



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 7.0.1, Avaya Aura® Session Manager 7.0.1 and Avaya Session Border Controller for Enterprise 7.1 with AT&T IP Toll Free SIP Trunk Service using IPv6 – Issue 1.1

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Session Manager 7.0.1, Avaya Aura® Communication Manager 7.0.1, and the Avaya Session Border Controller for Enterprise 7.1 with the AT&T IP Toll Free service using IPv6 and AT&T's **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager 7.0.1 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 7.0.1 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 7.1 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 7.0.1 (IPv4 address), Avaya Aura® Communication Manager 7.0.1 (IPv4 address), and the Avaya Session Border Controller for Enterprise 7.1 (IPv4/IPv6 address) with the AT&T IP Toll Free service (IPv6 address) using AT&T Virtual Private Network (AVPN) or Managed Internet Service Private Network Transport (MIS/PNT) connections¹.

Avaya Aura® Session Manager 7.0.1 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 7.0.1 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 7.1 (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 make adjustments to 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, as long as 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 from 14400-28800 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. But it should be noted 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. **Alphanumeric characters in IPv6 address** – The Avaya SBCE Server Configuration IP address field is case sensitive. The AT&T IPv6 address needs to be entered using lowercase characters. The Avaya SBCE will not match the source address of incoming SIP packets from AT&T if the IPv6 address is entered using uppercase characters (See **Section 7.6.2**).
10. **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.
11. **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 7.0.1 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 7.0.1 provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager 7.0.1 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), a Avaya 2420 Digital Telephone, as well as Avaya one-X® Agent soft phone (H323).
- The Avaya SBCE 7.1 provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the IPTF IPv6 service and the enterprise internal IPv4 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 IPv6 and 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 IPv4 and 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.

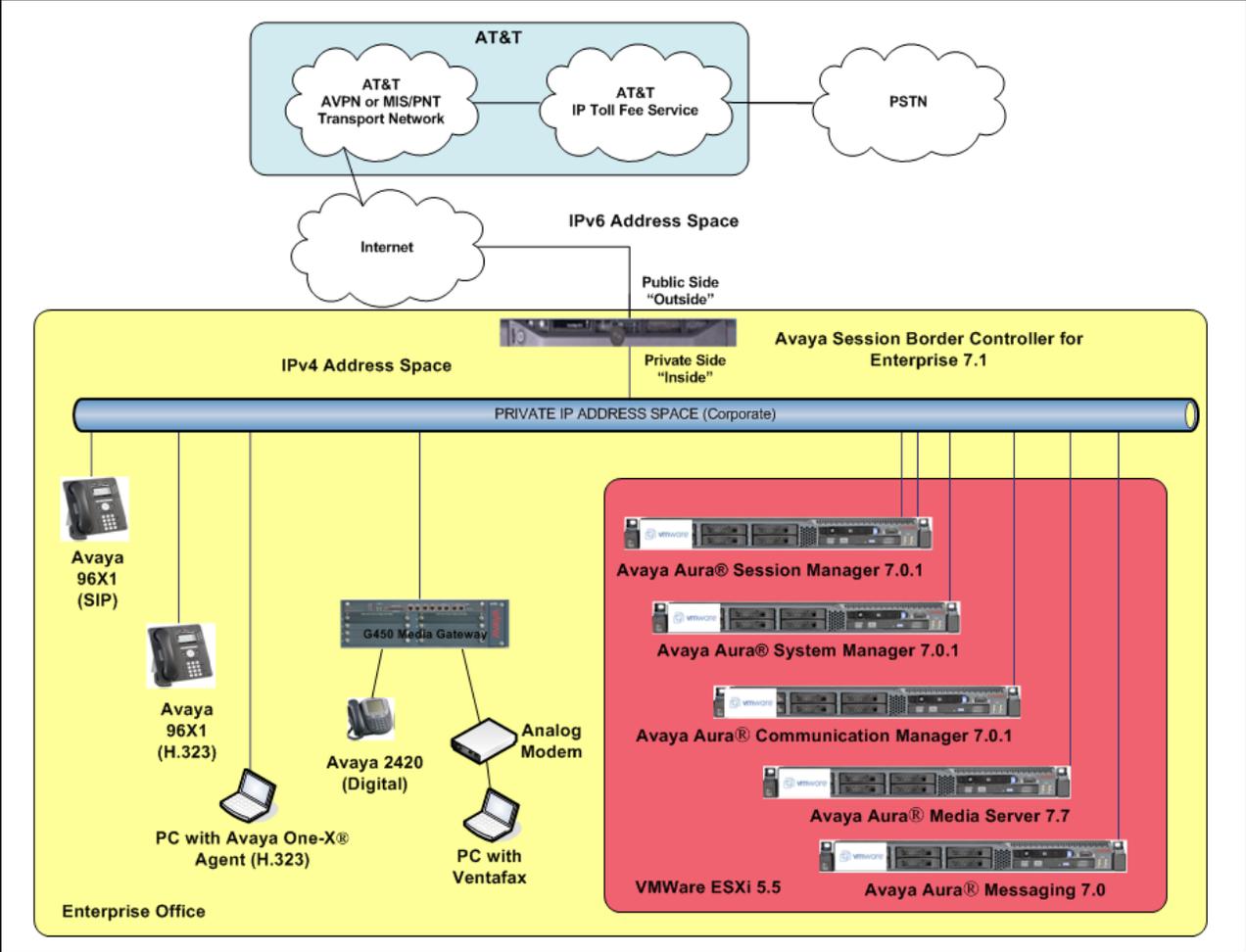


Figure 1: Reference configuration

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	10.64.90.62
Avaya Aura® Session Manager	
Management IP Address	10.64.90.61
Network IP Address	10.64.91.61
Avaya Aura® Communication Manager	
IP Address	10.64.91.65
Avaya Aura® Communication Manager extensions	14xxx = 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	3ffe:ffff:bb:bb::241 (see note below)
AT&T IP Toll Free Border Element	
IP Address	3ffe:ffff:aa:aa:10:10:172:80

Table 1: Illustrative Values Used in these Application Notes

Note – In the reference configuration, the IPTF service delivered 15 DNIS digits, with the format *0000xxxxxxxxxx*. 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 the majority of testing 15 digits were used, the length was also changed to 9 digits, and 21 digits to test compatibility. The total length supported by the IPTF service is 21 digits, including the five leading zeroes.

Note – For security reasons, the actual IPv6 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 **3ffe:ffff:bb:bb::241** (Avaya SBCE public interface) and **3ffe:ffff:aa:aa:10:10:172:80** (AT&T BE IPv6 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.

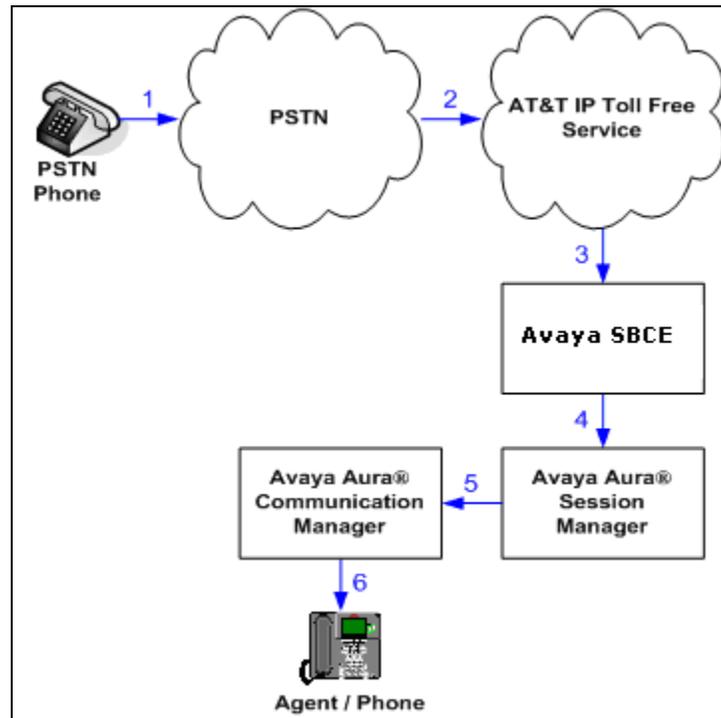


Figure 2: Inbound AT&T IP Toll Free Service Call to an Agent queue/telephone

Note that 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	<ul style="list-style-type: none">• 7.0.1.2.086007
Avaya Aura® Session Manager	<ul style="list-style-type: none">• 7.0.1.2.701230
Avaya Aura® Communication Manager	<ul style="list-style-type: none">• 7.0.1.2.0-R017x.00.0.441.0 (23523)
Avaya Aura® Media Server	<ul style="list-style-type: none">• 7.7.0.359
Avaya Aura® Messaging	<ul style="list-style-type: none">• 7.0-00.0.441.0-017_0004 (SP 0)
Avaya G450 Media Gateway	<ul style="list-style-type: none">• 37.41.0
Avaya Session Border Controller for Enterprise	<ul style="list-style-type: none">• 7.1.0.2-01-13249
Avaya 96x1 IP Telephones	<ul style="list-style-type: none">• H.323 Version 6.6401• SIP Version 7.0.1.4.6
Avaya one-X® Agent (H323)	<ul style="list-style-type: none">• 2.5.50022.65
Ventafax Home Version (Windows based Fax device)	<ul style="list-style-type: none">• 7.8.253.611

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.

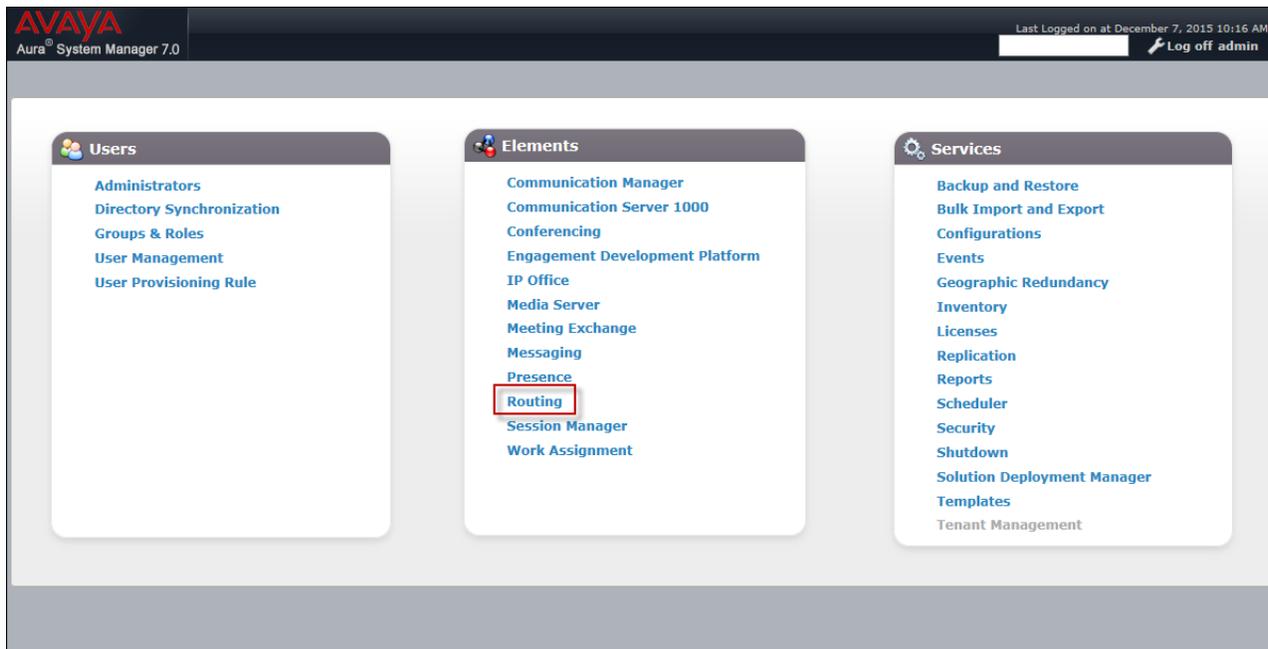
Session Manager serves as a central point for supporting SIP-based communication services in an enterprise. Session Manager connects and normalizes disparate SIP network components and provides a central point for external SIP trunking to the PSTN. The various SIP network components are represented as 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 in the center, 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 - In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern:** Leave blank.

