Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 7.2 with Verizon Business IP Trunking Service – Issue 1.0

Abstract

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 8.0, Avaya Aura® Communication Manager Release 8.0, Avaya Aura® Experience Portal 7.2, and Avaya Session Border Controller for Enterprise Release 7.2 with the Verizon Business IP Trunking service. These Application Notes update previously published Application Notes with newer versions of Communication Manager, Session Manager, and Avaya Session Border Controller for Enterprise.

The Verizon Business IP Trunking service offer referenced within these Application Notes is designed for business customers with an Avaya SIP trunk solution. The service provides local and/or long distance PSTN calling via standards-based SIP trunks directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

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

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab, utilizing a Verizon Business Private IP (PIP) circuit connection to the production Verizon Business IP Trunking service.

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1. Introduction

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 8.0, Avaya Aura® Communication Manager Release 8.0, Avaya Aura® Experience Portal 7.2, and Avaya Session Border Controller for Enterprise Release 7.2 with the Verizon Business IP Trunking service. The Verizon Business IP Trunking service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks.

2. General Test Approach and Test Results

The test approach was manual testing of inbound and outbound calls using the Verizon Business IP Trunking service on a production Verizon PIP access circuit, as shown in **Figure 1**.

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

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

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

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

Encryption (TLS/SRTP) was used internal to the enterprise between Avaya products wherever possible.

2.1. Interoperability Compliance Testing

Compliance testing scenarios for the configuration described in these Application Notes included the following:

- Inbound and outbound voice calls between telephones controlled by Communication Manager and the PSTN can be made using G.711MU or G.729A codecs.
- Direct IP-to-IP Media (also known as "Shuffling") when applicable.
- DTMF using RFC 2833
  - Outbound call to PSTN application requiring post-answer DTMF (e.g., an IVR or voice mail system)

    - Inbound call from PSTN to Avaya CPE application requiring post-answer DTMF (e.g., Aura® Messaging, Experience Portal, Avaya vector digit collection steps)
- Additional PSTN numbering plans (e.g., International, operator assist, 411)
- Hold / Retrieve with music on hold
- Call transfer using two approaches
    - REFER approach (Communication Manager Network Call Redirection flag on trunk group form set to "y")
    - INVITE approach (Communication Manager Network Call Redirection flag on trunk group form set to "n")
- Conference calls
- SIP Diversion Header for call redirection
    - Call Forwarding
    - EC500
- Inbound caller interaction with Experience Portal applications, including prompting, caller DTMF input, wait treatment (e.g., announcements and/or music on hold), Automatic Speech Recognition, and Text to Speech
- Experience Portal use of SIP REFER to redirect inbound calls, via the Avaya SBCE, to the appropriate Communication Manager agent extension
- Call and two-way talk path establishment between callers and Communication Manager agents following redirection from Experience Portal
- Inbound calls to a self-service Experience Portal application which forwards the call to 8YY or any other PSTN number over Verizon IPT service using SIP REFER
- Long hold time calls
- Remote Worker

2.2. Test Results

Interoperability testing of Verizon Business IP Trunking service was completed with successful results for all test cases. The following limitations are noted for the sample configuration described in these Application Notes.

1. Verizon provisioned T.38 fax on the production circuit used to verify these Application Notes. Verizon Business IP Trunking service will never send a re-Invite to T.38. If the **FAX Mode** field on the Communication Manager ip-codec-set form page 2 is set to "**t.38-standard**" (see **Section 6.6**), Communication Manager will send the proper re-Invite to T.38 for both inbound and outbound fax calls, but will not failback to G.711 should the Verizon network reject the Communication Manager attempt to transition to T.38 by sending a 488 Not Acceptable message. If the **FAX Mode** is set to "**t.38-G711-fallback**" setting[1], Communication Manager will send a re-Invite to T.38 for inbound fax calls only and relies on the far end to send a re-Invite to T.38 for outbound calls. Communication Manager assumes T.38 fax is not supported for an outbound fax call unless an Invite for T.38 is received. The result is an outbound fax sent using G.711, even though the circuit is provisioned for T.38. Inbound fax calls negotiate properly to T.38. With the limitations of T.38 on Verizon's network, it is recommended to use an AudioCodes MP-114 or MP-124

[1] The "T.38 Fax with Fallback to G.711 Pass-Through" feature requires G450 or G450 Media Gateways with release 33.13 or higher.

Gateway between Session Manager and the fax device when fax is used with Verizon Business IP Trunking service.

2. When the **Initial IP-IP Direct Media** field on the Communication Manager signaling group form page 1 is set to "**y**", Communication Manager sends a "183 Session Progress" without SDP during an inbound PSTN call that is forwarded to another PSTN call just before a 183 is sent with SDP information to the far end. This is undesirable to Verizon and could result in no audio. The recommendation in **Section 6.8.1.1** is to leave the **Initial IP-IP Direct Media** field to "**n**".

3. When TLS/SRTP is used within the enterprise, the SIP headers include the SIPS URI scheme for Secure SIP. The Avaya SBCE converts these header schemes from SIPS to SIP when it sends the SIP message toward Verizon. However, for call forward and EC500 calls, the Avaya SBCE was not changing the Diversion header scheme as expected. This caused these call types that require a Diversion header to fail since Verizon does not support Secure SIP. This anomaly is currently under investigation by the Avaya SBCE development team. A workaround is to include a SigMa script for the Verizon Server Configuration profile on the Avaya SBCE to convert "sips" to "sip" in the Diversion header. See **Section 8.3.3**.

4. Verizon Business IP Trunking service does not support an E.164 formatted number for the Calling Line Identification for outbound calls. An adaptation in Session Manager is used to convert the E.164 numbers Communication Manager used in the sample configuration for Calling Line Identification (e.g., From and P-Asserted Identity headers) into 10-digit numbers. See **Section 5.3.2**.

5. The Experience Portal test application used for compliance testing performs consultative call transfers using SIP INVITE with the original calling party number in the From and P-Asserted Identity headers, it does not include a Diversion header. Verizon requires a Diversion header for this scenario. This caused consultative call transfers out the Verizon Business IP Trunking service to fail. However, blind transfers out to Verizon using SIP REFER were successful. Also, consultative and blind transfers from Experience Portal to Communication Manager were successful as well.

6. Emergency 911/E911 Services Limitations and Restrictions - Although Verizon provides 911/E911 calling capabilities, 911 capabilities were not tested; therefore, it is the customer's responsibility to ensure proper operation with its equipment/software vendor.

7. Verizon Business IP Trunking service does not support G.711A codec for domestic service (EMEA only).

8. Verizon Business IP Trunking service does not support G.729B codec.

Note – These Application Notes describe the provisioning used for the sample configuration shown in **Figure 1**. Other configurations may require modifications to the provisioning described in this document.

2.3. History Info and Diversion Headers

The Verizon Business IP Trunking service does not support SIP History Info headers. Instead, the Verizon Business IP Trunking service requires that the SIP Diversion header be sent for redirected calls. The Communication Manager SIP trunk group form provides the options for specifying whether History Info headers or Diversion headers are sent.

If Communication Manager sends the History Info header, Session Manager can convert the History Info header into the Diversion header. This is performed by specifying the "*VerizonAdapter*" adaptation in Session Manager. See **Section 5.3.2**.

The Communication Manager Call Forwarding or Extension to Cellular (EC500) features may be used for the call scenarios testing the Diversion header.

2.4. SIP Header Removal

To support advanced SIP telephony features in the Avaya Aura® enterprise environment, certain proprietary headers may be included in the SIP message sent toward Verizon. These extra headers can cause the SIP message to become larger than the specified Maximum Transmission Unit (MTU) and create fragmented UDP packets. These fragmented packets may not be re-assembled properly on the far-end by Verizon's equipment, for instance, when packets arrive out of order. To prevent fragmented packets, any unnecessary or proprietary headers should be removed from the SIP message before being sent to Verizon. Session Manager can remove these headers by specifying the "*eRHdrs*" parameter within the "*VerizonAdapter*" adaptation. See **Section 5.3.2**.

In the sample configuration, the following headers were removed:

- AV-Global-Session-ID
- Alert-Info
- Endpoint-View
- P-AV-Message-Id
- P-Charging-vector
- P-Location
- AV-Secure-Indication

To help reduce the packet size further, the Avaya SBCE can remove the "*gsid*" and "*epv*" parameters that may be included within the Contact header by applying a Sigma script to the Verizon server configuration. See **Section 8.3.3** and **8.3.5**.

2.5. Support

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

For technical support on Verizon Business IP Trunking service offer, visit online support at http://www.verizonbusiness.com/us/customer/

3. Reference Configuration

3.1. Illustrative Configuration Information

Figure 1 illustrates the sample configuration used for the compliance testing. The Avaya CPE location simulates a customer site. The PIP service defines a secure MPLS connection between the Avaya CPE T1 connection and the Verizon service node.

The Avaya SBCE receives traffic from the Verizon Business IP Trunking service on port 5060 and sends traffic to the Verizon Business IP Trunking service on port 5071, using UDP protocol for network transport (required by the Verizon Business IP Trunking service). The Verizon Business IP Trunking service provided Direct Inward Dial (DID) 10-digit numbers. These DID numbers can be mapped by Session Manager or Communication Manager to Avaya telephone extensions.

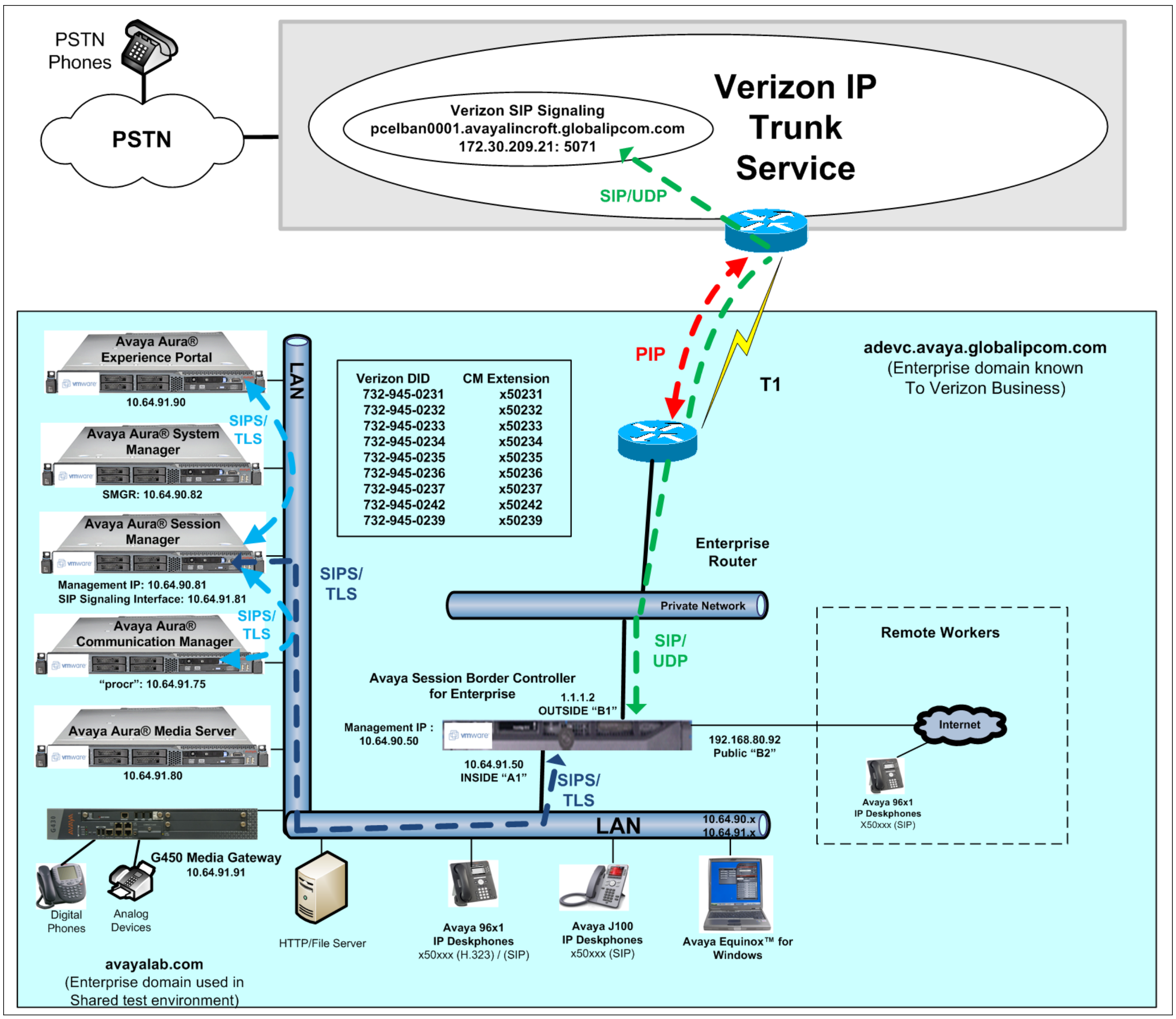


Figure 1: Avaya Interoperability Test Lab Configuration

The Verizon Business IP Trunking service used FQDN *pcelban0001.avayalincroft.globalipcom.com*. The Avaya CPE environment was known to Verizon Business IP Trunking service as FQDN *adevc.avaya.globalipcom.com.* Access to the Verizon Business IP Trunking service was added to a configuration that already used domain "avayalab.com" at the enterprise. As such, the Avaya SBCE is used to adapt the "avayalab.com" domain to the domain known to Verizon (see **Section 8.3.9**). These Application Notes indicate a configuration that would not be required in cases where the CPE domain in Communication Manager and Session Manager match the CPE domain known to the Verizon Business IP Trunking service.

Note – The Fully Qualified Domain Names and IP addressing specified in these Application Notes apply only to the reference configuration shown in **Figure 1**. Verizon Business customers will use their own FQDNs and IP addressing as required.

In summary, the following components were used in the reference configuration.

- Verizon Business IP Trunking network Fully Qualified Domain Name (FQDN)
  - *pcelban0001.avayalincroft.globalipcom.com*
- Avaya CPE Fully Qualified Domain Name (FQDN) known to Verizon
  - *adevc.avaya.globalipcom.com*
- Avaya Session Border Controllers for Enterprise
- Avaya Aura® Session Manager
- Avaya Aura® Communication Manager
- Avaya G450 Media Gateway
- Avaya Media Server
- Avaya Aura® Messaging
- Avaya Aura® Experience Portal
- Avaya 96X1 Series IP Deskphones using the SIP and H.323 software bundle
- J100 Series IP Deskphones using the SIP software bundle
- Avaya Equinox™ for Windows
- Avaya Digital Phones
- Ventafax fax software

3.2. Call Flows

To understand how Verizon Business IP Trunking service calls are handled by the Avaya CPE environment, several call flows are described in this section.

3.2.1 Communication Manager

The first call scenario illustrated is an inbound Verizon Business IP Trunking service call that arrives at the Avaya SBCE, to Session Manager, and is subsequently routed to Communication Manager, which in turn routes the call to a phone or fax endpoint.

1. A PSTN phone originates a call to a Verizon Business IP Trunking service number.
2. The PSTN routes the call to the Verizon Business IP Trunking service network.
3. The Verizon Business IP Trunking service routes the call to the Avaya SBCE.
4. The Avaya SBCE performs IP address translations and any necessary SIP header modifications and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines to where the call should be routed next. In this case, Session Manager routes the call to Communication Manager.
6. Depending on the called number, Communication Manager routes the call to a phone or fax endpoint.

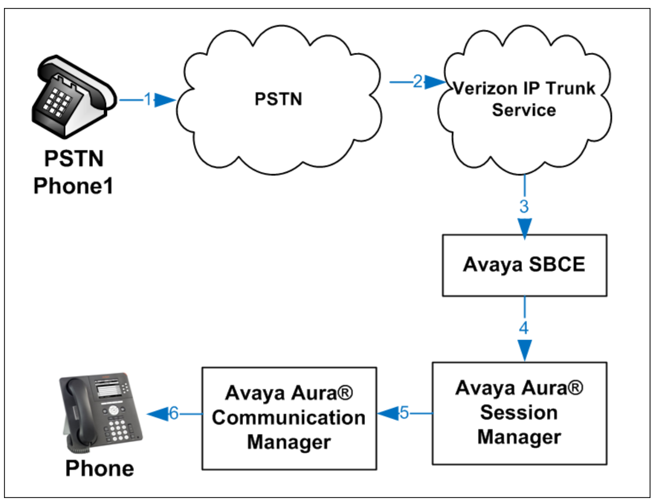


Figure 2: Inbound Verizon Call

The second call scenario illustrated is an outbound call initiated on Communication Manager, routed to Session Manager, and is subsequently sent to the Avaya SBCE for delivery to the Verizon Business IP Trunking service.

1. A Communication Manager phone or fax endpoint originates a call to a Verizon Business IP Trunking service number for delivery to the PSTN.
2. Communication Manager routes the call to Session Manager.
3. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines to where the call should be routed next. In this case, Session Manager routes the call to the Avaya SBCE.
4. The Avaya SBCE performs IP address translations and any necessary SIP header modifications and routes the call to the Verizon Business IP Trunking service.
5. The Verizon Business IP Trunking service delivers the call to the PSTN.

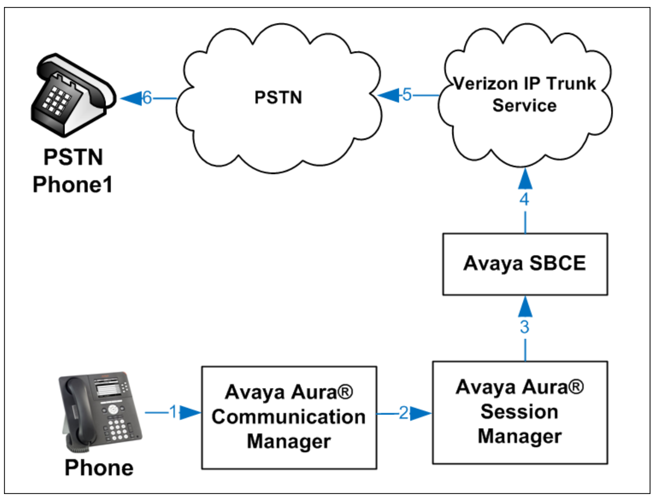


Figure 3: Outbound Verizon Call

The third call scenario illustrated is an inbound Verizon Business IP Trunking service call that arrives at the Avaya SBCE, to Session Manager, and subsequently Communication Manager. Communication Manager routes the call to a destination station; however, the station has set Call Forward to an alternate destination. Without answering the call, Communication Manager redirects the call back to the Verizon Business IP Trunking service for routing to the alternate destination.

Note – In cases where calls are forwarded to an alternate destination such as an 8xx numbers, the Verizon Business IP Trunking service requires the use of SIP Diversion Header for the redirected call to complete (see **Section 6.8**).

1. A PSTN phone originates a call to an IPFR-EF number.
2. The PSTN routes the call to the IPFR-EF network.
3. IPFR-EF routes the call to the Avaya SBCE.
4. The Avaya SBCE performs SIP Network Address Translation (NAT) and any necessary SIP header modifications and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to Communication Manager. Communication Manager routes the call to a station.
6. Because the Communication Manager phone has set Call Forward to another Verizon Business IP Trunking service number, Communication Manager initiates a new call back out to Session Manager, the Avaya SBCE, and to the Verizon Business IP Trunking service network.
7. The Verizon Business IP Trunking service places a call to the alternate destination, and upon answering Communication Manager connects the calling party to the target party.

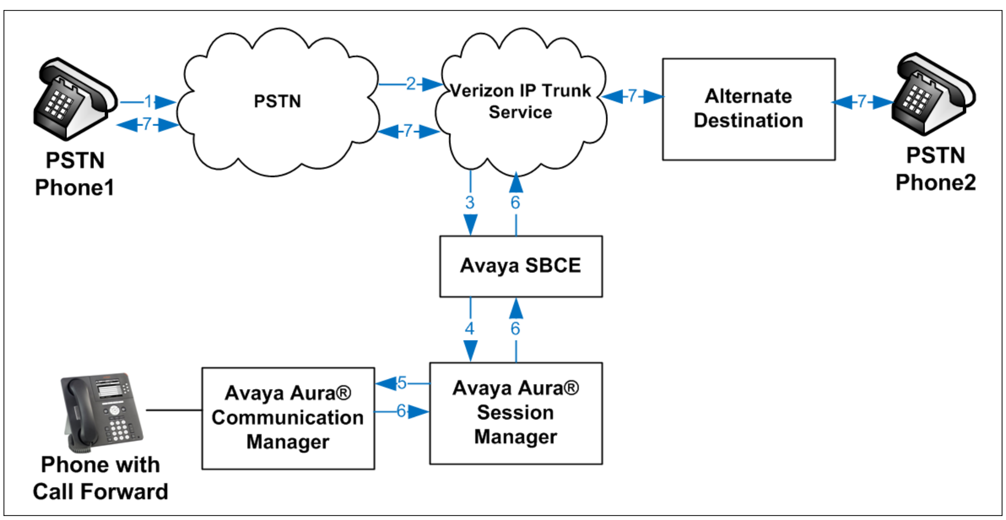


Figure 4: Station Re-directed (e.g., Call Forward) Verizon Call

3.2.2 Experience Portal

The first call scenario illustrated below is an inbound call arriving and remaining on Experience Portal.

1. A PSTN phone originates a call to a Verizon Business IP Trunking service number.
2. The PSTN routes the call to the Verizon Business IP Trunking service network.
3. The Verizon Business IP Trunking service routes the call to the Avaya SBCE.
4. The Avaya SBCE performs any necessary SIP header modifications and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to Experience Portal.
6. Experience Portal matches the called party number to a VXML and/or CCXML application script, answers the call, and handles the call according to the directives specified in the application. In this scenario, the application sufficiently meets the caller's needs or requests, and thus the call does not need to be transferred to Communication Manager.

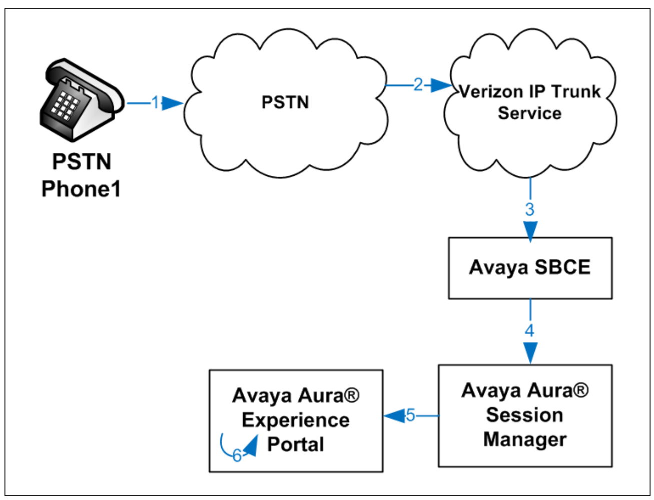


Figure 5: Inbound Call Handling Entirely by Avaya Aura® Experience Portal

The second call scenario illustrated below is an inbound call arriving on Experience Portal and transferred to Communication Manager without determining whether an agent is available or not.

1. Same as the first five steps from the first call scenario.
2. In this scenario, when the caller selects an option requesting an agent, Experience Portal redirects the call by sending a SIP REFER to the Avaya SBCE.
3. The Avaya SBCE sends a SIP INVITE to the Communication Manager (via Session Manager) for the selected skill. In addition, the Avaya SBCE places the inbound call on hold.
4. Communication Manager routes the call to the agent.
5. When the agent answers, the Avaya SBCE takes the call off hold and the caller is connected to the agent.

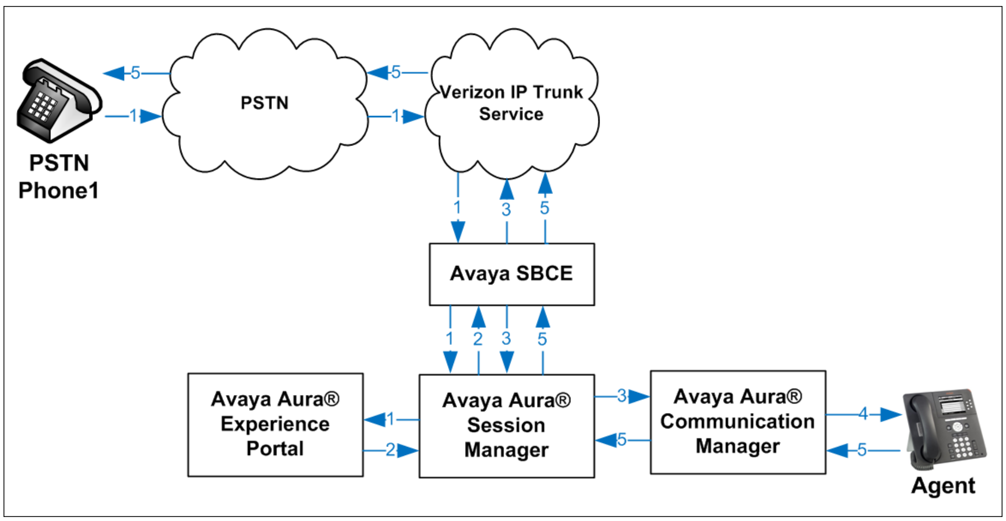


Figure 6: Avaya Aura® Experience Portal Transfers Call to Avaya Aura® Communication Manager

The third call scenario illustrated below is an inbound call arriving on Experience Portal and forwarded to an 8YY number or any other PSTN number over the Verizon network.

1. Same as the first six steps from the first call scenario.
2. In this scenario, the application is sufficient to meet the caller's requests, and thus the call needs to be forwarded to another PSTN number. Based upon the selection, Experience Portal forwards the call to an appropriate PSTN number which can be a regular PSTN number or an 8YY number.

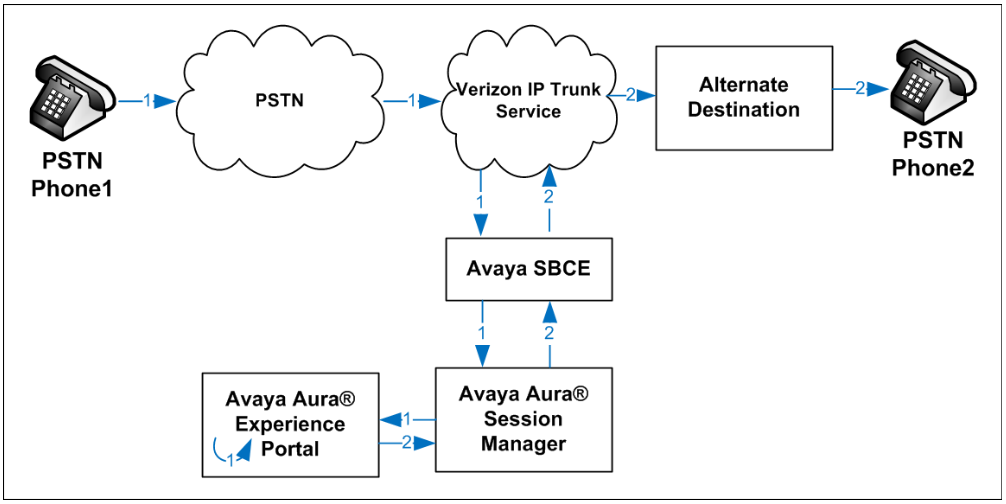


Figure 7: Inbound Call forwarded by Experience Portal to another PSTN number

4. Equipment and Software Validated

The following equipment and software were used in the sample configuration.

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	8.0.0.0-R018x.00.0.822.0
Avaya Aura® System Manager	8.0.0.0.098174
Avaya Aura® Session Manager	8.0.0.0.800035
Avaya Session Border Controller for Enterprise	7.2.2.0-11-15522
Avaya Aura® Messaging	7.0 SP 0
Avaya Aura® Experience Portal	7.2.0.0.1117
Avaya Aura® Media Server	8.0.0.117
G450 Gateway	40.10.0
Avaya 96X1– Series Telephones (SIP)	R7.1.3.0.11
Avaya 96X1– Series Telephones (H.323)	R6.66.04
Avaya J100 – Series Telephones (SIP)	3.0.0.2.2
Avaya Equinox™ for Windows	3.4.0.152.46
Avaya 2400 – Series Digital Telephones	N/A
Ventafax	7.9

Table 1: Equipment and Software Used in the Sample Configuration

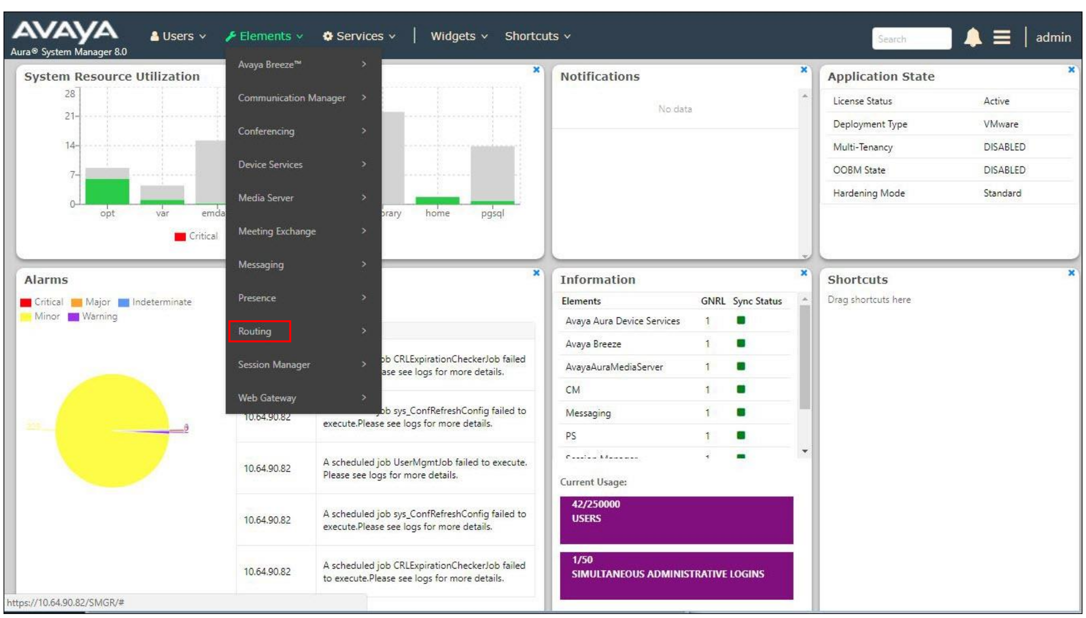
5. Configure Avaya Aura® Session Manager

Note – These Application Notes assume that basic System Manager and Session Manager administration has already been performed. Consult **[1]- [4]** for further details.

This section provides the procedures for configuring Session Manager to process inbound and outbound calls between Communication Manager and the Avaya SBCE. In the reference configuration, all Session Manager provisioning is performed via System Manager.

- Define a SIP Domain.
- Define a Location for Customer Premises Equipment (CPE).
- Configure the Adaptation Modules that will be associated with the SIP Entities for Communication Manager, the Avaya SBCE, and Messaging.
- Define SIP Entities corresponding to Session Manager, Communication Manager, Experience Portal, the Avaya SBCE, and Messaging.
- Define Entity Links describing the SIP trunks between Session Manager, Communication Manager, Experience Portal, and Messaging, as well as the SIP trunks between the Session Manager and the Avaya SBCE.
- Define Routing Policies associated with the Communication Manager, Experience Portal, Messaging, and the Avaya SBCE.
- Define Dial Patterns, which govern which Routing Policy will be selected for inbound and outbound call routing.
- Verify TLS Certificates.

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. In the **Log On** screen (not shown), enter appropriate **User ID** and **Password** and press the **Log On** button. Once logged in, **Home** screen is displayed. From the **Home** screen, under the **Elements** heading, select **Routing**.



5.1. SIP Domain

Step 1 - Select **Domains** from the left navigation menu. In the reference configuration, domain **avayalab.com** was defined.

Step 2 - Click **New**. Enter the following values and use default values for remaining fields.

- **Name:** Enter the enterprise SIP Domain Name. In the sample screen below, **avayalab.com** is shown.
- **Type:** Verify **sip** is selected.
- **Notes:** Add a brief description.

Step 3 - Click **Commit** (not shown) to save.

5.2. Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside. In the reference configuration, three Locations are specified:

- **Main** – The customer site containing System Manager, Session Manager, Communication Manager and local SIP endpoints.
- **Common** – Avaya SBCE

5.2.1 Main Location

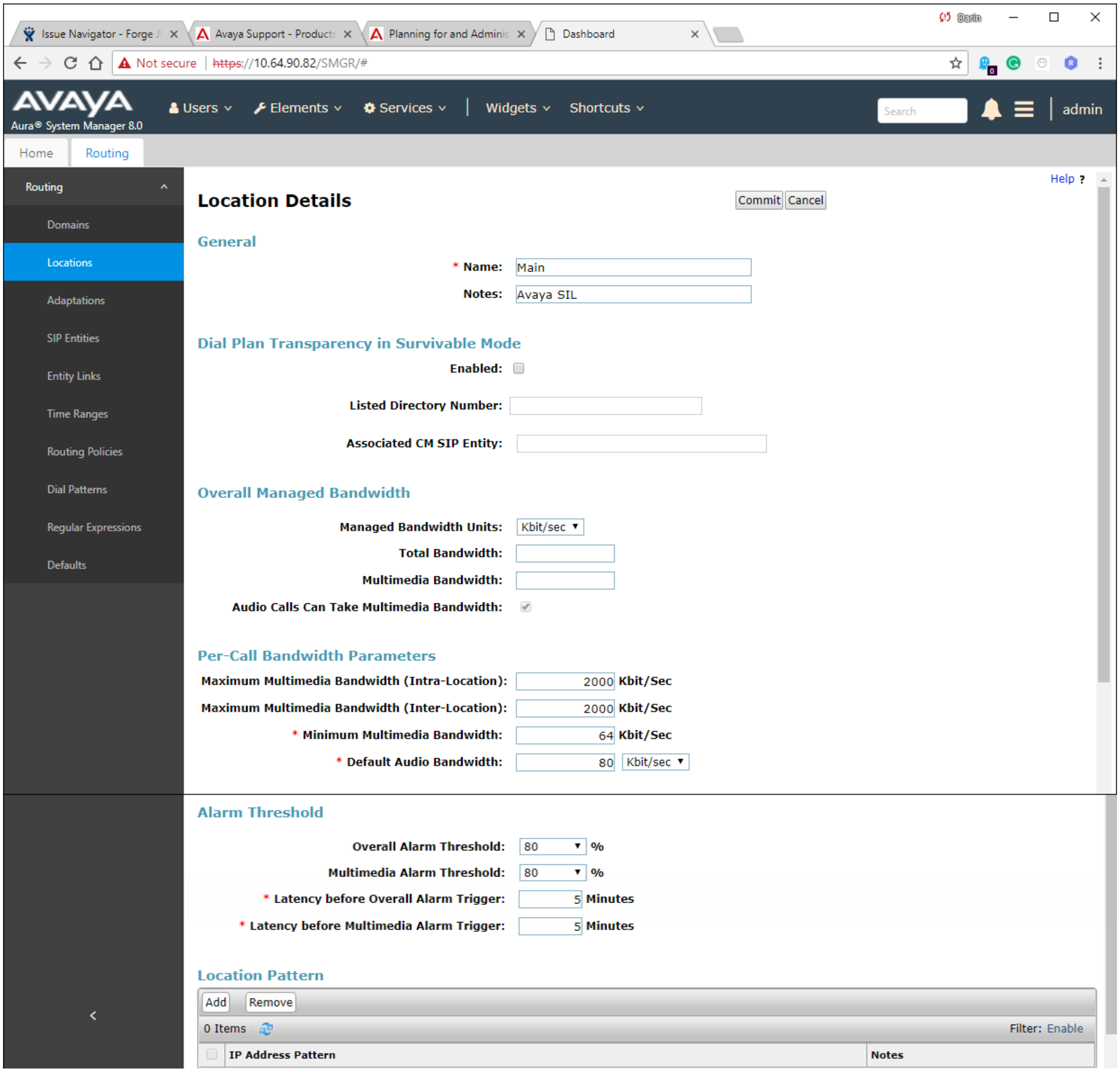
Step 1 - Select **Locations** from the left navigational menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Name:** Enter a descriptive name for the Location (e.g., **Main**).
- **Notes:** Add a brief description.

