** – Calls from the enterprise to the PSTN originated from Avaya H.323 telephones, the call display on H.323 telephones was updated with unrecognized letters and numbers (e.g., **f3412ebf016...**) when the call was answered at the PSTN, instead of being updated with the called number (the PSTN number being dialed). The issue was related to the “Contact” header in SIP messages received from Frontier, the content in the “Contact” headers of SIP messages received from Frontier

was being concealed by Frontier as part of their topology hiding. Changes were made by Frontier to allow the correct content in the Contact headers to be passed to the enterprise, thus solving the call display issue.

- **Incorrect Call Display on call transfers to the PSTN Phone** – Call display was not properly updated on PSTN phones involved in call transfers. After successful call transfers to the PSTN, the PSTN phone did not display the actual connected party, instead the DID number assigned to the Communication Manager station that initiated the transfer was displayed.
- **TLS/SRTP used within the enterprise** – When TLS/SRTP is used within the enterprise; the SIP headers include the SIPS URI scheme for Secure SIP. The Avaya SBCE converts these header schemes from SIPS to SIP when it sends the SIP message toward Frontier. However, for call forward and EC500 calls, the Avaya SBCE was not changing the Diversion header scheme as expected. This anomaly is currently under investigation by the Avaya SBCE team. A workaround is to include a SigMa script for the Service Provider Server Configuration profile on the Avaya SBCE to convert “sips” to “sip” in the Diversion header. See **Sections 8.8 and 13**.
- **Outbound call from an enterprise extension to a busy PSTN number** – Frontier Communications did not send a “486 Busy Here” response on outbound calls to busy PSTN numbers, as expected. There was no direct impact to the user, who heard busy tone.
- **Removal of unwanted xml element information from the SDP in SIP messages sent to Frontier Communications** – A Signaling Manipulation script (SigMa) on the Avaya SBCE was created to remove unwanted xml element information from the SDP in SIP messages the Avaya SBCE sent Frontier Communications, the xml elements were causing Frontier to respond with “415 Unsupported Media Type” to SIP messages sent by Communication Manager. Refer to **Sections 8.8 and 13**.
- **SIP header optimization** – There are multiple SIP headers and parameters used by Communication Manager and Session Manager, some of them Avaya proprietary, that had no significance in the service provider’s network. These headers were removed with the purpose of blocking enterprise information from being propagated outside of the enterprise boundaries, to reduce the size of the packets entering the service provider’s network and to improve the solution interoperability in general. The following headers were removed from outbound messages using an Adaptation in Session Manager: AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-id, P-Charging-Vector, AV-Global-Session-ID and P-Location (Refer to **Section 7.4**). To help reduce the packet size further, the Avaya SBCE can remove the “gsid” parameters that may be included within the Contact header by applying a Sigma script to the Frontier Communications server configuration. Refer to **Section 8.8, and 13**.

## 2.3. Support

For support of Frontier Communications SIP Trunking Service visit the corporate Web page at: <https://frontier.com/enterprise>

For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>

### 3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Frontier Communications SIP Trunking Service through a public Internet WAN connection.

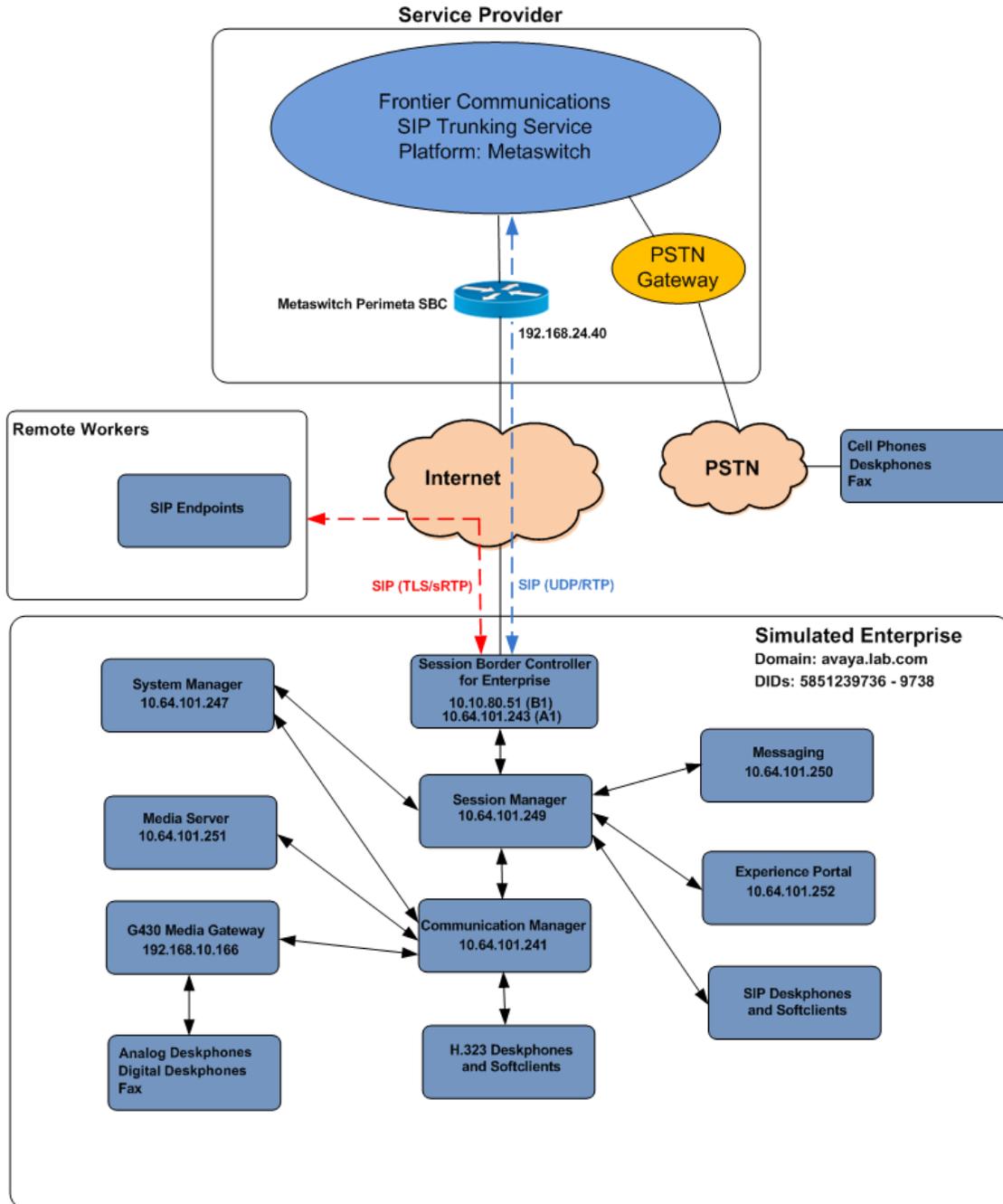


Figure 1: Avaya SIP Enterprise Solution connected to Frontier Communications SIP Trunking Service

The Avaya components used to create the simulated enterprise customer site included:

- Avaya Aura® Communication Manager.
- Avaya Aura® Session Manager.
- Avaya Aura® System Manager.
- Avaya Session Border Controller for Enterprise.
- Avaya Aura® Messaging.
- Avaya Aura® Media Server.
- Avaya Aura® Experience Portal.
- Avaya G430 Media Gateway.
- Avaya 96x1 Series IP Deskphones (H.323 and SIP).
- Avaya J179 IP Deskphones (H.323).
- Avaya one-X® Communicator softphones (H.323 and SIP).
- Avaya Equinox™ for Windows softphone (SIP).
- Avaya digital and analog telephones.
- Ventafax fax software.

Additionally, the reference configuration included remote worker functionality. A remote worker is a SIP endpoint that resides in the untrusted network, registered to Session Manager at the enterprise via the Avaya SBCE. Remote workers offer the same functionality as any other endpoint at the enterprise. This functionality was successfully tested during the compliance test using only the Avaya 96x1 SIP Deskphones. For signaling, Transport Layer Security (TLS) and for media, Secure Real-time Transport Protocol (SRTP) was used on Avaya 96x1 SIP Deskphones used to test remote worker functionality. Other Avaya SIP endpoints that are supported in a Remote Worker configuration deployment were not tested.

The configuration tasks required to support remote workers are beyond the scope of these Application Notes; hence they are not discussed in this document. Consult reference [11] in the **References** section for additional information on this topic.

The Avaya SBCE was located at the edge of the enterprise. Its public side was connected to the public Internet, while its private side was connected to the enterprise infrastructure. All signaling and media traffic entering or leaving the enterprise flowed through the Avaya SBCE, protecting in this way the enterprise against any SIP-based attacks. The Avaya SBCE also performed network address translation at both the IP and SIP layers.

For inbound calls, the calls flowed from the service provider to the Avaya SBCE then to Session Manager. Session Manager used the configured dial patterns (or regular expressions) and routing policies to determine the recipient (Communication Manager or Experience Portal) and on which link to send the call.

Outbound calls to the PSTN were first processed by Communication Manager for outbound feature treatment such as automatic route selection and class of service restrictions. Once Communication Manager selected the proper SIP trunk, the call was routed to Session Manager.

Session Manager once again used the configured dial patterns (or regular expressions) and routing policies to determine the route to the Avaya SBCE for egress to the Frontier Communications network.

A separate SIP trunk was created between Communication Manager and Session Manager to carry the service provider traffic. This was done so that any trunk or codec settings required by the service provider could be applied only to this trunk without affecting other enterprise SIP traffic. This trunk carried both inbound and outbound traffic.

As part of the Avaya Aura® version 8.0 release, Communication Manager incorporates the ability to use the Avaya Aura® Media Server (AAMS) as a media resource. The AAMS is a software-based, high density media server that provides DSP resources for IP-based sessions. Media resources from both the AAMS and a G430 Media Gateway were utilized during the compliance test. The configuration of the AAMS is not discussed in this document. For more information on the installation and administration of the AAMS in Communication Manager refer to the AAMS documentation listed in the **References** section.

The Avaya Aura® Messaging was used during the compliance test to verify voice mail redirection and navigation, as well as the delivery of Message Waiting Indicator (MWI) messages to the enterprise telephones. Since the configuration tasks for Messaging are not directly related to the interoperability tests with the Frontier Communications network SIP Trunking service, they are not included in these Application Notes.

The Avaya Aura® Experience Portal was also used during the compliance test to verify various SIP call flow scenarios with Frontier Communications SIP trunking service.

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
<b>Avaya</b>	
Avaya Aura® Communication Manager	8.0.1.1.0 (00.0.822.0-25183)
Avaya Aura® Session Manager	8.0.1.1 (8.0.1.1.801103)
Avaya Aura® System Manager	8.0.1.1 Build No. 8.0.0.0.931077 Software Update Rev. No. 8.0.1.1.039340
Avaya Session Border Controller for Enterprise	ASBCE 8.0 8.0.0.0-19-16991
Avaya Aura® Messaging	7.1 Patch 1
Avaya Aura® Media Server	8.0.0 SP3 8.0.0.15
Avaya G430 Media Gateway	g430_sw_40_25_0
Avaya Aura® Experience Portal	7.2.2.0.2065
Avaya 96x1 Series IP Deskphones (SIP)	Version 7.1.4.0.11
Avaya 96x1 Series IP Deskphones (H.323)	Version 6.8102
Avaya J179 IP Deskphones (H.323)	Version 6.8102
Avaya one-X® Communicator (H.323, SIP)	6.2.13.2-SP13-Patch1
Avaya Equinox for Windows (SIP)	3.5.7.30.1
Avaya 2420 Series Digital Deskphones	N/A
Avaya 6210 Analog Deskphones	N/A
<b>Frontier Communications</b>	
Metaswitch cCFS (Clustered Call Feature Server)	9.3.20
Metaswitch Perimeta SBC	4.3.40

The specific configuration above was used for the compliance testing. Note that this solution will be compatible with other Avaya Servers and Media Gateway platforms running similar versions of Communication Manager and Session Manager.

**Note** – The Avaya Aura® servers and the Avaya SBCE used in the reference configuration and shown on the previous table were deployed on a virtualized environment. These Avaya components ran as virtual machines over VMware® (ESXi 6.0.0) platforms. Consult the installation documentation on the **References** section for more information.

## 5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager to work with the Frontier Communications SIP Trunking Service. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from the service provider. It is assumed that the general installation of Communication Manager, the Avaya G430 Media Gateway and the Avaya Media Server has been previously completed and is not discussed here.

The Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. Some screens capture will show the use of the **change** command instead of the **add** command, since the configuration used for the testing was previously added.

### 5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to and from the service provider. The example shows that **30000** licenses are available and **120** are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

```
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 2
      Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
      Maximum Concurrently Registered IP eCons: 414 0
      Max Concur Registered Unauthenticated H.323 Stations: 100 0
      Maximum Video Capable Stations: 41000 0
      Maximum Video Capable IP Softphones: 18000 6
      Maximum Administered SIP Trunks: 30000 120
Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
      Maximum Number of DS1 Boards with Echo Cancellation: 688 0

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

## 5.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to *all* to allow incoming calls from the PSTN to be transferred to another PSTN endpoint. If for security reasons incoming calls should not be allowed to transfer back to the PSTN, then leave the field set to *none*.

```
display 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
```

On **Page 9** verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. This text string is entered in the two fields highlighted below. The compliance test used the value of *restricted* for restricted calls and *unavailable* for unavailable calls.

```
display system-parameters features Page 9 of 19
FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
CPN/ANI/ICLID Replacement for Restricted Calls: restricted
CPN/ANI/ICLID Replacement for Unavailable Calls: unavailable

DISPLAY TEXT
Identity When Bridging: principal
User Guidance Display? n
Extension only label for Team button on 96xx H.323 terminals? n

INTERNATIONAL CALL ROUTING PARAMETERS
Local Country Code:
International Access Code:

SCCAN PARAMETERS
Enable Enbloc Dialing without ARS FAC? n

CALLER ID ON CALL WAITING PARAMETERS
Caller ID on Call Waiting Delay Timer (msec): 200
```



## 5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, ip-codec-set 2 was used for this purpose. Enter the corresponding codec in the **Audio Codec** column of the table. Frontier Communications supports audio codecs *G.711MU* and *G.729*.

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

                                IP MEDIA PARAMETERS

Codec Set: 2

Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt    Size (ms)
1: G.711MU          n         2         20
2: G.729           n         2         20
3: _____      -           -           -
4: _____      -           -           -
5: _____      -           -           -
6: _____      -           -           -
7:                 -           -           -

Media Encryption                                Encrypted SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2: none
3: _____
4: _____
5: _____
```

On Page 2, set the Fax Mode to *t.38-G711-fallback*, ECM to *y* and FB-Timer set to *4*

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

IP MEDIA PARAMETERS

Allow Direct-IP Multimedia? n

Mode                Redun-                Packet
                   dancy                Size (ms)
FAX                 t.38-G711-fallback  0      ECM: y FB-Timer: 4
Modem               off                 0
TDD/TTY             US                 3
H.323 Clear-channel n                 0
SIP 64K Data        n                 0                20

Media Connection IP Address Type Preferences
1: IPv4
2:  
```

## 5.5. IP Network Regions

Create a separate IP network region for the service provider trunk group. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the service provider versus calls within the enterprise or elsewhere. For the compliance test, IP Network Region 2 was chosen for the service provider trunk. Use the **change ip-network-region 2** command to configure region 2 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is *avaya.lab.com* as assigned to the shared test environment in the Avaya test lab. This domain name appears in the “From” header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Leave both **Intra-region** and **Inter-region IP-IP Direct Audio** set to *yes*, the default setting. This will enable **IP-IP Direct Audio** (shuffling), to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway and Media Server. Shuffling can be further restricted at the trunk level on the Signaling Group form if needed.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values may be used for all other fields.

```
change ip-network-region 2                                     Page 1 of 20
                                     IP NETWORK REGION
Region: 2      NR Group: 2
Location: 1    Authoritative Domain: avaya.lab.com
Name: SP Region      Stub Network Region: n
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
Codec Set: 2          Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048    IP Audio Hairpinning? n
UDP Port Max: 3349
DIPFSERV/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
```

On **Page 4**, define the IP codec set to be used for traffic between region 2 and region 1 (the rest of the enterprise). Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The following example shows the settings used for the compliance test. It indicates that codec set **2** will be used for calls between region 2 (the service provider region) and region 1 (the rest of the enterprise).

change ip-network-region 2										Page 4 of 20		
Source Region: 2		Inter Network Region Connection Management							I	M		
dst rgn	codec set	direct WAN	WAN-BW-limits		Video		Intervening	Dyn CAC	A R	G L	t c e t	
1	<b>2</b>	<b>y</b>	<b>NoLimit</b>						<b>n</b>		<b>t</b>	
2	2									<b>all</b>		
3	_____											
4	_____											
5	_____											
6	_____											
7	_____											
8	_____											
9	_____											
10	_____											
11	_____											
12	_____											
13	_____											
14	_____											
15	_____											

## 5.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and Session Manager for use by the service provider trunk. This signaling group is used for inbound and outbound calls between the service provider and the enterprise. For the compliance test, signaling group 2 was used and was configured using the parameters highlighted below, shown on the screen on the next page:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*. This specifies the Communication Manager will serve as an Evolution Server for the Session Manager.
- Set the **Transport Method** to the transport protocol to be used between Communication Manager and Session Manager. For the compliance test, *tls* was used.
- Set the **Peer Detection Enabled** field to *y*. The **Peer-Server** field will initially be set to *Others* and cannot be changed via administration. Later, the **Peer-Server** field will automatically change to *SM* once Communication Manager detects its peer is a Session Manager.

**Note:** Once the **Peer-Server** field is updated to *SM*, the system changes the default values of the following fields, setting them to display-only:

- Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? is changed to *y*.
- Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? is changed to *n*.
- Set the **Near-end Node Name** to *procr*. This node name maps to the IP address of the Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to *SM*. This node name maps to the IP address of Session Manager, as defined in **Section 5.3**.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061). This is necessary so Session Manager can distinguish this trunk from the trunk used for other enterprise SIP traffic. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to *5071*.
- Set the **Far-end Network Region** to the IP network region defined for the Service Provider in **Section 5.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set the **DTMF over IP** field to *rtp-payload*. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Set **Direct IP-IP Audio Connections** to *y*. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the Avaya SBCE and the enterprise endpoint. If this value is set to *n*, then the Avaya Media Gateway or Media Server will remain in the media path of all calls between the SIP trunk and the endpoint. Depending on the number of media resources available in the Avaya Media Gateway and Media Server, these resources may be depleted during high call volume preventing additional calls from completing.
- Default values may be used for all other fields.

```

change signaling-group 2                               Page 1 of 2
                SIGNALING GROUP

Group Number: 2          Group Type: sip
IMS Enabled? n          Transport Method: tls
Q-SIP? n
IP Video? n
Peer Detection Enabled? y Peer Server: SM          Enforce SIPS URI for SRTP? y
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y          Clustered? n
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
Near-end Listen Port: 5071          Far-end Listen Port: 5071
Far-end Network Region: 2

Far-end Domain: avaya.lab.com

Incoming Dialog Loopbacks: eliminate          Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload          RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3          Direct IP-IP Audio Connections? y
Enable Layer 3 Test? n          IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n          Initial IP-IP Direct Media? n
Alternate Route Timer(sec): 6

```

## 5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.6**. For the compliance test, trunk group 2 was configured using the parameters highlighted below.

- Set the **Group Type** field to *sip*.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Service Type** field to *public-ntwrk*.
- Set the **Signaling Group** to the signaling group shown in **Section 5.6**.
- Set the **Number of Members** field to the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk.
- Default values were used for all other fields.