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

The screenshot displays the 'Location Details' configuration page in the Avaya System Manager. The left-hand navigation menu is open, showing 'Locations' selected. The main content area is titled 'Location Details' and includes the following sections:

- General:** Name: Main; Notes: Avaya SIL.
- Dial Plan Transparency in Survivable Mode:** Enabled: ; Listed Directory Number: [text box]; Associated CM SIP Entity: [text box].
- Overall Managed Bandwidth:** Managed Bandwidth Units: Kbit/sec; Total Bandwidth: [text box]; Multimedia Bandwidth: [text box]; Audio Calls Can Take Multimedia Bandwidth: .
- Per-Call Bandwidth Parameters:** Maximum Multimedia Bandwidth (Intra-Location): 2000 Kbit/Sec; Maximum Multimedia Bandwidth (Inter-Location): 2000 Kbit/Sec; Minimum Multimedia Bandwidth: 64 Kbit/Sec; Default Audio Bandwidth: 80 Kbit/sec.
- Alarm Threshold:** Overall Alarm Threshold: 80%; Multimedia Alarm Threshold: 80%; Latency before Overall Alarm Trigger: 5 Minutes; Latency before Multimedia Alarm Trigger: 5 Minutes.
- Location Pattern:** A table with columns for IP Address Pattern and Notes. The table is currently empty (0 Items).

Buttons for 'Commit' and 'Cancel' are present at the top right and bottom right of the configuration area.

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**).

Location Details [Commit] [Cancel]

General

* Name:
 Notes:

Dial Plan Transparency in Survivable Mode

Enabled:
 Listed Directory Number:
 Associated CM SIP Entity:

Overall Managed Bandwidth

Managed Bandwidth Units:
 Total Bandwidth:
 Multimedia Bandwidth:
 Audio Calls Can Take Multimedia Bandwidth:

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): Kbit/Sec
 Maximum Multimedia Bandwidth (Inter-Location): Kbit/Sec
 * Minimum Multimedia Bandwidth: Kbit/Sec
 * Default Audio Bandwidth:

Alarm Threshold

Overall Alarm Threshold: %
 Multimedia Alarm Threshold: %
 * Latency before Overall Alarm Trigger: Minutes
 * Latency before Multimedia Alarm Trigger: Minutes

Location Pattern

Add Remove Filter: En

0 Items

IP Address Pattern Notes

[Commit] [Cancel]

Location [New] [Edit] [Delete] [Duplicate] [More Actions] [Help ?]

3 Items [Filter: Enable]

<input type="checkbox"/>	Name	Correlation	Notes
<input type="checkbox"/>	Common	<input type="checkbox"/>	SBC to PSTN
<input type="checkbox"/>	Main	<input type="checkbox"/>	Avaya SIL

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**).

The screenshot shows a web application interface for configuring adaptations. On the left is a navigation menu with 'Routing' selected. The main content area is titled 'Adaptation Details' and has a 'General' tab. The form includes the following fields and values:

- Adaptation Name:** CM-TG4-IPTF
- Module Name:** DigitConversionAdapter (selected from a dropdown)
- Module Parameter Type:** (empty dropdown)
- Egress URI Parameters:** (empty text field)
- Notes:** CM - ATT - IPTF

At the top right of the form area, there are 'Commit' and 'Cancel' buttons. A 'Help ?' link is also visible in the top right corner of the page.

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). 00000111171057 is a DNIS string sent in the Request URI by the IPTF service that is associated with Communication Manager Agent/VDN skill queue 71057.

- Enter **00000111171057** in the **Matching Pattern** column.
- Enter **15** in the **Min/Max** columns.
- Enter **15** in the **Delete Digits** column.
- Enter **71057** in the **Insert Digits** column.
- Specify that this should be applied to the SIP **destination** headers in the **Address to modify** column.
- Enter any desired notes.

Step 4 - Repeat **Step 3** for all additional IPTF DNIS numbers.

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

Digit Conversion for Outgoing Calls from SM										
Add Remove										Filter: Enable
5 Items										
<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes	
<input type="checkbox"/>	*00001111171057	*15	*15		*15	71057	destination		DNIS to VDN Conversion	
<input type="checkbox"/>	*00000111118105	*15	*15		*15	71058	destination		DNIS to VDN Conversion	
<input type="checkbox"/>	*00000111119105	*15	*15		*15	71059	destination		DNIS to VDN Conversion	
<input type="checkbox"/>	*00000111116010	*15	*15		*15	71060	destination		DNIS to VDN Conversion	
<input type="checkbox"/>	*00001111111061	*15	*15		*15	71061	destination		DNIS to VDN Conversion (ADR)	

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., **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, and when using spaces, 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.

General

* Adaptation Name:

* Module Name:

Module Parameter Type:

Add		Remove	
<input type="checkbox"/>	Name	Value	
<input type="checkbox"/>	eRHdrs	*AV-Global-Session-ID, Alert-Info, Endpoint-View, P-AV-Message-Id, P-Charging-Vector, P-	

Select : All, None

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

Add		Remove							
0 Items									
<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes

Filter: Enable

Digit Conversion for Outgoing Calls from SM

Add		Remove							
0 Items									
<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes

Filter: Enable

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., **SessionManager**).

- **FQDN or IP Address** – Enter the IP address of Session Manager signaling interface, (*not* the management interface), provisioned during installation (e.g., **10.64.91.61**).
- **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.

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.

The screenshot shows the 'SIP Entity Details' configuration page. The 'General' section includes the following fields:

- Name:** SessionManager
- FQDN or IP Address:** 10.64.91.61
- Type:** Session Manager
- Notes:** Session Manager
- Location:** Main
- Outbound Proxy:** (blank)
- Time Zone:** America/Denver
- Credential name:** (blank)

The 'SIP Link Monitoring' section has a dropdown menu set to 'Use Session Manager Configuration'. Buttons for 'Commit' and 'Cancel' are located at the top right of the form.

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.

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

Step 7 - Click on **Commit**.

Listen Ports

TCP Failover port:

TLS Failover port:

Add Remove

4 Items Filter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	avayalab.com	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5061	TLS	avayalab.com	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5064	TCP	avayalab.com	<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5065	TLS	avayalab.com	<input type="checkbox"/>	<input type="text"/>

Select : All, None

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., **10.64.91.65**).
- **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**.

Home / Elements / Routing / SIP Entities

SIP Entity Details Commit Cancel Help ?

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

* SIP Timer B/F (in seconds):

Credential name:

Securable:

Call Detail Recording:

Loop Detection

Loop Detection Mode:

Loop Count Threshold:

Loop Detection Interval (in msec):

SIP Link Monitoring

SIP Link Monitoring:

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.

The screenshot shows the 'SIP Entity Details' configuration page for a Local Trunk. The page is titled 'SIP Entity Details' and has a breadcrumb trail 'Home / Elements / Routing / SIP Entities'. There are 'Commit' and 'Cancel' buttons at the top right, and a 'Help ?' link. The configuration is organized into sections: 'General', 'Loop Detection', and 'SIP Link Monitoring'.
General
* Name: CM-TG3
* FQDN or IP Address: 10.64.91.65
Type: CH
Notes: Trunk Group 3 - CM to Enterprise
Adaptation: (empty dropdown)
Location: Main
Time Zone: America/Denver
* SIP Timer B/F (in seconds): 4
Credential name: (empty text field)
Securable:
Call Detail Recording: none
Loop Detection
Loop Detection Mode: Off
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration

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-ipv6-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 **ATT** (**Section 5.3.2**).
- **Location** – Select location **Common** (**Section 5.2.2**).

Home / Elements / Routing / SIP Entities

SIP Entity Details

Commit Cancel

General

* Name: SBCE-ipv6-Toll Free

* FQDN or IP Address: 10.64.91.41

Type: SIP Trunk

Notes: SBCE for IPv6 testing

Adaptation: SBC1-Adaptation for ATT

Location: Common

Time Zone: America/Fortaleza

* SIP Timer B/F (in seconds): 4

Credential name:

Securable:

Call Detail Recording: egress

Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

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., **SessionManager**).
- **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**.

Home / Elements / Routing / Entity Links Help ?

Entity Links Commit Cancel

1 Item Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Not
<input type="checkbox"/>	* SM to CM TG4	* SessionManager	TLS	* 5064	* CM-TG4	<input type="checkbox"/>	* 5064	trusted	<input type="checkbox"/>	

Select : All, None

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**).

Home / Elements / Routing / Entity Links Help ?

Entity Links Commit Cancel

1 Item Filter: Enable

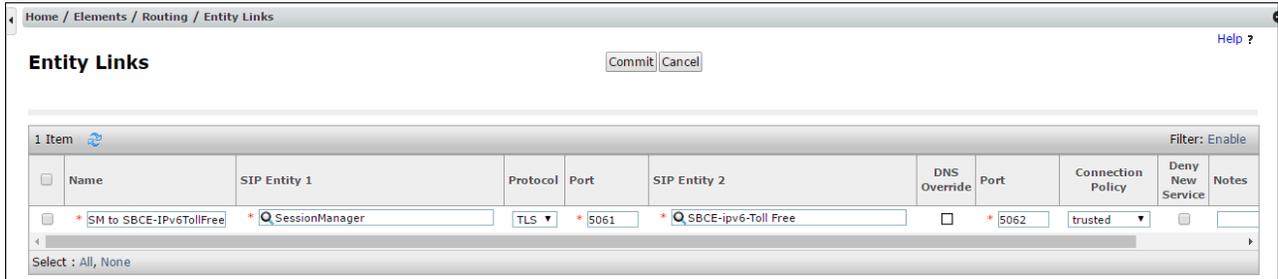
<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Not
<input type="checkbox"/>	* SM to CM TG3	* SessionManager	TLS	* 5061	* CM-TG3	<input type="checkbox"/>	* 5061	trusted	<input type="checkbox"/>	

Select : All, None

5.5.3 Entity Link for the AT&T IP Toll Free Service via the Avaya SBCE

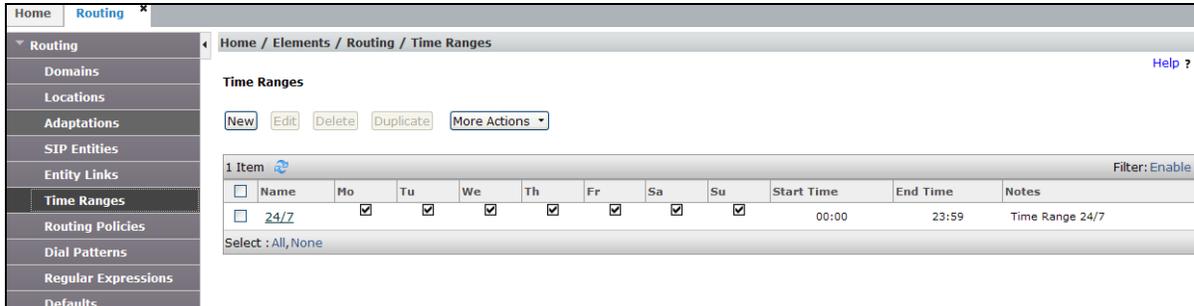
To configure this Entity Link, repeat the steps in **Section 5.5.1**, with the following changes:

- **Name** – Enter a descriptive name for this link to the Avaya SBCE (e.g., **SM to SBCE-IPv6TollFree**).
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.4** for the Avaya SBCE entity (e.g., **SBCE-ipv6-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.

