Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0 and Avaya Session Border Controller for Enterprise 7.2 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Session Manager 8.0, Avaya Aura® Communication Manager 8.0, and the Avaya Session Border Controller for Enterprise 7.2 with the AT&T IP Toll Free service using AT&T's **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager 8.0 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 8.0 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. The Avaya Session Border Controller for Enterprise 7.2 is the point of connection between Avaya Aura® Session Manager and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.

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

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program.

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1 Introduction

These Application Notes describe the steps for configuring Avaya Aura® Session Manager 8.0, Avaya Aura® Communication Manager 8.0, and the Avaya Session Border Controller for Enterprise 7.2 with the AT&T IP Toll Free service using AT&T Virtual Private Network (AVPN) or Managed Internet Service Private Network Transport (MIS/PNT) connections[1].

Avaya Aura® Session Manager 8.0 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 8.0 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. The Avaya Session Border Controller for Enterprise 7.2 (Avaya SBCE) is the point of connection between Avaya Aura® Session Manager and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to adjust the SIP signaling for interoperability.

The AT&T IP Toll Free service, (referred to in the remainder of this document as IPTF), is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks utilizing AVPN or MIS/PNT transport.

Note – These Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service. That solution is described in a separate document.

2 General Test Approach and Test Results

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.

The interoperability compliance testing focused on verifying inbound and outbound call flows between IPTF and the Customer Premises Equipment (CPE) containing Communication Manager, Session Manager, and the Avaya SBCE (see **Section 3.2** for call flow examples).

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.

[1] MIS/PNT transport does not support compressed RTP (cRTP), however AVPN transport does support cRTP.

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 these Application Notes, the interface between Avaya systems and the AT&T Toll Free service did not include use of any specific encryption features as requested by AT&T.

2.1 Interoperability Compliance Testing

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the IPTF network. Calls were made from the PSTN, across the IPTF network, to the CPE.
The following SIP trunking VoIP features were tested with the IPTF service:

- Inbound PSTN/IPTF calls to Communication Manager stations, Vector Directory Numbers (VDNs), Vectors, and Agents.
- Call and two-way talk path establishment between PSTN and Communication Manager telephones/Agents via IPTF.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729A and G.711Mu codecs.
- T.38 fax calls via IPTF to Communication Manager fax endpoints.
- G.711 pass-through fax calls via IPTF to Communication Manager fax endpoints.
- DTMF tone transmission using RFC 2833/4733 between Communication Manager and IPTF automated access systems.
- Inbound IPTF service calls to Communication Manager that are routed to Agent queues or directly to Agents.
- IPTF network features such as Legacy Transfer Connect and Alternate Destination Routing (ADR).
- Verify reception of IPTF SIP Multipart/NSS headers, including SDP and XML content.
- Long duration calls.

2.2 Test Results

The test objectives stated in **Section 2.1**, with limitations as noted below, were verified.

1. **IP Toll Free ADR Call Redirection feature in response to a ring-no-answer condition**. There is an anomaly in the VIT lab where the Ring No Answer did not get triggered due to Lab restrictions. However, in Production, if there is no answer for 20 seconds, the Ring No Answer will be invoked.

2. **IP Toll Free ADR Call Redirection feature based on SIP error code response**. Upon receiving an error response, IPTF service can be configured to invoke ADR Call Redirection. The following error codes were producible by the reference configuration and tested successfully, 480 Temporarily Unavailable, 486 Busy Here, 503Service Unavailable, and 500 Server Internal Error. The following error codes are also supported by IPTF

service, but were not producible by the reference configuration, and thus not tested, 408 Request Timeout, 504 Server Timeout, and 600 Busy Everywhere.

3. **G.726-32 codec support**. While Communication Manager supports G.726-32, the IPTF implementation of G.726-32 results in poor audio quality. Therefore, G.726-32 codec is not supported between Communication Manager and the IPTF service.

4. **T.38/G.729 fax is limited to 9600bps when using the G4xx Media Gateways**. A G450 Media Gateway is used in the reference configuration. As a result, T.38/G.729 fax was limited to 9600 bps. Also note that the sender and receiver of a T.38 fax call may use either Group 3 or Super Group 3 fax machines, but the T.38 fax protocol carries all fax transmissions as Group 3. Also note that inbound/outbound G.711 pass-through fax ran successfully at best line speed (rates of 14400 bps were observed).

5. **G.711 pass-through fax**. G.711 pass-through fax was tested in addition to T.38 fax. This was done by configuring a different Communication Manager **ip-codec-set** form (**Section 6.7.3**) to use **G.711 MU** codec as the first codec choice, and setting **Fax Mode** to **off**. The network region of the G450 Media Gateway hosting the fax machine was changed from the enterprise region, to one that utilized this ip-codec-set for IPTF service. Faxes using G.711 pass-through completed successfully during the test. It should be noted however, that due to the unpredictability of pass-through techniques, which only work well on networks with very few hops and with limited end-to-end delay, G.711 fax pass-through is delivered in Communication Manager on a "best effort" basis; its success is not guaranteed, and it should be used at the customer's discretion.

6. **IP Toll Free service Landline/Mobility test cases could not be executed**. The AT&T supplied IP Toll Free test plan specifies test cases to verify the transmission of Landline/Mobility data by the IP Toll Free service. Due to network access issues, these test cases could not be executed.

7. **Removal of unnecessary SIP headers**. In an effort to reduce packet size (or block a header containing private addressing), Session Manager is provisioned to remove SIP headers not required by the AT&T IPTF service (see **Section 5.3.2**). These headers are:
   - AV-Global-Session-ID, Alert-Info, Endpoint-View, P-AV-Message-Id, P-Charging-Vector, P-Location, Av-Secure-Indication

8. **Avaya SIP endpoints may generate three Bandwidth headers; b=TIAS:64000, b=CT:64, and b=AS:64, causing AT&T network issues**. Certain Avaya SIP endpoints (e.g., 9641, 9621, and 9608 models) may generate various Bandwidth headers depending on the call flow. It has been observed that sending these Bandwidth headers may cause issues with AT&T services. Therefore, an Avaya SBCE Signaling Manipulation Rule is used to remove these headers (see **Section 7.3.2**).

9. **Enhanced CID – NSS feature**. The inbound calls to Communication Manager are not exercising the Enhanced CID feature. Although Communication Manager is accepting SIP Multipart/NSS headers, it is neither passing nor acting upon it. It is simply being ignored.

10. **The version of Communication Manager used during testing specified a ptime value of 20 in the SIP SDP when the codec set was configured for 30**. Although no issues were found during testing, AT&T recommends that for maximum customer bandwidth utilization, a ptime value of 30 milliseconds should be specified.

2.3 Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (800) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting: http://support.avaya.com. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on http://support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

3 Reference Configuration

The reference configuration used in these Application Notes is shown in **Figure 1** and consists of several components:

- Session Manager 8.0 provides core SIP routing and integration services that enables communication between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Avaya SIP endpoints register to Session Manager.
- System Manager 8.0 provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager 8.0 provides the voice communication services for a particular enterprise site. Avaya H.323 endpoints register to Communication Manager.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G450 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya Aura® Media Server provides additional media resources for Communication Manager.
- Avaya desk telephones are represented with Avaya 96x1 Series IP Telephone (running H.323 firmware), a 96x1 Series IP Telephone (running SIP firmware), an Avaya 2420 Digital Telephone, as well as Avaya one-X® Agent soft phone (H323).
- The Avaya SBCE 7.2 provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the IPTF service and the enterprise internal network.

- Avaya Aura® Messaging was used in the reference configuration to provide voice mailbox capabilities. This solution is extensible to other Avaya messaging platforms. The provisioning of Avaya Aura® Messaging is beyond the scope of this document.
- The IPTF service uses SIP over UDP to communicate with enterprise edge SIP devices, e.g., the Avaya SBCE. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Avaya SBCE (e.g., UDP, TCP, or TLS) and Communication Manager (e.g., TCP or TLS). In the reference configuration, Session Manager uses SIP over TLS to communicate with the Avaya SBCE and Communication Manager.
- Inbound calls were placed from the PSTN via the IPTF service, through the Avaya SBCE to the Session Manager, which routed the call to Communication Manager. Communication Manager terminated the calls to the appropriate Agent queue, Agent phone, or fax extension.

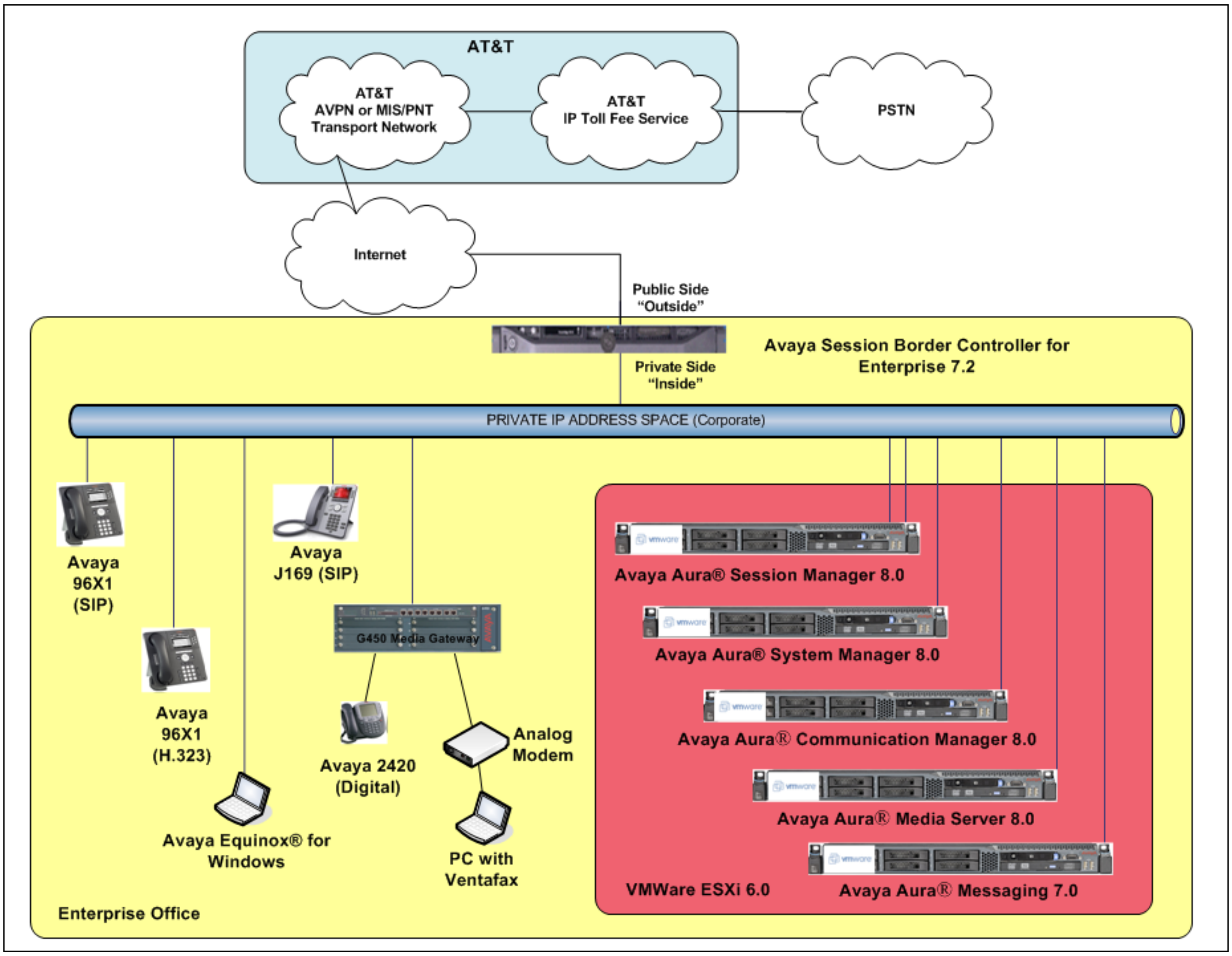


Figure 1: Reference configuration

3.1 Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

Note - The AT&T IP Toll Free service Border Element IP address and DNIS digits, (destination digits specified in the SIP Request URIs sent by the AT&T Toll Free service) are shown in this document as examples. AT&T Customer Care will provide the actual IP addresses and DNIS digits as part of the IP Toll Free provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura® System Manager	
IP Address	10.64.90.82
Avaya Aura® Session Manager	
Management IP Address	10.64.90.81
Network IP Address	10.64.91.81
Avaya Aura® Communication Manager	
IP Address	10.64.91.75
Avaya Aura® Communication Manager extensions	89xxx = Stations 2xxxx = Agents 71xxx = Agent skill queue VDNs
Avaya Session Border Controller for Enterprise (SBCE)	
IP Address of Inside (Private) Interface	10.64.91.41
IP Address of Outside (Public) Interface	192.168.80.43 (see note below)
AT&T IP Toll Free Border Element	
IP Address	192.168.225.210

Table 1: Illustrative Values Used in these Application Notes

Note – In the reference configuration, the IPTF service delivered 10 DNIS digits, with the format *00000xxxxx*. These DNIS digits are used in the provisioning defined in the following sections, not the dialed digits. The DNIS digit length can vary depending on the customer’s needs. Although during testing 10 digits were used, the total length supported by the IPTF service is 21 digits, including the five leading zeroes.

Note – For security reasons, the actual IP addresses of the Avaya SBCE and AT&T BE are not included in this document. However, as placeholders in the following configuration sections, the IP address of **192.168.80.43** (Avaya SBCE public interface) and **192.168.225.210** (AT&T BE IP address) are specified.

3.2 Call Flows

To understand how inbound AT&T IP Toll Free service calls are handled by the Avaya SBCE, Session Manager and Communication Manager, a general call flow is described below. In **Figure 2** an inbound IPTF service call arrives at the Avaya SBCE and is subsequently routed to Session Manager and to Communication Manager.

1. A PSTN telephone originates a call to an IPTF service number.
2. The PSTN routes the call to the IPTF service network.
3. The IPTF service 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 Routing Policies, determines 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 an Agent queue or telephone.

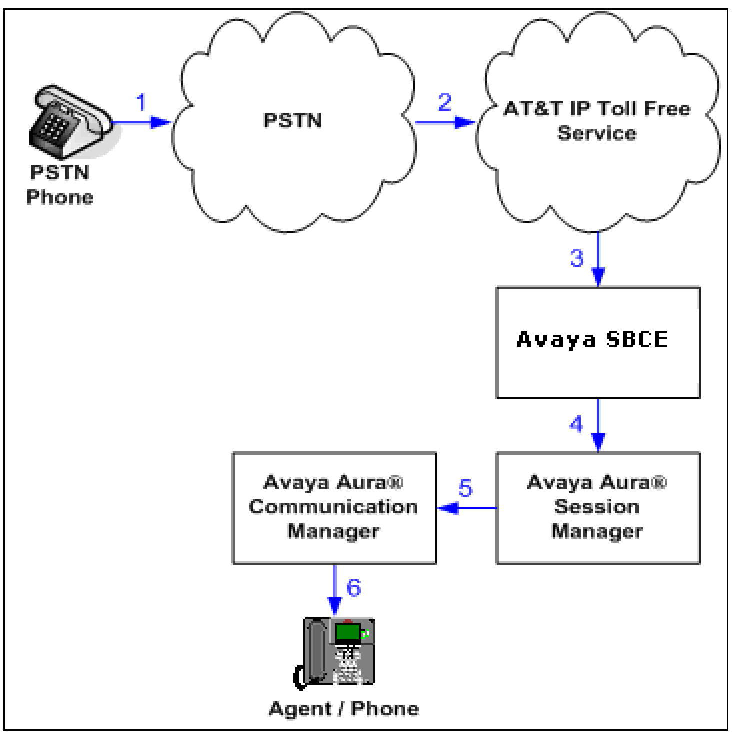


Figure 2: Inbound AT&T IP Toll Free Service Call to an Agent queue/telephone

Note: The IPTF service features such as Legacy Transfer Connect and Alternate Destination Routing utilize this call flow as well.

4 Equipment and Software Validated

The following equipment and software was used for the reference configuration described in these Application Notes.

Equipment/Software	Release/Version
Avaya Aura® System Manager	• 8.0.0.0.931077
Avaya Aura® Session Manager	• 8.0.0.0.800035
Avaya Aura® Communication Manager	• 8.0.0.0-R018x.00.0.822.0
Avaya Aura® Media Server	• 8.0.0.6 • 8.0.0.117
Avaya Aura® Messaging	• 7.0-00.0.441.0-017_0004 (SP 0)
Avaya G450 Media Gateway	• 40.10.0
Avaya Session Border Controller for Enterprise	• 7.2.2.0-07-14883
Avaya 96x1 IP Telephones	• H.323 Version 6.6604 • SIP Version 7.1.3.0.11
Avaya J100 Series IP Telephone	• 3.0.0.2.2
Ventafax Home Version (Windows based Fax device)	• 7.9.255.613

Table 2: Equipment and Software Versions

5 Configure Avaya Aura® Session Manager

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

This section provides the procedures for configuring Session Manager to receive calls from and route calls to the SIP trunk between Communication Manager and Session Manager, and the SIP trunk between Session Manager and the Avaya SBCE. In addition, provisioning for calls to Aura® Messaging are described.

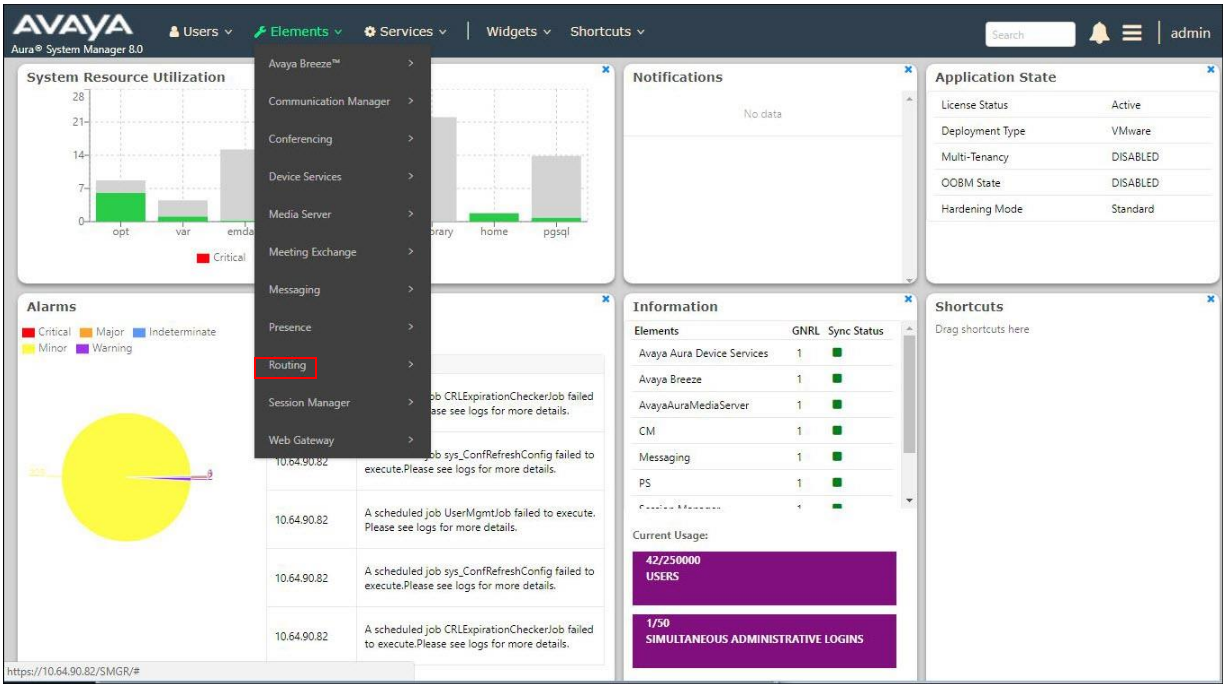
Session Manager serves as a central point for supporting SIP-based communication services in an enterprise. Session Manager connects and normalizes disparate SIP network components and provides a central point for external SIP trunking to the PSTN. The various SIP network components are represented as SIP Entities and the connections/trunks between Session Manager and those components are represented as Entity Links.

When calls arrive at Session Manager from a SIP Entity, Session Manager applies SIP protocol and numbering modifications to the calls. These modifications, referred to as Adaptations, are sometimes necessary to resolve SIP protocol differences between disparate SIP Entities, and also serve the purpose of normalizing the calls to a common or uniform numbering format, which allows for simpler administration of routing rules in Session Manager. Session Manager then matches the calls against certain criteria embodied in profiles termed Dial Patterns and determines the destination SIP Entities based on Routing Policies specified in the matching Dial Patterns. Lastly, before the calls are routed to the respective destinations, Session Manager again applies Adaptations in order to bring the calls into conformance with the SIP protocol interpretation and numbering formats expected by the destination SIP Entities.

The following administration activities will be described:

- Define a SIP Domain
- Define Locations
- Configure the Adaptation Modules that will be associated with digit manipulations for calls between the SIP Entities for Communication Manager, and the Avaya SBCE
- Define SIP Entities corresponding to Communication Manager, and the Avaya SBCE
- Define Entity Links describing the SIP trunk between Communication Manager and Session Manager, and the SIP Trunk between Session Manager and the Avaya SBCE
- Define Routing Policies associated with the Communication Manager, and the Avaya SBCE
- Define Dial Patterns, which govern which routing policy will be selected for call routing

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** (not shown). 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** to save (not shown).

5.2 Locations

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

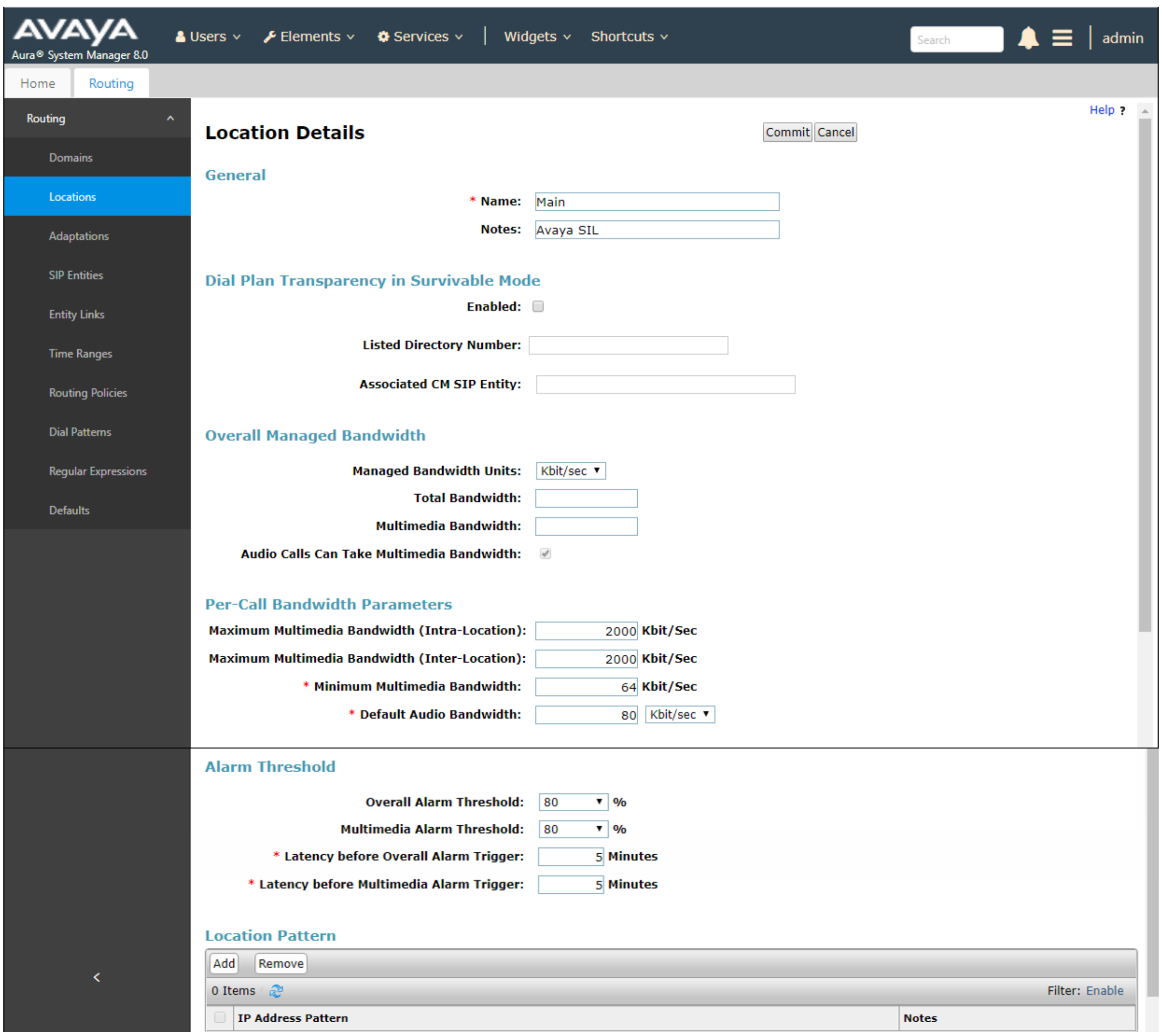
- **Main** – The customer site containing System Manager, Session Manager, Communication Manager, the G450 Media Gateway, and telephones.
- **Common** – This site contains the Avaya SBCE.