```
change trunk-group 2                                     Page 1 of 4
                                     TRUNK GROUP
Group Number: 2          Group Type: sip          CDR Reports: y
Group Name: Service Provider          COR: 1          TN: 1          TAC: 602
Direction: two-way          Outgoing Display? n
Dial Access? n          Night Service: _____
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                     Member Assignment Method: auto
                                     Signaling Group: 2
                                     Number of Members: 10
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. The default value of **600** seconds was used.

```
change trunk-group 2                                     Page 2 of 4
  Group Type: sip
TRUNK PARAMETERS
  Unicode Name: auto
  Redirect On OPTIM Failure: 5000
  SCCAN? n                                           Digital Loss Group: 18
  Preferred Minimum Session Refresh Interval(sec): 600
Disconnect Supervision - In? y Out? y
  XOIP Treatment: auto   Delay Call Setup When Accessed Via IGAR? n
Caller ID for Service Link Call to H.323 1xC: station-extension
```

On Page 3:

- Set the **Numbering Format** field to *public*. This field specifies the format of the calling party number (CPN) sent to the far-end. When *public* format is used, Communication Manager automatically inserts a “+” sign, preceding the numbers in the “From”, “Contact” and “P-Asserted Identity” (PAI) headers. To keep uniformity with the format used by Frontier Communications, the **Numbering Format** was set to *public* and the **Numbering Format** in the route pattern was set to *pub-unk* (see **Section 5.10**).
- Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to *y*. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.2**, if the inbound call has enabled CPN block.

```
change trunk-group 2                                     Page 3 of 4
TRUNK FEATURES
  ACA Assignment? n                                     Measured: none
  Maintenance Tests? y
  Suppress # Outpulsing? n   Numbering Format: public
  UI Treatment: service-provider
  Replace Restricted Numbers? y
  Replace Unavailable Numbers? y
  Hold/Unhold Notifications? y
  Modify Tandem Calling Number: no
  Show ANSWERED BY on Display? y
```

On Page 4:

- Set the **Network Call Redirection** field to *n*. With this setting, Communication Manager will not use the SIP REFER method, which is not supported by Frontier, for the redirection of PSTN calls that are transferred back to the SIP trunk (refer to **Section 2.1**).
- Set the **Send Diversion Header** field to *y* and **Support Request History** to *n*.
- Set the **Telephone Event Payload Type** to **101**, the value preferred by Frontier Communications.
- Verify that **Identity for Calling Party Display** is set to *P-Asserted-Identity*.
- Default values were used for all other fields.

```
change trunk-group 2                                     Page 4 of 4
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? y
Support Request History? n
Telephone Event Payload Type: 101
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
```

## 5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since public numbering was selected to define the format of this number (**Section 5.7**), use the **change public-unknown-numbering** command to create an entry for each extension which has a DID assigned. DID numbers are provided by the SIP service provider. Each DID number is assigned in this table to one enterprise internal extension or Vector Directory Numbers (VDNs). In the example below, two DID numbers assigned by the service provider are shown. Notice the “1” preceding each DID number, required by Frontier. These DID numbers were used as the outbound calling party information on the service provider trunk when calls were originated from the mapped extensions.

change public-unknown-numbering 1					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len	
4	3			4	Total Administered: 4
4	5			4	Maximum Entries: 9999
4	3041	2	15851239736	11	Note: If an entry applies to a SIP connection to Avaya Aura(R) Session Manager, the resulting number must be a complete E.164 number.
4	3044	2	15851239737	11	
					Communication Manager automatically inserts a '+' digit in this case.

## 5.9. Inbound Routing

In general, the “incoming call handling treatment” form for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can be used to perform digit conversion using an Adaptation, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. If the DID number sent by Frontier Communications is left unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. Use the **change inc-call-handling-trmt** command to create an entry for each DID.

```
change inc-call-handling-trmt trunk-group 2 Page 1 of 30
```

INCOMING CALL HANDLING TREATMENT				
Service/ Feature	Number Len	Number Digits	Del	Insert
public-ntwrk	10	5851239736	10	3041
public-ntwrk	10	5851239737	10	3044
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—
public-ntwrk	—	—	—	—

## 5.10.Outbound Routing

In these Application Notes, the Automatic Route Selection (ARS) feature is used to route outbound calls via the SIP trunk to the service provider. In the sample configuration, the single digit 9 is used as the ARS access code. Enterprise callers will dial 9 to reach an “outside line”. This common configuration is illustrated below with little elaboration. Use the **change dialplan analysis** command to define a dialed string beginning with 9 of length 1, as a feature access code (*fac*).

change dialplan analysis			DIAL PLAN ANALYSIS TABLE			Page 1 of 12		
			Location: all			Percent Full: 2		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
0	13	udp						
1	4	dac						
2	4	ext						
3	4	ext						
4	4	udp						
5	4	ext						
6	3	dac						
7	4	ext						
8	1	fac						
9	1	fac						
*	3	dac						
#	2	dac						

Use the `change feature-access-codes` command to configure 9 as the **Auto Route Selection (ARS) – Access Code 1**.

```
change feature-access-codes Page 1 of 10
FEATURE ACCESS CODE (FAC)
Abbreviated Dialing List1 Access Code: ____
Abbreviated Dialing List2 Access Code: ____
Abbreviated Dialing List3 Access Code: ____
Abbreviated Dial - Prgm Group List Access Code: ____
Announcement Access Code: #7
Answer Back Access Code: ____
Attendant Access Code: ____
Auto Alternate Routing (AAR) Access Code: 8
Auto Route Selection (ARS) - Access Code 1: 9 Access Code 2: ____
Automatic Callback Activation: ____ Deactivation: ____
Call Forwarding Activation Busy/DA: ____ All: ____ Deactivation: ____
Call Forwarding Enhanced Status: ____ Act: ____ Deactivation: ____
Call Park Access Code: ____
Call Pickup Access Code: ____
CAS Remote Hold/Answer Hold-Unhold Access Code: ____
CDR Account Code Access Code: ____
Change COR Access Code: ____
Change Coverage Access Code: ____
Conditional Call Extend Activation: ____ Deactivation: ____
Contact Closure Open Code: ____ Close Code: ____
```

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example below shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to route pattern 2, which contains the SIP trunk group to the service provider.

```
list ars analysis Page 8
```

ARS DIGIT ANALYSIS REPORT

Location: all

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Number	ANI Req
178	11	11	deny	fnpa		n
1786	11	11	2	fnpa		n
179	11	11	deny	fnpa		n
180	11	11	deny	fnpa		n
1800	11	11	2	fnpa		n
1800555	11	11	deny	fnpa		n
1809	11	11	2	hnpa		n
181	11	11	deny	fnpa		n
182	11	11	deny	fnpa		n
183	11	11	deny	fnpa		n
184	11	11	deny	fnpa		n
185	11	11	deny	fnpa		n

press CANCEL to quit -- press NEXT PAGE to continue

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the service provider trunk route pattern in the following manner. The example below shows the values used for route pattern 2 in the compliance test.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** Enter the outbound trunk group for the SIP service provider.
- **FRL:** Set the Facility Restriction Level (FRL) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **Pfx Mrk:** Set to **1** to ensure 1 + 10 digits are sent to the service provider for long distance numbers in the North American Numbering Plan (NANP).
- **Numbering Format:** Set to *pub-unk*. All calls using this route pattern will use the public numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.7**.

```

change route-pattern 2                                     Page 1 of 4
      Pattern Number: 2      Pattern Name: Serv. Provider
  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: 2 0  ___ 1  ___  ___  ___  _____  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
1: Y Y Y Y Y n n      rest      _____  -  pub-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: Y Y Y Y Y n n      rest      _____  -  _____  none
  
```

**Note -** Enter the **save translation** command (not shown) to save all the changes made to the Communication Manager configuration in the previous sections.

## 6. Configure Avaya Aura® Experience Portal

These Application Notes assume that the necessary Experience Portal licenses have been installed and basic Experience Portal administration has already been performed. Consult [9] in the **References** section for further details if necessary.

### 6.1. Background

Experience Portal consists of one or more Media Processing Platform (MPP) servers and an Experience Portal Manager (EPM) server. A single “server configuration” was used in the reference configuration. This consisted of a single MPP and EPM, running on a VMware environment, including an Apache Tomcat Application Server (hosting the Voice XML (VXML) and/or Call Control XML (CCXML) application scripts), that provide the directives to Experience Portal for handling the inbound calls.

References to the Voice XML and/or Call Control XML applications are administered on Experience Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Experience Portal, the called party DID number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match is found, Experience Portal informs the caller that the call cannot be handled and disconnects the call<sup>1</sup>.

For the sample configuration described in these Application Notes, a simple VXML test application was used to exercise various SIP call flow scenarios with the Frontier Communications SIP Trunking service. In production, enterprises can develop their own VXML and/or CCXML applications to meet specific customer self-service needs or consult Avaya Professional Services and/or authorized Avaya Business Partners. The development and deployment of VXML and CCXML applications is beyond the scope of these Application Notes.

---

<sup>1</sup> An application may be configured with “inbound default” as the called number, to process all inbound calls that do not match any other application references.

## 6.2. Logging in and Licensing

This section describes the steps on Experience Portal for administering a SIP connection to the Session Manager.

**Step 1** - Launch a web browser, enter `http://<IP address of the Avaya EPM server>/` in the URL, log in with the appropriate credentials and the following screen is displayed.

**Note** – All page navigation described in the following sections will utilize the menu shown on the left pane of the screenshot below.

The screenshot displays the Avaya Aura Experience Portal Manager interface. At the top, the Avaya logo is on the left, and the user is logged in as 'epadmin' with the last login time 'Jan 29, 2019 at 11:55:28 AM PST'. The main header reads 'Avaya Aura Experience Portal 7.2.0 (ExperiencePortal)'. A navigation menu on the left lists various categories: User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration, Security, Reports, and Multi-Media Configuration. The main content area is titled 'Avaya Aura Experience Portal Manager' and includes a 'You are here: Home' breadcrumb. Below this, there are sections for 'Installed Components' (Media Processing Platform, Email Service, HTML Service, SMS Service) and a 'Legal Notice' section. The 'Legal Notice' section is expanded to show the 'AVAYA GLOBAL SOFTWARE LICENSE TERMS', which were revised on May 1, 2017. The terms state that they govern the use of proprietary software and third-party proprietary software licensed through Avaya.

**Step 2** - In the left pane, navigate to **Security**→**Licensing**. On the **Licensing** page, verify that Experience Portal is properly licensed. If required licenses are not enabled, contact an authorized Avaya account representative to obtain the licenses.

The screenshot displays the Avaya Aura Experience Portal 7.2.0 interface. The left navigation pane shows the 'Security' menu expanded to 'Licensing'. The main content area is titled 'Licensing' and includes a 'Refresh' button. Below the title, there is a descriptive paragraph and two data sections: 'License Server Information' and 'Licensed Products'.

**License Server Information**

License Server URL:	https://10.64.101.247:52233/WebLM/LicenseServer
Last Updated:	Dec 4, 2018 3:20:00 PM PST
Last Successful Poll:	Feb 5, 2019 1:34:37 PM PST

**Licensed Products**

<b>Experience Portal</b>	
Announcement Ports:	100
ASR Connections:	100
Email Units:	10
Enable Media Encryption:	1
Enhanced Call Classification:	100
HTML Units:	10
SIP Signaling Connections:	100
SMS Units:	10
Telephony Ports:	100
TTS Connections:	100
Video Server Connections:	100
Zones:	1
Version:	7
Last Successful Poll:	Feb 5, 2019 1:34:37 PM PST
Last Changed:	Dec 4, 2018 3:19:59 PM PST

At the bottom of the page, there are buttons for 'Allocations' and 'Help'.

## 6.3. VoIP Connection

This section defines a SIP trunk between Experience Portal and Session Manager (Sections 7.5 and 7.6).

**Step 1** - In the left pane, navigate to **System Configuration** → **VoIP Connections**. On the **VoIP Connections** page, select the **SIP** tab and click **Add** to add a SIP trunk.

**Note** – Only *one* SIP trunk can be active at any given time on Experience Portal.

The screenshot shows the Avaya Aura Experience Portal 7.2.0 interface. The left navigation pane is expanded to 'System Configuration' > 'VoIP Connections'. The main content area displays the 'VoIP Connections' page with a breadcrumb trail: Home > System Configuration > VoIP Connections. Below the breadcrumb, there is a description: 'This page displays a list of Voice over Internet Protocol (VoIP) servers that Experience Portal communicates with. You can configure multiple SIP connections, but only one SIP connection can be enabled at any one given time.' A tab labeled 'SIP' is selected. Below the tab is a table with the following columns: Name, Enable, Proxy Transport, Proxy/DNS Server Address, Proxy Server Port, Listener Port, SIP Domain, and Maximum Simultaneous Calls. The table contains one entry: EP\_SIP, Yes, TLS, 10.64.101.249, 5061, 5061, avaya.lab.com, 100. Below the table are 'Add', 'Delete', and 'Help' buttons. The 'Add' button is highlighted with a red box.

Name	Enable	Proxy Transport	Proxy/DNS Server Address	Proxy Server Port	Listener Port	SIP Domain	Maximum Simultaneous Calls
EP_SIP	Yes	TLS	10.64.101.249	5061	5061	avaya.lab.com	100

**Step 2** - Configure a SIP connection as follows:

- **Name** – Set to a descriptive name (e.g., **EP\_SIP**).
- **Enable** – Set to **Yes**.
- **Proxy Server Transport** – Set to **TLS**.
- Select **Proxy Servers**, and enter:
  - **Proxy Server Address** = **10.64.101.249** (the IP address of the Session Manager signaling interface defined in **Section 7.5**).
  - **Port** = **5061**
  - **Priority** = **0** (default)
  - **Weight** = **0** (default)
- **Listener Port** – Set to **5061**.
- **SIP Domain** – Set to **avaya.lab.com** (see **Section 7.2**).
- **Consultative Transfer** – Select **REFER**.
- **SIP Reject Response Code** – Select **ASM (503)**.
- **Maximum Simultaneous Calls** – Set to a number in accordance with licensed capacity. In the reference configuration a value of **100** was used.
- Select **All Calls can be either inbound or outbound**.
- **SRTP Enable** = **Yes**

- **Encryption Algorithm = AES\_CM\_128**
- **Authentication Algorithm = HMAC\_SHA1\_80**
- **RTCP Encryption Enabled = No**
- **RTP Authentication Enabled = Yes**
- Click on **Add** to add SRTP settings to the **Configured SRTP List**
- Use default values for all other fields.
- Click **Save**.

**AVAYA** Welcome, epadmin  
Last logged in Jan 29, 2019 at 11:55:28 AM PST

Avaya Aura® Experience Portal 7.2.0 (ExperiencePortal)

Expand All | Collapse All

- ▼ User Management
  - Roles
  - Users
  - Login Options
- ▼ Real-time Monitoring
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ System Maintenance
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ System Management
  - Application Server
  - EPM Manager
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ System Configuration
  - Applications
  - EPM Servers
  - MPP Servers
  - SNMP
  - Speech Servers
  - VoIP Connections
  - Zones
- ▼ Security
  - Certificates
  - Licensing
- ▼ Reports
  - Standard
  - Custom
  - Scheduled
- ▼ Multi-Media Configuration
  - Email
  - HTML
  - SMS

Name: EP\_SIP  
 Enable:  Yes  No  
 Proxy Transport: TLS  
 Proxy Servers  DNS SRV Domain

Address	Port	Priority	Weight	
10.64.101.249	5061	0	0	Remove

Additional Proxy Server  
 Listener Port: 5061  
 SIP Domain: avaya.lab.com  
 P-Asserted-Identity:  
 Maximum Redirection Attempts: 0  
 Consultative Transfer:  INVITE with REPLACES  REFER  
 SIP Reject Response Code:  ASM (503)  SES (480)  Custom 503

SIP Timers  
 T1: 250 milliseconds  
 T2: 2000 milliseconds  
 B and F: 4000 milliseconds

Call Capacity  
 Maximum Simultaneous Calls: 100  
 All Calls can be either inbound or outbound  
 Configure number of inbound and outbound calls allowed

SRTP  
 Enable:  Yes  No  
 Encryption Algorithm:  AES\_CM\_128  NONE  
 Authentication Algorithm:  HMAC\_SHA1\_80  HMAC\_SHA1\_32  
 RTCP Encryption Enabled:  Yes  No  
 RTP Authentication Enabled:  Yes  No **Add**

Configured SRTP List  
 SRTP-Yes,AES\_CM\_128,HMAC\_SHA1\_80,RTCP Encryption-No,RTP Authentication-Yes **Remove**

**Save Apply Cancel Help**

## 6.4. Speech Servers

The installation and administration of the ASR and TSR Speech Servers are beyond the scope of this document. Some of the values shown below were defined during the Speech Server installations. Note that in the reference configuration the ASR and TTS servers used the same IP address.

## ASR speech server:

The screenshot shows the Avaya Aura Experience Portal 7.2.0 interface. The left navigation pane is expanded to 'System Configuration' > 'Speech Servers'. The main content area displays the 'Speech Servers' configuration page. At the top, there are tabs for 'ASR' and 'TTS', with 'ASR' selected. Below the tabs is a table with the following columns: Name, Enable, Network Address, Engine Type, MRCP, Base Port, Total Number of Licensed ASR Resources, and Languages. A single row is visible with the following values: Name: NuanceASR, Enable: Yes, Network Address: 10.64.101.154, Engine Type: Nuance, MRCP: MRCP V1, Base Port: 4900, Total Number of Licensed ASR Resources: 10, Languages: English(USA) en-US. Below the table are buttons for 'Add', 'Delete', 'Customize', and 'Help'.

## TTS speech server:

The screenshot shows the Avaya Aura Experience Portal 7.2.0 interface. The left navigation pane is expanded to 'System Configuration' > 'Speech Servers'. The main content area displays the 'Speech Servers' configuration page. At the top, there are tabs for 'ASR' and 'TTS', with 'TTS' selected. Below the tabs is a table with the following columns: Name, Enable, Network Address, Engine Type, MRCP, Base Port, Total Number of Licensed TTS Resources, and Voices. A single row is visible with the following values: Name: Nuance, Enable: Yes, Network Address: 10.64.101.154, Engine Type: Nuance, MRCP: MRCP V1, Base Port: 4900, Total Number of Licensed TTS Resources: 10, Voices: English(USA) en-US Jennifer F. Below the table are buttons for 'Add', 'Delete', 'Customize', and 'Help'.

## 6.5. Application References

This section describes the steps for administering a reference to the VXML and/or CCXML applications residing on the application server. In the sample configuration, the applications were co-resident on one Experience Portal server, with IP Address 10.64.101.252.

**Step 1** - In the left pane, navigate to **System Configuration** → **Applications**. On the **Applications** page (not shown), click **Add** to add an application and configure as follows:

- **Name** – Set to a descriptive name (e.g., **Test2\_APP**).
- **Enable** – Set to **Yes**. This field determines which application(s) will be executed based on their defined criteria.
- **Type** – Select **VoiceXML**, **CCXML**, or **CCXML/VoiceXML** according to the application type.
- **VoiceXML** and/or **CCXML URL** – Enter the necessary URL(s) to access the VXML and/or CCXML application(s) on the application server. In the sample screen below, the Experience Portal test application on a single server is referenced.

- **Speech Servers ASR and TTS** – Select the appropriate ASR and/or TTS servers as necessary.
- **Application Launch** – Set to **Inbound**.
- **Called Number** – Enter the number to match against an inbound SIP INVITE message and click **Add**. In the sample configuration illustrated in these Application Notes, the dialed DID number 5851239738 provided by Frontier Communications was used. Repeat to define additional called party numbers as needed. Inbound calls with these called party numbers will be handled by the application defined in this section.

**AVAYA**

Avaya Aura® Experience Portal 7.2.2 (ExperiencePortal)

Expand All | Collapse All

**Change Application**

Use this page to change the configuration of an application.

Name: Test2\_APP

Enable:  Yes  No

Type: CCXML

Reserved SIP Calls:  None  Minimum  Maximum

Requested: 5

URI

Single  Fail Over  Load Balance

CCXML URL: http://10.64.101.252/mpp/misc/avptestapp/root.cxmll **Verify**

Mutual Certificate Authentication:  Yes  No

Basic Authentication:  Yes  No

**ASR Speech Servers**

Engine Types: <None>

Selected Engine Types: Nuance

**Nuance**

Languages: <None>

Selected Languages: English(USA) en-US

Resources: Acquire on call start and retain

N Best List Length:

Speech Complete Timeout: 0 milliseconds

Speech Incomplete Timeout: milliseconds

Vendor Parameters:

**TTS Speech Servers**

**Application Launch**

Inbound  Inbound Default  Outbound

Number  Number Range  URI

Called Number: 5851239738 **Add**

**Remove**

**Speech Parameters**

**Reporting Parameters**

**Advanced Parameters**

**Save** **Apply** **Cancel** **Help**

## 6.6. MPP Servers and VoIP Settings

This section illustrates the procedure for viewing or changing the MPP Settings. In the sample configuration, the MPP Server is co-resident on a single server with the Experience Portal Management server (EPM).

**Step 1** - In the left pane, navigate to **System Configuration** → **MPP Servers** and the following screen is displayed. Click **Add**.

The screenshot shows the Avaya Aura Experience Portal 7.2.0 interface. The left navigation pane is expanded to 'System Configuration' > 'MPP Servers'. The main content area displays the 'MPP Servers' configuration page. At the top, it says 'You are here: Home > System Configuration > MPP Servers'. Below this is a table with columns: Name, Host Address, Network Address (VoIP), Network Address (MRCP), Network Address (AppSvr), Maximum Simultaneous Calls, and Trace Level. A single row is shown with the name 'MPP', Host Address '10.64.101.252', and other fields set to '<Default>'. Below the table are 'Add' and 'Delete' buttons. At the bottom, there are tabs for 'MPP Settings', 'Browser Settings', 'Video Settings', 'VoIP Settings', and 'Help'.