Step 2 - In the **Location Pattern** section, click **Add** and enter the following values (not shown).

- **IP Address Pattern:** Leave blank.
- **Notes:** Add a brief description.

Step 3 - Click **Commit** to save.



5.2.2 Common Location

To configure the Avaya SBCE Location, repeat the steps in **Section 5.2.1** with the following changes (not shown):

- **Name** – Enter a descriptive name (e.g., **Common**).

5.3. Configure Adaptations

Session Manager can be configured to use Adaptation Modules to convert SIP headers sent to/from Verizon. In the reference configuration the following Adaptations were used:

- Calls from Verizon (**Section 5.3.1**) - Modification of SIP messages sent to Communication Manager extensions.

- The Verizon DNIS number digit string in the Request URI is replaced with the associated Communication Manager extensions/VDN.

- Calls to Verizon (**Section 5.3.2**) - Modification of SIP messages sent by Communication Manager extensions.
    - The History-Info header is converted to a Diversion header automatically by the **VerizonAdapter**.
    - Avaya SIP headers not required by Verizon are removed (see **Section 2.4**).

5.3.1 Adaptation for Avaya Aura® Communication Manager

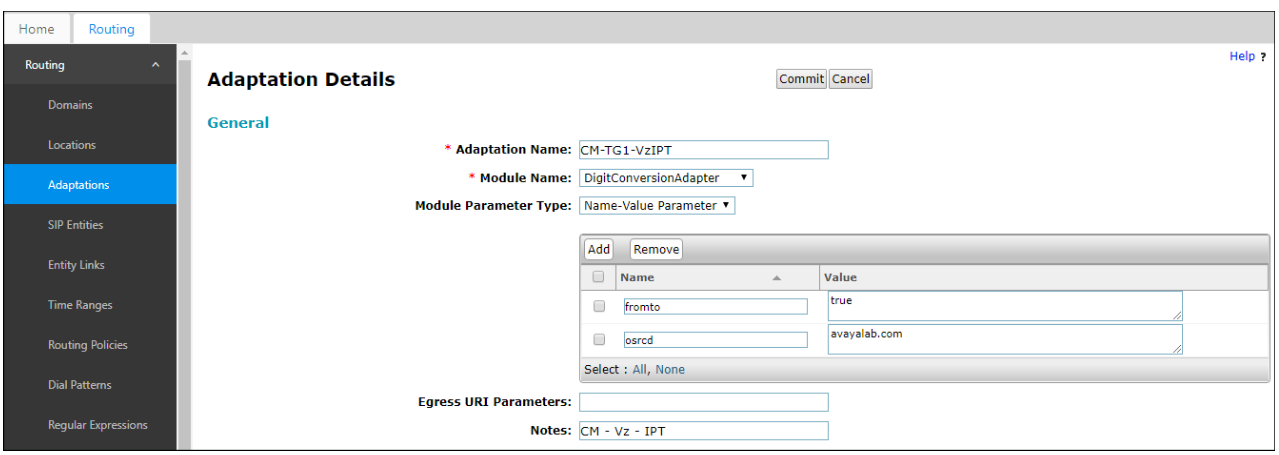
The Adaptation administered in this section is used for modification of SIP messages to Communication Manager extensions from Verizon.

Step 1 - In the **left** pane under **Routing**, click on **Adaptations**. In the **Adaptations** page, click on **New** (not shown).

Step 2 - In the **Adaptation Details** page, enter:

1. A descriptive **Name**, (e.g., **CM-TG1-VzIPT**).
2. Select **DigitConversionAdapter** from the **Module Name** drop down.
3. Select **Name-Value Parameter** from the **Module Parameter Type** drop down:
    - **Name**: "**fromto**" **Value**: "**true**"
        - This adapts the From and To headers along with the Request-Line and PAI headers.
    - **Name**: "**osrcd**" **Value**: "**avayalab.com**"
        - This enables the source domain to be overwritten with "avayalab.com". For example, for inbound PSTN calls from Verizon to Communication Manager, the PAI header will contain "avayalab.com".

Note – Depending on the Communication Manager configuration, it may not be necessary for Session Manager to adapt the domain in this fashion.



Step 3 - Scroll down to the **Digit Conversion for Outgoing Calls from SM** section (the *inbound* digits from Verizon that need to be replaced with their associated Communication Manager extensions before being sent to Communication Manager).

1. **Example 1 – destination extension**: 7329450231 is a DNIS string sent in the Request URI by the Verizon Business IP Trunking service that is associated with Communication Manager extension 12001.
    - Enter **7329450231** in the **Matching Pattern** column.
    - Enter **10** in the **Min/Max** columns.
    - Enter **10** in the **Delete Digits** column.
    - Enter **12001** in the **Insert Digits** column.
    - Specify that this should be applied to the SIP **destination** headers in the **Address to modify** column.
    - Enter any desired notes.

Step 4 - **Repeat Step 3** for all additional Verizon DNIS numbers/Communication manager extensions.

Step 5 - Click on **Commit**.

> **Note** – No **Digit Conversion for Incoming Calls to SM** were required in the reference configuration.

> **Note** – In the reference configuration, the Verizon Business IP Trunking service delivered 10-digit DNIS numbers.

Digit Conversion for Outgoing Calls from SM

Add Remove

4 Items Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* 7329450	* 10	* 10		* 5		destination		Verizon DIDs
	* 7329450228	* 10	* 10		* 10	12001	destination		
	* 7329450229	* 10	* 10		* 10	12000	destination		analog fax
	* 7329450231	* 10	* 10		* 10	12001	destination		

Select : All, None

Commit Cancel

5.3.2 Adaptation for the Verizon Business IP Trunking service

The Adaptation administered in this section is used for modification of SIP messages from Communication Manager to Verizon. Repeat the steps in **Section 5.3.1** with the following changes.

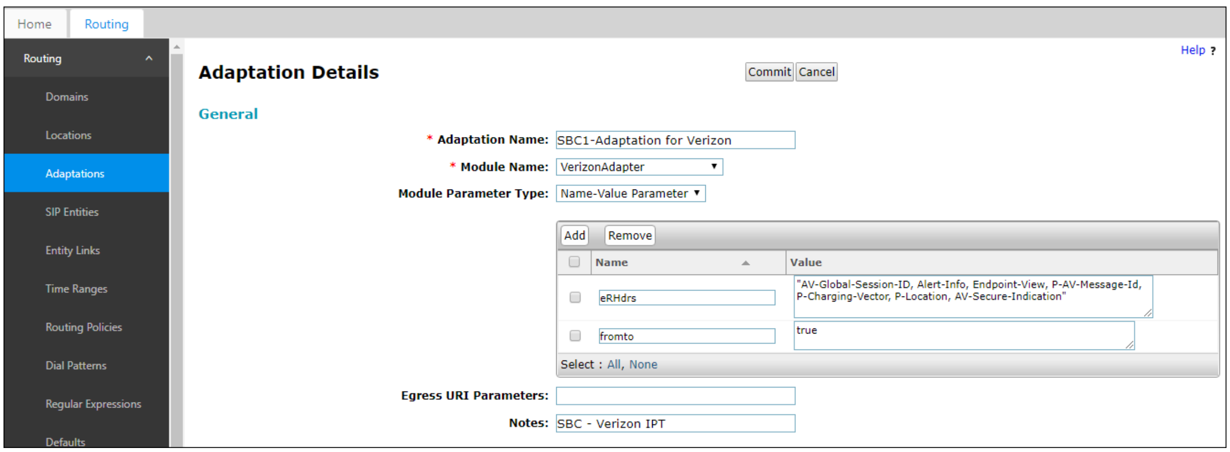
Step 1 - In the **Adaptation Details** page, enter:

1. A descriptive **Name**, (e.g., **SBC1-Adaptation for Verizon**).
2. Select **VerizonAdapter** from the **Module Name** drop down menu. The VerizonAdapter will automatically remove History-Info headers, (which the Verizon Business IP Trunking service does not support), sent by Communication Manager (see **Section 6.8.1.2**) and replace them with Diversion headers.

Step 2 - In the **Module Parameter Type**: field select **Name-Value Parameter** from the menu.

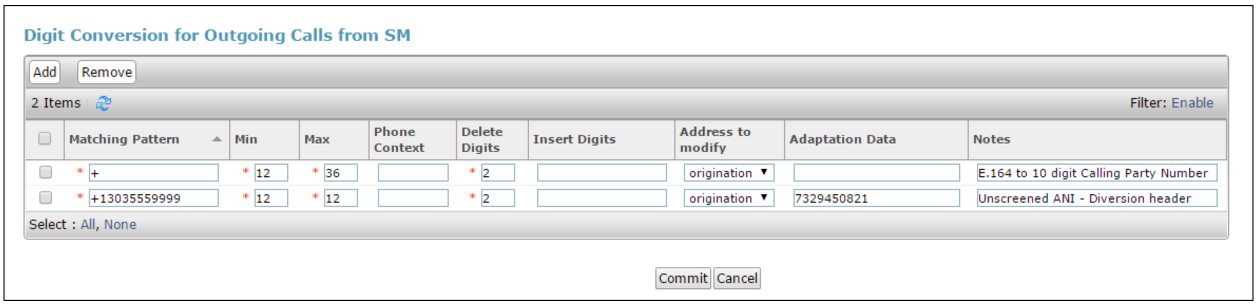
Step 3 - In the **Name-Value Parameter** table, enter the following:

1. **Name** – Enter **eRHdrs**
    - **Value** – Enter the following Avaya headers to be removed by Session Manager. **"AV-Global-Session-ID, Alert-Info, Endpoint-View, P-AV-Message-Id, P-Charging-Vector, P-Location, AV-Correlation-ID, Av-Secure-Indication"**



Step 3 - Scroll down to the **Digit Conversion for Outgoing Calls from SM** section (the *outbound* digits to Verizon that need to be converted to 10-digit numbers).

1. As described in **Section 2.2**, **Item 4**, the E.164 formatted numbers sent by Communication Manager's public-unknown numbering table (**Section 6.9**), needs to be converted to 10 digit numbers expected by Verizon.
    - Enter **+** in the **Matching Pattern** column.
    - Enter **12** in the **Min/Max** columns.
    - Enter **2** in the **Delete Digits** column.
    - Specify that this should be applied to the SIP **origination** headers in the **Address to modify** column.
    - Enter any desired notes



Digit Conversion for Outgoing Calls from SM

Add | Remove

2 Items — Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* +	* 12	* 36		* 2		origination		E.164 to 10 digit Calling Party Number
	* +13035559999	* 12	* 12		* 2		origination	7329450821	Unscreened ANI - Diversion header

Select : All, None

Commit | Cancel

Note – The Screened Telephone Number (STN) provided by Verizon for this test is 7329450821. Typically, customers would have one or more STN; one for every location. A central Session Manager could be used to pass multiple STNs to Verizon based on a **Matching Pattern** (i.e., a user's Calling Line Identification). The STN would then be entered in the **Adaptation Data** field as shown above.

5.4. SIP Entities

In this section, SIP Entities are administered for the following SIP network elements:

- Session Manager (**Section 5.4.1**).
- Communication Manager for Verizon trunk access (**Section 5.4.2**) – This entity, and its associated Entity Link (using TLS with port 5081), is for calls to/from Verizon and Communication Manager via the Avaya SBCE.
- Communication Manager for local trunk access (**Section 5.4.3**) – This entity, and its associated Entity Link (using TLS with port 5061), is primarily for traffic between Avaya SIP telephones and Communication Manager, as well as calls to Messaging.
- Avaya SBCE (**Section 5.4.4**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls to/from the Verizon Business IP Trunking service via the Avaya SBCE.
- Messaging (**Section 5.4.5**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls to/from Messaging.
- Experience Portal (**Section 5.4.6**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls to/from Experience Portal.

Note – In the reference configuration, TLS is used as the transport protocol between Session Manager and Communication Manager (ports 5061 and 5081), and to the Avaya SBCE (port 5061). The connection between the Avaya SBCE and the Verizon Business IP Trunking service uses UDP/5071 per Verizon requirements.

5.4.1 Avaya Aura® Session Manager SIP Entity

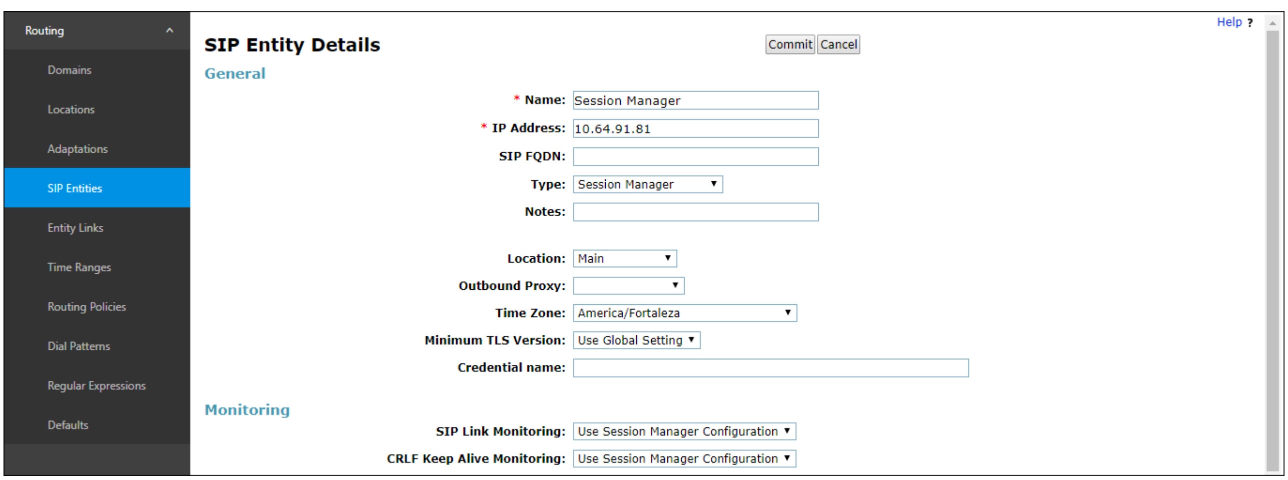
Step 1- In the left pane under **Routing**, click on **SIP Entities**. In the **SIP Entities** page click on **New** (not shown).

Step 2 - In the **General** section of the **SIP Entity Details** page, provision the following:

- **Name –** Enter a descriptive name (e.g., **SessionManager**).
- **FQDN or IP Address –** Enter the IP address of Session Manager signaling interface, (*not* the management interface), provisioned during installation (e.g., **10.64.91.81**).
- **Type –** Verify **Session Manager** is selected.
- **Location** – Select location **Main** (**Section 5.2.1**).
- **Outbound Proxy** – (Optional) Leave blank or select another SIP Entity. For calls to SIP domains for which Session Manager is not authoritative, Session Manager routes those calls to this **Outbound Proxy** or to another SIP proxy discovered through DNS if **Outbound Proxy** is not specified.
- **Time Zone** – Select the time zone in which Session Manager resides.
- **Minimum TLS Version** – Select the TLS version, or select **Use Global Settings** to use the default TLS version, configurable at the global level (**Elements→Session Manager→Global Settings**).

Step 3 - In the **Monitoring** section of the **SIP Entity Details** page configure as follows:

- Select **Use Session Manager Configuration** for **SIP Link Monitoring** field.
- Use the default values for the remaining parameters.



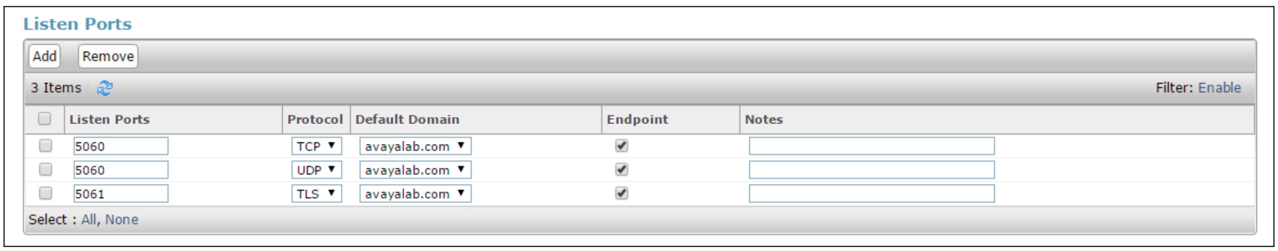
Step 4 - Scrolling down to the **Listen Port** section of the **SIP Entity Details** page. This section defines a default set of ports that Session Manager will use to listen for SIP requests, typically from registered SIP endpoints. Session Manager can also listen on additional ports defined elsewhere such as the ports specified in the SIP Entity Link definition in **Section 5.5.** Click on **Add** and provision entries as follows:

- **Port –** Enter **5061**
- **Protocol –** Select **TLS**
- **Default Domain –** Select a SIP domain administered in **Section 5.1** (e.g., **avayalab.com**)

Step 5 - **Repeat Step 4** to provision entries for any other listening ports used by Session Manager, for example:

- **5060** for **Port** and **TCP** for **Protocol**
- **5060** for **Port** and **UDP** for **Protocol**

Step 6 - Enter any notes as desired and leave all other fields on the page blank/default.
Step 7 - Click on **Commit**.



Note – The **Entity Links** section of the form (not shown) will be automatically populated when the Entity Links are defined in **Section 5.5**. The **SIP Responses to an OPTIONS Request** section of the form is not used in the reference configuration.

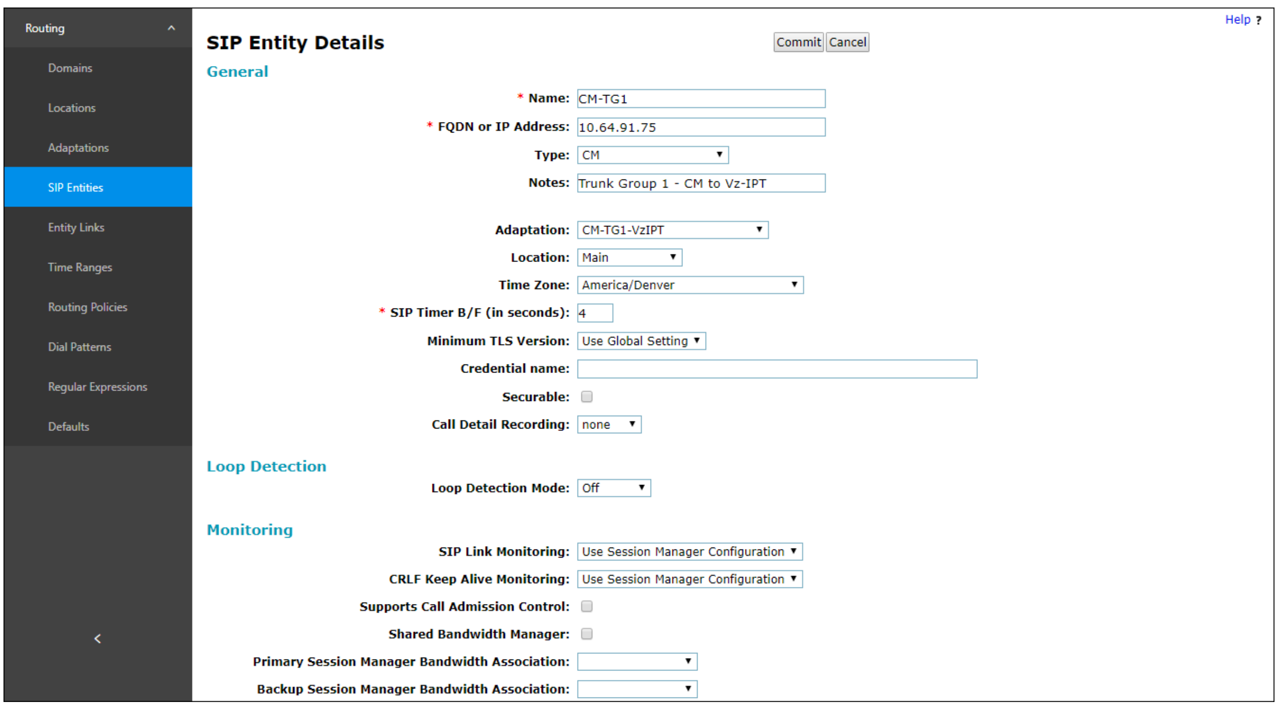
5.4.2 Avaya Aura® Communication Manager SIP Entity – Public Trunk

Step 1 - In the **SIP Entities** page, click on **New** (not shown).

Step 2 - In the **General** section of the **SIP Entity Details** page, provision the following:

- **Name –** Enter a descriptive name (e.g., **CM-TG1**).
- **FQDN or IP Address –** Enter the IP address of Communication Manager Processor Ethernet (procr) described in **Section 6.4** (e.g., **10.64.91.75**).
- **Type –** Select **CM**.
- **Adaptation** – Select the Adaptation **CM-TG1-VzIPT** administered in **Section 5.3.1**.
- **Location** – Select a Location **Main** administered in **Section 5.2.1**.
- **Time Zone** – Select the time zone in which Communication Manager resides.
- In the **SIP Link Monitoring** section of the **SIP Entity Details** page select:
    - Select **Use Session Manager Configuration** for **SIP Link Monitoring** field and use the default values for the remaining parameters.

Step 3 - Click on **Commit**.



5.4.3 Avaya Aura® Communication Manager SIP Entity – Local Trunk

To configure the Communication Manager Local trunk SIP Entity, repeat the steps in **Section 5.4.2** with the following changes:

- **Name –** Enter a descriptive name (e.g., **CM-TG3**).
- **Adaptations** – Leave this field blank.

5.4.4 Avaya Session Border Controller for Enterprise SIP Entity

Repeat the steps in **Section 5.4.2** with the following changes:

- **Name –** Enter a descriptive name (e.g., **SBC1**).
- **FQDN or IP Address –** Enter the IP address of the A1 (private) interface of the Avaya SBCE (e.g., **10.64.91.50**, see **Section 8.5.1**).
- **Type –** Select **SIP Trunk**.
- **Adaptations –** Select Adaptation **SBC1-Adaptation for Verizon** (**Section 5.3.2**).

5.4.5 Avaya Aura® Messaging SIP Entity

Repeat the steps in **Section 5.4.2** with the following changes:

- **Name –** Enter a descriptive name (e.g., **Aura Messaging**).
- **FQDN or IP Address –** Enter the IP address of Messaging (e.g., **10.64.91.54**).
- **Type –** Select **Messaging**.
- **Adaptations –** Leave this field blank.

5.4.6 Avaya Aura® Experience Portal SIP Entity

Repeat the steps in **Section 5.4.2** with the following changes:

- **Name –** Enter a descriptive name (e.g., **ExperiencePortal**).
- **FQDN or IP Address –** Enter the IP address of Experience Portal (e.g., **10.64.91.90**).
- **Type –** Select **Voice Portal**.
- **Adaptations –** Leave this field blank.

5.5. Entity Links

In this section, Entity Links are administered for the following connections:

- Session Manager to Communication Manager Public trunk (**Section 5.5.1**).
- Session Manager to Communication Manager Local trunk (**Section 5.5.2**).
- Session Manager to Avaya SBCE (**Section 5.5.3**).
- Session Manager to Messaging (**Section 5.5.4**).
- Session Manager to Experience Portal (**Section 5.5.5**).

Note – Once the Entity Links have been committed, the link information will also appear on the associated SIP Entity pages configured in **Section 5.4**.

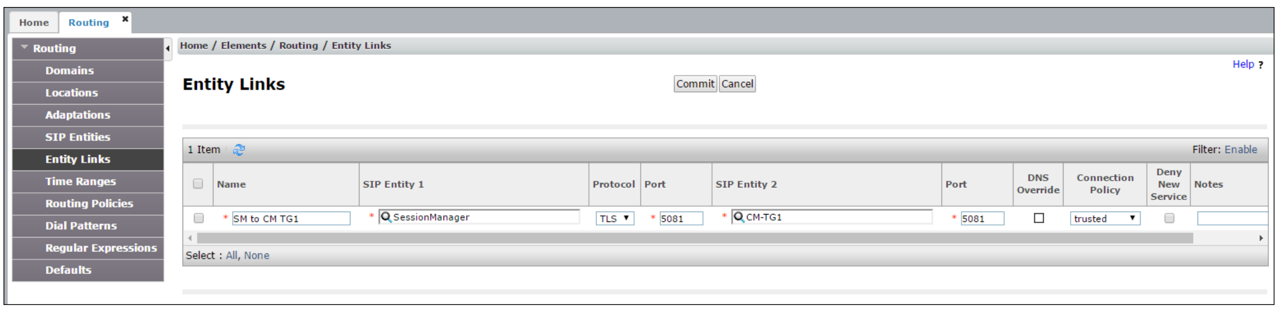
Note – See the information in **Section 5.4** regarding the transport protocols and ports used in the reference configuration.

5.5.1 Entity Link to Avaya Aura® Communication Manager – Public Trunk

Step 1 - In the left pane under **Routing**, click on **Entity Links**, then click on **New** (not shown).
Step 2 - Continuing in the **Entity Links** page, provision the following:

- **Name** – Enter a descriptive name for this link to Communication Manager (e.g., **SM to CM TG1**).
- **SIP Entity 1** – Select the SIP Entity administered in **Section 5.4.1** for Session Manager (e.g., **SessionManager**).
- **Protocol** – Select **TLS** (see **Section 6.8.1**).
- SIP Entity 1 **Port** – Enter **5081**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.2** for the Communication Manager public entity (e.g., **CM-TG1**).
- SIP Entity 2 **Port** – Enter **5081** (see **Section 6.8.1**).
- **Connection Policy** – Select **trusted**.
- Leave other fields as default.

Step 3 - Click on **Commit**.



5.5.2 Entity Link to Avaya Aura® Communication Manager – Local Trunk

To configure this Entity Link, repeat the steps in **Section 5.5.1**, with the following changes:

- **Name** – Enter a descriptive name for this link to Communication Manager (e.g., **SM to CM TG3**).
- SIP Entity 1 **Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.3** for the Communication Manager local entity (e.g., **CM-TG3**).
- SIP Entity 2 **Port** – Enter **5061** (see **Section 6.8.1**).

5.5.3 Entity Link for the Verizon Business IP Trunking service via the Avaya SBCE

To configure this Entity Link, repeat the steps in **Section 5.5.1**, with the following changes:

- **Name** – Enter a descriptive name for this link to the Avaya SBCE (e.g., **SM to SBC1**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.4** for the Avaya SBCE entity (e.g., **SBC1**).
- **SIP Entity 2 Port** – Enter **5061**.

5.5.4 Entity Link to Avaya Aura® Messaging

To configure this Entity Link, repeat the steps in **Section 5.5.1**, with the following changes:

- **Name** – Enter a descriptive name for this link to Messaging (e.g., **SM to AAM**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.5** for the Aura® Messaging entity (e.g., **Aura Messaging**).
- **SIP Entity 2 Port** – Enter **5061**.

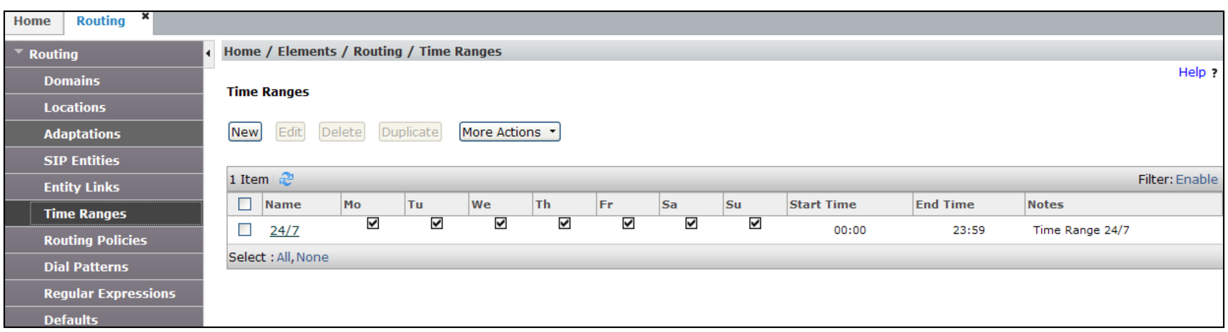
5.5.5 Entity Link to Avaya Aura® Experience Portal

To configure this Entity Link, repeat the steps in **Section 5.5.1**, with the following changes:

- **Name** – Enter a descriptive name for this link to Messaging (e.g., **SM to ExperiencePortal**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.6** for the Experience Portal entity (e.g., **ExperiencePortal**).
- **SIP Entity 2 Port** – Enter **5061**.

5.6. Time Ranges

Step 1 - In the left pane under **Routing**, click on **Time Ranges**. In the **Time Ranges** page click on **New**.
Step 2 - Continuing in the **Time Ranges** page, enter a descriptive **Name**, check the checkbox(s) for the desired day(s) of the week, and enter the desired **Start Time** and **End Time**.
Step 3 - Click on **Commit** (not shown). Repeat these steps to provision additional time ranges as required.



5.7. Routing Policies

In this section, the following Routing Policies are administered:

- Inbound calls to Communication Manager extensions (**Section 5.7.1**).
- Inbound calls to Messaging (**Section 5.7.2**).
- Inbound calls to Experience Portal (**Section 5.7.3**).
- Outbound calls to Verizon/PSTN (**Section 5.7.4**).

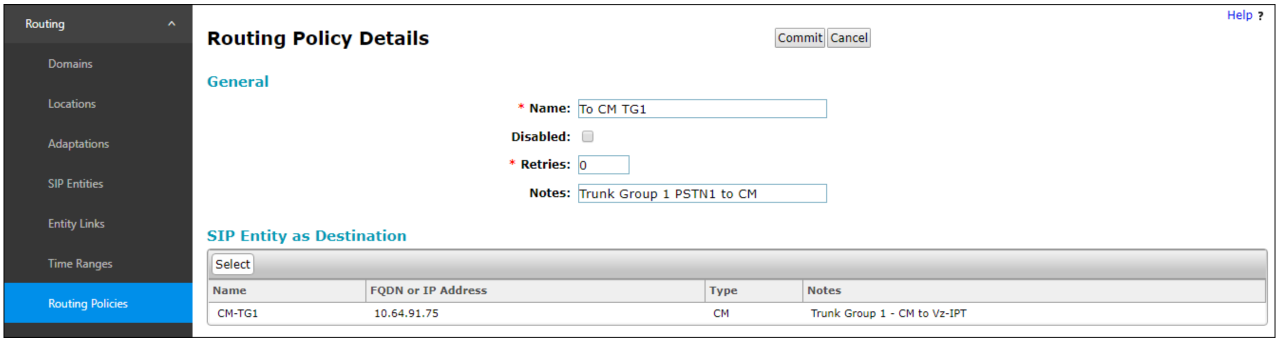
5.7.1 Routing Policy for Verizon Routing to Avaya Aura® Communication Manager

This Routing Policy is used for inbound calls from Verizon.

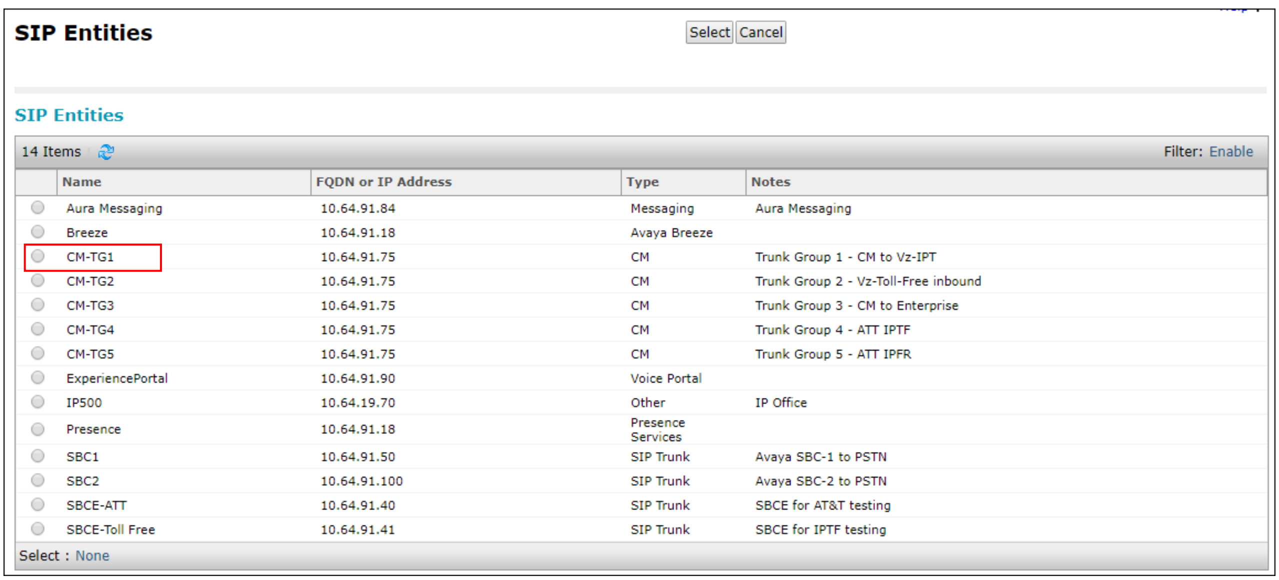
Step 1 - In the left pane under **Routing**, click on **Routing Policies**. In the **Routing Policies** page click on **New** (not shown).

Step 2 - In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing Verizon calls to Communication Manager (e.g., **To CM TG1**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.

Step 3 - In the **SIP Entity as Destination** section of the **Routing Policy Details** page, click on **Select** and the **SIP Entities** list page will open.



Step 4 - In the **SIP Entities** list page, select the SIP Entity administered in **Section 5.4.2** for the Communication Manager public SIP Entity (**CM-TG1**), and click on **Select**.



Name	FQDN or IP Address	Type	Notes
Aura Messaging	10.64.91.84	Messaging	Aura Messaging
Breeze	10.64.91.18	Avaya Breeze	
CM-TG1	10.64.91.75	CM	Trunk Group 1 - CM to Vz-IPT
CM-TG2	10.64.91.75	CM	Trunk Group 2 - Vz-Toll-Free inbound
CM-TG3	10.64.91.75	CM	Trunk Group 3 - CM to Enterprise
CM-TG4	10.64.91.75	CM	Trunk Group 4 - ATT IPTF
CM-TG5	10.64.91.75	CM	Trunk Group 5 - ATT IPFR
ExperiencePortal	10.64.91.90	Voice Portal	
IP500	10.64.19.70	Other	IP Office
Presence	10.64.91.18	Presence Services	
SBC1	10.64.91.50	SIP Trunk	Avaya SBC-1 to PSTN
SBC2	10.64.91.100	SIP Trunk	Avaya SBC-2 to PSTN
SBCE-ATT	10.64.91.40	SIP Trunk	SBCE for AT&T testing
SBCE-Toll Free	10.64.91.41	SIP Trunk	SBCE for IPTF testing

Step 5 - Returning to the **Routing Policy Details** page in the **Time of Day** section, click on **Add**.

