Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 8.0.1 with AT&T IP Flexible Reach - Enhanced Features – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 8.0.1, with the AT&T IP Flexible Reach - Enhanced Features service, using AT&T's **AVPN** or **MIS/PNT** transport connections.

The AT&T Flexible Reach is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for their voice communication needs. The AT&T IP Flexible Reach-Enhanced Features service is a SIP based service which includes additional network-based features which are not part of IP Flexible Reach 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® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise Release 8.0.1, with the AT&T IP Flexible Reach - Enhanced Features service using AVPN or MIS/PNT transport connections.

Avaya Aura® Communication Manager 8.1 (Communication Manager) is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. Avaya Aura® Session Manager 8.1 (Session Manager) is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise.

Avaya Aura® Experience Portal (Experience Portal) provides a single platform for automated voice and multimedia, self-service, and Interactive Voice Response (IVR) applications. In the sample configuration described in these Application Notes, a basic Experience Portal test application was used to exercise various SIP call flow scenarios.

The Avaya Session Border Controller for Enterprise 8.0.1 (Avaya SBCE) is the point of connection between Session Manager and the AT&T IP Flexible Reach - Enhanced Features (IPFR-EF) service and is used to not only secure the SIP trunk, but also to adjust the SIP signaling and media for interoperability.

The AT&T Flexible Reach service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for their voice communication needs. The AT&T IP Flexible Reach-Enhanced Features service is a SIP based service which includes additional network-based features which are not part of IP Flexible Reach service. The AT&T IP Flexible Reach - Enhanced Features service utilizes AT&T's AVPN[1] or MIS/PNT[2] transport services.

Note – The AT&T IP Flexible Reach - Enhanced Features service will be referred to as IPFR-EF in the remainder of this document.

[1] AVPN supports compressed RTP (cRTP).
[2] MIS/PNT does not support cRTP.

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 IPFR-EF and the Customer Premises Equipment (CPE) containing Communication Manager, Session Manager, Experience Portal and Avaya SBCE (see **Section 3.2** for call flow examples).

The test environment consisted of:

- A simulated enterprise with Communication Manager, Session Manager, System Manager (for Session Manager provisioning) Experience Portal, Avaya SBCE, Avaya phones, and fax machines (Ventafax application). Avaya Aura® Messaging (Messaging) is used to provide voicemail capabilities for the CPE.
- IPFR-EF service test lab circuit, connected to the simulated enterprise via AVPN transport.

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

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

For the testing associated with this Application Note, the interface between Avaya systems and the AT&T IP Flexible Reach 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 AT&T IPFR-EF network. Calls were made between the PSTN, via the AT&T IPFR-EF network, and the CPE.

The following SIP trunking VoIP features were tested with the IPFR-EF service:

- Inbound and outbound voice calls between telephones controlled by the CPE and the PSTN using G.729A and G.711MU codecs. Phone types included SIP, H.323, digital and analog telephones at the enterprise.
- DTMF using RFC 2833
    - Outbound call to PSTN application requiring post-answer DTMF (e.g., an IVR or voice mail system).
    - Inbound call from PSTN to Avaya CPE application requiring post-answer DTMF (e.g., Aura® Messaging, Experience Portal, Communication Manager vector digit collection steps).
- Requests for privacy for Communication Manager outbound calls to the PSTN, as well as privacy requests for inbound calls from the PSTN to Communication Manager users.
- SIP OPTIONS messages used to monitor the health of the SIP trunks between the CPE and AT&T.
- Additional PSTN numbering plans (e.g., operator assist, toll-free and International).
- Telephony features such as hold, transfer, and conference.
- SIP Diversion Header for call redirection.
    - Call Forwarding
    - EC500
- Long duration calls.
- Inbound/Outbound fax calls using T.38.
- Failover test between primary and secondary AT&T Border Elements.
- AT&T IPFR-EF service features such as:
    - Simultaneous Ring
    - Sequential Ring
    - Call Forward – Always
    - Call Forward – Busy
    - Call Forward – Ring No Answer
    - “Blind” and “Attended” transfers utilizing Refer messaging.
- Inbound caller interaction with Experience Portal applications, including prompting, caller DTMF input, wait treatment (e.g., announcements and/or music on hold) and Automatic Speech Recognition.
- Experience Portal use of SIP REFER to redirect inbound calls, via the Avaya SBCE, to the appropriate Communication Manager agent extension.
- Call and two-way talk path establishment between callers and Communication Manager agents following redirection from Experience Portal.
- Inbound calls to a self-service Experience Portal application which forwards the call to another PSTN number over the IPFR-EF service, using SIP INVITE and SIP REFER methods.