**Step 2** - Enter any descriptive name in the **Name** field (e.g., **MPP**) and the IP address of the MPP server in the **Host Address** field and click **Continue** (not shown). Note that the Host Address used is the same IP address assigned to Experience Portal.

**Step 3** - The certificate page will open. Check the **Trust this certificate** box (not shown). Once complete, click **Save**.

The screenshot shows the 'Change MPP Server' configuration page. The left navigation pane is expanded to 'System Configuration' > 'MPP Servers'. The main content area displays the 'Change MPP Server' configuration page. At the top, it says 'You are here: Home > System Configuration > MPP Servers > Change MPP Server'. Below this is a form with fields for Name, Host Address, Network Address (VoIP), Network Address (MRCP), Network Address (AppSvr), Maximum Simultaneous Calls, and Restart Automatically. The 'MPP Certificate' section is expanded, showing details for the certificate, including Owner, Issuer, Serial Number, Signature Algorithm, Valid from, Valid to, Certificate Fingerprints, and Subject. At the bottom, there are 'Save', 'Apply', 'Cancel', and 'Help' buttons.

**Step 4** - Click **VoIP Settings** tab on the screen displayed in **Step 1**, and the following screen is displayed.

- In the Port Ranges section, default ports were used.

**AVAYA** Welcome, epadmin  
Last logged in Jan 29, 2019 at 11:55:28 AM PST

Avaya Aura® Experience Portal 7.2.0 (ExperiencePortal) Home Help Logoff

Expand All | Collapse All You are here: Home > System Configuration > MPP Servers > VoIP Settings

### VoIP Settings

Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols such as H.323 and Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network. Note that if you make any changes to this page, you must restart all MPPs.

**Port Ranges**

	Low	High
UDP:	11000	30999
TCP:	31000	33499
MRCP:	34000	36499
H.323 Station:	37000	39499

**RTCP Monitor Settings**

Host Address:

Port:

**VoIP Audio Formats**

MPP Native Format:

**Codecs**

QoS Parameters ▶

Out of Service Threshold (% of VoIP Resources) ▶

Call Progress ▶

Miscellaneous ▶

**Save Apply Cancel Help**

- In the Codecs section set:
  - Set **Packet Time** to **20**.
  - Verify Codecs **G711uLaw** and **G729** are enabled (check marks). Set the **Offer** and Answer **Order** as shown. In the sample configuration **G711uLaw** is the preferred codec, with **Order 1**, followed by **G729**, with **Order 2**.
  - On the codec Offer set **G729 Discontinuous Transmission** to **No** (for G.729A).
- Use default values for all other fields.

**Step 5** - Click on **Save** (not shown).

The screenshot displays the Avaya Aura Experience Portal 7.2.2 (ExperiencePortal) interface. The user is logged in as 'epadmin' and is viewing the 'VoIP Settings' page under 'System Configuration > MPP Servers > VoIP Settings'. The page includes a navigation menu on the left with categories like User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration, Security, Reports, and Multi-Media Configuration. The 'System Configuration' menu item is highlighted. The main content area shows the 'VoIP Settings' configuration page. The 'Codecs' section is highlighted with a red box and contains two sub-sections: 'Offer' and 'Answer'. The 'Offer' section has a table with columns 'Enable', 'Codec', and 'Order'. G711uLaw is checked and ordered 1, G729 is checked and ordered 2, and G711aLaw is unchecked. The 'Answer' section has a similar table with G711uLaw checked and ordered 1, G711aLaw unchecked, and G729 checked and ordered 2. Below the tables, the 'Packet Time' is set to 20 milliseconds, and 'G729 Discontinuous Transmission' is set to 'No'. Other sections like 'Port Ranges', 'RTCP Monitor Settings', and 'VoIP Audio Formats' are also visible but not highlighted.

## 6.7. Configuring RFC2833 Event Value Offered by Experience Portal

The configuration change example noted in this section was not required for any of the call flows illustrated in these Application Notes. For incoming calls from Frontier Communications to Experience Portal, Frontier Communications specifies the value 101 for the RFC2833 telephone-events that signal DTMF digits entered by the user. When Experience Portal answers, the SDP from Experience Portal matches this Frontier Communications offered value.

When Experience Portal sends an INVITE with SDP as part of an INVITE-based transfer (e.g., bridged transfer), Experience Portal offers the SDP. By default, Experience Portal specifies the value 127 for the RFC2833 telephone-events. Optionally, the value that is offered by Experience Portal can be changed, and this section outlines the procedure that can be performed by an Avaya authorized representative.

- Access Experience Portal via the command line interface.
- Navigate to the following directory: /opt/Avaya/ ExperiencePortal/MPP/config
- Edit the file mppconfig.xml.
- Search for the parameter “mpp.sip.rfc2833.payload”. If there is no such parameter specified add a line such as the following to the file, where the value 101 is the value to be used for the RFC2833 events. If the parameter is already specified in the file, simply edit the value assigned to the parameter.  
`<parameter name="mpp.sip.rfc2833.payload">101</parameter>`
- In the verification of these Application Notes, the line was added directly above the line where the sip.session.expires parameter is configured.

After saving the file with the change, restart the MPP server for the change to take effect. As shown below, the MPP may be restarted using the **Restart** button available via the Experience Portal GUI at **System Management → MPP Manager**.

Note that the **State** column shows when the MPP is running after the restart.

The screenshot shows the Avaya Experience Portal MPP Manager interface. The page title is "MPP Manager (Feb 5, 2019 2:34:27 PM PST)". Below the title, there is a table displaying the current state of each MPP in the system. The table has columns for Server Name, Mode, State, Config, Auto Restart, Restart Schedule, and Active Calls. The MPP is currently in the "Running" state. Below the table, there are sections for "State Commands" (Start, Stop, Restart, Reboot, Halt, Cancel) and "Restart/Reboot Options" (One server at a time, All servers). The "Restart" button is highlighted in red.

Server Name	Mode	State	Config	Auto Restart	Restart Schedule	Active Calls
MPP	Online	Running	OK	Yes	No	0

## 7. 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.
- Adaptation module to perform header manipulations.
- SIP Entities corresponding to Communication Manager, Session Manager, Experience Portal and the Avaya SBCE.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Routing Policies, which control call routing between the SIP Entities.
- Dial Patterns, which govern to which SIP Entity a call is routed.

The following sections assume that the initial configuration of Session Manager and System Manager has already been completed, and that network connectivity exists between System Manager and Session Manager.

## 7.1. System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and click on **Log On** (not shown). The screen shown below is then displayed; under **elements** select **Routing** → **Domains**.

The screenshot displays the Avaya System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The 'Elements' menu is expanded, showing a list of system components. 'Routing' is highlighted in red, and its sub-menu is open, with 'Domains' also highlighted in red. Other visible components in the 'Elements' menu include Avaya Breeze™, Communication Manager, Communication Server 1000, Conferencing, Device Services, Equinox Conference, IP Office, Media Server, Meeting Exchange, Messaging, Presence, Session Manager, Web Gateway, and Work Assignment. The main dashboard area contains several widgets: 'System Resource Utilization' (a bar chart showing utilization for 'opt', 'var', and 'emdata'), 'Alarms', 'Notifications' (showing 'No data'), 'Application State' (listing License Status as Active, Deployment Type as VMware, Multi-Tenancy as DISABLED, OOBM State as DISABLED, and Hardening Mode as Standard), 'Information' (a table of system elements and their sync status), and 'Shortcuts' (a placeholder for drag-and-drop shortcuts). The 'Information' table is as follows:

Elements	GNRL	Sync Status
CM	1	■
Messaging	1	■
Session Manager	1	■
System Manager	1	■
UCM Applications	16	■

Below the table, the 'Current Usage' section shows two bars: '6/250000 USERS' and '1/50 SIMULTANEOUS ADMINISTRATIVE LOGINS'.

The navigation tree displayed in the left pane below will be referenced in subsequent sections to navigate to items requiring configuration. Most items discussed in this section will be located under the **Routing** link shown below.

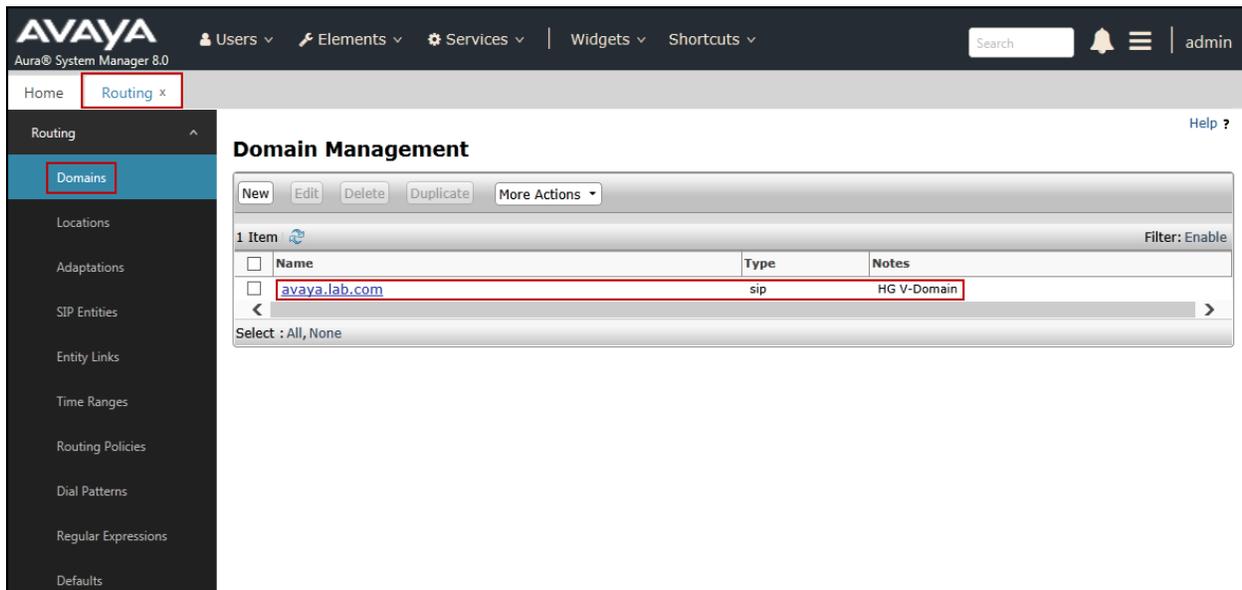
The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, user information (Users), and various menu options (Elements, Services, Widgets, Shortcuts). A search bar and a user profile (admin) are also visible. The main content area is titled "Domain Management" and features a table with one item: "avaya.lab.com" of type "sip" with the note "HG V-Domain". The left navigation pane is expanded to show the "Routing" section, which includes sub-items like Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The "Routing" link and its sub-items are highlighted with a red box.

## 7.2. SIP Domain

Create an entry for each SIP domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this was the enterprise domain, **avaya.lab.com**. Navigate to **Routing → Domains** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).
- Click **Commit** to save.

The screen below shows the entry for the enterprise domain.



The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left-hand navigation pane is open to 'Routing', with 'Domains' selected. The main content area is titled 'Domain Management' and shows a table with one item:

Name	Type	Notes
avaya.lab.com	sip	HG V-Domain

## 7.3. Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management, call admission control and location-based routing. To add a location, navigate to **Routing → Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the **General** section, enter the following values:

- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).
- Click **Commit** to save.

The following screen shows the location details for the location named *Session Manager*. Later, this location will be assigned to the SIP Entity corresponding to Session Manager. Other location parameters (not shown) retained the default values.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.0', and menu items for Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile 'admin' are also visible. The left sidebar shows a navigation tree with 'Routing' selected and 'Locations' highlighted. The main content area is titled 'Location Details' and contains the following fields:

- General:**
  - \* Name: Session Manager
  - Notes: VMware Session Manager
- Dial Plan Transparency in Survivable Mode:**
  - Enabled:
  - Listed Directory Number:
  - Associated CM SIP Entity:
- Overall Managed Bandwidth:**
  - Managed Bandwidth Units: Kbit/sec
  - Total Bandwidth:
  - Multimedia Bandwidth:

Buttons for 'Commit' and 'Cancel' are located at the top right of the form area.

The following screen shows the location details for the location named *Communication Manager*. Later, this location will be assigned to the SIP Entity corresponding to Communication Manager. Other location parameters (not shown) retained the default values.

The screenshot displays the Avaya Aura System Manager 8.0 interface, similar to the previous one. The top navigation bar and left sidebar are identical. The main content area is titled 'Location Details' and contains the following fields:

- General:**
  - \* Name: Communication Manager
  - Notes: VMware Communication Manager
- Dial Plan Transparency in Survivable Mode:**
  - Enabled:
  - Listed Directory Number:
  - Associated CM SIP Entity:
- Overall Managed Bandwidth:**
  - Managed Bandwidth Units: Kbit/sec
  - Total Bandwidth:
  - Multimedia Bandwidth:

Buttons for 'Commit' and 'Cancel' are located at the top right of the form area.

The following screen shows the location details for the location named *Avaya SBCE*. Later, this location will be assigned to the SIP Entity corresponding to the Avaya SBCE. Other location parameters (not shown) retained the default values.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar shows a menu with 'Locations' highlighted. The main content area is titled 'Location Details' and contains the following sections:

- General:** The 'Name' field is set to 'Avaya SBCE' and is highlighted with a red box. The 'Notes' field contains 'VMware Avaya SBCE'.
- Dial Plan Transparency in Survivable Mode:** The 'Enabled' checkbox is unchecked. The 'Listed Directory Number' and 'Associated CM SIP Entity' fields are empty.
- Overall Managed Bandwidth:** The 'Managed Bandwidth Units' dropdown is set to 'Kbit/sec'. The 'Total Bandwidth' and 'Multimedia Bandwidth' fields are empty.

Buttons for 'Commit' and 'Cancel' are visible in the top right corner of the form area.

The following screen shows the location details for the location named *Lab Others*. Later, this location will be assigned to the SIP Entity corresponding to the Experience Portal. Other location parameters (not shown) retained the default values.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar shows a menu with 'Locations' highlighted. The main content area is titled 'Location Details' and contains the following sections:

- General:** The 'Name' field is set to 'Lab Others' and is highlighted with a red box. The 'Notes' field contains 'VMware Lab others'.
- Dial Plan Transparency in Survivable Mode:** The 'Enabled' checkbox is unchecked. The 'Listed Directory Number' and 'Associated CM SIP Entity' fields are empty.
- Overall Managed Bandwidth:** The 'Managed Bandwidth Units' dropdown is set to 'Kbit/sec'. The 'Total Bandwidth' and 'Multimedia Bandwidth' fields are empty.

Buttons for 'Commit' and 'Cancel' are visible in the top right corner of the form area.

## 7.4. Adaptations

In order to improve interoperability with third party elements, Session Manager 8.0 incorporates the ability to use Adaptation modules to remove specific headers that are either Avaya proprietary or deemed excessive/unnecessary for non-Avaya elements.

For the compliance test, an Adaptation named *CM\_Outbound\_Header\_Removal* was created to block the following headers from outbound messages, before they were forwarded to the Avaya SBCE: AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-ID, P-Charging-Vector and P-Location. These headers contain private information from the enterprise, which should not be propagated outside of the enterprise boundaries. They also add unnecessary size to outbound messages, while they have no significance to the service provider.

Navigate to **Routing** → **Adaptations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Adaptation Name:** Enter an appropriate name.
- **Module Name:** Select the *DigitConversionAdapter* option.
- **Module Parameter Type:** Select *Name-Value Parameter*.

Click **Add** to add the name and value parameters, as follows:

- **Name:** Enter *eRHdrs*. This parameter will remove the specified headers from messages in the egress direction.
- **Value:** Enter “*Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-Id, P-Location, Endpoint-View*”
- Click **Commit** to save.

The screen below shows the adaptation created for the compliance test. This adaptation will later be applied to the SIP Entity corresponding to the Avaya SBCE. All other fields were left at their default values.

**AVAYA**  
Aura® System Manager 8.0

Users | Elements | Services | Widgets | Shortcuts | Search | admin

Home | Routing x | Routing x

Routing

Domains

Locations

Conditions

Adaptations

Adaptations

Regular Expression...

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

**Adaptation Details** [Commit] [Cancel] Help ?

**General**

\* Adaptation Name: CM\_Outbound\_Header\_Removal

\* Module Name: DigitConversionAdapter

Module Parameter Type: Name-Value Parameter

Name	Value
eRHdrs	"Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-id,"

Select : All, None

Egress URI Parameters: \_\_\_\_\_

Notes: \_\_\_\_\_

**Digit Conversion for Incoming Calls to SM**

Add Remove

0 Items Filter: Enable

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
------------------	-----	-----	---------------	---------------	---------------	-------------------	-----------------	-------

## 7.5. SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Communication Manager, Avaya SBCE and the Experience Portal. Navigate to **Routing** → **SIP Entities** in the left navigation pane and click on the **New** button in the right pane (not shown). In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling (see **Figure 1**).
- **Type:** Select *Session Manager* for Session Manager, *CM* for Communication Manager, *SIP Trunk* (or *Other*) for the Avaya SBCE and *Voice Portal* for the Experience Portal.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If Adaptations were to be created, here is where they would be applied to the entity.
- **Location:** Select the location that applies to the SIP Entity being created, defined in **Section 7.3**.
- **Time Zone:** Select the time zone for the location above.
- Click **Commit** to save.

The following screen shows the addition of the *Session Manager* SIP Entity for Session Manager. The IP address of the Session Manager Security Module is entered in the **FQDN or IP Address** field.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The left navigation pane shows 'Routing' selected, with 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and is divided into 'General' and 'Monitoring' sections. The 'General' section contains the following fields:

- Name:** Session Manager
- IP Address:** 10.64.101.249
- SIP FQDN:** (empty)
- Type:** Session Manager
- Notes:** VMware Session Manager
- Location:** Session Manager
- Outbound Proxy:** (empty)
- Time Zone:** America/New\_York
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)

The 'Monitoring' section contains the following fields:

- SIP Link Monitoring:** Use Session Manager Configuration
- CRLF Keep Alive Monitoring:** CRLF Monitoring Disabled

Buttons for 'Commit' and 'Cancel' are visible at the top right of the form area.

The following screen shows the addition of the *Communication Manager Trunk 2* SIP Entity for Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, the creation of a separate SIP entity for Communication Manager is required. This SIP Entity should be different than the one created during the Session Manager installation, used by all other enterprise SIP traffic. The **FQDN or IP Address** field is set to the IP address of the “**procr**” interface in Communication Manager, as seen in **Section 5.3**. Select the location that applies to the SIP Entity being created, defined in **Section 7.3**. Select the **Time Zone**.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 8.0', and various menu items like 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are also visible. The main content area is titled 'SIP Entity Details' and is currently on the 'General' tab. The left sidebar shows a navigation menu with 'SIP Entities' highlighted. The form fields are as follows:

- Name:** Communication Manager Trunk 2
- FQDN or IP Address:** 10.64.101.241
- Type:** CM
- Notes:** Used for SP Testing
- Adaptation:** (empty dropdown)
- Location:** Communication Manager
- Time Zone:** America/New\_York
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty text field)
- Securable:**
- Call Detail Recording:** none

The following screen shows the addition of the *Avaya SBCE* SIP Entity for the Avaya SBCE:

- The **FQDN or IP Address** field is set to the IP address of the SBC private network interface (see **Figure 1**).
- On the **Adaptation** field, the adaptation module *CM\_Outbound\_Header\_Removal* previously defined in **Section 7.4** was selected.
- Select the location that applies to the SIP Entity being created, defined in **Section 7.3**.
- Select the **Time Zone**.

The screenshot displays the 'SIP Entity Details' configuration page in the Avaya Aura System Manager 8.0 interface. The left sidebar shows the 'Routing' menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and includes a 'General' tab. The configuration fields are as follows:

- Name:** Avaya SBCE
- FQDN or IP Address:** 10.64.101.243
- Type:** SIP Trunk
- Notes:** VMware Avaya SBCE
- Adaptation:** CM\_Outbound\_Header\_Removal
- Location:** Avaya SBCE
- Time Zone:** America/New\_York
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty field)
- Securable:**
- Call Detail Recording:** none

The following screen shows the addition of the *Avaya Experience Portal* SIP Entity:

- The **FQDN or IP Address** field is set to the IP address of the Experience Portal (see **Figure 1**).
- Select the location that applies to the SIP Entity being created, defined in **Section 7.3**.
- Select the **Time Zone**.