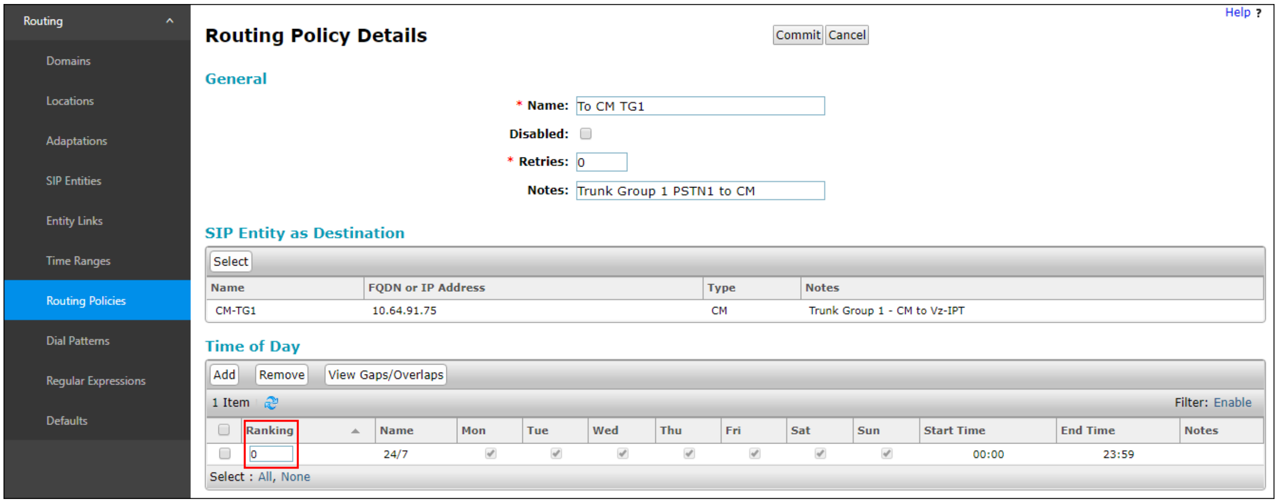
Step 6 - In the **Time Range List** page (not shown), check the checkbox(s) corresponding to one or more Time Ranges administered in **Section 5.6**, and click on **Select**.

Step 7 - Returning to the **Routing Policy Details** page in the **Time of Day** section, enter a **Ranking** of **0**.

Step 8 - No **Regular Expressions** were used in the reference configuration.

Step 9 - Click on **Commit**.

Note – Once the **Dial Patterns** are defined (**Section 5.8**) they will appear in the **Dial Pattern** section of this form.



5.7.2 Routing Policy for Inbound Routing to Avaya Aura® Messaging

This routing policy is for inbound calls to Aura® Messaging for message retrieval. Repeat the steps in **Section 5.7.1** with the following differences:

- Enter a descriptive **Name** (e.g., **To AAM**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.
- In the **SIP Entities** list page, select the SIP Entity administered in **Section 5.4.5** for Aura® Messaging (e.g., **AAM**).

5.7.3 Routing Policy for Inbound Routing to Experience Portal

This routing policy is for inbound calls to Experience Portal. Repeat the steps in **Section 5.7.1** with the following differences:

- Enter a descriptive **Name** (e.g., **To Experience Portal**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.
- In the **SIP Entities** list page, select the SIP Entity administered in **Section 5.4.6** for Experience Portal (e.g., **ExperiencePortal**).

5.7.4 Routing Policy for Outbound Calls to Verizon

This Routing Policy is used for outbound calls to Verizon. Repeat the steps in **Section 5.7.1** with the following differences:

- Enter a descriptive **Name** for routing calls to the Verizon Business IP Trunking service via the Avaya SBCE (e.g., **To SBC1**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.
- In the **SIP Entities** list page, select the SIP Entity administered in **Section 5.4.4** for the Avaya SBCE SIP Entity (e.g., **SBC1**).

5.8. Dial Patterns

In this section, Dial Patterns are administered matching the following calls:

- Inbound PSTN calls via the Verizon Business IP Trunking service to Communication Manager (**Section 5.8.1**).
- Outbound calls to Verizon/PSTN (**Section 5.8.2**).

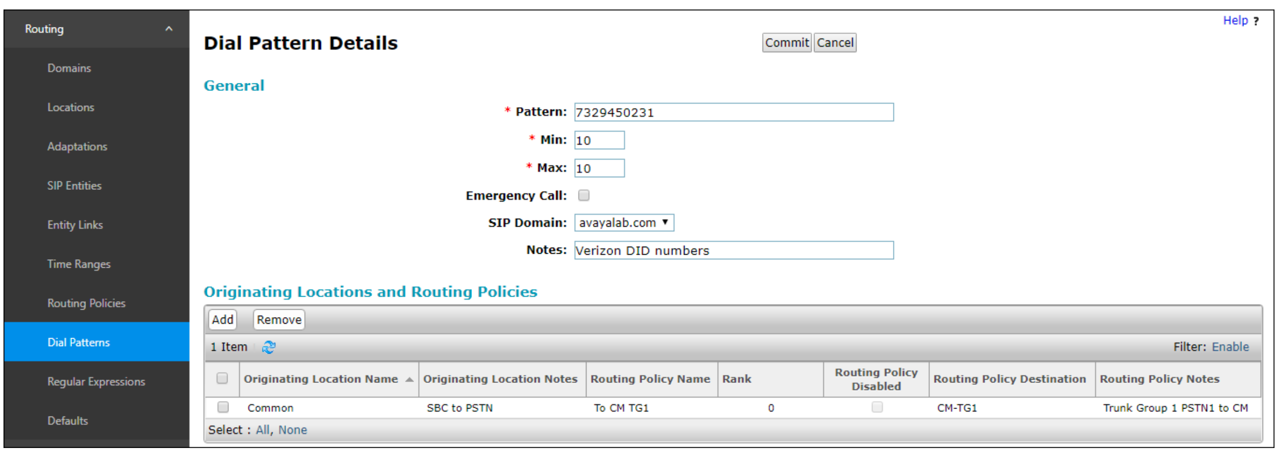
5.8.1 Matching Inbound PSTN Calls to Avaya Aura® Communication Manager

In the reference configuration inbound calls from the Verizon Business IP Trunking service sent 10 DNIS digits in the SIP Request URI. The DNIS pattern must be matched for further call processing.

Step 1 - In the left pane under **Routing**, click on **Dial Patterns**. In the **Dial Patterns** page click on **New** (not shown).

Step 2 - In the **General** section of the **Dial Pattern Details** page, provision the following:

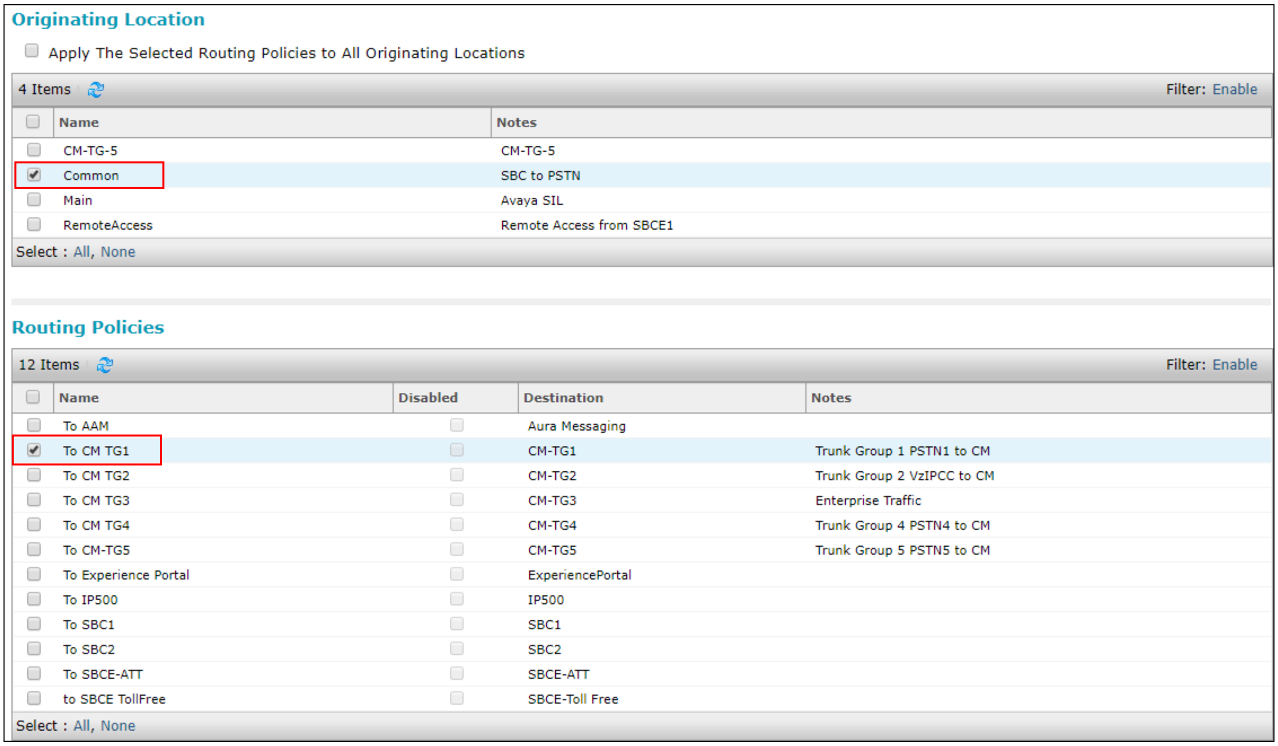
- **Pattern** – Enter **7329450231**. Note – The Adaptation defined for Communication Manager in **Section 5.3.1** will convert the various 732-945-0xxx numbers into their corresponding Communication Manager extensions.
- **Min** and **Max** – Enter **10**.
- **SIP Domain** – Select the enterprise SIP domain, e.g., **avayalab.com**.



Step 3 - Scrolling down to the **Originating Location and Routing Policies** section of the **Dial Pattern Details** page and click on **Add**.

Step 4 - In the **Originating Location**, check the checkbox corresponding to the Avaya SBCE location, e.g., **Common**.

Step 5 - In the **Routing Policies** section, check the checkbox corresponding to the Routing Policy administered for routing calls to the Communication Manager public trunk in **Section 5.7.1** (e.g., **To CM TG1**) and click on **Select** (not shown).



Originating Location

Apply The Selected Routing Policies to All Originating Locations

4 Items — Filter: Enable

Name	Notes
CM-TG-5	CM-TG-5
Common	SBC to PSTN
Main	Avaya SIL
RemoteAccess	Remote Access from SBCE1

Select : All, None

Routing Policies

12 Items — Filter: Enable

Name	Disabled	Destination	Notes
To AAM		Aura Messaging	
To CM TG1		CM-TG1	Trunk Group 1 PSTN1 to CM
To CM TG2		CM-TG2	Trunk Group 2 VzIPCC to CM
To CM TG3		CM-TG3	Enterprise Traffic
To CM TG4		CM-TG4	Trunk Group 4 PSTN4 to CM
To CM-TG5		CM-TG5	Trunk Group 5 PSTN5 to CM
To Experience Portal		ExperiencePortal	
To IP500		IP500	
To SBC1		SBC1	
To SBC2		SBC2	
To SBCE-ATT		SBCE-ATT	
to SBCE TollFree		SBCE-Toll Free	

Select : All, None

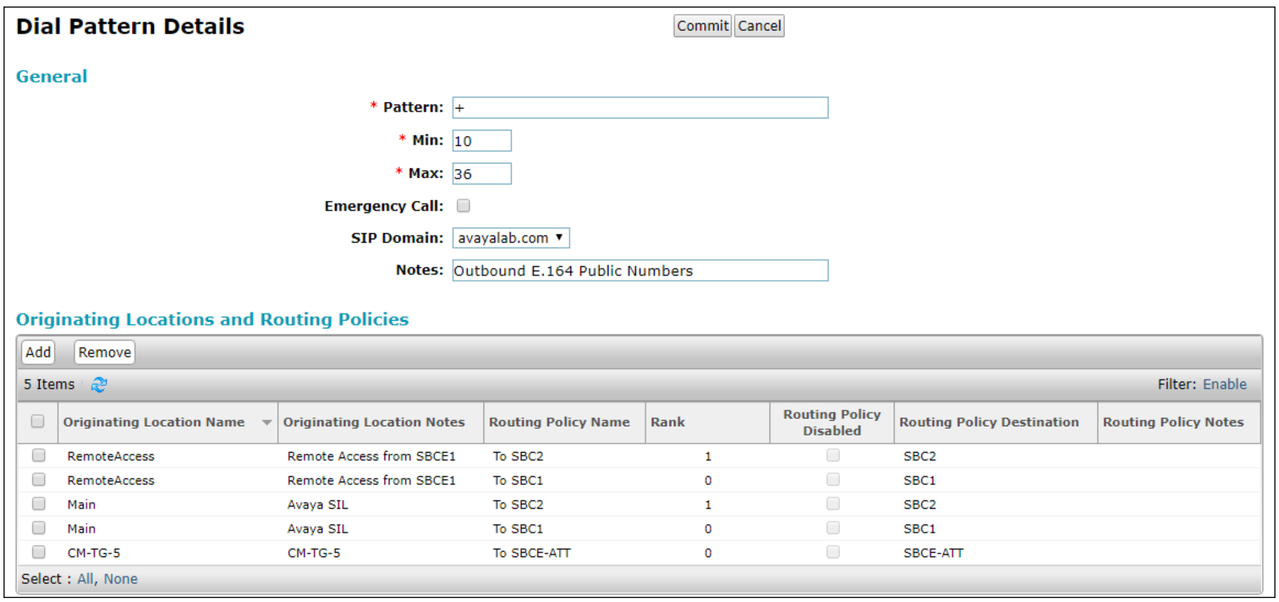
Step 6 - Returning to the Dial Pattern Details page and click on **Commit**.
Step 7 - Repeat **Steps 1-6** for any additional inbound dial patterns from Verizon.

5.8.2 Matching Outbound Calls to Verizon/PSTN

In this section, Dial Patterns are administered for all outbound calls to Verizon/PSTN. In the reference configuration E.164 numbers were used for national and international calls. Non-E.164 numbers were used for service numbers, e.g., x11, 1411, 5551212, etc.

Step 1 - Repeat the steps shown in **Section 5.8.1**, with the following changes:

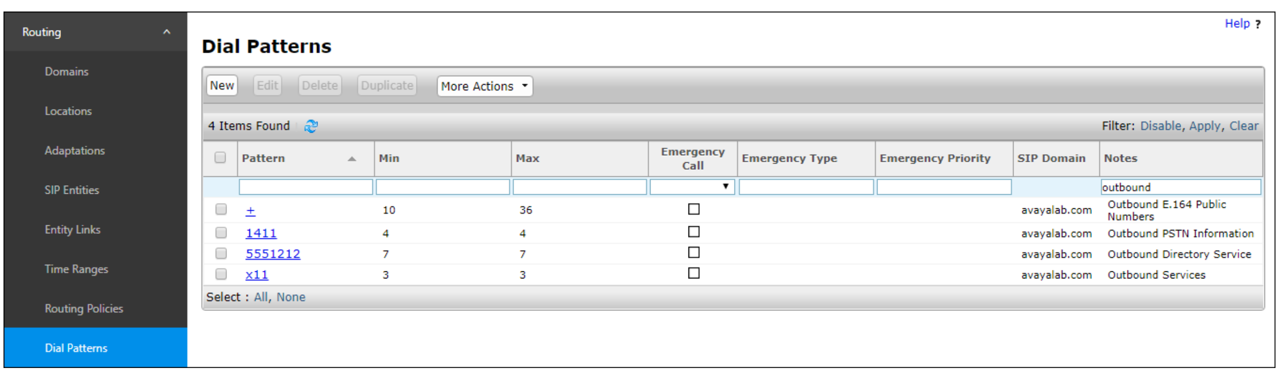
- In the **General** section of the **Dial Pattern Details** page, enter a dial pattern for routing calls to Verizon/PSTN (e.g., +). This will match any outbound call prefixed with a plus sign (+), such as an E.164 formatted number.
- Enter a **Min** pattern of **10**.
- Enter a **Max** pattern of **36**.
- In the **Routing Policies** section of the **Originating Locations and Routing Policies** page, check the checkboxes corresponding to the Communication Manager Originating Location (e.g., **Main**) and the Routing Policy administered for routing calls to Verizon in **Section 5.7.4** (e.g., **To SBC1**).



Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
RemoteAccess	Remote Access from SBCE1	To SBC2	1		SBC2	
RemoteAccess	Remote Access from SBCE1	To SBC1	0		SBC1	
Main	Avaya SIL	To SBC2	1		SBC2	
Main	Avaya SIL	To SBC1	0		SBC1	
CM-TG-5	CM-TG-5	To SBCE-ATT	0		SBCE-ATT	

Select : All, None

Step 2 - Repeat **Step 1** to add any additional outbound patterns as required.



Pattern	Min	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
							outbound
+	10	36	☐			avayalab.com	Outbound E.164 Public Numbers
1411	4	4	☐			avayalab.com	Outbound PSTN Information
5551212	7	7	☐			avayalab.com	Outbound Directory Service
x11	3	3	☐			avayalab.com	Outbound Services

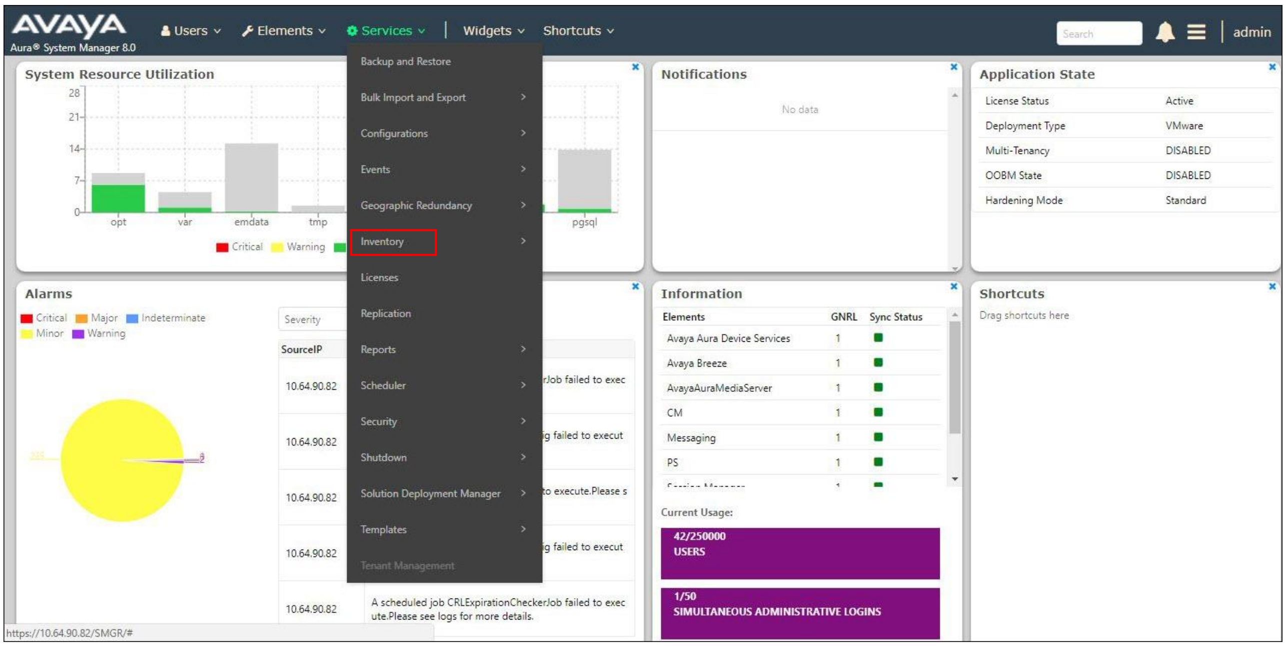
Select : All, None

5.9. Verify TLS Certificates – Session Manager

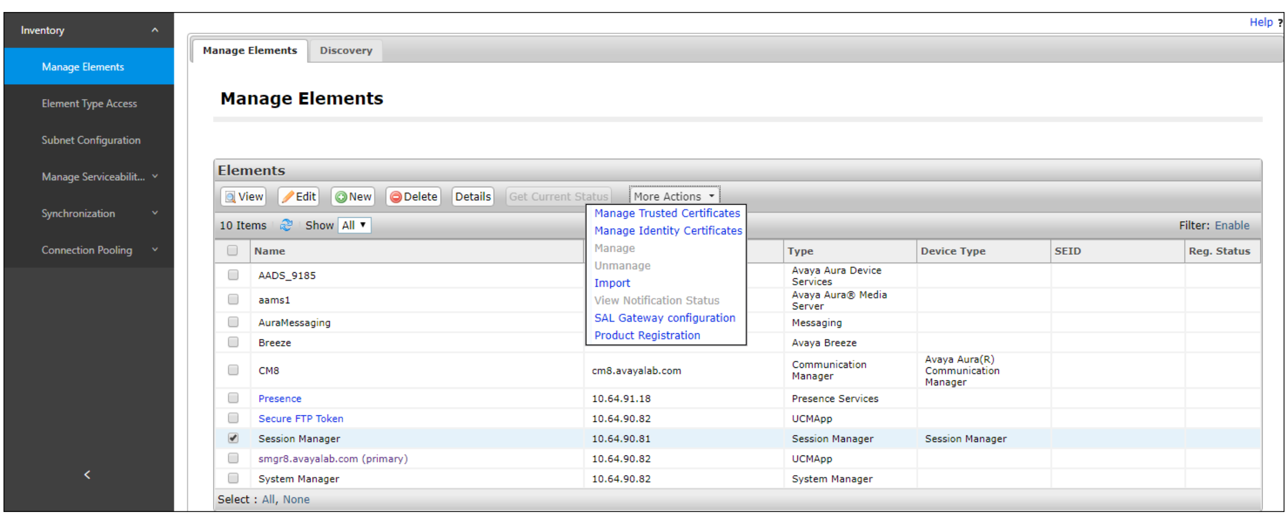
Note – Testing was done with System Manager signed identity certificates. The procedure to obtain and install certificates is outside the scope of these Application Notes.

The following procedures show how to verify the certificates used by Session Manager.

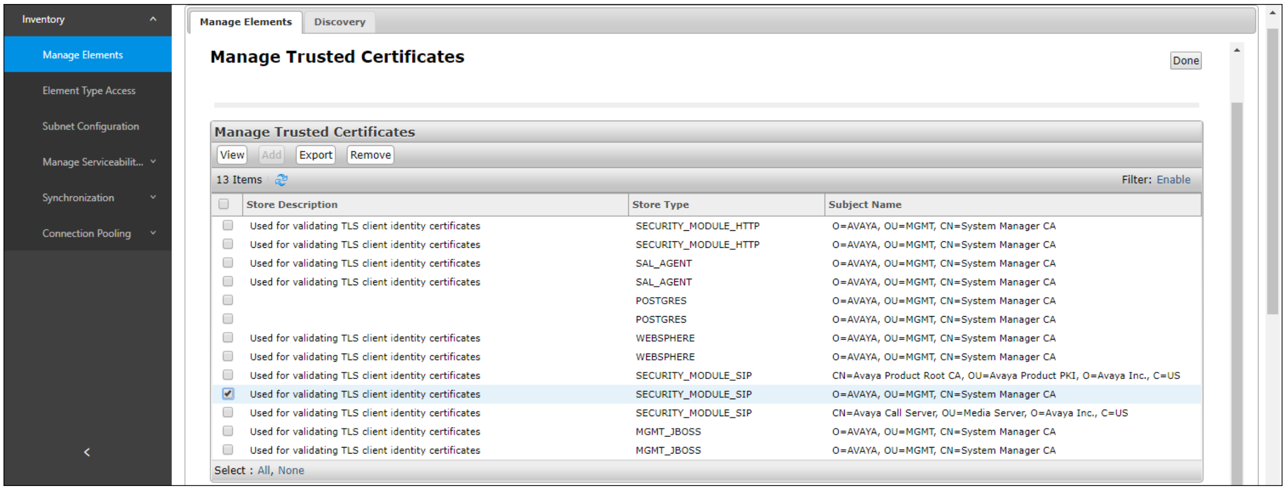
Step 1 - From the **Home** screen, under the **Services** heading, select **Inventory**.



Step 2 - In the left pane under **Inventory**, click on **Manage Elements** and select the Session Manager element, e.g., **SessionManager**. Click on **More Actions → Manage Trusted Certificates**.

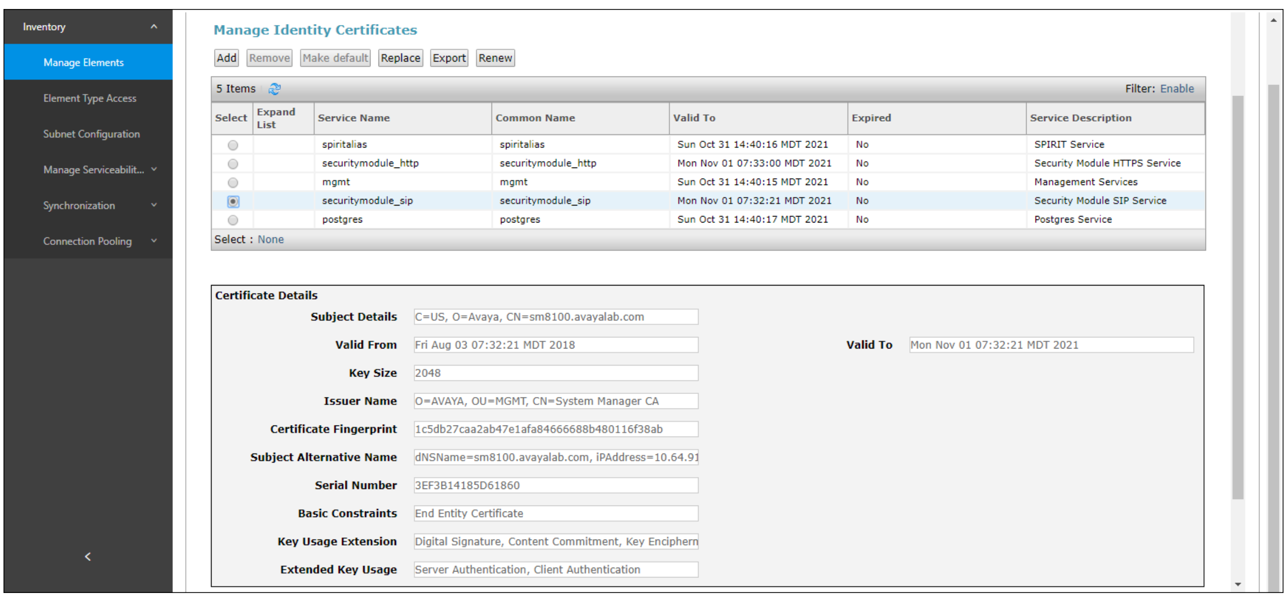


Step 3 - Verify the **System** Manager Certificate Authority certificate is listed in the trusted store, **SECURITY_MODULE_SIP**. Click **Done** to return to the previous screen.



Step 4 - With **Session Manager** selected, click on **More Actions → Manage Identity Certificates** (not shown).

Step 5 - Verify the **Security Module SIP** service has a valid identity certificate signed by System Manager. If the **Subject Details** and **Subject Alternative Name** fields of the System Manager signed certificate need to be updated, click **Replace**, otherwise click **Done** (not shown).



6. Configure Avaya Aura® Communication Manager Release 8.0

This section illustrates an example configuration allowing SIP signaling via the "Processor Ethernet" of Communication Manager to Session Manager.

Note – The initial installation, configuration, and licensing of the Avaya servers and media gateways for Communication Manager are assumed to have been previously completed and are not discussed in these Application Notes.

6.1. Verify Licensed Features

Note – This section describes steps to verify Communication Manager feature settings that are required for the reference configuration described in these Application Notes. Depending on access privileges and licensing, some or all of the following settings might only be viewed, and not modified. If any of the required features are not set, and cannot be configured, contact an authorized Avaya account representative to obtain the necessary licenses/access.

Step 1 - Enter the **display system-parameters customer-options** command. On **Page 2** of the form, verify that the **Maximum Administered SIP Trunks** number is sufficient for the number of expected SIP trunks.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  0
           Maximum Concurrently Registered IP Stations: 2400  1
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  3
                   Maximum Video Capable IP Softphones: 2400  10
                       Maximum Administered SIP Trunks: 4000  60
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
```

Step 2 - On **Page 4** of the form, verify that **ARS** is enabled.

```
display system-parameters customer-options                      Page   4 of  12
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                         CAS Main? n
Answer Supervision by Call Classifier? y                Change COR by FAC? n
                                 ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? n                       DCS (Basic)? y
          ASAI Link Core Capabilities? n                 DCS Call Coverage? y
          ASAI Link Plus Capabilities? n                DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                           DS1 MSP? y
                                ATMS? y             DS1 Echo Cancellation? y
                  Attendant Vectoring? y
```

Step 3 - On **Page 5** of the form, verify that the **Enhanced EC500**, **IP Trunks**, and **ISDN-PRI,** features are enabled. If the use of SIP REFER messaging will be required verify that the **ISDN/SIP Network Call Redirection** feature is enabled. If the use of SRTP will be required verify that the **Media Encryption Over IP** feature is enabled.

```
display system-parameters customer-options                      Page   5 of  12
                                OPTIONAL FEATURES
    Emergency Access to Attendant? y                             IP Stations? y
            Enable 'dadmin' Login? y
            Enhanced Conferencing? y                      ISDN Feature Plus? n
                   Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
     Enterprise Survivable Server? n                        ISDN-BRI Trunks? y
        Enterprise Wide Licensing? n                               ISDN-PRI? y
               ESS Administration? y              Local Survivable Processor? n
           Extended Cvg/Fwd Admin? y                    Malicious Call Trace? y
      External Device Alarm Admin? y               Media Encryption Over IP? y
   Five Port Networks Max Per MCC? n    Mode Code for Centralized Voice Mail? n
                 Flexible Billing? n
    Forced Entry of Account Codes? y                Multifrequency Signaling? y
       Global Call Classification? y        Multimedia Call Handling (Basic)? y
              Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y               Multimedia IP SIP Trunking? y
                        IP Trunks? y

            IP Attendant Consoles? y
```

Step 4 - On **Page 6** of the form, verify that the **Processor Ethernet** field is set to **y**.

```
display system-parameters customer-options                      Page   6 of  12
                                OPTIONAL FEATURES

                Multinational Locations? n            Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n      Station as Virtual Extension? y
                     Multiple Locations? n
                                             System Management Data Transfer? n
          Personal Station Access (PSA)? y               Tenant Partitioning? y
                        PNC Duplication? n        Terminal Trans. Init. (TTI)? y
                   Port Network Support? y               Time of Day Routing? y
                        Posted Messages? y        TN2501 VAL Maximum Capacity? y
                                                         Uniform Dialing Plan? y
                     Private Networking? y      Usage Allocation Enhancements? y
               Processor and System MSP? y
                     Processor Ethernet? y                 Wideband Switching? y
                                                                    Wireless? n
                          Remote Office? y
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y
```

6.2. System-Parameters Features

Step 1 - Enter the **display system-parameters features** command. On **Page 1** of the form, verify that the **Trunk-to-Trunk Transfer** is set to **all**.

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

              Music (or Silence) on Transferred Trunk Calls? all
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n


             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

6.3. Dial Plan

The dial plan defines how digit strings will be used locally by Communication Manager. The following dial plan was used in the reference configuration.

Step 1 - Enter the **change dialplan analysis** command to provision the following dial plan.

- 5-digit extensions with a **Call Type** of **ext** beginning with:
  - The digits **1**, **5**, **7** and **8** for Communication Manager extensions.
- 3-digit dial access code (indicated with a **Call Type** of **dac**), e.g., access code ***xx** for SIP Trunk Access Codes (TAC). See the trunk forms in **Section 6.8**.

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

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
 1             5  ext
 2             5  ext
 3             5  ext
 4             5  ext
 5             5  ext
 60            3  ext
 66            2  fac
 7             5  ext
 8             5  ext
 9             1  fac
 *             3  dac
```

6.4. Node Names

Node names define IP addresses to various Avaya components in the enterprise. In the reference configuration a Processor Ethernet (procr) based Communication Manager platform is used. Note that the Communication Manager procr name and IP address are entered during installation. The procr IP address was used to define the Communication Manager SIP Entities in **Section 5.4**.

Step 1 - Enter the **change node-names ip** command, and add a node name and IP address for the following:

- Session Manager SIP signaling interface (e.g., **SM** and **10.64.91.81**).
- Media Server (e.g., **AMS** and **10.64.91.80**). The Media Server node name is only needed if a Media Server is present.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                IP Address
AMS                   10.64.91.80
SM                    10.64.91.81
default               0.0.0.0
procr                 10.64.91.75
procr6                ::
```

6.5. Processor Ethernet Configuration

The **display ip-interface procr** command can be used to verify the Processor Ethernet (procr) parameters defined during installation.

- Verify that **Enable Interface?**, **Allow H.323 Endpoints?**, and **Allow H248 Gateways?** fields are set to **y**.
- In the reference configuration the procr is assigned to **Network Region: 1**.
- The default values are used for the remaining parameters.

```
change ip-interface procr                                      Page   1 of   2
                                 IP INTERFACES


                  Type: PROCR
                                                        Target socket load: 4800

      Enable Interface? y                           Allow H.323 Endpoints? y
                                                     Allow H.248 Gateways? y
        Network Region: 1                             Gatekeeper Priority: 5


                                IPV4 PARAMETERS
             Node Name: procr                    IP Address: 10.64.91.75


           Subnet Mask: /24
```

6.6. IP Codec Sets

6.6.1 Codecs for IP Network Region 1 (calls within the CPE)

Step 1 - Enter the **change ip-codec-set x** command, where **x** is the number of an IP codec set used for internal calls (e.g., **1**). On **Page 1** of the **ip-codec-set** form, ensure that **G.711MU**, and **G.729A** are included in the codec list.

```
change ip-codec-set 1                                          Page   1 of   2
                         IP Codec Set
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.722-64K                   2        20
 2: G.711MU           n         2        20
 3: G.729A            n         2        20

     Media Encryption                  Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: none
```

Step 2 - On **Page 2** of the ip-codec-set form, set **FAX Mode** to **t.38-standard**, and **ECM** to **y**.

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

                          IP MEDIA PARAMETERS

                              Allow Direct-IP Multimedia? y
              Maximum Call Rate for Direct-IP Multimedia: 15360:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia: 15360:Kbits

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

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

6.6.2 Codecs for IP Network Region 2 (calls to/from Verizon)

This IP codec set will be used for Verizon Business IP Trunking calls. Repeat the steps in **Section 6.6.1** with the following changes:

- Provision the codecs in the order shown below.
- On **Page 2**, set **FAX Mode** to **t.38-G711-fallback**, **ECM** to **y**, and **FB-Timer** to **4**. See **Section 2.2** for limitations regarding fax.

```
change ip-codec-set 2                                           Page   1 of   2
                          IP CODEC SET
    Codec Set: 2

    Audio         Silence      Frames   Packet
    Codec         Suppression  Per Pkt  Size(ms)
 1: G.729A            n          2        20
 2: G.711MU           n          2        20
 3:

     Media Encryption                  Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: none

change ip-codec-set 2                                           Page   2 of   2

                          IP MEDIA PARAMETERS

                              Allow Direct-IP Multimedia? y
              Maximum Call Rate for Direct-IP Multimedia:   384:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia:   384:Kbits

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

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

6.7. Network Regions

Network regions provide a means to logically group resources. In the shared Communication Manager configuration used for the testing, the Avaya G450 Media Gateway and Avaya Media Server are in region 1. To provide testing flexibility, network region 2 was associated with other components used specifically for the Verizon testing.