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.

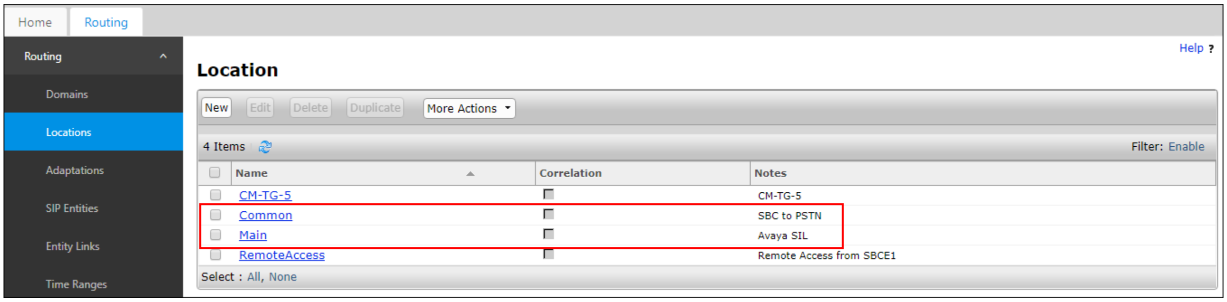
Step **2** - Click **Commit** to save.



5.2.2 Common Location

Follow the steps from **Section 5.2.1** with the following changes:

- **Name**: Enter a descriptive name for the Location (e.g., **Common**).



5.3 Configure Adaptations

Session Manager can be configured to use Adaptation Modules to convert SIP headers sent from AT&T to Communication Manager.

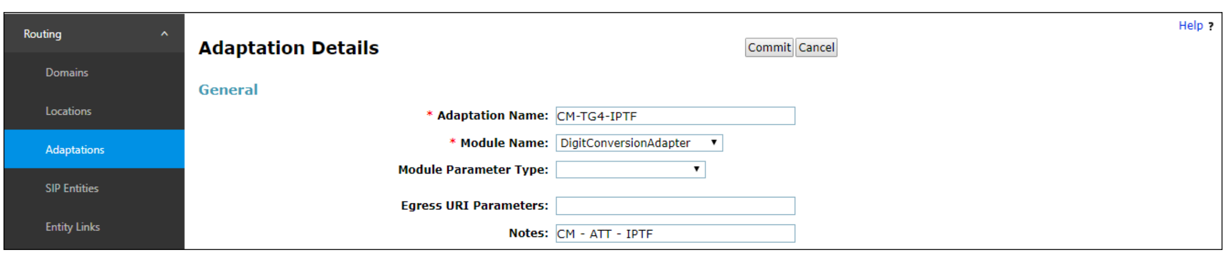
- Calls from AT&T - Modification of SIP messages sent to Communication Manager extensions.
- The AT&T called number digit string in the Request URI is replaced with the associated Communication Manager extensions defined for Agent skill queue VDNs/telephones.

5.3.1 Adaptation for Avaya Aura® Communication Manager Extensions

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:

- A descriptive **Name**, (e.g., **CM-TG4-IPTF**).
- Select **DigitConversionAdapter** from the **Module Name** drop down menu (if no module name is present, select **<click to add module>** and enter **DigitConversionAdapter**).



Step 3 - Scroll down to the **Digit Conversion for Outgoing Calls from SM** section (the *inbound* digits from AT&T that need to be replaced with their associated Communication Manager extensions before being sent to Communication Manager). 0000011041 is a DNIS string sent in the Request URI by the IPTF service that is associated with Communication Manager Agent/VDN skill queue 71041.

- Enter **0000011041** in the **Matching Pattern** column.
- Enter **10** in the **Min/Max** columns.
- Enter **10** in the **Delete Digits** column.
- Enter **7** in the **Insert Digits** column to convert the number to 71041, a Vector Directory Number (VDN) in Communication Manager.
- 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 IPTF DNIS numbers.
Step 5 - Click on **Commit** (not shown).

Digit Conversion for Outgoing Calls from SM

Add | Remove

5 Items — Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* 0000011041	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion
	* 0000021042	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion
	* 0000031043	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion
	* 0000041044	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion
	* 0000051045	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion

Select : All, None

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

5.3.2 Adaptation for the AT&T IP Toll Free Service

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

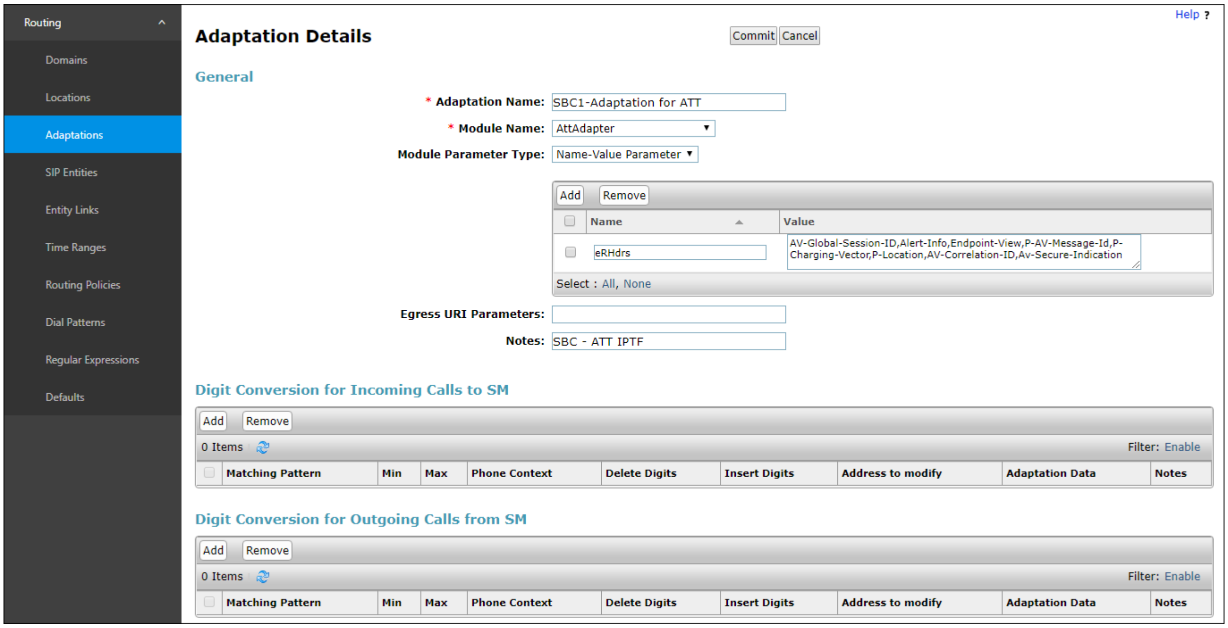
Step 1 - In the **Adaptation Details** page, enter:

1. A descriptive **Name**, (e.g., **SBC1-Adaptation for ATT**).
2. Select **AttAdapter** from the **Module Name** drop down menu (if no module name is present, select **<click to add module>** and enter **AttAdapter**). The AttAdapter will automatically remove History-Info headers, (which the IPFR-EF service does not support), sent by Communication Manager (see **Section 6.8.1**).

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**
2. **Value** – Enter the following Avaya headers to be removed by Session Manager. Note that each header name is separated by a comma with no spaces in between. If spaces are used after the comma, the string needs to be enclosed in quotes.
   - **AV-Global-Session-ID,Alert-Info, Endpoint-View,P-AV-Message-Id,P-Charging-Vector,P-Location,Av-Secure-Indication**

Note – As shown in the screen below, no Incoming or Outgoing Digit Conversion was required in the reference configuration.



5.4 SIP Entities

Note – The **Entity Links** section of these forms (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.

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

- Session Manager (**Section 5.4.1**). Note that this Entity is normally created during Session Manager installation but is shown here for completeness.
- Communication Manager for AT&T access (**Section 5.4.2**) – This entity, and its associated Entity Link (using TLS with port 5064, is for calls from the IPTF service to Communication Manager via the Avaya SBCE.
- Communication Manager for local access (**Section 5.4.3**) – This entity, and its associated Entity Link (using TLS with port 5061), is primarily used for traffic between Avaya SIP telephones and Communication Manager.
- Avaya SBCE (**Section 5.4.4**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls from the IPTF service via the Avaya SBCE.

5.4.1 Avaya Aura® Session Manager SIP Entity

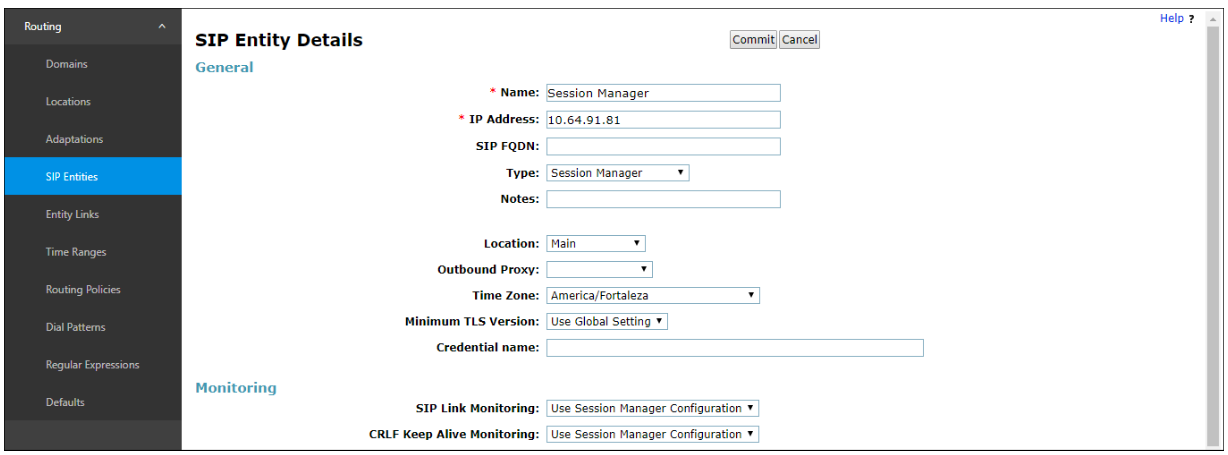
Step 1 - In the left pane under **Routing**, click on **SIP Entities**. In the **SIP Entities** page click on **New** (not shown).

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

- **Name** – Enter a descriptive name (e.g., **Session Manager**).
- **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 **Use Global Setting**. In the reference configuration, the Session Manager Global Setting TLS version is 1.0 (not shown).

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

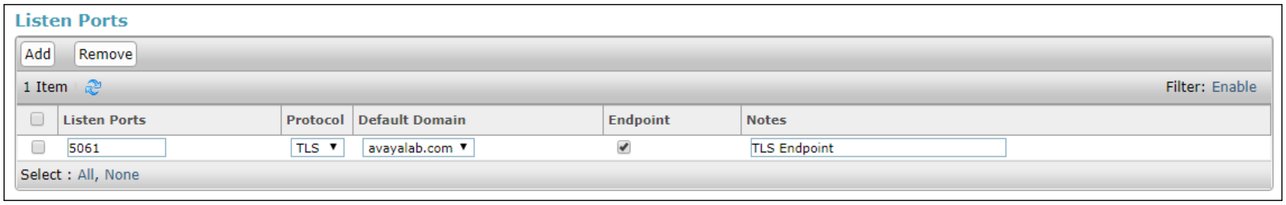
- Select **Use Session Manager Configuration** for **SIP Link Monitoring** field.
- Use the default values for the remaining parameters.



Step 4 - Scrolling down to the **Port** section of the **SIP Entity Details** page, click on **Add** and provision entries as follow:

- **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 SIP telephones. These are separate from the ports defined for the Entity Links in **Section 5.5**.
Step 6 - Enter any notes as desired and leave all other fields on the page blank/default.
Step 7 - Click on **Commit**.



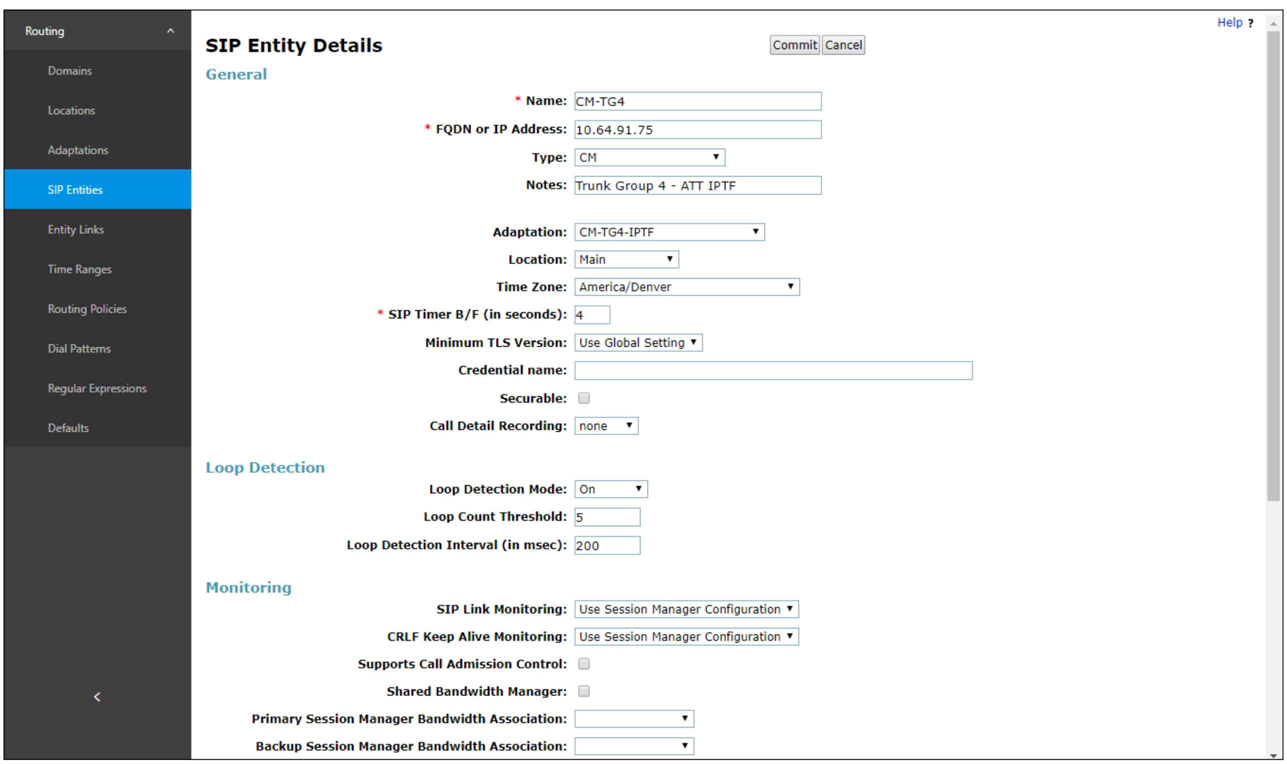
5.4.2 Avaya Aura® Communication Manager SIP Entity – Public Trunk

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-TG4**).
- **FQDN or IP Address –** Enter the IP address of Communication Manager Processor Ethernet (procr) described in **Sections 6.4** and **6.5** (e.g., **10.64.91.75**).
- **Type –** Select **CM**.
- **Adaptation –** Select the Adaptation **CM-TG4-IPTF** 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.

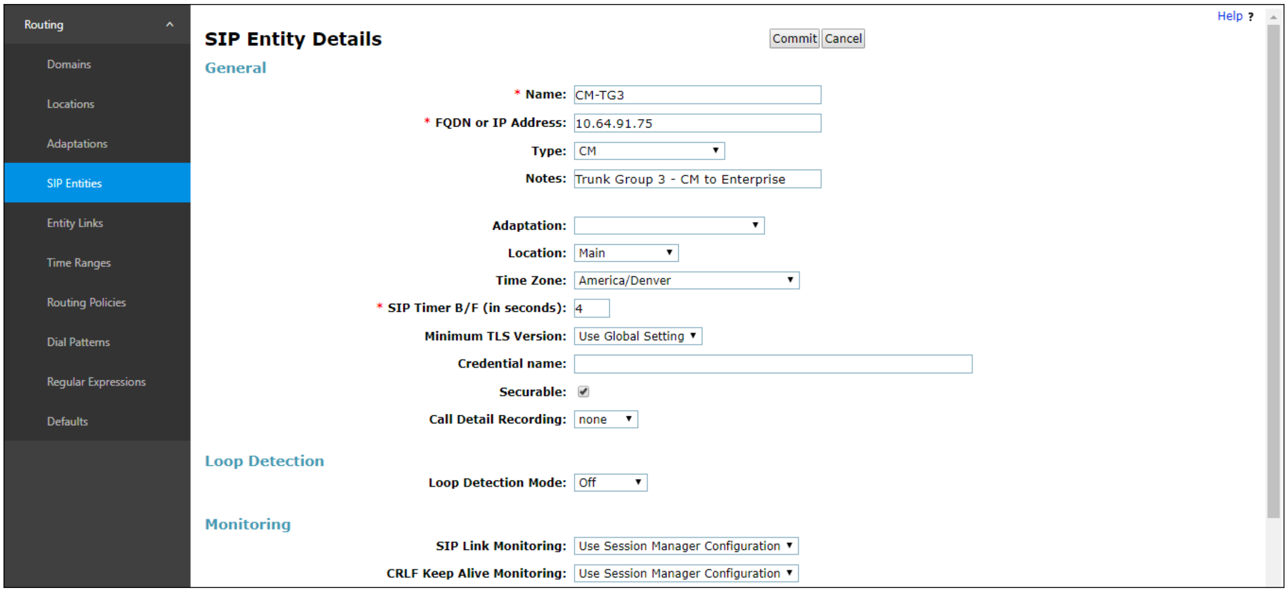
Step 3 - Click on **Commit**.



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:

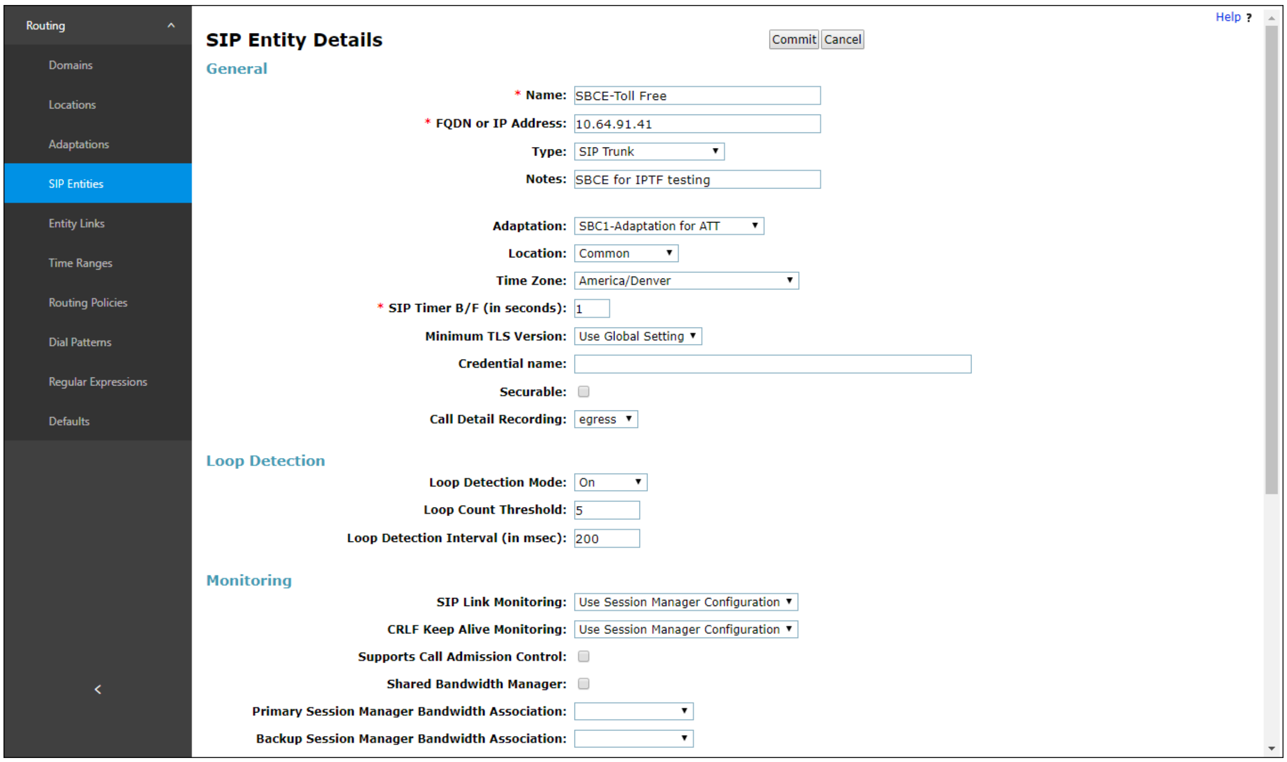
- **Name** – Enter a descriptive name (e.g., **CM-TG3**).
- Note that this Entity has no Adaptation defined.



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., **SBCE-Toll Free**).
- **FQDN or IP Address –** Enter the IP address of the A1 (private) interface of the Avaya SBCE (e.g., **10.64.91.41**, see **Section 7.5.1**).
- **Type –** Verify **SIP Trunk** is selected.
- **Adaptations –** Select Adaptation **SBC1-Adaptation for ATT** (**Section 5.3.2**).
- **Location** – Select location **Common** (**Section 5.2.2**).



5.5 Entity Links

In this section, Entity Links are administered between Session Manager and the following SIP Entities:

- Avaya Aura® Communication Manager – Public (**Section 5.5.1**).
- Avaya Aura® Communication Manager – Local (**Section 5.5.2**).
- Avaya SBCE (**Section 5.5.3**).

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**.

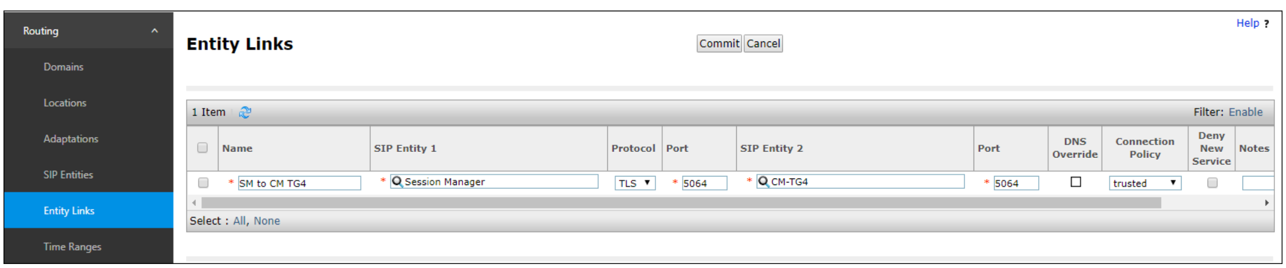
Note – See the information in **Section 5.4** regarding the transport protocols and ports used in the reference configuration.