An Avaya Remote Worker endpoint (Avaya Equinox SIP softphone) was used in the reference configuration. The Remote Worker endpoint resides on the public side of the Avaya SBCE (via a TLS connection), and registers/communicates with Avaya Session Manager via Avaya SBCE as though it was an endpoint residing in the private CPE space. The configuration of the Remote Worker environment is beyond the scope of this document.

Note – Documents used to provision the test environment are listed in **Section 12**. In the following sections, references to these documents are indicated by the notation **[x]**, where *x* is the document reference number.

2.2. Test Results

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

1) **IPFR-EF Simultaneous Ring and Sequential Ring - Loss of calling display information on Communication Manager stations**. If the Communication Manager station associated with these IPFR-EF "secondary" number for Simultaneous Ring and Sequential Ring scenarios answers the call, the phone may not display all the calling information. By default, Communication Manager expects a display update from the network in the P-Asserted-Identity (PAI) header. However, the subsequent network signaling does not contain a PAI header, and the From header must be used instead. The recommended workaround is described in **Section 5.8.1**, where Communication Manager will retrieve the display information using the *From* header.

2) **T.38/G.729 fax is limited to 9600bps when using the G4xx Media Gateways**. A G430 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.

3) **Avaya SBCE inserts a=ptime:20 in the SIP SDP toward Communication Manager**. AT&T includes a=maxptime:30 in the SIP SDP to recommend a ptime value of 30ms, but does not specify a ptime value in the SDP. If no media packetization attribute (ptime) is included in the SIP Session Description Protocol (SDP), Avaya SBCE inserts "a=ptime:20", specifying 20 milliseconds. Although Communication Manager can be configured to send ptime with a value of 30ms (See **Section 5.7.2**), it will send a ptime value of 20ms when it receives "a=ptime:20" from the Avaya SBCE. This causes the media packetization to be set to 20ms. No issues were found during testing due to this behavior.

4) **Avaya SBCE does not change the Diversion header from sips to sip**. When TLS/SRTP is used within the enterprise, the SIP headers include the SIPS URI scheme for Secure SIP. The Avaya SBCE converts these header schemes from SIPS to SIP when it sends the SIP message toward AT&T. However, for call forward and EC500 calls, the Avaya SBCE was not changing the Diversion header scheme as expected. This caused these call types that require a Diversion header to fail, since Secure SIP is not supported on the SIP trunk to AT&T. This anomaly is currently under investigation by the Avaya SBCE development team. A workaround is to include an Avaya SBCE Signaling Manipulation (SigMa) script on the AT&T SIP Server profile on the Avaya SBCE, to convert "sips" to "sip" in the Diversion header. See **Section 8.8**.

5) **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 SigMa script (**Section 8.8)** is used to remove these headers.

6) **SIP OPTIONS** – The AT&T IPFR-EF service is configured to send SIP OPTIONS messages with a Max-Forwards header value of "0". This is by design from AT&T and Avaya SBCE correctly responded with "483 Too Many Hops". AT&T considers this response acceptable to keep the trunk in service. However, an incident is logged on the Avaya SBCE for each OPTIONS message received with Max-Forwards=0. To prevent the incident log from being filled with these route failure messages, an optional SigMa script (**Section 8.8)** can be added to the Avaya SBCE to change the Max-Forwards value to an acceptable value to reach Communication Manager.