6.7.1 IP Network Region 1 – Local CPE Region

Step 1 - Enter **change ip-network-region x**, where **x** is the number of an unused IP network region (e.g., region **1**). This IP network region will be used to represent the local CPE. Populate the form with the following values:

- Enter a descriptive name (e.g., **Enterprise**).
- Enter the enterprise domain (e.g., **avayalab.com**) in the **Authoritative Domain** field (see **Section 5.1**).

- Enter **1** for the **Codec Set** parameter.
- **Intra-region IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible within the same region.
- **Inter-region IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible between regions.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avayalab.com
    Name: Enterprise            Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Step 2 - On **page 2** of the form:

- Verify that **RTCP Reporting to Monitor Server Enabled** is set to **y**.

```
change ip-network-region 1                                      Page   2 of  20
                               IP NETWORK REGION

 RTCP Reporting to Monitor Server Enabled? y

 RTCP MONITOR SERVER PARAMETERS
   Use Default Server Parameters? y
```

Step 3 - On **page 4** of the form:

- Verify that next to region **1** in the **dst rgn** column, the codec set is **1**.
- Next to region **2** in the **dst rgn** column, enter **2** for the codec set (this means region 1 is permitted to talk to region 2 and it will use codec set 2 to do so). The **direct WAN** and **Units** columns will self-populate with **y** and **No Limit** respectively.
- Let all other values default for this form.

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

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

6.7.2 IP Network Region 2 – Verizon Trunk Region

Repeat the steps in **Section 6.7.1** with the following changes:

Step 1 - On **Page 1** of the form (not shown):

- Enter a descriptive name (e.g., **Verizon**).
- Enter **2** for the **Codec Set** parameter.

Step 2 - On **Page 4** of the form:

- Set codec set **2** for **dst rgn 1**.
- Note that **dst rgn 2** is pre-populated with codec set **2** (from page 1 provisioning).

```
change ip-network-region 2                                      Page   4 of  20

 Source Region: 2     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    2      y    NoLimit                                            n       t
 2    2                                                                all
 3
```

6.8. SIP Trunks

SIP trunks are defined on Communication Manager by provisioning a Signaling Group and a corresponding Trunk Group. Two SIP trunks are defined on Communication Manager in the reference configuration:

- Inbound/outbound Verizon access – SIP Trunk 1
  - Note that this trunk will use TLS port 5081 as described in **Section 5.5.1**.
- Internal CPE access (e.g., Avaya SIP telephones, Messaging, etc.) – SIP Trunk 3
  - Note that this trunk will use TLS port 5061 as described in **Section 5.5.2**.

Note – Although TLS is used as the transport protocols between the Avaya CPE components, UDP was used between the Avaya SBCE and the Verizon IP Trunk service. See the note in **Section 5.4** regarding the use of TLS transport protocols in the CPE.

6.8.1 SIP Trunk for Inbound/Outbound Verizon calls

This section describes the steps for administering the SIP trunk to Session Manager used for Verizon IP Trunk service calls. Trunk 1 is defined. This trunk corresponds to the **CM-TG1** SIP Entity defined in **Section 5.4.2**.

6.8.1.1 Signaling Group 1

Step 1 - Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **1**), and provision the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tls**.
- Verify that **IMS Enabled?** is set to **n**.
- Verify that **Peer Detection Enabled?** is set to **y**. The system will auto detect and set the **Peer Server** to **SM**.
- **Near-end Node Name** – Set to the node name of the **procr** noted in **Section 6.4**.
- **Far-end Node Name** – Set to the node name of Session Manager as administered in **Section 6.4** (e.g., **SM**).
- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5081**.
- **Far-end Network Region** – Set the IP network region to **2**, as set in **Section 6.6.2**.
- **Far-end Domain** – Enter **avayalab.com**. This is the domain provisioned for Session Manager in **Section 5.1**.
- **DTMF over IP** – Set to **rtp-payload** to enable Communication Manager to use DTMF according to RFC 2833.
- **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized directly to the associated stations, to reduce the use of media resources on the Avaya Media Gateway when possible (known as shuffling).
- **Initial IP-IP Direct Media** is set to **n**. See **Section 2.2** for details.
- **H.323 Station Outgoing Direct Media** is set to **n**.

Use the default parameters on **page 2** of the form (not shown).

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

 Group Number: 1             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM                      Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: SM
 Near-end Listen Port: 5081                 Far-end Listen Port: 5081
                                         Far-end Network Region: 2

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

6.8.1.2 Trunk Group 1

Step 1 - Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g., **1**). On **Page 1** of the **trunk-group** form, provision the following:

- **Group Type** – Set to **sip**.
- **Group Name** – Enter a descriptive name (e.g., **Verizon IPT**).
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., ***01**).
- **Direction** – Set to **two-way**.
- **Service Type** – Set to **public-ntwrk**.
- **Signaling Group** – Set to the signaling group administered in **Section 6.8.1.1** (e.g., **1**).
- **Number of Members** – Enter the maximum number of simultaneous calls desired on this trunk group (based on licensing) (e.g., **10**).

```
add trunk-group 1                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: Verizon IPT                COR: 1        TN: 1        TAC: *01
   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: 1
                                                     Number of Members: 10
```

Step 2 - On **Page 2** of the **Trunk Group** form:

- Set the **Preferred Minimum Session Refresh Interval(sec):** to **900**. This entry will actually cause a value of 1800 to be generated in the SIP Session-Expires header pertaining to active call session refresh.

```
add trunk-group 1                                              Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900

 Disconnect Supervision - In? y  Out? y

             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n

Caller ID for Service Link Call to H.323 1xC: station-extension
```

Step 3 - On **Page 3** of the **Trunk Group** form:

- Set **Numbering Format** to **public**.

```
add trunk-group 1                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y

   Suppress # Outpulsing? n  Numbering Format: public
                                                UUI 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
```

Step 4 - On **Page 4** of the **Trunk Group** form:

- Verify **Network Call Redirection** is set to **y**.
- Set **Telephone Event Payload Type** to the RTP payload type recommended by Verizon (e.g., **101**).
- Set **Convert 180 to 183 for Early Media** to **y**. Verizon prefers to have Communication Manager send 183 with SDP rather than a 180 with SDP.

Note – The Verizon Business IP Trunking service does not support History Info header. As shown below, by default this header is supported by Communication Manager. In the reference configuration, the History Info header is automatically removed from SIP signaling by Session Manager, as part of the *VerizonAdapter* (see **Section 5.3.2**). Alternatively, History Info may be disabled here with the Diversion Header enabled.

```
add trunk-group 1                                               Page   4 of  21
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 101
                                       Shuffling with SDP? n

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

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

6.8.2 Local SIP Trunk (Avaya SIP Telephone and Messaging Access)

Trunk 3 corresponds to the **CM-TG3** SIP Entity defined in **Section 5.4.3**.

6.8.2.1 Signaling Group 3

Repeat the steps in **Section 6.8.1.1** with the following changes:

Step 1 - Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **3**).

Step 2 - Set the following parameters on page 1:

- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5061**
- **Far-end Network Region** – Set to the IP network region **1**, as defined in **Section 6.6.1**.

6.8.2.2 Trunk Group 3

Repeat the steps in **Section 6.8.1.2** with the following changes:

Step 1 - Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g., **3**). On **Page 1** of the **trunk-group** form:

- **Group Name** – Enter a descriptive name (e.g., **SM Enterprise**).
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., ***03**).
- **Service Type** – Set to **tie**.
- **Signaling Group** – Set to the number of the signaling group administered in **Section 6.8.2.1** (e.g., **3**).

Step 2 - On **Page 2** of the **Trunk Group** form:

- Same as **Section 6.8.1.2**

Step 3 - On **Page 3** of the **Trunk Group** form:

- Set **Numbering Format** to **private**.

Step 4 - On **Page 4** of the **Trunk Group** form:

- Set **Network Call Redirection** to **n**.
- Set **Send Diversion Header** to **n**.
- Verify **Identity for Calling Party Display** is set to **P-Asserted-Identity** (default).

Use default values for all other settings.

6.9. Public Numbering

In the reference configuration, the public-unknown-numbering form, (used in conjunction with the **Numbering Format: public** setting in **Section 6.8.1.2**), is used to convert Communication Manager local extensions to Verizon public numbers, for inclusion in any SIP headers directed to the Verizon Business IP Trunking service via the public trunk.

Step 1 - Enter **change public-unknown-numbering 5 ext-digits xxxxx**, where xxxxx is the 5-digit extension number to change.

Step 2 - Add each Communication Manager station extension and their corresponding Verizon DNIS numbers (for the public trunk to Verizon). Communication Manager will insert these Verizon DNIS numbers in E.164 format into the From, Contact, and PAI headers as appropriate:

- **Ext Len** – Enter the total number of digits in the local extension range (e.g., **5**).
- **Ext Code –** Enter a Communication Manager extension (e.g., **12002**.
- **Trk Grp(s)** – Enter the number of the Public trunk group (e.g., **1**).
- **Private Prefix** – Enter the corresponding Verizon DNIS number (e.g., 1**7329450232**).
- **Total Len** – Enter the total number of digits after the digit conversion (e.g., **11**).

```
change public-unknown-numbering 5 ext-digits 12001             Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk       CPN             CPN
Len Code            Grp(s)    Prefix          Len
                                                Total Administered: 46
 5  12001           1         17329450231     11      Maximum Entries: 240
 5  14006           1         17329450236     11
 5  14007           1         17329450237     11    Note: If an entry applies to
 5  14008           1         17329450238     11    a SIP connection to Avaya
 5  50              1         173294          11    Aura(R) Session Manager,
                                                    the resulting number must
                                                    be a complete E.164 number.
```

6.10. Private Numbering

In the reference configuration, the private-numbering form, (used in conjunction with the **Numbering Format: private** setting in **Section 6.8.2.2**), is used to send Communication Manager local extension numbers to Session Manager, for inclusion in any SIP headers directed to SIP endpoints and Messaging.

Step 1 - Add all Communication Manager local extension patterns (for the local trunk).

- **Ext Len** – Enter the total number of digits in the local extension range (e.g., **5**).
- **Ext Code –** Enter the Communication Manager extension patterns defined in the Dial Plan in **Section 6.3** (e.g., **5**, **14** and **20**).
- **Trk Grp(s)** – Enter the number of the Local trunk group (e.g., **3**).
- **Total Len** - Enter the total number of digits after the digit conversion (e.g., **5**).

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

Ext Ext                 Trk        Private          Total
Len Code                Grp(s)     Prefix           Len
 5  1                   11                          5  Total Administered: 11
 5  5                   3                           5     Maximum Entries: 540
 5  14                  3                           5
 5  20                  3                           5
```

6.11. Route Patterns

Route Patterns are used to direct outbound calls via the public or local CPE SIP trunks.

6.11.1 Route Pattern for National Calls to Verizon

This form defines the public SIP trunk, based on the route-pattern selected by the ARS table in **Section 6.12**. The routing defined in this section is simply an example and not intended to be prescriptive. Other routing policies may be appropriate for different customer networks. In the reference configuration, route pattern 1 is used for national calls, route pattern 2 is used for international calls, and route pattern 4 is used for service calls.

Step 1 - Enter the **change route-pattern 1** command to configure a route pattern for national calls and enter the following parameters:

- In the **Grp No** column, enter **1** for public trunk 1, and the **FRL** column enter **0** (zero).
- In the **Pfx mrk** column, enter **1** to ensure a 1 + 10 digits are sent to the service provider for FNPA calls.
- In the **Inserted Digits** column, enter **p** to have Communication Manager insert a plus sign (+) in front of the number dialed to convert it to an E.164 formatted number.

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

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

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

6.11.2 Route Pattern for International Calls to Verizon

Repeat the steps in **Section 6.11.1** to add a route pattern for international calls with the following changes:

Step 1 - Enter the **change route-pattern 2** command and enter the following parameters:

- In the **Grp No** column, enter **1** for public trunk 1, and the **FRL** column enter **0** (zero).
- In the **Pfx mrk** column, leave blank (default).
- In the **No. Del Digits** column, enter **3** to have Communication Manager remove the international 011 prefix from the number.
- In the **Inserted Digits** column, enter **p** to have Communication Manager insert a plus sign (+) in front of the number dialed to convert it to an E.164 formatted number.

```
change route-pattern 2                                          Page   1 of   3
                    Pattern Number: 2      Pattern Name: 011 to E.164
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

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

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

6.11.3 Route Pattern for Service Calls to Verizon

Repeat the steps in **Section 6.11.1** to add a route pattern for x11 and other service numbers that do not require a leading plus sign:

Step 1 - Enter the **change route-pattern 4** command and enter the following parameters:

- In the **Grp No** column, enter **1** for public trunk 1, and the **FRL** column enter **0** (zero).
- In the **Pfx mrk** column, leave blank (default).
- In the **Inserted Digits** column, leave blank (default).

```
change route-pattern 4                                          Page   1 of   3
                    Pattern Number: 4      Pattern Name: Service Numbers
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

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

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

6.11.4 Route Pattern for Calls within the CPE

This form defines the Route pattern for the local SIP trunk, based on the route-pattern selected by the AAR table in **Section 6.13** (e.g., calls to Avaya SIP telephone extensions or Messaging).

Step 1 - Repeat the steps in **Section 6.11.1** with the following changes:

- In the **Grp No** column enter **3** for SIP trunk 3 (local trunk).
- In the **FRL** column enter **0** (zero).
- In the **Pfx mrk** column, leave blank (default).
- In the **Inserted Digits** column, leave blank (default).
- In the **Numbering Format** column, across from line **1:** enter **lev0-pvt**.

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3      Pattern Name: ToSM Enterprise
    SCCAN? n    Secure SIP? n     Used for SIP stations? y
    Primary SM: SM              Secondary SM:
    Grp FRL NPA Pfx Hop Toll No.  Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               QSIG
                             Dgts                                      Intw
 1: 3    0                                                              n   user
 2:                                                                     n   user
 3:                                                                     n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
1: y y y y y n  n            rest                                lev0-pvt  none
```

6.12. Automatic Route Selection (ARS) Dialing

The ARS table is selected based on the caller dialing the ARS access code (e.g., **9**) as defined in **Section 6.3**. The access code is removed and the ARS table matches the remaining outbound dialed digits and sends them to the designated route-pattern (see **Section 6.11**).

Step 1 - Enter the **change ars analysis 1720** command and enter the following:

- In the **Dialed String** column enter a matching dial pattern (e.g., **1720**). Note that the best match will route first, that is 1720555xxxx will be selected before 17xxxxxxxxx.
- In the **Min** and **Max** columns enter the corresponding digit lengths, (e.g., **11** and **11**).
- In the Route Pattern column select a route-pattern to be used for these calls (e.g., **1**).
- In the **Call Type** column enter **fnpa** (selections other than **fnpa** may be appropriate, based on the digits defined here).

Step 2 - Repeat **Step 1** for all other outbound call strings.

```
change ars analysis 1720                                        Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 1

          Dialed          Total      Route     Call   Node  ANI
          String         Min  Max   Pattern    Type   Num   Reqd
    1720                 11   11     1         fnpa         n
    18                   11   11     1         fnpa         n
    19                   11   11     1         fnpa         n
    1900                 11   11     deny      fnpa         n
    1900555              11   11     deny      fnpa         n
    1xxx976              11   11     deny      fnpa         n
    311                  3    3      4         svcl         n
    011                  10   18     2         intl         n
    411                  3    3      4         svcl         n
    5                    10   10     1         fnpa         n
    511                  3    3      4         svcl         n
    555                  7    7      deny      hnpa         n
    5551212              7    7      1         svcl         n
```

6.13. Automatic Alternate Routing (AAR) Dialing

AAR is used for outbound calls within the CPE.

Step 1 - Enter the **change aar analysis 0** command and enter the following:

- **Dialed String -** In the reference configuration all SIP telephones used extensions in the range 50xxx, therefore enter **50**.
- **Min** & **Max** – Enter **5**
- **Route Pattern** – Enter **3**
- **Call Type** – Enter **lev0**

Step 2 - Repeat **Step 1** and create an entry for Messaging access extension (not shown).

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

          Dialed            Total      Route     Call   Node   ANI
          String          Min  Max   Pattern    Type   Num    Reqd
    50                    5    5      3         lev0          n
```

6.14. Avaya G450 Media Gateway Provisioning

In the reference configuration, a G450 Media Gateway is provisioned. The G450 is located in the Main site and is used for local DSP resources, announcements, Music On Hold, etc.

Note – Only the Media Gateway provisioning associated with the G450 registration to Communication Manager is shown below. For additional information on G450 provisioning, see **[7]**.

Step 1 - Use SSH to connect to the G450 (not shown). Note that the Media Gateway prompt will contain "???" if the Media Gateway is not registered to Communication Manager (e.g., ***G450-???(super)#***).

Step 2 - Enter the **show system** command and copy down the G450 serial number (e.g., **11N507727041**).

Step 3 - Enter the **set mgc list x.x.x.x** command where x.x.x.x is the IP address of the Communication Manager Procr (e.g., **10.64.91.75**, see **Section 6.5**).

Step 4 - Enter the **copy run start** command to save the G450 configuration.

Step 5 - From Communication Manager SAT, enter **add media-gateway x** where x is an available Media Gateway identifier (e.g., **1**).

Step 6 – On the Media Gateway form (not shown), enter the following parameters:

- Set **Type** = **g450**
- Set **Name** = a descriptive name (e.g., **G450-1**)
- Set **Serial Number** = the serial number copied from **Step 2** (e.g., **11N507727041**)
- Set the **Link Encryption Type** parameter as desired (**any-ptls/tls** was used in the reference configuration)

Set **Network Region** = 1

Wait a few minutes for the G450 to register to Communication Manager. When the Media Gateway registers, the G450 SSH connection prompt will change to reflect the Media Gateway Identifier assigned in **Step 5** (e.g., ***G450-001(super)#***).

Step 7 - Enter the **display media-gateway 1** command and verify that the G450 has registered.

```
display media-gateway 1                                         Page   1 of   2
                             MEDIA GATEWAY 1

                     Type: g450
                     Name: G450-1
                Serial No: 11N507727041
    Link Encryption Type: any-ptls/tls       Enable CF? n
          Network Region: 1                   Location: 1
         Use for IP Sync? y                  Site Data:
           Recovery Rule: 1

              Registered?  y
   FW Version/HW Vintage: 40 .10 .0  /1
       MGP IPV4 Address: 10.64.91.91
       MGP IPV6 Address:
   Controller IP Address: 10.64.91.75
             MAC Address: b4:b0:17:90:61:d8

   Mutual Authentication? optional
```

6.15. Avaya Aura® Media Server Provisioning

In the reference configuration, an Avaya Aura® Media Server is provisioned. The Media Server is located in the Main site and is used, along with the G450 Media Gateway, for local DSP resources, announcements, and Music On Hold.

Note – Only the Media Server provisioning associated with Communication Manager is shown below. See **[8]** and **[9]** for additional information.

Step 1 - Access the Media Server Element Manager web interface by typing "**https://x.x.x.x:8443**" (where x.x.x.x is the IP address of the Media Server) (not shown).

Step 2 - On the Media Server Element Manager, navigate to **Home → System Configuration → Signaling Protocols → SIP →Node and Routes** and add the Communication Manager Procr interface IP address (e.g., **10.64.91.75**, see **Section 6.4**) as a trusted node (not shown).

Step 3 - On Communication Manager, enter the **add signaling-group x** command where x is an unused signaling group (e.g., **60**), and provision the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tls**
- Verify that **Peer Detection Enabled?** – Set to **n**.
- **Peer Server** to **AMS**.
- **Near-end Node Name** – Set to the node name of the **procr** noted in **Section 6.4**.
- **Far-end Node Name** – Set to the node name of Media Server as administered in **Section 6.4** (e.g., **AMS**).
- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5061**.
- **Far-end Network Region** – Set the IP network region to **1**, as set in **Section 6.6.1**.
- **Far-end Domain** – Automatically populated with the IP address of the Media Server.

```
add signaling-group 60                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 60             Group Type: sip
                        Transport Method: tls


  Peer Detection Enabled? n  Peer Server: AMS



   Near-end Node Name: procr                  Far-end Node Name: AMS
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain: 10.64.91.80
```

Step 4 - On Communication Manager, enter the **add media-server x** command where x is an available Media Server identifier (e.g., **1**). Enter the following parameters:

- Signaling **Group** – Enter the signaling group previously configured for Media Server (e.g., **60**).
- **Voip Channel License Limit** – Enter the number of VoIP channels for this Media Server (based on licensing) (e.g., **300**).
- **Dedicated Voip Channel Licenses** – Enter the number of VoIP channels licensed to this Media Server (e.g., **300**)
- Remaining fields are automatically populated based on the signaling group provisioning for the Media Server.

```
add media-server 1                                              Page   1 of   1
                                  MEDIA SERVER

                    Media Server ID: 1

                    Signaling Group: 60
         Voip Channel License Limit: 300
    Dedicated Voip Channel Licenses: 300

                          Node Name: AMS
                     Network Region: 1
                           Location: 1
          Announcement Storage Area: ANNC-be99ad1a-1f39-41e5-ba04-000c29f8f3f3
```

6.16. Save Translations

After the Communication Manager provisioning is completed, enter the command **save translation**.

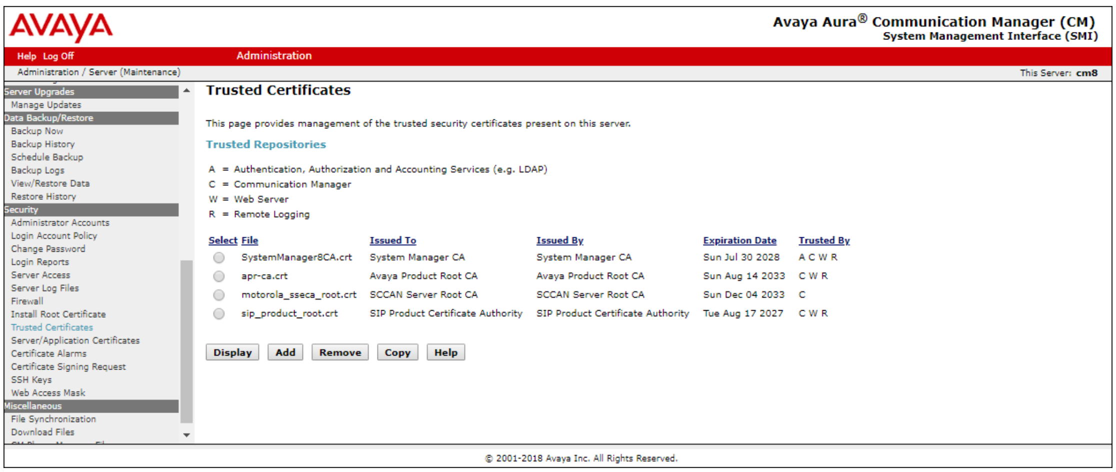
6.17. Verify TLS Certificates – Communication Manager

Note – Testing was done with System Manager signed identity certificates. The procedure to create and obtain these certificates is outside the scope of these Application Notes.

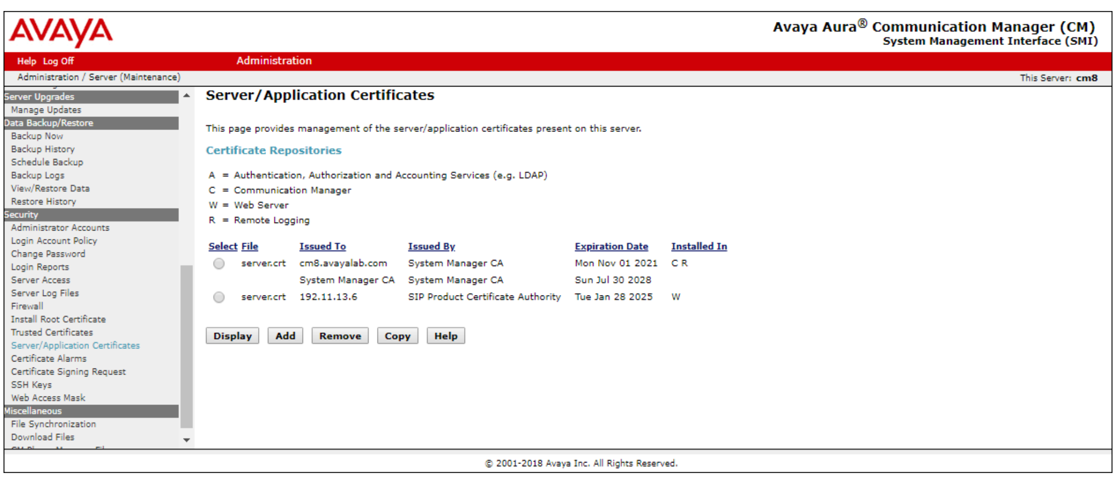
In the reference configuration, TLS transport is used for the communication between Session Manager and Communication Manager. The following procedures show how to verify the certificates used by Communication Manager.

Step 1 - **From** a web browser, type in “https://<ip-address>”, where “<ip-address>” is the IP address or FQDN of Communication Manager. Follow the prompted steps to enter appropriate **Logon ID** and **Password** credentials to log in (not shown).

Step 2 - **Click** on **Administration** at the top of the page and select **Server (Maintenance)** (not shown). Click on **Security → Trusted Certificate** and verify the System Manager CA certificate is present in the Communication Manager trusted repository.



Step 3 - Click on **Security → Server/Application Certificates** and verify the System Manager CA certificate is present in the Communication Manager certificate repository.



7. Avaya Aura® Experience Portal

These Application Notes assume that the necessary Experience Portal licenses have been installed and basic Experience Portal administration has already been performed. Consult [**13**] and [**14**] for further details if necessary.

7.1. Background

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

References to the Voice XML and/or Call Control XML applications are administered on Experience Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Experience Portal, the called party DNIS number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match is found, Experience Portal informs the caller that the call cannot be handled, and disconnects the call[2].

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

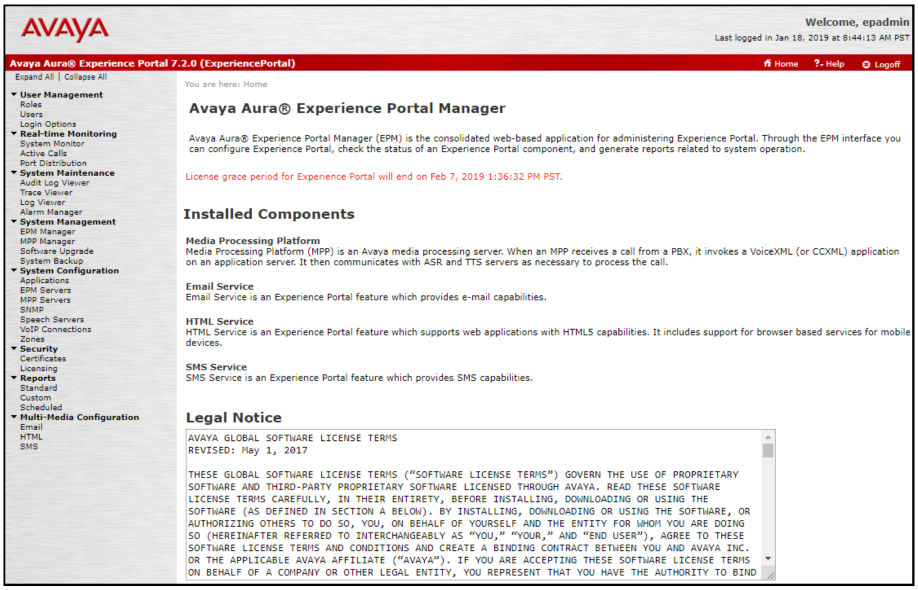
[2] An application may be configured with "inbound default" as the called number, to process all inbound calls that do not match any other application references.

7.2. Logging In and Licensing

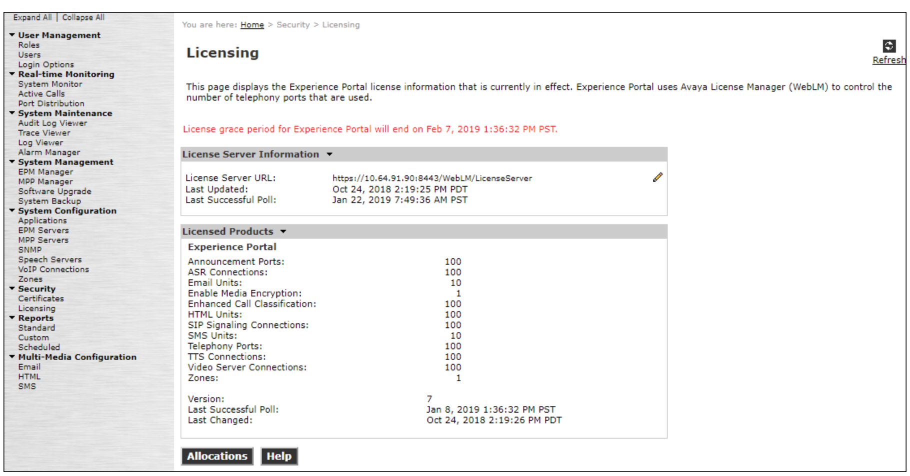
This section describes the steps on Experience Portal for administering a SIP connection to the Session Manager.

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

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



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

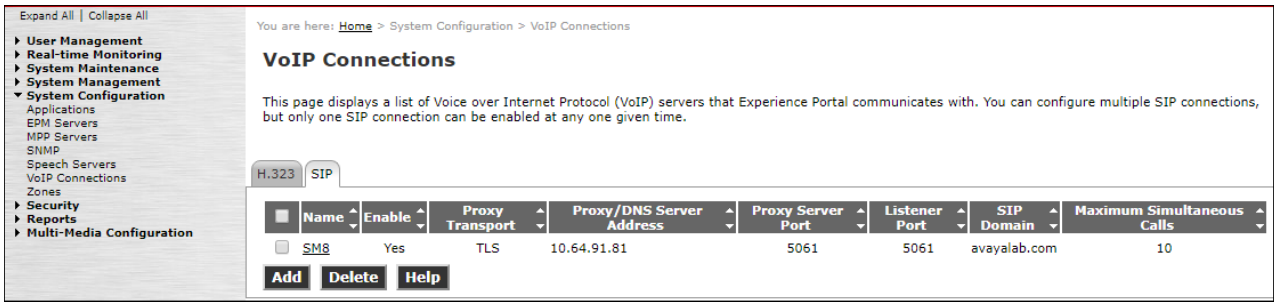


7.3. VoIP Connection

This section defines a SIP trunk between Experience Portal and Session Manager (**Section 5.5.5**).

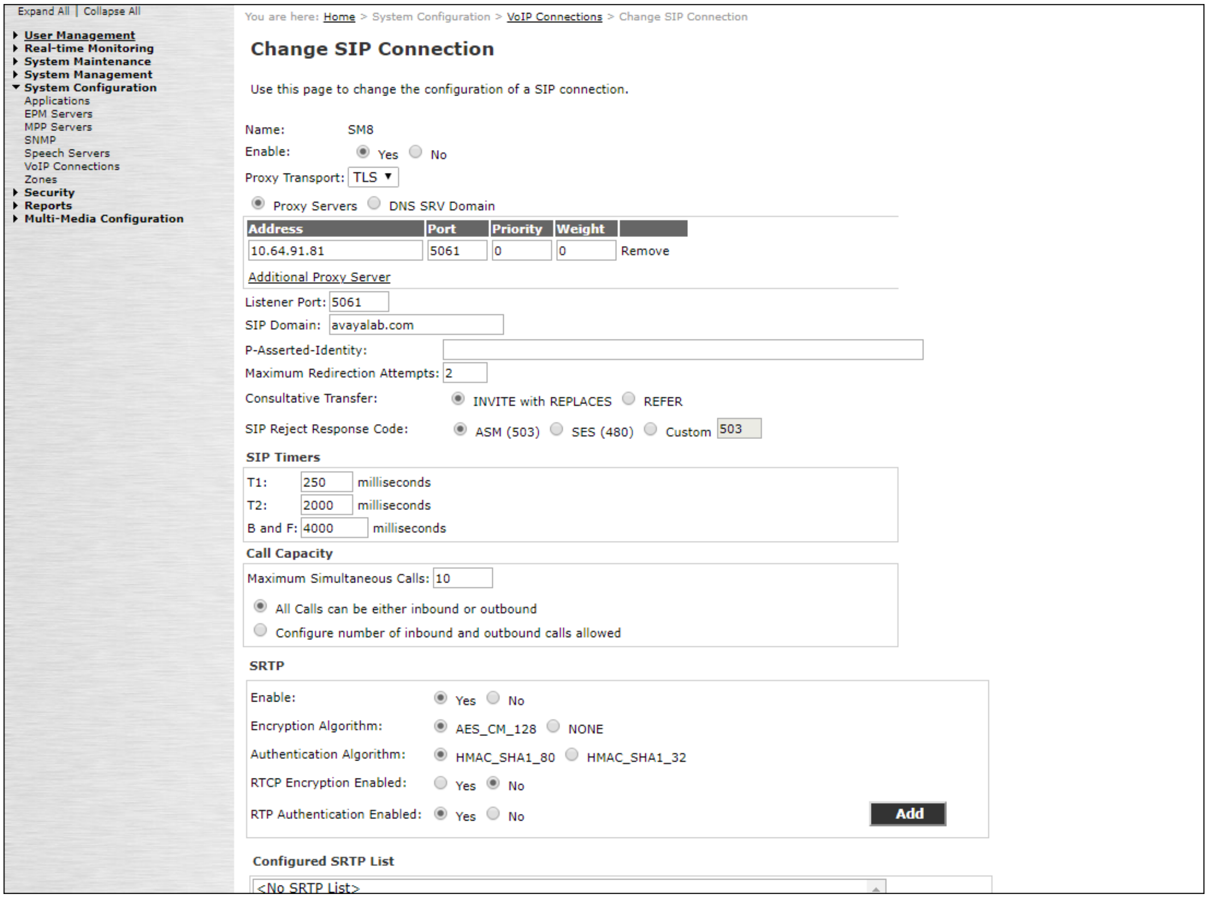
Step 1 - In the left pane, navigate to **System Configuration→VoIP Connections**. On the **VoIP Connections** page, select the **SIP** tab and click **Add** to add a SIP trunk.

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



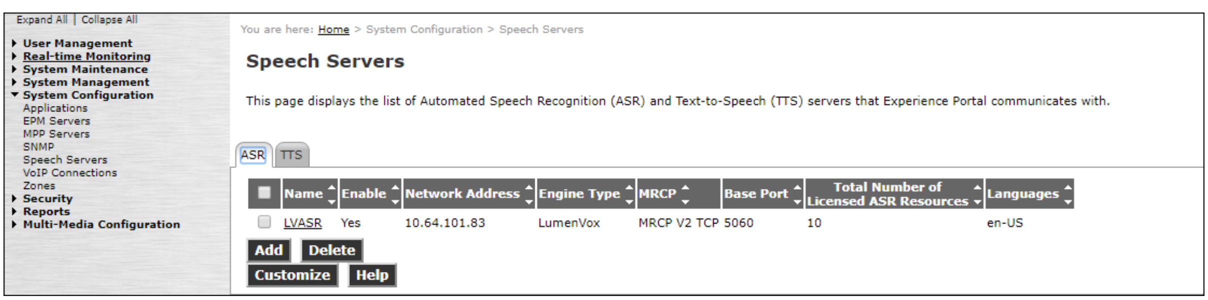
Step 2 - Configure a SIP connection as follows:

- **Name** – Set to a descriptive name (e.g., **SM8**).
- **Enable** – Set to **Yes**.
- **Proxy Server Transport** – Set to **TLS**.
- Select **Proxy Servers**, and enter:
  - **Proxy Server Address** = **10.64.91.81** (the IP address of the Session Manager signaling interface defined in **Section 5.4.1**).
  - **Port** = **5061**
  - **Priority** = **0** (default)
  - **Weight** = **0** (default)
- **Listener Port** – Set to **5061**.
- **SIP Domain** – Set to **avayalab.com** (see **Section 5.1)**.
- **Consultative Transfer** – Select **INVITE with REPLACES**.
- **SIP Reject Response Code** – Select **ASM (503)**.
- **Maximum Simultaneous Calls** – Set to a number in accordance with licensed capacity. In the reference configuration a value of **10** was used.
- Select **All Calls can be either inbound or outbound**.
- **SRTP Enable** = **Yes**
- **Encryption Algorithm** = **AES_CM_128**
- **Authentication Algorithm** = **HMAC_SHA1_80**
- **RTCP Encryption Enabled** = **No**
- **RTP Authentication Enabled** = **Yes**
- Use default values for all other fields.
- Click **Save**.



