



Application Notes for Configuring Avaya Aura® Communication Manager Rel. 7.1, Avaya Aura® Session Manager Rel. 7.1 and Avaya Session Border Controller for Enterprise Rel. 7.2 to support 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 Rel. 7.1, Avaya Aura® Session Manager Rel. 7.1 and Avaya Session Border Controller for Enterprise Rel. 7.2, to interoperate with the Frontier Communications SIP Trunking service.

The Frontier Communications SIP Trunking service provide customers with PSTN access via a SIP trunk between the enterprise and the service provider's 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 Rel. 7.1 (Communication Manager), Avaya Aura® Session Manager Rel. 7.1 (Session Manager), Avaya Session Border Controller for Enterprise Rel. 7.2 (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 Communications” or “Frontier” 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 Frontier Communications 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.

2.1. Interoperability Compliance Testing

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

- Response to SIP OPTIONS queries.
- Incoming calls from the PSTN were routed to DID numbers assigned by Frontier.
Incoming PSTN calls were terminated to the following endpoints: Avaya 96x1 Series IP Deskphones (H.323 and SIP), 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's 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.
- Voicemail and DTMF tone support (leaving and retrieving voice mail messages, etc.).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response systems).
- 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.
- 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.

The following items were not tested:

- Inbound toll-free calls, outbound Toll-Free calls, 911 calls (emergency), “0” calls (Operator), 411 Directory Assistance, International calls and 0+10 digits calls (Operator Assisted) were not tested.
- The SIP REFER method for call redirection is not supported by Frontier, refer to **Section 2.2**.

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:

- **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 were successfully tested using the G.711 pass-through method. 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.
- **SIP OPTIONS:** SIP OPTIONS messages sent by Frontier to the enterprise contained a no-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.
- **SIP REFER:** During call transfers scenarios to the PSTN, with REFER enabled in Communication Manager (Network Call Redirection set to Y, refer to **Section 5.7**), Frontier accepted the REFER messages sent by Communication Manager with 202 Accepted, as expected, after the 202 Accepted message Frontier would send a NOTIFY message to Communication Manager with “403 Forbidden” embedded within the NOTIFY message, this resulted in the SIP trunk resources (SIP trunk channels) not being release after the call was successfully transferred to the PSTN. This issue was reported to Frontier, currently Frontier does not support the SIP REFER method for call redirection. The testing was done with SIP REFER disabled in Communication Manager (Network Call Redirection set to N, refer to **Section 5.7**).
- **Incorrect Call Display on call transfers to the PSTN Phone:** Call display was not properly updated on PSTN phones involved in a 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 IP Office station that initiated the transfer was displayed.
- **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.
- **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-Global-Session-ID, AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-id, P-Charging-Vector and P-Location (**Section 6.4**).

2.3. Support

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

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.

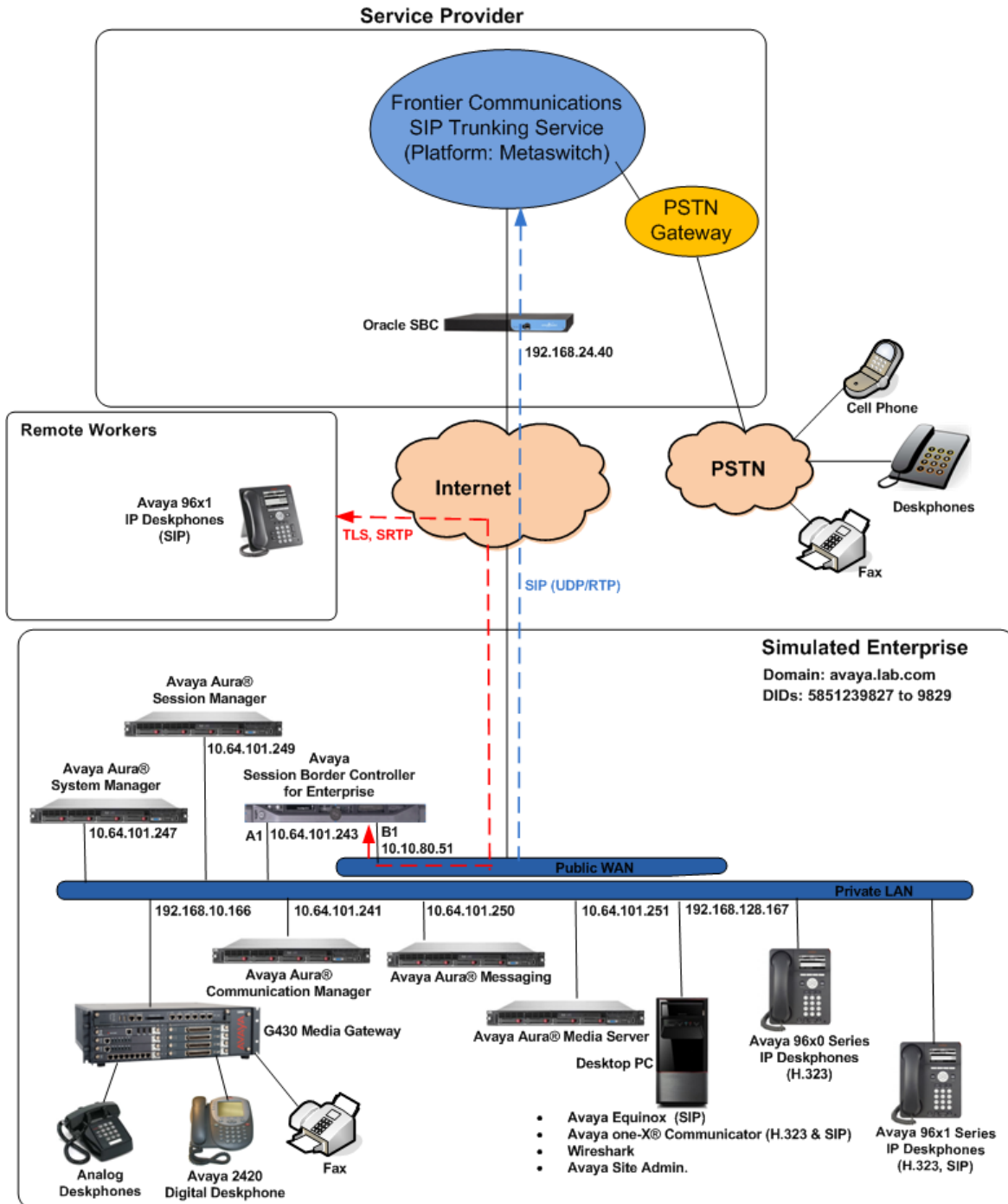


Figure 1: Avaya SIP Enterprise Solution connected to Frontier SIP Trunking Service

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 G430 Media Gateway.
- Avaya 96x1 Series IP Deskphones (H.323 and SIP).
- Avaya one-X® Communicator softphones (H.323 and SIP).
- Avaya Equinox softphone (SIP).
- Avaya digital and analog telephones.

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 [9] 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 (in this case Communication Manager) and on which link to send the call. Once the call arrived at Communication Manager, further incoming call treatment, such as incoming digit translation was performed.

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 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 7.1 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.

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 network SIP Trunking service, they are not included in these Application Notes.

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.

For the compliance testing associated with this Application Notes, the interface between the Avaya system and the Frontier network did not include the use of any specific encryption features, UDP Transport for signaling and RTP for media was used between the Avaya system and the Frontier network across the SIP trunk. TLS transport for signaling and SRTP for media was used inside of the enterprise (private network side, in between Avaya components).

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Communication Manager	7.1.1.0.0 (01.0.532.0-23985)
Avaya Aura® Session Manager	7.1.1.0 (7.1.1.0.711008)
Avaya Aura® System Manager	7.1.1.0 Build No. 7.1.0.0.1125193 Software Update Rev. No. 7.1.1.0.046931
Avaya Session Border Controller for Enterprise	ASBCE 7.2 7.2.0.0-18-13712
Avaya Aura® Messaging	7.0 Service Pack 0 (MSG-00.0.441.0-017_0004)
Avaya Aura® Media Server	7.8.0.333 SP5 7.8.0.333_2017.07.17
Avaya G430 Media Gateway	G430_sw_38_20_1
Avaya 96x1 Series IP Deskphones (SIP)	Version 7.1.1.0.9
Avaya 96x1 Series IP Deskphones (H.323)	Version 6.6506
Avaya one-X® Communicator (H.323, SIP)	6.2.12.04-SP12
Avaya Equinox (SIP)	3.3.1.60
Avaya 2420 Series Digital Deskphones	N/A
Avaya 6210 Analog Deskphones	N/A
Frontier Communications	
Metaswitch cCFS (Clustered Call Feature Server)	9.3.20
Oracle 3820 Session Border Controller	6.4

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 network 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 Aura® 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 captures 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 **24000** 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 1
      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 7
      Maximum Administered SIP Trunks: 24000 120
Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
      Maximum Number of DS1 Boards with Echo Cancellation: 522 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**.

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

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

      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.

```
change 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
```

5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager (**procr**) and the Session Manager security module (**SM**). These node names will be needed for defining the service provider signaling group in **Section 5.6**.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
ASBCE A1	10.64.101.243	
SM	10.64.101.249	
default	0.0.0.0	
media server	10.64.101.251	
procr	10.64.101.241	
procr6	::	
(6 of 6 administered node-names were displayed)		
Use 'list node-names' command to see all the administered node-names		
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name		

5.4. 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 supports audio codecs *G.711MU* and *G.729*.

change ip-codec-set 2 Page 1 of 2

IP MEDIA PARAMETERS

Codec Set: 2

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

Media Encryption

1: 1-srtp-aescm128-hmac80

2: none

3:

4:

5:

Encrypted SRTCP: best-effort

On **Page 2**, set the **Fax Mode** to *t.38-G711-fallback* (refer to **Section 2.2**).