The screenshot displays the 'SIP Entity Details' configuration page in the Avaya Aura System Manager 8.0 interface. The left sidebar shows the 'Routing' menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and includes a 'General' tab. The configuration fields are as follows:

- Name:** Avaya Experience Portal
- FQDN or IP Address:** 10.64.101.252
- Type:** Voice Portal
- Notes:** SIP Trunk to Avaya Experience Poi
- Adaptation:** (empty dropdown)
- Location:** Lab Others
- Time Zone:** America/Fortaleza
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty field)
- Securable:**
- Call Detail Recording:** none

## 7.6. Entity Links

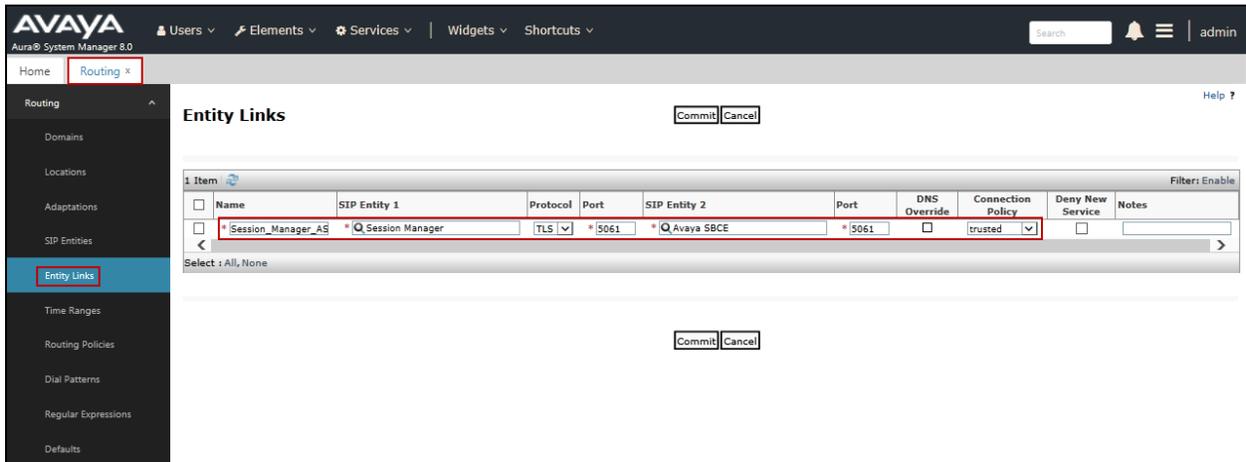
A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Three Entity Links were created; an entity link to Communication Manager for use only by service provider traffic, an entity link to the Avaya SBCE and an entity link to Experience Portal. To add an Entity Link, navigate to **Routing** → **Entity Links** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager from the drop-down menu (**Section 7.5**).
- **Protocol:** Select the transport protocol used for this link (**Section 5.6**).
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end (**Section 5.6**).
- **SIP Entity 2:** Select the name of the other system from the drop-down menu (**Section 7.5**).
- **Port:** Port number on which the other system receives SIP requests from Session Manager (**Section 5.6**).
- **Connection Policy:** Select **Trusted** to allow calls from the associated SIP Entity.
- Click **Commit** to save.

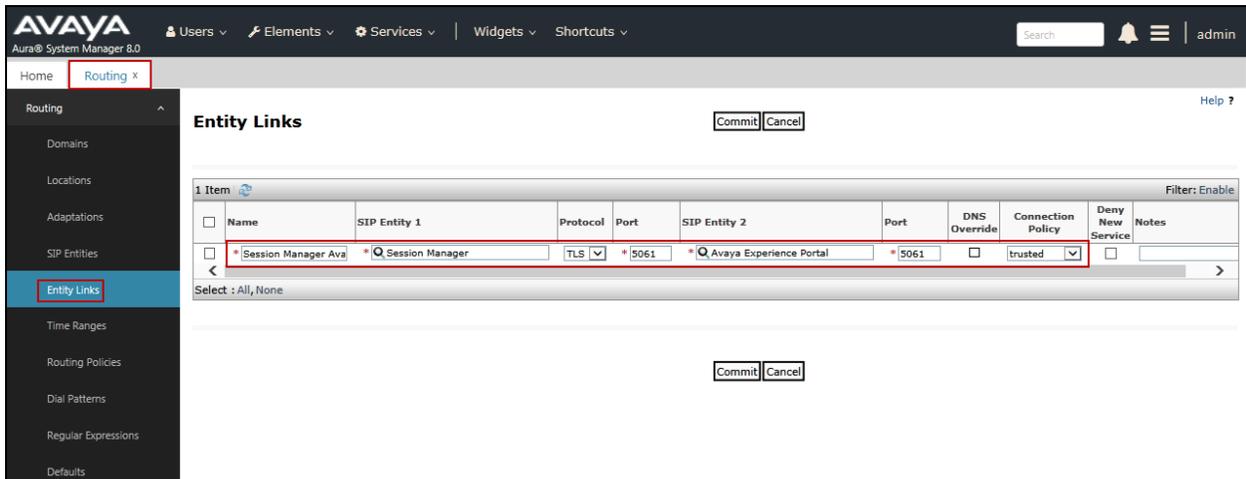
The screen below shows the Entity Link to Communication Manager. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.6**. *TLS* transport and port *5071* were used.

Entity Links										
Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy	Deny New Service	Notes	Filters: Enable
<input type="checkbox"/>	*Session_Manager_Ch	*Q Session Manager	TLS	*5071	*Q Communication Manager Trunk 2	*5071	<input type="checkbox"/>	trusted	<input type="checkbox"/>	

The Entity Link to the Avaya SBCE is shown below; *TLS* transport and port *5061* were used.



The Entity Link to the Experience Portal is shown below; *TLS* transport and port *5061* were used.



## 7.7. Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 7.5**. Three routing policies were added; an incoming policy with Communication Manager as the destination, an outbound policy to the Avaya SBCE as the destination, an incoming policy with Experience Portal as the destination. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed:

- In the **General** section, enter a descriptive **Name** and add a brief description under **Notes** (optional).
- In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Choose the appropriate SIP entity to which this routing policy applies (**Section 7.5**) and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page as shown below.
- Use default values for remaining fields.
- Click **Commit** to save.

The following screens show the Routing Policies for Communication Manager, the Avaya SBCE and the Experience Portal.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar shows the 'Routing' menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and contains the following sections:

- General:**
  - \* Name: To CM Trunk 2
  - Disabled:
  - \* Retries: 0
  - Notes: For inbound calls to CM via Trunk
- SIP Entity as Destination:**

Name	FQDN or IP Address	Type	Notes
Communication Manager Trunk 2	10.64.101.241	CM	Used for SP Testing
- Time of Day:**

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7						

The screenshot shows the 'Routing Policy Details' page in Avaya Aura System Manager 8.0. The left navigation pane has 'Routing Policies' selected. The main content area is divided into sections: 'General', 'SIP Entity as Destination', and 'Time of Day'. In the 'General' section, the name is 'Avaya SBCE', it is not disabled, and it has 0 retries. The notes specify 'For outbound calls to SP via ASB'. The 'SIP Entity as Destination' table lists one entry: 'Avaya SBCE' with FQDN '10.64.101.243', Type 'SIP Trunk', and Notes 'VMware Avaya SBCE'. The 'Time of Day' section shows a single time range for 24/7 from 00:00 to 23:59.

The screenshot shows the 'Routing Policy Details' page in Avaya Aura System Manager 8.0. The left navigation pane has 'Routing Policies' selected. The main content area is divided into sections: 'General', 'SIP Entity as Destination', and 'Time of Day'. In the 'General' section, the name is 'To Avaya Experience Portal', it is not disabled, and it has 0 retries. The notes specify 'To Avaya Experience Portal'. The 'SIP Entity as Destination' table lists one entry: 'Avaya Experience Portal' with FQDN '10.64.101.252', Type 'Voice Portal', and Notes 'SIP Trunk to Avaya Experience Portal'. The 'Time of Day' section shows a single time range for 24/7 from 00:00 to 23:59.

## 7.8. Dial Patterns

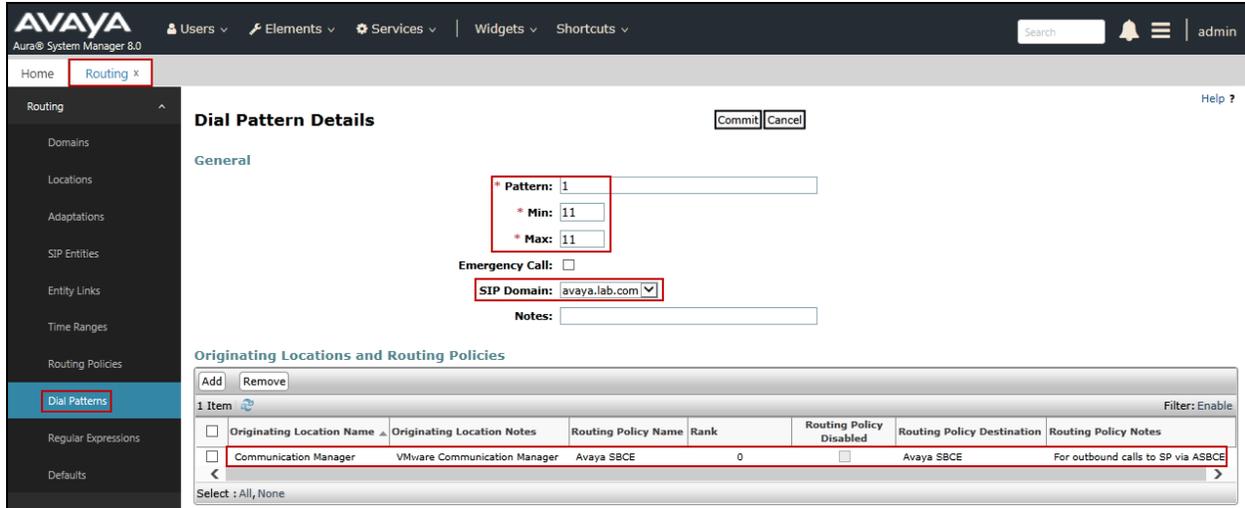
Dial Patterns are needed to route specific calls through Session Manager. For the compliance test, dial patterns were needed to route calls from Communication Manager to the service provider and vice versa. Also, a dial pattern was created to route calls from service provider to Experience Portal. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing** → **Dial Patterns** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

In the **General** section, enter the following values:

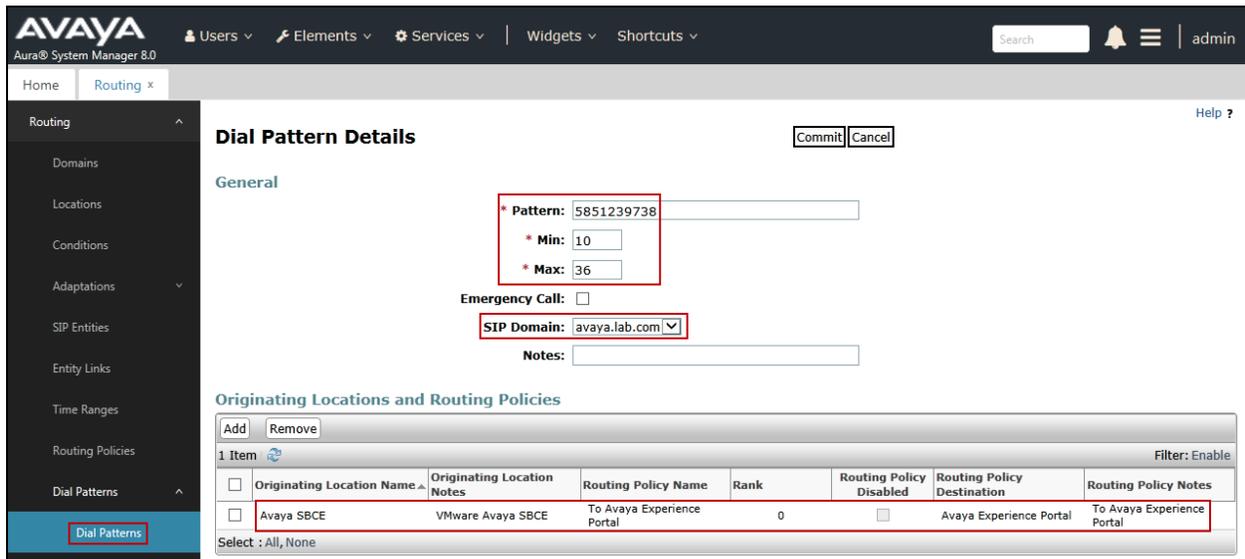
- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call.



The example in this screen shows the 11-digit dialed numbers for outbound calls, beginning with *1*, arriving from the **Communication Manager** location, will use route policy **Avaya SBCE**, which sends the call out to the PSTN via Avaya SBCE and the service provider SIP trunk. The SIP Domain was set to *avaya.lab.com*.



The following screen illustrates an example dial pattern used to verify inbound PSTN calls to Experience Portal. In the sample configuration one of the DID numbers provided by Frontier Communications was used as a test number to route calls from the PSTN to Experience Portal, arriving from location **Avaya SBCE**, used routing policy **To Avaya Experience Portal**. The SIP Domain was set to *avaya.lab.com*.



Repeat the above procedures as needed to define additional dial patterns.

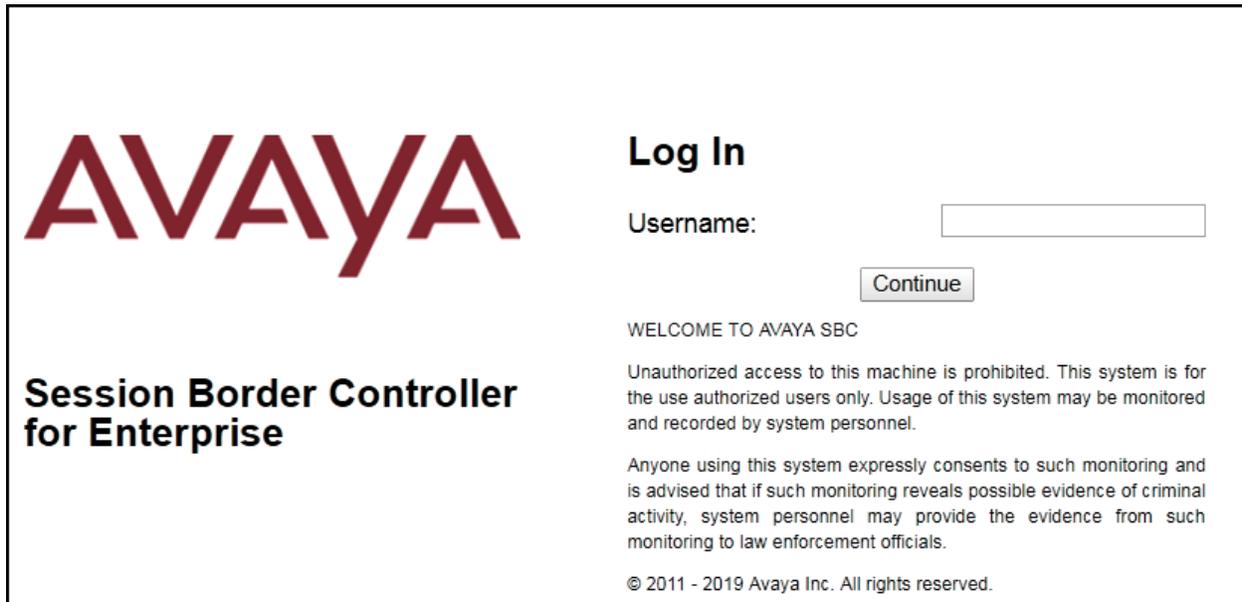
## 8. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Avaya SBCE. It is assumed that the initial installation of the Avaya SBCE, the assignment of the management interface IP Address and license installation have already been completed; hence these tasks are not covered in these Application Notes. For more information on the installation and initial provisioning of the Avaya SBCE consult the Avaya SBCE documentation in the **References** section.

**Note** - The configuration tasks required to support TLS transport for signaling and SRTP for media are beyond the scope of these Application Notes; hence it's not discussed in detail in this document. Consult reference [8] in the **References** section for additional information on this topic.

### 8.1. System Access

Access the Session Border Controller web management interface by using a web browser and entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation. Log in using the appropriate credentials.



The screenshot shows the login interface for the Avaya Session Border Controller for Enterprise. On the left, the Avaya logo is displayed in a dark red color, with the text "Session Border Controller for Enterprise" below it. On the right, the "Log In" section features a "Username:" label, a text input field, and a "Continue" button. Below the login fields, there is a "WELCOME TO AVAYA SBC" message, a warning about unauthorized access, a consent statement, and a copyright notice: "© 2011 - 2019 Avaya Inc. All rights reserved."

Once logged in, on the top left of the screen, under **Device:** select the device being managed, *Avaya\_SBCE* in the sample configuration.

The screenshot shows the Avaya SBCE dashboard for device EMS. The top navigation bar includes 'Device: EMS', 'Alarms 1', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The left sidebar shows 'EMS' and 'Avaya\_SBCE' (highlighted with a red box). The main content area is titled 'Session Border Controller for Enterprise' and features the AVAYA logo. The dashboard includes an 'Information' table, an 'Installed Devices' list, and 'Active Alarms' and 'Incidents' sections.

Information		
System Time	08:13:13 AM MDT	Refresh
Version	8.0.0.0-19-16991	
Build Date	Sat Jan 26 21:58:11 UTC 2019	
License State	OK	
Aggregate Licensing Overages	0	
Peak Licensing Overage Count	0	
Last Logged in at	04/01/2019 08:11:58 MDT	
Failed Login Attempts	0	

Installed Devices
EMS 1
Avaya_SBCE

Active Alarms (past 24 hours): None found.

Incidents (past 24 hours): None found.

The left navigation pane contains the different available menu items used for the configuration of the Avaya SBCE. Verify that the status of the **License State** field is **OK**, indicating that a valid license is present. Contact an authorized Avaya sales representative if a license is needed.

The screenshot shows the Avaya SBCE dashboard for device Avaya\_SBCE. The top navigation bar includes 'Device: Avaya\_SBCE', 'Alarms 1', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The left sidebar shows 'EMS Dashboard' and 'Avaya\_SBCE' (highlighted with a red box). The main content area is titled 'Session Border Controller for Enterprise' and features the AVAYA logo. The dashboard includes an 'Information' table, an 'Installed Devices' list, and 'Active Alarms' and 'Incidents' sections.

Information		
System Time	04:06:22 PM MDT	Refresh
Version	8.0.0.0-19-16991	
Build Date	Sat Jan 26 21:58:11 UTC 2019	
License State	OK	
Aggregate Licensing Overages	0	
Peak Licensing Overage Count	0	
Last Logged in at	03/28/2019 15:55:54 MDT	
Failed Login Attempts	0	

Installed Devices
EMS 1
Avaya_SBCE

Active Alarms (past 24 hours): None found.

Incidents (past 24 hours): Avaya\_SBCE: No Subscriber Flow Matched

## 8.2. Device Management

To view current system information, select **Device Management** on the left navigation pane. In the reference configuration, the device named *Avaya\_SBCE* is shown. The management IP address that was configured during installation is blurred out for security reasons, the current software version is shown. The management IP address needs to be on a subnet separate from the ones used in all other interfaces of the Avaya SBCE, segmented from all VoIP traffic. Verify that the **Status** is *Commissioned*, indicating that the initial installation process of the device has been previously completed, as shown on the screen below.

The screenshot shows the Avaya Session Border Controller for Enterprise (SBCE) web interface. The top navigation bar includes 'Device: Avaya\_SBCE', 'Alarms 1', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Session Border Controller for Enterprise' with the AVAYA logo. The left navigation pane lists various management options, with 'Device Management' highlighted. The main content area is titled 'Device Management' and contains several tabs: 'Devices', 'Updates', 'SSL VPN', 'Licensing', and 'Key Bundles'. The 'Devices' tab is active, showing a table with the following data:

Device Name	Management IP	Version	Status						
Avaya_SBCE	[Blurred]	8.0.0.0-19-16991	Commissioned	Reboot	Shutdown	Restart Application	View	Edit	Uninstall

To view the network configuration assigned to the Avaya SBCE, click **View** on the screen above. The **System Information** window is displayed, containing the current device configuration and network settings.