7.4. Speech Servers

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

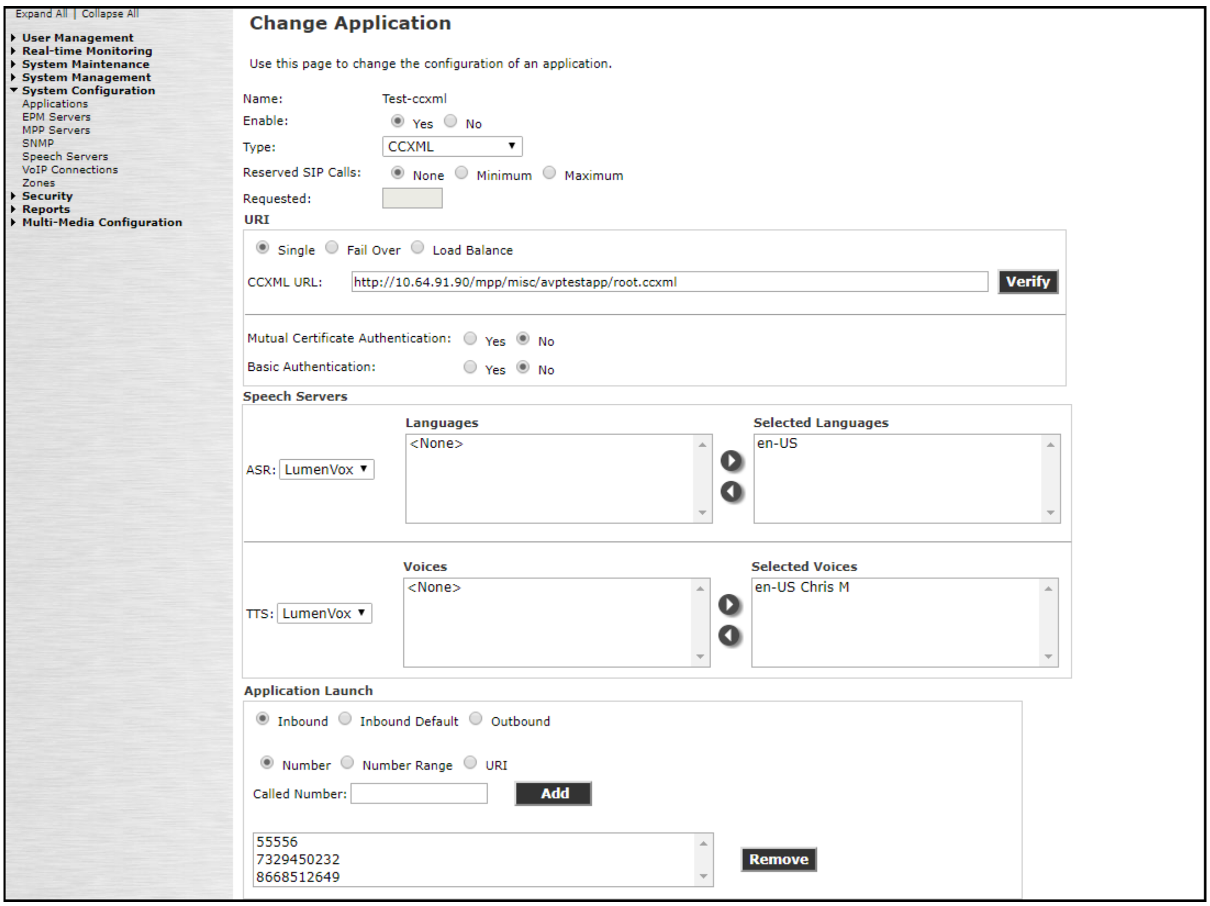


7.5. Application References

This section describes the steps for administering a reference to the VXML and/or CCXML applications residing on the application server. In the sample configuration, the applications were co-resident on one Experience Portal server, with IP Address 10.64.90.91.

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

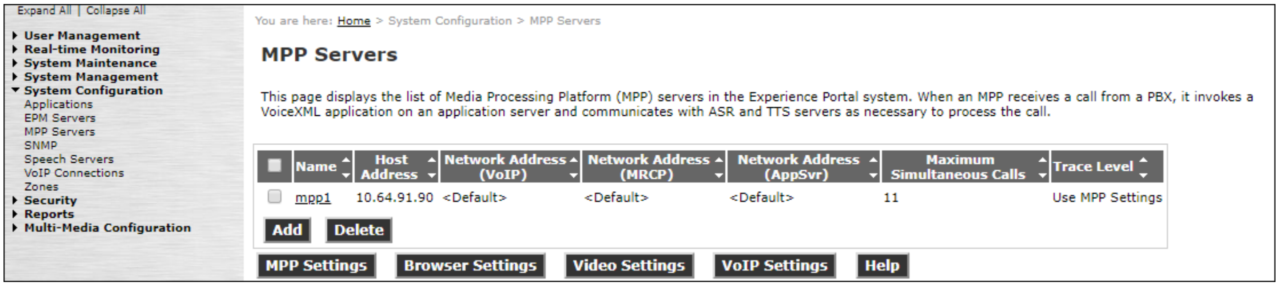
- **Name** – Set to a descriptive name (e.g., **Test-ccxml**).
- **Enable** – Set to **Yes**. This field determines which application(s) will be executed based on their defined criteria.
- **Type** – Select **VoiceXML**, **CCXML**, or **CCXML/VoiceXML** according to the application type.
- **VoiceXML** and/or **CCXML URL** – Enter the necessary URL(s) to access the VXML and/or CCXML application(s) on the application server. In the sample screen below, the Experience Portal test application on a single server is referenced.
- **Speech Servers ASR** and **TTS** – Select the appropriate ASR and/or TTS servers as necessary.
- **Application Launch** – Set to **Inbound**.
- **Called Number** – Enter the number to match against an inbound SIP INVITE message, and click **Add**. In the sample configuration illustrated in these Application Notes, the dialed Verizon IP Trunk DID number 732-945-0232 was used. Repeat to define additional called party numbers as needed. Inbound Verizon Business calls with these called party numbers will be handled by the application defined in this section.



7.6. MPP Servers and VoIP Settings

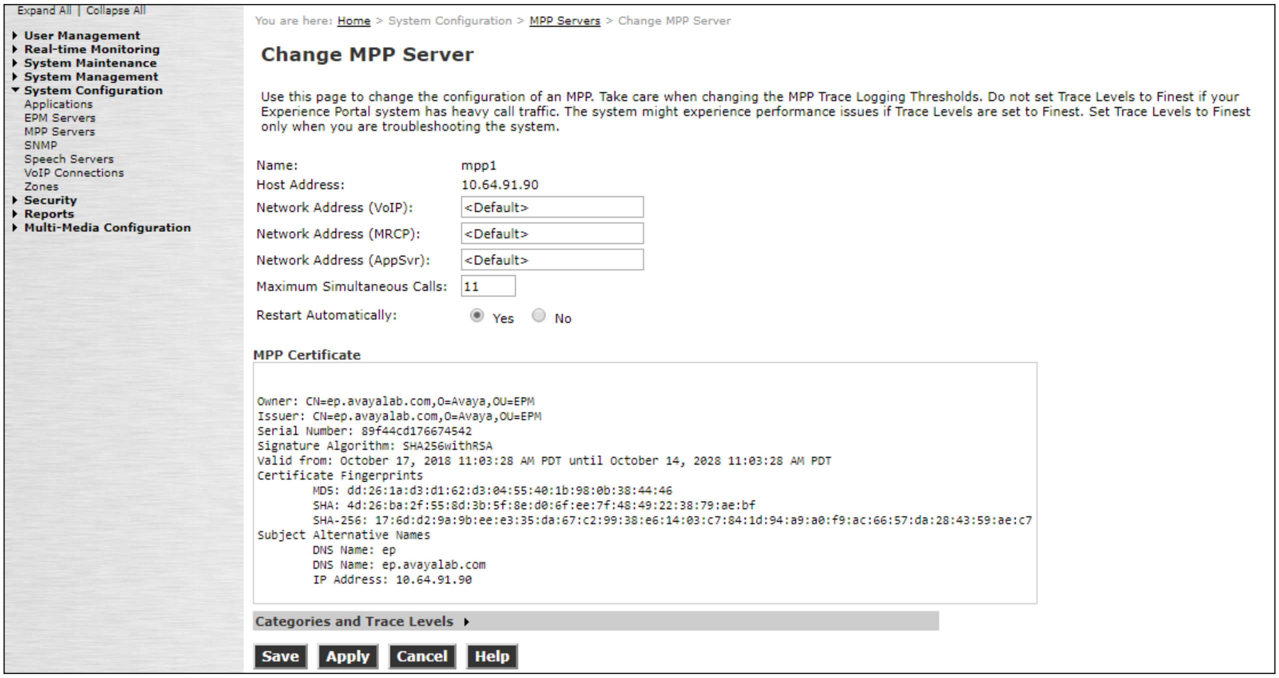
This section illustrates the procedure for viewing or changing the MPP Settings. In the sample configuration, the MPP Server is co-resident on a single server with the Experience Portal Management server (EPM).

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



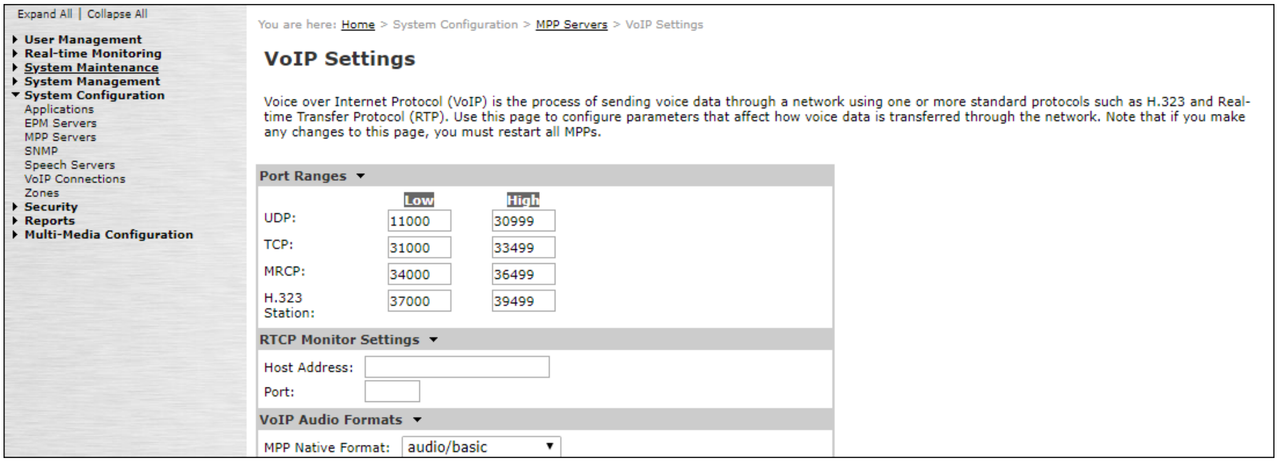
Step 2 - Enter any descriptive name in the **Name** field (e.g., **mpp1**) and the IP address of the MPP server in the **Host Address** field and click **Continue** (not shown).

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



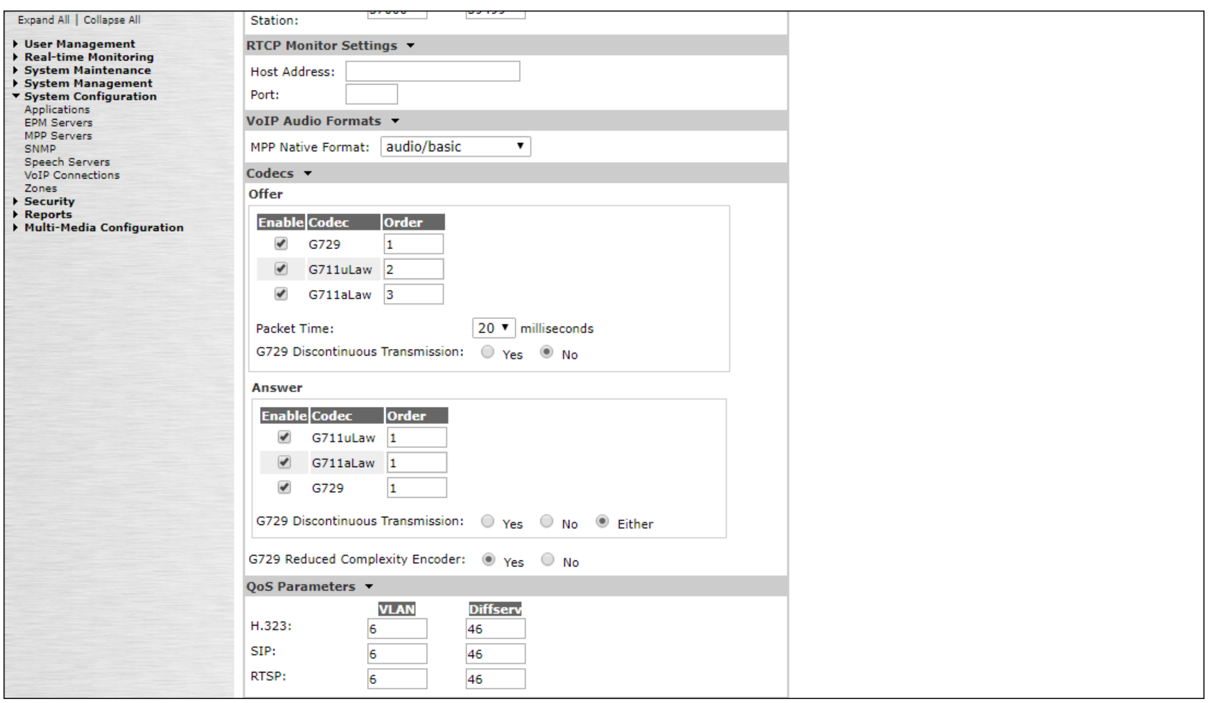
Step 4 - Click **VoIP Settings** tab on the screen displayed in **Step 1**, and the following screen is displayed.

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



- In the Codecs section set:
  - Set **Packet Time** to **20**.
  - Verify the **G729 Codec** is enabled.
  - Set **G729 Discontinuous Transmission** to **No** (G.729A).
  - Set the **Offer Order** to the preferred codec. In the sample configuration, **G729** is the first codec, followed by **G711ulaw**, then **G711aLaw**.
- Use default values for all other fields.

Step 5 - Click on **Save**.



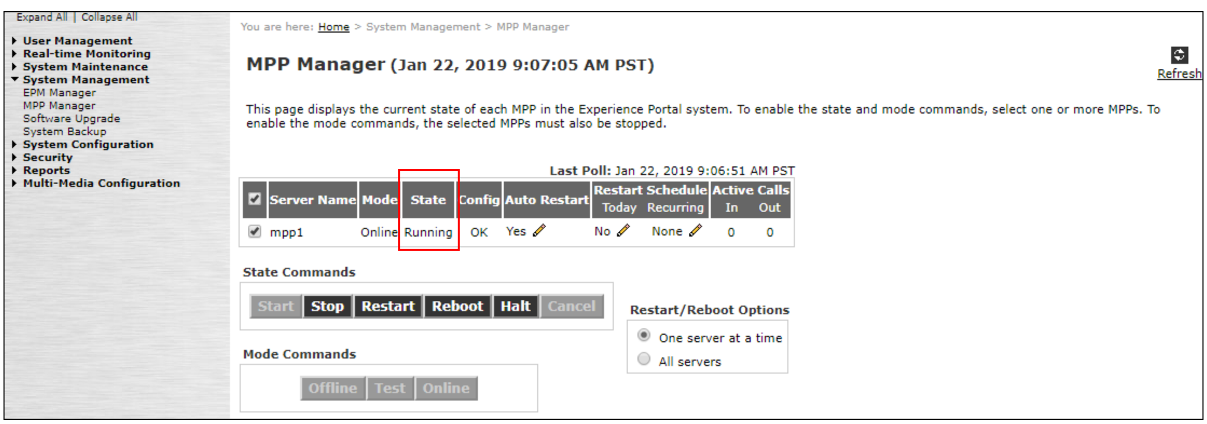
7.7. Configuring RFC2833 Event Value Offered by Experience Portal

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

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

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

After saving the file with the change, restart the MPP server for the change to take effect. As shown below, the MPP may be restarted using the **Restart** button available via the Experience Portal GUI at **System Management → MPP Manager**.
Note that the **State** column shows when the MPP is running after the restart.



8. Configure Avaya Session Border Controller for Enterprise Release 7.2

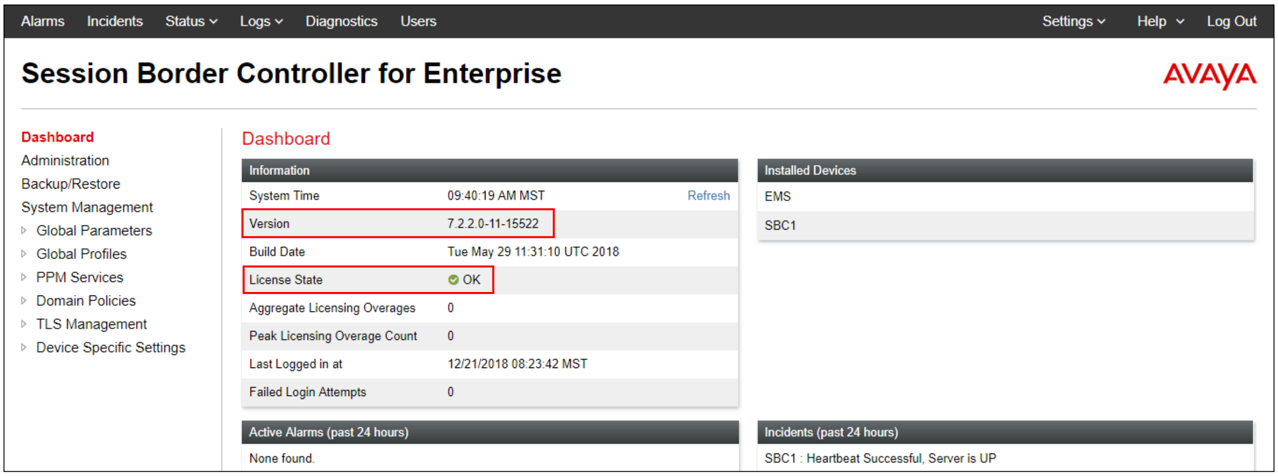
These Application Notes assume that the installation of the Avaya SBCE and the assignment of all IP addresses have already been completed, including the management IP address.

In the sample configuration, the management IP is 10.64.90.50. Access the web management interface by entering https://<ip-address> where <ip-address> is the management IP address assigned during installation. Enter the **Username** and click on **Continue**.

Enter the password and click on **Log In**.

The main page of the Avaya SBCE will appear. Note that the installed software version is displayed. Verify that the **License State** is **OK**. The SBCE will only operate for a short time without a valid license. Contact your Avaya representative to obtain a license.

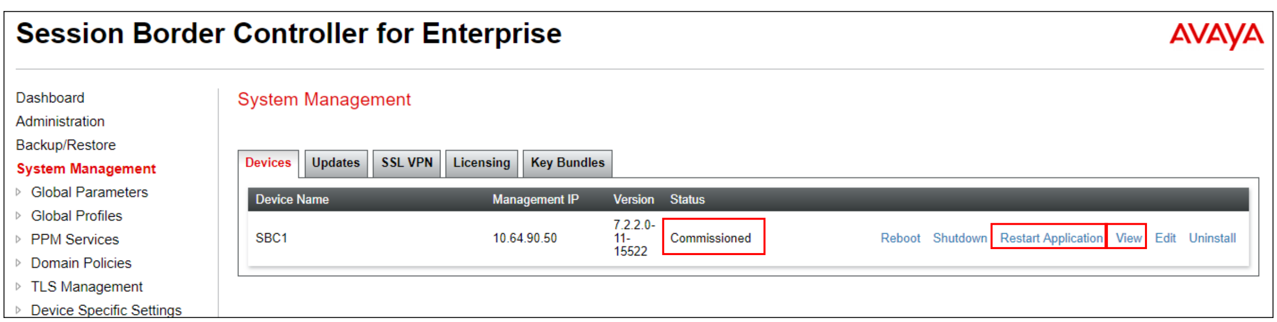
Note – The provisioning described in the following sections use the menu options listed in the left-hand column shown below.



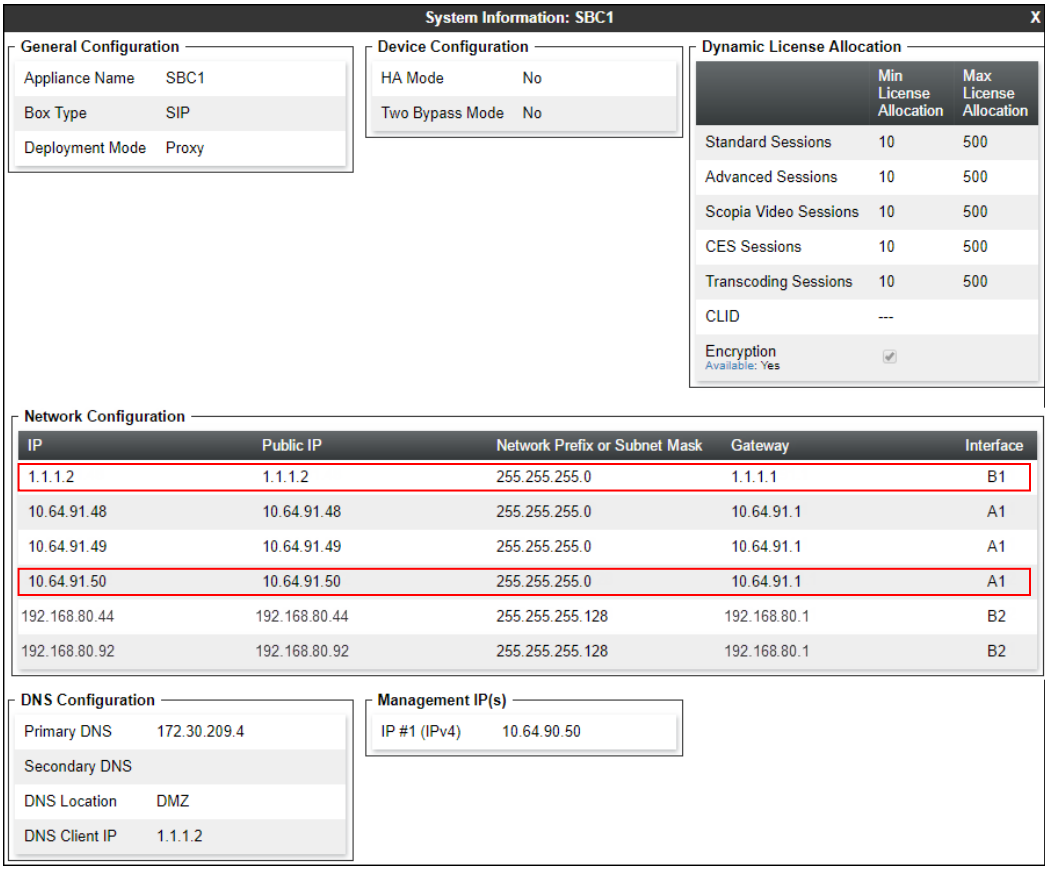
8.1. System Management – Status

Step 1 - Select **System Management** and verify that the **Status** column says **Commissioned**. If not, contact your Avaya representative.

Note – Certain Avaya SBCE configuration changes require that the underlying application be restarted. To do so, click on **Restart Application** shown below.



Step 2 - Click on **View** (shown above) to display the **System Information** screen. The following shows the relevant IP information highlighted in the shared test environment. The highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to Verizon. Other IP addresses assigned to these interfaces and interface **B2** on the screen below are used to support remote workers and are not the focus of these Application Notes. Note that the **Management IP** must be on a separate subnet from the IP interfaces designated for SIP traffic.



8.2. TLS Management

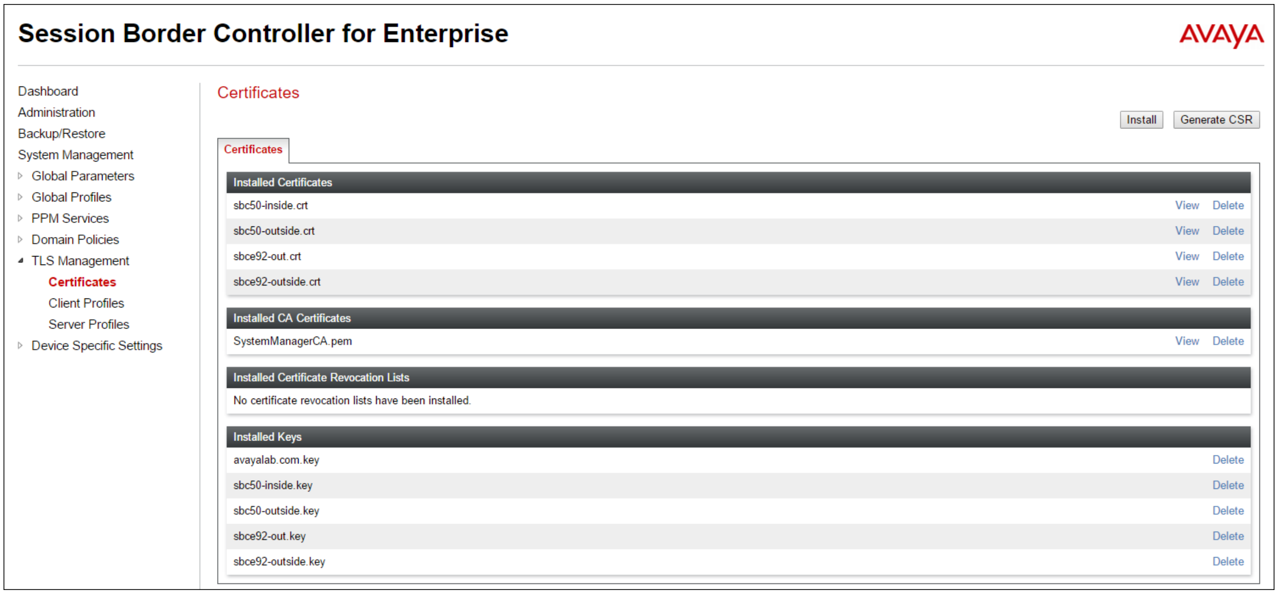
Note – Testing was done with System Manager signed identity certificates. The procedure to create and obtain these certificates is outside the scope of these Application Notes.

In the reference configuration, TLS transport is used for the communication between Session Manager and Avaya SBCE. The following procedures show how to create the client and server profiles.

8.2.1 Verify TLS Certificates – Avaya Session Border Controller for Enterprise

Step 1 - Select **TLS Management → Certificates** from the left-hand menu. Verify the following:

- System Manager CA certificate is present in the **Installed CA Certificates** area.
- System Manager CA signed identity certificate is present in the **Installed Certificates** area.
- Private key associated with the identity certificate is present in the **Installed Keys** area.



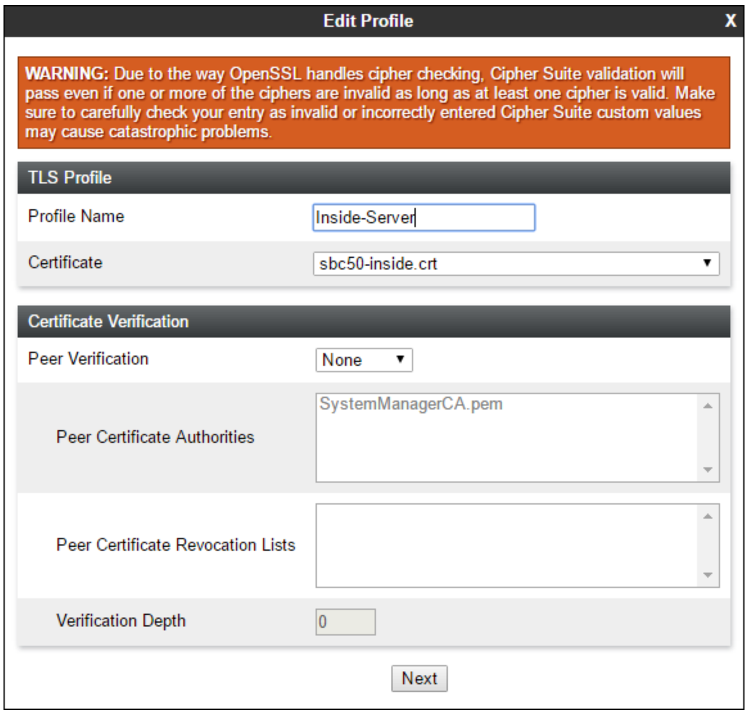
Session Border Controller for Enterprise
AVAYA
Dashboard
Administration
Backup/Restore
System Management
Global Parameters
Global Profiles
PPM Services
Domain Policies
TLS Management
Certificates
Client Profiles
Server Profiles
Device Specific Settings
Certificates
Install
Generate CSR
Certificates
Installed Certificates
sbc50-inside.crt View Delete
sbc50-outside.crt View Delete
sbce92-out.crt View Delete
sbce92-outside.crt View Delete
Installed CA Certificates
SystemManagerCA.pem View Delete
Installed Certificate Revocation Lists
No certificate revocation lists have been installed.
Installed Keys
avayalab.com.key Delete
sbc50-inside.key Delete
sbc50-outside.key Delete
sbce92-out.key Delete
sbce92-outside.key Delete

8.2.2 Server Profiles

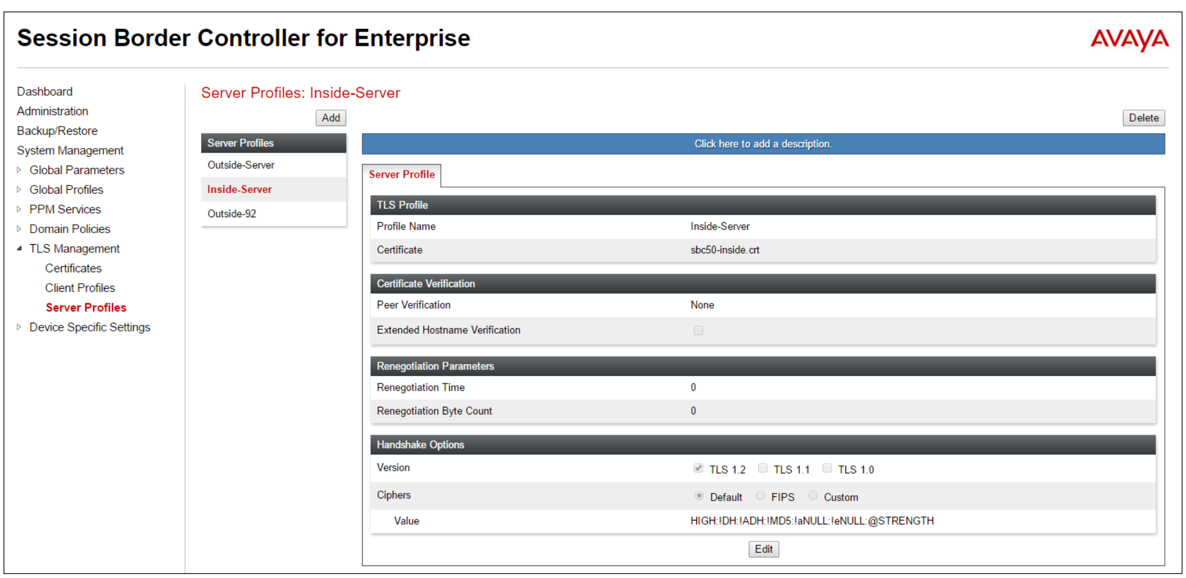
Step 1 - Select **TLS Management → Server Profiles** and click on **Add**. Enter the following:

- **Profile Name:** enter descriptive name.
- **Certificate:** select the identity certificate, e.g., **Inside-Server**, from pull down menu.
- **Peer Verification** = **None**.
- Click **Next**.

Step 2 - Accept default values for the next screen (not shown) and click **Finish**.



The following screen shows the completed TLS **Server Profile** form:

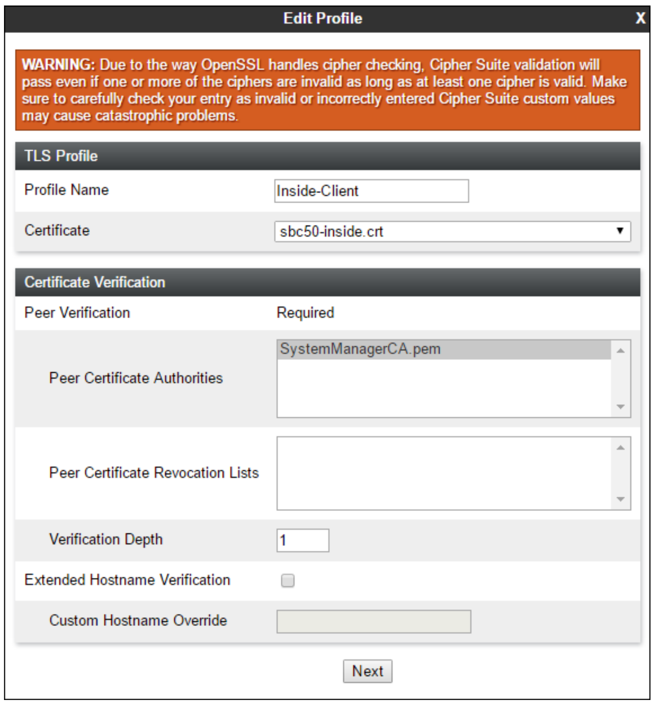


8.2.3 Client Profiles

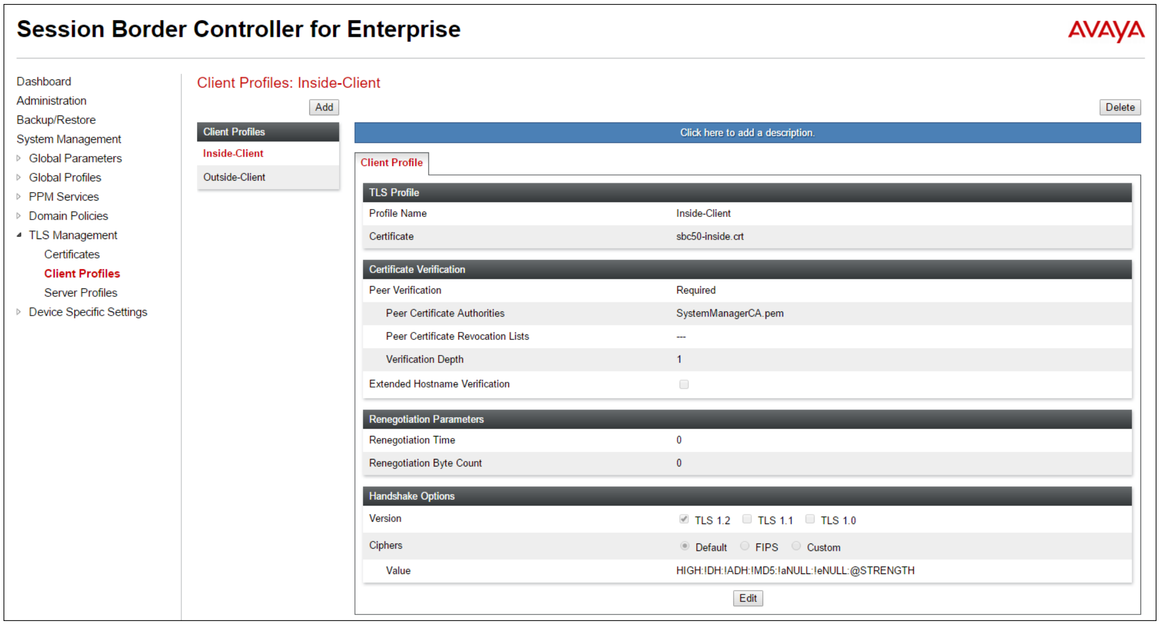
Step 1 - Select **TLS Management → Server Profiles** and click on **Add**. Enter the following:

- **Profile Name:** enter descriptive name.
- **Certificate:** select the identity certificate, e.g., **Inside-Client**, from pull down menu.
- **Peer Verification** = **Required**.
- **Peer Certificate Authorities:** select the CA certificate used to verify the certificate received from Session Manager, e.g., **SystemManagerCA.pem**.
- **Verification Depth:** enter **1**.
- Click **Next**.