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

IP MEDIA PARAMETERS

Allow Direct-IP Multimedia? n

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

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: <u>avaya.lab.com</u>	
Name: <u>SP Region</u>	Stub Network Region: <u>n</u>	
MEDIA PARAMETERS		
Codec Set: <u>2</u>	Intra-region IP-IP Direct Audio: <u>yes</u>	
	Inter-region IP-IP Direct Audio: <u>yes</u>	
UDP Port Min: <u>2048</u>	IP Audio Hairpinning? <u>n</u>	
UDP Port Max: <u>3349</u>		
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: <u>46</u>		
Audio PHB Value: <u>46</u>		
Video PHB Value: <u>26</u>		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: <u>6</u>		
Audio 802.1p Priority: <u>6</u>		
Video 802.1p Priority: <u>5</u>		
H.323 IP ENDPOINTS		AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? <u>y</u>		RSVP Enabled? <u>n</u>
Idle Traffic Interval (sec): <u>20</u>		
Keep-Alive Interval (sec): <u>5</u>		
Keep-Alive Count: <u>5</u>		

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	A	M		
dst rgn	codec set	direct WAN	WAN-BW-limits Units	Video Total Norm	Intervening Prio Shr Regions	Dyn CAC	A	G	R	L	t	c	e	t
1	<u>2</u>	y	NoLimit				n							
2	2									all				
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? <u>n</u>	Transport Method: <u>tls</u>		
Q-SIP? <u>n</u>			
IP Video? <u>n</u>	Enforce SIPS URI for SRTP? <u>y</u>		
Peer Detection Enabled? <u>y</u>	Peer Server: SM		
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? <u>y</u>			
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? <u>n</u>			
Alert Incoming SIP Crisis Calls? <u>n</u>			
Near-end Node Name: <u>procr</u>	Far-end Node Name: <u>SM</u>		
Near-end Listen Port: <u>5071</u>	Far-end Listen Port: <u>5071</u>		
	Far-end Network Region: <u>2</u>		
Far-end Domain: <u>avaya.lab.com</u>			
Incoming Dialog Loopbacks: <u>eliminate</u>	Bypass If IP Threshold Exceeded? <u>n</u>		
DTMF over IP: <u>rtp-payload</u>	RFC 3389 Comfort Noise? <u>n</u>		
Session Establishment Timer(min): <u>3</u>	Direct IP-IP Audio Connections? <u>y</u>		
Enable Layer 3 Test? <u>n</u>	IP Audio Hairpinning? <u>n</u>		
H.323 Station Outgoing Direct Media? <u>n</u>	Initial IP-IP Direct Media? <u>n</u>		
	Alternate Route Timer(sec): <u>6</u>		

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 21
                                     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 21
Group Type: sip	
TRUNK PARAMETERS	
Unicode Name: <u>auto</u>	
Redirect On OPTIM Failure: <u>5000</u>	
SCCAN? <u>n</u>	Digital Loss Group: <u>18</u>
Preferred Minimum Session Refresh Interval(sec): <u>600</u>	
Disconnect Supervision - In? <u>y</u> Out? <u>y</u>	
XOIP Treatment: <u>auto</u> Delay Call Setup When Accessed Via IGAR? <u>n</u>	
Caller ID for Service Link Call to H.323 1xC: <u>station-extension</u>	

On Page 3:

- Set the **Numbering Format** field to *private*. 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, the **Numbering Format** was set to *private* and the **Numbering Format** in the route pattern was set to *unk-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 21
TRUNK FEATURES
  ACA Assignment? n                                     Measured: none
                                                         Maintenance Tests? y

  Suppress # Outpulsing? n   Numbering Format: private
                                                         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 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** and **2.2**).
- 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.
- Set the **Convert 180 to 183 for Early Media?** to *y*.
- 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 21
PROTOCOL VARIATIONS		
Mark Users as Phone?	<u>n</u>	
Prepend '+' to Calling/Alerting/Diverting/Connected Number?	<u>n</u>	
Send Transferring Party Information?	<u>n</u>	
Network Call Redirection?	<u>n</u>	
Send Diversion Header?	<u>y</u>	
Support Request History?	<u>n</u>	
Telephone Event Payload Type:	<u>101</u>	
Convert 180 to 183 for Early Media?	<u>y</u>	
Always Use re-INVITE for Display Updates?	<u>n</u>	
Identity for Calling Party Display:	<u>P-Asserted-Identity</u>	
Block Sending Calling Party Location in INVITE?	<u>n</u>	
Accept Redirect to Blank User Destination?	<u>n</u>	
Enable Q-SIP?	<u>n</u>	
Interworking of ISDN Clearing with In-Band Tones:	<u>keep-channel-active</u>	
Request URI Contents:	<u>may-have-extra-digits</u>	

5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since private numbering was selected to define the format of this number (**Section 5.7**), use the **change private-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, three DID numbers were assigned by the service provider for testing. 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 private-numbering 1					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len	
4	3			4	Total Administered: 5
4	5			4	Maximum Entries: 540
4	3041	2	5851239827	10	
4	3042	2	5851239828	10	
4	3044	2	5851239829	10	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	

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 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	5851239827	10	3041	
public-ntwrk	10	5851239828	10	3042	
public-ntwrk	10	5851239829	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	—	—	—	—	

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 **unk-unk**. All calls using this route pattern will use the private 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 3	
Pattern Number: 2															Pattern Name: <u>Serv. Provider</u>	
SCCAN? <u>n</u>															Secure SIP? <u>n</u>	
															Used for SIP stations? <u>n</u>	
Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Dgts					DCS/ QSIG Intw	IXC			
1:	<u>2</u>	<u>0</u>	<u>1</u>								<u>n</u>	<u>user</u>				
2:											<u>n</u>	<u>user</u>				
3:											<u>n</u>	<u>user</u>				
4:											<u>n</u>	<u>user</u>				
5:											<u>n</u>	<u>user</u>				
6:											<u>n</u>	<u>user</u>				

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

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® Session Manager

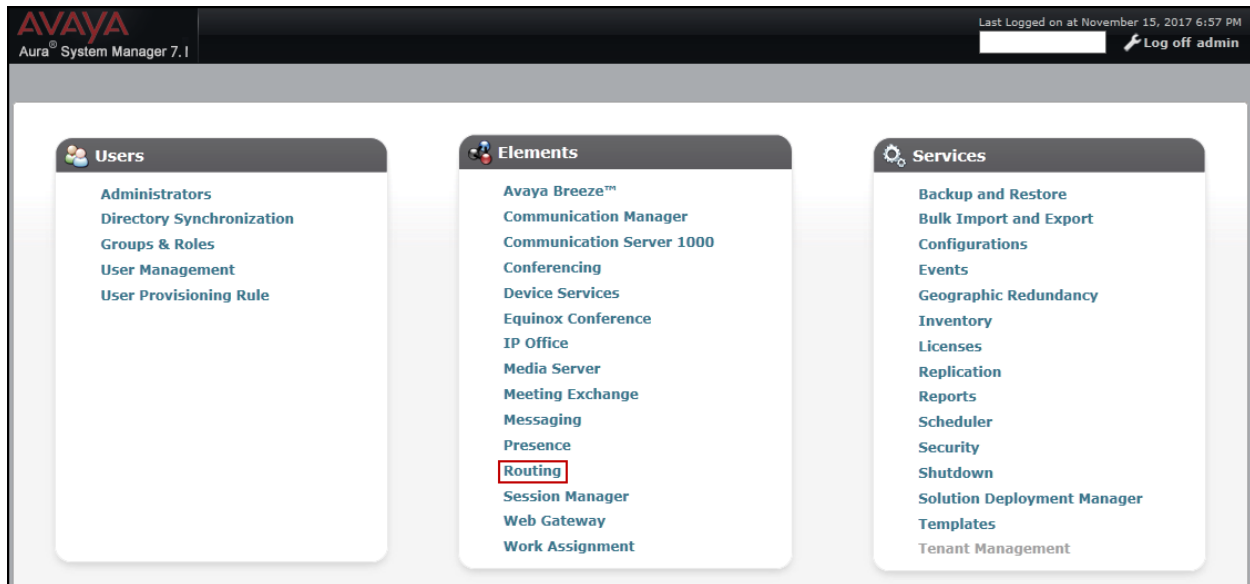
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- 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 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.

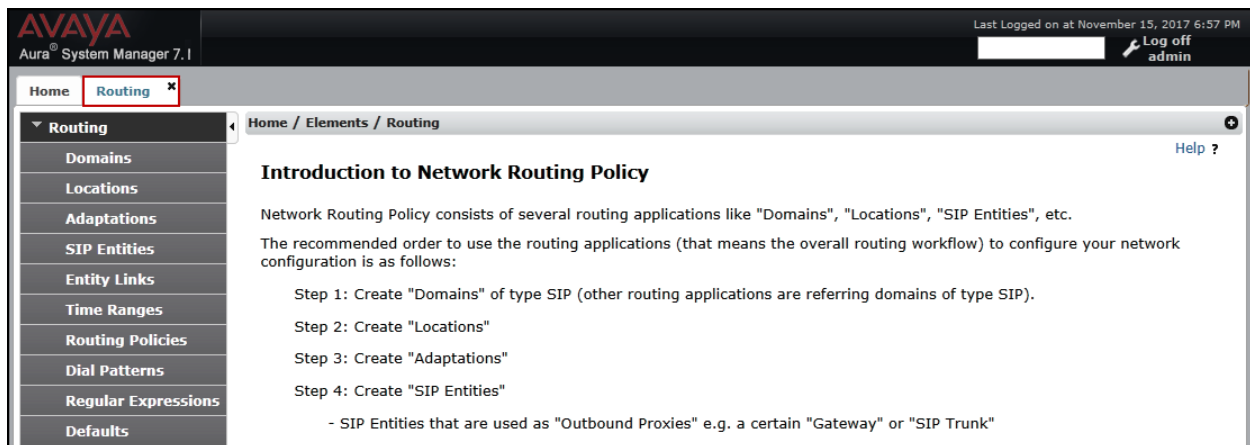
The following sections assume that the initial configuration of Session Manager and System Manager has already been completed, and that network connectivity exists between System Manager and Session Manager.

6.1. 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; click on **Routing**.



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.



6.2. 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 7.1 interface. The top header includes the Avaya logo and 'Aura System Manager 7.1'. The left navigation pane has 'Routing' selected, and 'Domains' is highlighted. The main area is titled 'Domain Management' and shows a table with one entry: 'avaya.lab.com' of type 'sip' with notes 'HG V-Domain'.

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

6.3. 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 7.1 web interface. The top navigation bar includes the Avaya logo, the text "Aura® System Manager 7.1", and a "Last Logged on at November 15, 2017 6:57 PM" timestamp. A "Log off admin" button is visible. The left sidebar contains a menu with "Routing" selected and highlighted. The main content area shows the "Location Details" page for "Session Manager". The "General" section includes a red-bordered input field for "* Name:" containing "Session Manager" and a "Notes:" field containing "VMware Session Manager". Below this is the "Dial Plan Transparency in Survivable Mode" section, which has an "Enabled:" checkbox (unchecked), a "Listed Directory Number:" field, and an "Associated CM SIP Entity:" field. "Commit" and "Cancel" buttons are located in the top right corner 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.