7) **Experience Portal consultative call transfer**. The Experience Portal test application used for compliance testing performs consultative call transfer of inbound calls that are transferred back to AT&T using SIP INVITE, with the original calling party number in the From and P-Asserted Identity headers, and it does not contain a Diversion header. In this scenario, since none of the headers in the outbound INVITE contains a number recognizable by the AT&T network, consultative call transfers out the AT&T IPFR-EF service fail. As a workaround, a SigMa script file (**Section 8.8)** was created to modify the P-Asserted-Identity header on outbound INVITEs from Experience Portal to the PSTN, with the DID number assigned to Experience Portal, known to AT&T.
In addition, Experience Portal blind transfers out to AT&T using SIP REFER were tested successfully. Also, consultative and blind transfers from Experience Portal to Communication Manager were successful as well.

8) **Some Avaya SIP endpoints use different RFC2833 DTMF Payload Type than defined in CM trunk provisioning**. Although Communication Manager can specify the default RFC2833 payload type to be used on the SIP trunk to AT&T (payload type 100 in the compliance test), outbound calls originating from some Avaya SIP Deskphones and Avaya Equinox for Windows softphones used payload type 101 in requests on the SIP trunk to AT&T. The payload type on these SIP endpoints is ultimately controlled by the 46xxsettings file that is loaded on the phones. Calls originating from these endpoints had the payload type dynamically negotiated with AT&T to type 101 during the call setup. No issues were found during testing as a result of this behavior.

9) **Emergency 911/E911 Services Limitations and Restrictions**. Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) documented in these Application Notes will properly operate with AT&T IPFR-EF to complete 911/E911 calls. 911/E991 was not tested during the compliance test; therefore, it is the customer's responsibility to ensure proper operation with the equipment/software vendor. While AT&T IPFR-EF services support 911/E911 calling capabilities under certain Calling Plans, there are circumstances when the 911/E911 service

may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.

2.3. Support

For more information on the AT&T IP Flexible Reach service visit: https://www.business.att.com/products/sip-trunking.html. AT&T customers may obtain support for the AT&T IP Flexible Reach service by calling (877) 288-8362.

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.

2.4. SIP Message – Packet Optimization

To support advanced SIP telephony features in the Avaya Aura® enterprise environment, certain proprietary headers may be included in the SIP message sent toward the AT&T IPFR-EF service. These extra headers can cause the SIP message to become larger than the specified Maximum Transmission Unit (MTU) in some network equipment and create fragmented UDP packets. These fragmented packets may not be re-assembled properly on the far-end when packets arrive out of order. To prevent fragmented packets, any unnecessary or proprietary headers should be removed from the SIP message before being sent to AT&T. Session Manager can remove these headers by specifying the "*eRHdrs*" parameter within the "AttAdapter" adaptation. See **Section 6.4.2**.

In the sample configuration, the following headers were removed:

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

To help reduce the packet size further, the Avaya SBCE can remove the Avaya "*gsid*" and "*epv*" parameters that may be included within the Contact header of outbound messages, by applying a Sigma script to the AT&T SIP server profile. See **Sections 8.8** and **8.9.2**.

3. Reference Configuration

Figure 1 illustrates the sample configuration used for the compliance testing, with the AT&T IPFR-EF service test lab circuit, connected to the simulated enterprise site via AVPN transport.