Step 2 - Accept default values for the next screen (not shown) and click **Finish**.



The following screen shows the completed TLS **Client Profile** form:



8.3. Global Profiles

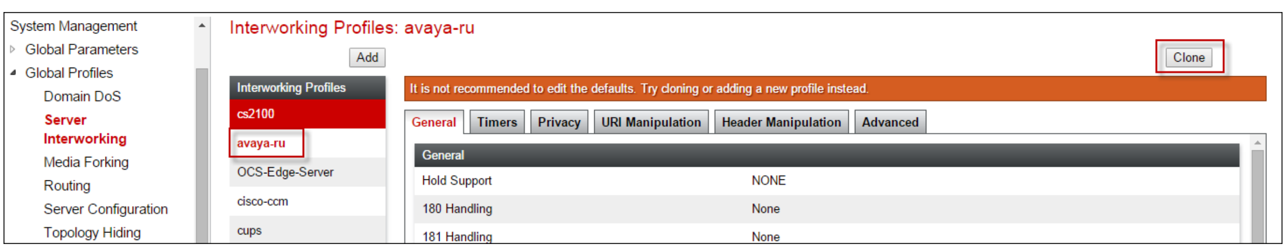
Global Profiles allow for configuration of parameters across the Avaya SBCE appliances.

8.3.1 Server Interworking – Avaya

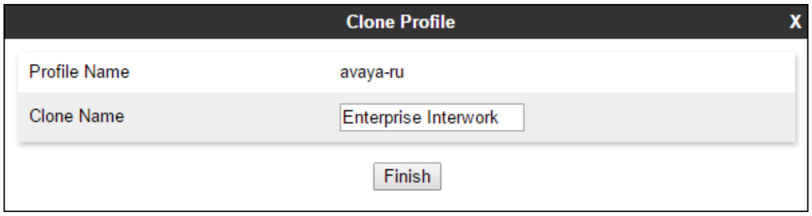
Server Interworking allows users to configure and manage various SIP call server-specific capabilities such as call hold and T.38 faxing. This section defines the connection to Session Manager.

Step 1 - Select **Global Profiles → Server Interworking** from the left-hand menu.

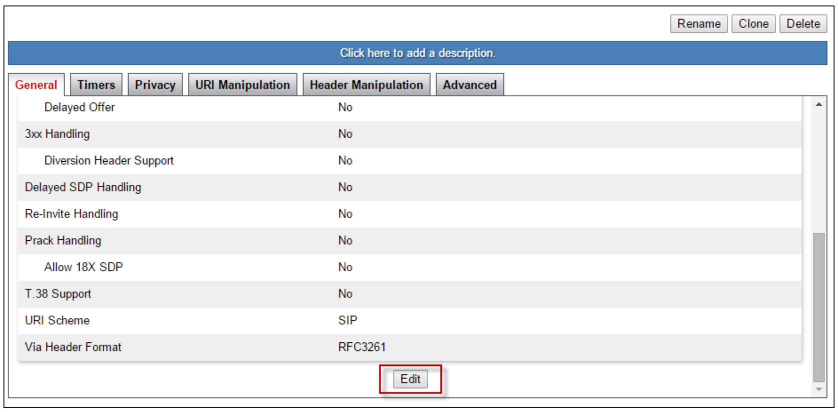
Step 2 - Select the pre-defined **avaya-ru** profile and click the **Clone** button.



Step 3 - Enter profile name: (e.g., **Enterprise Interwork**), and click **Finish**.

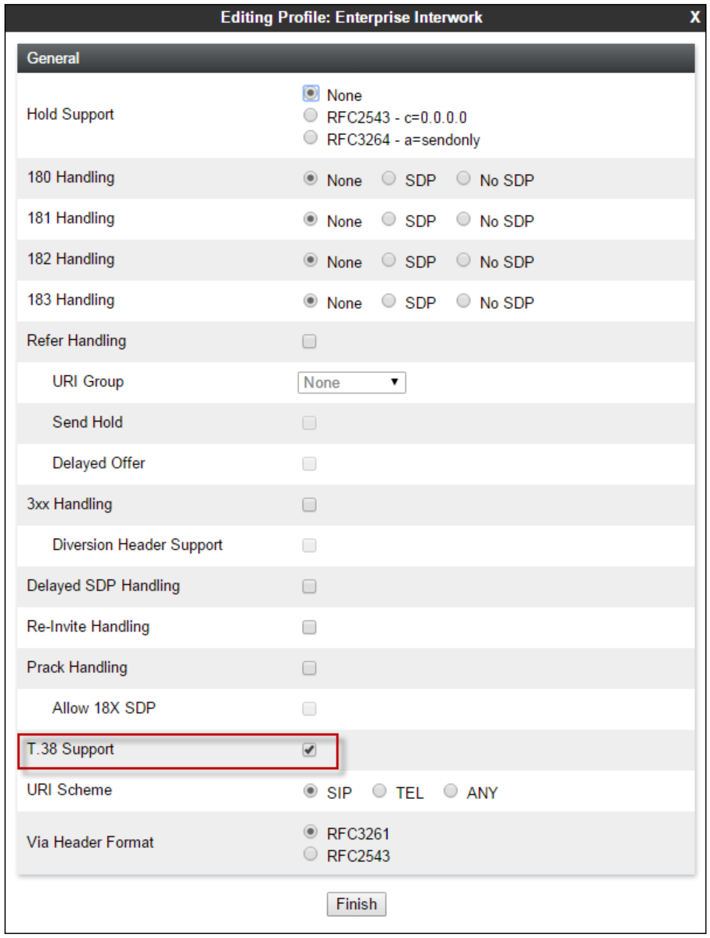


Step 4 - The new Enterprise Interwork profile will be listed. Select it, scroll to the bottom of the Profile screen, and click on **Edit**.

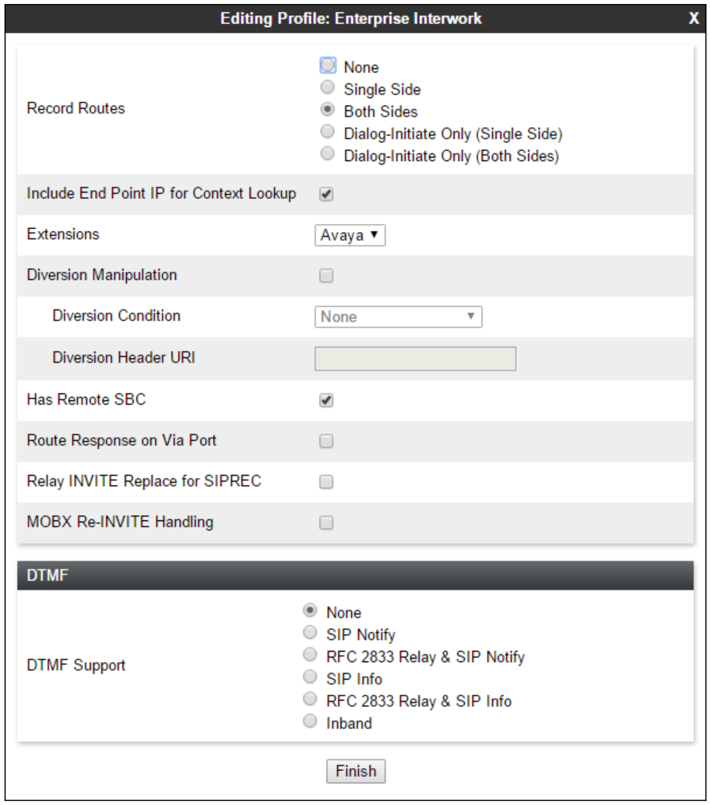


Step 5 - The **General** screen will open.

- Check **T38 Support**.
- All other options can be left with default values.
- Click **Finish**.



Step 6 - Returning to the Interworking Profile screen, select the **Advanced** tab, accept the default values, and click **Finish**.



8.3.2 Server Interworking – Verizon

Repeat the steps shown in **Section 8.3.1** to add an Interworking Profile for the connection to Verizon via the public network, with the following changes:

Note – See **Section 13** for additional steps necessary for Experience Portal to redirect calls to Communication Manager using SIP REFER.

Step 1 - Select **Add Profile** (not shown) and enter a profile name: (e.g., **SIP Provider Interwk**) and click **Next** (not shown).

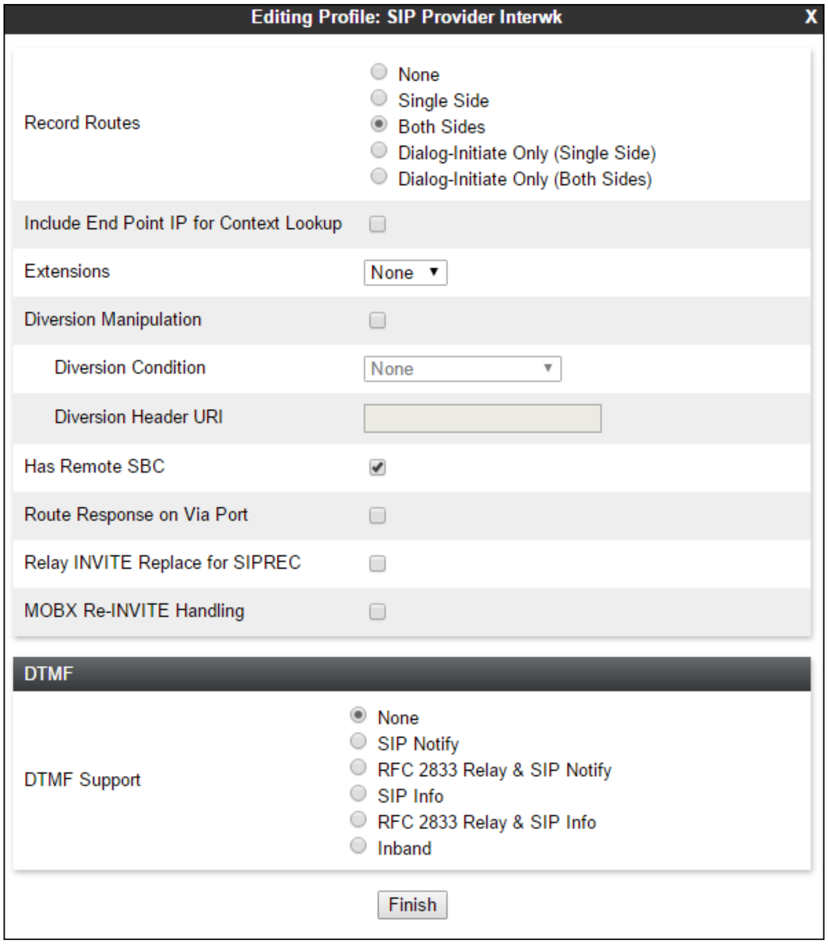
Step 2 - The **General** screen will open (not shown):

- Check **T38 Support**.
- All other options can be left as default.
- Click **Next**.

Step 3 - The **SIP Timers** and **Privacy** screens will open (not shown), accept default values for these screens by clicking **Next**.

Step 4 - The **Advanced/DTMF** screen will open:

- In the **Record Routes** field, check **Both Sides**.
- All other options can be left as default.
- Click **Finish**.



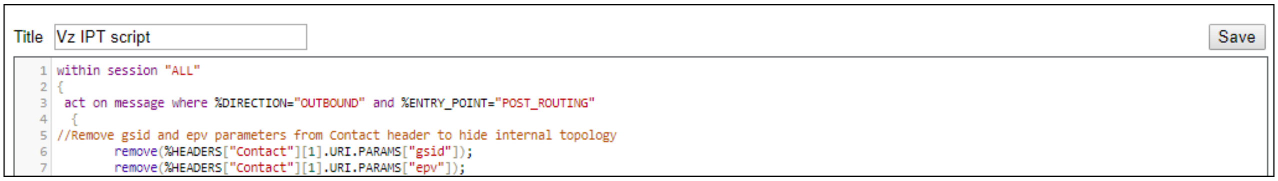
8.3.3 Signaling Manipulation

Signaling Manipulations are SigMa scripts the Avaya SBCE can use to manipulate SIP headers/messages. In the reference configuration, one signaling manipulation script is used.

Note – Use of the Signaling Manipulation scripts require higher processing requirements on the Avaya SBCE. Therefore, this method of header manipulation should only be used in cases where the use of Signaling Rules or Interworking Profiles does not meet the desired result. Refer to **[10]** for information on the Avaya SBCE scripting language.

Step 1 - As described in **Section 2.4**, Avaya SIP endpoints may send requests with Endpoint-View headers containing private network information. These are removed by Session Manager, as shown in **Section 5.3.2**. However, an "epv" parameter is also inserted into the Contact header of these requests. This parameter also contains private network information. The following signaling manipulation is used to remove this "epv" parameter from the Contact header, along with the "gsid" parameter. The "gsid" parameter was removed to further reduce packet size.

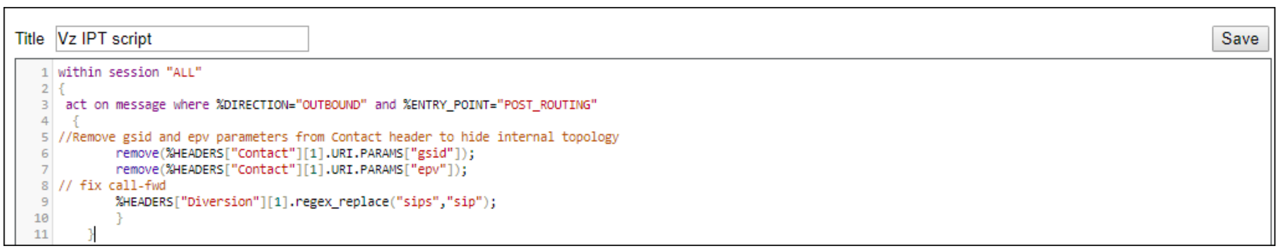
- Select **Global Profiles** from the menu on the left-hand side.
- Select **Signaling Manipulation**.
- Click **Add Script** (not shown) and the script editor window will open.
- Enter a name for the script in the **Title** box (e.g., **Vz IPT script**). The following script is defined:



```
Title Vz IPT script                                   Save
within session "ALL"
{
 act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
  {
//Remove gsid and epv parameters from Contact header to hide internal topology
        remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
        remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);
```

Step 2 - As described in **Section 2.2**, **Item 3**, the Diversion header includes the SIPS URI scheme toward Verizon. The following signaling manipulation script is added to the script defined in **Step 1** above, to convert "sips" to "sip".

- The following script is added:



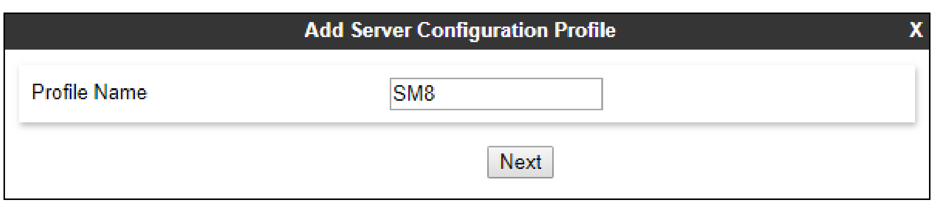
```
Title Vz IPT script                                   Save
within session "ALL"
{
 act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
  {
//Remove gsid and epv parameters from Contact header to hide internal topology
        remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
        remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);
// fix call-fwd
        %HEADERS["Diversion"][1].regex_replace("sips","sip");
        }
    }
```

Step 3 - Click on **Save**. The script editor will test for any errors, and the window will close. This script is applied to the Verizon Server Configuration in **Section 8.3.5**, **Step 3**.

8.3.4 Server Configuration – Session Manager

This section defines the Server Configuration for the Avaya SBCE connection to Session Manager.

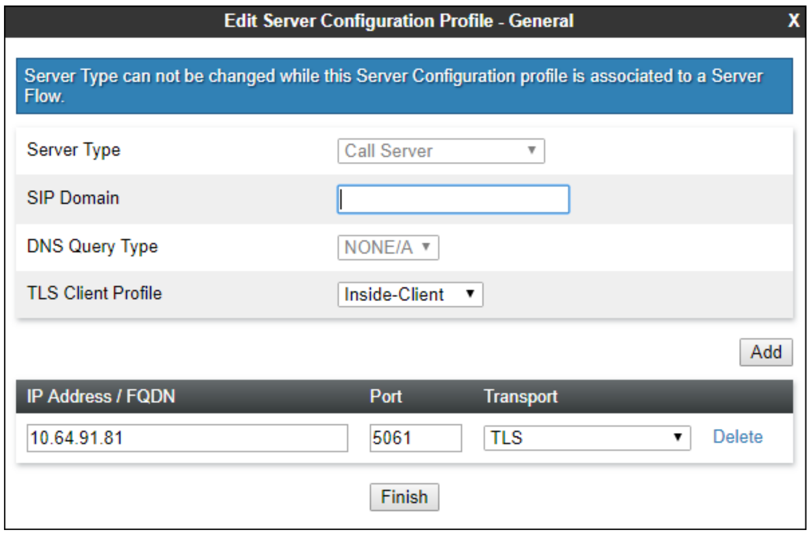
Step 1 - Select **Global Profiles → Server Configuration** from the left-hand menu.
Step 2 - Select **Add Profile** and the **Profile Name** window will open. Enter a Profile Name (e.g., **SM8**) and click **Next**.



Step 3 - The **Add Server Configuration Profile** window will open.

- Select **Server Type**: **Call Server**
- **SIP Domain**: Leave blank (default)
- **DNS Query Type**: Select **NONE/A** (default)

- **TLS Client Profile**: Select the profile create in **Section 8.2.3** (e.g., **Inside-Client**)
- **IP Address**: **10.64.91.81** (Session Manager network IP address)
- **Transport**: Select **TLS**
- **Port**: **5061**
- Select **Next** (not shown)



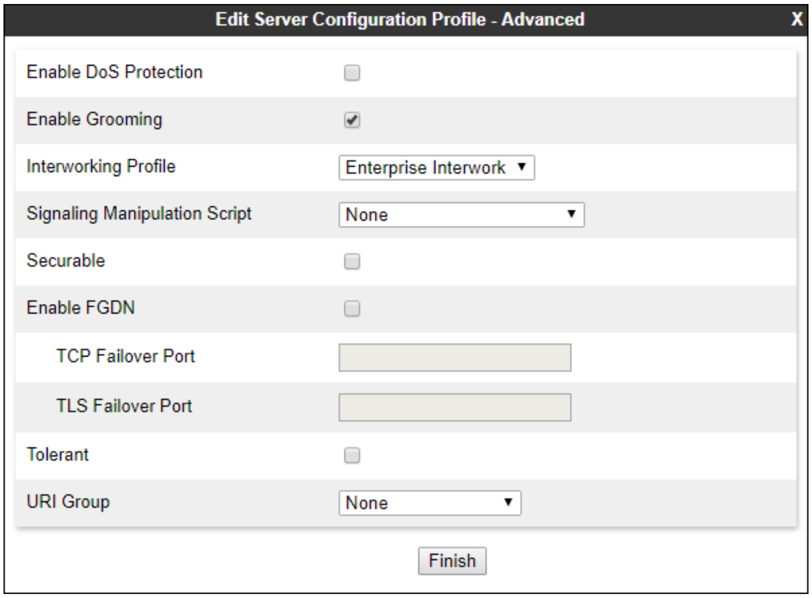
Step 4 - The **Authentication**, **Heartbeat**, **Registration** and **Ping** windows will open (not shown).

- Select **Next** to accept default values

Step 5 - The **Advanced** window will open.

- Select **Enterprise Interwork** (created in **Section 8.3.1**), for **Interworking Profile**
- Check **Enable Grooming**
- In the **Signaling Manipulation Script** field select **none**
- Select **Finish**

Note – Since TLS transport is specified in **Step 3**, then the **Enable Grooming** option should be enabled.



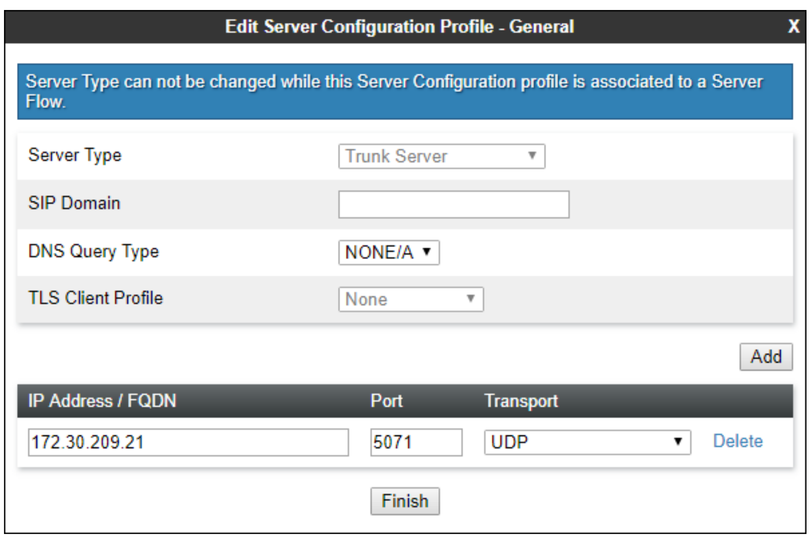
8.3.5 Server Configuration – Verizon

Repeat the steps in **Section 8.3.4**, with the following changes, to create a Server Configuration for the Avaya SBCE connection to Verizon.

Step 1 - Select **Add** and enter a Profile Name (e.g., **Verizon IPT**) and select **Next** (not shown).

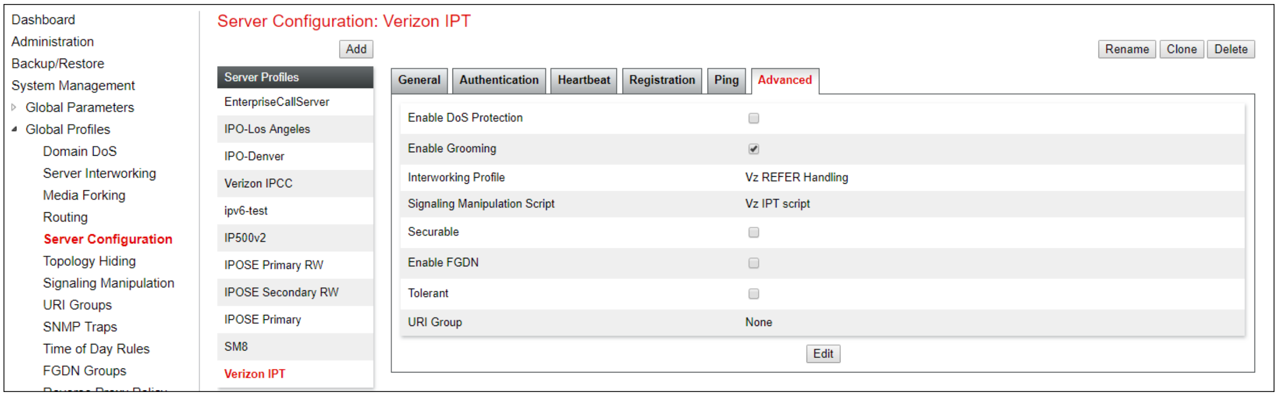
Step 2 - On the **General** window, enter the following:

- **Server Type:** Select **Trunk Server**
- **IP Address: 172.30.209.21** (Verizon-provided IP address)
- **Transport**: Select **UDP**
- **Port: 5071**
- Select **Next** (not shown) until the Advanced tab is reached



Step 3 - On the **Advanced** window, enter the following:

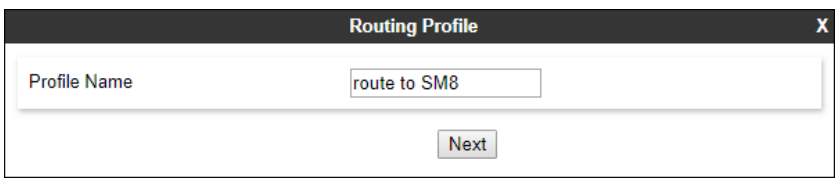
- Select **SIP Provider Interwk** (created in **Section 8.3.2**), for **Interworking Profile**.
- Select **Vz IPT script** (created in **Section 8.3.3**) for **Signaling Manipulation Script**.
- Select **Finish** (not shown)



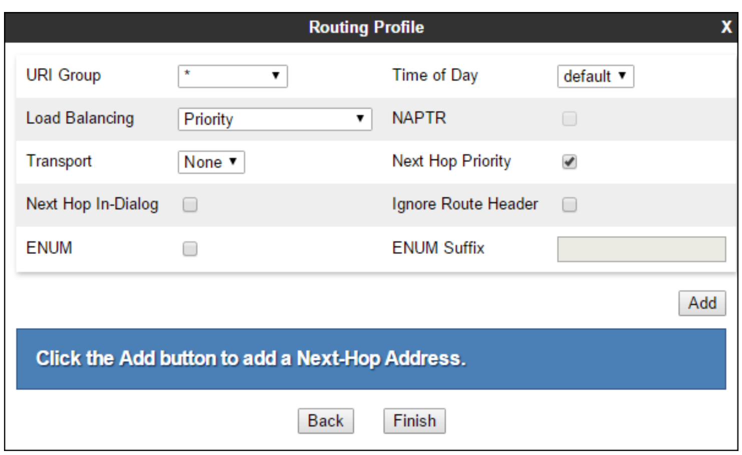
8.3.6 Routing – To Session Manager

This provisioning defines the Routing Profile for the connection to Session Manager.

Step 1 - Select **Global Profiles → Routing** from the left-hand menu, and select **Add** (not shown)
Step 2 - Enter a **Profile Name**: (e.g., **route to SM8**) and click **Next**.



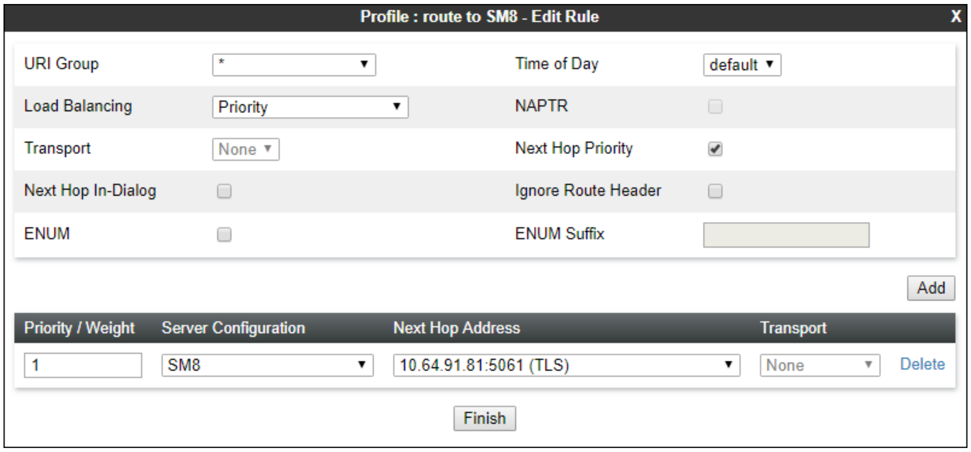
Step 3 - The Routing Profile window will open. Using the default values shown, click on **Add**.



Step 4 - The **Next-Hop Address** window will open. Populate the following fields:

- **Priority/Weight** = **1**
- **Server Configuration** = **SM8** (from **Section 8.3.4**).

- **Next Hop Address:** Verify that the **10.64.91.81:5061 (TLS)** entry from the drop-down menu is selected (Session Manager IP address). Also note that the **Transport** field is grayed out.
- Click on **Finish**.

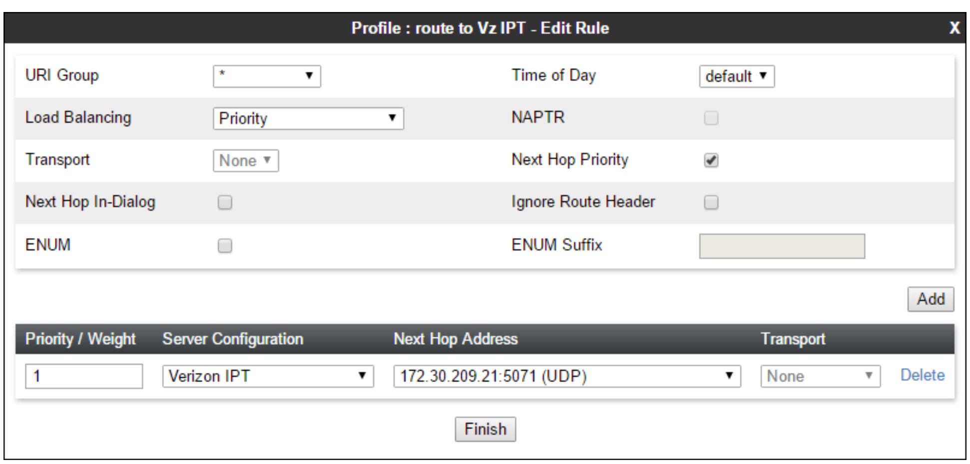


8.3.7 Routing – To Verizon

Repeat the steps in **Section 8.3.6**, with the following changes, to add a Routing Profile for the Avaya SBCE connection to Verizon.

Step 1 - On the **Global Profiles → Routing Profile** window, enter a Profile Name: (e.g., **route to Vz IPT**).
Step 2 - On the **Next-Hop Address** window, populate the following fields:
- **Priority/Weight** = **1**
- **Server Configuration** = **Verizon IPT** (**from Section 8.3.5**).
- **Next Hop Address:** select **172.30.209.21:5071 (UDP)**.

Step 3 - Click **Finish**.



8.3.8 Topology Hiding – Enterprise Side

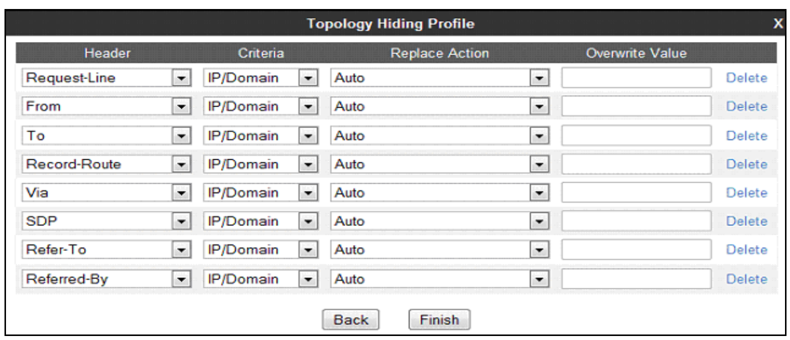
The **Topology Hiding** screen allows users to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the security of the network. It hides the topology of the enterprise network from external networks.

Step 1 - Select **Global Profiles → Topology Hiding** from the left-hand side menu.

Step 2 - Select the **Add** button, enter Profile Name: (e.g., **Enterprise-Topology**), and click **Next**.

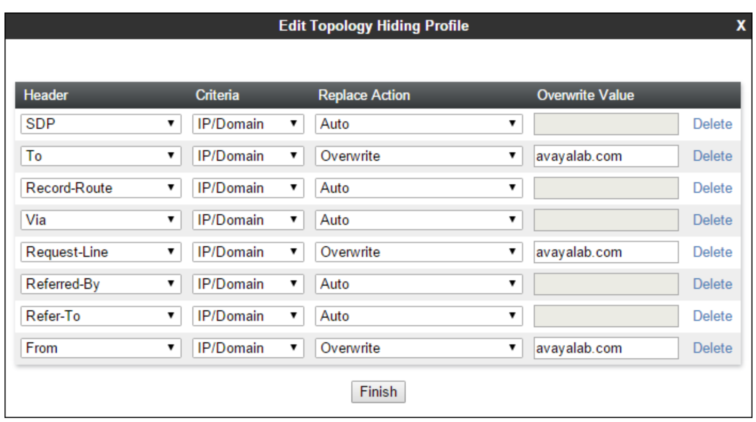
Step 3 - The **Topology Hiding Profile** window will open. Click on the **Add Header** button repeatedly until no new headers are added to the list, and the **Add Header** button is no longer displayed.

Header	Criteria	Replace Action	Overwrite Value	
Request-Line	IP/Domain	Auto		Delete



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

Step 4 - Populate the fields as shown below and click **Finish**. Note that **avayalab.com** is the domain used by the CPE (see **Sections 5.1**, **6.7**, and **6.8**).

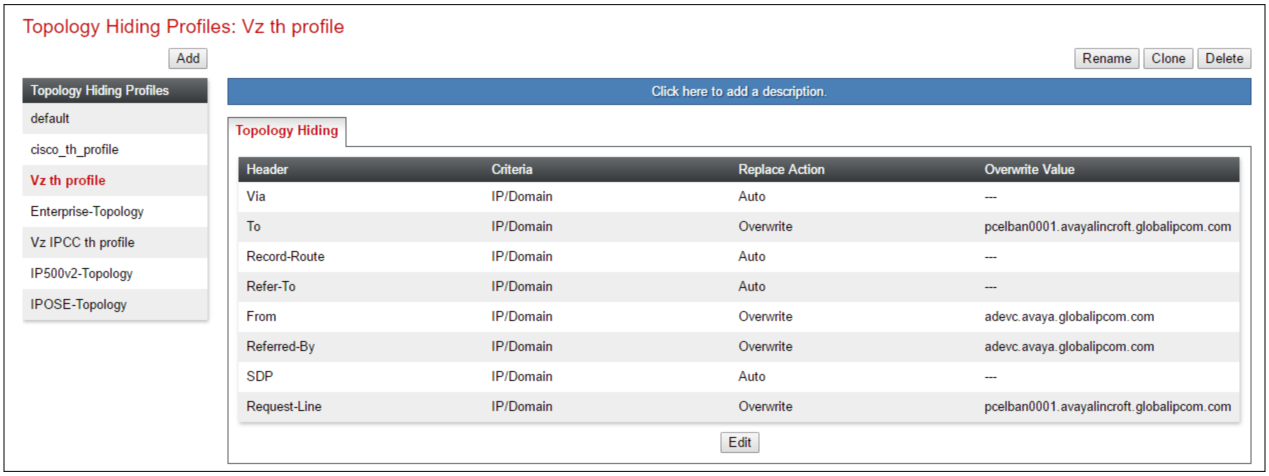


Header	Criteria	Replace Action	Overwrite Value	
SDP	IP/Domain	Auto		Delete
To	IP/Domain	Overwrite	avayalab.com	Delete
Record-Route	IP/Domain	Auto		Delete
Via	IP/Domain	Auto		Delete
Request-Line	IP/Domain	Overwrite	avayalab.com	Delete
Referred-By	IP/Domain	Auto		Delete
Refer-To	IP/Domain	Auto		Delete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

8.3.9 Topology Hiding – Verizon Side

Repeat the steps in **Section 8.3.8**, with the following changes, to create a Topology Hiding Profile for the Avaya SBCE connection to Verizon.