This screenshot shows the same Avaya Aura System Manager 7.1 interface as the previous one, but with the "Location Details" page for "Communication Manager". The "General" section features a red-bordered input field for "* Name:" containing "Communication Manager" and a "Notes:" field containing "VMware Communication Manager". The "Dial Plan Transparency in Survivable Mode" section remains the same, with the "Enabled:" checkbox unchecked and empty fields for "Listed Directory Number:" and "Associated CM SIP Entity:". The "Commit" and "Cancel" buttons are still present in the top right.

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 7.1 web interface. The top header shows the Avaya logo and 'Aura System Manager 7.1'. The user is logged in as 'admin' on November 15, 2017, at 6:57 PM. The breadcrumb trail is 'Home / Elements / Routing / Locations'. The left sidebar menu has 'Locations' selected under the 'Routing' category. The main content area is titled 'Location Details' and includes 'Commit' and 'Cancel' buttons. The 'General' section contains a required field '* Name:' with the value 'Avaya SBCE' and a 'Notes:' field with the value 'VMware Avaya SBCE'. The 'Dial Plan Transparency in Survivable Mode' section has an 'Enabled:' checkbox (unchecked), a 'Listed Directory Number:' field, and an 'Associated CM SIP Entity:' field.

AVAYA
Aura System Manager 7.1

Last Logged on at November 15, 2017 6:57 PM
Log off admin

Home Routing

Home / Elements / Routing / Locations

Location Details

Commit Cancel

Help ?

General

* Name: Avaya SBCE

Notes: VMware Avaya SBCE

Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

6.4. Adaptations

In order to improve interoperability with third party elements, Session Manager 7.0 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-Global-Session-ID, 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.

The screenshot displays the Avaya Aura System Manager 7.1 web interface. The top navigation bar includes the Avaya logo, the text 'Aura® System Manager 7.1', and a user session summary showing 'Last Logged on at November 15, 2017 6:57 PM' and a 'Log off admin' button. The left sidebar contains a menu with 'Routing' selected, and sub-items like 'Domains', 'Locations', 'Adaptations', 'SIP Entities', 'Entity Links', 'Time Ranges', 'Routing Policies', 'Dial Patterns', 'Regular Expressions', and 'Defaults'. The main content area is titled 'Adaptation Details' and includes 'Commit' and 'Cancel' buttons. Under the 'General' tab, the 'Adaptation Name' is 'CM_Outbound_Header_Removal', the 'Module Name' is 'DigitConversionAdapter', and the 'Module Parameter Type' is 'Name-Value Parameter'. Below this is a table with columns 'Name' and 'Value'. The table contains one entry with 'Name' as 'eRHdrs' and 'Value' as '"Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-id, P-'. The interface also features an 'Add' button, a 'Remove' button, and a 'Select : All, None' option at the bottom of the table.

Home / Elements / Routing / Adaptations

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, P-

Select : All, None

6.5. SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Communication Manager and the Avaya SBCE. 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 and *SIP Trunk* (or *Other*) for the Avaya SBCE.
- **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 6.3**.
- **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 shows the Avaya Aura System Manager 7.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 7.1', and a user session bar showing 'Last Logged on at November 15, 2017 6:57 PM' and a 'Log off admin' button. The left sidebar contains a navigation menu with 'Routing' selected, and sub-items: Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area shows the 'SIP Entity Details' form. The breadcrumb trail is 'Home / Elements / Routing / SIP Entities'. The form has a 'Commit' and 'Cancel' button at the top right. The 'General' tab is active. The form fields are: '* Name: Session Manager', '* FQDN or IP Address: 10.64.101.249', 'Type: Session Manager' (dropdown), 'Notes: VMware Session Manager', 'Location: Session Manager' (dropdown), 'Outbound Proxy: ' (empty dropdown), 'Time Zone: America/New_York' (dropdown), and 'Minimum TLS Version: Use Global Setting' (dropdown).

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 6.3**.

The screenshot displays the Avaya Aura System Manager 7.1 web interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.1', and a user session summary showing 'Last Logged on at November 15, 2017 6:57 PM' and a 'Log off admin' button. A breadcrumb trail reads 'Home / Elements / Routing / SIP Entities'. The left sidebar contains a menu with 'Routing' selected, and sub-items including Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and includes 'Commit' and 'Cancel' buttons. Under the 'General' tab, the form contains the following fields: '* 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), and 'Time Zone' (America/New_York). Red rectangular boxes highlight the 'Name', 'FQDN or IP Address', 'Location', and 'Time Zone' fields.

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 6.4** was selected.
- Select the location that applies to the SIP Entity being created, defined in **Section 6.3**.

AVAYA
Aura® System Manager 7.1

Last Logged on at November 15, 2017 6:57 PM
Log off
admin

Home Routing

Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Home / Elements / Routing / SIP Entities

SIP Entity Details

Commit Cancel

Help ?

General

* 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

6.6. Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created; one to the Communication Manager for use only by service provider traffic and one to the Avaya SBCE. 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 6.5**).
- **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 6.5**).
- **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.

AVAYA
Aura® System Manager 7.1

Last Logged on at November 15, 2017 6:57 PM
Log off admin

Home Routing

Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Home / Elements / Routing / Entity Links

Entity Links

Commit Cancel

1 Item Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy
<input type="checkbox"/>	* Session_Manager_Ch	* Session Manager	TLS	* 5071	* Communication Manager Trunk 2	* 5071	<input type="checkbox"/>	trusted

Select : All, None

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

AVAYA
Aura® System Manager 7.1

Last Logged on at November 15, 2017 6:57 PM
Log off admin

Home Routing

Home / Elements / Routing / Entity Links

Entity Links

Commit Cancel

1 Item Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy
<input type="checkbox"/>	* Session_Manager_AS *	Session Manager	TLS	* 5061 *	Avaya SBCE	* 5061 *	<input type="checkbox"/>	trusted

Select : All, None

6.7. Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Two routing policies were added: an incoming policy with Communication Manager as the destination, and an outbound policy to the Avaya SBCE. 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 6.5**) 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 and the Avaya SBCE.

AVAYA
Aura® System Manager 7.1

Last Logged on at November 15, 2017 6:57 PM

Home Routing

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel

Help ?

General

* Name: To CM Trunk 2

Disabled: ☐

* Retries: 0

Notes: For inbound calls to CM via Trunk

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Communication Manager Trunk 2	10.64.101.241	CM	Used for SP Testing

AVAYA
Aura® System Manager 7.1

Last Logged on at November 21, 2017 8:32 AM

Home Routing

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel

Help ?

General

* Name: Avaya SBCE

Disabled: ☐

* Retries: 0

Notes: For outbound calls to SP via ASB

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Avaya SBCE	10.64.101.243	SIP Trunk	VMware Avaya SBCE

6.8. 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. 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.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria, or select “**ALL**” to route incoming calls to all SIP domains.
- **Notes:** Add a brief description (optional).
- In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria (**Section 6.3**).
- Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria (**Section 6.7**). Click **Select** (not shown).
- Click **Commit** to save.

The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. In the example, calls to 10 digit numbers starting with **585**, arriving from location **Avaya SBCE**, used route policy **To CM Trunk 2** to Communication Manager. The SIP Domain was set to **avaya.lab.com**

AVAYA
Aura® System Manager 7.1

Last Logged on at April 23, 2018 8:16 PM
Log off admin

Home Routing

Home / Elements / Routing / Dial Patterns

Dial Pattern Details [Commit] [Cancel] Help ?

General

* Pattern: 585
* Min: 3
* Max: 10

Emergency Call: ☐
Emergency Priority: 1
Emergency Type:
SIP Domain: avaya.lab.com
Notes:

Originating Locations and Routing Policies

Add Remove Filter: Enable

1 Item

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Avaya SBCE	VMware Avaya SBCE	To CM Trunk 2	0	<input type="checkbox"/>	Communication Manager Trunk 2	For inbound calls to CM via Trunk 2

Select : All, None

Denied Originating Locations

Add Remove Filter: Enable

0 Items

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

Repeat this procedure as needed to define additional dial patterns for other range of numbers assigned by the service provider to the enterprise, to be routed to Communication Manager.

The example in this screen shows the 11 digit dialed numbers for outbound international 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**.

AVAYA
Aura® System Manager 7.1

Last Logged on at April 23, 2018 8:16 PM
Log off admin

Home Routing

Home / Elements / Routing / Dial Patterns

Dial Pattern Details [Commit] [Cancel] Help ?

General

* Pattern: 1
* Min: 11
* Max: 11

Emergency Call: ☐
Emergency Priority: 1
Emergency Type:
SIP Domain: avaya.lab.com
Notes:

Originating Locations and Routing Policies

Add Remove Filter: Enable

1 Item

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Communication Manager	VMware Communication Manager	Avaya SBCE	0	<input type="checkbox"/>	Avaya SBCE	For outbound calls to SP via ASBCE

Select : All, None

Denied Originating Locations

Add Remove Filter: Enable

0 Items

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

Repeat this procedure as needed, to define additional dial patterns for PSTN numbers to be routed to the service provider's network via the Avaya SBCE.

7. 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 **Additional References** section.

7.1. 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 image shows the login page for the Avaya Session Border Controller for Enterprise. On the left, there is a large red 'AVAYA' logo and the text 'Session Border Controller for Enterprise' below it. On the right, there is a 'Log In' section with a 'Username:' label, a text input field, and a 'Continue' button. Below the login fields, there is a 'WELCOME TO AVAYA SBC' message, a disclaimer about unauthorized access, a consent statement, and a copyright notice for 2011-2017 Avaya Inc.

AVAYA

**Session Border Controller
for Enterprise**

Log In

Username:

WELCOME TO AVAYA SBC

Unauthorized access to this machine is prohibited. This system is for the use authorized users only. Usage of this system may be monitored and recorded by system personnel.

Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible evidence of criminal activity, system personnel may provide the evidence from such monitoring to law enforcement officials.

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Once logged in, the Dashboard screen is presented. 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.

AlarmsIncidentsStatus ▾Logs ▾DiagnosticsUsersSettings ▾Help ▾Log Out

Session Border Controller for EnterpriseAVAYA

DashboardAdministrationBackup/RestoreSystem Management▶ Global Parameters▶ Global Profiles▶ PPM Services▶ Domain Policies▶ TLS Management▶ Device Specific Settings

Dashboard

This system contains one or more Avaya demo certificates. These certificates have been compromised and should not be used for any production traffic.