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 TG4**).
- **SIP Entity 1** – Select the SIP Entity administered in **Section 5.4.1** for Session Manager (e.g., **Session Manager**).
- **SIP Entity 1 Port** – Enter **5064**.
- **Protocol –** Select **TLS** (see **Section 6.8.1**). **SIP Entity 2** –Select the SIP Entity administered in **Section 5.4.2** for the Communication Manager public entity (e.g., **CM-TG4**).
- **SIP Entity 2 Port –** Enter **5064** (see **Section 6.8.1**).
- **Connection Policy –** Select **trusted**.

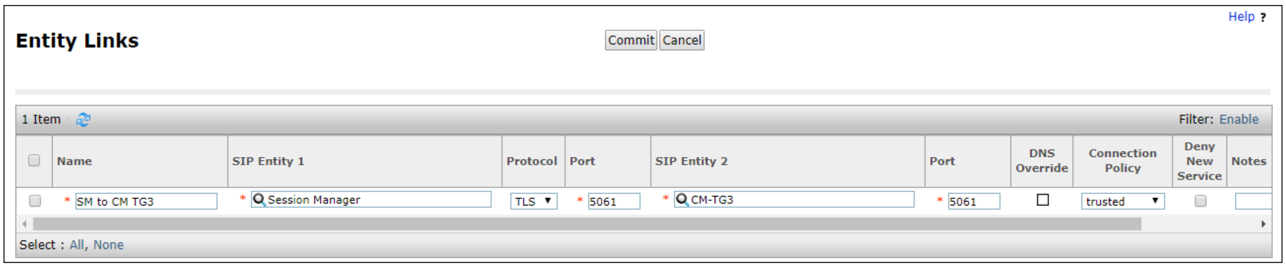
Step 3 - Click on **Commit**.



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:

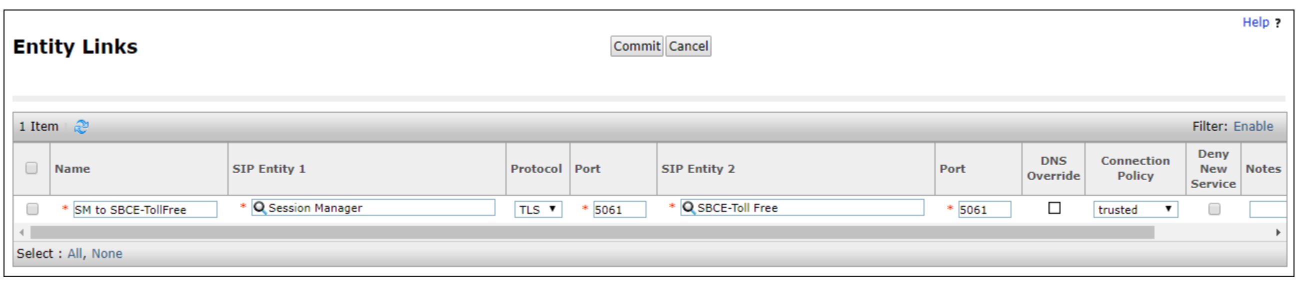
- **Name** – Enter a descriptive name for this link to Communication Manager (e.g., **SM to CM TG3**).
- **Protocol** – Select **TLS**.
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** –Select the SIP Entity administered in **Section 5.4.3** for the Communication Manager local entity (e.g., **CM-TG3**).
- **SIP Entity 2 Port** – Enter **5061** (see **Section 6.8.2**).



5.5.3 Entity Link for the AT&T IP Toll Free 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 SBCE-TollFree**).
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.4** for the Avaya SBCE entity (e.g., **SBCE-Toll Free**).

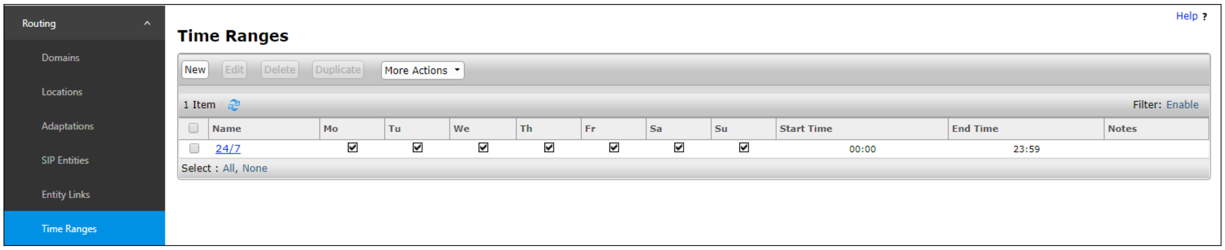


5.6 Time Ranges

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

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**. 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.

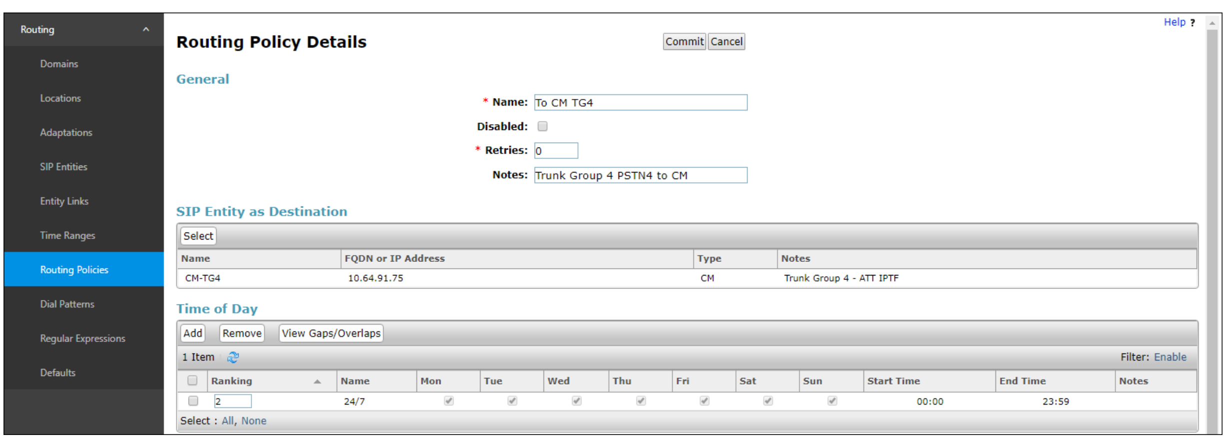
5.7.1 Routing Policy for AT&T Routing to Avaya Aura® Communication Manager

This Routing Policy is used for inbound calls from IPTF.

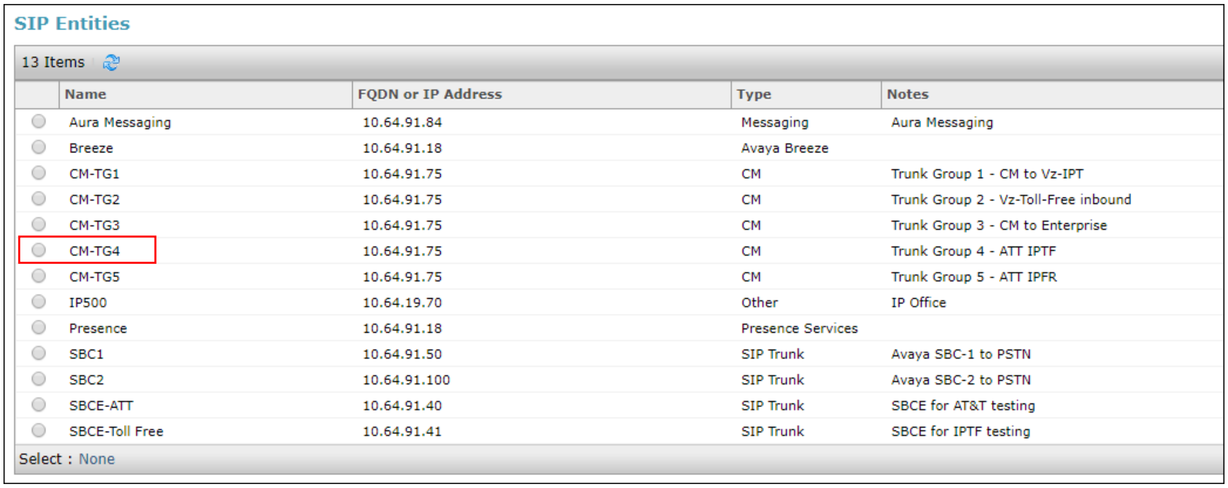
Step 1 - In the left pane under **Routing**, click on **Routing Policies**. In the **Routing Policies** page click on **New** (not shown).

Step 2 - In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing AT&T calls to Communication Manager (e.g., **To CM TG4**), 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 Entity 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-TG4**), and click on **Select**.



SIP Entities

13 Items

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
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

Select : None

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

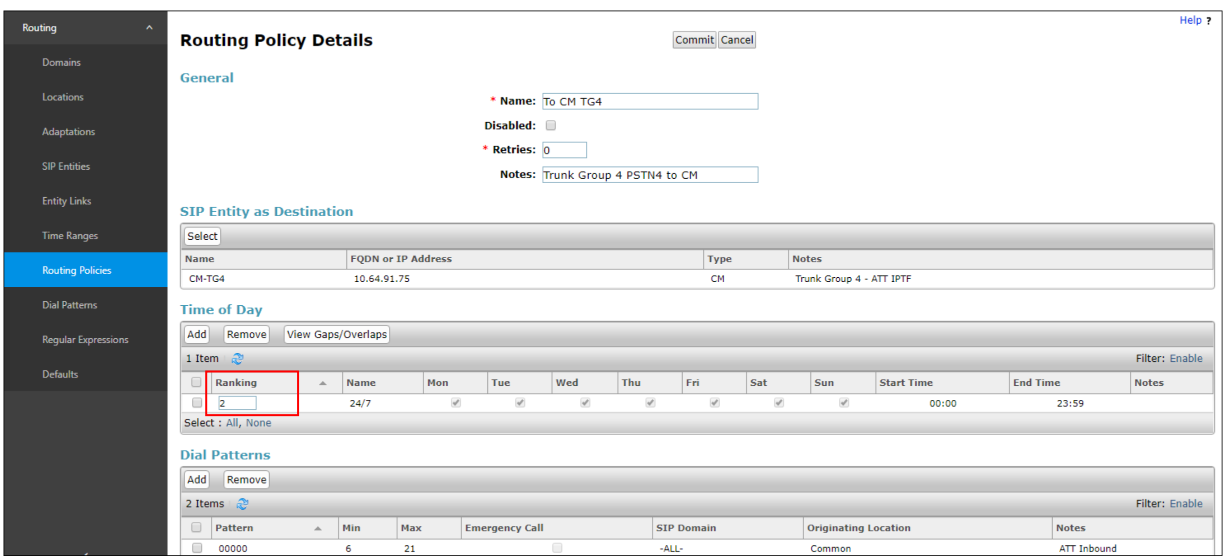
Step 6 - In the **Time Range List** page (not shown), check the checkbox(s) corresponding to one or more Time Ranges administered in **Section 5.6**, and click on **Select**.

Step 7 - Returning to the **Routing Policy Details** page in the **Time of Day** section, enter a **Ranking** of **2**, and click on **Commit**.

Step 8 - Note that once the **Dial Patterns** are defined (**Section 5.8**) they will appear in the **Dial Pattern** section of this form.

Step 9 - No **Regular Expressions** were used in the reference configuration.

Step 10 - Click on **Commit**.



5.8 Dial Patterns

In this section, Dial Patterns are administered to match inbound PSTN calls via the IPTF service to Communication Manager. In the reference configuration inbound calls from the IPTF service sent 15 digits in the SIP Request URI. This pattern must be matched for further call processing.

Note – Be sure to match on the digit string specified in the AT&T Request URI, not the digit string that is dialed. They may be different.

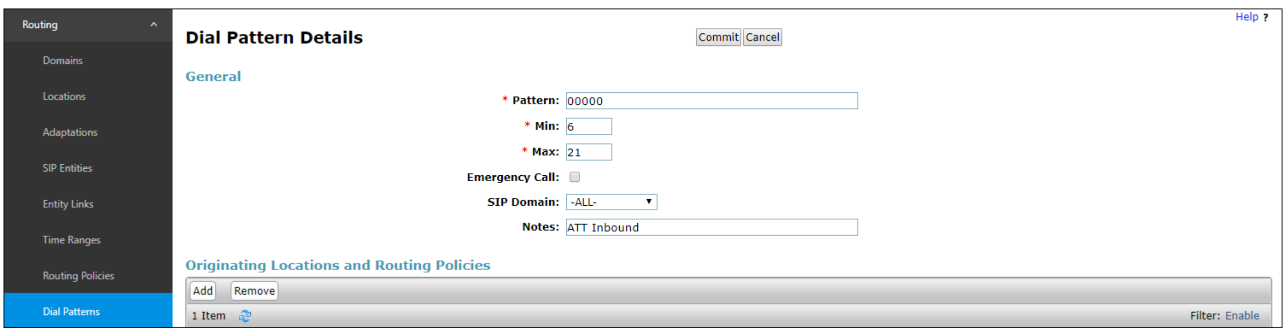
Step 1 - In the left pane under **Routing**, click on **Dial Patterns**. In the **Dial Patterns** page click on **New** (not shown).

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

- **Pattern** – In the reference configuration, AT&T sends a 10-digit number in the Request URI with the format 00000xxxxx. Enter **00000**.

Note – The Adaptation defined for Communication Manager in **Section 5.3.1** will convert the various 00000xxxxx numbers into their corresponding Communication Manager extensions.

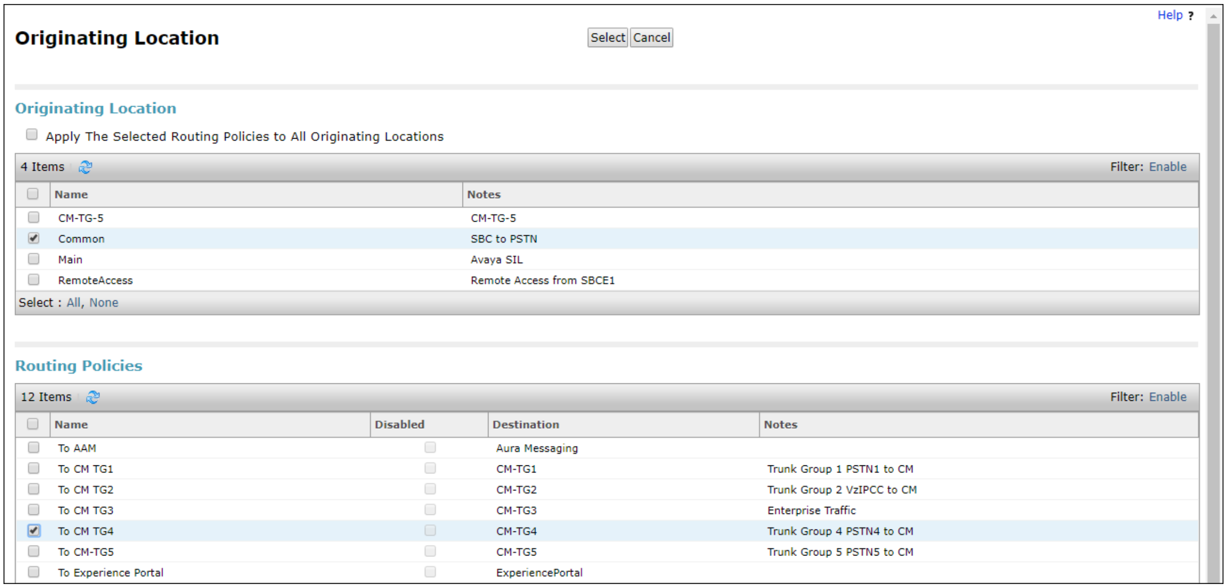
- **Min** – Enter **6**.
- **Max** – Enter **21**.
- **SIP Domain** – Select **-ALL-**, to select all of the administered SIP Domains.



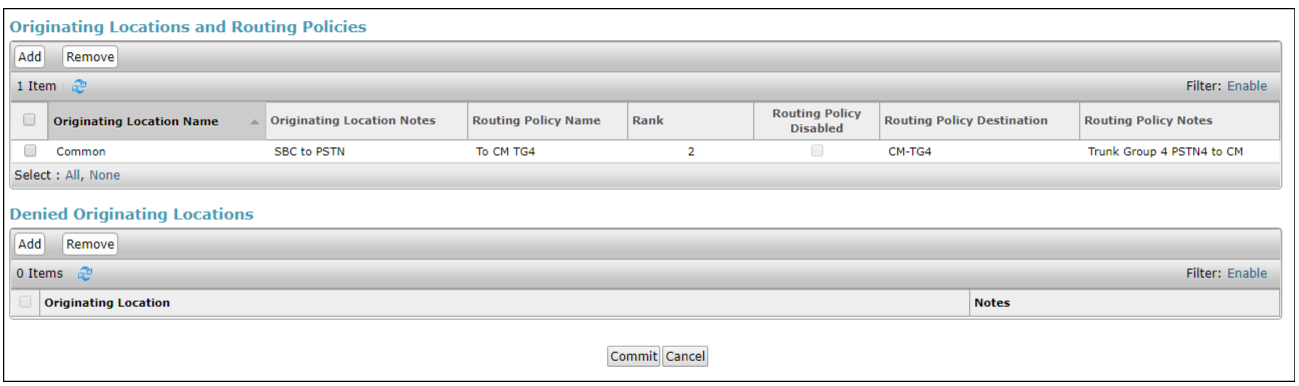
Step 3 - Scrolling down to the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page, click on **Add**.

Step 4 - In the **Originating Location** section of the **Originating Locations and Routing Policies** page, check the checkbox corresponding to the location assigned to the Avaya SBCE in **Section 5.4.4** (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** (e.g., **To CM TG4**). Click on **Select** (not shown).



Step 6 - Returning to the Dial Pattern Details page click on **Commit**.



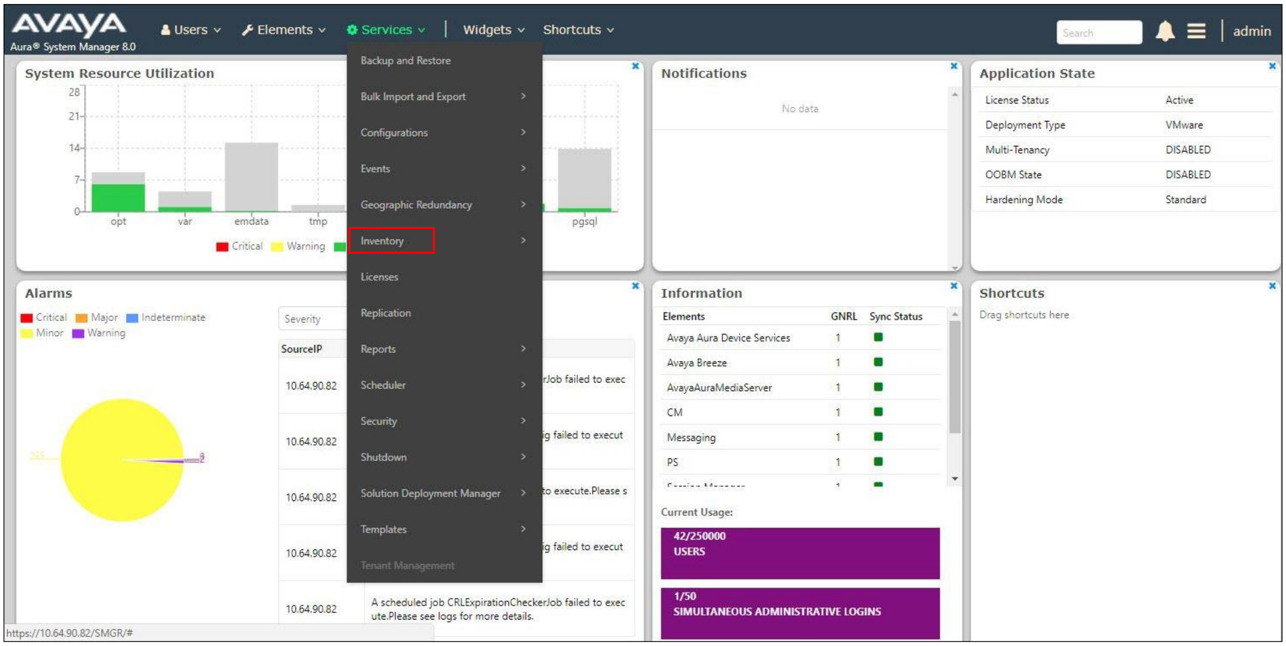
Step 7 - Repeat **Steps 1-6** for any additional inbound dial patterns from AT&T.

5.9 Verify TLS Certificates – Session Manager

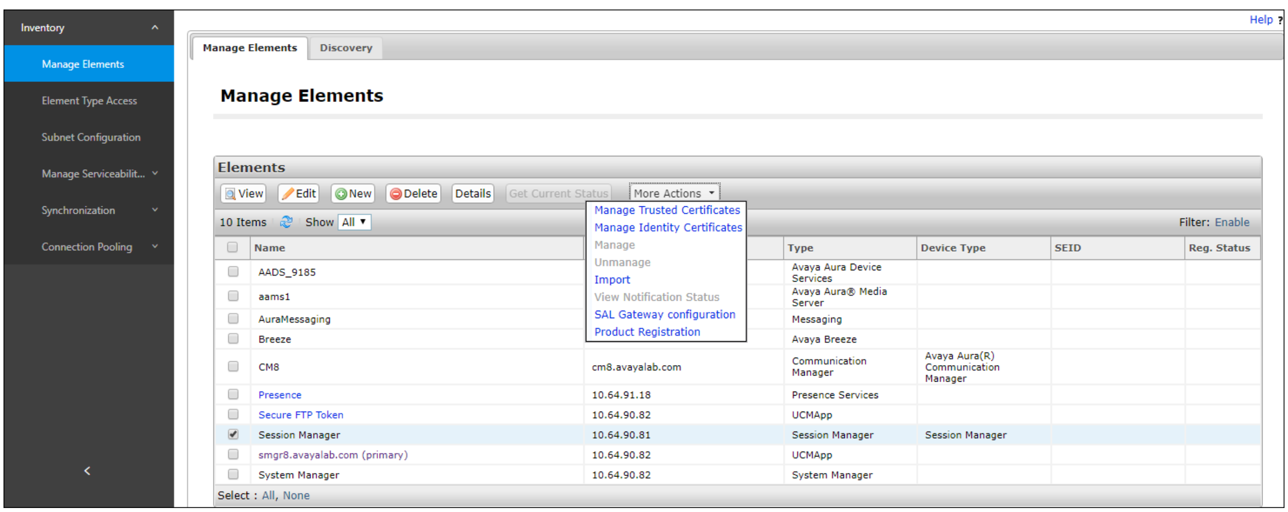
Note – Testing was done with System Manager signed identity certificates. The procedure to obtain and install certificates is outside the scope of these Application Notes.