The screenshot shows the 'Routing Policy Details' page. The left sidebar is expanded to 'Routing Policies'. The main content area is titled 'Routing Policy Details' and has 'Commit' and 'Cancel' buttons. Under the 'General' section, there are fields for: * Name: To CM TG4, Disabled: , * Retries: 0, and Notes: Trunk Group 4 PSTN4 to CM. Below this is the 'SIP Entity as Destination' section with a 'Select' button.

Step 4 - In the **SIP Entity 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**.

The screenshot shows the 'SIP Entities' page. The left sidebar is expanded to 'Routing Policies'. The main content area is titled 'SIP Entities' and has 'Select' and 'Cancel' buttons. Below this is a table of SIP Entities. The table has columns for Name, FQDN or IP Address, Type, and Notes. The row for CM-TG4 is selected and highlighted in blue.

Name	FQDN or IP Address	Type	Notes
<input type="radio"/> Aura Messaging	10.64.91.54	Modular Messaging	Aura Messaging
<input type="radio"/> CM-TG1	10.64.91.65	CM	Trunk Group 1
<input type="radio"/> CM-TG2	10.64.91.65	CM	Trunk Group 2
<input type="radio"/> CM-TG3	10.64.91.65	CM	Trunk Group 3 - CM to Enterprise
<input checked="" type="radio"/> CM-TG4	10.64.91.65	CM	Trunk Group 4 - ATT IPTF
<input type="radio"/> CS1000	10.80.140.103	Other	CS1000 7.65
<input type="radio"/> SBC1	10.64.91.50	SIP Trunk	Avaya SBC-1 to PSTN
<input type="radio"/> SBC2	10.64.91.100	SIP Trunk	Avaya SBC-2 to PSTN

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**.

Home / Elements / Routing / Routing Policies

Routing Policy Details Commit Cancel Help ?

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
CM-TG4	10.64.91.65	CM	Trunk Group 4 - ATT IPTF

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="text" value="2"/>	24/7	<input checked="" type="checkbox"/>	00:00	23:59							

Select: All, None

Dial Patterns

Add Remove

2 Items Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/> 00000	15	15	<input type="checkbox"/>	-ALL-	Common	ATT Inbound

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 15 digit number in the Request URI with the format 00000xxxxxxxxx. Enter **00000**.

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

- **Min and Max** – Enter **15**.
- **SIP Domain** – Select **-ALL-**, to select all of the administered SIP Domains.

Home / Elements / Routing / Dial Patterns

Dial Pattern Details Commit Cancel Help ?

General

* Pattern: 00000

* Min: 15

* Max: 15

Emergency Call:

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes: ATT Inbound

Originating Locations and Routing Policies

Add Remove

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 all Locations).

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).

Originating Location

Apply The Selected Routing Policies to All Originating Locations

3 Items Filter: Enable

<input type="checkbox"/>	Name	Notes
<input type="checkbox"/>	Common	SBC to PSTN
<input type="checkbox"/>	Main	Avaya SIL
<input type="checkbox"/>	RemoteAccess	Remote Access from SBCE1

Select : All, None

Routing Policies

8 Items Filter: Enable

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	To AAM	<input type="checkbox"/>	Aura Messaging	
<input type="checkbox"/>	To CM TG1	<input type="checkbox"/>	CM-TG1	Trunk Group 1 PSTN1 to CM
<input type="checkbox"/>	To CM TG2	<input type="checkbox"/>	CM-TG2	Trunk Group 2 PSTN2 to CM
<input type="checkbox"/>	To CM TG3	<input type="checkbox"/>	CM-TG3	Enterprise Traffic
<input checked="" type="checkbox"/>	To CM TG4	<input type="checkbox"/>	CM-TG4	Trunk Group 4 PSTN4 to CM
<input type="checkbox"/>	To CS1000	<input type="checkbox"/>	CS1000	
<input type="checkbox"/>	To SBC1	<input type="checkbox"/>	SBC1	

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

Originating Locations and Routing Policies

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Common	SBC to PSTN	To CM TG4	2	<input type="checkbox"/>	CM-TG4	Trunk Group 4 PSTN4 to CM

Select : All, None

Denied Originating Locations

Add Remove

0 Items Filter: Enable

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

Commit Cancel

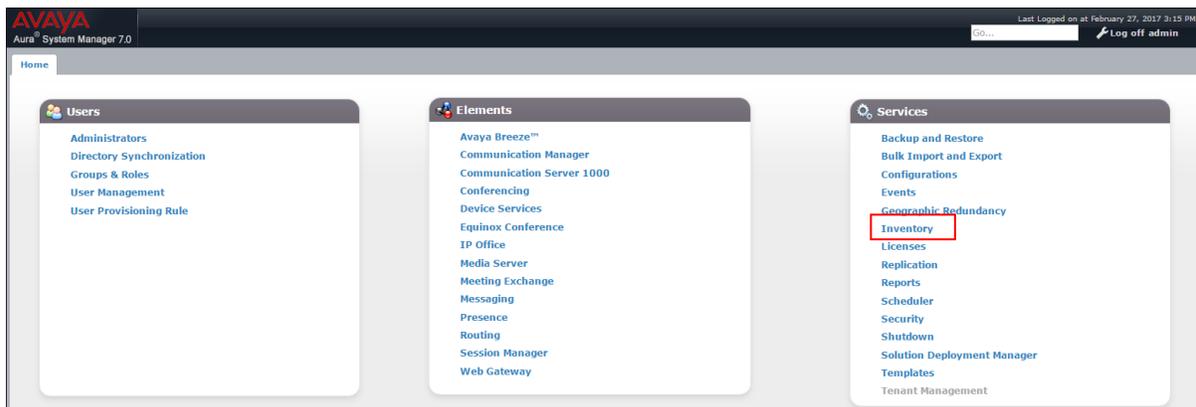
Step 7 - Repeat **Steps 1-6** for any additional inbound dial patterns from AT&T.

5.9 Verify TLS Certificates – Session Manager

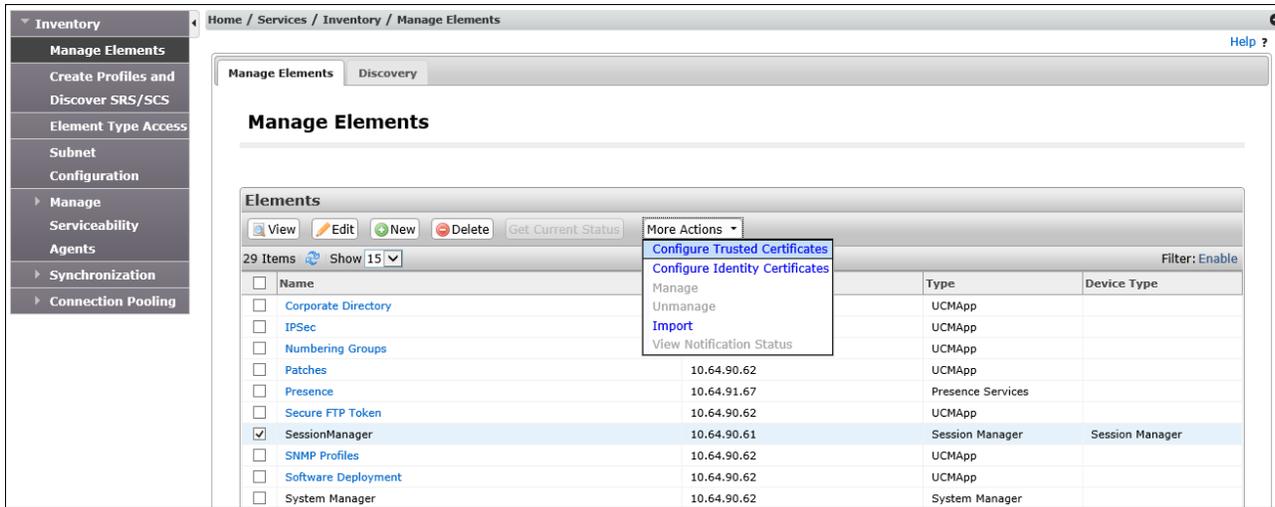
Note – Testing was done with System Manager signed identity certificates. The procedure to obtain and install certificates is outside the scope of these Application Notes.

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

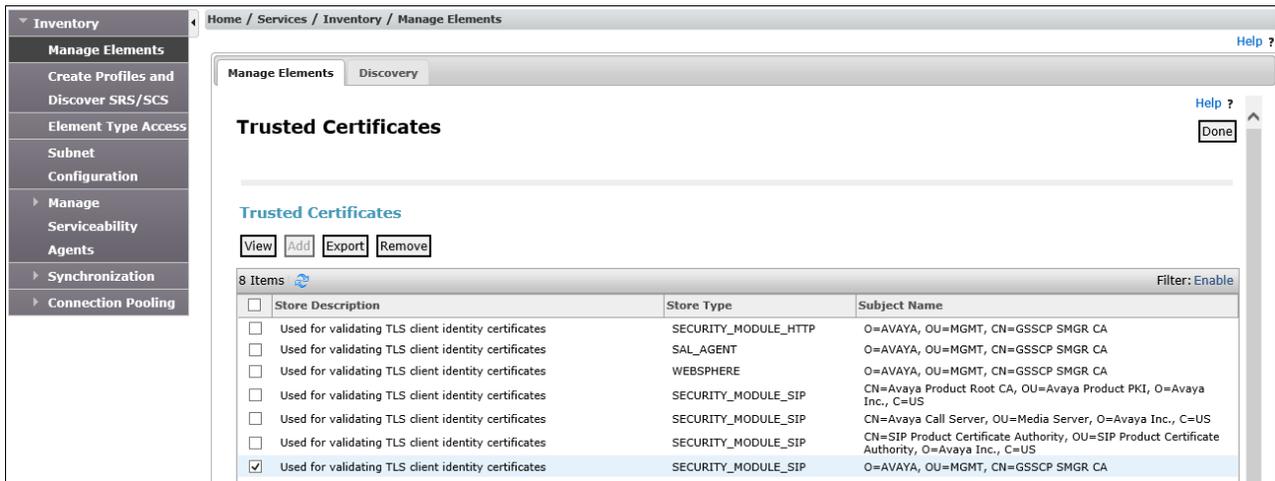
Step 1 - From the **Home** screen, under the **Services** heading in the right column, select **Inventory**.