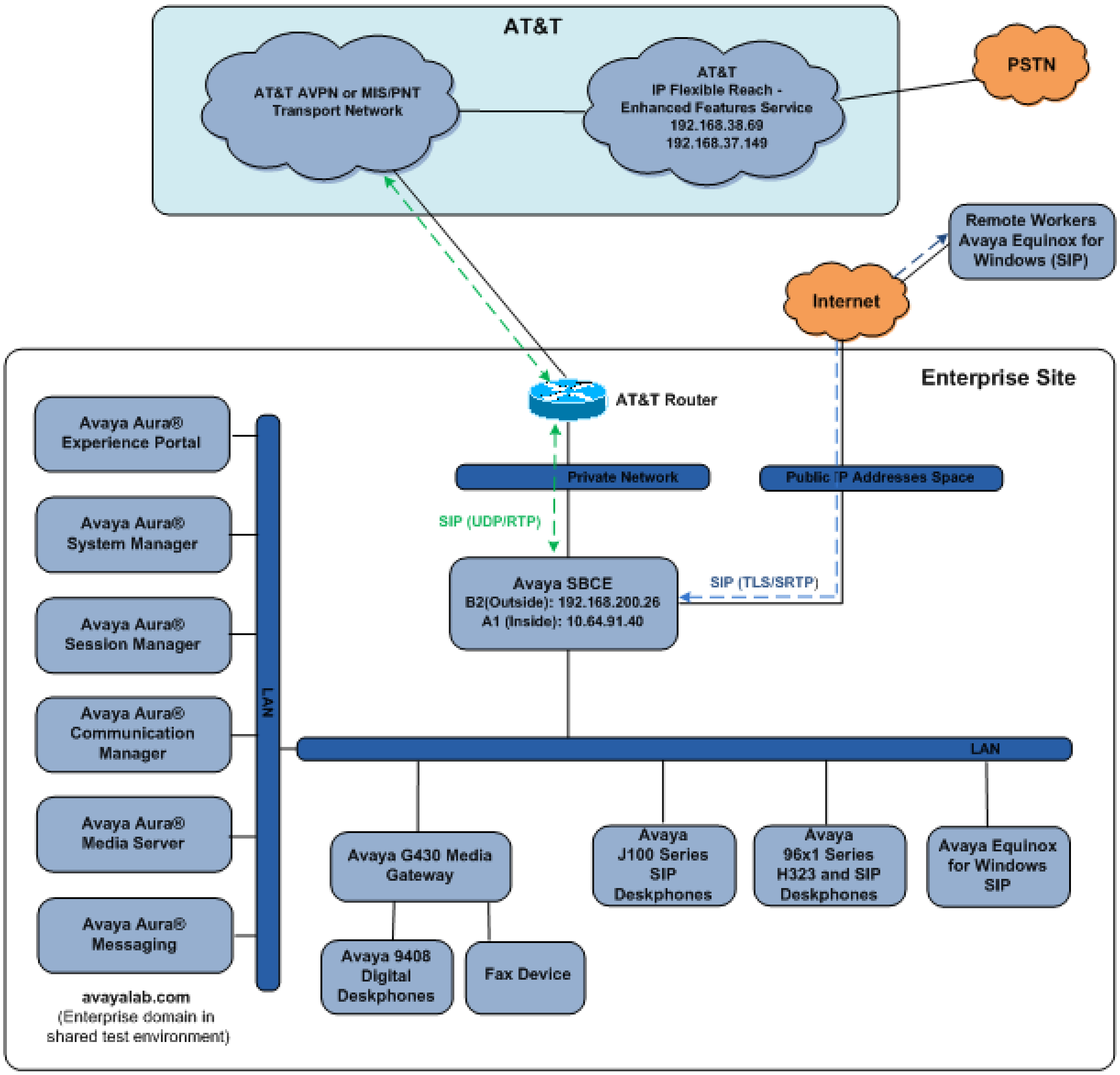


Figure 1: Reference configuration

The following components were used in the reference configuration:

- Avaya Session Border Controllers for Enterprise
- Avaya Aura® Session Manager
- Avaya Aura® System Manager
- Avaya Aura® Communication Manager
- Avaya G430 Media Gateway
- Avaya Media Server
- Avaya Aura® Messaging
- Avaya Aura® Experience Portal
- Avaya 96X1 Series IP Deskphones using the SIP and H.323 software bundle
- J100 Series IP Deskphones using the SIP software bundle
- Avaya Equinox™ for Windows
- Avaya Digital Phones
- Ventafax fax software

Avaya Aura® System Manager provides a common administration interface for centralized management of Session Manager and Communication Manager. Avaya Aura® Messaging was used in the reference configuration to provide voice mailbox capabilities. This solution is extensible to other messaging platforms. The provisioning of Avaya Aura® Messaging is beyond the scope of this document.

Note that while an Avaya G430 Media Gateway and an Avaya Media Server are used in the reference configuration, this solution is extensible to other Avaya Media Gateways.

The Avaya SBCE provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the IPFR-EF service and the enterprise internal network.

The IPFR-EF service Border Element (BE) uses SIP over UDP to communicate with enterprise edge SIP devices, (e.g., the Avaya SBCE in this sample configuration). Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements. In the reference configuration, Session Manager uses SIP over TLS to communicate with the Avaya SBCE, Experience Portal, Messaging and Communication Manager.