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

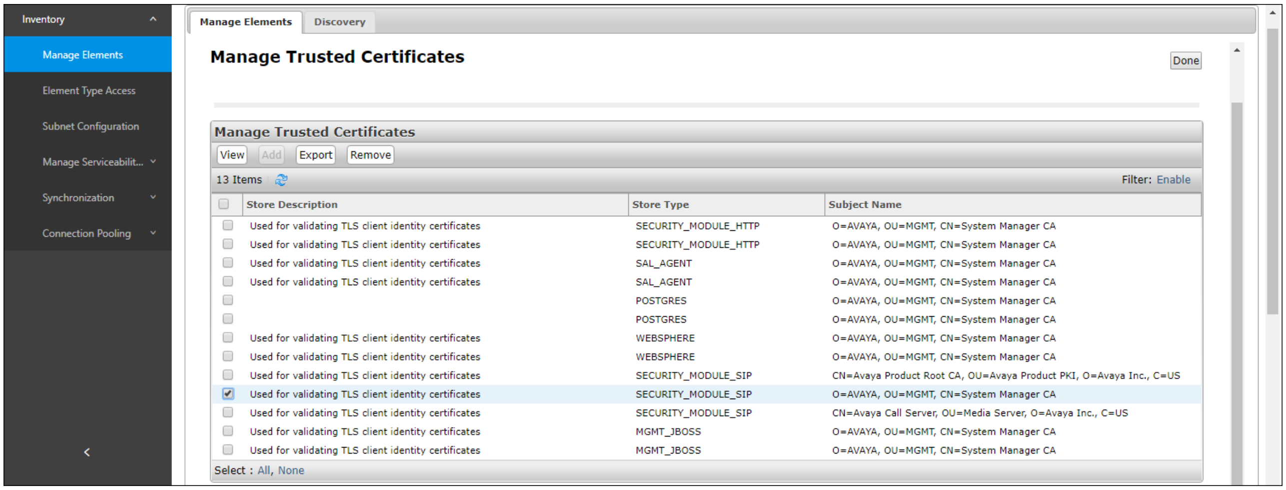
Step 1 - From the **Home** screen, under the **Services** heading, select **Inventory**.



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 → Configure Trusted Certificates**.

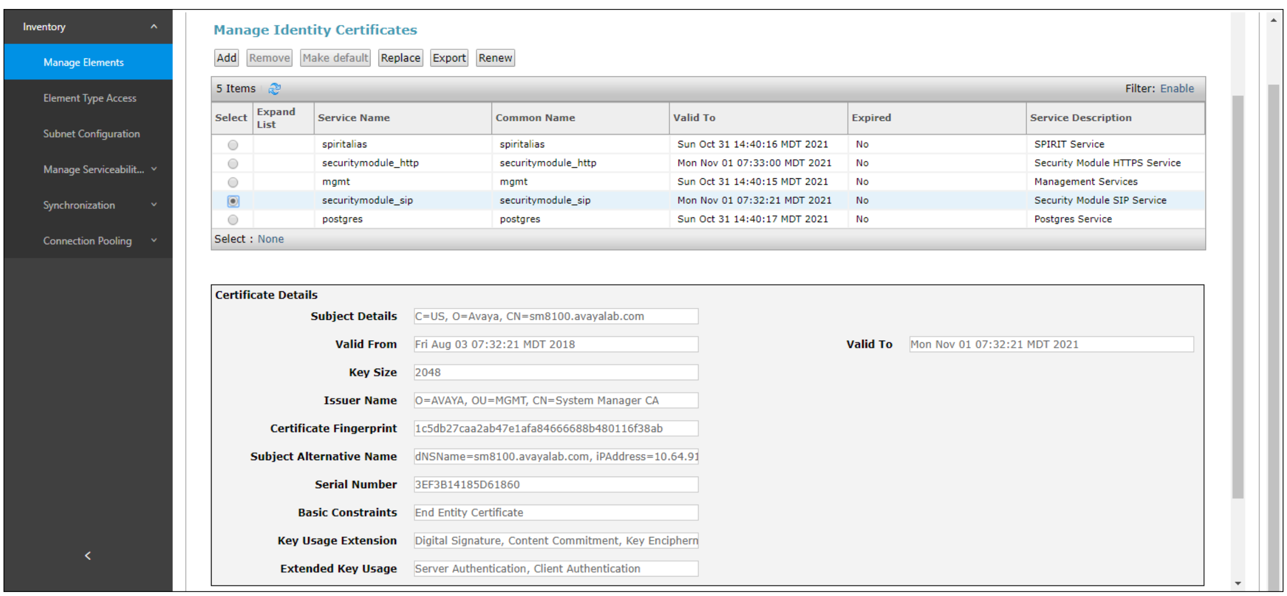


Step 3 - Verify the System Manager Certificate Authority certificate is listed in the trusted store, **SECURITY_MODULE_SIP**. Click **Done** to return to the previous screen.



Step 4 - With Session Manager selected, click on **More Actions → Configure 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**.



6 Avaya Aura® Communication Manager

This section describes the administration steps for Communication Manager in support of the reference configuration described in these Application Notes. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager administration has already been performed. Consult **[5]** and **[6]** for further details if necessary.

Note – In the following sections, only the parameters that are highlighted in **bold** text are applicable to these application notes. Other parameter values may or may not match based on local configurations.

6.1 System-Parameters Customer-Options

This section reviews the Communication Manager licenses and features that are required for the reference configuration described in these Application Notes.

NOTE - For any required features that cannot be enabled in the steps that follow, contact an authorized Avaya account representative to obtain the necessary licenses.

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  2
              Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
               Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 2400  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 5** of the form, verify that the **Media Encryption Over IP** field is set to **y**.

```
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
```

Step 2 - 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? n               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


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

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 IP 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 IP Interface for procr

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.

```
display 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 Network Regions

Network Regions are used to group various Communication Manager resources such as codecs, UDP port ranges, and inter-region communication. In the reference configuration, two network regions are used. Region 1 for the CPE access, and region 4 for SIP trunk access.

6.6.1 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.
- **UDP Port Min**: – Set to **16384** (**AT&T requirement**).
- **UDP Port Max**: – Set to **32767** (**AT&T requirement**).

Note – The port range for Region 1 does not have to be in the range required by AT&T. However, the same range was used here in the reference configuration.

```
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: 16384                     IP Audio Hairpinning? n
   UDP Port Max: 32767
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 **4** in the **dst rgn** column, enter **4** for the codec set (this means region 1 is permitted to talk to region 4 and it will use codec set 4 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
 3    1     y    NoLimit                                             n       t
 4    4     y    NoLimit                                             n       t
```

6.6.2 IP Network Region 4 – SIP Trunk Region

Repeat the steps in **Section 6.6.1** with the following changes:

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

- Enter a descriptive name (e.g., **AT&T**).
- Enter **4** for the **Codec Set** parameter.

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

- Set codec set **4** for **dst rgn 1**.
- Note that **dst rgn 4** is pre-populated with codec set **4** (from page 1 provisioning).

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

 Source Region: 4     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    4     y    NoLimit                                             n       t
 2    4     y    NoLimit                                             n       t
 3    3     y    NoLimit                                             n       t
 4    4                                                                all
```

Note – Region 3 was created to test G.711 pass-through fax (not shown), and is permitted to talk to region 4 using codec set 3.

6.7 IP Codec Parameters

Note – The IPTF service offers G.729A, G.726-32, and G.711MU codecs in their Invite SDP. G.726-32 codec is supported by Communication Manager, but testing found issues when G.726-32 codec is used (see **Section 2.2**, **item 3**). In addition, some calls could require support of G.729B (silence suppression). Therefore G.729B is also included in the codec lists.

6.7.1 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**, **G.729A**, and **G.729B** are included in the codec list. Note that the packet interval size will default to 20ms. Under **Media Encryption**, ensure **1-srtp-aescm128-hmac80** is included to support Secure Real-time Transport Protocol (SRTP).

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

                          IP CODEC SET
    Codec Set: 1

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

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

                                                                      Packet
                          Mode                    Redundancy          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
```

6.7.2 Codecs for IP Network Region 4 (calls from AT&T)

Step 1 - Repeat the steps in **Section 6.8.0** with the following changes.

- Provision the codecs in the order shown below. Note that the order of G.729A and G.729B codecs may be reversed as required.
- Set **Frames Per Pkt** to **3**. This will auto-populate **30** for the **Packet Size (ms)** field, and specify a PTIME value of 30 in the SDP (recommended by AT&T). See **Section 2.2**, **Item 10** for limitations with the packet size.

```
change ip-codec-set 4                                          Page   1 of   2

                          IP CODEC SET
    Codec Set: 4

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729A           n          3        30
 2: G.729B           n          3        30
 3: G.711MU          n          3        30

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

change ip-codec-set 4                                          Page   2 of   2
                          IP CODEC SET
                              Allow Direct-IP Multimedia? n
                                                                        Packet
                          Mode                   Redundancy             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
```

6.7.3 Codecs for G.711 Pass-Through Fax

During G.711 pass-through fax testing, the network region assigned to the G450 Media Gateway was changed from region 1 to region 3 (**Section 6.14**). This network region utilized **ip-codec-set 3** for calls between region 3 and region 4 (IPTF calls). This codec set is shown here for completeness and is only needed if G.711 pass-through is preferred to T.38 fax. See **Section 2.2** for limitations. For this codec set, **G.711MU** is listed as the preferred codec, and on **Page 2**, the **Fax Mode** is set to **off**. Creating a dedicated network region and ip-codec-set for G.711 pass-through fax allowed for fax calls from this G450 Media Gateway to begin with G.711MU, while voice calls to other Media Gateways, Media Servers, and IP endpoints belonging to region 1, will continue to request G.729A as the first codec choice (**Section 6.7.2**).

```
change ip-codec-set 3                                          Page   1 of   2

                          IP CODEC SET
    Codec Set: 3
    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         3        30
 2: G.729A            n         3        30
 3: G.729B            n         3        30

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

change ip-codec-set 3                                          Page   2 of   2
                          IP CODEC SET
                              Allow Direct-IP Multimedia? n
                                                                       Packet
                          Mode                  Redundancy             Size(ms)
    FAX                   off                    0
    Modem                 off                    0
    TDD/TTY               US                     3
    H.323 Clear-channel   n                      0
```

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 IPTF access – SIP Trunk 4
  - Note that this trunk will use TLS port 5064 as described in **Section 5.5.1**.
- Internal CPE access (e.g., Avaya SIP telephones, etc.) – SIP Trunk 3
  - Note that this trunk will use TLS port 5061 as described in **Section 5.5.2**.

6.8.1 SIP Trunk for Inbound AT&T calls

This section describes the steps for administering the SIP trunk to Session Manager used for inbound IPTF calls. This trunk corresponds to the **CM-TG4** SIP Entity defined in **Section 5.4.2**.

Step 1 - Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **4**), 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 systems 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 **5064**.
- **Far-end Network Region** – Set the IP network region to **4**, 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).
- **Enable Layer 3 Test** – Set to **y**. This directs Communication Manager to send SIP OPTIONS messages to Session Manager to check link status.
- **OPTIONAL**: If desired, set **Initial IP-IP Direct Media** to **y**. Otherwise leave it disable (default).

Note - Enabling the **Initial IP-IP Direct Media** parameter allows Communication Manager to signal the IP address of Avaya SIP telephones during the initial setup of a call. This permits the Avaya SIP telephone and the AT&T caller to exchange media directly, without allocating Communication Manager media resources. However, unless network routing permits direct IP access between the Avaya SIP telephone and the "inside" interface of the Avaya SBCE, a loss of audio can occur when this option is enabled. In addition, when this option is enabled, Communication Manager will not send SDP in 180 messages, and will not send 183 messages (if enabled).

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

```
add signaling-group 4                                           Page   1 of   2
                                SIGNALING GROUP

 Group Number: 4              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: SM
 Near-end Listen Port: 5064                Far-end Listen Port: 5064
                                        Far-end Network Region: 4

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? y
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Step 2 - Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g., **4**). On **Page 1** of the **trunk-group** form, provision the following:

- **Group Type** – Set to **sip**.
- **Group Name** – Enter a descriptive name (e.g., **ATT IPTF**).
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., ***04**).
- **Direction** – Set to **incoming**.
- **Service Type** – Set to **public-ntwrk**.
- **Signaling Group** – Set to the signaling group administered in **Step 1** (e.g., **4**).
- **Number of Members** – Enter the maximum number of simultaneous calls desired on this trunk group (based on licensing) (e.g., **20**).

```
add trunk-group 4                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 4                   Group Type: sip          CDR Reports: y
  Group Name: ATT IPTF                   COR: 1        TN: 1        TAC: *04
   Direction: incoming      Outgoing Display? n
 Dial Access? n                                    Night Service:

Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 4
                                                     Number of Members: 20
```

Step 3 - On **Page 2** of the **Trunk Group** form:

- Set the **Preferred Minimum Session Refresh Interval (sec):** to **900**.

```
add trunk-group 4                                               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 4 - On **Page 3** of the **Trunk Group** form:

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

```
add trunk-group 4                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y

                                                   Hold/Unhold Notifications? y


 Show ANSWERED BY on Display? y
```

Step 5 - On **Page 4** of the **Trunk Group** form:

- Set **Telephone Event Payload Type** to the RTP payload type recommended by the IPTF service (e.g., **100**).

Note – The IPTF service does not support History Info header. As shown below, by default this header is supported by Communication Manager. In the reference configuration, any History Info headers sent by Communication Manager are automatically removed from SIP signaling by Session Manager, as part of the AttAdapter (see **Section 5.3.2**). Alternatively, History Info may be disabled here.

```
add trunk-group 4                                               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? n

                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 100
                                       Shuffling with SDP? n

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

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

6.8.2 Local SIP Trunk (Avaya SIP Telephone Access)

This trunk corresponds to the **CM-TG3** SIP Entity defined in **Section 5.4.3**.

Step 1 - Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **3**), and repeat the steps in **Section 6.8.1** with the following changes:

- **Transport Method** – Set to **tls**.
- **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**.
- **Initial IP-IP Direct Media –** Set to **y**.

```
add signaling-group 3                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 3             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? y        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: SM
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

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? y
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Step 2 - 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, repeat the steps in **Section 6.8.1** with the following changes:

- **Group Name** – Enter a descriptive name (e.g., **To 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 **Step 1** (e.g., **3**).

```
add trunk-group 3                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 3                  Group Type: sip          CDR Reports: y
  Group Name: To SM Enterprise          COR: 1        TN: 1        TAC: *03
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 3
                                                     Number of Members: 10
```

Step 3 - On **Page 2** of the **Trunk Group** form:

- Same as **Section 6.8.1**.

Step 4 - On **Page 3** of the **Trunk Group** form:

- **Same** as **Section 6.8.1**.
- **Step 5** - On **Page 4** of the **Trunk Group** form:

- Use default values for all settings.

```
add trunk-group 3                                               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? 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? n
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n

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

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**), is used to convert Communication Manager local extensions to IPTF DNIS numbers, for inclusion in any SIP headers directed to the IPTF 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 any Communication Manager station extensions and their corresponding IPTF DNIS number (for the public trunk):

- **Ext Len** – Enter the total number of digits in the local extension range (e.g., **5**).
- **Ext Code –** Enter the Communication Manager station extension (e.g., SIP phone **14006**).
- **Trk Grp(s)** – Enter the number of the Public trunk group (e.g., **4**).
- **CPN Prefix** – Enter the corresponding IPTF DNIS number (e.g., **0000011041**).
- **CPN Len** – Enter the total number of digits after the digit conversion (e.g., **10**).

Step 3 - Add any Communication Manager Agent skill VDN extensions and their corresponding IPTF DNIS number (for the public trunk):

- **Ext Len** – Enter the total number of digits in the local extension range (e.g., **5**).
- **Ext Code –** Enter the Communication Manager extension (e.g., Skill VDN **71041**).
- **Trk Grp(s)** – Enter the number of the Public trunk group (e.g., **4**).
- **CPN Prefix** – Enter the corresponding IPTF DNIS number (e.g., **0000011041**).
- **CPN Len** – Enter the total number of digits after the digit conversion (e.g., **15**).

Step 4 - Repeat **Steps 2** and **3** for all IPTF DNIS numbers and their corresponding Communication Manager station, Skill, or Agent extensions.

```
change public-unknown-numbering 5 ext-digits 71041              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext              Trk       CPN             CPN
Len Code             Grp(s)    Prefix          Len
                                                     Total Administered: 20
 5  71041            4         0000011041      15       Maximum Entries: 240
 5  71042            4         0000021042      15
 5  71043            4         0000031043      15    Note: If an entry applies to
 5  71044            4         0000041044      15    a SIP connection to Avaya
                                                     Aura(R) Session Manager,
                                                     the resulting number must
                                                     be a complete E.164 number.

                                                     Communication Manager
                                                     automatically inserts
                                                     a '+' digit in this case.
```

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**), 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., **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  10                3                           5     Total Administered: 6
 5  11                3                           5        Maximum Entries: 540
 5  12                3                           5
 5  14                3                           5
 5  20                3                           5
```

6.11 Route Patterns for Local SIP Trunk

Route Patterns are used to direct calls to the Local SIP trunk for access to SIP phones or other destinations in the CPE. This form specifies the local SIP trunk (e.g., 3), based on the route-pattern selected by the AAR table in **Section 6.12** (e.g., calls SIP phone extensions).

Note – As IPTF is an inbound only service, no outbound route patterns are defined for the public SIP trunk.

Step 1 - Enter the **change route-pattern 3** command and enter the following:

- In the **Grp No** column enter **3** for SIP trunk 3 (local trunk).
- In the **FRL** column enter **0** (zero).
- 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 Alternate Routing (AAR) Dialing

AAR is used to direct calls to the local SIP trunk for Avaya SIP telephones, using the route pattern defined in **Section 6.11**.

Step 1 - Enter the following:

- **Dialed String -** In the reference configuration all SIP telephones used extensions in the range 89xxx, therefore enter **89**.
- **Min** & **Max** – Enter **5**.
- **Route Pattern** – Enter **3**.
- **Call Type** – Enter **lev0**.

```
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
    20                    5    5      3       lev0         n
    89                    5    5      3       lev0         n
```

6.13 Provisioning for Simulated Call Center Functionality

In the reference configuration, a Call Center environment (skill queues and Agents) was simulated on Communication Manager. The administration of Communication Manager Call Center type elements – Agents, skills (hunt groups), vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Consult **[6** and **10]** for further details. The samples that follow are provided for reference purposes only.

- Agent form – **Page 1**

```
display agent-loginID 20001                                 Page   1 of   2
                                 AGENT LOGINID

                Login ID: 20001                                 AAS? n
                    Name: Agent 1                             AUDIX? n
                      TN: 1        Check skill TNs to match agent TN? n
                     COR: 2
           Coverage Path: 1                            LWC Reception: spe
           Security Code:                     LWC Log External Calls? n
           Attribute:                       AUDIX Name for Messaging:

                                        LoginID for ISDN/SIP Display? n
                                                            Password:
                                              Password (enter again):
                                                         Auto Answer: acd
 AUX Agent Remains in LOA Queue: system            MIA Across Skills: system
AUX Agent Considered Idle (MIA): system    ACW Agent Considered Idle: system
             Work Mode on Login: system    Aux Work Reason Code Type: system
                                              Logout Reason Code Type: system
                         Maximum time agent in ACW before logout (sec): system
                                             Forced Agent Logout Time:   :
    WARNING:  Agent must log in again before changes take effect
```

- Agent form – **Page 2**

```
display agent-loginID 20001                                 Page   2 of   2
                                 AGENT LOGINID
      Direct Agent Skill:                             Service Objective? n
Call Handling Preference: skill-level             Local Call Preference? n

    SN   RL SL          SN   RL SL
 1: 1       1       16:
```

- Skill 1 Hunt Group form – **Page 1**

```
display hunt-group 1                                            Page   1 of   4
                                  HUNT GROUP

            Group Number: 1                                  ACD? y
              Group Name: Agent Group                      Queue? y
         Group Extension: 19991                           Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                      MM Early Answer? n
           Security Code:                 Local Agent Preference? n
 ISDN/SIP Caller Display: grp-name

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

- Skill 1 Vector form – **Page 1**

```
display vector 4                                                Page   1 of   6
                                  CALL VECTOR

    Number: 4                Name: Call Center
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 #    Wait hearing ringback
02 wait-time    2   secs hearing ringback
03 #    Play greeting and collect 1 digit
04 collect      1    digits after announcement 11001    for none
05 goto step    7             if digits           =      1
06 stop
07 #    Simple queue to skill with recurring announcement until available
08 queue-to     skill 1    pri m
09 announcement 11004
10 wait-time    30  secs hearing music
11 goto step    8             if unconditionally
12 stop
```

- Skill 1 VDN form – **Page 1**

```
display vdn 71041                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 71041
                                 Name*: ATT Toll-Free 1
                           Destination: Vector Number        4
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none


        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
```

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, etc.

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

Step 1 - SSH 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 note 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 6.4**).
Step 4 - Enter the **copy running-config startup-config** command to save the G450 configuration.
Step 5 - On Communication Manager, enter the **add media-gateway x** command where x is an available Media Gateway identifier (e.g., **1**). The Media Gateway form will open (not shown). Enter the following parameters:

- Set **Type** = **g450**
- Set **Name** = Enter a descriptive name (e.g., **G450**)
- Set **Serial Number** = Enter 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**

When the Media Gateway registers, the SSH connection prompt will change to reflect the Media Gateway Identifier assigned in **Step 5** (e.g., ***G450-001(super)#***).

Step 6 - Enter the **display media-gateway 1** command and verify that the G450 has registered.

```
display media-gateway 1                                         Page   1 of   2
                             MEDIA GATEWAY 10

                    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** – Set to **9061**
- **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: 9061                 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
```

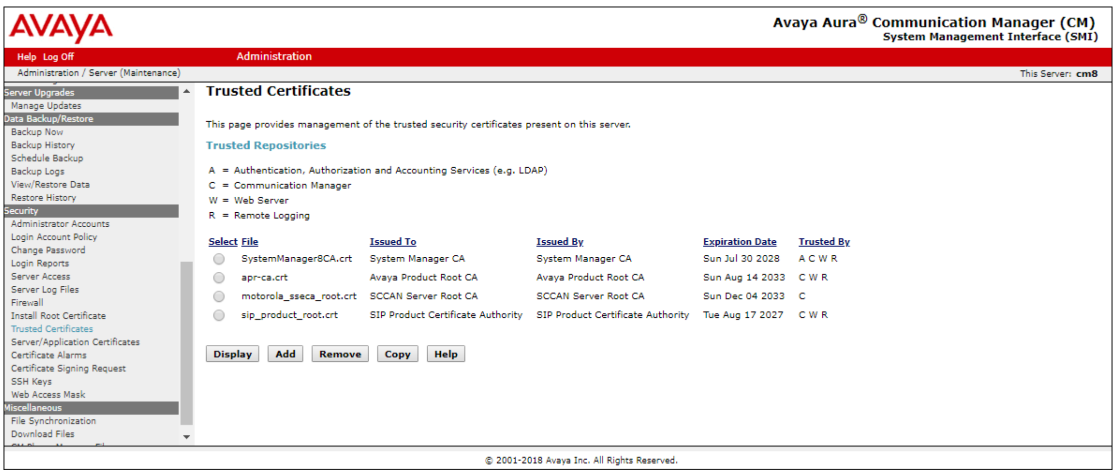
6.16 Verify TLS Certificates – Communication Manager

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.

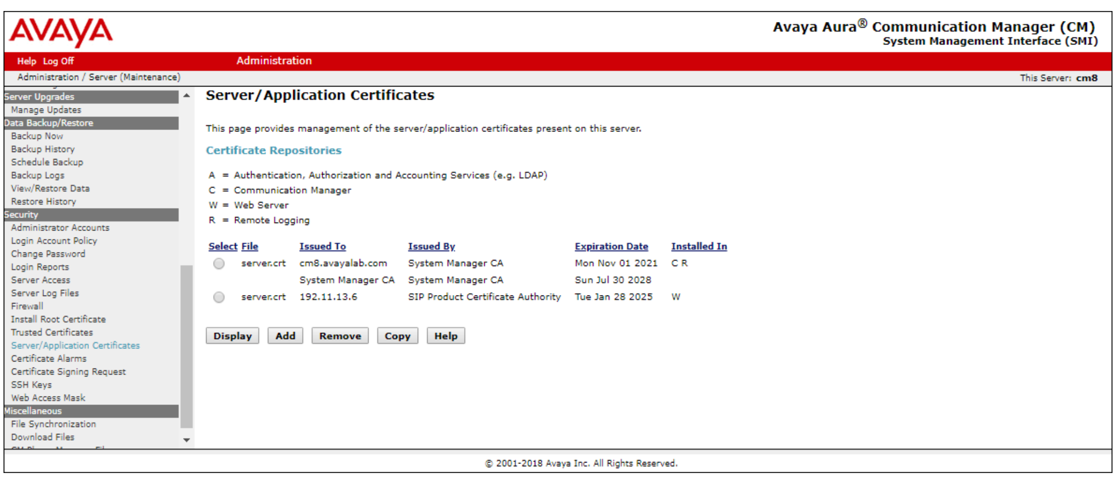
In the reference configuration, TLS transport is used for the communication between Session Manager and Communication Manager. The following procedures show how to verify the certificates used by Communication Manager.