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**.

The screenshot shows the Avaya System Manager interface. On the left is a navigation menu with options like 'Inventory', 'Manage Elements', 'Create Profiles and Discover SRS/SCS', 'Element Type Access', 'Subnet Configuration', 'Manage Serviceability Agents', 'Synchronization', and 'Connection Pooling'. The main window is titled 'Home / Services / Inventory / Manage Elements' and contains a 'Manage Elements' tab with a 'Discovery' sub-tab. Under 'Identity Certificates', there are buttons for 'Replace', 'Export', and 'Renew'. A table lists 5 items with columns for Service Name, Common Name, Valid To, Expired, and Service Description. The 'Security Module SIP' service is selected. Below the table is a 'Certificate Details' section with fields for Subject Details, Valid From, Key Size, Issuer Name, Certificate Fingerprint, and Subject Alternative Name.

Service Name	Common Name	Valid To	Expired	Service Description
<input type="radio"/> WebSphere	websphere	Fri Jan 11 16:35:30 MST 2019	No	Internal TLS communication between Security Module and WebSphere
<input type="radio"/> SPIRIT	spiritalias	Fri Jan 11 16:35:28 MST 2019	No	SPIRIT Service
<input type="radio"/> Security Module HTTPS	securitymodule_http	Fri Jan 11 16:47:22 MST 2019	No	Security Module HTTPS Service
<input checked="" type="radio"/> Security Module SIP	securitymodule_sip	Fri Jan 11 16:46:30 MST 2019	No	Security Module SIP Service
<input type="radio"/> Management	mgmt	Fri Jan 11 16:35:27 MST 2019	No	Management Service

Select : None

Certificate Details

Subject Details C=US, O=Avaya, CN=avayalab.com

Valid From Wed Jan 11 16:46:30 MST 2017

Key Size 2048

Valid To Fri Jan 11 16:46:30 MST 2019

Issuer Name O=AVAYA, OU=MGMT, CN=GSSCP SMGR CA

Certificate Fingerprint 8a3e73d4f869ec0b2f9485a7cb074f3199dfa689

Subject Alternative Name dNSName=vz-sm-7-sm100.avayalab.com, IPAdd

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 1
      Maximum Administered Remote Office Trunks: 4000 0
Maximum Concurrently Registered Remote Office Stations: 2400 0
      Maximum Concurrently Registered IP eCons: 68 0
    Max Concur Registered Unauthenticated H.323 Stations: 100 0
      Maximum Video Capable Stations: 2400 3
      Maximum Video Capable IP Softphones: 2400 4
      Maximum Administered SIP Trunks: 4000 30
    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 Private Networking and Processor Ethernet fields are 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
Personal Station Access (PSA)? y                                  System Management Data Transfer? n
  PNC Duplication? n                                             Tenant Partitioning? y
  Port Network Support? n                                         Terminal Trans. Init. (TTI)? y
  Posted Messages? y                                             Time of Day Routing? y
  Private Networking? y                                         TN2501 VAL Maximum Capacity? y
  Processor and System MSP? y                                     Uniform Dialing Plan? y
  Processor Ethernet? y                                         Usage Allocation Enhancements? y
  Remote Office? y                                               Wideband Switching? y
  Restrict Call Forward Off Net? y                                 Wireless? n
  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.

```

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, 2** and **7** 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**.

```

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
7       5    ext
8        1     fac
9        1     fac
*        3     dac
#        3     fac

```

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 **10.64.91.61**).
- Media Server (e.g., **AMS** and **10.64.91.60**). 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                 10.64.91.60
SM                  10.64.91.61
default             0.0.0.0
procr               10.64.91.65
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
Network Region: 1                                   Allow H.248 Gateways? y
                                                    Gatekeeper Priority: 5
                                                    IPV4 PARAMETERS
Node Name: procr                                     IP Address: 10.64.91.65
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., **Main**).
- 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			
dst rgn	codec set	direct	WAN	Units	WAN-BW-limits	Video	Intervening	Dyn	CAC	I	G	A	M
1	1				Total Norm	Prio	Shr	Regions		R	L	e	
											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			
dst rgn	codec set	direct	WAN	Units	WAN-BW-limits	Video	Intervening	Dyn	CAC	I	G	A	M
1	4	y	NoLimit		Total Norm	Prio	Shr	Regions		R	L	e	
											all		
2	4	y	NoLimit						n				t
3	3	y	NoLimit						n				t
4	4												

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

FAX      Mode      Redundancy      Packet      ECM: y
Modem    t.38-standard  0               Size(ms)
TDD/TTY  off             0
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 6.7.1** 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).

```

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

Mode      Redundancy      Packet
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

Mode                                     Redundancy   Packet
FAX                                     0             Size(ms)
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

Incoming Dialog Loopbacks: eliminate                Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                RFC 3389 Comfort Noise? n
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 lxC: 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

  UI 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 5.3.1**). 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
                                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

Incoming Dialog Loopbacks: eliminate                Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3                Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y                IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n                Initial IP-IP Direct Media? y
                                                    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., **601**).
- **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
Signalng 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
                                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
                                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., **000001111101061**).
- **CPN Len** – Enter the total number of digits after the digit conversion (e.g., **15**).

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 **71057**).
- **Trk Grp(s)** – Enter the number of the Public trunk group (e.g., **4**).
- **CPN Prefix** – Enter the corresponding IPTF DNIS number (e.g., **000001111171057**).
- **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 14006
                                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                     Total
Ext  Ext      Trk      CPN      Total
Len  Code      Grp(s)  Prefix   CPN
                                     Len
5  14006      4        000001111101061 15
5  71057      4        000001111171057 15
5  71058      4        000001111181058 15
5  71059      4        000001111191059 15
5  71060      4        000001111101060 15
Total Administered: 20
Maximum Entries: 240
Note: If an entry applies to
a SIP connection to Avaya
Aura(R) Session Manager,
the resulting number must
be a complete E.164 number.
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
                          NUMBERING - PRIVATE FORMAT
                          Total
Ext  Ext      Trk      Private   Total
Len  Code      Grp(s)  Prefix    Len
5  10        3        5        5
5  11        3        5        5
5  12        3        5        5
5  14        3        5        5
5  20        3        5        5
Total Administered: 6
Maximum Entries: 540

```

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
2:
3:
                n      user
                n      user
                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 14xxx, therefore enter **14**.
- **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
14      5      5      3      lev0      n
20      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:                                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
                                MIA Across Skills: system
                                ACW Agent Considered Idle: 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 71057                                     Page 1 of 3
VECTOR DIRECTORY NUMBER
Extension: 71057
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., **08IS38199678**).

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., **10.64.91.65**, 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., **08IS38199678**).
- 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 1

                                     Type: g450
                                     Name: G450-1
                                     Serial No: 08IS38199678
Link Encryption Type: any-ptls/tls                       Enable CF? n
Network Region: 1                                       Location: 1
                                                         Site Data:

Recovery Rule: 1

Registered? y
FW Version/HW Vintage: 37 .41 .0 /1
MGP IPV4 Address: 10.64.19.81
MGP IPV6 Address:
Controller IP Address: 10.64.91.65
```

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., **10.64.91.65**, 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: 10.64.91.60
```

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
```

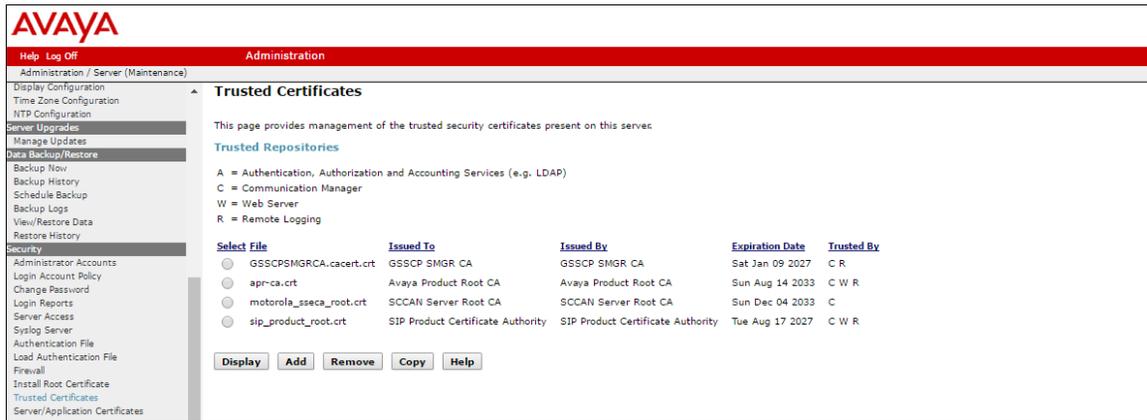
6.16 Verify TLS Certificates – Communication Manager

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.

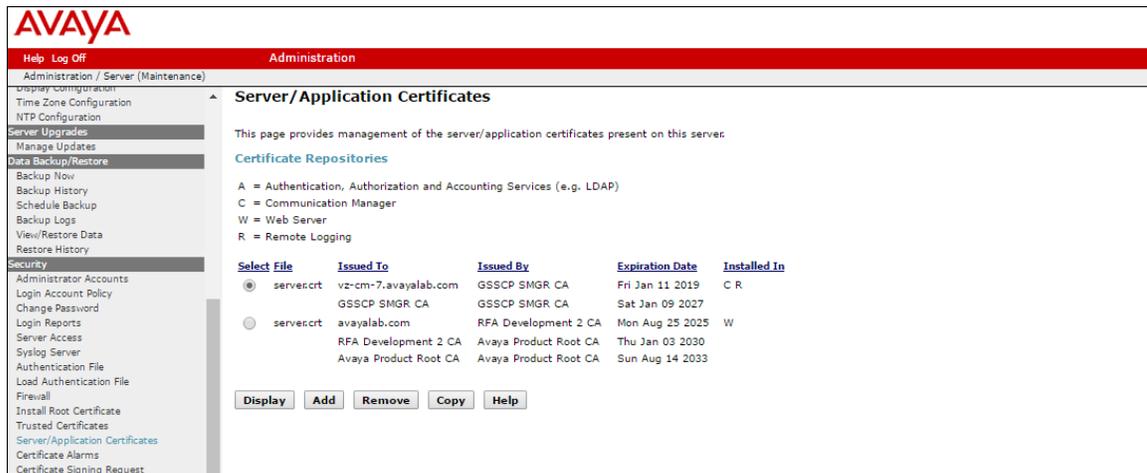
In the reference configuration, TLS transport is used for the communication between Session Manager and Communication Manager. The following procedures show how to verify the certificates used by Communication Manager.

Step 1 - From a web browser, type in “[https:// x.x.x.x](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 must 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, (IPv4 address 10.64.91.41), with access to the Main site. The connection to AT&T uses the Avaya SBCE public interface B1 (IPv6 address 3ffe:ffff:bb:bb::241). The follow 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.