Testing was performed using an IPFR-EF service test lab circuit.

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 configurations.

Note – The IPFR-EF service Border Element IP address and DID/DNIS digits are shown in this document as examples. AT&T Customer Care will provide the actual IP addresses and DID/DNIS digits as part of the IPFR-EF provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura® Session Manager	
IP Address	10.64.91.81
Avaya Aura® Communication Manager	
IP Address	10.64.91.75
Avaya Aura® System Manager	
IP Address	10.64.90.82
Avaya Aura® Messaging	
IP Address	10.64.91.84
Avaya Aura® Experience Portal	
IP Address	10.64.91.90
Avaya Session Border Controller for Enterprise (SBCE)	
IP Address of Inside (Private) Interface	10.64.91.40
IP Address of Outside (Public) Interface	192.168.200.26 (see note below)
AT&T Border Element	
IP Addresses	192.168.38.69 192.168.37.149

Table 1: Network Values Used in these Application Notes

Note – The Avaya SBCE Outside interface communicates with AT&T Border Elements (BEs) located in the AT&T IP Flexible Reach network. For security reasons, the actual IP addresses of the Avaya SBCE and AT&T BE are not included in this document. However, as placeholders in the following configuration sections, the IP addresses of **192.168.200.26** (Avaya SBCE public interface), **192.168.38.69** and **192.168.37.149** (AT&T BE IP addresses) are specified.

3.2. Call Flows

To understand how IPFR-EF service calls are handled by the Avaya CPE environment, several basic call flows are described in this section. However, for brevity, not all possible call flows are described.

3.2.1. Communication Manager Basic Flows

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

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

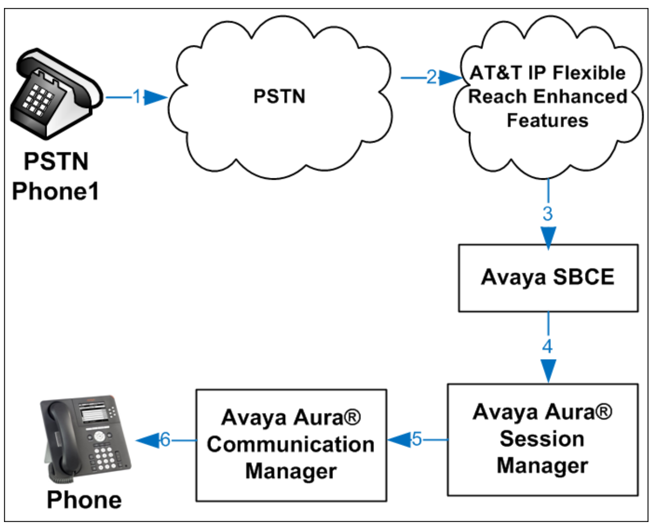


Figure 2: Inbound IPFR-EF Call

The second call scenario illustrated is an outbound call initiated on Communication Manager, routed to Session Manager, and is subsequently sent to the Avaya SBCE for delivery to the IPFR-EF service.

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

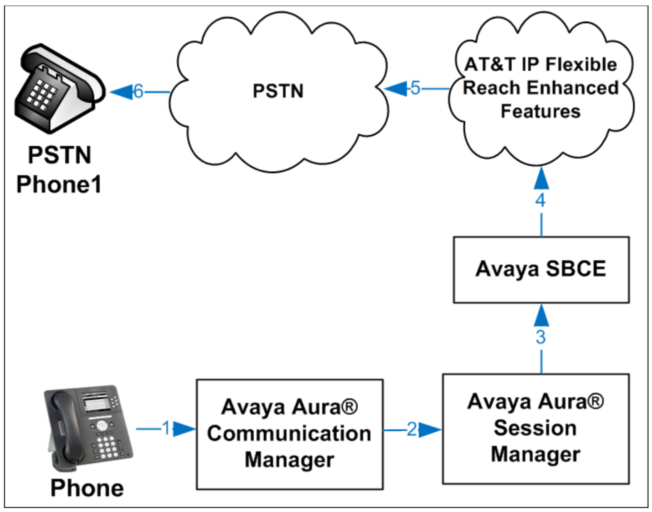


Figure 3: Outbound IPFR-EF Call

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

Note – In cases where calls are forwarded to an alternate destination such as an 8xx numbers, the IPFR-EF service requires the use of SIP Diversion Header for the redirected call to complete (see **Section 5.8.1**).