The following certificates are expired:

- Rapid_SSL_Cert.crt (Certificate)

System Time	12:45:10 PM EST	Refresh
Version	7.2.0.0-18-13712	
Build Date	Thu Jun 1 00:12:50 UTC 2017	
License State	✔ OK	
Aggregate Licensing Overages	0	
Peak Licensing Overage Count	0	
Last Logged in at	11/21/2017 12:42:59 EST	
Failed Login Attempts	0	

EMS
Avaya_SBCE

None found.

Avaya_SBCE : No Subscriber Flow Matched

HG; Reviewed:
SPOC 5/23/2018

Solution & Interoperability Test Lab Application Notes
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FronCMSM71SBC72

7.2. System Management

To view current system information, select **System Management** on the left navigation pane. A list of installed devices is shown in the **Devices** tab on the right pane. In the reference configuration, a single 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 web interface. The top navigation bar includes 'Alarms 1', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header displays 'Session Border Controller for Enterprise' and the 'AVAYA' logo. The left sidebar contains a navigation menu with 'Dashboard', 'Administration', 'Backup/Restore', 'System Management' (highlighted with a red box), and several expandable sections: 'Global Parameters', 'Global Profiles', 'PPM Services', 'Domain Policies', 'TLS Management', and 'Device Specific Settings'. The main content area is titled 'System Management' and features five tabs: 'Devices' (selected), 'Updates', 'SSL VPN', 'Licensing', and 'Key Bundles'. Below the tabs is a table with the following data:

Device Name	Management IP	Version	Status						
Avaya_SBCE	[Blurred]	7.2.0.0-18-13712	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 2000
Requested: 2000
Advanced Sessions 2000
Requested: 2000
Scopia Video Sessions 500
Requested: 500
CES Sessions 0
Requested: 0
Transcoding Sessions 0
Requested: 0
Encryption ☒

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 are the ones used for the SIP trunk to Frontier, 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.

7.3. 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 **Device Specific Settings** on the left-side menu. Under **Devices** in the center pane, select the device being managed, *Avaya_SBCE* in the sample configuration. 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.

Alarms 1 Incidents Status Logs Diagnostics Users Settings Help Log Out

Session Border Controller for Enterprise

AVAYA

Dashboard
Administration
Backup/Restore
System Management
 Global Parameters
 Global Profiles
 PPM Services
 Domain Policies
 TLS Management
 Device Specific Settings
 Network Management
Media Interface

Network Management: Avaya_SBCE

Devices
Avaya_SBCE

Interfaces Networks

Add

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

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.

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

7.4. 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 **Device Specific Settings** menu on the left-hand side, select the device being managed and 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**.

Add Media Interface

Name: Private_med

IP Address: Network_A1 (A1, VLAN 0)

Port Range: 35000 - 40000

TLS Profile: None

Finish

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 the following fields and values:

- Name:** Public_med
- IP Address:** Network_B1 (B1, VLAN 0) (dropdown menu) and 10.10.80.51 (text field)
- Port Range:** 35000 (text field) - 40000 (text field)
- TLS Profile:** None (dropdown menu)

A red rectangle highlights the Name, IP Address, and Port Range fields. A "Finish" button is located at the bottom center of the dialog.

7.5. 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 **Device Specific Settings** menu on the left-hand side, select the device being managed and 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 6.6**.
- Select a **TLS Profile** (See Note below).
- Click **Finish**.

The screenshot shows the 'Add Signaling Interface' configuration window. The form includes the following fields and options:

- Name:** Private_sig
- IP Address:** Network_A1 (A1, VLAN 0) (dropdown), 10.64.101.243 (dropdown)
- TCP Port:** (text field), Leave blank to disable
- UDP Port:** (text field), Leave blank to disable
- TLS Port:** 5061 (text field), Leave blank to disable
- TLS Profile:** New_ServiceProvider_Server_TLS (dropdown)
- Enable Shared Control:** ☐
- Shared Control Port:** (text field)
- Finish:** (button)

Note - The configuration tasks required to support TLS transport for signaling and SRTP for media inside of the enterprise (private network side) are beyond the scope of these Application Notes; hence they are not discussed in this document

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 this is the protocol and port used by the Avaya SBCE to listen to the service provider's SIP traffic.
- 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 Leave blank to disable	5060
TLS Port Leave blank to disable	
TLS Profile	None
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

Finish

7.6. 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).

7.6.1. 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 lists various configuration areas, with 'Global Profiles' expanded and 'Server Interworking' selected. The main content area is titled 'Interworking Profiles: avaya-ru' and features an 'Add' button and a 'Clone' button. A warning message 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 configuration parameters:

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 and a clear 'x' button)
- Finish** button

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**.

Editing Profile: Avaya-SM

General

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

Send Hold ☐

Delayed Offer ☐

3xx Handling ☐

Diversion Header Support ☐

Delayed SDP Handling ☐

Re-Invite Handling ☐

Prack Handling ☐

Allow 18X SDP ☐

T.38 Support ☒

URI Scheme ☒ SIP ☐ TEL ☐ ANY

Via Header Format ☒ RFC3261
☐ RFC2543

Finish

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

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

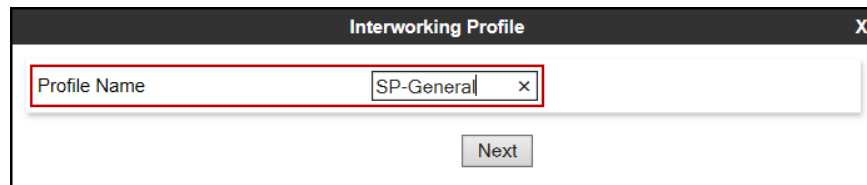
The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. A left-hand navigation menu lists various configuration areas, with 'Global Profiles' expanded to show 'Server Interworking' selected. The main content area is titled 'Interworking Profiles: Avaya-SM' and features a list of profiles on the left, including 'Avaya-SM' which is highlighted. The 'Advanced' tab is active, showing settings for '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), and 'Relay INVITE Replace for SIPREC' (No). Below these, the 'DTMF' section shows 'DTMF Support' set to 'None'. Buttons for 'Add', 'Rename', 'Clone', 'Delete', and 'Edit' are visible throughout the interface.

Interworking Profiles	General	Timers	Privacy	URI Manipulation	Header Manipulation	Advanced
cs2100	Click here to add a description.					
avaya-ru	Record Routes: Both Sides					
OCS-Edge-Server	Include End Point IP for Context Lookup: Yes					
cisco-ccm	Extensions: Avaya					
cups	Diversion Manipulation: No					
OCS-FrontEnd...	Has Remote SBC: Yes					
Avaya-SM	Route Response on Via Port: No					
SP-General	Relay INVITE Replace for SIPREC: No					
Avaya-IPO	DTMF					
Avaya-CS1000	DTMF Support: None					
Avaya-CM	Edit					

7.6.2. Server Interworking Profile – Service Provider

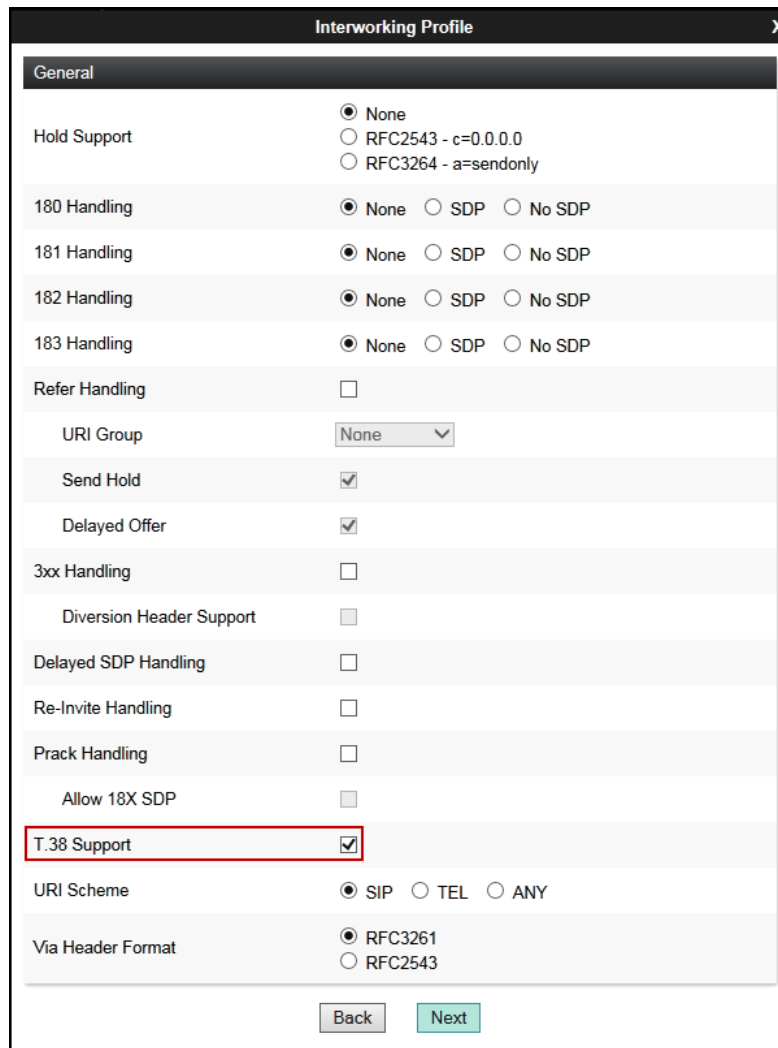
A second interworking profile in the direction of the SIP trunk was created, by adding a new profile in this case. Select **Global Profiles → Server Interworking** on the left navigation pane and click **Add** (not shown).

- 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" which contains the text "SP-General". A red rectangular box highlights the "Profile Name" field and its value. Below the input field, there is a "Next" button.

On the **General** tab, check **T.38 Support**. Click **Next**, 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 rectangular box. Other settings are at their defaults: "Hold Support" is set to "None", "180 Handling" through "183 Handling" are set to "None", "Refer Handling" is unchecked, "URI Group" is set to "None", "Send Hold" and "Delayed Offer" are checked, "3xx Handling", "Diversion Header Support", "Delayed SDP Handling", "Re-Invite Handling", "Prack Handling", and "Allow 18X SDP" are unchecked, "URI Scheme" is set to "SIP", and "Via Header Format" is set to "RFC3261". The "Next" button is highlighted in green.