The screenshot shows the Avaya Session Border Controller for Enterprise dashboard. The left sidebar contains a navigation menu with 'System Management' selected. The main content area is titled 'Dashboard' and includes several sections:

- Information:** A table showing system details:

System Time	01:23:43 PM MST	Refresh
Version	7.1.0.1-07-12368	
Build Date	Fri Nov 11 09:21:54 EST 2016	
License State	OK	
Aggregate Licensing Overages	0	
Peak Licensing Overage Count	0	
Last Logged in at	02/13/2017 10:09:44 MST	
Failed Login Attempts	2	
- Installed Devices:** A list showing 'EMS' and 'SBCE'.
- Alarms (past 24 hours):** A section stating 'None found.'
- Incidents (past 24 hours):** A section showing 'SBCE : Heartbeat Successful, Server is UP' with an 'Add' button.
- Notes:** A section stating 'No notes found.'

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.

The screenshot shows the 'System Management' interface. The left sidebar has 'System Management' selected. The main content area has tabs for 'Devices', 'Updates', 'SSL VPN', 'Licensing', and 'Key Bundles'. The 'Devices' tab is active, displaying a table of device information:

Device Name	Management IP	Version	Status						
SBCE	10.64.90.40	7.1.0.1-07-12368	Commissioned	Reboot	Shutdown	Restart Application	View	Edit	Uninstall

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.

System Information: SBCE

General Configuration

Appliance Name	SBCE
Box Type	SIP
Deployment Mode	Proxy

Device Configuration

HA Mode	No
Two Bypass Mode	No

License Allocation

Standard Sessions Requested: 50	50
Advanced Sessions Requested: 50	50
Scopia Video Sessions Requested: 5	5
CES Sessions Requested: 0	0
Transcoding Sessions Requested: 50	50
Encryption	<input checked="" type="checkbox"/>

Network Configuration

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.64.91.41	10.64.91.41	255.255.255.0	10.64.91.1	A1
10.64.91.41	10.64.91.41	255.255.255.0	10.64.91.1	A1
10.64.91.41	10.64.91.41	255.255.255.0	10.64.91.1	B2
3ffe:ffff:bb:bb::241	3ffe:ffff:bb:bb::241	64	3ffe:ffff:bb:bb::1	B1
3ffe:ffff:bb:bb::241	3ffe:ffff:bb:bb::241	64	3ffe:ffff:bb:bb::1	B1

DNS Configuration

Primary DNS	10.64.90.201
Secondary DNS	
DNS Location	DMZ
DNS Client IP	10.64.91.40

Management IP(s)

IP #1 (IPv4)	10.64.90.40
--------------	-------------

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**.

WARNING: Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems.

TLS Profile

Profile Name: sbc40-server

Certificate: sbc40.crt

Certificate Verification

Peer Verification: None

Peer Certificate Authorities: GSSCPSMGRCA.pem

Peer Certificate Revocation Lists:

Verification Depth:

Next

The following screen shows the completed TLS Server Profile form:

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The main heading is "Session Border Controller for Enterprise" with the Avaya logo in the top right corner. A left-hand navigation menu includes: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, PPM Services, Domain Policies, TLS Management (expanded), Certificates, Client Profiles, **Server Profiles**, and Device Specific Settings. The main content area is titled "Server Profiles: sbc40-server" and contains a list of server profiles with "sbc40-server" selected. An "Add" button is visible above the list, and a "Delete" button is in the top right of the main content area. The "Server Profile" form for "sbc40-server" is shown with the following details:

Server Profile	
Click here to add a description.	
TLS Profile	
Profile Name	sbc40-server
Certificate	sbc40.crt
Certificate Verification	
Peer Verification	None
Extended Hostname Verification	<input type="checkbox"/>
Renegotiation Parameters	
Renegotiation Time	0
Renegotiation Byte Count	0
Handshake Options	
Version	<input checked="" type="checkbox"/> TLS 1.2 <input type="checkbox"/> TLS 1.1 <input type="checkbox"/> TLS 1.0
Ciphers	<input checked="" type="radio"/> Default <input type="radio"/> FIPS <input type="radio"/> Custom
Value	HIGH:DH:1ADH:IMD5:1aNULL:1eNULL:@STRENGTH

An "Edit" button is located at the bottom of the 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., **GSSCPSMGRCA.pem**.
- **Verification Depth:** enter **1**
- Click **Next**.

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

Edit Profile X

WARNING: Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems.

TLS Profile

Profile Name: sbc40-client

Certificate: sbc40.crt

Certificate Verification

Peer Verification: Required

Peer Certificate Authorities: GSSCPSMGRCA.pem

Peer Certificate Revocation Lists:

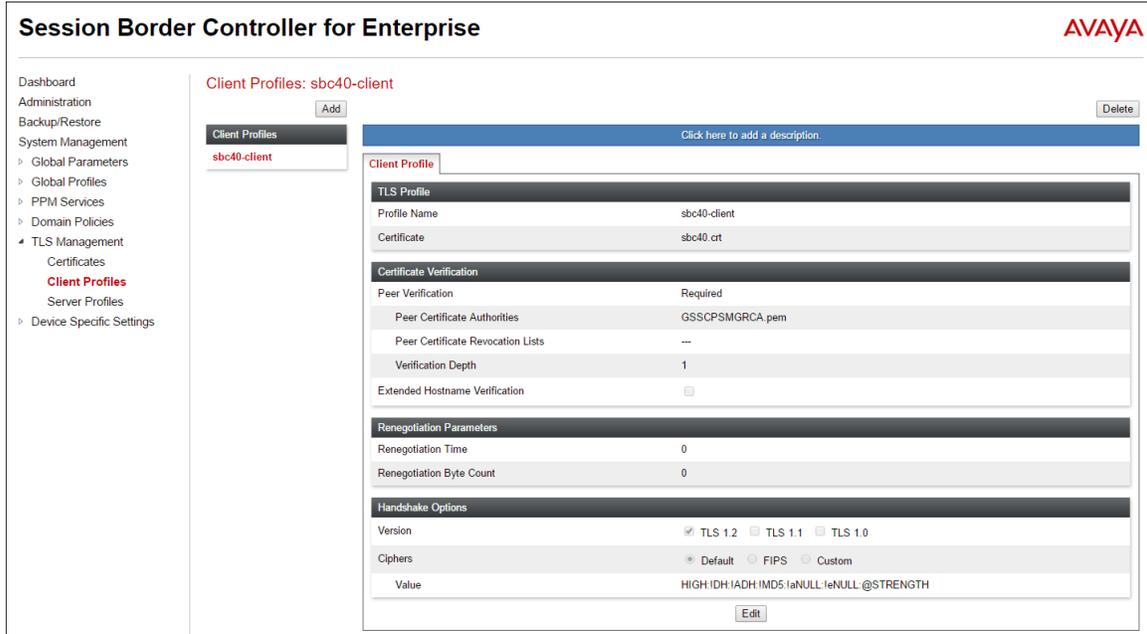
Verification Depth: 1

Extended Hostname Verification:

Custom Hostname Override:

Next

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



7.3 Global Profiles

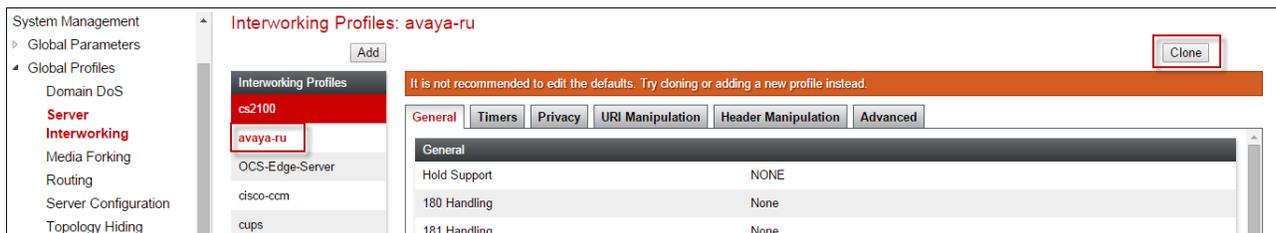
Global Profiles allow for configuration of parameters across the Avaya SBCE appliances.

7.3.1 Server Interworking – Avaya

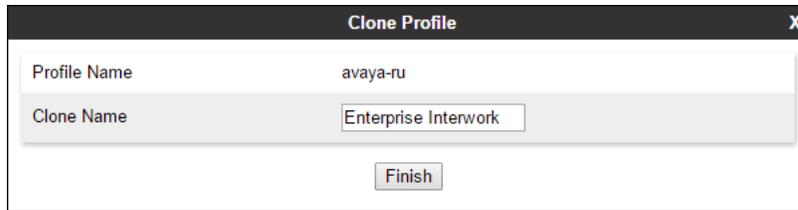
Server Interworking allows users to configure and manage various SIP call server-specific capabilities such as call hold and T.38 faxing. This section defines the connection to Session Manager.

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.

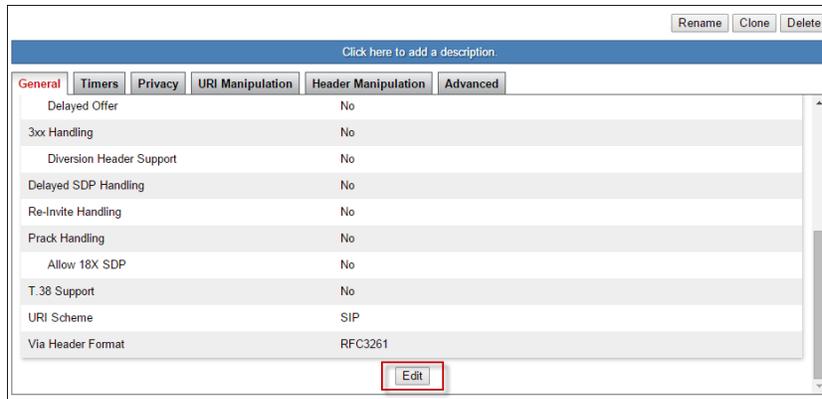


Step 3 - Enter profile name: (e.g., **Enterprise Interwork**), and click **Finish**.



The image shows a 'Clone Profile' dialog box with a dark header and a light body. It contains two text input fields: 'Profile Name' with the value 'avaya-ru' and 'Clone Name' with the value 'Enterprise Interwork'. A 'Finish' button is located at the bottom center.

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



The image shows a profile configuration screen with a blue header and a white body. At the top right are 'Rename', 'Clone', and 'Delete' buttons. Below the header is a blue bar with the text 'Click here to add a description.' Below this are several tabs: 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'General' tab is selected. The main area contains a table of settings:

Delayed Offer	No
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
Prack Handling	No
Allow 18X SDP	No
T.38 Support	No
URI Scheme	SIP
Via Header Format	RFC3261

An 'Edit' button is located at the bottom center of the screen, highlighted with a red box.

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

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

The screenshot shows a window titled "Editing Profile: Enterprise Interwork" with a close button (X) in the top right corner. The window contains a "General" tab with the following settings:

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

At the bottom of the window is a "Finish" button.