**System Information: Avaya\_SBCE** X

**General Configuration**

Appliance Name	Avaya_SBCE
Box Type	SIP
Deployment Mode	Proxy

**Device Configuration**

HA Mode	No
Two Bypass Mode	No

**License Allocation**

Standard Sessions <small>Requested: 2000</small>	2000
Advanced Sessions <small>Requested: 2000</small>	2000
Scopia Video Sessions <small>Requested: 500</small>	500
CES Sessions <small>Requested: 0</small>	0
Transcoding Sessions <small>Requested: 0</small>	0
CLID	---
Encryption <small>Available: Yes</small>	<input checked="" type="checkbox"/>

**Network Configuration**

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.64.101.243	10.64.101.243	255.255.255.0	10.64.101.1	A1
				A1
				A1
				B1
				B1
10.10.80.51	10.10.80.51	255.255.255.128	10.10.80.1	B1

**DNS Configuration**

Primary DNS	8.8.8.8
Secondary DNS	7.7.7.7
DNS Location	DMZ
DNS Client IP	10.10.80.51

**Management IP(s)**

IP #1 (IPv4)

The highlighted IP addresses in the **System Information** screen shown above are the ones used for the SIP trunk to Frontier Communications and are the ones relevant to these Application Notes. Other IP addresses assigned to the Avaya SBCE **A1** and **B1** interfaces are used to support remote workers and other SIP trunks, and they are not discussed in this document. Also note that for security purposes, any public IP addresses used during the compliance test have been masked in this document.

In the reference configuration, the private interface of the Avaya SBCE (10.64.101.243) was used to connect to the enterprise network, while its public interface (10.10.80.51) was used to connect to the public network. See **Figure 1**.

On the **License Allocation** area of the **System Information**, verify that the number of **Standard Sessions** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise. The number of sessions and encryption features are primarily controlled by the license file installed.

### 8.3. TLS Management

Transport Layer Security (TLS) is a standard protocol that is used extensively to provide a secure channel by encrypting communications over IP networks. It enables clients to authenticate servers or, optionally, servers to authenticate clients. UC-Sec security products utilize TLS primarily to facilitate secure communications with remote servers.

It is assumed that generation and installation of certificates and the creation of TLS Profiles on the Avaya SBCE have been previously completed, as it's not discussed in this document. Refer to item [8] in **Section 12**.

### 8.4. Network Management

The network configuration parameters should have been previously specified during installation of the Avaya SBCE. In the event that changes need to be made to the network configuration, they can be entered here.

Select **Network Management** from the **Network & Flows** on the left-side menu. On the **Networks** tab, verify or enter the network information as needed.

Note that in the configuration used during the compliance test, the IP addresses assigned to the private (**10.64.101.243**) and public (**10.10.80.51**) sides of the Avaya SBCE are the ones relevant to these Application Notes.

Device: Avaya\_SBCE ▾ Alarms 1 Incidents Status ▾ Logs ▾ Diagnostics Users Settings ▾ Help ▾ Log Out

## Session Border Controller for Enterprise

AVAYA

EMS Dashboard  
 Device Management  
 Backup/Restore  
 ▸ System Parameters  
 ▸ Configuration Profiles  
 ▸ Services  
 ▸ Domain Policies  
 ▸ TLS Management  
 ▾ Network & Flows  
   **Network Management**  
 Media Interface  
 Signaling Interface

### Network Management

Interfaces Networks Add

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	Edit	Delete
Network_A1	10.64.101.1	255.255.255.0	A1	10.64.101.243		
Network_B1	10.10.80.1	255.255.255.128	B1	10.10.80.51		

On the **Interfaces** tab, verify the **Administrative Status** is **Enabled** for the **A1** and **B1** interfaces. Click the buttons under the **Status** column if necessary to enable the interfaces.

Device: Avaya\_SBCE ▾ Alarms 1 Incidents Status ▾ Logs ▾ Diagnostics Users Settings ▾ Help ▾ Log Out

## Session Border Controller for Enterprise

AVAYA

EMS Dashboard  
 Device Management  
 Backup/Restore  
 ▸ System Parameters  
 ▸ Configuration Profiles  
 ▸ Services  
 ▸ Domain Policies  
 ▸ TLS Management  
 ▾ Network & Flows  
   **Network Management**  
 Media Interface  
 Signaling Interface

### Network Management

Interfaces Networks Add VLAN

Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Disabled

## 8.5. Media Interfaces

Media Interfaces were created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address, and one of the ports in this range as the listening IP address and port in which it will accept media from the Call Server or the trunk server.

To add the Media Interface in the enterprise direction, select **Media Interface** from the **Network & Flows** menu on the left-hand side, click the **Add** button (not shown).

- On the **Add Media Interface** screen, enter an appropriate **Name** for the Media Interface.

- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- The **Port Range** was left at the default values of **35000-40000**.
- Click **Finish**.

The screenshot shows a dialog box titled "Add Media Interface" with a close button (X) in the top right corner. The dialog contains three rows of input fields:
 

- Name:** A text box containing "Private\_med".
- IP Address:** A dropdown menu showing "Network\_A1 (A1, VLAN 0)" selected, with a sub-dropdown showing "10.64.101.243".
- Port Range:** Two text boxes containing "35000" and "40000" separated by a hyphen.

 A red rectangular box highlights the Name, IP Address, and Port Range fields. At the bottom center of the dialog is a "Finish" button.

A Media Interface facing the public side was similarly created with the name **Public\_med**, as shown below.

- Under **IP Address**, the network and IP address to be associated with this interface was selected.
- The **Port Range** was left at the default values.
- Click **Finish**.

The screenshot shows a dialog box titled "Add Media Interface" with a close button (X) in the top right corner. The dialog contains three rows of input fields:
 

- Name:** A text box containing "Public\_med".
- IP Address:** A dropdown menu showing "Network\_B1 (B1, VLAN 0)" selected, with a sub-dropdown showing "10.10.80.51".
- Port Range:** Two text boxes containing "35000" and "40000" separated by a hyphen.

 A red rectangular box highlights the Name, IP Address, and Port Range fields. At the bottom center of the dialog is a "Finish" button.

## 8.6. Signaling Interfaces

Signaling Interfaces are created to specify the IP addresses and ports in which the Avaya SBCE will listen for signaling traffic in the connected networks.

To add the Signaling Interface in the enterprise direction, select **Signaling Interface** from the **Network & Flows** menu on the left-hand side, click the **Add** button (not shown).

- On the **Add Signaling Interface** screen, enter an appropriate **Name** for the interface.
- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- Enter **5061** for **TLS Port**, since TLS port 5061 is used to listen for signaling traffic from Session Manager in the sample configuration, as defined in **Section 7.6**.
- Select a **TLS Profile**.
- Click **Finish**.

The screenshot shows the 'Add Signaling Interface' configuration window. The form is titled 'Add Signaling Interface' and has a close button 'X' in the top right corner. The form contains the following fields and values:

- Name:** Private\_sig
- IP Address:** Network\_A1 (A1, VLAN 0) (dropdown menu), 10.64.101.243 (dropdown menu)
- TCP Port:** (text box), Leave blank to disable
- UDP Port:** (text box), Leave blank to disable
- TLS Port:** 5061 (text box), Leave blank to disable
- TLS Profile:** New\_ServiceProvider\_Server\_TLS (dropdown menu)
- Enable Shared Control:**
- Shared Control Port:** (text box)

A 'Finish' button is located at the bottom center of the form.

A second Signaling Interface with the name **Public\_sig** was similarly created in the service provider's direction.

- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- Enter **5060** for **UDP Port**, since UDP port 5060 is used to listen for signaling traffic from Frontier Communications in the sample configuration.
- Click **Finish**.

**Add Signaling Interface** X

Name: Public\_sig

IP Address: Network\_B1 (B1. VLAN 0) | 10.10.80.51

TCP Port: Leave blank to disable

UDP Port: 5060 | Leave blank to disable

TLS Port: Leave blank to disable

TLS Profile: None

Enable Shared Control:

Shared Control Port:

Finish

## 8.7. Server Interworking

Interworking Profile features are configured to facilitate the interoperability between the enterprise SIP-enabled solution (Call Server) and the SIP trunk service provider (Trunk Server).

### 8.7.1. Server Interworking Profile – Enterprise

Interworking profiles can be created by cloning one of the pre-defined default profiles, or by adding a new profile. To configure the interworking profile in the enterprise direction, select **Global Profiles** → **Server Interworking** on the left navigation pane. Under **Interworking Profiles**, select *avaya-ru* from the list of pre-defined profiles. Click **Clone**.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Alarms 1', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header is 'Session Border Controller for Enterprise' with the AVAYA logo. The left navigation pane is expanded to 'Global Profiles' > 'Server Interworking'. The main content area is titled 'Interworking Profiles: avaya-ru' and features a list of profiles on the left, including 'cs2100', 'avaya-ru', 'OCS-Edge-Server', 'cisco-ccm', 'cups', 'OCS-FrontEnd...', 'Avaya-SM', 'SP-General', 'Avaya-IPO', 'Avaya-CS1000', and 'Avaya-CM'. A 'Clone' button is visible in the top right. A warning banner states: 'It is not recommended to edit the defaults. Try cloning or adding a new profile instead.' Below this, there are tabs for 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'General' tab is active, showing a table of settings:

General	
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
Send Hold	No
Delayed Offer	No
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No

- Enter a descriptive name for the cloned profile.
- Click **Finish**.

The 'Clone Profile' dialog box is shown with the following fields:

- Profile Name: avaya-ru
- Clone Name: Avaya-SM (with a red box around the input field)

A 'Finish' button is located at the bottom of the dialog.

Click **Edit** on the newly cloned *Avaya-SM* interworking profile:

- On the **General** tab, check *T.38 Support*.
- Leave remaining fields with default values.
- Click **Finish**.

The screenshot shows a dialog box titled "Editing Profile: Avaya-SM" with a close button (X) in the top right corner. The "General" tab is selected. The following table represents the configuration options visible in the dialog:

Field	Value / Option
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
URI Group	None (dropdown menu)
Send Hold	<input type="checkbox"/>
Delayed Offer	<input checked="" type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
Re-Invite Handling	<input type="checkbox"/>
Prack Handling	<input type="checkbox"/>
Allow 18X SDP	<input type="checkbox"/>
<b>T.38 Support</b>	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

At the bottom of the dialog, there is a "Finish" button.

The **Timers**, **Privacy**, **URI Manipulation** and **Header Manipulation** tabs contain no entries.

The **Advanced** tab settings are shown on the screen below:

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Alarms 3', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Session Border Controller for Enterprise' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with the following items: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles (selected), Domain DoS, Server, Server Interworking (selected), Media Forking, Routing, Server Configuration, Topology Hiding, Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, FGDN Groups, and Reverse Proxy.

The main content area is titled 'Interworking Profiles: Avaya-SM'. It features an 'Add' button and three action buttons: 'Rename', 'Clone', and 'Delete'. Below this is a blue bar with the text 'Click here to add a description.'.

The configuration is shown in the 'Advanced' tab, which is highlighted with a red box. The configuration table is as follows:

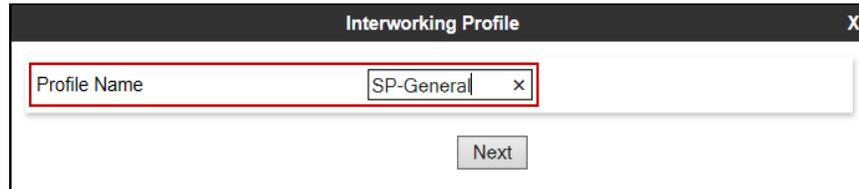
General	Timers	Privacy	URI Manipulation	Header Manipulation	Advanced
Record Routes			Both Sides		
Include End Point IP for Context Lookup		Yes			
Extensions		Avaya			
Diversion Manipulation		No			
Has Remote SBC		Yes			
Route Response on Via Port		No			
Relay INVITE Replace for SIPREC		No			
MOBX Re-INVITE Handling		No			
<b>DTMF</b>					
DTMF Support		None			

An 'Edit' button is located at the bottom right of the configuration area.

## 8.7.2. Server Interworking Profile – Service Provider

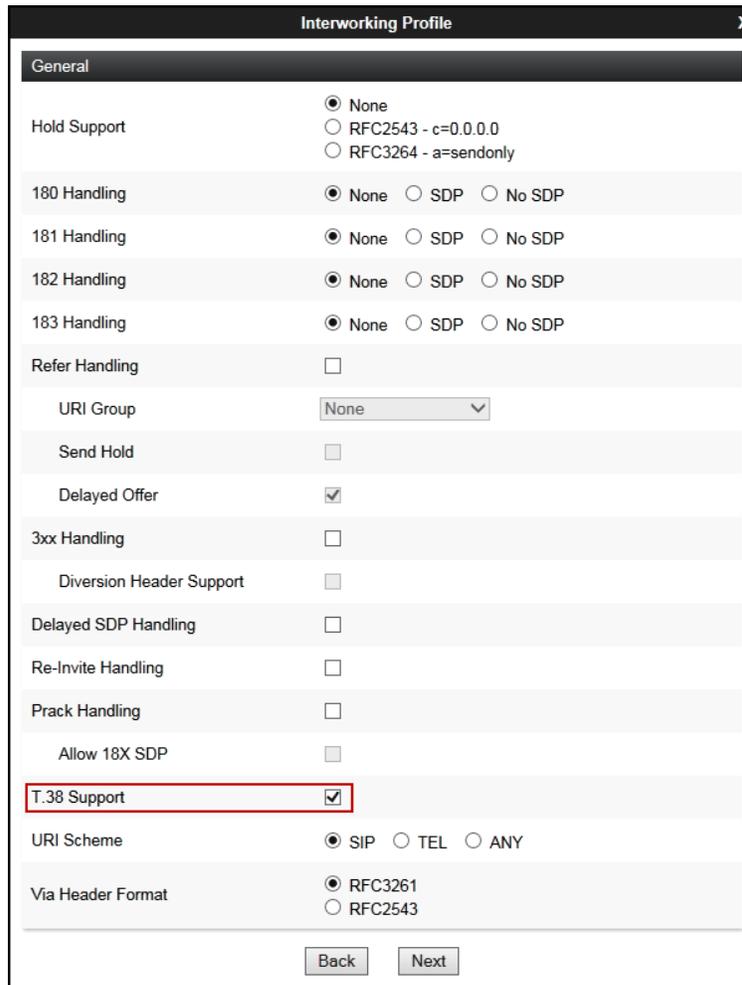
A second interworking profile in the direction of the SIP trunk was created, by adding a new profile in this case. Select **Global Profiles** → **Server Interworking** on the left navigation pane and click **Add** (not shown).

- Enter a descriptive name for the new profile.
- Click **Next**.



The screenshot shows a dialog box titled "Interworking Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" containing the text "SP-General". A red rectangular box highlights the "Profile Name" field. Below the input field, there is a "Next" button.

- On the General tab, check **T.38 Support**.
- Click **Next** until the last tab is reached then click **Finish** on the last tab leaving remaining fields with default values (not shown).



The screenshot shows the "Interworking Profile" dialog box with the "General" tab selected. The "T.38 Support" checkbox is checked and highlighted with a red box. Other options include:

- Hold Support:  None,  RFC2543 - c=0.0.0.0,  RFC3264 - a=sendonly
- 180 Handling:  None,  SDP,  No SDP
- 181 Handling:  None,  SDP,  No SDP
- 182 Handling:  None,  SDP,  No SDP
- 183 Handling:  None,  SDP,  No SDP
- Refer Handling:
- URI Group: None (dropdown)
- Send Hold:
- Delayed Offer:
- 3xx Handling:
- Diversion Header Support:
- Delayed SDP Handling:
- Re-Invite Handling:
- Prack Handling:
- Allow 18X SDP:
- URI Scheme:  SIP,  TEL,  ANY
- Via Header Format:  RFC3261,  RFC2543

At the bottom of the dialog, there are "Back" and "Next" buttons.

## 8.8. Signaling Manipulation

The Signaling Manipulation feature of the Avaya SBCE allows an administrator to perform granular header manipulations on the headers of the SIP messages, which sometimes is not possible by direct configuration on the web interface. This ability to configure header manipulation in such a highly flexible manner is achieved by the use of a proprietary scripting language called SigMa.

The script can be created externally as a regular text file and imported in the Signaling Manipulation screen, or they can be written directly in the page using the embedded Sigma Editor. In the reference configuration, the Editor was used. A detailed description of the structure of the SigMa scripting language and details on its use is beyond the scope of these Application Notes. Consult reference [8] in the **References** section for more information on this topic.

A single Sigma script was created during the compliance test to correct the following interoperability issues (refer to **Section 2.2**):

- Remove the unused “gsid” parameter and P-Location from the Contact header.
- Change the Diversion header scheme from SIPS to SIP.
- Remove unwanted xml element information from the SDP in SIP messages sent to Frontier Communications.

The scripts will later be applied to the Server Configuration profiles corresponding to the Service Provider (toward Frontier Communications) in **Section 8.9.2**.

To create the SigMa script on the left navigation pane, select **Configuration Profiles** → **Signaling Manipulation**. From the **Signaling Manipulation Scripts** list, select **Add**.

- For **Title** enter a name, the name *Frontier\_Sigma* was chosen in this example.
- Copy and paste the entire script shown below or from **Appendix A**.
- Click **Save**.

---

```
within session "ALL"
{
act on message where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
{
//Remove gsid parameter in Contact header
remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);

//Remove P-Location parameter
remove(%HEADERS["P-Location"][1]);

//Changes the Diversion header scheme from SIPS to SIP.
%HEADERS["Diversion"][1].regex_replace("sips","sip");
```

//Remove unwanted xml element information from the SDP in SIP messages sent to Service Provider.

```
remove(%BODY[1]);
```

```
}  
}
```

---

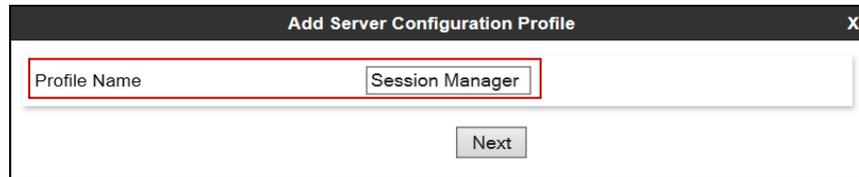
## 8.9. Server Configuration

Server Profiles are created to define the parameters for the Avaya SBCE peers; Session Manager (Call Server) at the enterprise and Frontier Communications SIP Proxy (Trunk Server).

### 8.9.1. Server Configuration Profile – Enterprise

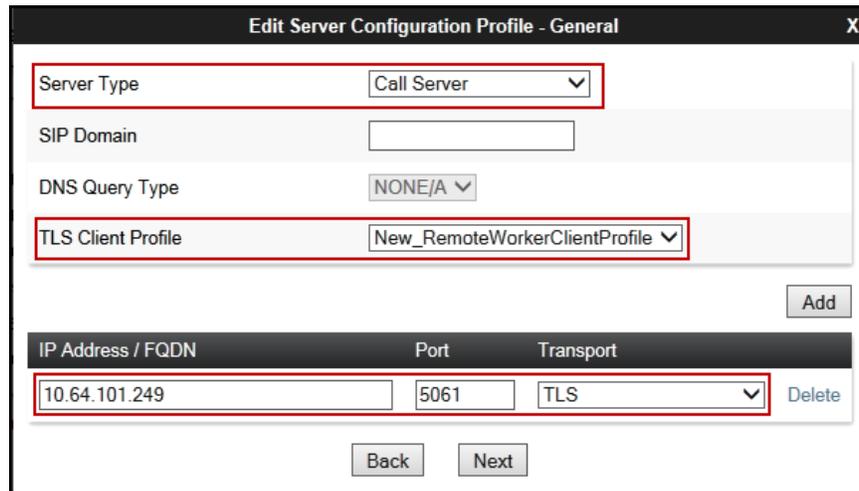
From the **Services** menu on the left-hand navigation pane, select **SIP Servers** and click the **Add** button (not shown) to add a new profile for the Call Server.

- Enter an appropriate **Profile Name** similar to the screen below.
- Click **Next**.



The screenshot shows a dialog box titled "Add Server Configuration Profile". It has a close button (X) in the top right corner. The main content area contains a text input field labeled "Profile Name" with the value "Session Manager" entered. Below the input field is a "Next" button.

- On the **Edit SIP Server Profile – General** tab select **Call Server** from the drop-down menu under the **Server Type**.
- On the **IP Addresses / FQDN** field, enter the IP address of the Session Manager Security Module (**Section 7.5**).
- Enter **5061** under **Port** and select **TLS** for **Transport**. The transport protocol and port selected here must match the values defined for the Entity Link to the Session Manager previously created in **Section 7.6**.
- Select a **TLS Profile**.
- Click **Next**.



The screenshot shows a dialog box titled "Edit Server Configuration Profile - General". It has a close button (X) in the top right corner. The main content area contains several fields: "Server Type" (dropdown menu set to "Call Server"), "SIP Domain" (text input field), "DNS Query Type" (dropdown menu set to "NONE/A"), and "TLS Client Profile" (dropdown menu set to "New\_RemoteWorkerClientProfile"). Below these fields is an "Add" button. At the bottom, there is a table with three columns: "IP Address / FQDN", "Port", and "Transport". The first row contains the values "10.64.101.249", "5061", and "TLS" (dropdown menu). A "Delete" button is located to the right of the table. At the very bottom are "Back" and "Next" buttons.