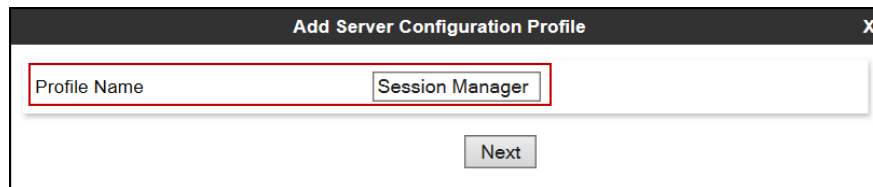
7.7. Server Configuration

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

7.7.1. Server Configuration Profile – Enterprise

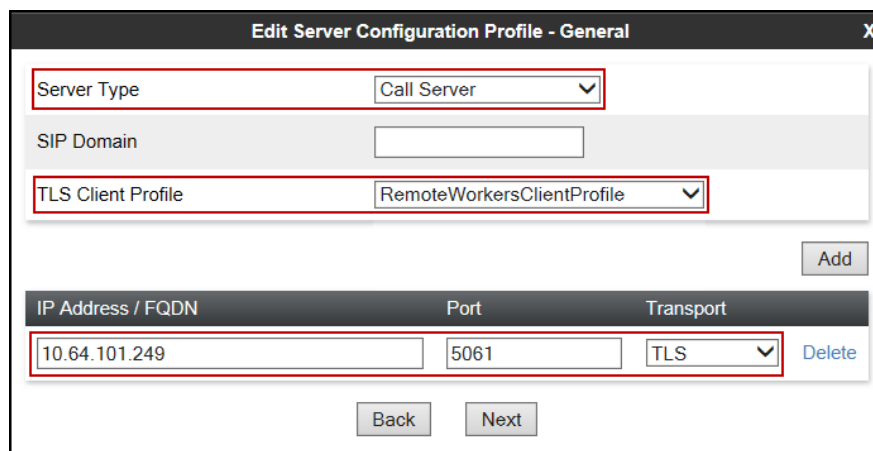
From the **Global Profiles** menu on the left-hand navigation pane, select **Server Configuration** and click the **Add** button (not shown) to add a new profile for the Call Server.

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



The screenshot shows a dialog box titled "Add Server Configuration Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" which contains the text "Session Manager". Below this field is a "Next" button.

- On the **Edit Server Configuration Profile – General** tab select **Call Server** from the drop down menu under the **Server Type**.
- Select a **TLS Profile**.
- On the **IP Addresses / FQDN** field, enter the IP address of the Session Manager Security Module (**Section 6.5**).
- 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 6.6**.
- Click **Next**.



The screenshot shows a dialog box titled "Edit Server Configuration Profile - General" with a close button (X) in the top right corner. The dialog contains several fields and buttons:

- Server Type**: A dropdown menu set to "Call Server".
- SIP Domain**: An empty text input field.
- TLS Client Profile**: A dropdown menu set to "RemoteWorkersClientProfile".
- Add**: A button located to the right of the TLS Client Profile dropdown.
- IP Address / FQDN**: A text input field containing "10.64.101.249".
- Port**: A text input field containing "5061".
- Transport**: A dropdown menu set to "TLS".
- Delete**: A button located to the right of the Transport dropdown.
- Back** and **Next**: Buttons located at the bottom center of the dialog.

- Click **Next** until the **Add Server Configuration Profile – Advanced** tab is reached (not shown).
- On the **Add Server Configuration Profile – Advanced** tab select *Avaya-SM* from the **Interworking Profile** drop down menu (Section 7.6.1).
- Click **Finish**.

Add Server Configuration Profile - Advanced X

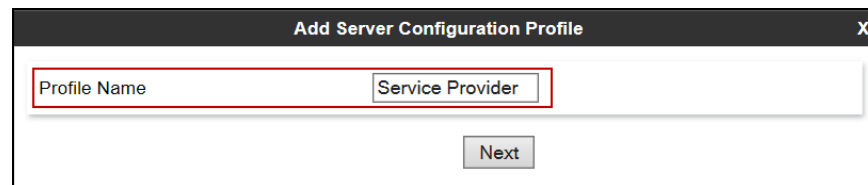
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Avaya-SM ▼
Signaling Manipulation Script	None ▼
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	5060
TLS Failover Port	5061
Tolerant	<input type="checkbox"/>
URI Group	None ▼

Back Finish

7.7.2. Server Configuration Profile – Service Provider

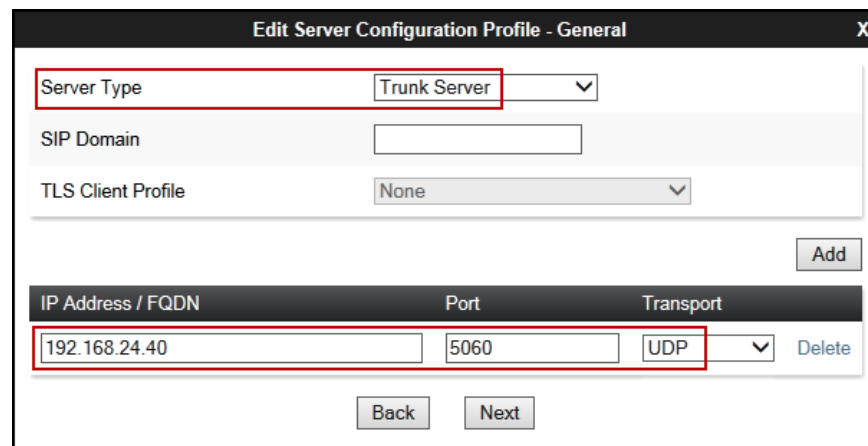
Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown).

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



The screenshot shows a dialog box titled "Add Server Configuration Profile". It contains two text input fields: "Profile Name" and "Service Provider". A "Next" button is located at the bottom right of the dialog.

- 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 the IP address of the Frontier SIP proxy server. This information was provided by Frontier.
- Enter **5060** under **Port**, and select **UDP** for **Transport** for both entries.
- Click **Next** until the **Add Server Configuration Profile – Advanced** tab is reached (not shown).



The screenshot shows a dialog box titled "Edit Server Configuration Profile - General". It contains several fields: "Server Type" (dropdown menu with "Trunk Server" selected), "SIP Domain" (text input), and "TLS Client Profile" (dropdown menu with "None" selected). Below these is an "Add" button. At the bottom, there is a table with columns "IP Address / FQDN", "Port", and "Transport". The first row contains "192.168.24.40", "5060", and "UDP". There is a "Delete" button next to the first row. At the very bottom are "Back" and "Next" buttons.

- On the **Add Server Configuration Profile – Advanced** tab select *SP-General* from the **Interworking Profile** drop down menu (Section 7.6.2).
- Click **Finish**.

Add Server Configuration Profile - Advanced X

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SP-General ▼
Signaling Manipulation Script	None ▼
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	5060
TLS Failover Port	5061
Tolerant	<input type="checkbox"/>
URI Group	None ▼

Back Finish

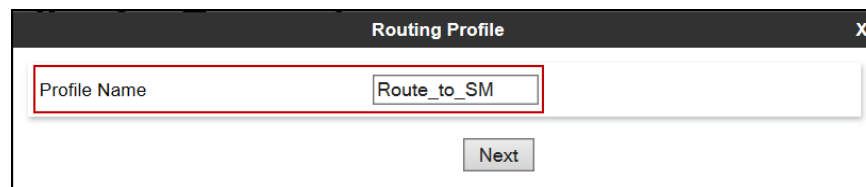
7.8. Routing

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.

7.8.1. Routing Profile – Enterprise

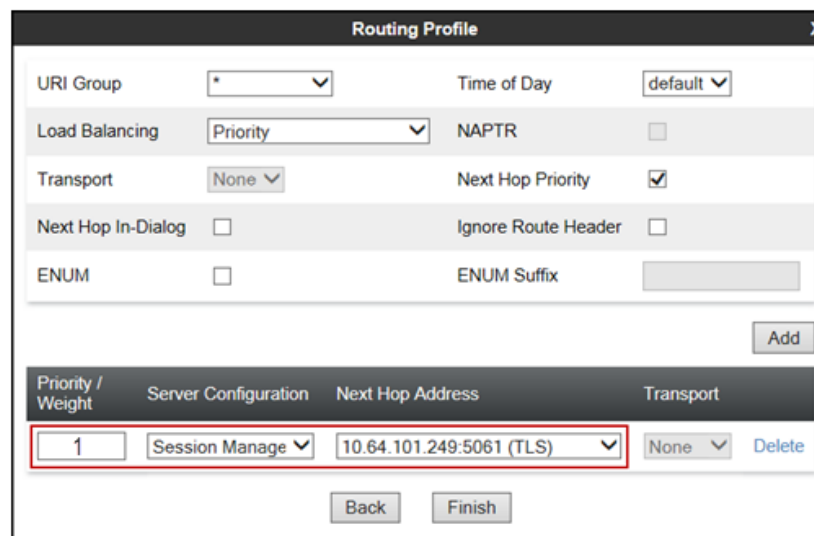
To create the inbound route, select the **Routing** tab from the **Global Profiles** menu on the left-hand side and select **Add** (not shown).

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



The screenshot shows 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 "Route_to_SM". Below the input field is a button labeled "Next".

- On the **Routing Profile** tab, click the **Add** button to enter the next-hop address.
- Under **Priority/Weight** enter **1**.
- Under **Server Configuration**, 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 7.8.1**.
- Defaults were used for all other parameters.
- Click **Finish**.



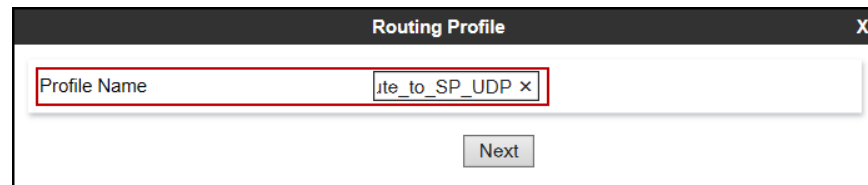
The screenshot shows the "Routing Profile" dialog box with various configuration options. The "URI Group" is set to "*", "Time of Day" is "default", "Load Balancing" is "Priority", "NAPTR" is unchecked, "Transport" is "None", "Next Hop Priority" is checked, "Next Hop In-Dialog" is unchecked, "Ignore Route Header" is unchecked, "ENUM" is unchecked, and "ENUM Suffix" is empty. An "Add" button is visible. Below these options is a table with the following columns: "Priority / Weight", "Server Configuration", "Next Hop Address", and "Transport". The table contains one row with the following values: "1", "Session Manage", "10.64.101.249:5061 (TLS)", and "None". A "Delete" button is next to the "None" value. At the bottom of the dialog are "Back" and "Finish" buttons.