Step 1 - From a web browser, type in "**https:// x.x.x.x**", where x.x.x.x 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.



6.17 Save Communication Manager Translations

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

7 Configure Avaya Session Border Controller for Enterprise

Note: Only the Avaya SBCE provisioning required for the reference configuration is described in these Application Notes.

Note: The installation and initial provisioning of the Avaya SBCE is beyond the scope of this document. Refer to **[11]** and **[12]** for additional information.

IMPORTANT! – During the Avaya SBCE installation, the Management interface of the Avaya SBCE <u>must</u> be provisioned on a different subnet than either of the Avaya SBCE private and public network interfaces (e.g., A1 and B1).

As described in **Section 3**, the reference configuration places the private interface (A1) of the Avaya SBCE in the Common site, (IP address 10.64.91.41), with access to the Main site. The connection to AT&T uses the Avaya SBCE public interface B1 (IP address 192.168.80.43).
The following provisioning is performed via the Avaya SBCE GUI interface, using the "M1" management LAN connection.

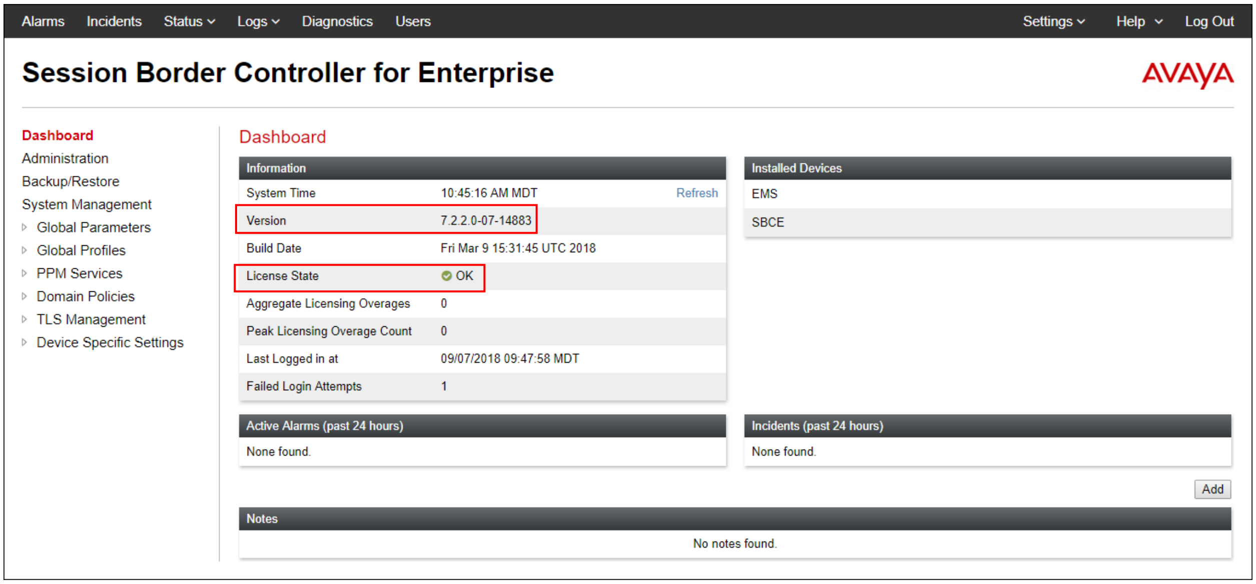
Step 1 - Access the web interface by typing "**https://x.x.x.x**", where x.x.x.x is the management IP address of the Avaya SBCE.

Step 2 - Enter the **Username** and click on **Continue**.

Step 3 - Enter the password and click on **Log In**.

Step 4 - The main menu window will open. 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.

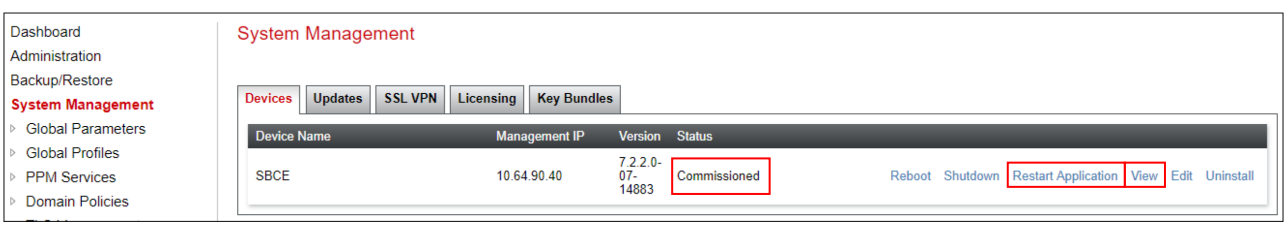
Note – The provisioning described in the following sections use the menu options listed in the left-hand column shown below.



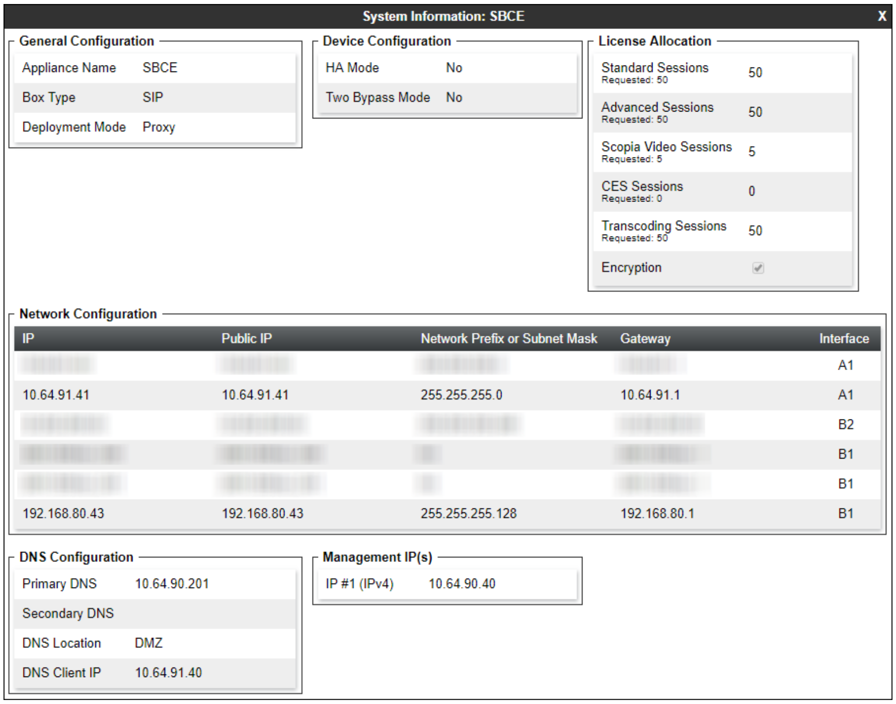
7.1 System Management – Status

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 in the shared test environment.



7.2 TLS Management

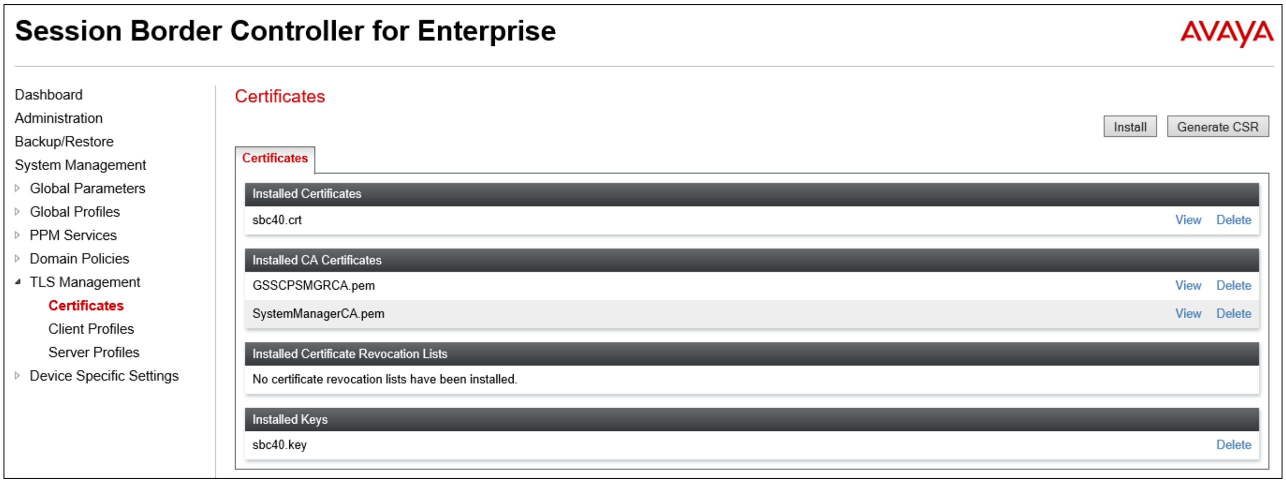
Note – Testing was done with System Manager signed identity certificates. The procedure to create and obtain these certificates is outside the scope of these Application Notes.

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.

7.2.1 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.

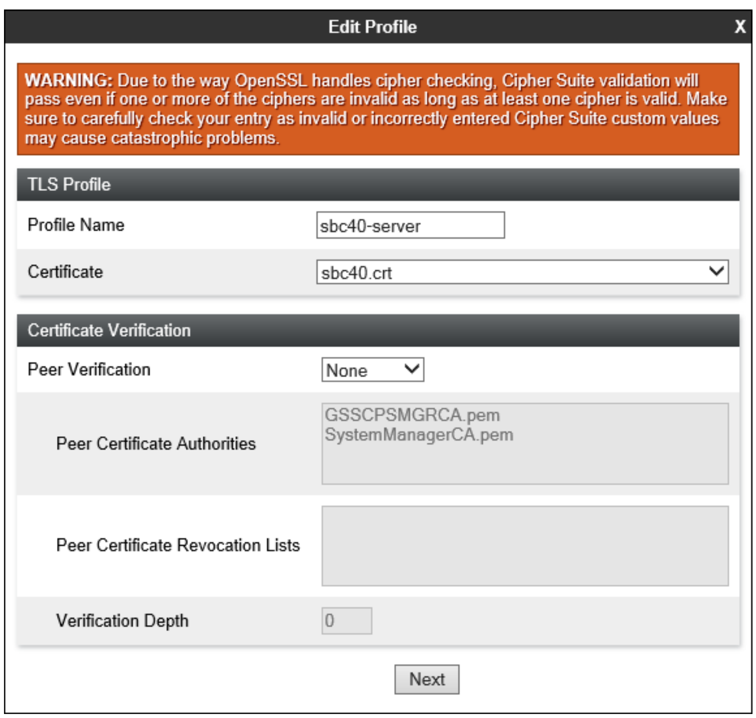


7.2.2 Server Profiles

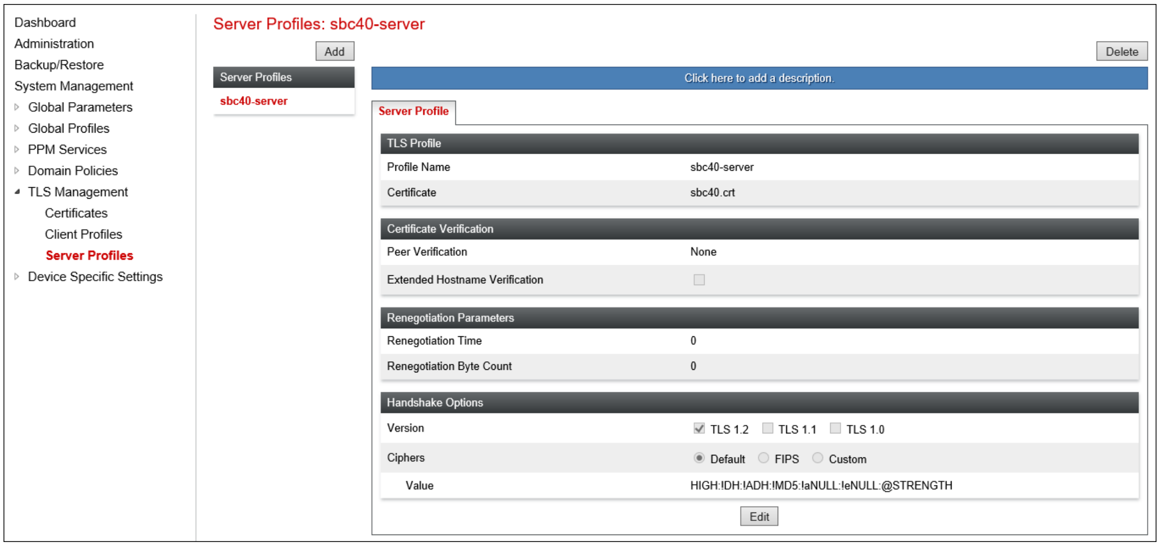
Step 1 - Select **TLS Management → Server Profiles** and click on **Add**. Enter the following:

- **Profile Name:** enter descriptive name.
- **Certificate:** select the identity certificate, e.g., **sbc40.crt**, 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:

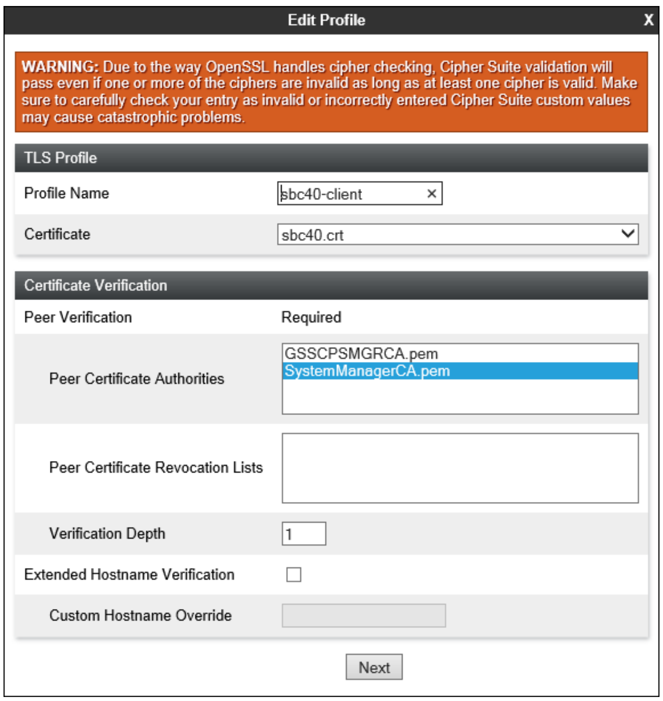


7.2.3 Client Profiles

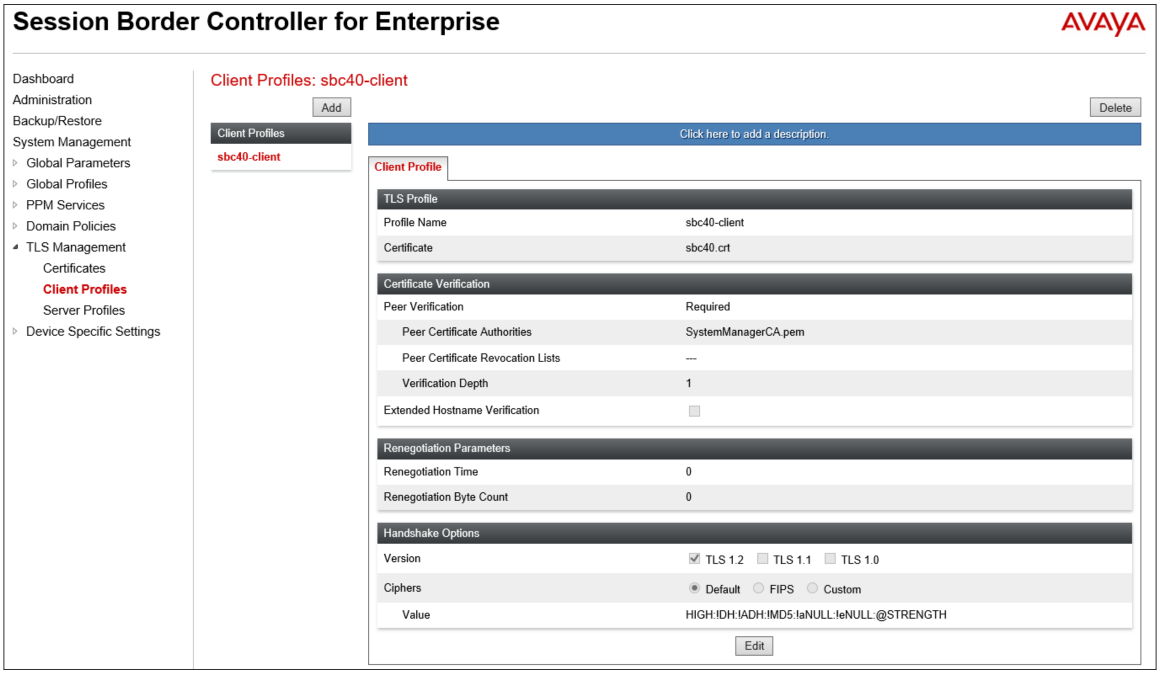
Step 1 - Select **TLS Management → Server Profiles** and click on **Add**. Enter the following:

- **Profile Name:** enter descriptive name.
- **Certificate:** select the identity certificate, e.g., **sbc40.crt**, 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:



7.3 Global Profiles

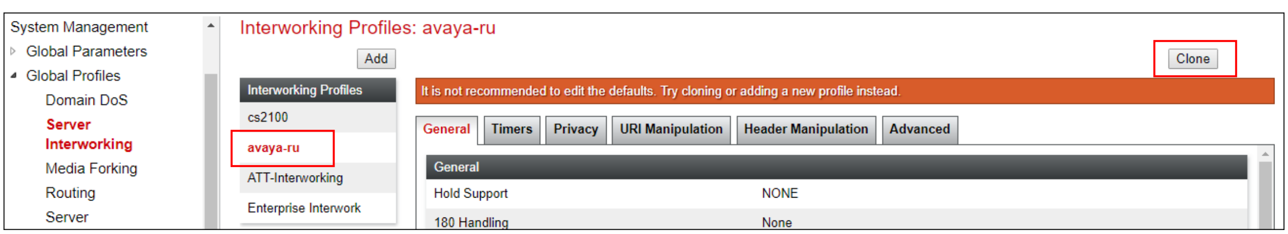
Global Profiles allow for configuration of parameters across the Avaya SBCE appliances.

7.3.1 Server Interworking – Avaya

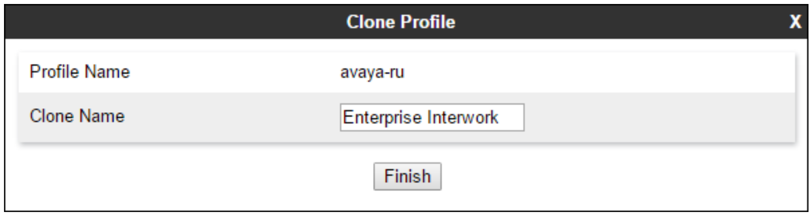
Server Interworking allows users to configure and manage various SIP call server-specific capabilities such as call hold and T.38 faxing. This section defines the connection to Session Manager.

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

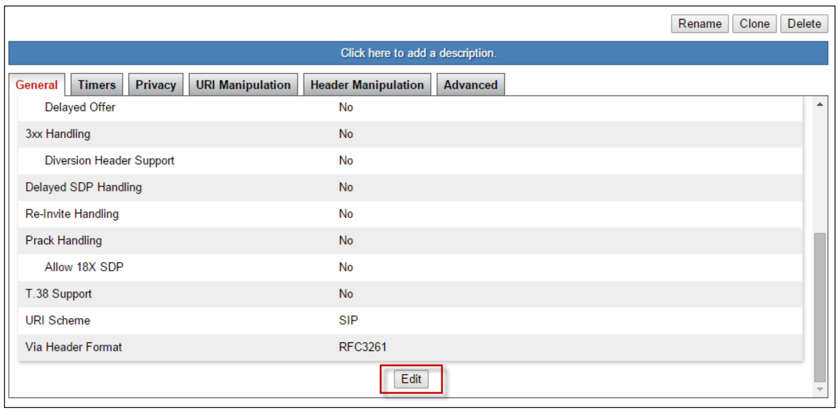
Step 2 - Select the pre-defined **avaya-ru** profile and click the **Clone** button.



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

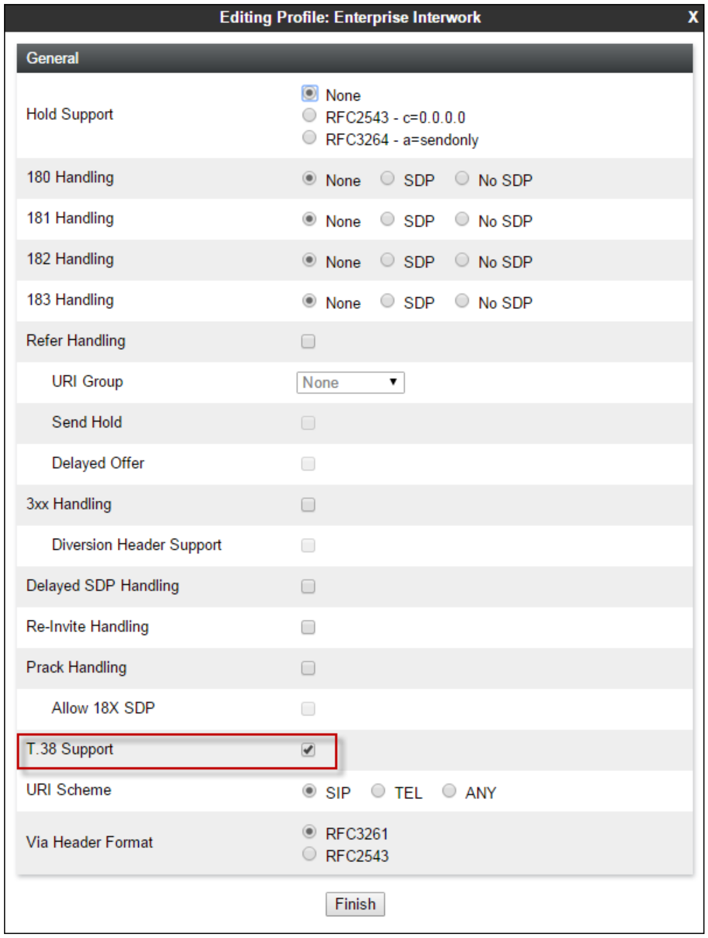


Step 4 - The new Enterprise Interwork profile will be listed. Select it, scroll to the bottom of the Profile screen, and click on **Edit**.