- Enter a Profile Name (e.g., **Vz th profile**).
- Overwrite the headers as shown below with the FQDNs known by Verizon.



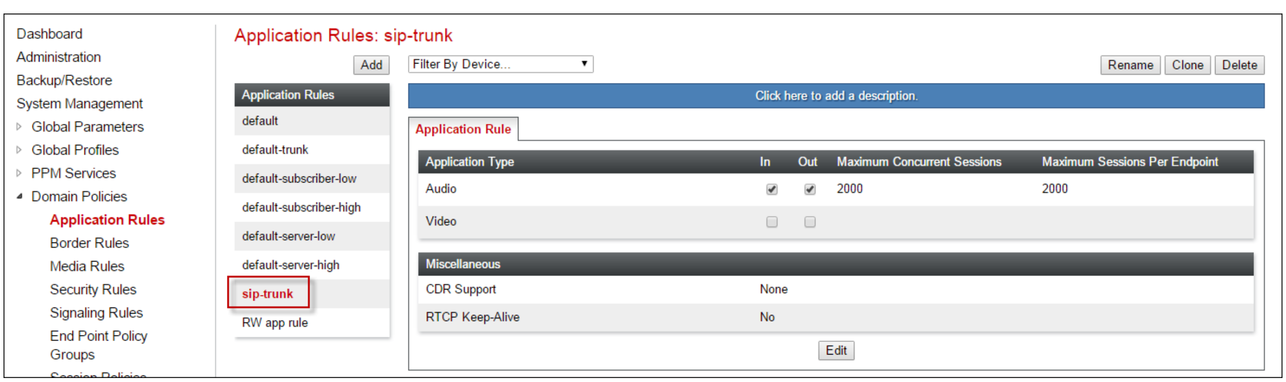
8.4. Domain Policies

The Domain Policies feature allows users to configure, apply, and manage various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating in the enterprise.

8.4.1 Application Rules

Step 1 - Select **Domain Policies →Application Rules** from the left-hand side menu (not shown).
Step 2 - Select the **default-trunk** rule (not shown).
Step 3 - Select the **Clone** button (not shown), and the **Clone Rule** window will open (not shown).

- In the **Clone Name** field enter **sip-trunk**
- Click **Finish** (not shown). The completed **Application Rule** is shown below.



8.4.2 Media Rules

Media Rules are used to define QoS parameters. Separate media rules are created for Verizon and Session Manager.

8.4.2.1 Enterprise – Media Rule

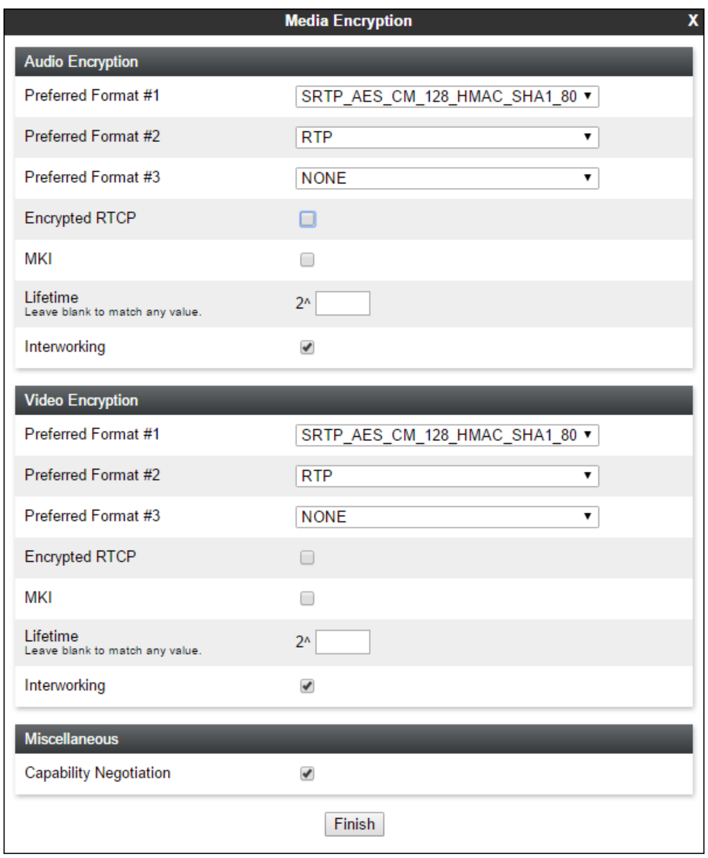
Step 1 - Select **Domain Policies → Media Rules** from the left-hand side menu (not shown).
Step 2 - From the Media Rules menu, select the **avaya-low-med-enc** rule.
Step 3 - Select **Clone** button (not shown), and the **Clone Rule** window will open.

- In the **Clone Name** field enter **enterprise med rule**
- Click **Finish.** The newly created rule will be displayed.

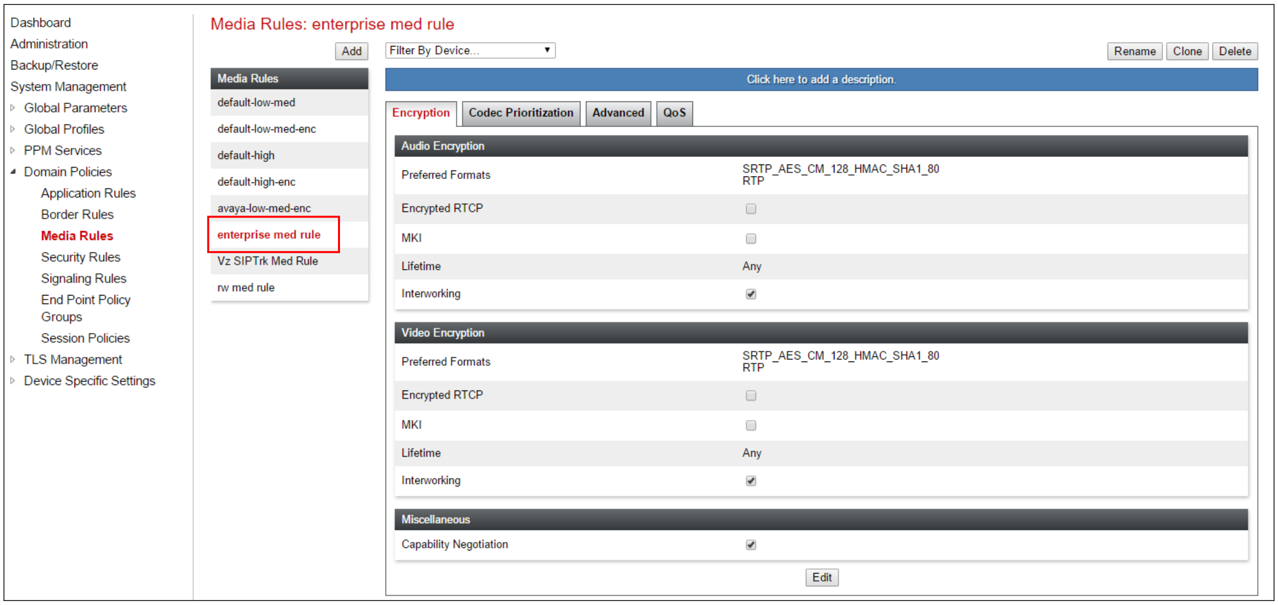
Step 4 - Highlight the **enterprise med rule** just created (not shown):

- Select the **Encryption** tab (not shown).
- Click the **Edit** button and the **Media Encryption** window will open.
- In the **Audio Encryption** section, select **RTP** for **Preferred Format #2**.
- In the **Video Encryption** section, select **RTP** for **Preferred Format #2**.
- In the **Miscellaneous** section, select **Capability Negotiation**.

Step 5 - Click **Finish**.



The completed **enterprise med rule** screen is shown below.

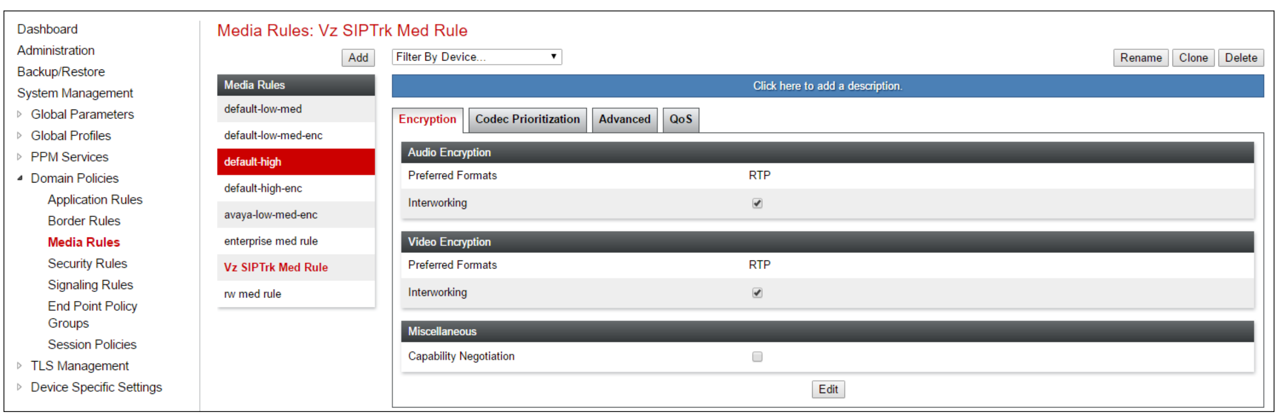


8.4.2.2 Verizon – Media Rule

Repeat the steps in **Section 8.4.2.1**, with the following changes, to create a Media Rule for Verizon.

1. Clone the **default-low-med** profile
2. In the **Clone Name** field enter **Vz SIPTrk Med Rule**

The completed **Vz SIPTrk Med Rule** screen is shown below.



8.4.3 Signaling Rules

In the reference configuration, Signaling Rules are used to define QoS parameters.

8.4.3.1 Enterprise – Signaling Rules

Step 1 - Select **Domain Policies → Signaling Rules** from the left-hand side menu (not shown).

Step 2 - **The Signaling Rules** window will open (not shown). From the Signaling Rules menu, select the **default** rule.

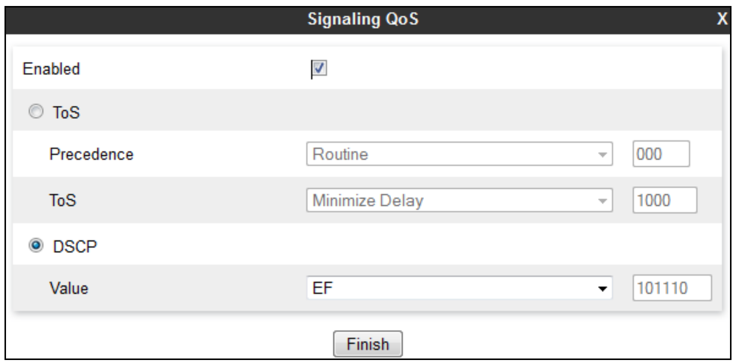
Step 3 - Select the **Clone** button and the **Clone Rule** window will open (not shown).

- In the **Rule Name** field enter **enterprise sig rule**
- Click **Finish**. The newly created rule will be displayed (not shown).

Step 4 - Highlight the **enterprise sig rule**, select the **Signaling QoS** tab and enter the following:

- Click the **Edit** button and the **Signaling QOS** window will open.
- Verify that **Enabled** is selected.
- Select **DCSP**
- Select **Value** = **EF**

Step 5 - Click **Finish**.



8.4.3.2 Verizon – Signaling Rule

Step 1 - Select **Domain Policies** from the menu on the left-hand side menu (not shown).
Step 2 - Select **Signaling Rules** (not shown).
Step 3 - From the Signaling Rules menu, select the **default** rule.
Step 4 - Select **Clone Rule** button

- Enter a name**: Vz SIPTrk Sig Rule**

Step 5 - Click **Finish** (not shown).
Step 6 - Highlight the **Vz SIPTrk Sig Rule**, select the **Signaling QoS** tab and enter the following:

- Click the **Edit** button and the **Signaling QOS** window will open.
- Verify that **Enabled** is selected.
- Select **DCSP**
- Select **Value** = **AF32**

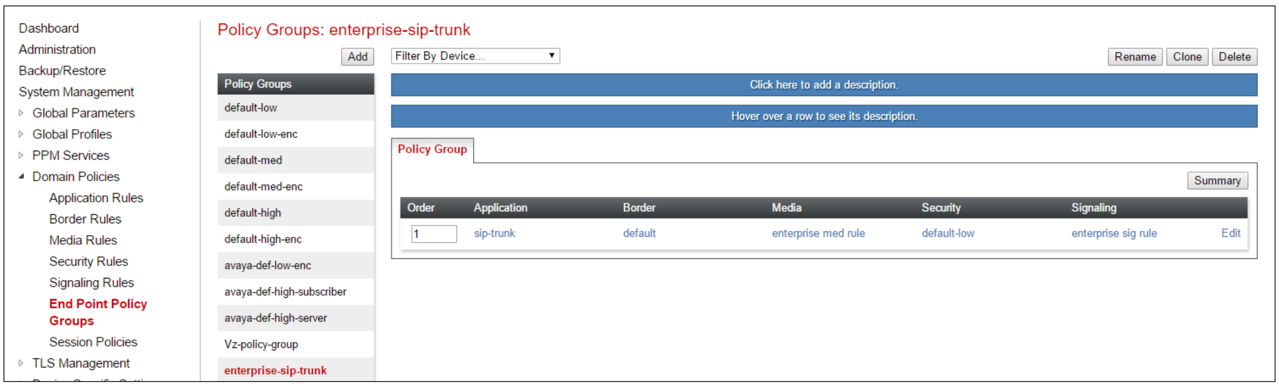
Step 5 - Click **Finish**.

8.4.4 Endpoint Policy Groups – Enterprise Connection

Step 1 - Select **Domain Policies** from the menu on the left-hand side.
Step 2 - Select **End Point Policy Groups**.
Step 3 - Select **Add**.

- **Name**: **enterprise-sip-trunk**
- **Application Rule**: **sip-trunk** (created in **Section 8.4.1**)
- **Border Rule**: **default**
- **Media Rule**: **enterprise med rule** (created in **Section 8.4.2.1**)
- **Security Rule**: **default-low**
- **Signaling Rule**: **enterprise sig rule** (created in **Section 8.4.3.1**)

Step 4 - Select **Finish** (not shown). The completed **Policy Groups** screen is shown below.

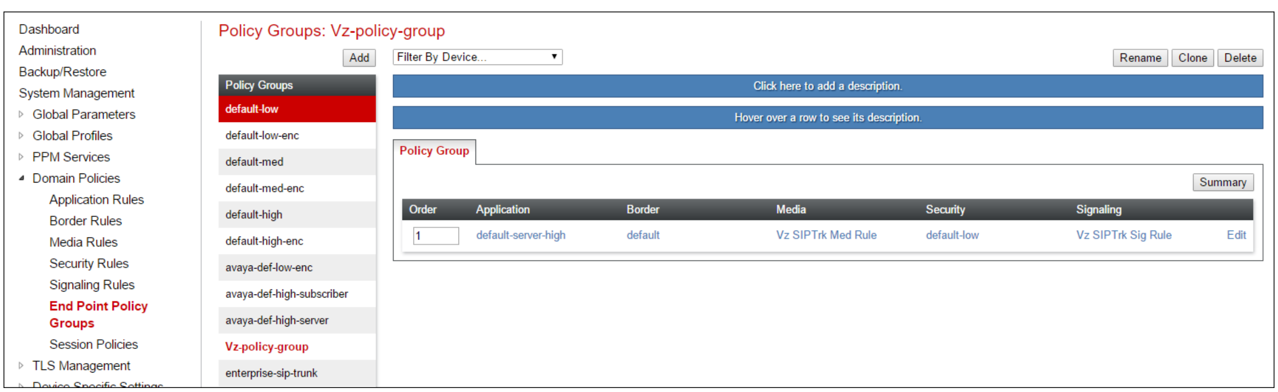


8.4.5 Endpoint Policy Groups – Verizon Connection

Step 1 - Repeat steps **1** through **4** from **Section 8.4.4** with the following changes:

- **Group Name**: **Vz-policy-group**
- **Media Rule**: **Vz SIPTrk Med Rule** (created in **Section 8.4.2.2**)
- **Signaling Rule**: **Vz SIPTrk Sig Rule** (created in **Section 8.4.3.2**)

Step 2 - Select **Finish** (not shown).



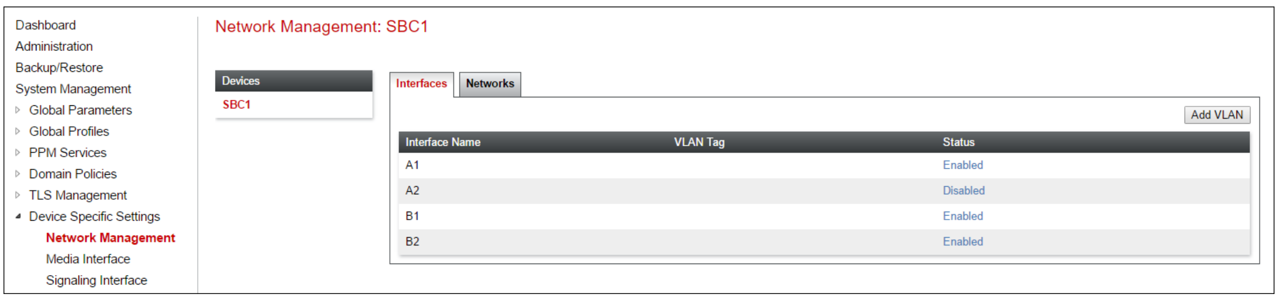
8.5. Device Specific Settings

Device Specific Settings allows aggregate system information to be viewed and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network. Specifically, it gives the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality and protocol scrubber rules, end-point and session call flows, as well as the ability to manage system logs and control security features.

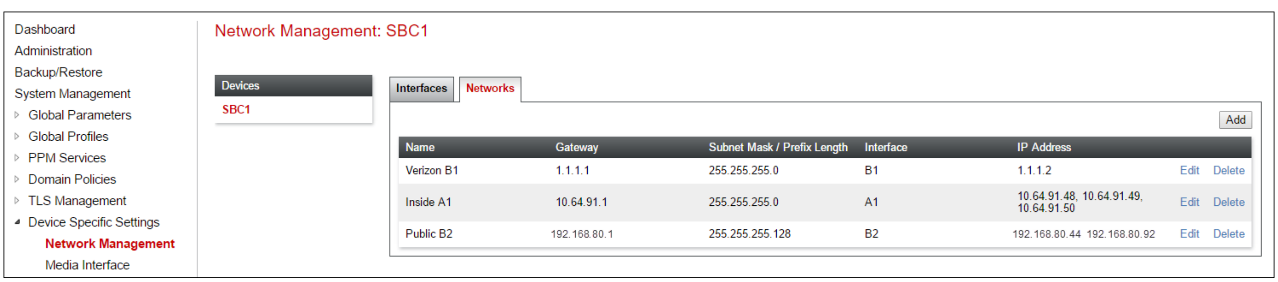
8.5.1 Network Management

Step 1 - Select **Device Specific Settings → Network Management** from the menu on the left-hand side.

Step 2 - The **Interfaces** tab displays the enabled/disabled interfaces. In the reference configuration, interfaces A1 (private) and B1 (public) interfaces are used.



Step 3 - Select the **Networks** tab to display the IP provisioning for the A1 and B1 interfaces. These values are normally specified during installation. These can be modified by selecting **Edit**; however, some of these values may not be changed if associated provisioning is in use.



8.5.2 Media Interfaces

The Media Interface screen is where the SIP media ports are defined. Avaya SBCE will send SIP media on the defined ports. Create a SIP Media Interface for both the inside and outside IP interfaces.

Step 1 - Select **Device Specific Settings** from the menu on the left-hand side.

Step 2 - Select **Media Interface**.

Step 3 - Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:

- **Name**: **Inside-Med-50**
- **IP Address**: Select **Inside-A1 (A1,VLAN0)** and **10.64.91.50**
- **Port Range**: **35000 – 40000**

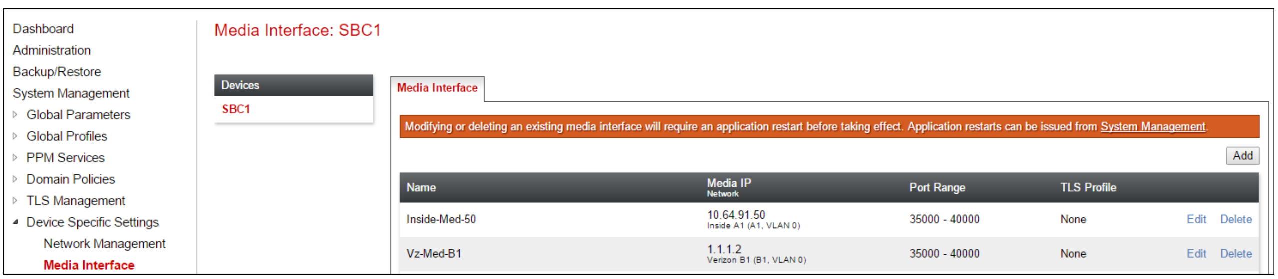
Step 4 - Click **Finish** (not shown).

Step 5 - Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:

- **Name**: **Vz-Med-B1**
- **IP Address**: Select **Verizon-B1 (B1,VLAN0)** and **1.1.1.2**
- **Port Range**: **35000 – 40000**

Step 6 - Click **Finish** (not shown). Note that changes to these values require an application restart (see **Section 8.1**).

The completed **Media Interface** screen in the shared test environment is shown below.



8.5.3 Signaling Interface

The Signaling Interface screen is where the SIP signaling ports are defined. Avaya SBCE will listen for SIP requests on the defined ports. Create a Signaling Interface for both the inside and outside IP interfaces.

Step 1 - Select **Device Specific Settings** from the menu on the left-hand side.

Step 2 - Select **Signaling Interface**.

Step 3 - Select **Add** (not shown) and enter the following:

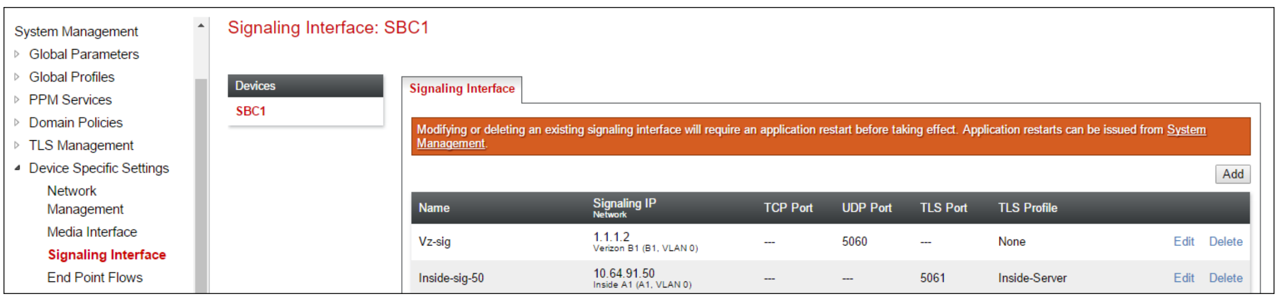
- **Name**: **Inside-Sig-50**
- **IP Address**: Select **Inside A1 (A1,VLAN0)** and **10.64.91.50**
- **TLS Port**: **5061**
- **TLS Profile**: Select the TLS server profile created in **Section 8.2.2** (e.g., **Inside-Server**)

Step 4 - Click **Finish** (not shown).

Step 5 - Select **Add** again, and enter the following:

- **Name**: **Vz-sig**
- **IP Address**: Select **Verizon B1 (B1,VLAN0)** and **1.1.1.2**
- **UDP Port**: **5060**

Step 6 - Click **Finish** (not shown). Note that changes to these values require an application restart (see **Section 8.1**).



8.5.4 Server Flows – For Session Manager

Step 1 - Select **Device Specific Settings → Endpoint Flows** from the menu on the left-hand side (not shown).

Step 2 - Select the **Server Flows** tab (not shown).

Step 3 - Select **Add**, (not shown) and enter the following:

- **Flow Name**: **SM8 to Vz IPT**.
- **Server Configuration**: **SM8** (**Section 8.3.4**).
- **URI Group**: *
- **Transport**: *
- **Remote Subnet**: *
- **Received Interface**: **Vz-sig** (**Section 8.5.3**).
- **Signaling Interface**: **Inside-sig-50** (**Section 8.5.3**).
- **Media Interface**: **Inside-Med-50** (**Section 8.5.2**).
- **End Point Policy Group**: **enterprise-sip-trunk** (**Section 8.4.4**).
- **Routing Profile**: **route to Vz IPT** (**Section 8.3.7**).
- **Topology Hiding Profile**: **Enterprise-Topology** (**Section 8.3.8**).
- Let other values default.

Step 4 - Click **Finish** (not shown).

View Flow: SM8 to Vz IPT

Criteria

Flow Name	SM8 to Vz IPT
Server Configuration	SM8
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Vz-sig

Profile

Signaling Interface	Inside-sig-50
Media Interface	Inside-Med-50
Secondary Media Interface	None
End Point Policy Group	enterprise-sip-trunk
Routing Profile	route to Vz IPT
Topology Hiding Profile	Enterprise-Topology
Signaling Manipulation Script	None
Remote Branch Office	Any

8.5.5 Server Flows – For Verizon

Step 1 - Repeat steps **1** through **4** from **Section 8.5.4**, with the following changes:

- **Flow Name**: **Verizon IPT Flow**.
- **Server Configuration**: **Verizon IPT** (**Section 8.3.5**).
- **URI Group**: *
- **Transport**: *
- **Remote Subnet**: *
- **Received Interface**: **Inside-sig-50** (**Section 8.5.3**).
- **Signaling Interface**: **Vz-sig (Section 8.5.3**).
- **Media Interface**: **Vz-Med-B1** (**Section 8.5.2**).
- **End Point Policy Group**: **Vz-policy-group** (**Section 8.4.5**).
- **Routing Profile**: **route to SM8** (**Section 8.3.6**).
- **Topology Hiding Profile**: **Vz th profile** (**Section 8.3.9**).

View Flow: Verizon IPT Flow

Criteria

Flow Name	Verizon IPT Flow
Server Configuration	Verizon IPT
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Inside-sig-50

Profile

Signaling Interface	Vz-sig
Media Interface	Vz-Med-B1
Secondary Media Interface	None
End Point Policy Group	Vz-policy-group
Routing Profile	route to SM8
Topology Hiding Profile	Vz th profile
Signaling Manipulation Script	None
Remote Branch Office	Any

The completed **End Point Flows** screen in the shared test environment is shown below.

Server Configuration: SM8

Update

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	SM8 to Vz IPT	*	Vz-sig	Inside-sig-50	enterprise-sip-trunk	route to Vz IPT	View	Clone	Edit	Delete

Server Configuration: Verizon IPT

Update

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Verizon IPT Flow	*	Inside-sig-50	Vz-sig	Vz-policy-group	route to SM8	View	Clone	Edit	Delete

9. Verizon Business IP Trunking Services Suite Configuration

Information regarding the Verizon Business IP Trunking Services suite offer can be found at http://www.verizonbusiness.com/Products/communications/ip-telephony/ or by contacting a Verizon Business sales representative.

The reference configuration described in these Application Notes is located in the Avaya Solutions and Interoperability Test Lab. Access to the Verizon Business IP Trunking Services suite was via a Verizon Private IP (PIP) T1 connection. Verizon Business provided all of the necessary service provisioning.

9.1. Service Access Information

The following service access information (FQDN, ports, DID numbers) was provided by Verizon for the sample configuration.

CPE (Avaya)	Verizon Network
adevc.avaya.globalipcom.com ***UDP port 5060***	***pcelban0001.avayalincroft.globalipcom.com*** ***UDP Port 5071***

IP DID Numbers
732-945-0231
732-945-0232
732-945-0233
732-945-0234
732-945-0235
732-945-0236
732-945-0237
732-945-0238
732-945-0239

10. Verification Steps

This section provides example verifications of the Avaya configuration with Verizon Business Private IP (PIP) Trunk service.

10.1. Avaya Aura® Communication Manager Verifications

This section illustrates verifications from Communication Manager.

The following edited Communication Manager ***list trace tac*** trace output shows a call incoming on trunk group 1. The PSTN telephone dialed 732-945-0233. Session Manager mapped the number received from Verizon to the extension of a Communication Manager telephone (x50233).

```
list trace tac *01                                                    Page   1

                                LIST TRACE

time            data

11:38:55 TRACE STARTED 01/02/2019 CM Release String R018x.00.0.822.0
11:39:09 SIP<INVITE sips:50233@avayalab.com SIP/2.0
11:39:09     Call-ID: 98150601e5575f4ef875381771400a3f
11:39:09     active trunk-group 1 member 1    cid 0x7b6
11:39:09     dial 50233
11:39:09     term station      50233 cid 0x7b6
11:39:09     Called party  uses private-numbering
11:39:09 SIP>INVITE sips:50233@avayalab.com SIP/2.0
11:39:09     Call-ID: b0f526d6ebd41e9b82c0c29742c4c
11:39:09 SIP<SIP/2.0 100 Trying
11:39:09     Call-ID: b0f526d6ebd41e9b82c0c29742c4c
11:39:09 SIP<INVITE sips:50233@avayalab.com SIP/2.0
11:39:09     Call-ID: b0f526d6ebd41e9b82c0c29742c4c
11:39:09 SIP>INVITE sips:50233@avayalab.com SIP/2.0
11:39:09     Call-ID: b0f526d6ebd41e9b82c0c29742c4c
11:39:09 SIP<SIP/2.0 100 Trying
11:39:09     Call-ID: b0f526d6ebd41e9b82c0c29742c4c
11:39:09 SIP>SIP/2.0 100 Trying
11:39:09     Call-ID: b0f526d6ebd41e9b82c0c29742c4c
11:39:09 SIP<SIP/2.0 180 Ringing
11:39:09     Call-ID: b0f526d6ebd41e9b82c0c29742c4c
11:39:09 SIP>SIP/2.0 180 Ringing
```

The following screen shows **Page 2** of the output of the ***status trunk*** command pertaining to this same call. Note the signaling using port 5081 between Communication Manager and Session Manager. Note the media is "ip-direct" from the IP Telephone (**10.64.91.154**) to the inside IP address of Avaya SBCE (**10.64.91.50**) using codec G.729a.

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

Near-end Signaling Loc: PROCR
  Signaling   IP Address                              Port
   Near-end:  10.64.91.75                           : 5081
    Far-end:  10.64.91.81                           : 5081
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.729
   Audio      IP Address                              Port
   Near-end:  10.64.91.154                          : 5004
    Far-end:  10.64.91.50                           : 35938
```

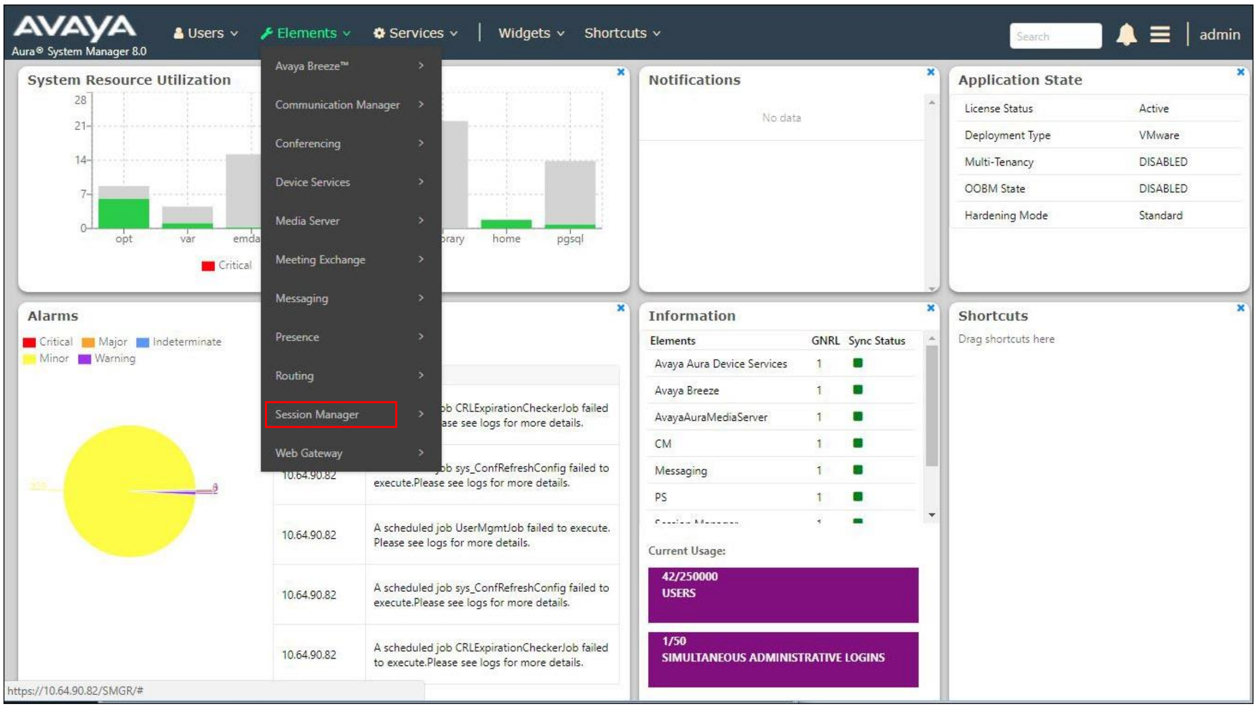
The following screen shows **Page 3** of the output of the ***status trunk*** command pertaining to this same call. Here it can be observed that G.729 codec is used.