Priority / Weight	Server Configuration	Next Hop Address	Transport
1	Session Manage	10.64.101.249:5061 (TLS)	None

7.8.2. Routing Profile – Service Provider

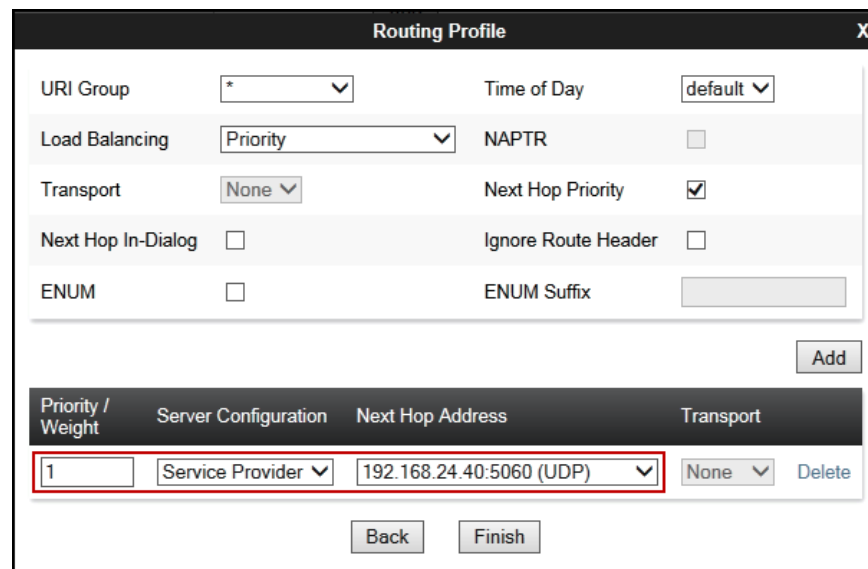
Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route.

- Enter an appropriate **Profile Name** similar to the example below.
- 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". To the right of the input field is a small "X" icon. Below the input field is a "Next" button.

- On the **Routing Profile** tab, click the **Add** button to enter the next-hop address.
- Click on the **Add** button to add a **Next-Hop Address**.
- **Server Configuration:** Select *Service Provider UDP*.
- The **Next Hop Address** is populated automatically with *192.168.24.40:5060 (UDP)* Frontier's SIP Proxy IP address, Port and Transport, Server Configuration Profile defined in **Section 7.8.2**
- Click **Finish**.



The screenshot shows a dialog box titled "Routing Profile" with a close button (X) in the top right corner. The dialog contains several configuration options:

- URI Group: * (dropdown)
- Time of Day: default (dropdown)
- Load Balancing: Priority (dropdown)
- NAPTR: ☐
- Transport: None (dropdown)
- Next Hop Priority: ☒
- Next Hop In-Dialog: ☐
- Ignore Route Header: ☐
- ENUM: ☐
- ENUM Suffix: (text input)

Below these options is an "Add" button. Under the "Add" button is a table with the following columns: Priority / Weight, Server Configuration, Next Hop Address, and Transport.

Priority / Weight	Server Configuration	Next Hop Address	Transport
1	Service Provider	192.168.24.40:5060 (UDP)	None

At the bottom of the dialog are "Back" and "Finish" buttons. The "Add" button and the table row are highlighted with a red border.

7.9. Topology Hiding

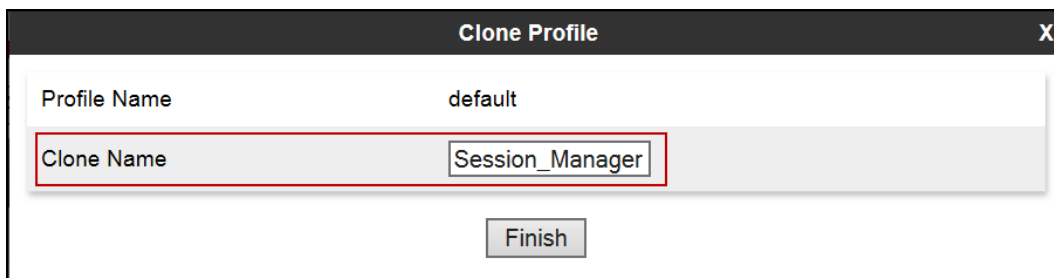
Topology Hiding is a security feature that allows the modification of several SIP headers, preventing private enterprise network information from being propagated to the untrusted public network.

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.

7.9.1. 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 'Clone Profile' dialog box. It has a title bar with 'Clone Profile' and a close button 'X'. Inside, there are two input fields: 'Profile Name' with the value 'default' and 'Clone Name' with the value 'Session_Manager'. The 'Clone Name' field is highlighted with a red border. Below the fields is a 'Finish' button.

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

- For the, **From**, **To** and **Request-Line** headers, select **Overwrite** in the **Replace Action** column and enter the enterprise SIP domain **avaya.lab.com**, in the **Overwrite Value** column of these headers, as shown below. This is the domain known by Session Manager, defined in **Section 6.2**.
- Default values were used for all other fields.
- Click **Finish**.

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

Finish

7.9.2. 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**.



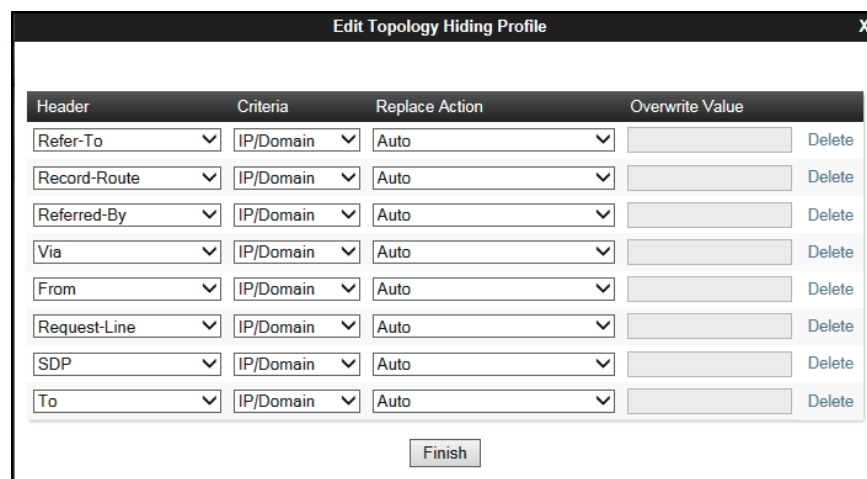
Clone Profile

Profile Name: default

Clone Name: Service_Provider

Finish

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.



Edit Topology Hiding Profile

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

Finish

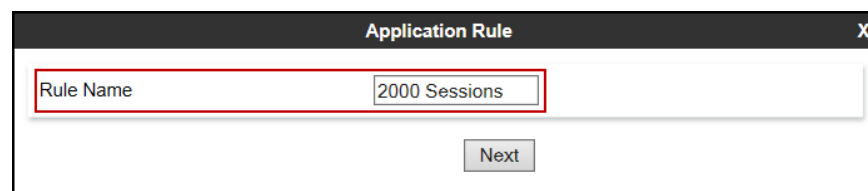
7.10. Domain Policies

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.

7.10.1. 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 and video 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**.

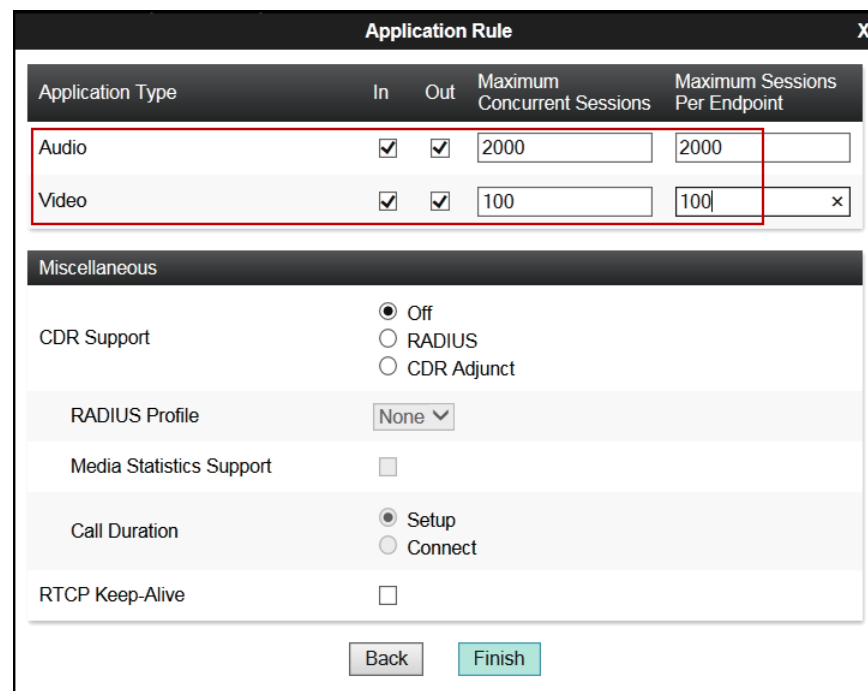


Application Rule

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 values of **2000** for Audio and **100** for Video were used in the sample configuration.
- 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 checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	100	100

Miscellaneous

CDR Support: ☒ Off, ☐ RADIUS, ☐ CDR Adjunct

RADIUS Profile: None

Media Statistics Support: ☐

Call Duration: ☒ Setup, ☐ Connect

RTCP Keep-Alive: ☐

Back Finish

7.10.2. 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, two media rules (shown below) were used; one toward Session Manager and one 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.
- Under Miscellaneous verify that **Capability Negotiation** is checked.
- Click **Next**.

Media Encryption
X

Audio Encryption

Preferred Format #1

SRTP_AES_CM_128_HMAC_SHA1_80

Preferred Format #2

RTP

Preferred Format #3

NONE

Encrypted RTCP

☐

MKI

☐

Lifetime

2^

Leave blank to match any value.

Interworking

☒

Video Encryption

Preferred Format #1

SRTP_AES_CM_128_HMAC_SHA1_80

Preferred Format #2

RTP

Preferred Format #3

NONE

Encrypted RTCP

☐

MKI

☐

Lifetime

2^

Leave blank to match any value.

Interworking

☒

Miscellaneous

Capability Negotiation

☒

Finish

- 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.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms (1), Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. A left-hand navigation menu lists various system management options, with 'Domain Policies' and 'Media Rules' highlighted. The main content area is titled 'Media Rules: default-low-med' 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 'Encryption', 'Codec Prioritization', 'Advanced', and 'QoS'. The 'Encryption' tab is active, showing sections for 'Audio Encryption', 'Video Encryption', and 'Miscellaneous'. The 'Audio Encryption' section has 'Preferred Formats' set to 'RTP' and 'Interworking' checked. The 'Video Encryption' section also has 'Preferred Formats' set to 'RTP' and 'Interworking' checked. The 'Miscellaneous' section has 'Capability Negotiation' unchecked. An 'Edit' button is located at the bottom right of the configuration area.