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

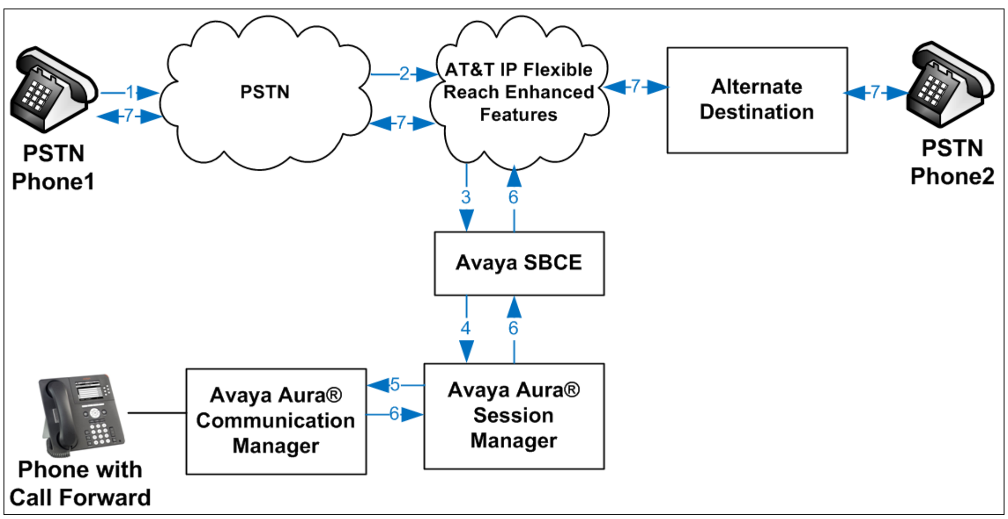


Figure 4: Station Re-directed (e.g., Call Forward) IPFR-EF Call

3.2.2. Network Based Blind Transfer Call Flow (Communication Manager Vector)

This section describes the call flow for IPFR-EF using SIP REFER to perform Network Based Blind Transfer. The REFER is generated by an inbound call to a Communication Manager Vector. The call scenario illustrated in **Figure 5** below is an inbound IPFR-EF call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector. The vector answers the call and, using REFER (without the Replaces parameter) redirects the call back to the IPFR-EF service for routing to an alternate destination.

1. A PSTN phone originates a call to an IPFR-EF number.
2. The PSTN routes the call to the IPFR-EF network.
3. IPFR-EF routes the call to the Avaya SBCE.
4. The Avaya SBCE performs SIP Network Address Translation (NAT) and any necessary SIP header modifications and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to Communication Manager.
6. Communication Manager routes the call to a VDN/Vector, which answers the call and plays an announcement, and attempts to redirect the call using a SIP REFER message. The REFER message specifies the alternate destination in its Refer-To header, and is routed back through Session Manager on to the Avaya SBCE. The Avaya SBCE sends the REFER to the IPFR-EF service.
7. IPFR-EF places a call to the alternate destination specified in the REFER, and upon answer, connects the calling party to the alternate party.
8. IPFR-EF clears the call on the redirecting/referring party (Communication Manager).