- Click **Next** until the **Add Server Configuration Profile – Advanced** tab is reached (not shown).
- On the **Add Server Configuration Profile – Advanced** tab:
  - Check **Enable Grooming**.
  - Select **Avaya-SM** from the **Interworking Profile** drop-down menu (**Section 8.7.1**).
- Click **Finish**.

The screenshot shows a configuration window titled "Add SIP Server Profile - Advanced". The window contains several settings:

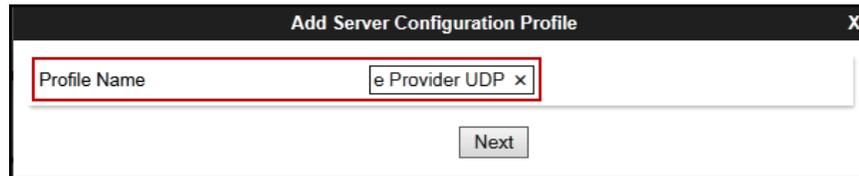
- Enable DoS Protection:
- Enable Grooming:  (highlighted with a red box)
- Interworking Profile: Avaya-SM (dropdown menu, highlighted with a red box)
- Signaling Manipulation Script: None (dropdown menu)
- Securable:
- Enable FGDN:
- TCP Failover Port: 5060 (text input)
- TLS Failover Port: 5061 (text input)
- Tolerant:
- URI Group: None (dropdown menu)

At the bottom of the window, there are two buttons: "Back" and "Finish".

## 8.9.2. Server Configuration Profile – Service Provider

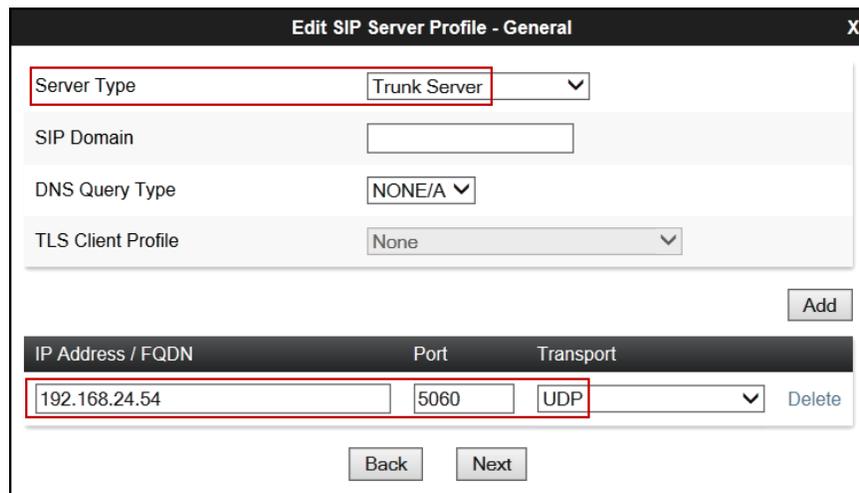
Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown).

- Enter an appropriate **Profile Name** similar to the screen below (*Service Provider UDP* was used).
- Click **Next**.



The screenshot shows a dialog box titled "Add Server Configuration Profile". It has a close button (X) in the top right corner. The main area contains a text input field labeled "Profile Name" with the text "e Provider UDP" inside. A red rectangular box highlights the input field. Below the input field is a "Next" button.

- On the **Edit Server Configuration Profile - General** Tab select *Trunk Server* from the drop-down menu for the **Server Type**.
- On the **IP Addresses / FQDN** field, enter *192.168.24.54* (the IP address of Frontier’s SIP proxy server. This information was provided by Frontier).
- Enter *5060* under **Port** and select **UDP** for **Transport**.
- Click **Next** until the **Add Server Configuration Profile – Advanced** tab is reached (not shown).



The screenshot shows a dialog box titled "Edit SIP Server Profile - General". It has a close button (X) in the top right corner. The main area contains several fields: "Server Type" (dropdown menu set to "Trunk Server"), "SIP Domain" (text input field), "DNS Query Type" (dropdown menu set to "NONE/A"), and "TLS Client Profile" (dropdown menu set to "None"). Below these fields is an "Add" button. At the bottom, there is a table with three columns: "IP Address / FQDN", "Port", and "Transport". The table contains one row with the values "192.168.24.54", "5060", and "UDP". A red rectangular box highlights the "IP Address / FQDN", "Port", and "Transport" fields. To the right of the table is a "Delete" button. Below the table are "Back" and "Next" buttons.

On the **Add Server Configuration Profile - Advanced** window:

- Uncheck **Enable Grooming**.
- Select **SP-General** from the **Interworking Profile** drop-down menu (**Section 8.7.2**).
- Select the **Frontier\_Sigma** from the **Signaling Manipulation Script** drop down menu (**Sections 8.8** and **Section 13**).
- Click **Finish**.

The screenshot shows the 'Add SIP Server Profile - Advanced' window. The 'Enable Grooming' checkbox is unchecked and highlighted with a red box. The 'Interworking Profile' dropdown is set to 'SP-General' and the 'Signaling Manipulation Script' dropdown is set to 'Frontier\_Sigma', both also highlighted with a red box. Other options include 'Enable DoS Protection', 'Securable', 'Enable FGDN', 'TCP Failover Port' (5060), 'TLS Failover Port' (5061), 'Tolerant', and 'URI Group' (None). 'Back' and 'Finish' buttons are at the bottom.

## 8.10. Routing

Routing profiles define a specific set of routing criteria that is used, in addition to other types of domain policies, to determine the path that the SIP traffic will follow as it flows through the Avaya SBCE interfaces. Two Routing Profiles were created in the test configuration, one for inbound calls, with Session Manager as the destination, and the second one for outbound calls, which are routed to the service provider SIP trunk.

### 8.10.1. Routing Profile – Enterprise

To create the inbound route, select the **Routing** tab from the **Configuration Profiles** menu on the left-hand side and select **Add** (not shown).

- Enter an appropriate **Profile Name** similar to the example below.
- Click **Next**.

The screenshot shows the 'Routing Profile' window. The 'Profile Name' field contains the text 'Route\_to\_SM' and is highlighted with a red box. A 'Next' button is located below the field.

- On the **Routing Profile** tab, click the **Add** button to enter the next-hop address.
- Under **Priority/Weight** enter **1**.
- Under **SIP Server Profile**, select **Session Manager**. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the Session Manager Server Configuration Profile in **Section 8.9.1**.
- Defaults were used for all other parameters.
- Click **Finish**.

The screenshot shows the 'Routing Profile' configuration window. The main configuration area includes fields for URI Group, Time of Day, Load Balancing, Transport, LDAP Server Profile, LDAP Base DN (Search), Matched Attribute Priority, Next Hop Priority, Ignore Route Header, ENUM, and ENUM Suffix. An 'Add' button is located at the bottom right of this section.

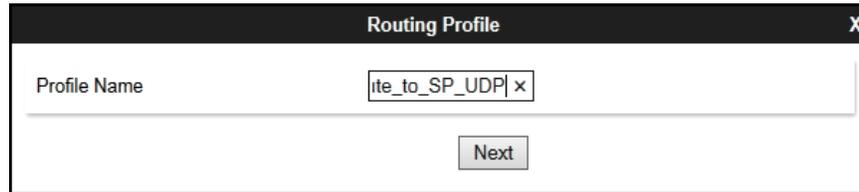
Below the configuration area is a table with the following columns: Priority / Weight, LDAP Search Attribute, LDAP Search Regex Pattern, LDAP Search Regex Result, SIP Server Profile, Next Hop Address, and Transport. The first row in the table has the following values: 1, (empty), (empty), (empty), Session Manage, 10.64.101.249:5061 (TLS), and None. The '1' in the Priority / Weight column is highlighted with a red box.

At the bottom of the window are 'Back' and 'Finish' buttons.

## 8.10.2. Routing Profile – Service Provider

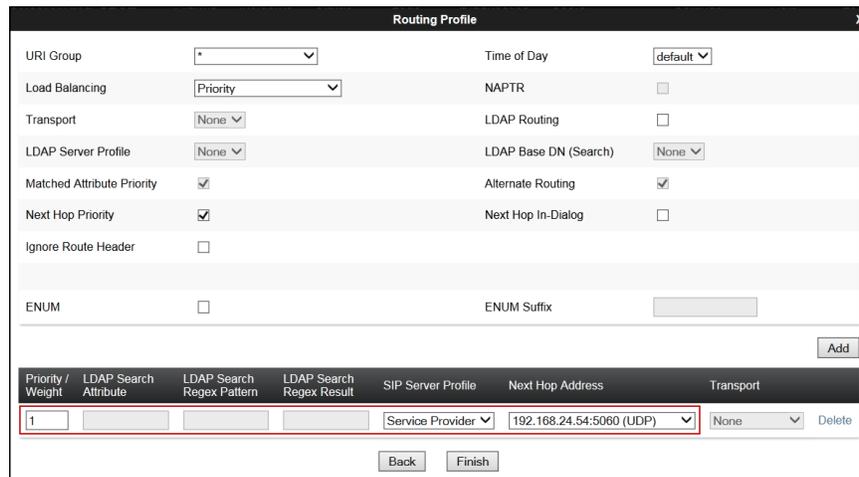
Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route.

- Enter an appropriate **Profile Name** similar to the example below (*Route\_to\_SP\_UDP* was used).
- Click **Next**.



The screenshot shows a dialog box titled "Routing Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" containing the text "ite\_to\_SP\_UDP". Below the input field is a "Next" button.

- On the **Routing Profile** tab, click the **Add** button to enter the next-hop address.
- Under **Priority/Weight** enter *1*.
- Under **SIP Server Profile**, select *Service Provider UDP*.
- The **Next Hop Address** is populated automatically with *192.168.24.54:5060 (UDP)* Frontier's SIP Proxy IP address, Port and Transport, Server Configuration Profile defined in **Section 8.9.2**.
- Click **Finish**



The screenshot shows the "Routing Profile" dialog box with various configuration options. The "Add" button is visible at the bottom right. Below the configuration options is a table with the following columns: Priority / Weight, LDAP Search Attribute, LDAP Search Regex Pattern, LDAP Search Regex Result, SIP Server Profile, Next Hop Address, and Transport. The first row in the table is highlighted with a red border and contains the following values: 1, (empty), (empty), (empty), Service Provider, 192.168.24.54:5060 (UDP), and None. Below the table are "Back" and "Finish" buttons.

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport
1				Service Provider	192.168.24.54:5060 (UDP)	None

## 8.11. Topology Hiding

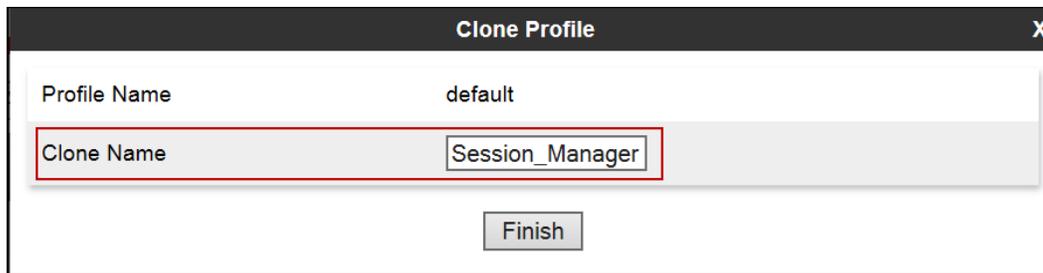
Topology Hiding is a security feature that allows the modification of several SIP headers, preventing private enterprise network information from being propagated to the untrusted public network.

Topology Hiding can also be used as an interoperability tool to adapt the host portion in the SIP headers to the IP addresses or domains expected on the service provider and the enterprise networks. For the compliance test, the default Topology Hiding Profile was cloned and modified accordingly. Only the minimum configuration required to achieve interoperability on the SIP trunk was performed. Additional steps can be taken in this section to further mask the information that is sent from the enterprise to the public network.

### 8.11.1. Topology Hiding Profile – Enterprise

To add the Topology Hiding Profile in the enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side, select *default* from the list of pre-defined profiles and click the **Clone** button (not shown).

- Enter a **Clone Name** such as the one shown below.
- Click **Finish**.



The screenshot shows a dialog box titled "Clone Profile" with a close button (X) in the top right corner. Inside the dialog, there are two input fields. The first field is labeled "Profile Name" and contains the text "default". The second field is labeled "Clone Name" and contains the text "Session\_Manager"; this field is highlighted with a red rectangular border. Below the input fields is a button labeled "Finish".

On the newly cloned *Session\_Manager* profile screen, click the **Edit** button (not shown).

- For the, **From**, **To** and **Request-Line** headers, select **Override** in the **Replace Action** column and enter the enterprise SIP domain **avaya.lab.com**, in the **Override Value** column of these headers, as shown below. This is the domain known by Session Manager, defined in **Section 7.2**.
- Default values were used for all other fields.
- Click **Finish**.

Header	Criteria	Replace Action	Overwrite Value	
To	IP/Domain	Override	avaya.lab.com	Delete
Record-Route	IP/Domain	Auto		Delete
Request-Line	IP/Domain	Override	avaya.lab.com	Delete
From	IP/Domain	Override	avaya.lab.com	Delete
Referred-By	IP/Domain	Auto		Delete
SDP	IP/Domain	Auto		Delete
Via	IP/Domain	Auto		Delete
Refer-To	IP/Domain	Auto		Delete

### 8.11.2. Topology Hiding Profile – Service Provider

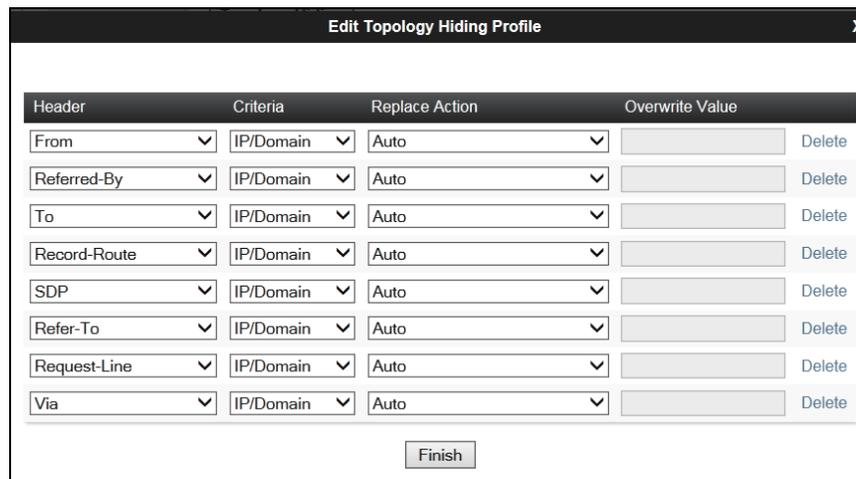
To add the Topology Hiding Profile in the service provider direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side, select *default* from the list of pre-defined profiles and click the **Clone** button (not shown).

- Enter a **Clone Name** such as the one shown below.
- Click **Finish**.



The screenshot shows a dialog box titled "Clone Profile" with a close button (X) in the top right corner. Inside the dialog, there are two input fields: "Profile Name" with the value "default" and "Clone Name" with the value "Service\_Provider". The "Clone Name" field is highlighted with a red rectangular border. Below the input fields is a "Finish" button.

During the compliance test, IP addresses and not domains names were used in all SIP messages between the service provider and the Avaya SBCE. Note that since the default action of *Auto* implies the insertion of IP addresses in the host portion of these headers, it was not necessary to modify any of the headers sent to the service provider. The screen below shows the *Service\_Provider* profile once the configuration was completed.



The screenshot shows a dialog box titled "Edit Topology Hiding Profile" with a close button (X) in the top right corner. It contains a table with the following columns: Header, Criteria, Replace Action, and Overwrite Value. The table lists several SIP headers with their respective criteria and actions. A "Finish" button is located at the bottom of the dialog.

Header	Criteria	Replace Action	Overwrite Value
From	IP/Domain	Auto	
Referred-By	IP/Domain	Auto	
To	IP/Domain	Auto	
Record-Route	IP/Domain	Auto	
SDP	IP/Domain	Auto	
Refer-To	IP/Domain	Auto	
Request-Line	IP/Domain	Auto	
Via	IP/Domain	Auto	

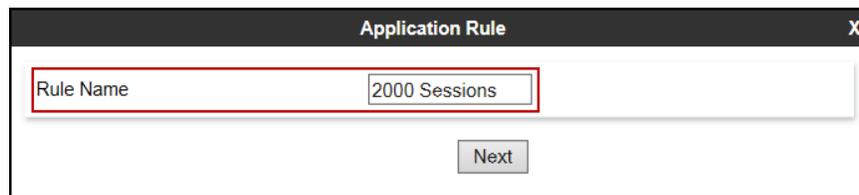
## 8.12.Domain Policies

Domain Policies allow the configuration of sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise. Domain Policies include rules for Application, Media, Signaling, Security, etc.

### 8.12.1.Application Rules

Application Rules define which types of SIP-based Unified Communications (UC) applications the UC-Sec security device will protect: voice, video, and/or Instant Messaging (IM). In addition, Application Rules define the maximum number of concurrent voice sessions the network will process in order to prevent resource exhaustion. From the menu on the left-hand side, select **Domain Policies** → **Application Rules**, click on the **Add** button to add a new rule.

- Under **Rule Name** enter the name of the profile, e.g., **2000 Sessions**.
- Click **Next**.

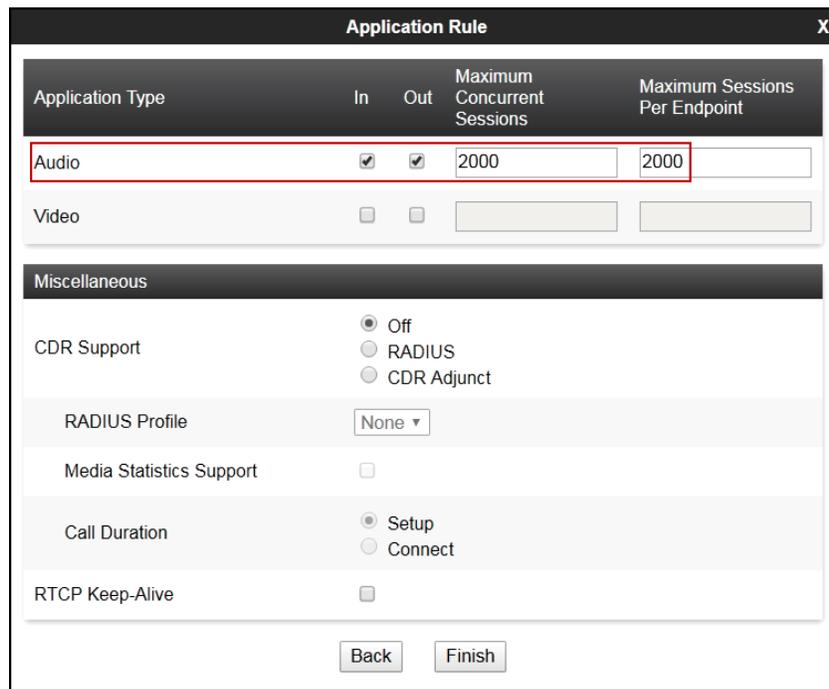


Application Rule

Rule Name: 2000 Sessions

Next

- Under **Audio** check **In** and **Out** and set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** to recommended values, the value of **2000** for Audio. Repeat for video if needed.
- Click **Finish**.



Application Rule

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Audio	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	2000	2000
Video	<input type="checkbox"/>	<input type="checkbox"/>		

Miscellaneous

CDR Support:  Off,  RADIUS,  CDR Adjunct

RADIUS Profile: None

Media Statistics Support:

Call Duration:  Setup,  Connect

RTCP Keep-Alive:

Back Finish

## 8.12.2. Media Rules

Media Rules allow one to define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBCE security product. For the compliance test, one media rule (shown below) was created toward Session Manager and a default media rule was used toward the Service Provider.

To add a media rule in the Session Manager direction, from the menu on the left-hand side, select **Domain Policies** → **Media Rules**.

- Click on the **Add** button to add a new media rule (not shown).
- Under **Rule Name** enter **SM\_SRTP**.
- Click **Next** (not shown).
- Under Audio Encryption, **Preferred Format #1**, select **SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80**.
- Under Audio Encryption, **Preferred Format #2**, select **RTP**.
- Under Audio Encryption, uncheck **Encrypted RTCP**.
- Under Audio Encryption, check **Interworking**.
- Repeat the above steps under Video Encryption, if needed.
- Under Miscellaneous verify that **Capability Negotiation** is checked.
- Click **Next**.