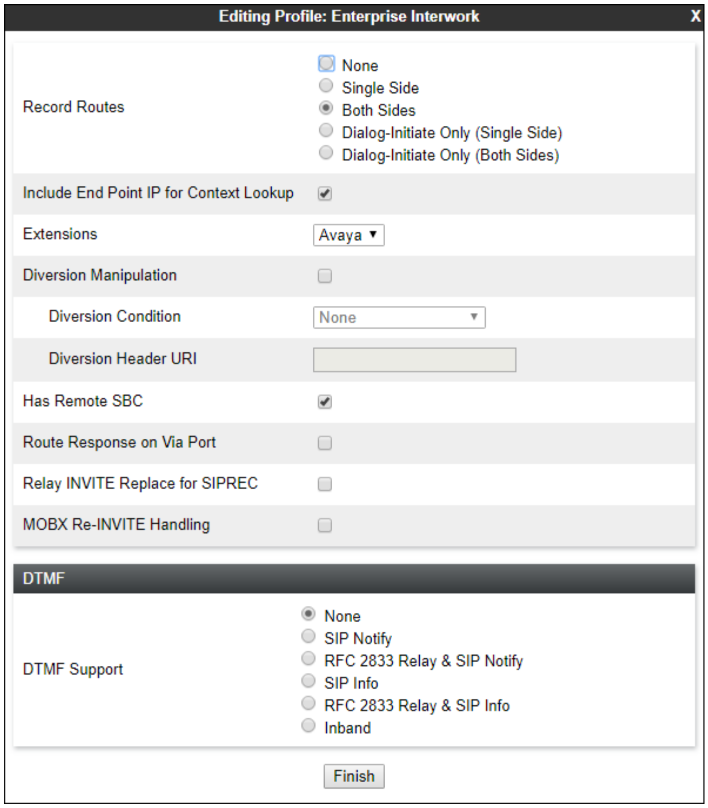


Step 5 - The **General** screen will open.

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



Step 6 - Select the **Advanced** tab, accept the default values, and click **Finish**.



7.3.2 Signaling Manipulation

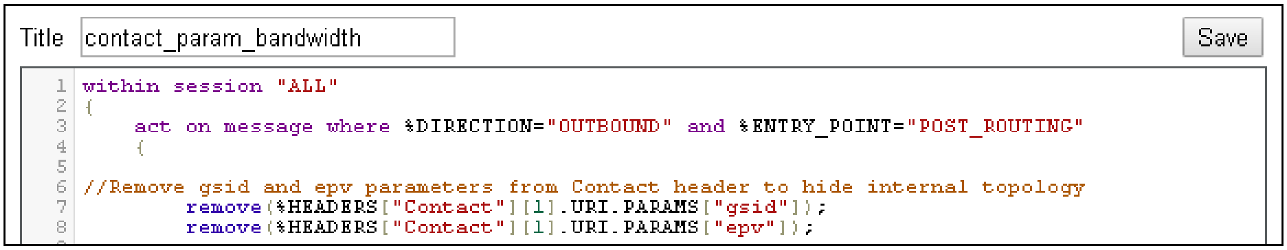
Signaling Manipulations are SigMa scripts the Avaya SBCE can use to manipulate SIP headers/messages.
In the reference configuration, one signaling manipulation script is used.

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 (**Section 7.4.3**) does not meet the desired result. Refer to **[11]** for information on the Avaya SBCE scripting language.

Step 1 - As described in **Section 2.2, Item 7**, Avaya SIP endpoints may send requests with Endpoint-View headers containing private network information. These are removed 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.

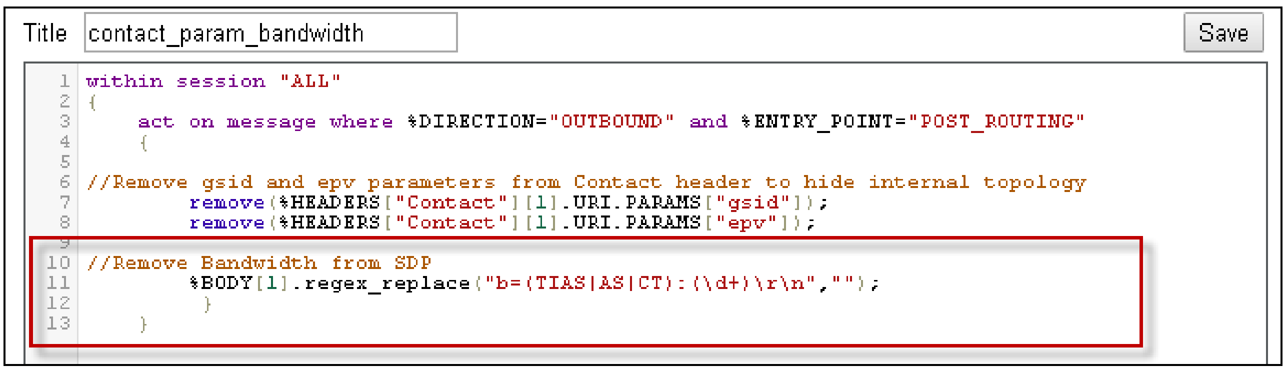
1. Select **Global Profiles** from the menu on the left-hand side.
2. Select **Signaling Manipulation**.

3. Click **Add Script** (not shown) and the script editor window will open.
4. Enter a name for the script in the **Title** box (e.g., **contact_param_bandwidth**). The following script is defined:



Step 2 - As described in **Section 2.2**, **Item 8**, some Avaya SIP endpoints may send Bandwidth headers that may cause issues with the AT&T network. The following signaling manipulation script is added to the script defined in **Step 1** above, to remove these Bandwidth headers.

1. The following script is added:



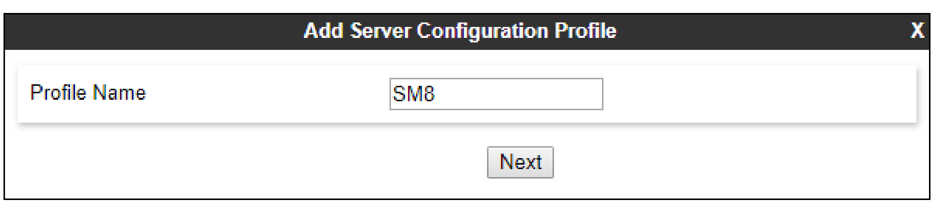
Step 3 - Click on **Save**. The script editor will test for any errors, and the window will close. This script is applied to the AT&T Server Configuration in **Section 7.3.4**, **Step 3**.

7.3.3 Server Configuration – Session Manager

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

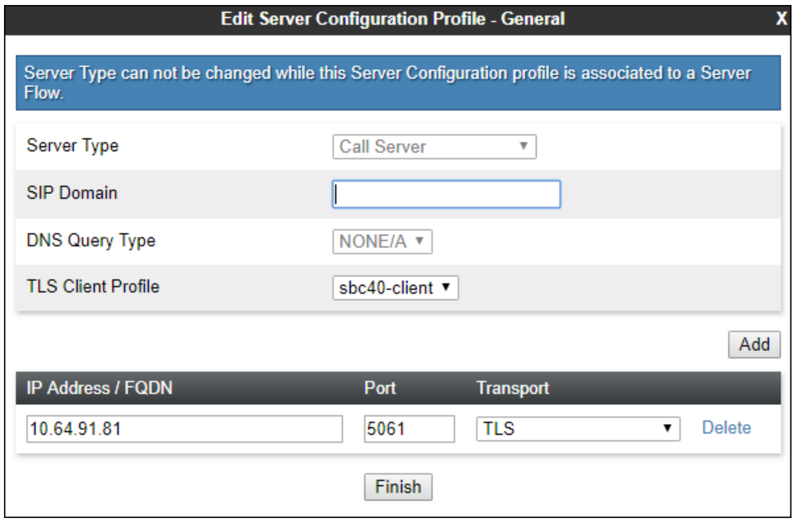
Step 1 - Select **Global Profiles → Server Configuration** from the left-hand menu.

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 7.2.3** (e.g., **sbc40-client**)
- **IP Address/FQDN**: **10.64.91.81** (Session Manager network IP address)
- **Transport**: Select **TLS**
- **Port**: **5061**
- Select **Next** (not shown)

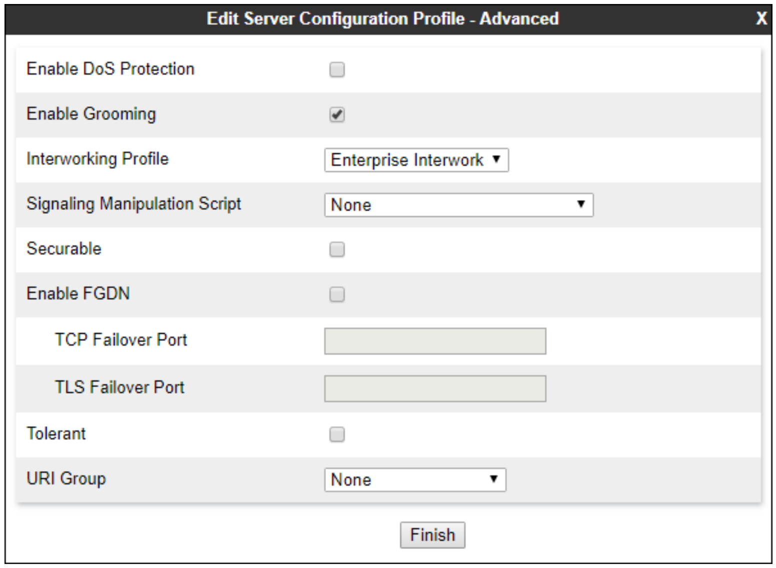


Step 4 - The **Authentication** and **Heartbeat** windows will open (not shown).

- Select **Next** to accept default values.

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

- Check **Enable Grooming**.
- Select **Enterprise Interwork** (created in **Section 7.3.1**), for **Interworking Profile**.
- In the **Signaling Manipulation Script** field select **None**.
- Select **Finish**.



7.3.4 Server Configuration – AT&T

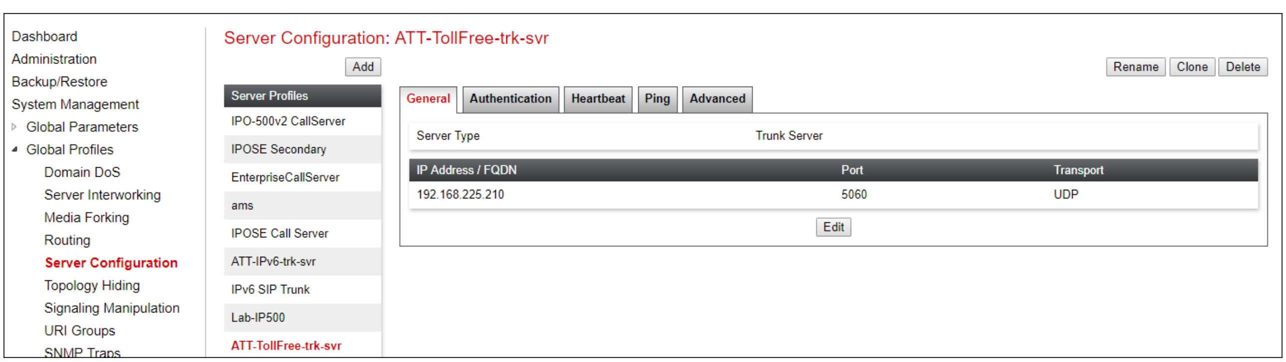
Note – The AT&T IPTF service may provide a Primary and Secondary Border Element. This section describes the connection to a single (Primary) Border Element. See **Addendum 1** for information on configuring two IPTF Border Elements (Primary & Secondary).

Repeat the steps in **Section 7.3.3**, with the following changes, to create a Server Configuration for the Avaya SBCE connection to AT&T.

Step 1 - Select **Add Profile** and enter a Profile Name (e.g., **ATT-TollFree-trk-svr**) and select **Next**.

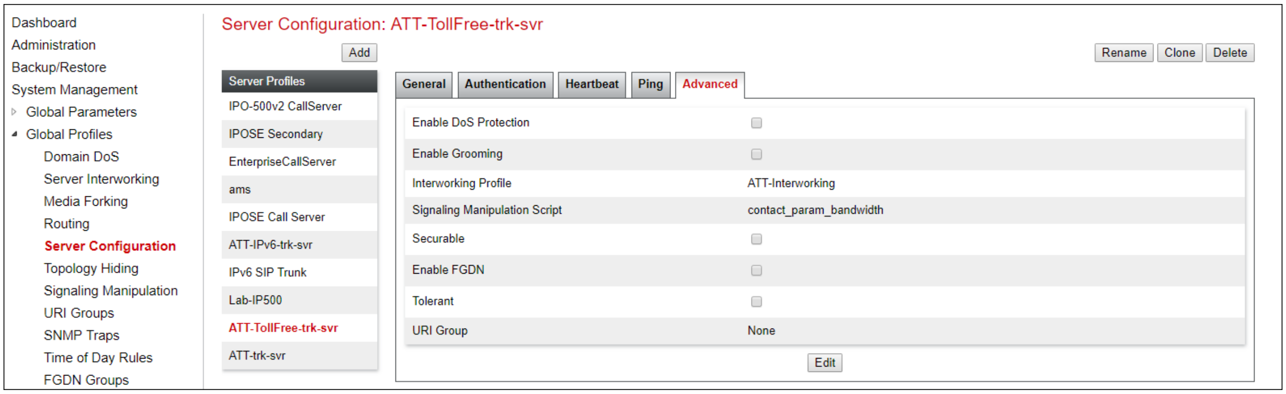
Step 2 - On the **General** window (not shown), enter the following.

- Select **Server Type**: **Trunk Server**
- **IP Address/FQDN**: **192.168.225.210** (AT&T Border Element IP address)
- **Port**: **5060**
- Select **Transport**: **UDP**
- Select **Next** until the **Advanced** tab is reached



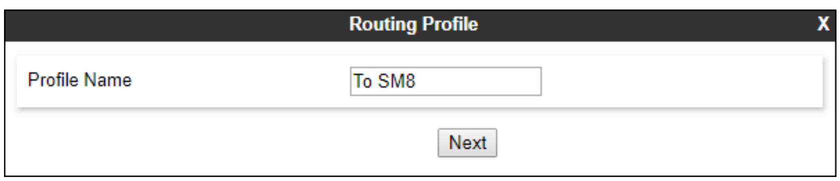
Step 3 - On the **Advanced** window, enter the following.

- For the **Signaling Manipulation Script** select the **contact_param_bandwidth** script defined in **Section 7.3.2**.
- Select **Finish** (not shown).



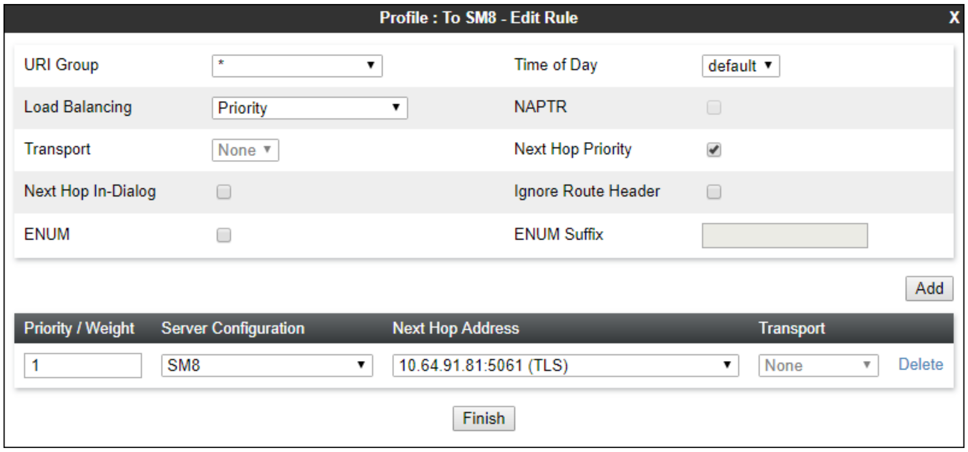
7.3.5 Routing – To Session Manager

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., **To SM)** 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** = **EnterpriseCallServer** (from **Section 7.3.3**).
- **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** fields are grayed out.
- Click on **Finish**.



7.3.6 Routing – To AT&T

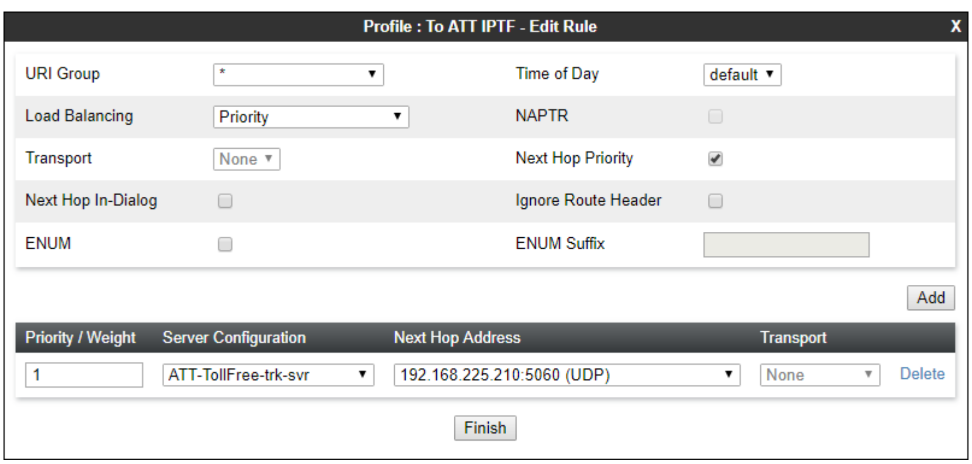
Repeat the steps in **Section 7.3.5**, with the following changes, to add a Routing Profile for the Avaya SBCE connection to AT&T.

Step 1 - On the **Global Profiles → Routing window (not shown),** enter a Profile Name: (e.g., **To ATT IPTF**).

Step 2 - On the Next-Hop Address window (not shown), populate the following fields:

- **Priority/Weight** = **1**
- **Server Configuration** = **ATT-TollFree-trk-svr** (from **Section 7.3.4**).
- **Next Hop Address**: Verify that the **192.168.225.210:5060 (UDP)** entry from the drop-down menu is selected (AT&T Border Element IP address).
- Use default values for the rest of the parameters.

Step 4 - Click **Finish**.



7.3.7 Topology Hiding – Enterprise Side

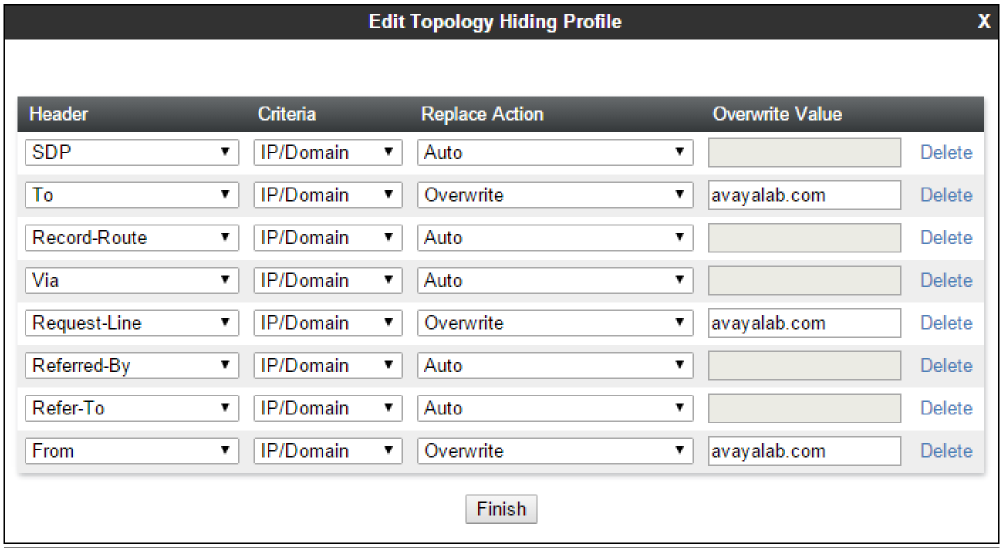
The **Topology Hiding** screen allows users to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the security of the network. It hides the topology of the enterprise network from external networks.

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.

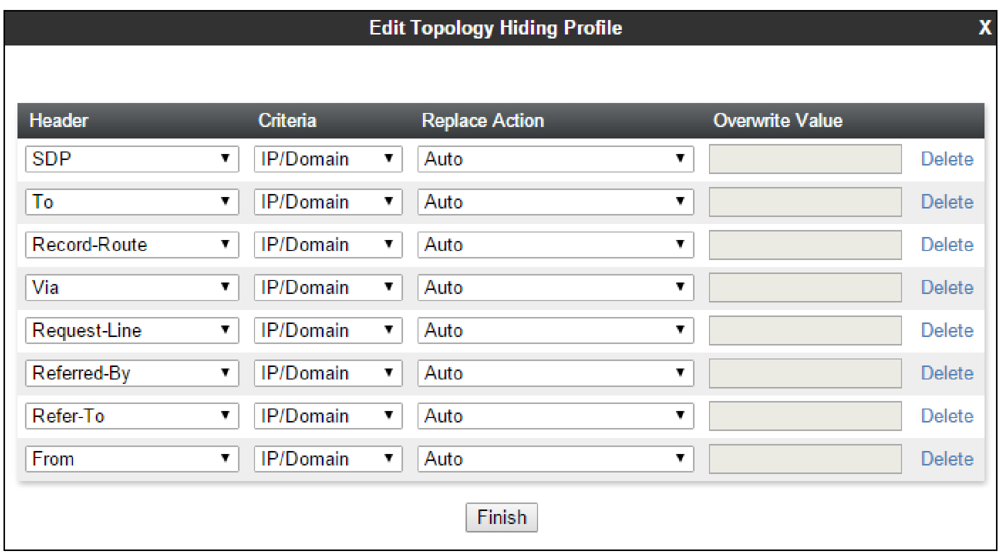
Step 4 - Populate the fields as shown below and click **Finish**. Note that **avayalab.com** is the domain used by the CPE (see **Sections 5.1** and **6.6**).



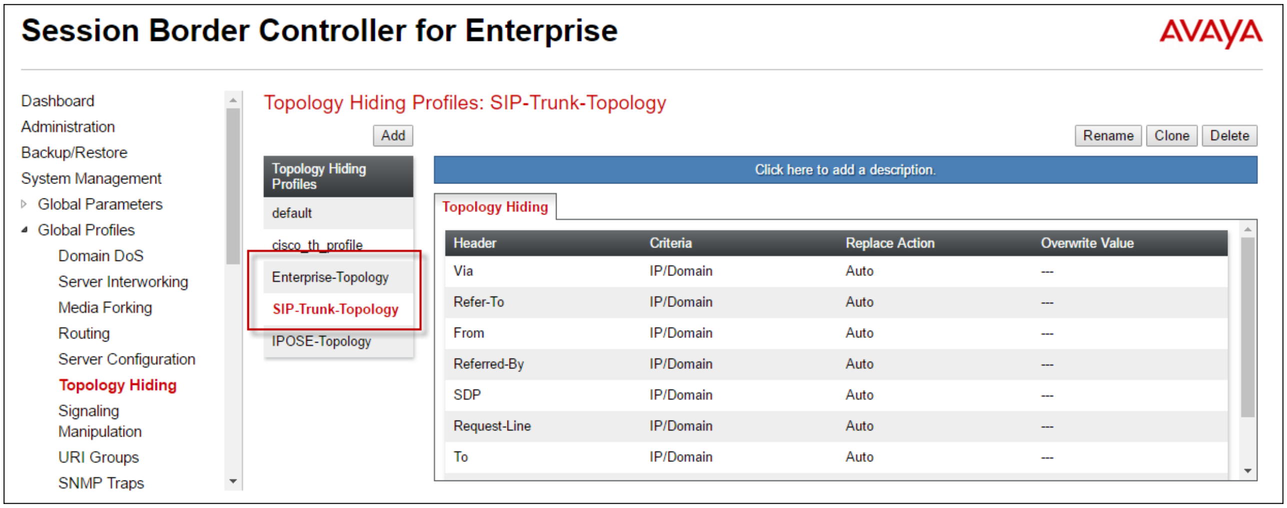
7.3.8 Topology Hiding – AT&T Side

Repeat the steps in **Section 7.3.7**, with the following changes, to create a Topology Hiding Profile for the Avaya SBCE connection to AT&T.

1. Enter a Profile Name (e.g., **SIP-Trunk-Topology**).
2. Use the default values for all fields and click **Finish**.



The following screen shows the completed **Topology Hiding Profile** forms in the shared test environment.