```
status trunk 1/1                                                Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00001
T00001:TX:10.64.91.50:35938/g729/20ms/1-srtp-aescm128-hmac80
T00028:RX:10.64.91.154:5004/g729/20ms/1-srtp-aescm128-hmac80
```

10.2. Avaya Aura® Session Manager Verification

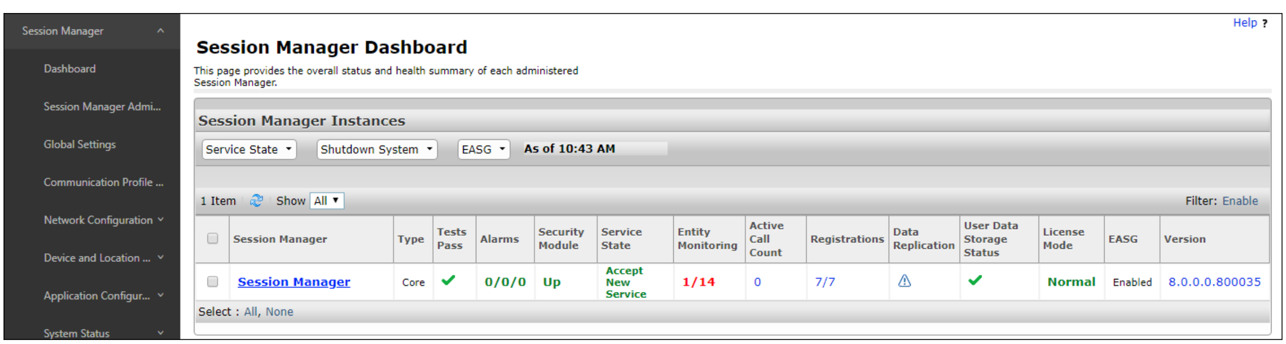
The Session Manager configuration may be verified via System Manager.

Step 1 - Using the procedures described in **Section 5**, access the System Manager GUI. From the **Home** screen, under the **Elements** heading, select **Session Manager**.

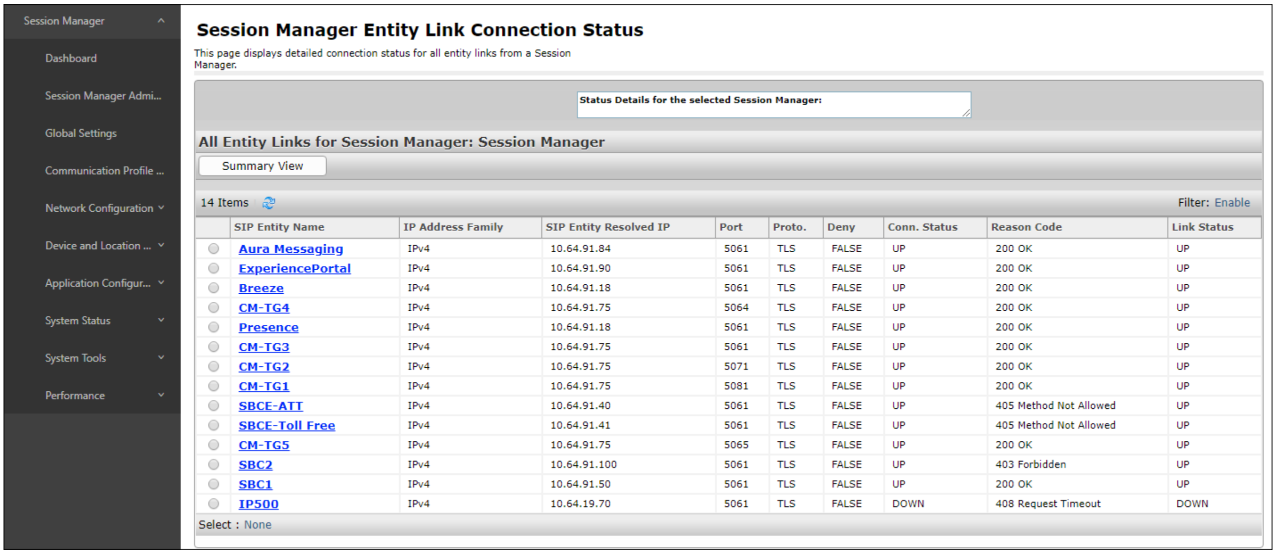


Step 2 - The Session Manager Dashboard is displayed. Note that the **Test Passed**, **Alarms**, **Service State**, and **Data Replication** columns all show good status.

In the **Entity Monitoring** column, Session Manager shows that there is **1** alarm out of the **14** Entities defined.



Step 3 - Clicking on the **1/14** entry (shown above) in the **Entity Monitoring** column, results in the following display:



Session Manager | Dashboard | Session Manager Admi... | Global Settings | Communication Profile ... | Network Configuration | Device and Location ... | Application Configur... | System Status | System Tools | Performance

Session Manager Entity Link Connection Status

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

Status Details for the selected Session Manager:

All Entity Links for Session Manager: Session Manager

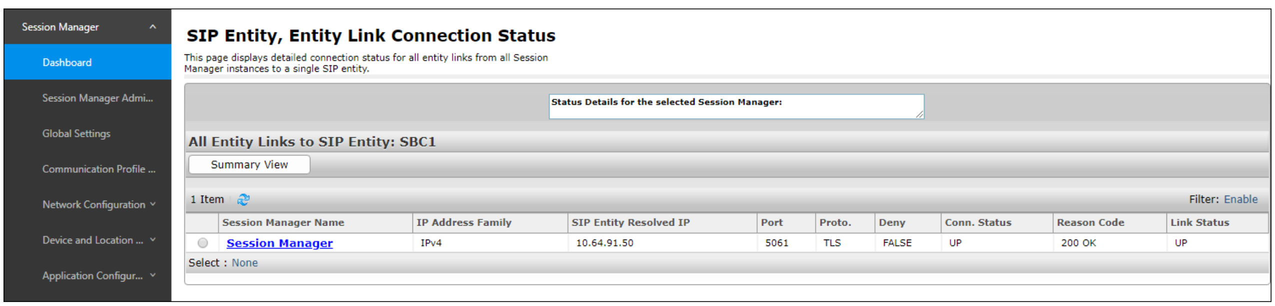
Summary View

14 Items Filter: Enable

	SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	Aura Messaging	IPv4	10.64.91.84	5061	TLS	FALSE	UP	200 OK	UP
○	ExperiencePortal	IPv4	10.64.91.90	5061	TLS	FALSE	UP	200 OK	UP
○	Breeze	IPv4	10.64.91.18	5061	TLS	FALSE	UP	200 OK	UP
○	CM-TG4	IPv4	10.64.91.75	5064	TLS	FALSE	UP	200 OK	UP
○	Presence	IPv4	10.64.91.18	5061	TLS	FALSE	UP	200 OK	UP
○	CM-TG3	IPv4	10.64.91.75	5061	TLS	FALSE	UP	200 OK	UP
○	CM-TG2	IPv4	10.64.91.75	5071	TLS	FALSE	UP	200 OK	UP
○	CM-TG1	IPv4	10.64.91.75	5081	TLS	FALSE	UP	200 OK	UP
○	SBCE-ATT	IPv4	10.64.91.40	5061	TLS	FALSE	UP	405 Method Not Allowed	UP
○	SBCE-Toll Free	IPv4	10.64.91.41	5061	TLS	FALSE	UP	405 Method Not Allowed	UP
○	CM-TG5	IPv4	10.64.91.75	5065	TLS	FALSE	UP	200 OK	UP
○	SBC2	IPv4	10.64.91.100	5061	TLS	FALSE	UP	403 Forbidden	UP
○	SBC1	IPv4	10.64.91.50	5061	TLS	FALSE	UP	200 OK	UP
○	IP500	IPv4	10.64.19.70	5061	TLS	FALSE	DOWN	408 Request Timeout	DOWN

Select : None

From the list of monitored entities, select an entity of interest, such as **SBC1**. Under normal operating conditions, the **Link Status** should be **UP** as shown in the example screen below.



Session Manager | Dashboard | Session Manager Admi... | Global Settings | Communication Profile ... | Network Configuration | Device and Location ... | Application Configur...

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

Status Details for the selected Session Manager:

All Entity Links to SIP Entity: SBC1

Summary View

1 Item Filter: Enable

	Session Manager Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	Session Manager	IPv4	10.64.91.50	5061	TLS	FALSE	UP	200 OK	UP

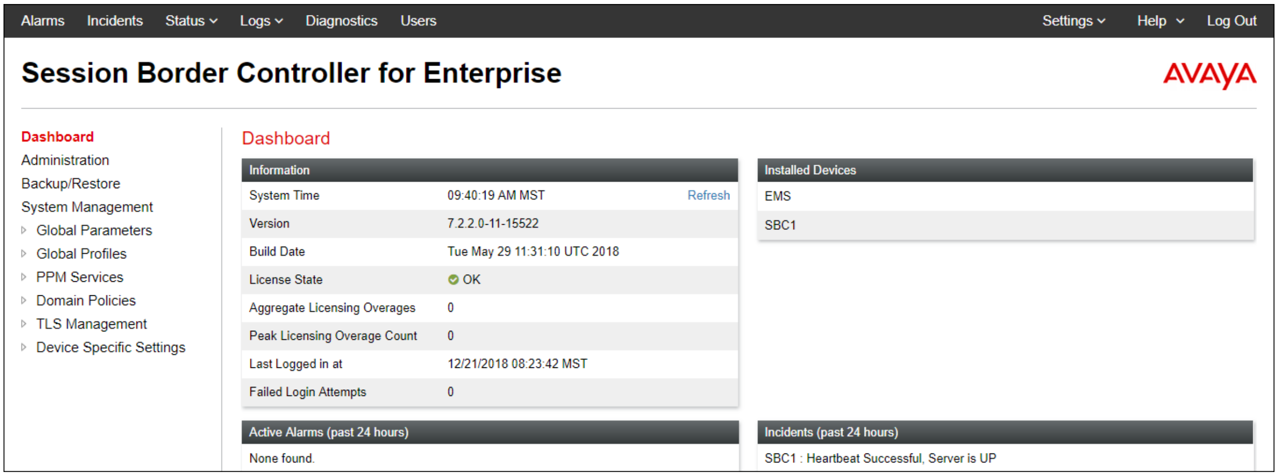
Select : None

Another useful tool is to select **System Tools → Call Routing Test** (not shown) from the left-hand menu. This tool allows specific call criteria to be entered, and the simulated routing of this call through Session Manager is then verified.

10.3. Avaya Session Border Controller for Enterprise Verification

10.3.1 Welcome Screen

The welcome screen shows alarms, incidents, and the status of all managed Avaya SBCEs at a glance.



10.3.2 Alarms

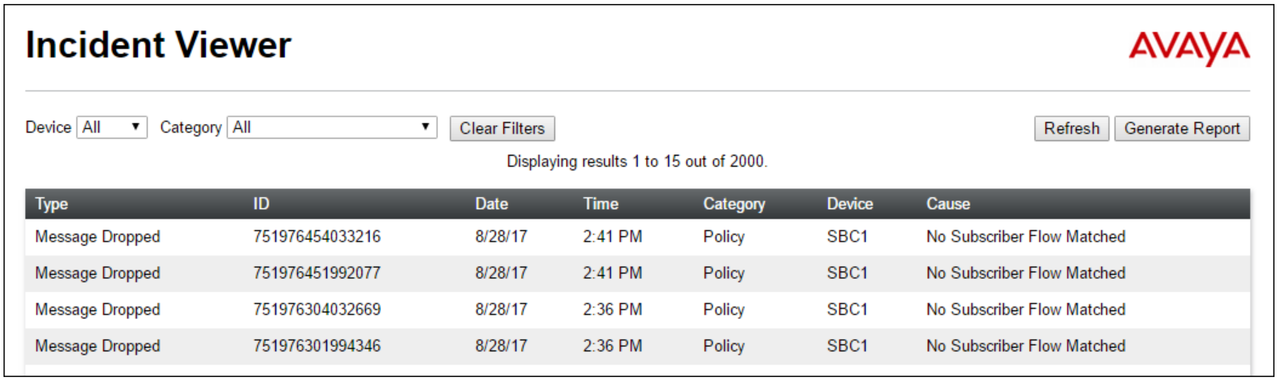
A list of the most recent alarms can be found under the **Alarms** tab on the top left bar.

Alarm Viewer:

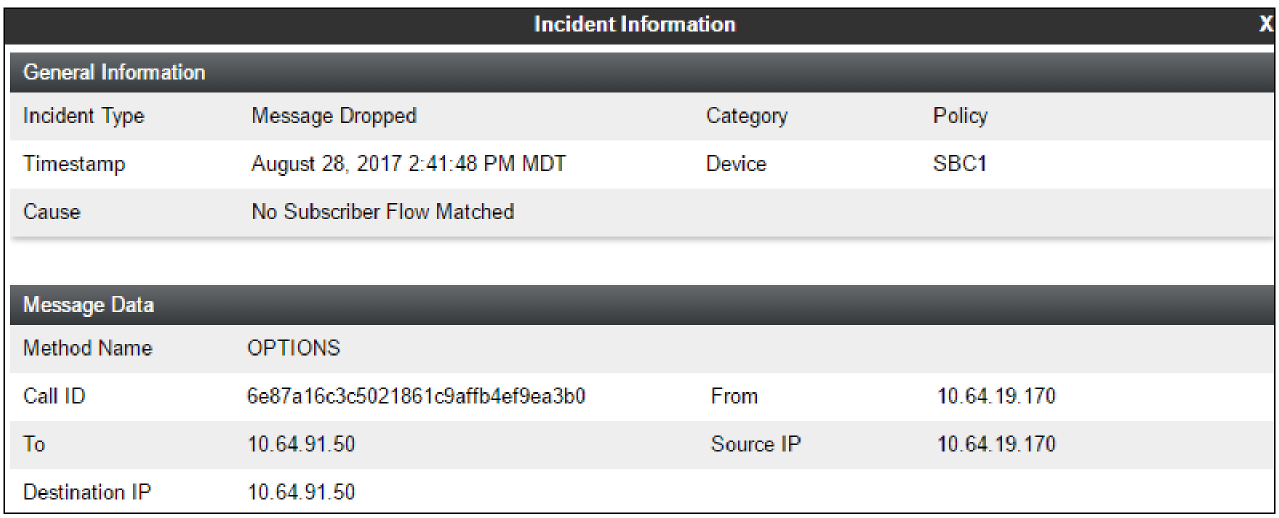
10.3.3 Incidents

A list of all recent incidents can be found under the **Incidents** tab at the top left next to the Alarms.

Incident Viewer:



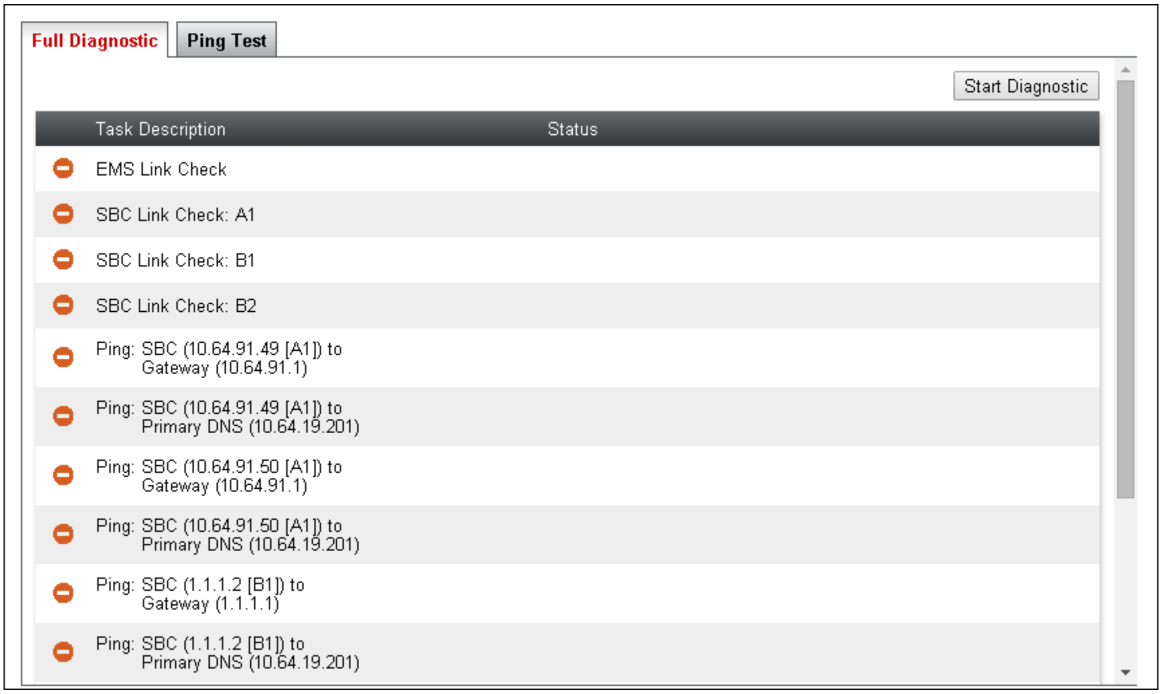
Further Information can be obtained by clicking on an incident in the incident viewer.



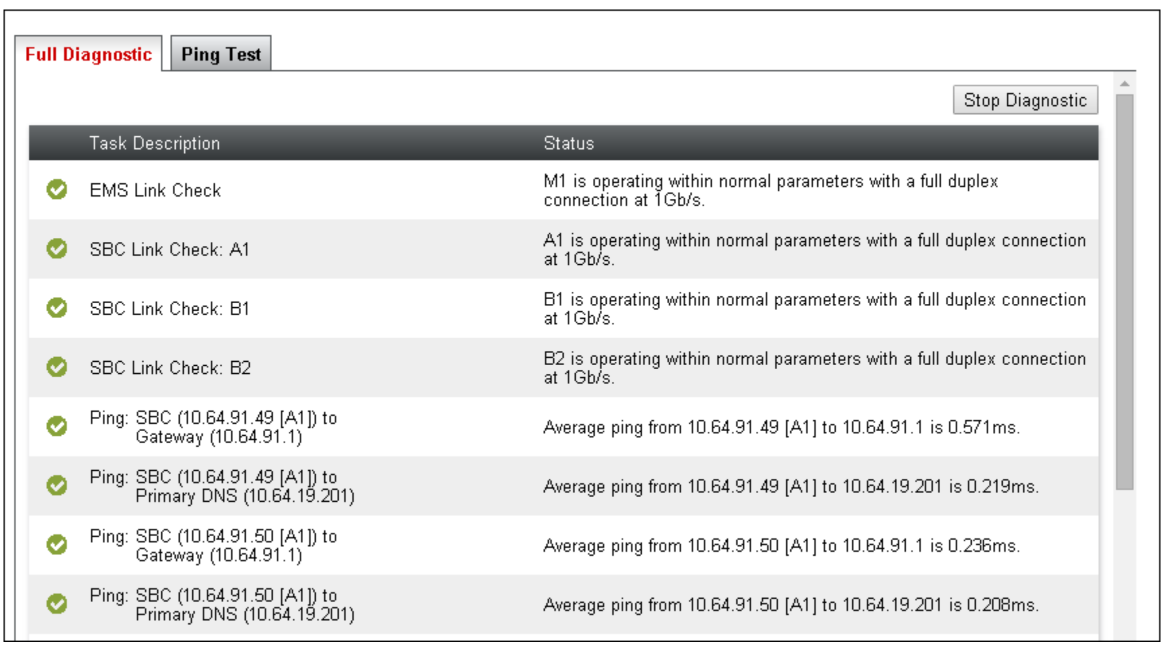
10.3.4 Diagnostics

The full diagnostics check will verify the link of each interface and ping the configured next-hop gateways and DNS servers.

Click on **Diagnostics** on the top bar, select the Avaya SBCE from the list of devices and then click "**Start Diagnostics**".

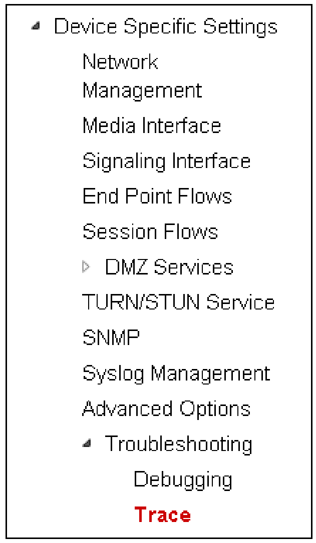


A green check mark or a red x will indicate success or failure.

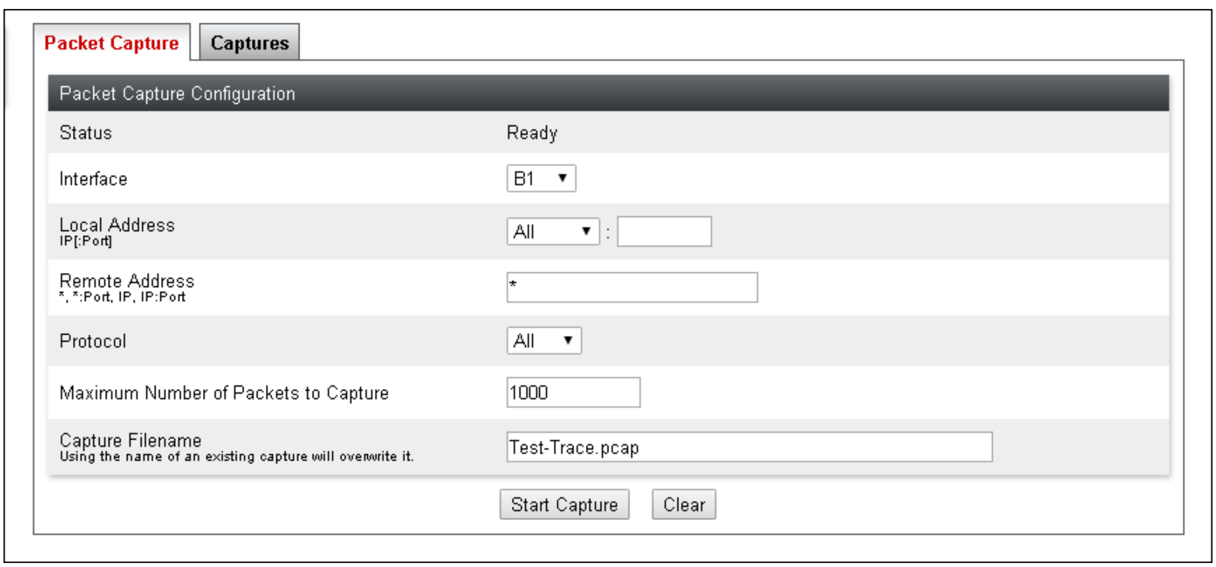


10.3.5 Tracing

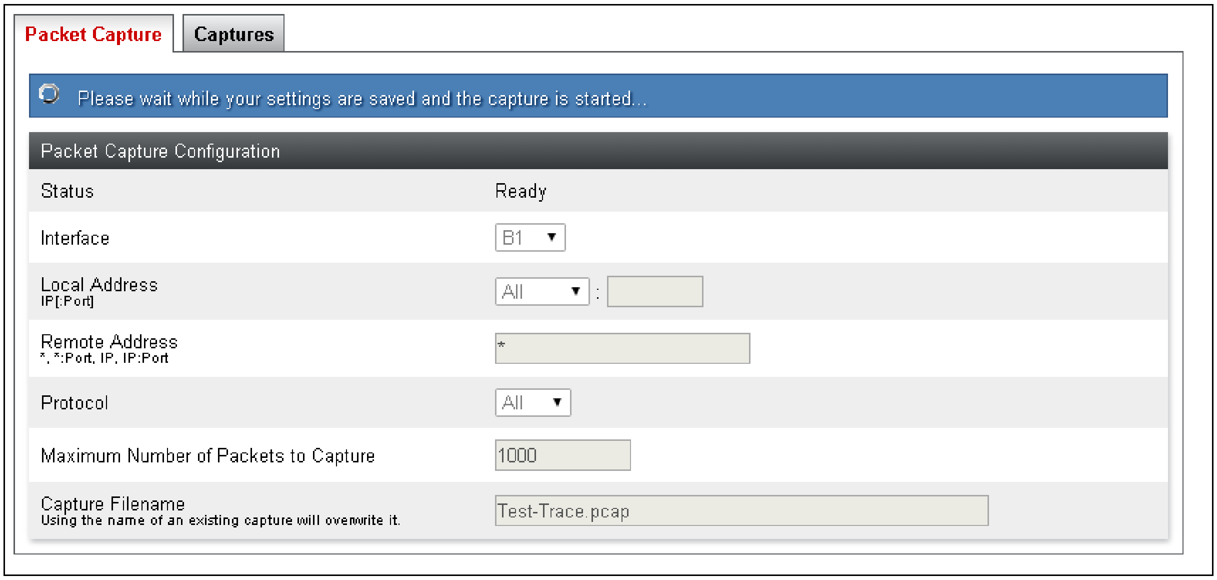
To take a call trace, Select **Device Specific Settings → Troubleshooting → Tracing** from the left-side menu as shown below.



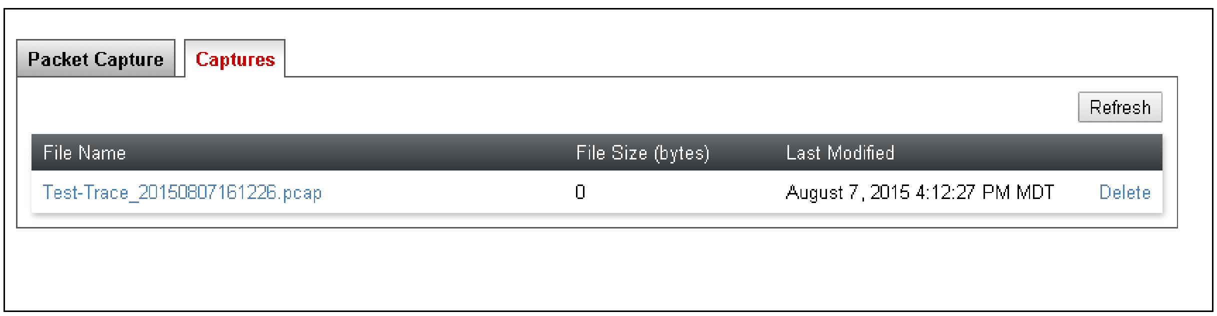
Select the **Packet Capture** tab and set the desired configuration for a call trace and click **Start Capture**.



When tracing has reached the desired number of packets the trace will stop automatically, or alternatively, click the **Stop Capture** button at the bottom.



Select the **Captures** tab at the top and the capture will be listed; select the **File Name** and choose to open it with an application like Wireshark.



11. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, Avaya Aura® Experience Portal 7.2, and Avaya Session Border Controller for Enterprise 7.2 can be configured to interoperate successfully with Verizon Business IP Trunking service. This solution allows Avaya Aura® Communication Manager and Avaya Aura® Session Manager users access to the PSTN using a Verizon Business IP Trunking public SIP trunk service connection.

12. Additional References

12.1. Avaya

Avaya product documentation, including the following, is available at http://support.avaya.com

Avaya Aura® Session Manager/System Manager

[1] *Deploying Avaya Aura® Session Manager and Branch Session Manager in Virtualized Environment*, Release 8.0, Issue 2, August 2018

[2] *Administering Avaya Aura® Session Manager*, Release 8.0, Issue 2, August 2018

[3] *Deploying Avaya Aura® System Manager in Virtualized Environment*, Release 8.0, Issue 2, September 2018

[4] *Administering Avaya Aura® System Manager for Release 8.0*, Issue 4, September 2018

Avaya Aura® Communication Manager

[5] *Deploying Avaya Aura® Communication Manager in Virtualized Environment*, Release 8.0, Issue 4, September 2018

[6] *Administering Avaya Aura® Communication Manager*, Release 8.0, Issue 1, July 2018

[7] *Administering Avaya G450 Branch Gateway*, Release 8.0, Issue 1, July 2018

[8] *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 8.0, Issue 2, July 2018

[9] *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager*, August 2015

Avaya Session Border Controller for Enterprise

[10] *Administering Avaya Session Border Controller for Enterprise*, Release 7.2.2, Issue 9, April 2018

[11] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.2.2, Issue 7, April 2018

Avaya Aura® Messaging

[12] *Administering Avaya Aura® Messaging*, Release 7.0.0, Issue 4, April 2018

Avaya Aura® Experience Portal

[13] *Administering Avaya Aura® Experience Portal*, Release 7.2.1, Issue 1, March 2018

[14] *Implementing Avaya Aura® Experience Portal on a single server*, Release 7.2, Issue 1, July 2017

12.2. Verizon Business

The following documents may be obtained by contacting a Verizon Business Account Representative.

[15] *Retail VoIP Interoperability Test Plan*

[16] *Network Interface Specification Retail VoIP Trunk Interface (for non-registering devices)*

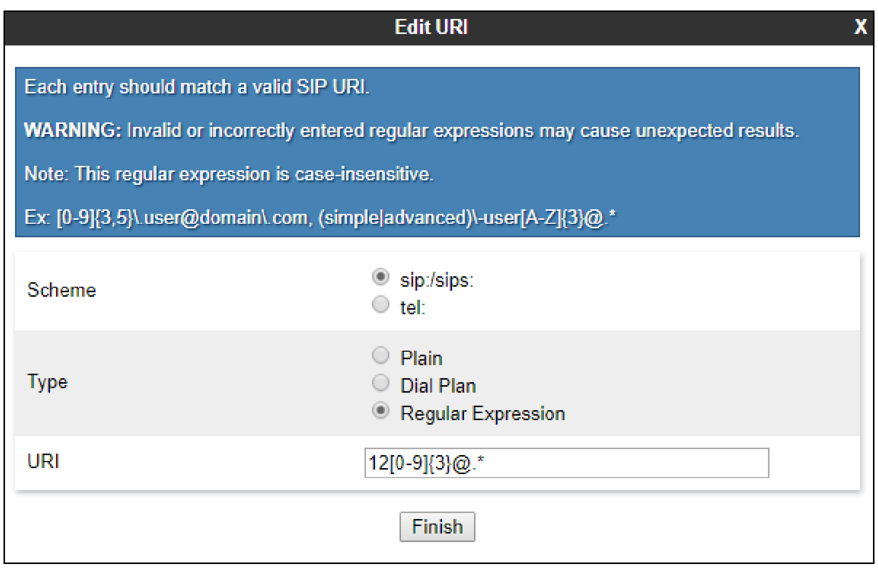
13. Appendix A – Avaya Session Border Controller for Enterprise – Refer Handling

One of the capabilities important to the Experience Portal environment is the Avaya SBCE Refer Handling option. As described in **Section 3.2.2**, Experience Portal inbound call processing may include call redirection to Communication Manager agents, or other CPE destinations. This redirection is accomplished by having Experience Portal send SIP REFER messaging to the Avaya SBCE. Enabling the Refer Handling option causes the Avaya SBCE to intercept and process the REFER and generate a new SIP INVITE messages back to the CPE (e.g., Communication Manager).

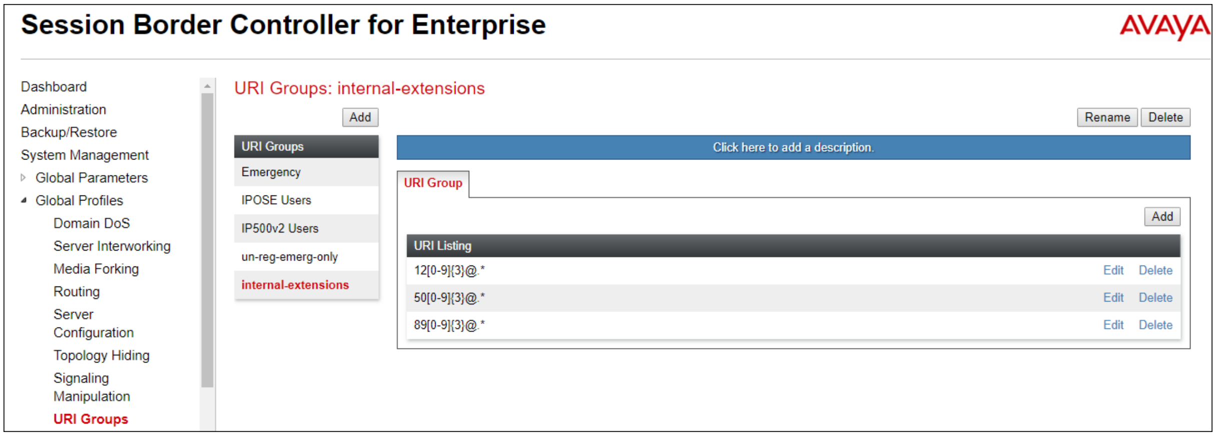
As an additional option, the Refer Handling feature can also specify *URI Group* criteria as a discriminator, whereby SIP REFER messages matching the URI Group criteria are processed by the Avaya SBCE, while SIP REFER messages that do not match the URI Group criteria, are passed through to Verizon.

Create a URI Group for numbers intended for Communication Manager.
Step 1 - Select **Global Profiles → URI Groups** from the left-hand menu.
Step 2 - Select **Add** and enter a descriptive **Group Name**, e.g., **internal-extension**, and select **Next** (not shown).
Step 3 - Enter the following:

- **Scheme**: **sip:/sips:**
- **Type: Regular Expression**
- **URI: 12[0-9]{3}@.*** This will match 5-digit local extensions starting with 12, e.g., 12001.
- Select **Finish**.



Step 4 - For additional entries, select **Add** on the right-hand side of the URI Group tab and repeat **Step 3**.

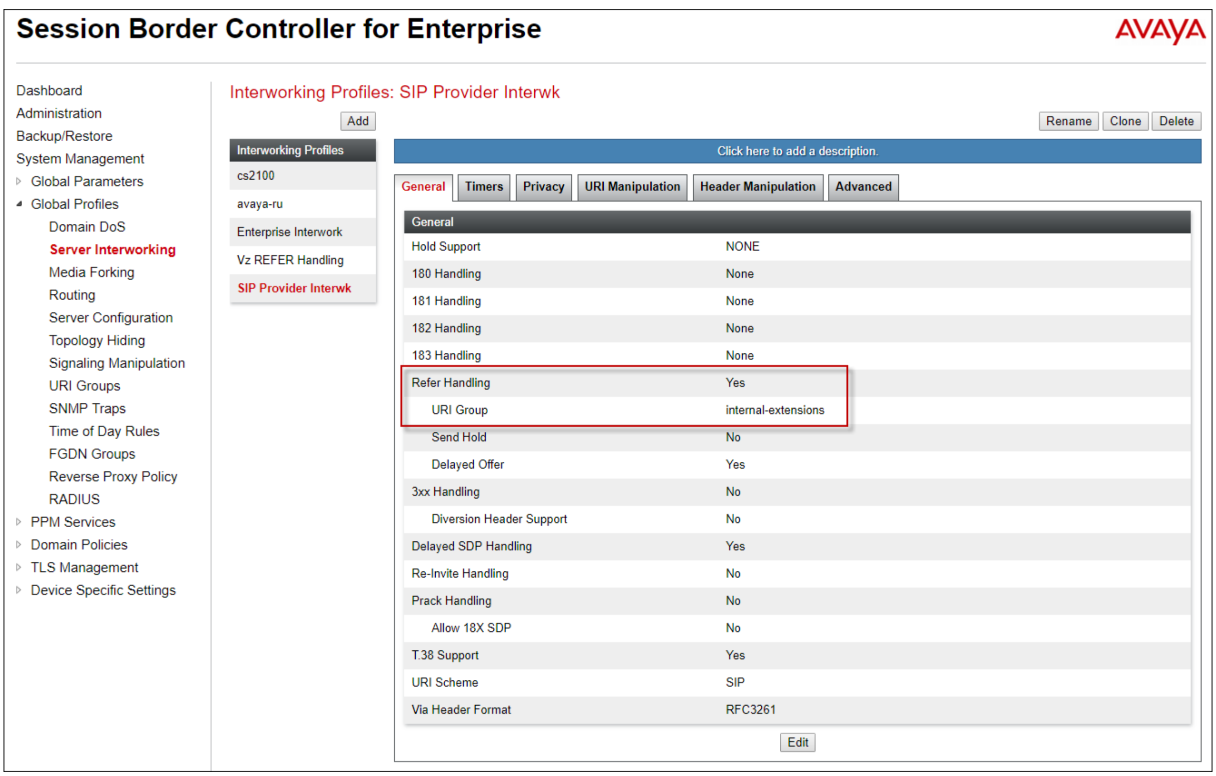


Edit the existing Verizon Server Interworking Profile to enable Refer Handling and assign the newly created URI Group.

Step 1 - Select **Global Profiles → Server Interworking** from the left-hand menu

Step 2 - Select the Verizon Server Interworking Profile created in **Section 8.3.2** and click **Edit**

- Check **Refer Handling**.
- **URI Group: internal-extensions**
- Select **Finish**.



Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.