The screenshot shows a 'Media Rule' configuration window with three main sections: Audio Encryption, Video Encryption, and Miscellaneous. Each section contains several configuration options. In the Audio Encryption section, Preferred Format #1 is set to SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80, Preferred Format #2 is set to RTP, Encrypted RTCP is unchecked, and Interworking is checked. The Video Encryption section has identical settings. In the Miscellaneous section, Capability Negotiation is checked. At the bottom of the window are 'Back' and 'Next' buttons.

Section	Option	Value
Audio Encryption	Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80
	Preferred Format #2	RTP
	Preferred Format #3	NONE
	Encrypted RTCP	<input type="checkbox"/>
	MKI	<input type="checkbox"/>
	Lifetime	2 <sup>n</sup>
Video Encryption	Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80
	Preferred Format #2	RTP
	Preferred Format #3	NONE
	Encrypted RTCP	<input type="checkbox"/>
	MKI	<input type="checkbox"/>
	Lifetime	2 <sup>n</sup>
Miscellaneous	Capability Negotiation	<input checked="" type="checkbox"/>

- Accept default values in the remaining sections by clicking **Next** (not shown), and then click **Finish** (not shown).

- For the compliance test, the **default-low-med** Media Rule was used in the Service Provider direction.

Media Encryption	
<b>Audio Encryption</b>	
Preferred Format #1	RTP
Preferred Format #2	NONE
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime <small>Leave blank to match any value.</small>	2^ <input type="text"/>
Interworking	<input checked="" type="checkbox"/>
<b>Video Encryption</b>	
Preferred Format #1	RTP
Preferred Format #2	NONE
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime <small>Leave blank to match any value.</small>	2^ <input type="text"/>
Interworking	<input checked="" type="checkbox"/>
<b>Miscellaneous</b>	
Capability Negotiation	<input type="checkbox"/>
<input type="button" value="Finish"/>	

### 8.12.3. Signaling Rules

For the compliance test, the **default** signaling rule was used.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header displays 'Session Border Controller for Enterprise' and the 'AVAYA' logo. A left sidebar menu lists various configuration categories, with 'Domain Policies' and 'Signaling Rules' highlighted. The main content area is titled 'Signaling Rules: default' and features an 'Add' button, a 'Filter By Device...' dropdown, and a 'Clone' button. A warning message states: 'It is not recommended to edit the defaults. Try cloning or adding a new rule instead.' Below this, there are tabs for 'General', 'Requests', 'Responses', 'Request Headers', 'Response Headers', 'Signaling QoS', and 'UCID'. The 'General' tab is active, showing 'Inbound' and 'Outbound' sections with a table of settings. The 'Content-Type Policy' section includes a checked 'Enable Content-Type Checks' checkbox and an 'Action' table.

Inbound	
Requests	Allow
Non-2XX Final Responses	Allow
Optional Request Headers	Allow
Optional Response Headers	Allow

Outbound	
Requests	Allow
Non-2XX Final Responses	Allow
Optional Request Headers	Allow
Optional Response Headers	Allow

Content-Type Policy			
Enable Content-Type Checks	<input checked="" type="checkbox"/>		
Action	Allow	Multipart Action	Allow
Exception List	Exception List		

## 8.13. End Point Policy Groups

End Point Policy Groups associate the different sets of rules under Domain Policies (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBCE. Please note that changes should not be made to any of the default rules used in these End Point Policy Groups.

### 8.13.1. End Point Policy Group – Enterprise

To create an End Point Policy Group for the enterprise, select **End Point Policy Groups** under the **Domain Policies** menu and select **Add** (not shown).

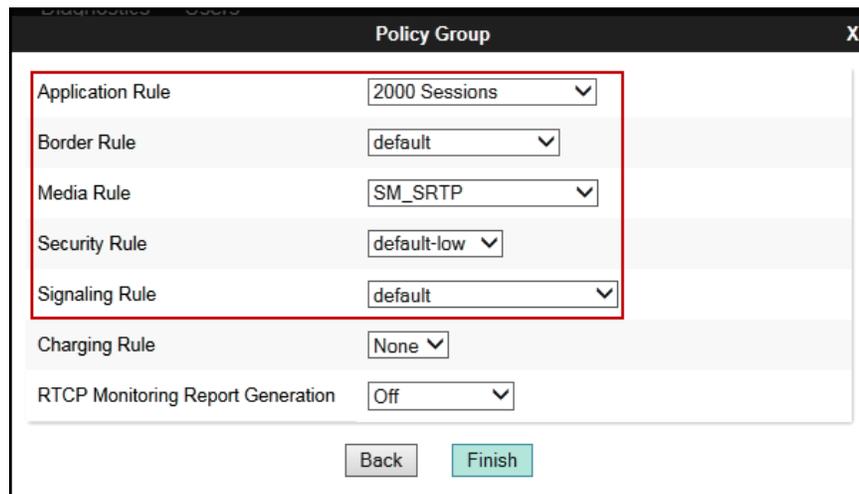
- Enter an appropriate name in the **Group Name** field.
- Click **Next**.



The screenshot shows a dialog box titled "Policy Group" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Group Name" containing the text "Enterprise". Below the input field is a "Next" button.

Under the **Policy Group** tab enter the following:

- **Application Rule:** *2000 Sessions* (Section 8.12.1).
- **Border Rule:** *default*.
- **Media Rule:** *SM\_SRTP* (Section 8.12.2).
- **Security Rule:** *default-low*.
- **Signaling Rule:** *default* (Section 8.12.3).
- Click **Finish**.

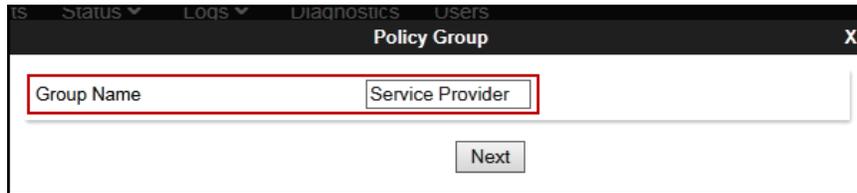


The screenshot shows the "Policy Group" dialog box with several dropdown menus. A red box highlights the first five rules: Application Rule (2000 Sessions), Border Rule (default), Media Rule (SM\_SRTP), Security Rule (default-low), and Signaling Rule (default). Below these are Charging Rule (None) and RTCP Monitoring Report Generation (Off). At the bottom, there are "Back" and "Finish" buttons.

### 8.13.2. End Point Policy Group – Service Provider

To create an End Point Policy Group for the Service Provider, select **End Point Policy Groups** under the **Domain Policies** menu and select **Add** (not shown).

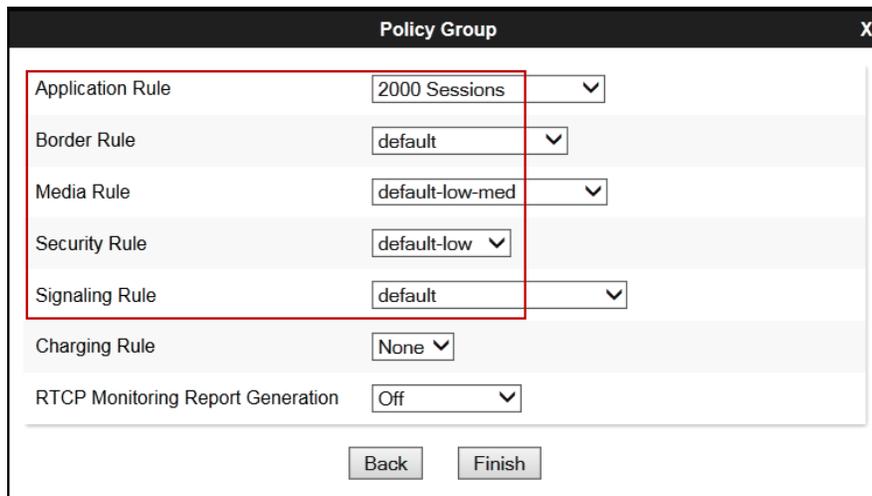
- Enter an appropriate name in the **Group Name** field (*Service Provider* was used).
- Click **Next**.



The screenshot shows a window titled "Policy Group" with a close button (X) in the top right corner. At the top, there are navigation tabs: "ts", "Status", "Logs", "Diagnostics", and "Users". Below the tabs is a text input field labeled "Group Name" containing the text "Service Provider". A red rectangular box highlights this field. Below the input field is a "Next" button.

Under the **Policy Group** tab enter the following:

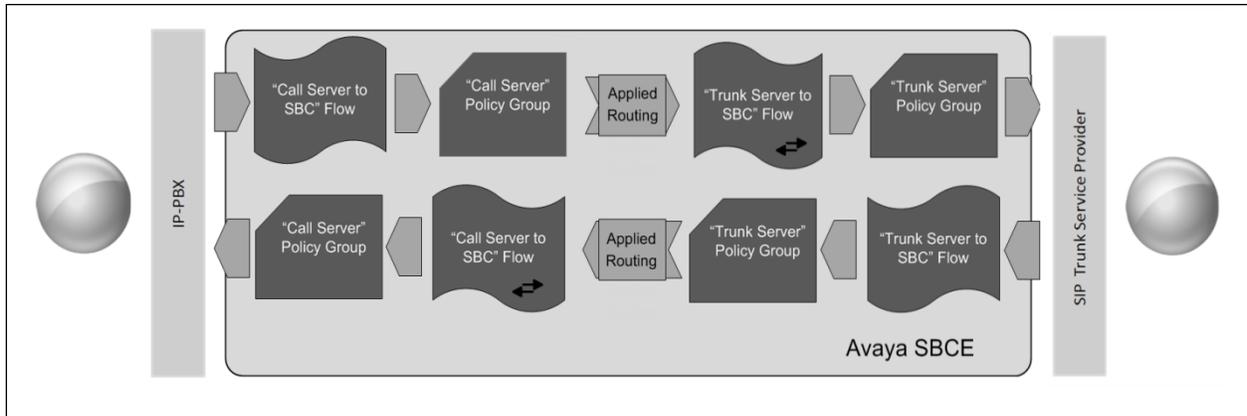
- **Application Rule:** *2000 Sessions* (Section 8.12.1).
- **Border Rule:** *default*.
- **Media Rule:** *default-low-med* (Section 8.12.2).
- **Security Rule:** *default-low*.
- **Signaling Rule:** *default* (Section 8.12.3).
- Click **Finish**.



The screenshot shows a window titled "Policy Group" with a close button (X) in the top right corner. The window contains several configuration options, each with a dropdown menu. A red rectangular box highlights the first five options: "Application Rule" (2000 Sessions), "Border Rule" (default), "Media Rule" (default-low-med), "Security Rule" (default-low), and "Signaling Rule" (default). Below these are "Charging Rule" (None) and "RTCP Monitoring Report Generation" (Off). At the bottom of the window are "Back" and "Finish" buttons.

## 8.14. End Point Flows

When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy group which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screen illustrates the flow through the Avaya SBCE to secure a SIP trunk call.



The **End-Point Flows** defines certain parameters that pertain to the signaling and media portions of a call, whether it originates from within the enterprise or outside of the enterprise.

### 8.14.1. End Point Flow – Enterprise

To create the call flow toward the enterprise, from the **Device Specific** menu, select **End Point Flows**, then select the **Server Flows** tab. Click **Add** (not shown). The screen below shows the flow named *Session\_Manager\_Flow* created in the sample configuration. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for the Service Provider in **Section 8.10.2**, which is the reverse route of the flow. Click **Finish**.

Flow Name	Session_Manager_Flow
SIP Server Profile	Session Manager
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Public_sig
Signaling Interface	Private_sig
Media Interface	Private_med
Secondary Media Interface	None
End Point Policy Group	Enterprise
Routing Profile	Route_to_SP_UDP
Topology Hiding Profile	Session_Manager
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>

Finish

### 8.14.2. End Point Flow – Service Provider

A second Server Flow with the name *SIP\_Trunk\_Flow\_UDP* was similarly created in the Service Provider direction. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for Session Manager in **Section 8.10.1**, which is the reverse route of the flow. Also note that there is no selection under the **Signaling Manipulation Script** field. Click **Finish**.

Field	Value
Flow Name	SIP_Trunk_Flow_UDP
SIP Server Profile	Service Provider UDP
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Private_sig
Signaling Interface	Public_sig
Media Interface	Public_med
Secondary Media Interface	None
End Point Policy Group	Service Provider
Routing Profile	Route_to_SM
Topology Hiding Profile	Service_Provider
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>

Finish

## 9. Frontier Communications SIP Trunking Service Configuration

To use Frontier Communications SIP Trunking Service, a customer must request the service from Frontier Communications using the established sales processes. The process can be started by contacting Frontier Communications via the corporate web site at:

<https://frontier.com/enterprise>

During the signup process, Frontier Communications and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to Frontier Communications network.

Frontier will provide the following information:

- Frontier SIP proxy server IP address.
- DID numbers.
- Supported codecs and order of preference.
- Etc.

## 10. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of commands that can be used to troubleshoot the solution.

### 10.1. General Verification Steps

- Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
- Verify that the user on the PSTN can end an active call by hanging up.
- Verify that an endpoint at the enterprise site can end an active call by hanging up.

### 10.2. Communication Manager Verification

The following commands can be entered in the Communication Manager SAT terminal to verify the SIP trunk functionality:

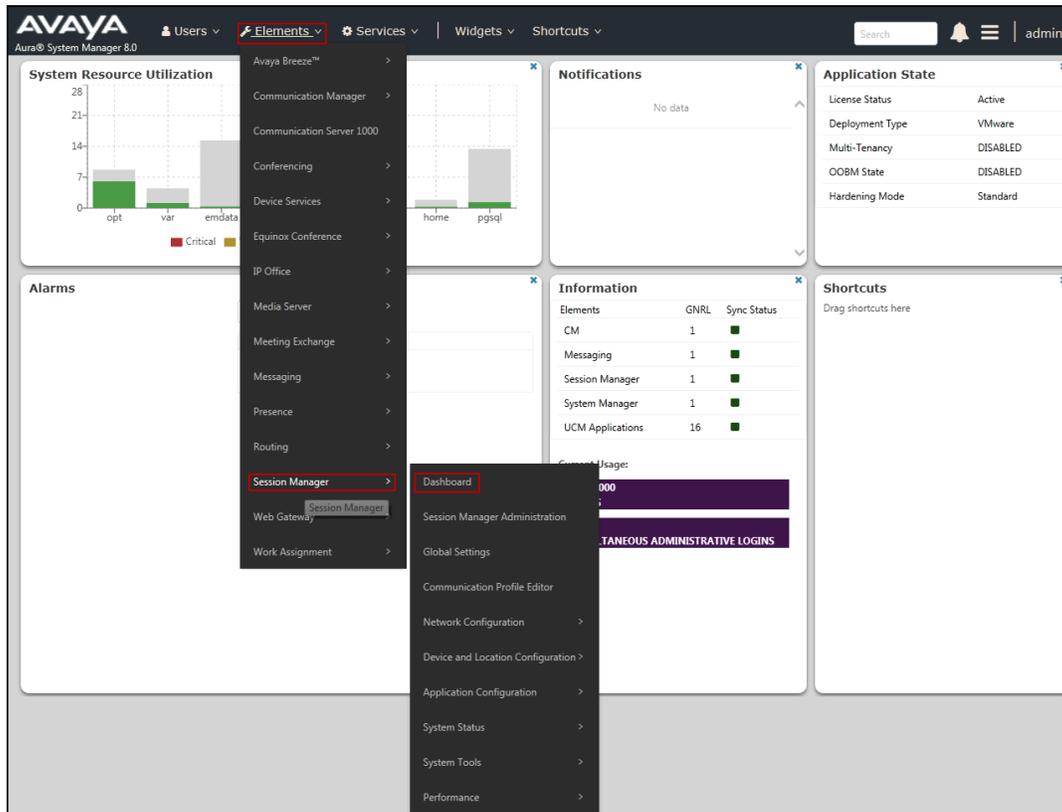
- **list trace station** <extension number>  
Traces calls to and from a specific station.
- **list trace tac** <trunk access code number>  
Trace calls over a specific trunk group.
- **status signaling-group** <signaling group number>  
Displays signaling group service state.
- **status trunk** <trunk group number>  
Displays trunk group service state.
- **status station** <extension number>

Displays signaling and media information for an active call on a specific station.

### 10.3.Session Manager Verification

The Session Manager configuration may be verified via System Manager.

**Step 1** - Using the procedures described in **Section 7**, access the System Manager GUI. From the **Home** screen, under the **Elements** heading, select **Session Manager**, then select **Dashboard**.



**Step 2** - The Session Manager Dashboard is displayed. Note that the **Test Passed**, **Alarms**, **Service State**, and **Data Replication** columns all show good status.

In the **Entity Monitoring** column, Session Manager shows that there are **2** alarms out of the **7** Entities defined.

**Session Manager Dashboard**

This page provides the overall status and health summary of each administered Session Manager.

**Session Manager Instances**

Service State: Shutdown System EASG: As of 1:40 PM

1 Item Show All Filter: Enable

	Session Manager	Type	Tests Pass	Alarms	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Data Replication	User Data Storage Status	License Mode	EASG	Version
<input type="checkbox"/>	<a href="#">Session Manager</a>	Core	0/0/0	0/0/0	Up	Accept New Service	2/7	0	1/1	✓	✓	Normal	Enabled	8.0.1.1.801103

Select: All, None

Verify that the state of the Session Manager links under the **Conn. Status** and **Link Status** columns are **UP**, like shown on the screen below

**Session Manager Entity Link Connection Status**

This page displays detailed connection status for all entity links from a Session Manager.

Status Details for the selected Session Manager:

**All Entity Links for Session Manager: Session Manager**

Summary View

7 Items Filter: Enable

	SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	<a href="#">Avaya_SBCE</a>	IPv4	10.64.101.243	5061	TLS	FALSE	UP	200 OK	UP
<input type="radio"/>	<a href="#">Avaya Experience Portal</a>	IPv4	10.64.101.252	5061	TLS	FALSE	UP	200 OK	UP
<input type="radio"/>	<a href="#">Communication Manager Trunk 1</a>	IPv4	10.64.101.241	5061	TLS	FALSE	UP	200 OK	UP
<input type="radio"/>	<a href="#">AA-Messaging</a>	IPv4	10.64.101.250	5060	TCP	FALSE	UP	200 OK	UP
<input type="radio"/>	<a href="#">Communication Manager Trunk 2</a>	IPv4	10.64.101.241	5071	TLS	FALSE	UP	200 OK	UP
<input type="radio"/>	<a href="#">Communication Manager Trunk 98</a>	IPv4	10.64.101.241	5065	TLS	FALSE	UP	200 OK	UP
<input type="radio"/>	<a href="#">CS1K7.6</a>	IPv4	172.16.5.60	5085	UDP	FALSE	DOWN	408 Request Timeout	DOWN

Select: None

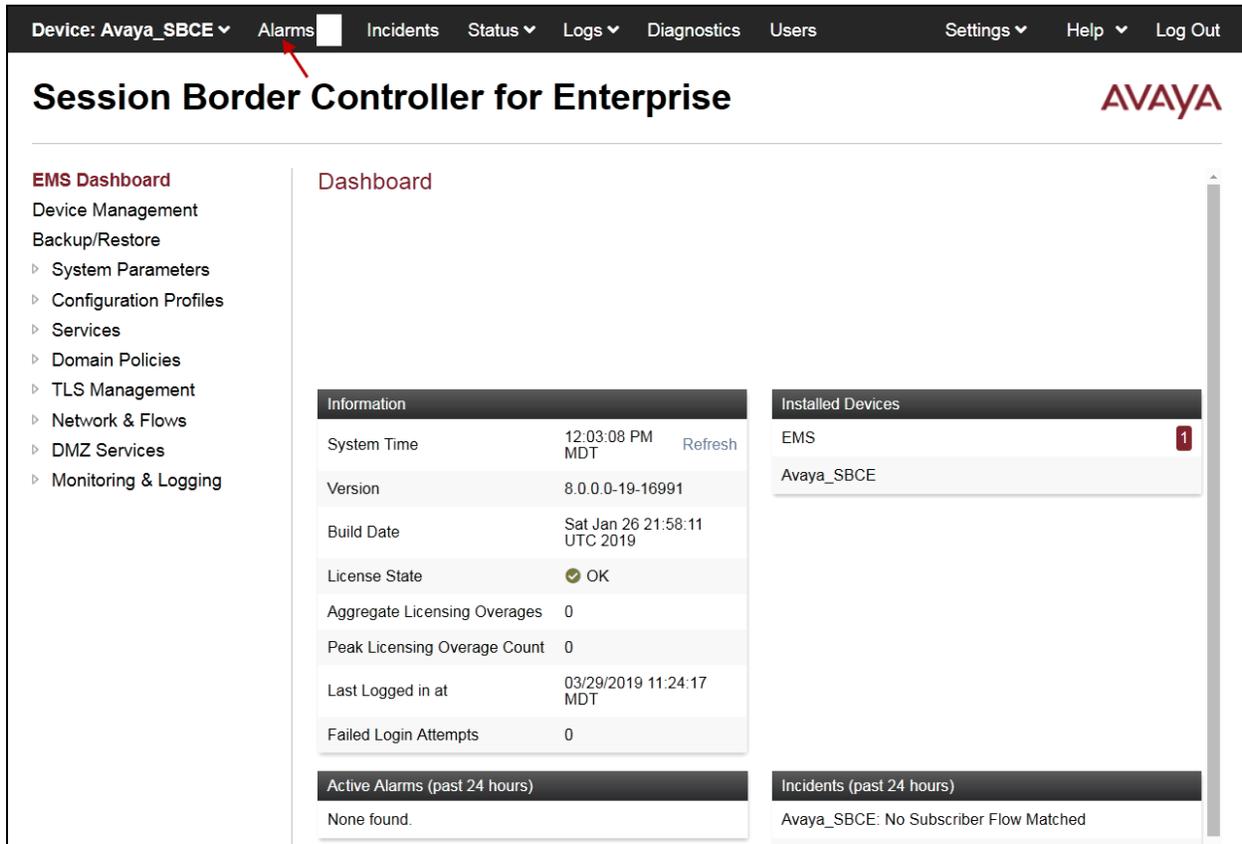
Other Session Manager useful verification and troubleshooting tools include:

- **traceSM** – Session Manager command line tool for traffic analysis. Login to the Session Manager command line management interface to run this command.
- **Call Routing Test** – The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, from the System Manager Home screen navigate to **Elements** → **Session Manager** → **System Tools** → **Call Routing Test**. Enter the requested data to run the test.