Alarms 1 Incidents Status Logs Diagnostics Users Settings Help Log Out

Session Border Controller for Enterprise

AVAYA

Dashboard
Administration
Backup/Restore
System Management
‣ Global Parameters
‣ Global Profiles
‣ PPM Services
‣ **Domain Policies**
  Application Rules
  Border Rules
  Media Rules
  Security Rules
  Signaling Rules
  End Point Policy Groups
  Session Policies
‣ TLS Management
‣ Device Specific Settings

Media Rules: default-low-med

Add Filter By Device... Clone

Media Rules
default-low-med
default-low-me...
default-high
default-high-enc
avaya-low-med...
Rem_Workers_...
IPO_SRTP
ServiceProvide...
SM_SRTP

It is not recommended to edit the defaults. Try cloning or adding a new rule instead.

Encryption Codec Prioritization Advanced QoS

Audio Encryption
Preferred Formats RTP
Interworking ☒

Video Encryption
Preferred Formats RTP
Interworking ☒

Miscellaneous
Capability Negotiation ☐

Edit

7.10.3. Signaling Rules

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

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. A left-hand navigation menu lists various configuration areas, with 'Domain Policies' and 'Signaling Rules' highlighted. The main content area is titled 'Signaling Rules: default' and features a 'default' rule selected from a list. 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', and 'Signaling QoS'. The 'General' tab is active, showing a table of inbound and outbound rules. The 'Content-Type Policy' section includes a checkbox for 'Enable Content-Type Checks' and a table for 'Action' and 'Exception List'.

Session Border Controller for Enterprise AVAYA

Alarms 1 Incidents Status Logs Diagnostics Users Settings Help Log Out

Dashboard
Administration
Backup/Restore
System Management
‣ Global Parameters
‣ Global Profiles
‣ PPM Services
‣ **Domain Policies**
‣ Application Rules
‣ Border Rules
‣ Media Rules
‣ Security Rules
‣ **Signaling Rules**
‣ End Point Policy Groups
‣ Session Policies
‣ TLS Management
‣ Device Specific Settings

Signaling Rules: default

Add Filter By Device... Clone

Signaling Rules
default
No-Content-Ty...
SessMgr_CM_...
OPTIONS
Remote Workers
Remove_Update
Contact
Remove PAI
Remove PAI_1
Remove_headers
Remove Recor...
Test

It is not recommended to edit the defaults. Try cloning or adding a new rule instead.

General Requests Responses Request Headers Response Headers Signaling QoS

UCID

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 ☒

Action	Allow	Multipart Action	Allow
Exception List		Exception List	

Edit

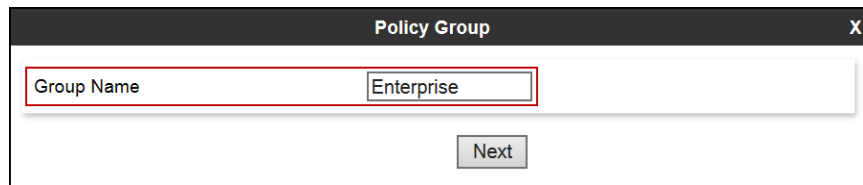
7.11. End Point Policy Groups

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.

7.11.1. 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).

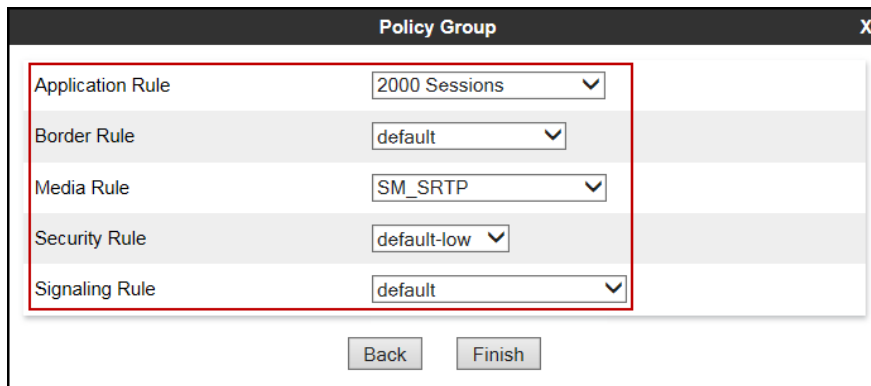
- Enter an appropriate name in the **Group Name** field.
- Click **Next**.



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" which contains the text "Enterprise". This field is highlighted with a red rectangular border. Below the input field, there is a "Next" button.

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

- **Application Rule:** *2000 Sessions* (Section 7.10.1).
- **Border Rule:** *default*.
- **Media Rule:** *SM_SRTP* (Section 7.10.2).
- **Security Rule:** *default-low*.
- **Signaling Rule:** *default* (Section 7.10.3).
- Click **Finish**.



The screenshot shows the "Policy Group" dialog box with several dropdown menus. The "Application Rule" is set to "2000 Sessions", "Border Rule" is set to "default", "Media Rule" is set to "SM_SRTP", "Security Rule" is set to "default-low", and "Signaling Rule" is set to "default". These dropdowns are highlighted with a red rectangular border. At the bottom of the dialog, there are "Back" and "Finish" buttons.

7.11.2. End Point Policy Group – Service Provider

A second End Point Policy Group was created for the service provider, repeating the steps previously described. In the Policy Group tab, all fields used one of the default sets already pre-defined in the configuration, except for the Application Rule, which was set to **2000 Sessions** (Section 7.10.1).

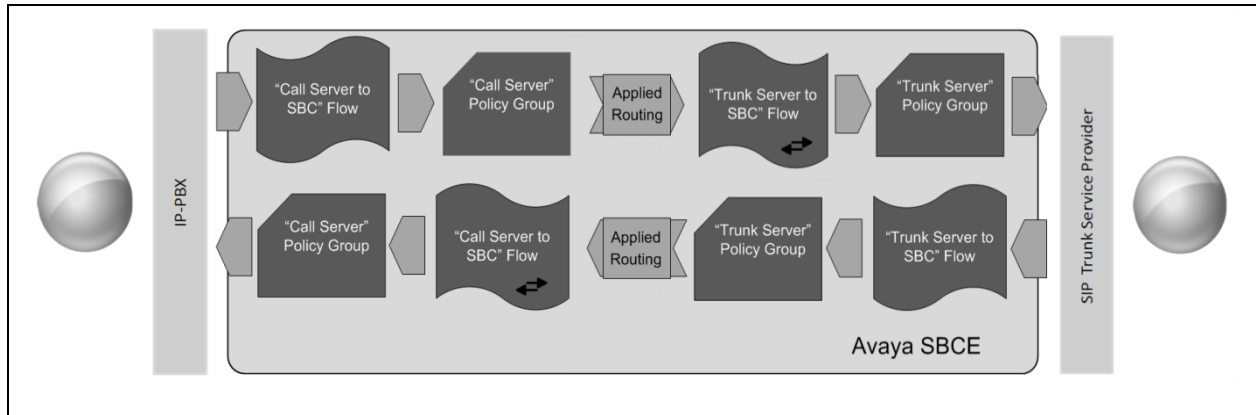
The screen below shows the End Point Policy Group named **Service Provider** after the configuration was completed.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. A left-hand navigation menu lists various configuration areas, with 'Domain Policies' and 'End Point Policy Groups' highlighted. The main content area is titled 'Policy Groups: Service Provider' and features a list of policy groups on the left, including 'default-low', 'default-low-enc', 'default-med', 'default-med-enc', 'default-high', 'default-high-enc', 'OCS-default...', 'avaya-def-low...', 'avaya-def-hig...', 'avaya-def-hig...', 'Enterprise', and 'Service Prov...'. The 'Service Prov...' group is selected. The right-hand pane shows the configuration details for this group, including a table with columns for Order, Application, Border, Media, Security, and Signaling. The table contains one row with the following values: Order 1, Application 2000 Sessions, Border default, Media default-low-med, Security default-low, and Signaling default. The '2000 Sessions' value is highlighted with a red box.

Order	Application	Border	Media	Security	Signaling
1	2000 Sessions	default	default-low-med	default-low	default

7.12. 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.

7.12.1. 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 7.8.2**, which is the reverse route of the flow. Click **Finish**.

Edit Flow: Session_Manager_Flow	
Flow Name	Session_Manager_Flow
Server Configuration	Session Manager
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Public_siq
Signaling Interface	Private_siq
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

Finish

7.12.2. 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 7.8.1**, which is the reverse route of the flow. Also note that there is no selection under the **Signaling Manipulation Script** field. Click **Finish**.

Edit Flow: SIP_Trunk_Flow_UDP	
Flow Name	SIP_Trunk_Flow_UDP
Server Configuration	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

Finish

8. Frontier Communications SIP Trunking Service Configuration

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

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

Frontier will provide the following information:

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

9. 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.

9.1. 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.

9.2. 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.

9.3. Session Manager Verification

Log in to System Manager. Under the **Elements** section, navigate to **Session Manager** → **System Status** → **SIP Entity Monitoring**. Click the Session Manager instance (*Session Manager* in the example below).

The screenshot shows the Avaya Aura System Manager 7.1 interface. The top navigation bar includes 'Home', 'Session Manager', and a notification for '1 New import page(s)'. The breadcrumb trail is 'Home / Elements / Session Manager / System Status / SIP Entity Monitoring'. The left sidebar contains a tree view with categories like Session Manager, Network, Device and Location, Application, System Status, and Managed. The 'System Status' category is expanded, and 'SIP Entity Monitoring' is selected. The main content area is titled 'SIP Entity Link Monitoring Status Summary' and includes a 'Run Monitor' button. Below this is a table showing monitoring data for 'Session Manager'.

		Monitored Entities						
Session Manager	Type	Down	Partially Up	Up	Not Monitored	Deny	Total	
<input type="checkbox"/> Session Manager	Core	1	0	5	0	0	6	

1 Items Refresh Filter: Enable

Select: All, None

[All Monitored SIP Entities](#)

Verify that the state of the Session Manager links to Communication Manager and the Avaya SBCE under the **Conn. Status** and **Link Status** columns is **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.