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

The screenshot shows a configuration window titled "Editing Profile: Enterprise Interwork". It contains several sections with radio buttons and checkboxes. The "Record Routes" section has "None" selected. The "Include End Point IP for Context Lookup" checkbox is unchecked. The "Extensions" dropdown is set to "Avaya". The "Diversion Manipulation" checkbox is unchecked, and the "Diversion Condition" dropdown is set to "None". The "Diversion Header URI" field is empty. The "Has Remote SBC" checkbox is checked. The "Route Response on Via Port" checkbox is unchecked. The "DTMF" section has "DTMF Support" with "None" selected. A "Finish" button is located at the bottom of the window.

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 8.4.3**. 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:

```
Title contact_param_bandwidth Save
1 within session "ALL"
2 {
3     act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
4     {
5
6     //Remove gsid and epv parameters from Contact header to hide internal topology
7     remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
8     remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);
9
```

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:

```
Title contact_param_bandwidth Save
1 within session "ALL"
2 {
3     act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
4     {
5
6     //Remove gsid and epv parameters from Contact header to hide internal topology
7     remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
8     remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);
9
10    //Remove Bandwidth from SDP
11    %BODY[1].regex_replace("b=(TIAS|AS|CT): (\d+)\r\n", "");
12    }
13
```

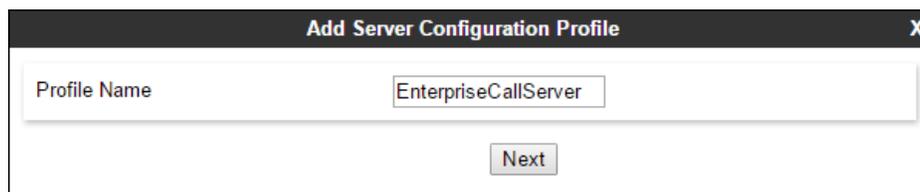
Step 3 - Click on **Save**. The script editor will test for any errors, and the window will close. This script is applied to the AT&T Server Configuration in **Section 7.3.4, Step 3**.

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., **EnterpriseCallServer**) and click **Next**.



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

- Select **Server Type: Call Server**.

- **IP Address/FQDN: 10.64.91.61** (Session Manager network IP Address)
- **Port: 5061.**
- **Select Transport: TLS.**
- **TLS Client Profile: sbc40-client.**
- **Select Next.**

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

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-IPv6-trk-svr**) and select **Next**.

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

- Select **Server Type: Trunk Server**
- **IP Address/FQDN: 3ffe:ffff:aa:aa:10:10:172:80** (AT&T Border Element IPv6 address)
- **Port: 5060**
- Select **Transport: UDP**
- Select **Next** until the **Advanced** tab is reached

Note – The IPv6 address needs to be entered using lowercase characters. See **Section 2.2, Item 9** for limitations in entering an IPv6 address.

The screenshot shows the 'Server Configuration: ATT-IPv6-trk-svr' window with the 'General' tab selected. The 'Server Type' is set to 'Trunk Server'. Below this, a table displays the configuration details:

IP Address / FQDN	Port	Transport
3ffe:ffff:aa:aa:10:10:172:80	5060	UDP

An 'Edit' button is located below the table. The left sidebar shows the navigation menu with 'Server Configuration' highlighted.

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).

The screenshot shows the 'Server Configuration: ATT-IPv6-trk-svr' window with the 'Advanced' tab selected. The configuration options are as follows:

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	ATT-Interworking
Signaling Manipulation Script	contact_param_bandwidth
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>

An 'Edit' button is located at the bottom right of the configuration area. The left sidebar shows the navigation menu with 'Server Configuration' highlighted.

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**.

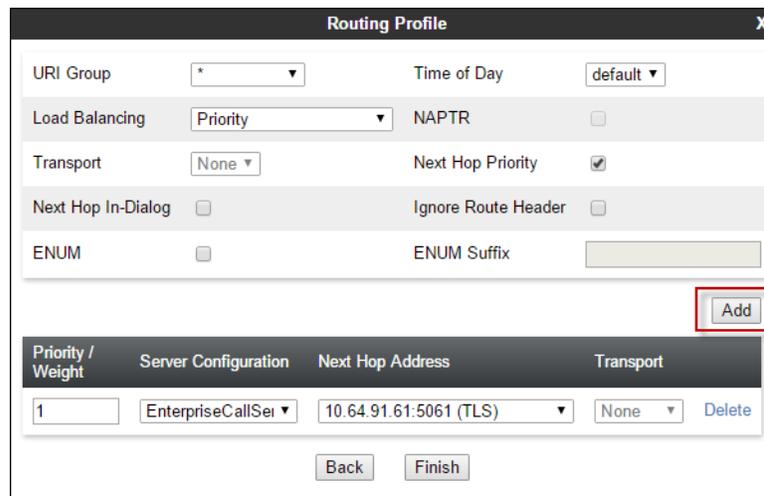


The screenshot shows a window titled "Routing Profile" with a close button (X) in the top right corner. Below the title bar, there is a text input field labeled "Profile Name" containing the text "To SM". Below the input field is a "Next" button.

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 **10.64.91.61:5061 (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**.



The screenshot shows the "Routing Profile" window with various configuration options. The "Add" button is highlighted with a red box. Below the configuration options is a table with the following columns: Priority / Weight, Server Configuration, Next Hop Address, and Transport. The table contains one entry with the following values: Priority / Weight: 1, Server Configuration: EnterpriseCallSer, Next Hop Address: 10.64.91.61:5061 (TLS), and Transport: None. There is also a "Delete" button next to the entry. At the bottom of the window are "Back" and "Finish" buttons.

Priority / Weight	Server Configuration	Next Hop Address	Transport
1	EnterpriseCallSer	10.64.91.61:5061 (TLS)	None

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 IPv6**).

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

- **Priority/Weight** = 1
- **Server Configuration** = **ATT-IPv6-trk-svr** (from **Section 7.3.4**).

- **Next Hop Address:** Verify that the [3ffe:ffff:aa:aa:10:10:172:80]: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**).

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

Finish

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**.

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

Finish

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

Session Border Controller for Enterprise AVAYA

Dashboard
Administration
Backup/Restore
System Management
‣ Global Parameters
‣ Global Profiles
‣ Domain DoS
‣ Server Interworking
‣ Media Forking
‣ Routing
‣ Server Configuration
Topology Hiding
‣ Signaling
‣ Manipulation
‣ URI Groups
‣ SNMP Traps

Topology Hiding Profiles: SIP-Trunk-Topology Rename Clone Delete

Topology Hiding Profiles: default cisco_th_profile Enterprise-Topology **SIP-Trunk-Topology** IPOSE-Topology Add

Click here to add a description.

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

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.

Application Rules: sip-trunk Rename Clone Delete

Application Rules: default default-trunk default-subscriber-low default-subscriber-high default-server-low default-server-high **sip-trunk** RW app rule Add Filter By Device...

Click here to add a description.

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

Miscellaneous

CDR Support	None
RTCP Keep-Alive	No

Edit

7.4.2 Media Rules

Media Rules are used to define QoS parameters. Separate media rules are create 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.

The screenshot displays the configuration page for the 'enterprise med rule' under the 'Media Rules' section. The left-hand navigation menu includes options like Dashboard, Administration, System Management, and Domain Policies, with 'Media Rules' highlighted. The main content area shows a list of media rules, with 'enterprise med rule' selected. The configuration details for this rule are shown in a tabbed interface with the 'Encryption' tab active. The 'Audio Encryption' section is expanded, showing 'Preferred Formats' set to 'SRTP_AES_CM_128_HMAC_SHA1_80 RTP', 'Encrypted RTCP' and 'MKI' as unchecked, 'Lifetime' as 'Any', and 'Interworking' as checked. The 'Video Encryption' section shows 'Preferred Formats' as 'RTP' and 'Interworking' as checked. The 'Miscellaneous' section shows 'Capability Negotiation' as checked. An 'Edit' button is visible at the bottom of the configuration area.

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.

The screenshot shows the configuration page for the 'att med rule' Media Rule. The left sidebar contains a navigation menu with categories like Administration, System Management, and Domain Policies. The main content area is titled 'Media Rules: att med rule' and includes a list of rules on the left and a configuration panel on the right. The configuration panel has tabs for Encryption, Codec Prioritization, Advanced, and QoS. The QoS tab is active, showing settings for Media QoS Marking (Enabled), Audio QoS (Audio DSCP: EF), and Video QoS (Video DSCP: EF). Buttons for 'Add', 'Filter By Device...', 'Rename', 'Clone', 'Delete', and 'Edit' are visible.

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**.

Signaling QoS

Enabled

ToS

Precedence Routine 000

ToS Minimize Delay 1000

DSCP

Value EF 101110

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**.

Signaling QoS

Enabled

ToS

Precedence Routine 000

ToS Minimize Delay 1000

DSCP

Value EF 101110

Finish

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.

The screenshot shows the 'Policy Groups: enterprise policy' configuration page. On the left is a navigation menu with 'End Point Policy Groups' highlighted. The main area shows a list of policy groups with 'enterprise policy' selected and highlighted in red. Below the list is a detailed view of the selected policy group, including a table of rules.

Order	Application	Border	Media	Security	Signaling
1	sip-trunk	default	enterprise med rule	default-low	enterprise sig rule

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).

The screenshot shows the 'Policy Groups: att-policy-group' configuration page. The navigation menu is the same as in the previous screenshot. The main area shows a list of policy groups with 'att-policy-group' selected and highlighted in red. Below the list is a detailed view of the selected policy group, including a table of rules.

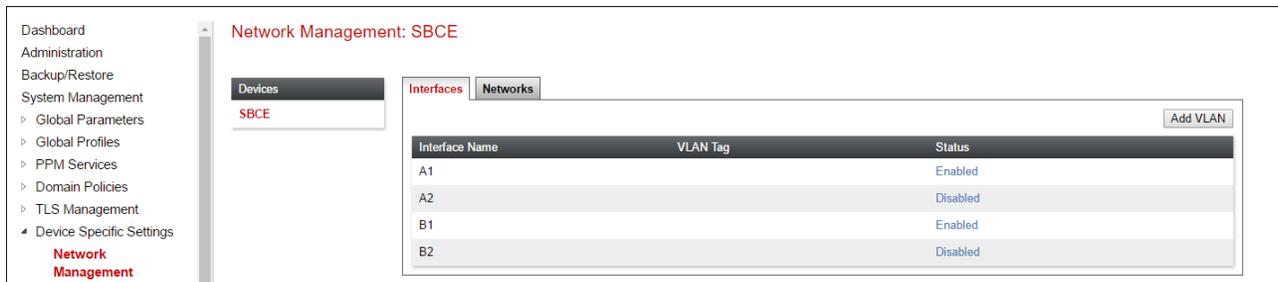
Order	Application	Border	Media	Security	Signaling
1	sip-trunk	default	att med rule	default-low	att sig rule

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 (IPv6 public) interfaces are used.