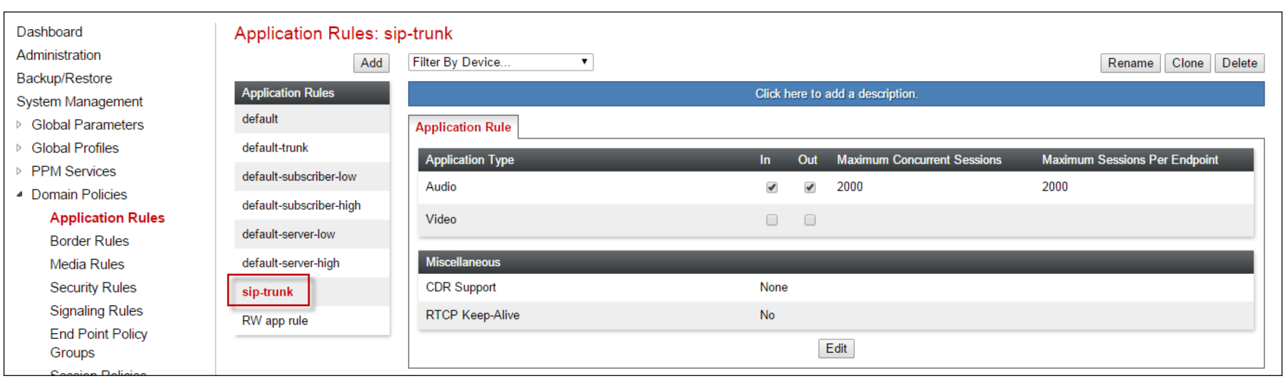
7.4 Domain Policies

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.

7.4.1 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.



7.4.2 Media Rules

Media Rules are used to define QOS parameters. Separate media rules are created for AT&T and Session Manager.

7.4.2.1 Enterprise – Media Rule

Step 1 - Select **Domain Policies → Media Rules** from the left-hand side menu.
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.

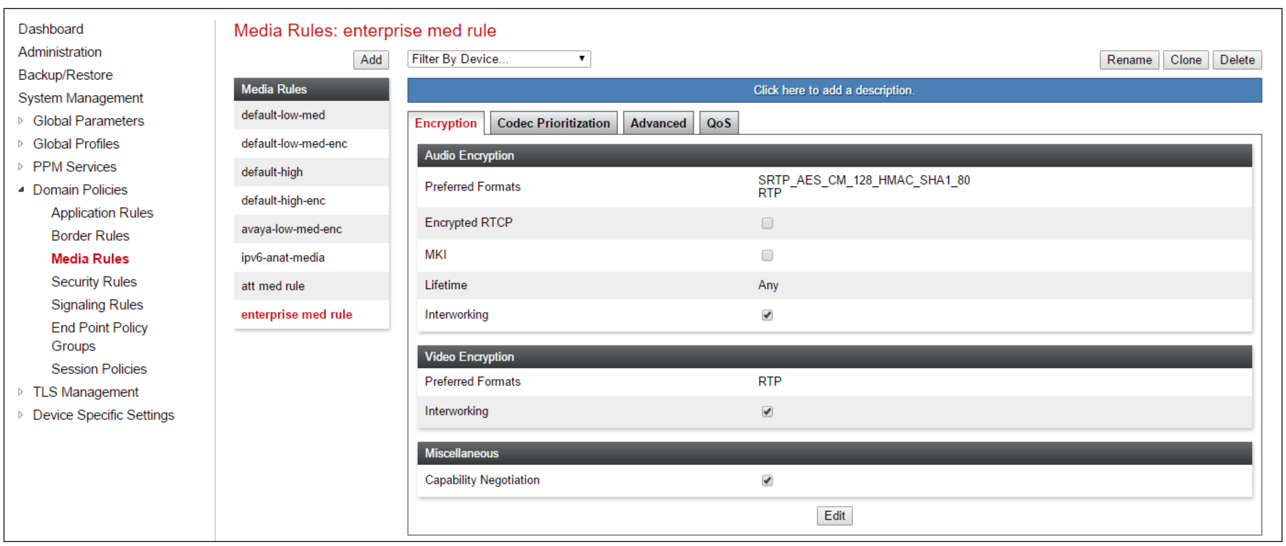
- In the **Clone Name** field enter **enterprise med rule**
- Click **Finish**. The newly created rule will be displayed.

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

- Select the **Encryption** tab.
- Click the **Edit** button and the **Media Encryption** window will open (not shown).
- **Preferred Format #2**: Select **RTP** from the drop-down.
- Select the **Capability Negotiation** box.

Step 5 - Click **Finish**.

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

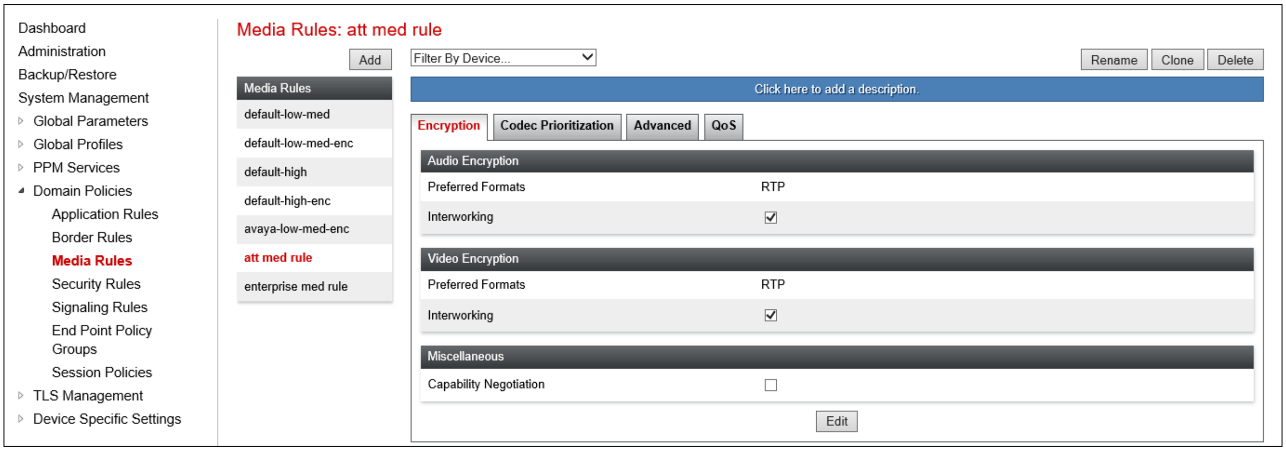


7.4.2.2 AT&T – Media Rule

Repeat the steps in **Section 7.4.2.1**, with the following changes, to create a Media Rule for AT&T.

1. From the Media Rules menu, select the **default-low-med** rule.
2. In the **Clone Name** field enter **att med rule**.
3. Use default values for all settings.

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



7.4.3 Signaling Rules

In the reference configuration, Signaling Rules are used to filter various SIP headers.

7.4.3.1 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.

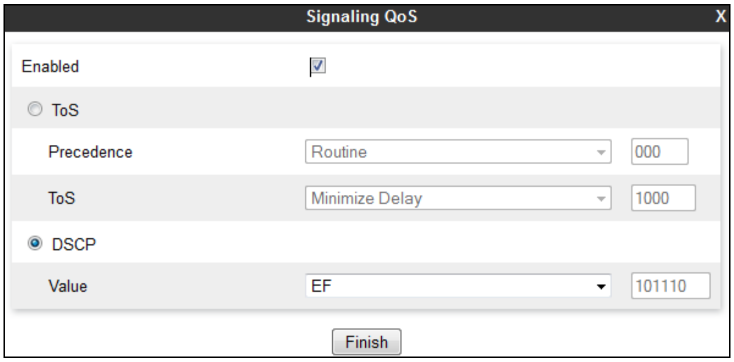
Step 3 - Select the **Clone** button and the **Clone Rule** window will open (not shown).

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

Step 4 - Highlight **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**.

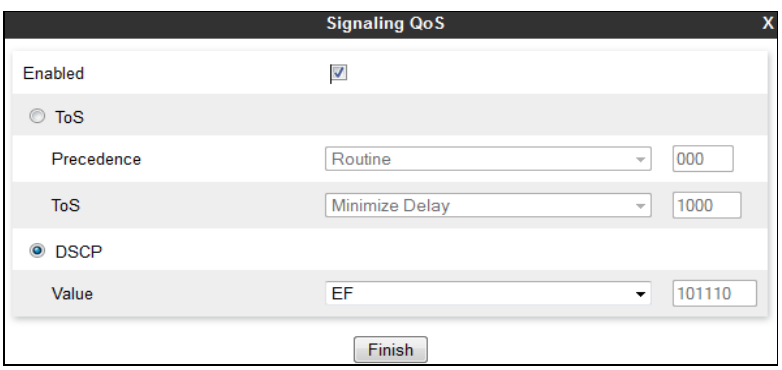


7.4.3.2 AT&T – Signaling Rule

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**: att sig rule**

Step 5 - Click **Finish**.
Step 6 - Highlight **att sig rule**, select the **Signaling QoS** tab and repeat **Steps 4** & **5** from **Section 7.4.3.1**.



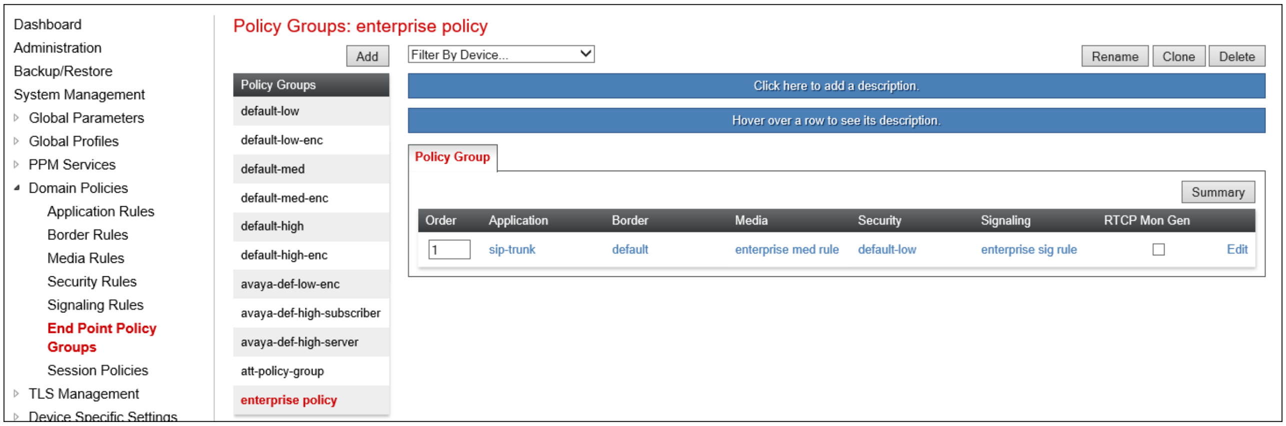
7.4.4 Endpoint Policy Groups – Enterprise Connection

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 policy**.

- **Application Rule**: **sip-trunk** (created in **Section 7.4.1**).
- **Border Rule**: **default**.
- **Media Rule**: **enterprise med rule** (created in **Section 7.4.2.1**).
- **Security Rule**: **default-low**.
- **Signaling Rule**: **enterprise sig rule** (created in **Section 7.4.3.1**).

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

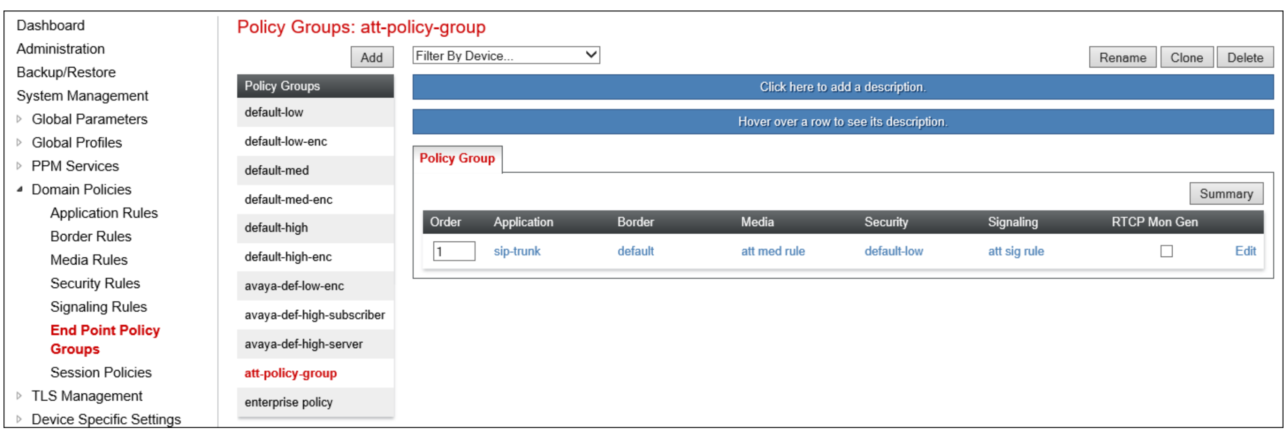


7.4.5 Endpoint Policy Groups – AT&T Connection

Step 1 - Repeat steps **1** through **4** from **Section 7.4.4**with the following changes:

- **Group Name**: **att-policy-group**.
- **Media Rule**: **att med rule** (created in **Section 7.4.2.2**).
- **Signaling Rule**: **att sig rule** (created in **Section 7.4.3.2**).

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

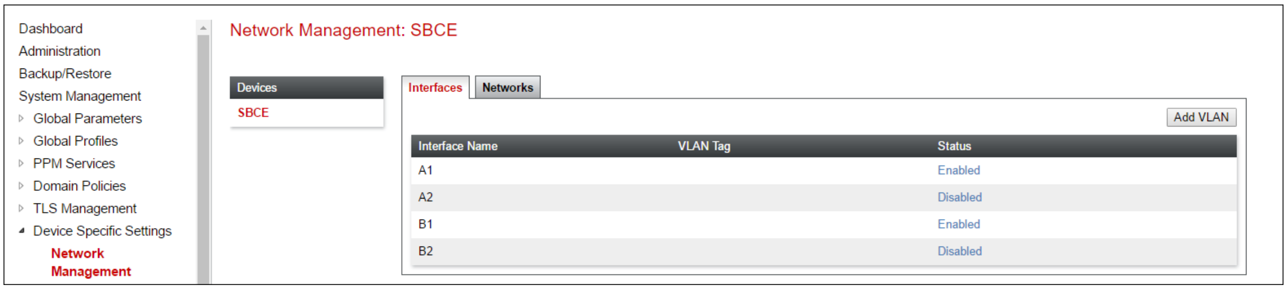


7.5 Device Specific Settings

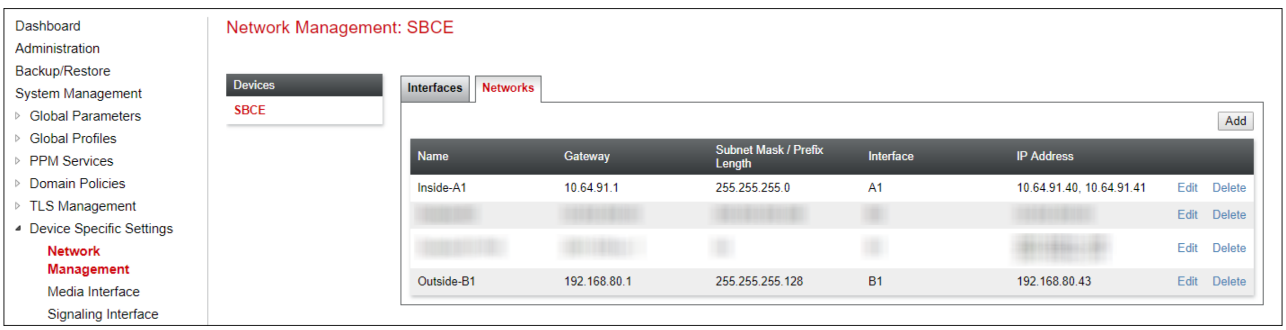
7.5.1 Network Management

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.



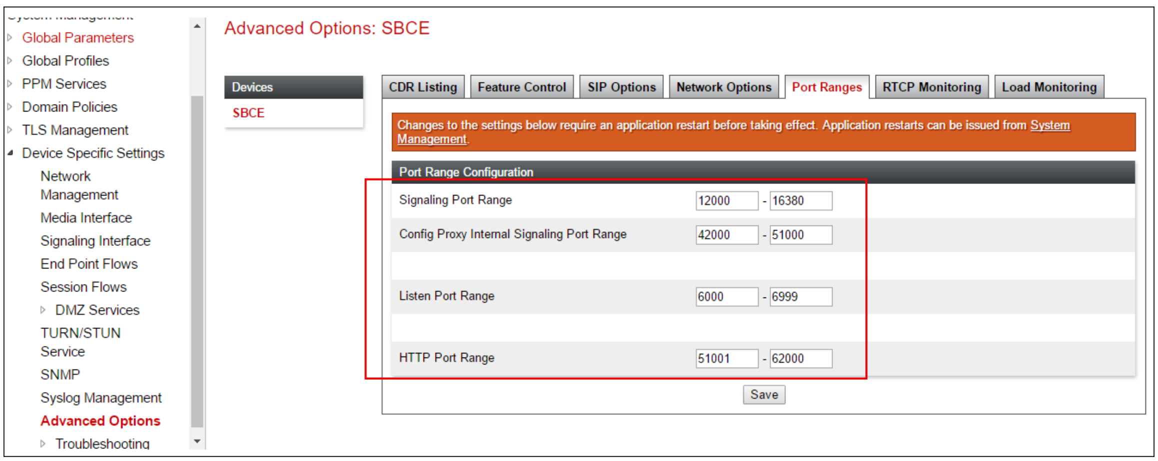
Step 3 - Select the **Networks** tab to display the IP provisioning for the A1 and B1 interfaces. These values are normally specified during installation. These can be modified by selecting **Edit**; however, some of these values may not be changed if associated provisioning is in use.



7.5.2 Advanced Options

In **Section 7.5.3**, the media UDP port ranges required by AT&T are configured (**16384 – 32767**). However, by default part of this range is already allocated by the Avaya SBCE for internal use (22000 - 31000). The following steps reallocate the port ranges used by the Avaya SBCE so the range required by AT&T can be defined in **Section 7.5.3**.

Step 1 - Select **Device Specific Settings → Advanced Options** from the menu on the left-hand side.
Step 2 - Select the **Port Ranges** tab.
Step 3 - In the **Signaling Port Range** row, change the range to **12000** – **16380**
Step 4 - In the **Config Proxy Internal Signaling Port Range** row, change the range to **42000** – **51000**.
Step 5 - In the **Listen Port Range** row, change the range to **6000** – **6999**.
Step 6 - In the **HTTP Port Range** row, change the range to **51001** – **62000**.
Step 7 - Scroll to the bottom of the window and select **Save**. Note that changes to these values require an application restart (see **Section 7.1**).



7.5.3 Media Interfaces

As mentioned in **Section 7.4.2**, the IPTF service specifies that customers use RTP ports in the range of **16384** – **32767**. Both inside and outside ports have been changed to this range, but only the outside is required by the IPTF service.

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

Step 2 - Select **Media Interface**.

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

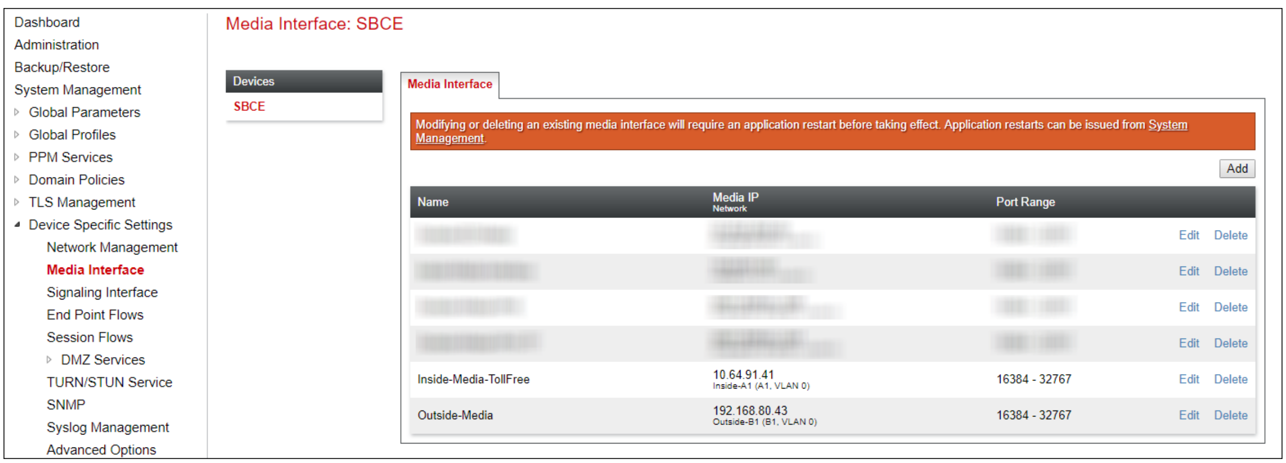
- **Name**: **Inside-Media-TollFree**.
- **IP Address**: **10.64.91.41** (Avaya SBCE A1 IP address).
- **Port Range**: **16384** – **32767**.

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

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

- **Name**: **Outside-Media**.
- **IP Address**: **192.168.80.43** (Avaya SBCE B1 IP address).
- **Port Range**: **16384** – **32767**.

Step 6 - Click **Finish** (not shown). Note that changes to these values require an application restart (see **Section 7.1**). The completed **Media Interface** screen in the shared test environment is shown below.



7.5.4 Signaling Interface

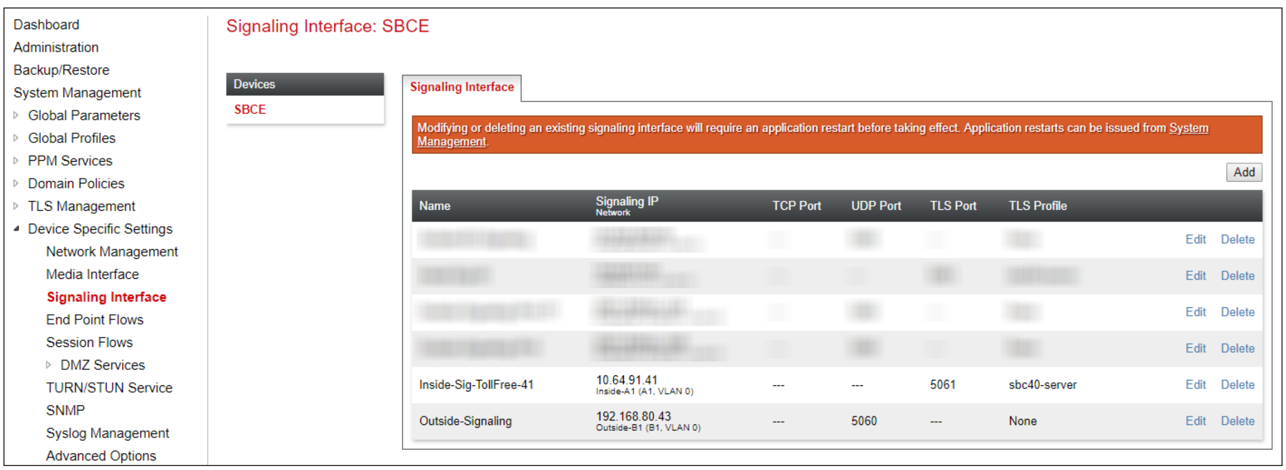
Step 1 - Select **Device Specific Settings** from the menu on the left-hand side (not shown).
Step 2 - Select **Signaling Interface**.
Step 3 - Select **Add** (not shown) and enter the following:

- **Name**: **Inside-Sig-TollFree-41**.
- **IP Address**: **10.64.91.41** (Avaya SBCE A1 IP address).
- **TLS Port**: **5061**.
- **TLS Profile**: **sbc40-server**.

Step 4 - Click **Finish** (not shown).
Step 5 - Select **Add** again, and enter the following:

- **Name**: **Outside-Signaling**.
- **IP Address**: **192.168.80.43** (Avaya SBCE B1 IP address).
- **UDP Port**: **5060**.