All Entity Links for Session Manager: Session Manager

Summary View

Status Details for the selected Session Manager:

SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
CS1K7.6	IPv4	172.16.5.60	5085	UDP	FALSE	DOWN	408 Request Timeout	DOWN
Avaya SBCE	IPv4	10.64.101.243	5061	TLS	FALSE	UP	200 OK	UP
Communication Manager Trunk 1	IPv4	10.64.101.241	5061	TLS	FALSE	UP	200 OK	UP
AA-Messaging	IPv4	10.64.101.250	5060	TCP	FALSE	UP	200 OK	UP
Communication Manager Trunk 2	IPv4	10.64.101.241	5071	TLS	FALSE	UP	200 OK	UP
Communication Manager Trunk 98	IPv4	10.64.101.241	5065	TLS	FALSE	UP	200 OK	UP

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.

9.4. 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.

Session Border Controller for Enterprise

Dashboard

This system contains one or more Avaya demo certificates. These certificates have been compromised and should not be used for any production traffic.

The following certificates are expired:

- Rapid_SSL_Cert.crt (Certificate)

Information	
System Time	12:27:04 PM EST Refresh
Version	7.2.0.0-18-13712
Build Date	Thu Jun 1 00:12:50 UTC 2017
License State	OK
Aggregate Licensing Overages	0
Peak Licensing Overage Count	0
Last Logged in at	11/22/2017 10:47:24 EST
Failed Login Attempts	0

Installed Devices	
EMS	
Avaya_SBCE	1

Active Alarms (past 24 hours)

Avaya_SBCE : Disk utilization is more than 75% for /archive/log/ipcs

Incidents (past 24 hours)

Avaya_SBCE : No Subscriber Flow Matched

Avaya_SBCE : No Subscriber Flow Matched

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

Alarm Viewer

Devices

EMS

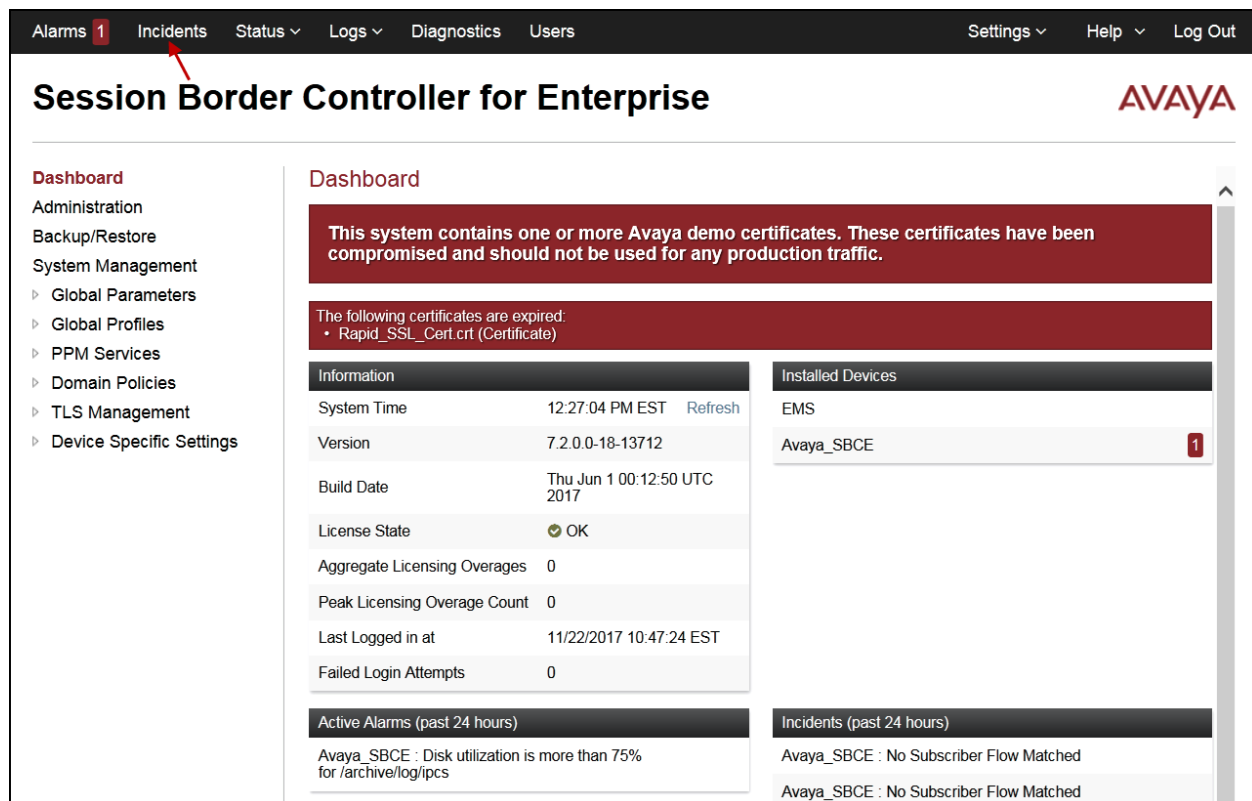
Avaya_SBCE

Alarms

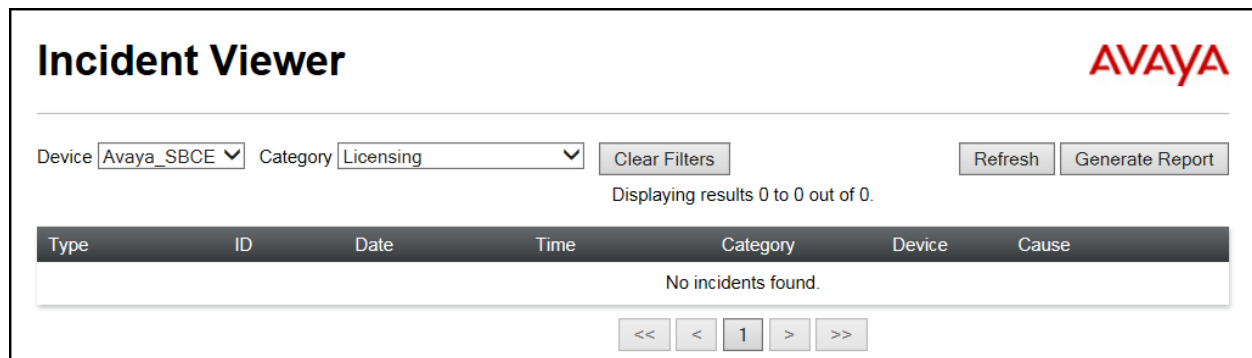
<input checked="" type="checkbox"/>	ID	Details	State	Time	Device
No alarms found for this device.					

[Clear Selected](#) [Clear All](#)

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



The following screen shows the Incident Viewer page.



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

Session Border Controller for Enterprise

Dashboard

This system contains one or more Avaya demo certificates. These certificates have been compromised and should not be used for any production traffic.

The following certificates are expired:

- Rapid_SSL_Cert.crt (Certificate)

Information	
System Time	12:27:04 PM EST Refresh
Version	7.2.0.0-18-13712
Build Date	Thu Jun 1 00:12:50 UTC 2017
License State	OK
Aggregate Licensing Overages	0
Peak Licensing Overage Count	0
Last Logged in at	11/22/2017 10:47:24 EST
Failed Login Attempts	0

Installed Devices	
EMS	
Avaya_SBCE	1

Active Alarms (past 24 hours)

Avaya_SBCE : Disk utilization is more than 75% for /archive/log/ipcs

Incidents (past 24 hours)

Avaya_SBCE : No Subscriber Flow Matched

Avaya_SBCE : No Subscriber Flow Matched

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

Diagnostics

Pinging 10.64.101.249

Average ping from 10.64.101.244 [A1] to 10.64.101.249 is 0.369ms.

Full Diagnostic **Ping Test**

Outgoing pings from this device can only be sent via the primary IP (determined by the OS) of each respective interface or VLAN.

Source Device / IP: A1

Destination IP: 10.64.101.249

Ping

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 **Device Specific Settings** → **Troubleshooting** → **Trace**. Select the **Packet Capture** tab, set the desired configuration for the trace and click **Start Capture**.

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" with the Avaya logo. A left-hand navigation menu lists various system management options, with "Device Specific Settings" and "Trace" highlighted. The main content area is titled "Trace: Avaya_SBCE" and features two tabs: "Devices" and "Avaya_SBCE". The "Packet Capture" tab is active, showing a configuration window with the following settings: Status (Ready), Interface (Any), Local Address (All), Remote Address (*), Protocol (All), Maximum Number of Packets to Capture (10000), and Capture Filename (Test.pcap). "Start Capture" and "Clear" buttons are at the bottom of the configuration window.

Packet Capture Configuration	
Status	Ready
Interface	Any
Local Address IP[:Port]	All
Remote Address *, *:Port, IP, IP:Port	*
Protocol	All
Maximum Number of Packets to Capture	10000
Capture Filename <small>Using the name of an existing capture will overwrite it.</small>	Test.pcap
<div>Start Capture Clear</div>	

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 displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. A left-hand navigation menu lists various system management options, with 'Device Specific Settings' 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, showing a table of captured files. The table has columns for File Name, File Size (bytes), Last Modified, and a Delete button. One file is listed: 'Test_20171122123225.pcap' with a size of 176,128 bytes and a timestamp of November 22, 2017, 12:32:47 PM EST.

File Name	File Size (bytes)	Last Modified	
Test_20171122123225.pcap	176,128	November 22, 2017 12:32:47 PM EST	Delete

10. Conclusion

These Application Notes describe the procedures required to configure Avaya Aura® Communication Manager 7.1, Avaya Aura® Session Manager 7.1 and Avaya Session Border Controller for Enterprise 7.2, 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**.

11. 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*, Release 7.1.1, Issue 2, August 2017.
- [2] *Administering Avaya Aura® Communication Manager*, Release 7.1.1, Issue 2, August 2017.
- [3] *Administering Avaya Aura® System Manager* for Release 7.1.1, Issue 7, October 2017.
- [4] *Deploying Avaya Aura® System Manager*, Release 7.1.1, Issue 3, August 2017.
- [5] *Deploying Avaya Aura® Session Manager*, Release 7.1, Issue 1, May 2017.
- [6] *Administering Avaya Aura® Session Manager*, Release 7.1.1, Issue 2, August 2017.
- [7] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.2.1, Issue 4, November 2017.
- [8] *Administering Avaya Session Border Controller for Enterprise*, Release 7.2.1, Issue 4, November 2017.
- [9] *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*.
- [10] *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 7.8, Issue 3, August 2017.
- [11] *Implementing and Administering Avaya Aura® Media Server*. Release 7.8, Issue 5, October 2017.
- [12] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [13] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>

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