## 10.4. Avaya SBCE Verification

There are several links and menus located on the taskbar at the top of the screen of the web interface that can provide useful diagnostic or troubleshooting information.

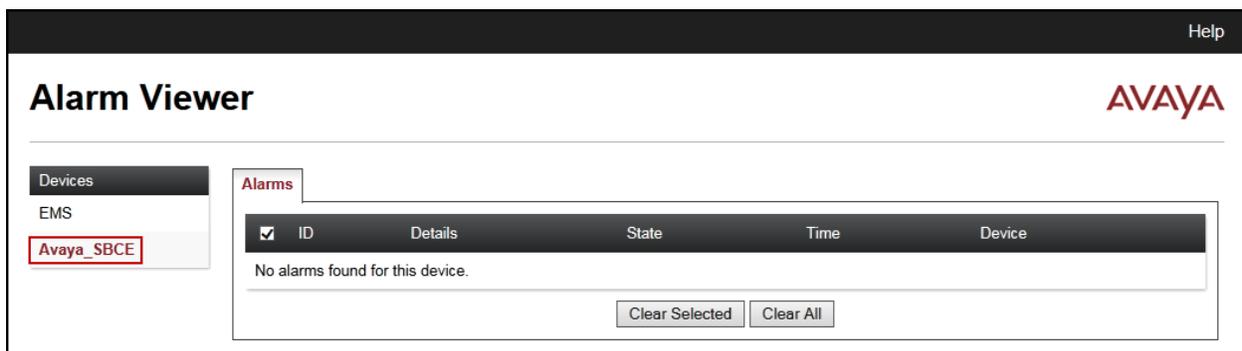
**Alarms:** This screen provides information about the health of the SBC.



The screenshot shows the Avaya SBCE Dashboard. The top navigation bar includes: Device: Avaya\_SBCE, Alarms (highlighted with a red arrow), Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" with the AVAYA logo on the right. The left sidebar lists the EMS Dashboard menu items: Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, TLS Management, Network & Flows, DMZ Services, and Monitoring & Logging. The main content area is titled "Dashboard" and contains several panels:

- Information:** System Time (12:03:08 PM MDT), Version (8.0.0.0-19-16991), Build Date (Sat Jan 26 21:58:11 UTC 2019), License State (OK), Aggregate Licensing Overages (0), Peak Licensing Overage Count (0), Last Logged in at (03/29/2019 11:24:17 MDT), Failed Login Attempts (0).
- Installed Devices:** EMS (1), Avaya\_SBCE.
- Active Alarms (past 24 hours):** None found.
- Incidents (past 24 hours):** Avaya\_SBCE: No Subscriber Flow Matched.

The following screen shows the **Alarm Viewer** page.



The screenshot shows the Avaya Alarm Viewer page. The top navigation bar includes: Help. The main header reads "Alarm Viewer" with the AVAYA logo on the right. The left sidebar lists the Devices menu items: EMS, Avaya\_SBCE (highlighted with a red box). The main content area is titled "Alarms" and contains a table with the following columns: ID, Details, State, Time, Device. The table is empty, and the text "No alarms found for this device." is displayed. Below the table are two buttons: "Clear Selected" and "Clear All".

**Incidents** : Provides detailed reports of anomalies, errors, policies violations, etc.

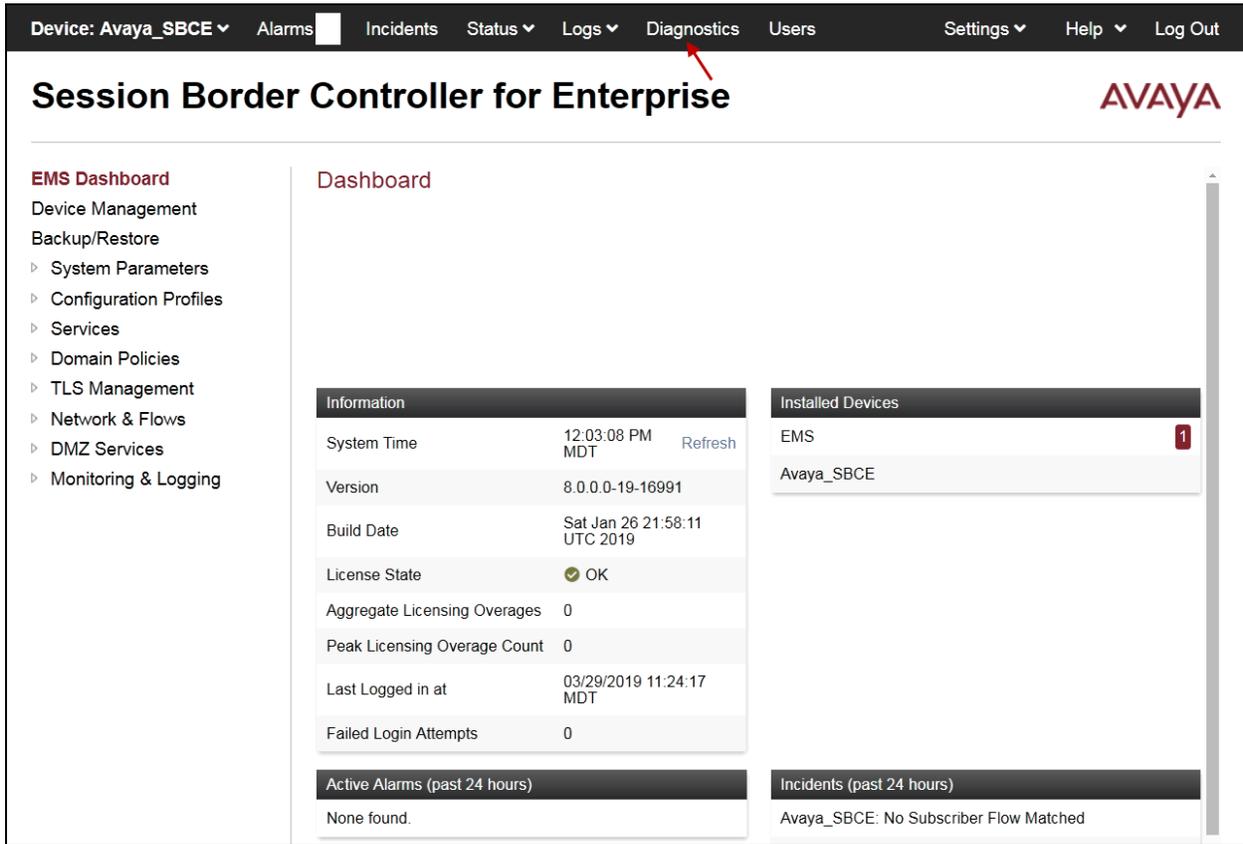
The screenshot shows the Avaya Session Border Controller for Enterprise dashboard. The top navigation bar includes 'Device: Avaya\_SBCE', 'Alarms', 'Incidents' (highlighted with a red arrow), 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Session Border Controller for Enterprise' with the AVAYA logo on the right. A left sidebar lists 'EMS Dashboard' options: Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, TLS Management, Network & Flows, DMZ Services, and Monitoring & Logging. The main content area is titled 'Dashboard' and contains several panels:
 

- Information**: System Time (12:03:08 PM MDT), Version (8.0.0.0-19-16991), Build Date (Sat Jan 26 21:58:11 UTC 2019), License State (OK), Aggregate Licensing Overages (0), Peak Licensing Overage Count (0), Last Logged in at (03/29/2019 11:24:17 MDT), Failed Login Attempts (0).
- Installed Devices**: Shows 'EMS' with a red '1' indicator and 'Avaya\_SBCE'.
- Active Alarms (past 24 hours)**: None found.
- Incidents (past 24 hours)**: Avaya\_SBCE: No Subscriber Flow Matched.

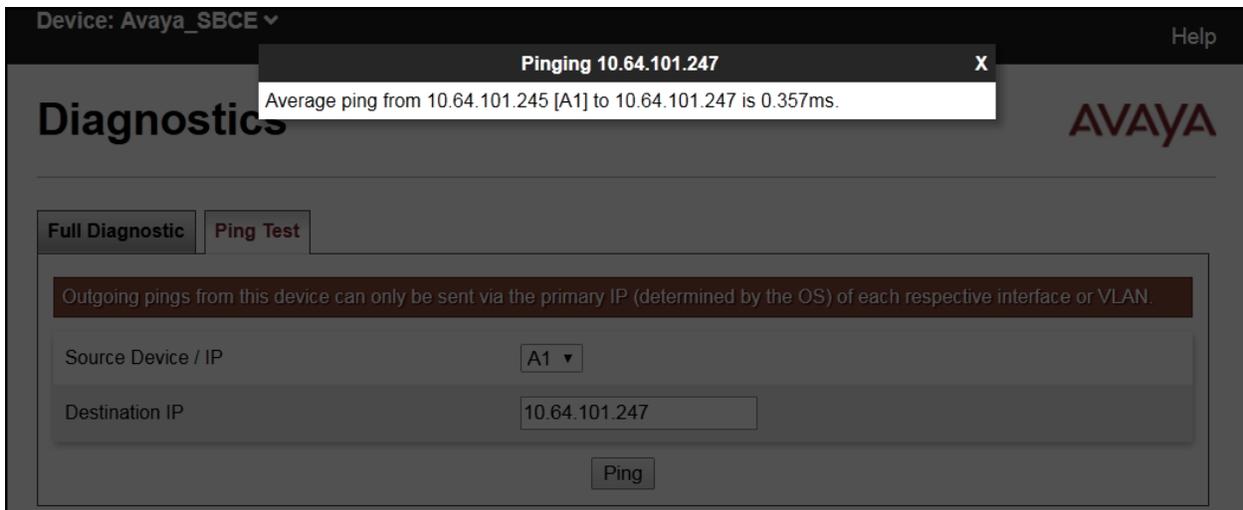
The following screen shows the Incident Viewer page.

The screenshot shows the Avaya Incident Viewer page. The top right corner has a 'Help' link. The main header reads 'Incident Viewer' with the AVAYA logo on the right. Below the header are filter controls: 'Device' set to 'All', 'Category' set to 'Authentication', and a 'Clear Filters' button. There are also 'Refresh' and 'Generate Report' buttons. Below the filters, it says 'Displaying results 0 to 0 out of 0.' A table with columns 'ID', 'Device', 'Date & Time', 'Category', 'Type', and 'Cause' is shown, with the message 'No incidents found.' below it. At the bottom, there are pagination controls: '<<', '<', '1', '>', '>>'.

**Diagnostics:** This screen provides a variety of tools to test and troubleshoot the Avaya SBCE network connectivity.



The following screen shows the Diagnostics page with the results of a ping test.



Additionally, the Avaya SBCE contains an internal packet capture tool that allows the capture of packets on any of its interfaces, saving them as *pcap* files. Navigate to **Monitor & Logging** → **Trace**. Select the **Packet Capture** tab, set the desired configuration for the trace and click **Start Capture**.

The screenshot displays the Avaya SBCE web interface. At the top, a navigation bar shows 'Device: Avaya\_SBCE', 'Alarms 1', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. Below this, the main header reads 'Session Border Controller for Enterprise' with the AVAYA logo on the right. A left-hand navigation menu lists various management options, with 'Monitoring & Logging' selected and 'Trace' highlighted. The main content area is titled 'Trace: Avaya\_SBCE' and features two tabs: 'Packet Capture' (active) and 'Captures'. The 'Packet Capture Configuration' form includes the following fields: Status (Ready), Interface (Any), Local Address (All), Remote Address (\*), Protocol (All), Maximum Number of Packets to Capture (10000), and Capture Filename (Blind\_Xfer.pcap). 'Start Capture' and 'Clear' buttons are located at the bottom of the form.

Once the capture is stopped, click the **Captures** tab and select the proper *pcap* file. Note that the date and time is appended to the filename specified previously. The file can now be saved to the local PC, where it can be opened with an application such as Wireshark.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Device: Avaya\_SBCE', 'Alarms 1', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Session Border Controller for Enterprise' with the AVAYA logo on the right. A left-hand navigation menu lists various system management options, with 'Monitoring & Logging' and its sub-item 'Trace' highlighted. The main content area is titled 'Trace: Avaya\_SBCE' and contains two tabs: 'Packet Capture' and 'Captures'. The 'Captures' tab is active and displays a table of captured files. The table has columns for 'File Name', 'File Size (bytes)', and 'Last Modified'. A single entry is shown: 'Blind\_Xfer\_20190325155823.pcap' with a size of 1,859,584 bytes and a timestamp of March 25, 2019 3:59:11 PM MDT. A 'Delete' button is visible next to the entry. A 'Refresh' button is located in the top right corner of the table area.

File Name	File Size (bytes)	Last Modified
Blind_Xfer_20190325155823.pcap	1,859,584	March 25, 2019 3:59:11 PM MDT

Also, the **traceSBC** tool can be used to monitor the SIP signaling messages between the Service provider and the Avaya SBCE.

## 11. Conclusion

These Application Notes describe the procedures required to configure Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, Avaya Aura® Experience Portal 7.2, and Avaya Session Border Controller for Enterprise 8.0, to connect to the Frontier Communications SIP Trunking service, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with the observations/limitations described in **Sections 2.1** and **2.2**.

## 12. References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Deploying Avaya Aura® Communication Manager in a Virtualized Environment*, Release 8.0.1, Issue 4, February 2019.
- [2] *Administering Avaya Aura® Communication Manager*, Release 8.0.1, Issue 3, December 2018.
- [3] *Administering Avaya Aura® System Manager* for Release 8.0.1, Issue 7, January 2019.
- [4] *Deploying Avaya Aura® System Manager in a Virtualized Environment*, Release 8.0.1, Issue 4, February 2019.
- [5] *Deploying Avaya Aura® Session Manager and Avaya Aura® Branch Session Manager in a Virtualized Environment*, Release 8.0.1, Issue 4, February 2019.
- [6] *Administering Avaya Aura® Session Manager*, Release 8.0.1, Issue 3, December 2018.
- [7] *Deploying Avaya Session Border Controller in a Virtualized Environment*, Release 8.0, Issue 2, March 2019.
- [8] *Administering Avaya Session Border Controller for Enterprise*, Release 8.0, Issue 1, February 2019.
- [9] *Administering Avaya Aura® Experience Portal*, Release 7.2.2, Issue 1, March 2019
- [10] *Implementing Avaya Aura® Experience Portal on a single server*, Release 7.2.2, Issue 1, July 2019
- [11] *Configuring Remote Workers with Avaya Session Border Controller for Enterprise Rel. 7.0, Avaya Aura® Communication Manager Rel. 7.0 and Avaya Aura® Session Managers Rel. 7.0 - Issue 1.0*.
- [12] *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 8.0, Issue 6, March 2019.
- [13] *Implementing and Administering Avaya Aura® Media Server*. Release 8.0, Issue 3, November 2018.
- [14] *Planning for and Administering Avaya Equinox for Android, iOS, Mac, and Windows*. Release 3.5.5, Issue 1, March 2019.
- [15] *Administering Avaya one-X® Communicator*. Release 6.2, Feature Pack 10, November 2015.
- [16] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [17] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>

## 13. Appendix A: SigMa Scripts

Following are the Signaling Manipulation scripts that were used in the configuration of the Avaya SBCE, **Section 8.8**. When adding these scripts as instructed in **Sections 8.9.2** enter a name for the script in the Title (e.g., *Frontier\_Sigma*) and copy/paste the entire scripts shown below.

---

The following SigMa scripts will:

- Remove the unused “gsid” parameter and P-Location from the Contact header.
  - Change the Diversion header scheme from SIPS to SIP.
  - Remove unwanted xml element information from the SDP in SIP messages sent to Frontier Communications.
- 

**Title:** *Frontier\_Sigma*

This script is to be applied to the Service Provider Server Configuration

within session "ALL"

```
{
act on message where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
{
//Remove gsid parameter in Contact header
remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);

//Remove P-Location parameter
remove(%HEADERS["P-Location"][1]);

//Changes the Diversion header scheme from SIPS to SIP.
%HEADERS["Diversion"][1].regex_replace("sips","sip");

//Remove unwanted xml element information from the SDP in SIP messages sent to Service
Provider.
remove(%BODY[1]);

}
}
```

---

## 14. Appendix B – Avaya Session Border Controller for Enterprise – Refer Handling

One of the capabilities important to the Experience Portal environment is the Avaya SBCE Refer Handling option. Experience Portal inbound call processing may include call redirection to Communication Manager agents, or other CPE destinations. This redirection is accomplished by having Experience Portal send SIP REFER messaging to the Avaya SBCE. Enabling the Refer Handling option causes the Avaya SBCE to intercept and process the REFER and generate a new SIP INVITE messages back to the CPE (e.g., Communication Manager).

As an additional option, the Refer Handling feature can also specify *URI Group* criteria as a discriminator, whereby SIP REFER messages matching the URI Group criteria are processed by the Avaya SBCE, while SIP REFER messages that do not match the URI Group criteria, are passed through to the Service Provider. Since the SIP REFER method for call redirection is not fully supported by Frontier (refer to **Section 2.1**) the *URI Group* criteria method for SIP REFER handling was not used.

Edit the existing **SP-General** Server Interworking Profile to enable Refer Handling.

**Step 1** - Select **Configuration Profiles → Server Interworking** from the left-hand menu (not shown).

**Step 2** - Select the **SP-General** Server Interworking Profile created in **Section 8.7.2** and click **Edit**

- Check **Refer Handling**.
- Select **Finish**.

(Note that URI Group was left as *None* (not used, as mentioned above)).

General	
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input checked="" type="checkbox"/>
URI Group	None
Send Hold	<input type="checkbox"/>
Delayed Offer	<input checked="" type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
Re-Invite Handling	<input type="checkbox"/>
Prack Handling	<input type="checkbox"/>
Allow 18X SDP	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Finish

Following is the SP-General Server Interworking profile after editing.

Device: Avaya\_SBCE ▾ Alarms Incidents Status ▾ Logs ▾ Diagnostics Users

## Session Border Controller for Enterprise

EMS Dashboard

Device Management

Backup/Restore

▸ System Parameters

▸ **Configuration Profiles**

Domain DoS

**Server Interworking**

Media Forking

Routing

Topology Hiding

Signaling Manipulation

URI Groups

SNMP Traps

Time of Day Rules

FGDN Groups

Reverse Proxy Policy

▸ Services

▸ Domain Policies

▸ TLS Management

▸ Network & Flows

▸ DMZ Services

▸ Monitoring & Logging

### Interworking Profiles: SP-General

[Add](#)

Interworking Profiles	
cs2100	
avaya-ru	
OCS-Edge-Server	
cisco-ccm	
cups	
OCS-FrontEnd-Server	
Avaya-SM	
Avaya-IPO	
Avaya-CS1000	
Avaya-CM	
<b>SP-General</b>	

[Click here to add a description.](#)

**General**

Timers

Privacy

URI Manipulation

Header Manipulation

Advanced

General	
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
<b>Refer Handling</b>	<b>Yes</b>
URI Group	None
Send Hold	No
Delayed Offer	Yes
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
Prack Handling	No
Allow 18X SDP	No
T.38 Support	Yes
URI Scheme	SIP
Via Header Format	RFC3261

[Edit](#)

---

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