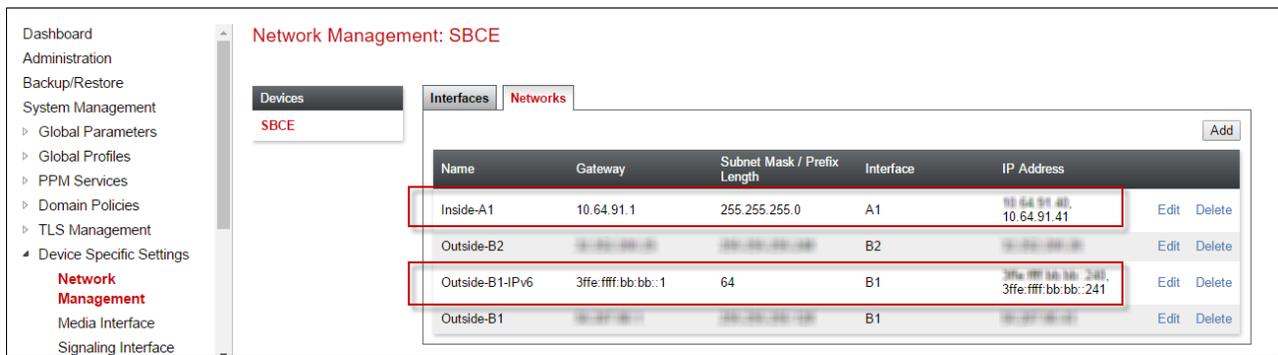
Network Management: SBCE

Devices: SBCE

Interfaces Networks

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

Step 3 - Select the **Networks** tab to display the IP provisioning for the A1 and B2 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.



Network Management: SBCE

Devices: SBCE

Interfaces Networks

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	Edit	Delete
Inside-A1	10.64.91.1	255.255.255.0	A1	10.64.91.40, 10.64.91.41		
Outside-B2	10.255.255.254	255.255.255.255	B2	10.255.255.254	Edit	Delete
Outside-B1-IPv6	3ffe:ffff:bb:bb::1	64	B1	3ffe:ffff:bb:bb::240, 3ffe:ffff:bb:bb::241	Edit	Delete
Outside-B1	10.255.255.1	255.255.255.255	B1	10.255.255.1	Edit	Delete

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 **7000 – 16000**

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**).

The screenshot shows the 'Advanced Options: SBCE' configuration page. The 'Port Ranges' tab is selected. A warning message states: 'Changes to the settings below require an application restart before taking effect. Application restarts can be issued from System Management.' The 'Port Range Configuration' table is as follows:

Port Range Configuration	
Signaling Port Range	12000 - 16380
Config Proxy Internal Signaling Port Range	42000 - 51000
Listen Port Range	6000 - 6999
HTTP Port Range	51001 - 62000

A 'Save' button is located at the bottom of the configuration area.

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 IPv4 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-IPv6-TF**.
- **IP Address:** **33fe:fff:bb:bb::241** (Avaya SBCE B1 IPv6 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.

The screenshot displays the 'Media Interface: SBCE' configuration page. On the left is a navigation menu with 'Device Specific Settings' expanded to 'Media Interface'. The main content area shows a table of media interfaces. A red box highlights the 'Outside-Media-IPv6-TF' and 'Inside-Media-TollFree' entries.

Name	Media IP Network	Port Range	Edit	Delete
Outside-B1-IPV6	33fe:fff:bb:bb::241 Outside-B1-IPV6 (B1, VLAN 0)	16384 - 32767	Edit	Delete
Inside-Media-TollFree	10.64.91.40 Inside-A1 (A1, VLAN 0)	16384 - 32767	Edit	Delete
Outside-Media-IPV4	10.64.91.41 Outside-B1-IPV4 (B1, VLAN 0)	16384 - 32767	Edit	Delete
Outside-Media-IPV6-TF	33fe:fff:bb:bb::241 Outside-B1-IPV6 (B1, VLAN 0)	16384 - 32767	Edit	Delete
Outside-Media-IPV4	10.64.91.40 Outside-B1-IPV4 (B1, VLAN 0)	16384 - 32767	Edit	Delete
Outside-Media-IPV6-TF	33fe:fff:bb:bb::241 Outside-B1-IPV6 (B1, VLAN 0)	16384 - 32767	Edit	Delete
Inside-Media-TollFree	10.64.91.41 Inside-A1 (A1, VLAN 0)	16384 - 32767	Edit	Delete

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 IPv4 address).
- **TLS Port: 5062.**
- **TLS Profile: sbc40-server.**

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

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

- **Name: Outside-Signaling-IPv6-TF.**
- **IP Address: 3ffe:ffff:bb:bb::241** (Avaya SBCE B1 IPv6 address).
- **UDP Port: 5060.**

Step 6 - Click **Finish** (not shown). Note that changes to these values require an application restart (see **Section 7.1**).

Signaling Interface: SBCE

Devices: SBCE

Signaling Interface

Modifying or deleting an existing signaling interface will require an application restart before taking effect. Application restarts can be issued from System Management.

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile	
Inside-Sig-TollFree-41	10.64.91.41 Inside-A1 (A1, VLAN 0)	---	5060	---	None	Edit Delete
Outside-Sig-TollFree-41	10.64.91.41 Outside-B1 (B1, VLAN 0)	---	5060	---	None	Edit Delete
Inside-Sig-TF	10.64.91.41 Inside-A1 (A1, VLAN 0)	---	---	5061	sbc40-server	Edit Delete
Outside-Signaling-IPv6-TF	3ffe:ffff:bb:bb::241 Outside-B1-IPv6 (B1, VLAN 0)	---	5060	---	None	Edit Delete
Inside-Sig-TollFree-41	10.64.91.41 Inside-A1 (A1, VLAN 0)	---	---	5062	sbc40-server	Edit Delete
Outside-Signaling-IPv6-TF	3ffe:ffff:bb:bb::241 Outside-B1-IPv6 (B1, VLAN 0)	---	5060	---	None	Edit Delete

7.5.5 Endpoint Flows – For Enterprise

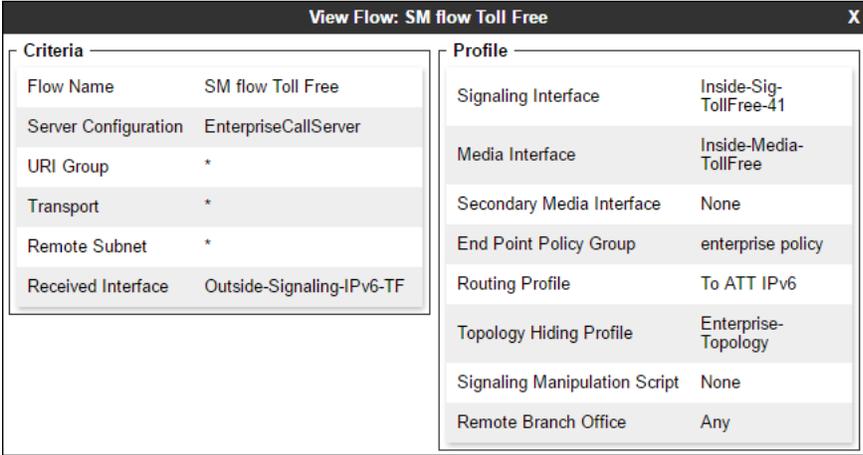
Step 1 - Select **Device Specific Settings** → **Endpoint Flows** from the menu on the left-hand side (not shown).

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

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

- **Name:** SM flow Toll Free
- **Server Configuration:** EnterpriseCallServer (Section 7.3.3)
- **URI Group:** *
- **Transport:** *
- **Remote Subnet:** *
- **Received Interface:** Outside-Signaling-IPv6-TF (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 IPv6 (Section 7.3.6)
- **Topology Hiding Profile:** Enterprise-Topology (Section 7.3.7)
- Let other values default

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



Criteria		Profile	
Flow Name	SM flow Toll Free	Signaling Interface	Inside-Sig-TollFree-41
Server Configuration	EnterpriseCallServer	Media Interface	Inside-Media-TollFree
URI Group	*	Secondary Media Interface	None
Transport	*	End Point Policy Group	enterprise policy
Remote Subnet	*	Routing Profile	To ATT IPv6
Received Interface	Outside-Signaling-IPv6-TF	Topology Hiding Profile	Enterprise-Topology
		Signaling Manipulation Script	None
		Remote Branch Office	Any

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-IPv6-Toll-Free
- **Server Configuration:** ATT-IPv6-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-IPv6-TF (Section 7.5.4)

- **Media Interface: Outside-Media-IPv6-TF (Section 7.5.3)**
- **End Point Policy Group: att-policy-group (Section 7.4.5)**
- **Routing Profile: To SM (Section 7.3.5)**
- **Topology Hiding Profile: SIP-Trunk-Topology (Section 7.3.8)**

Criteria		Profile	
Flow Name	ATT-IPv6-Toll-Free	Signaling Interface	Outside-Signaling-IPv6-TF
Server Configuration	ATT-IPv6-trk-svr	Media Interface	Outside-Media-IPv6-TF
URI Group	*	Secondary Media Interface	None
Transport	*	End Point Policy Group	att-policy-group
Remote Subnet	*	Routing Profile	To SM
Received Interface	Inside-Sig-TollFree-41	Topology Hiding Profile	SIP-Trunk-Topology
		Signaling Manipulation Script	None
		Remote Branch Office	Any

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

End Point Flows: SBCE

Devices: SBCE

Subscriber Flows | Server Flows

Click here to add a row description.

Server Configuration: ATT-IPv6-trk-svr

Update

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	ATT-IPv6 Flow	*	Inside-Sig-40	Outside-Signaling-IPv6	att-policy-group	To SM	View	Clone	Edit	Delete
2	ATT-IPv6-Toll-Free	*	Inside-Sig-TollFree-41	Outside-Signaling-IPv6-TF	att-policy-group	To SM	View	Clone	Edit	Delete

Server Configuration: ATT-TollFree-trk-svr

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	ATT-IP TF	*	Inside-Sig-TollFree-41	Outside-Signaling	att-policy-group	To IPOSE	View	Clone	Edit	Delete

Server Configuration: ATT-trk-svr

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	IPFR flow	*	Inside-Sig-40	Outside-B2-Signaling	att-policy-group	To IPOSE	View	Clone	Edit	Delete

Server Configuration: EnterpriseCallServer

Update

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Session Manager flow	*	Outside-Signaling-IPv6	Inside-Sig-40	enterprise policy	To ATT IPv6	View	Clone	Edit	Delete
2	SM Production	*	Outside-B2-Signaling	Inside-Sig-40	enterprise policy	To ATT IPFR	View	Clone	Edit	Delete
3	SM flow Toll Free	*	Outside-Signaling-IPv6-TF	Inside-Sig-TollFree-41	enterprise policy	To ATT IPv6	View	Clone	Edit	Delete

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 71060, before sending the INVITE to Communication Manager.