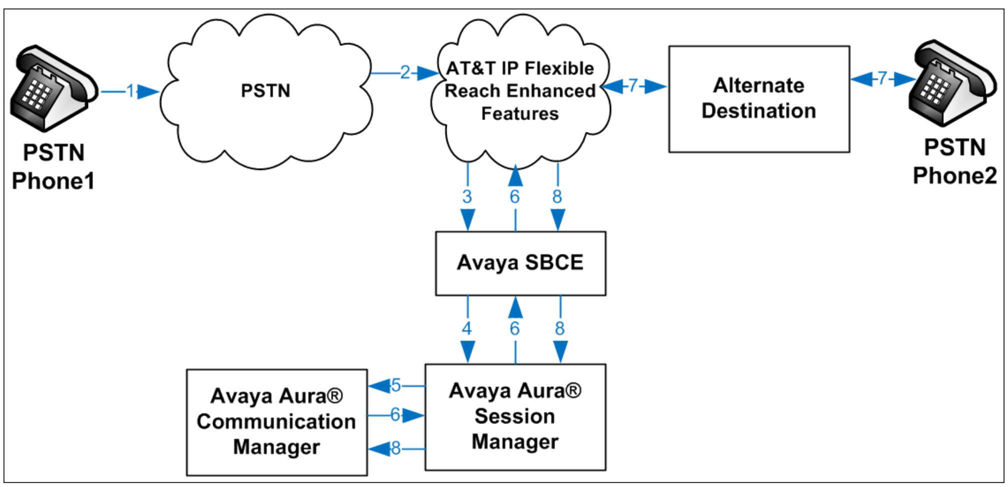


Figure 5: Network Based Blind Transfer Using REFER (Communication Manager Vector)

3.2.3. Network Based Attended/Unattended Transfer Call Flow initiated by Communication Manager Station

This section describes the call flow for IPFR-EF using SIP REFER to perform an Attended or Unattended Transfer. The call scenario illustrated in **Figure 6** below is an inbound IPFR-EF call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a station. The station answers the call and transfers it back out to a second PSTN destination. Communication Manager initiates a new call back out to Session Manager, the Avaya SBCE, and to the IPFR-EF service network. Communication Manager completes the transfer, using REFER (with the Replaces parameter), to the IPFR-EF service to connect the two active calls together.

1. A PSTN phone originates a call to an IPFR-EF number.
2. The PSTN routes the call to the IPFR-EF network.
3. IPFR-EF routes the call to the Avaya SBCE.
4. The Avaya SBCE performs SIP Network Address Translation (NAT) and any necessary SIP header modifications and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to Communication Manager. Communication Manager routes the call to a station.
6. The station answers the call and then transfers it to a new PSTN destination. Communication Manager initiates a new call back out to Session Manager, the Avaya SBCE, and to the IPFR-EF service network. Communication Manager redirects the call using a SIP REFER message when the transfer is completed by the station. The REFER message specifies the active call to replace and is routed back through Session Manager on to the Avaya SBCE. The Avaya SBCE sends the REFER to the IPFR-EF service.
7. IPFR-EF replaces the call with the alternate destination specified in the REFER and connects the calling party to the alternate party.
8. IPFR-EF clears the existing calls to Communication Manager.

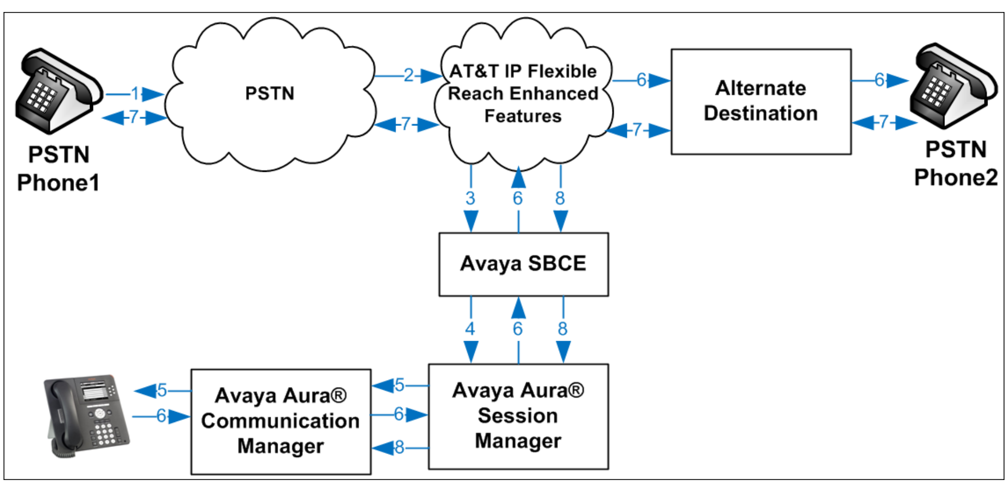


Figure 6: Attended/Unattended Transfer Using REFER (Communication Manager Station)