Step 6 - Click **Finish** (not shown). Note that changes to these values require an application restart (see **Section 8.0**).



7.5.5 Endpoint Flows – For Enterprise

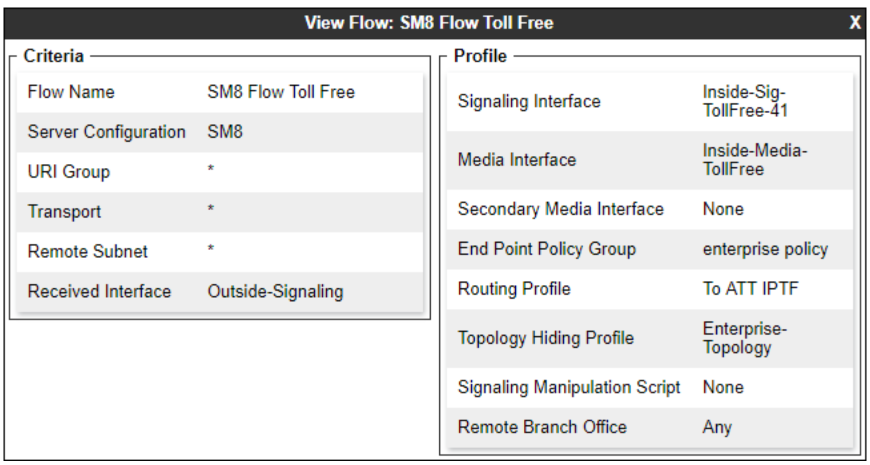
Step 1 - Select **Device Specific Settings → Endpoint Flows** from the menu on the left-hand side (not shown).

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

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

- **Name**: **SM8 Flow Toll Free**
- **Server Configuration**: **SM8** (**Section 7.3.3**)
- **URI Group**: *
- **Transport**: *
- **Remote Subnet**: *
- **Received Interface**: **Outside-Signaling** (**Section 7.5.4**)
- **Signaling Interface**: **Inside-Sig-TollFree-41** (**Section 7.5.4**)
- **Media Interface**: **Inside-Media-TollFree** (**Section 7.5.3**)
- **End Point Policy Group**: **enterprise policy** (**Section 7.4.4**)
- **Routing Profile**: **To ATT IPTF** (**Section 7.3.6**)
- **Topology Hiding Profile**: **Enterprise-Topology** (**Section 7.3.7**)
- Let other values default

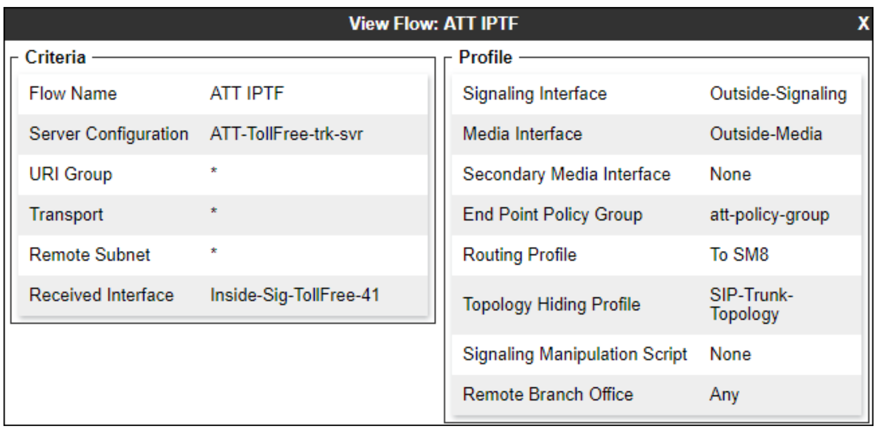
Step 4 - Click **Finish** (not shown).



7.5.6 Endpoint Flows – For AT&T

Step 1 - Repeat steps **1** through **4** from **Section 7.4.5**, with the following changes:

- **Name**: **ATT-IPTF**
- **Server Configuration**: **ATT-TollFree-trk-svr** (**Section 7.3.4**)
- **URI Group**: *
- **Transport**: *
- **Remote Subnet**: *
- **Received Interface**: **Inside-Sig-TollFree-41** (**Section 7.5.4**)
- **Signaling Interface**: **Outside-Signaling (Section 7.5.4**)
- **Media Interface**: **Outside-Media** (**Section 7.5.3**)
- **End Point Policy Group**: **att-policy-group** (**Section 7.4.5**)
- **Routing Profile**: **To SM8 (Section 7.3.5**)
- **Topology Hiding Profile**: **SIP-Trunk-Topology** (**Section 7.3.8**)



8 Verification Steps

The following steps may be used to verify the configuration:

8.1 AT&T IP Toll Free Service

1. Place an inbound call, answer the calls, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnects properly.
2. Verify basic call functions such as hold, transfer, and conference.
3. Verify the use of DTMF signaling.
4. Using the appropriate IPTF access numbers and DTMF codes, verify that the following IPTF features are successful:
    a. Legacy Transfer Connect DTMF triggered Agent Hold, Conference and Transfer capabilities
    b. Alternate Destination Routing call redirection capabilities based on Busy, Ring-No-Answer, and other SIP error codes.

8.2 Avaya Aura® Communication Manager

The following examples are only a few of the monitoring commands available on Communication Manager. See **[6]** for more information.

- Tracing a SIP trunk.
    a. From the Communication Manager console connection enter the command ***list trace tac xxx***, where ***xxx*** is a trunk access code defined for the SIP trunk to AT&T (e.g., 602). Note that in the trace shown below, Session Manager has previously converted the IPTF DNIS number included in the Request URI, to the Communication Manager VDN 71041, before sending the INVITE to Communication Manager.

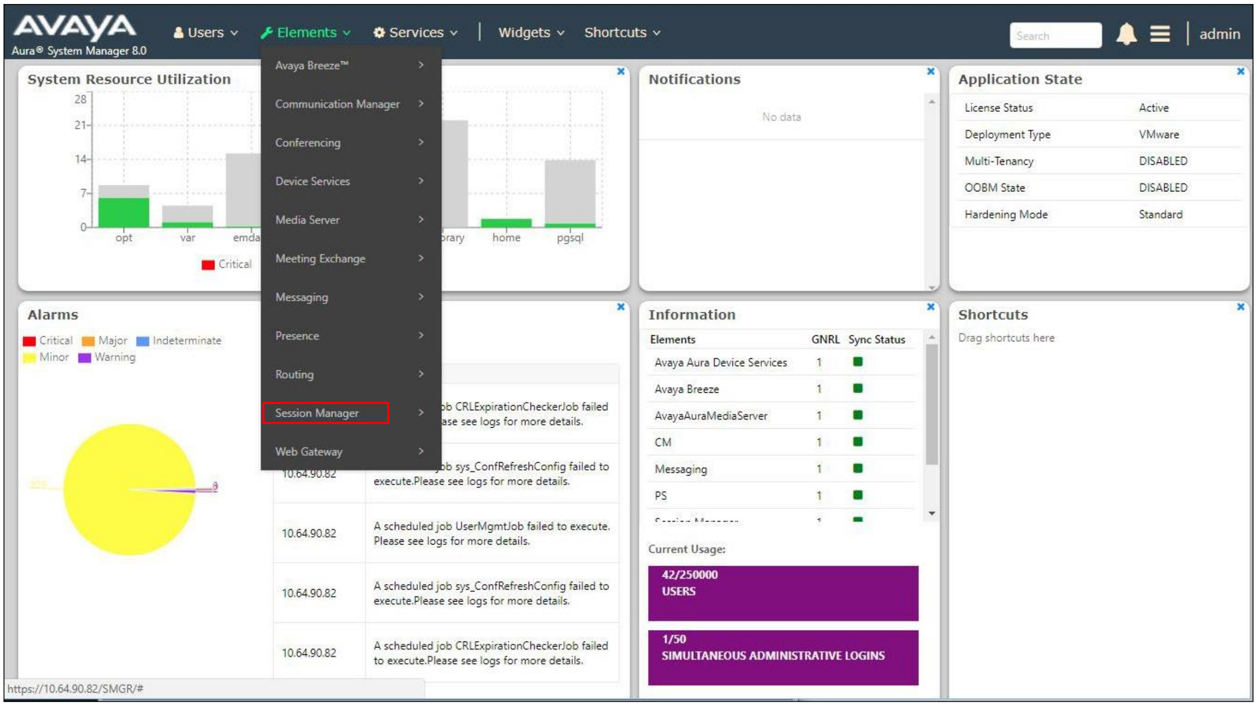
```
list trace tac *04                                                Page   1
                                LIST TRACE
time            data

11:35:39 TRACE STARTED 11/14/2018 CM Release String R018x.00.0.822.0
11:35:48 SIP<INVITE sips:71041@avayalab.com SIP/2.0
11:35:48     Call-ID: ea7abb3fb2ecff41cb203c03f7ed425e
11:35:48     active trunk-group 4 member 1    cid 0x59b
11:35:48    0  0 ENTERING TRACE cid 1435
11:35:48    4  1 vdn e71041 bsr appl   0 strategy 1st-found override n
11:35:48    4  1 AVDN: 71041 AVRD:
11:35:48    4  2 wait 2 secs hearing ringback
11:35:48 SIP>SIP/2.0 180 Ringing
11:35:48     Call-ID: ea7abb3fb2ecff41cb203c03f7ed425e
11:35:48     dial 71041
11:35:48     ring vector 4      cid 0x59b
11:35:48     G729 ss:off ps:30
             rgn:4 [10.64.91.41]:17044
             rgn:1 [10.64.91.91]:16388
```

- Other useful Communication Manager commands are, ***list trace station***, ***list trace vdn***, ***list trace vector***, ***list trace trunk***, ***list trace station***, ***status trunk***, and ***status station***.

8.3 Avaya Aura® Session Manager Status

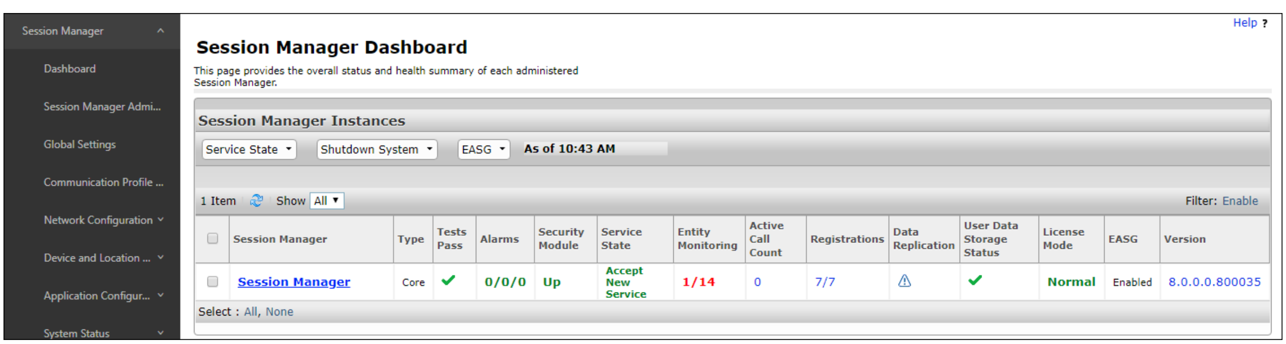
The Session Manager configuration may be verified via System Manager.

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**.

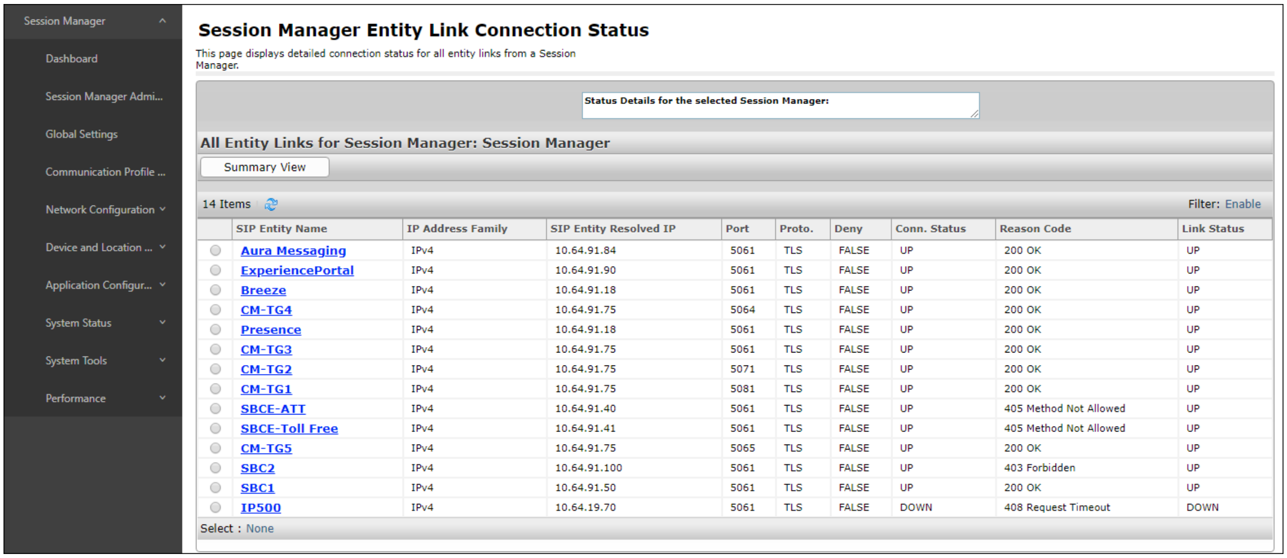


Step 2 - The Session Manager Dashboard is displayed. Note that the **Test Passed**, **Alarms**, **Service State**, and **Data Replication** columns, all show good status.

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



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



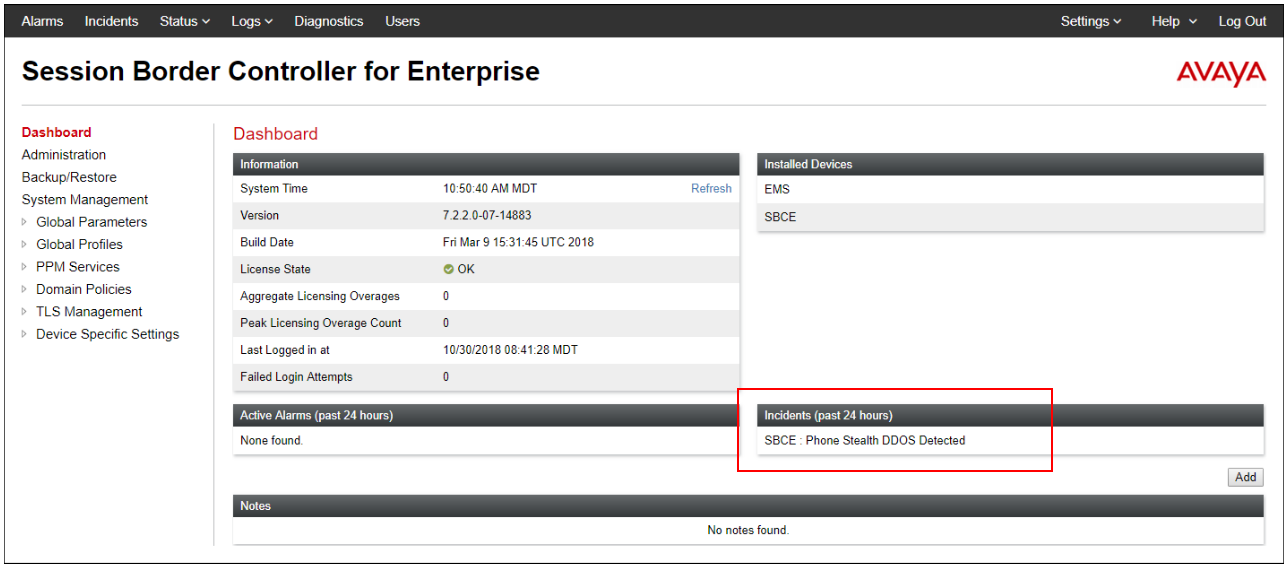
SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
Aura Messaging	IPv4	10.64.91.84	5061	TLS	FALSE	UP	200 OK	UP
ExperiencePortal	IPv4	10.64.91.90	5061	TLS	FALSE	UP	200 OK	UP
Breeze	IPv4	10.64.91.18	5061	TLS	FALSE	UP	200 OK	UP
CM-TG4	IPv4	10.64.91.75	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

Note – The **SBCE-Toll Free** Entity from the list of monitored entities above. The **Reason Code** column indicates that Session Manager has received a SIP **405 Method Not Allowed** response to the SIP OPTIONS it generated. This response is sufficient for SIP Link Monitoring to consider the link up. Also note that the Avaya SBCE sends the Session Manager generated OPTIONS on to the AT&T IPTF Border Element, and it is the AT&T Border Element that is generating the 405 response, and the Avaya SBCE sends it back to Session Manager.

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.

8.4 Avaya Session Border Controller for Enterprise Verification

Step 1 - Log into the Avaya SBCE as shown in **Section 7**. Across the top of the display are options to display **Alarms**, **Incidents**, **Status**, **Logs**, and **Diagnostics**. In addition, the most recent Incidents are listed in the lower right of the screen.



8.4.1 Protocol Traces

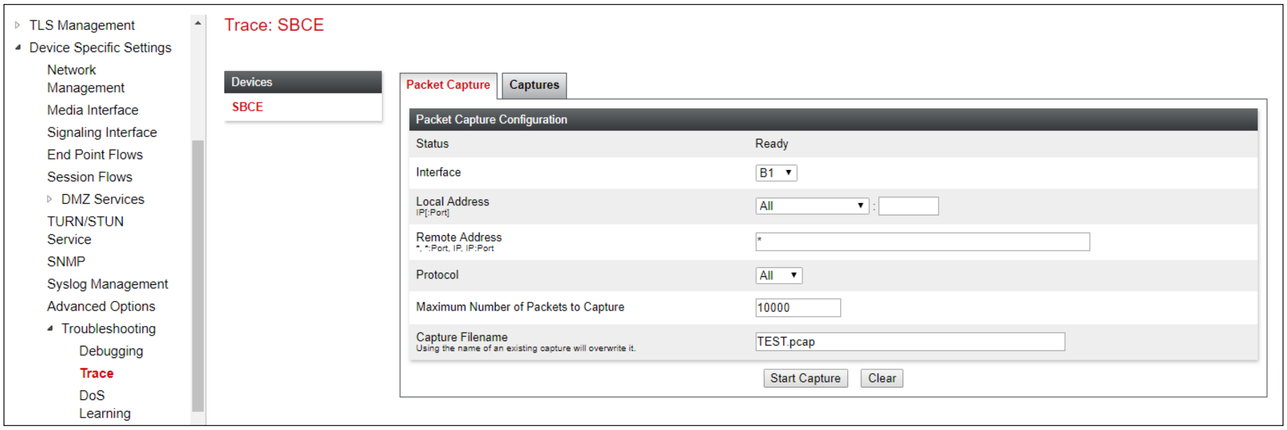
The Avaya SBCE can take internal traces of specified interfaces.

Step 1 - Navigate to **Device Specific Settings → Troubleshooting → Trace**.

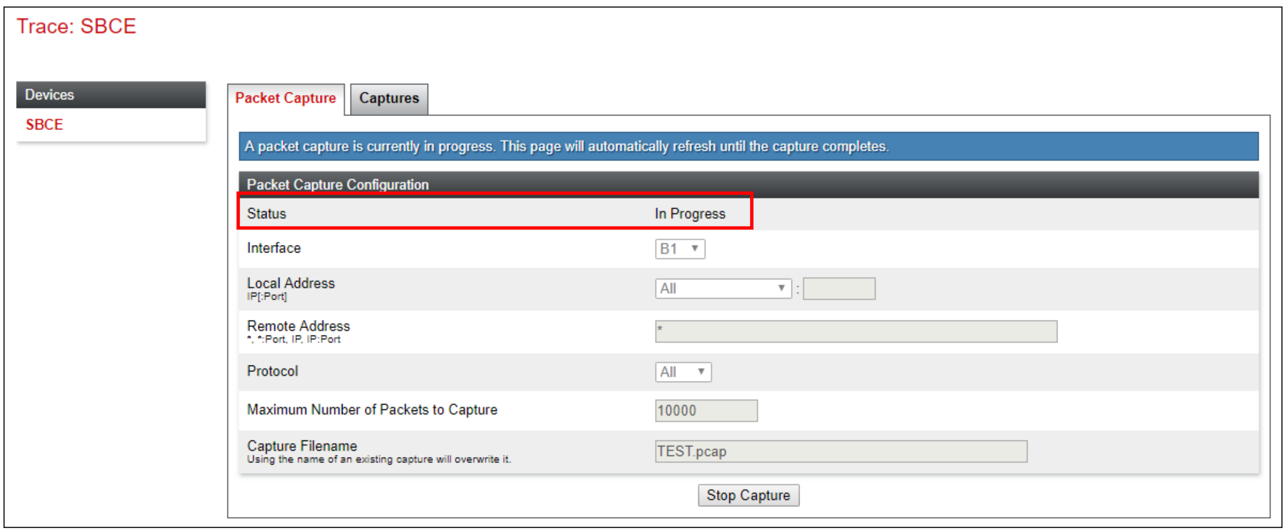
Step 2 - Select the **Packet Capture** tab and select the following:

- Select the desired **Interface** from the drop-down menu (e.g., **All**).
- Specify the **Maximum Number of Packets to Capture** (e.g., **10000**).
- Specify a **Capture Filename** (e.g., **TEST.pcap**).
- Unless specific values are required, the default values may be used for the **Local Address**, **Remote Address**, and **Protocol** fields.
- Click **Start Capture** to begin the trace.

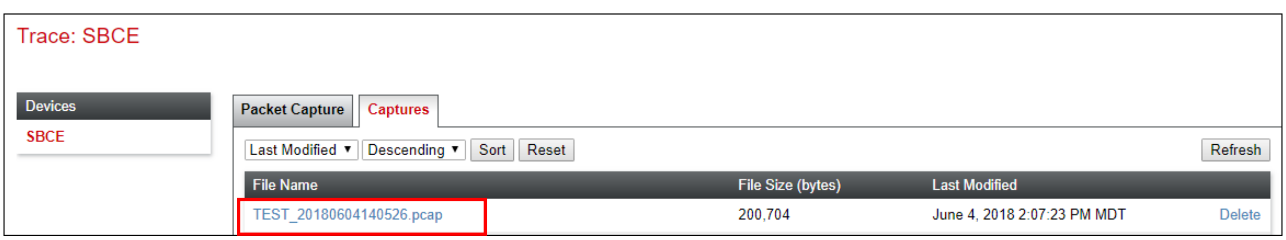
Note – Specifying **All** in the **Interface** field will result in the Avaya SBCE capturing traffic from both the A1 and B1 interfaces defined in the reference configuration. Also, when specifying the **Maximum Number of Packets to Capture**, estimate a number large enough to include all packets for the duration of the test.



The capture process will initialize and then display the following **In Progress** status window:



Step 3 - Run the test.
Step 4 - When the test is completed, select **Stop Capture** button shown above.
Step 5 - Click on the **Captures** tab and the packet capture is listed as a *.pcap* file with the date and time added to filename specified in **Step 2**.
Step 6 - Click on the **File Name** link to download the file and use Wireshark to open the trace.



9 Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, and the Avaya Session Border Controller for Enterprise 7.2, can be configured to interoperate successfully with the AT&T IP Toll Free service, within the constraints described in **Section 2.2.**

Testing was performed on a simulated AT&T IP Toll Free service circuit. The reference configuration shown in these Application Notes is intended to provide configuration guidance to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program.

10 References

The Avaya product documentation is available at http://support.avaya.com unless otherwise noted.

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

[10] Programming Call Vectors in Avaya Aura® Call Center, 6.0, June 2010

Avaya Session Border Controller for Enterprise

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

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

AT&T IP Toll Free Service:

- AT&T IP Toll Free Service description - http://www.business.att.com/enterprise/Service/voice-services/null/ip-toll-free/
- AT&T IP Toll Free service support: (800) 325-5555.

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.