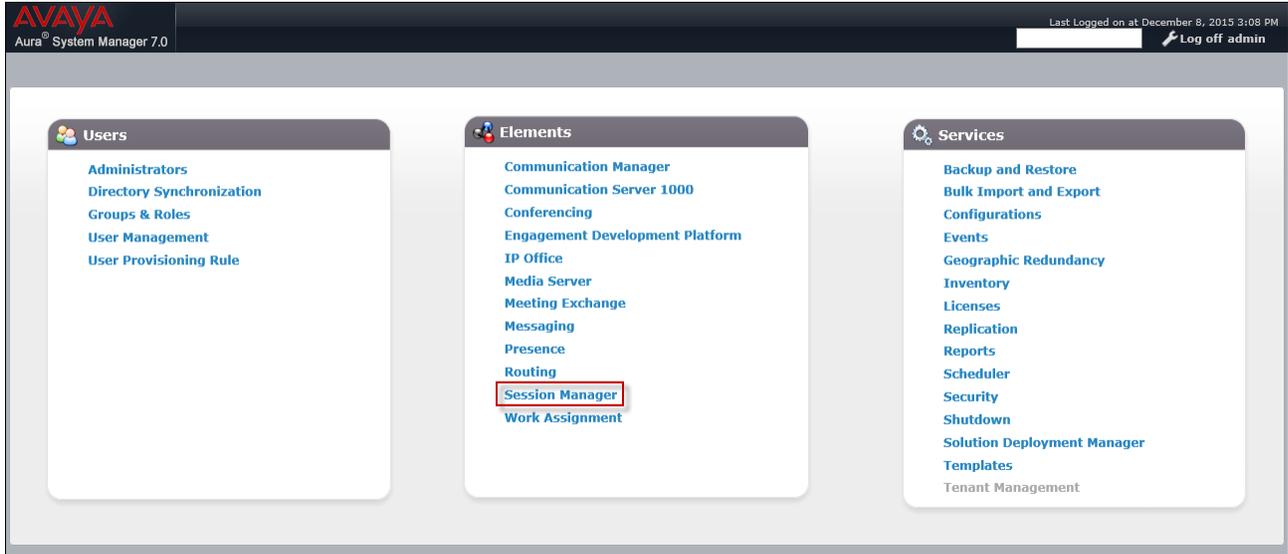
```
list trace tac *04                                     Page 1
                                                    LIST TRACE
time          data
10:07:02 TRACE STARTED 04/10/2017 CM Release String cold-00.0.441.0-23523
10:07:27 SIP<INVITE sip:71060@avayalab.com SIP/2.0
10:07:27      Call-ID: 2c36d935b7113b662d50240d11ec4e78
10:07:27      active trunk-group 4 member 1      cid 0xc90
10:07:27      0 0 ENTERING TRACE cid 3216
10:07:27      60 1 vdn e71060 bsr appl 0 strategy 1st-found override n
10:07:27      60 1 wait 2 secs hearing ringback
10:07:27 SIP>SIP/2.0 180 Ringing
10:07:27      Call-ID: 2c36d935b7113b662d50240d11ec4e78
10:07:27      dial 71060
10:07:27      ring vector 60      cid 0xc90
10:07:27      G729 ss:off ps:30
10:07:27      rgn:4 [10.64.91.41]:16580
10:07:27      rgn:1 [10.64.91.60]:6128
10:07:29      60 2 collect 1 digits after annc 11001 for none
```

- 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

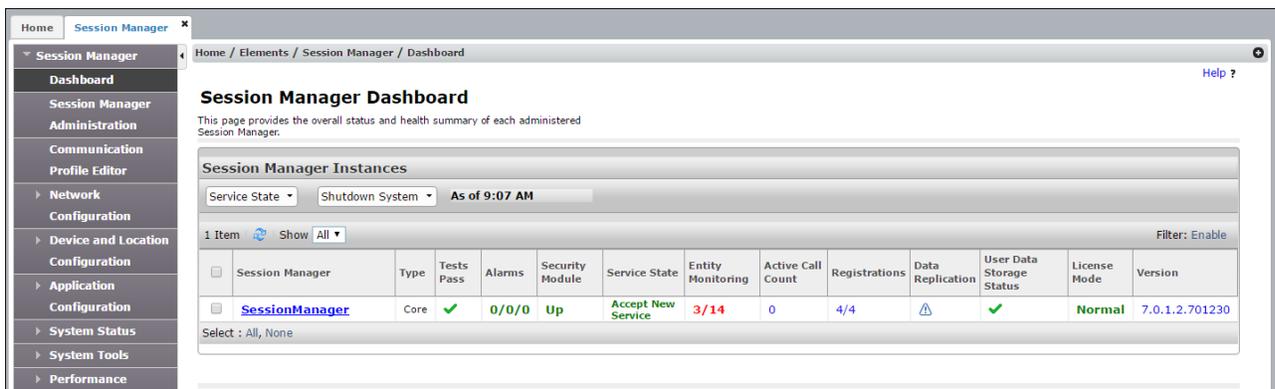
The Session Manager configuration may be verified via System Manager.

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 are **0** (zero) alarms out of the **8** Entities defined in the shared test environment.



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

All Entity Links for Session Manager: SessionManager

Summary View

Status Details for the selected Session Manager:

11 Items | Refresh Filter: Disable, Apply, Clear

SIP Entity Name		SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
Aura Messaging	<input type="radio"/>	10.64.91.54	5061	TLS	FALSE	UP	200 OK	UP
Breeze	<input type="radio"/>	10.64.91.67	5061	TLS	FALSE	UP	200 OK	UP
CM-TG1	<input type="radio"/>	10.64.91.65	5081	TLS	FALSE	UP	200 OK	UP
CM-TG2	<input type="radio"/>	10.64.91.65	5071	TLS	FALSE	UP	200 OK	UP
CM-TG3	<input type="radio"/>	10.64.91.65	5061	TLS	FALSE	UP	200 OK	UP
CM-TG4	<input type="radio"/>	10.64.91.65	5064	TLS	FALSE	UP	200 OK	UP
CM-TG5	<input type="radio"/>	10.64.91.65	5065	TLS	FALSE	UP	200 OK	UP
IP500	<input type="radio"/>	10.64.19.70	5061	TLS	FALSE	UP	200 OK	UP
Presence	<input type="radio"/>	10.64.91.67	5061	TLS	FALSE	UP	200 OK	UP
SBCE-ipv6	<input type="radio"/>	10.64.91.40	5061	TLS	FALSE	UP	405 Method Not Allowed	UP

< Previous | Page 1 of 2 | Next >

Note – The **SBCE-ipv6** 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.

The screenshot shows the Avaya SBCE dashboard. At the top, there is a navigation bar with 'Alarms 1', 'Incidents', 'Status', 'Logs', 'Diagnostics', and 'Users'. The main header reads 'Session Border Controller for Enterprise' with the AVAYA logo on the right. A sidebar on the left contains navigation links under 'Dashboard', including 'Administration', 'Backup/Restore', 'System Management', and 'Device Specific Settings'. The main content area is divided into several sections: 'Information' (System Time, Version, Build Date, License State, etc.), 'Installed Devices' (listing EMS and SBCE), 'Alarms (past 24 hours)' (None found), and 'Incidents (past 24 hours)' (listing heartbeat status for SBCE). A red box highlights the incident list.

8.4.1 Protocol Traces

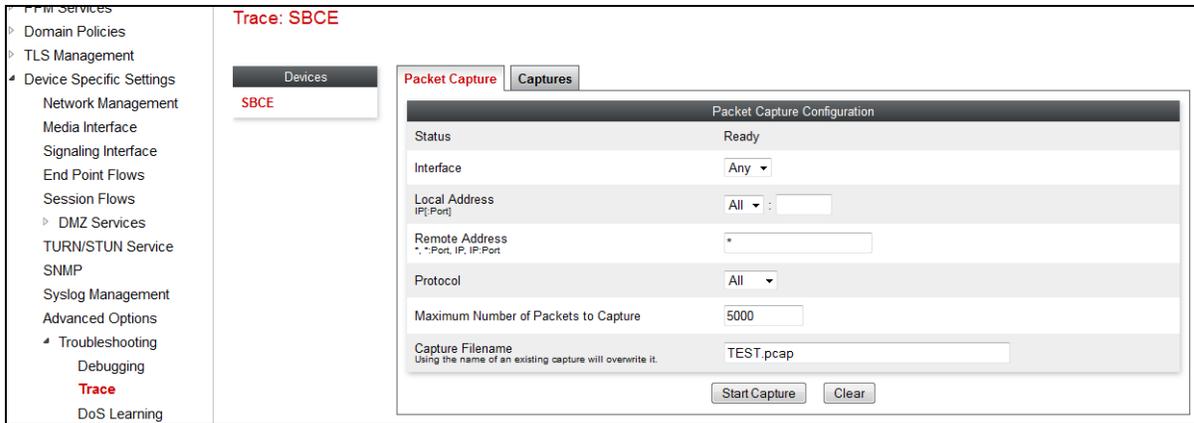
The Avaya SBCE can take internal traces of specified interfaces.

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

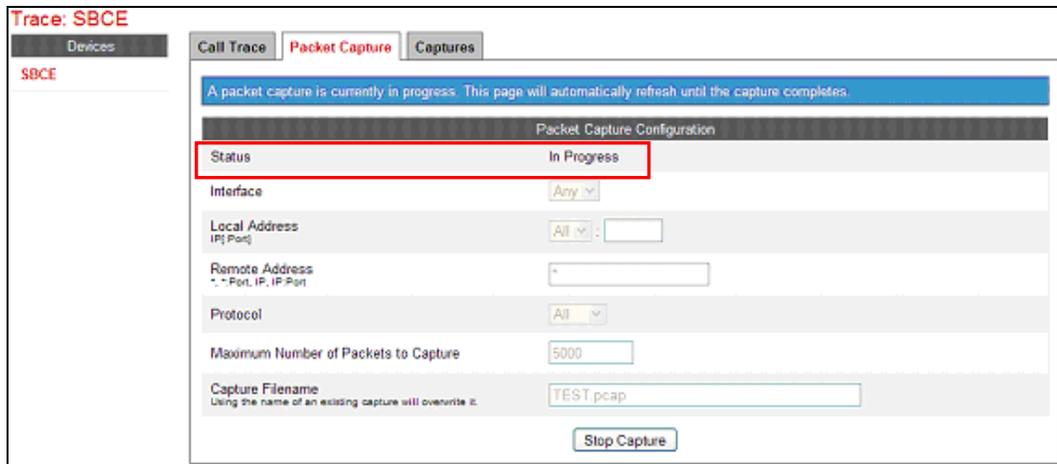
Step 2 - Select the **Packet Capture** tab and select the following:

- Select the desired **Interface** from the drop down menu (e.g., **All**).
- Specify the **Maximum Number of Packets to Capture** (e.g., **5000**).
- 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 7.0.1, Avaya Aura® Session Manager 7.0.1, and the Avaya Session Border Controller for Enterprise 7.1, can be configured to interoperate successfully with the AT&T IP Toll Free service using IPv6, 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, Release 7.0.1, Issue 3, November 2016
- [2] Administering Avaya Aura® Session Manager, Release 7.0.1, Issue 2, May 2016
- [3] Deploying Avaya Aura® System Manager, Release 7.0.1, Issue 2, August 2016
- [4] Administering Avaya Aura® System Manager for Release 7.0.1, Issue 4, April 2017

Avaya Aura® Communication Manager

- [5] Deploying Avaya Aura® Communication Manager, Release 7.0.1, Issue 3, April 2017
- [6] Administering Avaya Aura® Communication Manager, Release 7.0.1, 03-300509, Issue 2.1, August 2016
- [7] Administering Avaya G450 Branch Gateway, Release 7.0.1, 03-603228, Issue 2, May 2016
- [8] Deploying and Updating Avaya Aura® Media Server Appliance, Release 7.7, Issue 3, May 2016
- [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 7.1, Issue 3, May 2017
- [12] Deploying Avaya Session Border Controller for Enterprise, Release 7.1, Issue 2, January 2017

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.

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