3.2.4. Experience Portal Call Flows

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

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

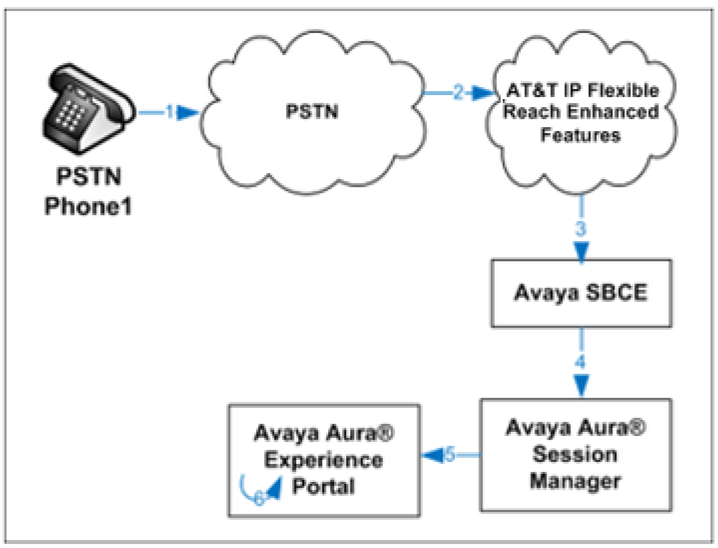


Figure 7: Inbound Call Handling Entirely by Avaya Aura® Experience Portal

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

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

Note: See **Appendix A**, **Section 13** for configuration information on the Avaya SBCE Refer Handling option for Experience Portal

4. Communication Manager routes the call to the agent.
5. When the agent answers, the Avaya SBCE takes the call off hold and the caller is connected to the agent.

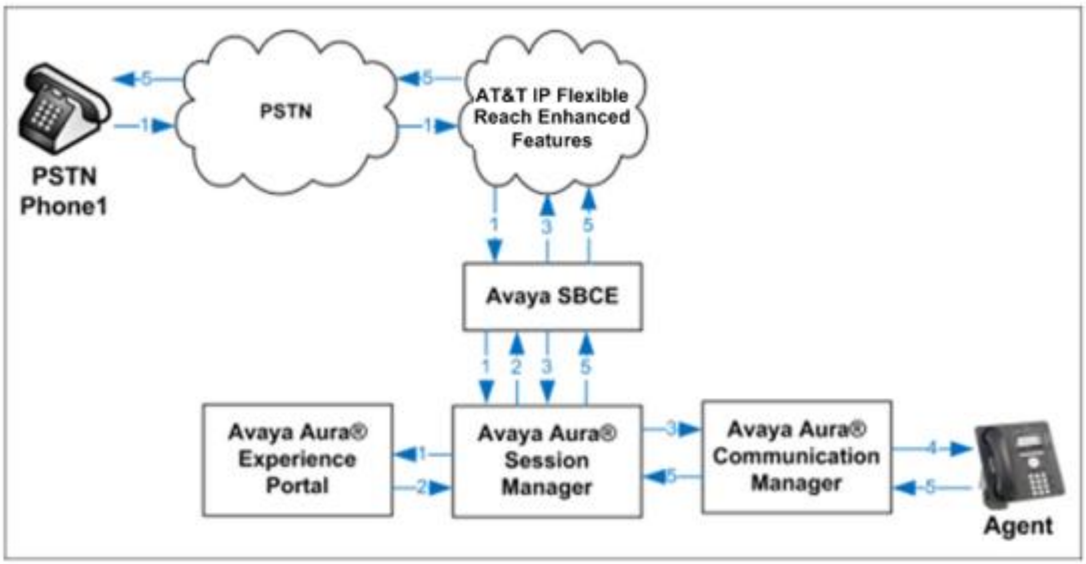


Figure 8: Avaya Aura® Experience Portal Transfers Call to Avaya Aura® Communication Manager

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

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

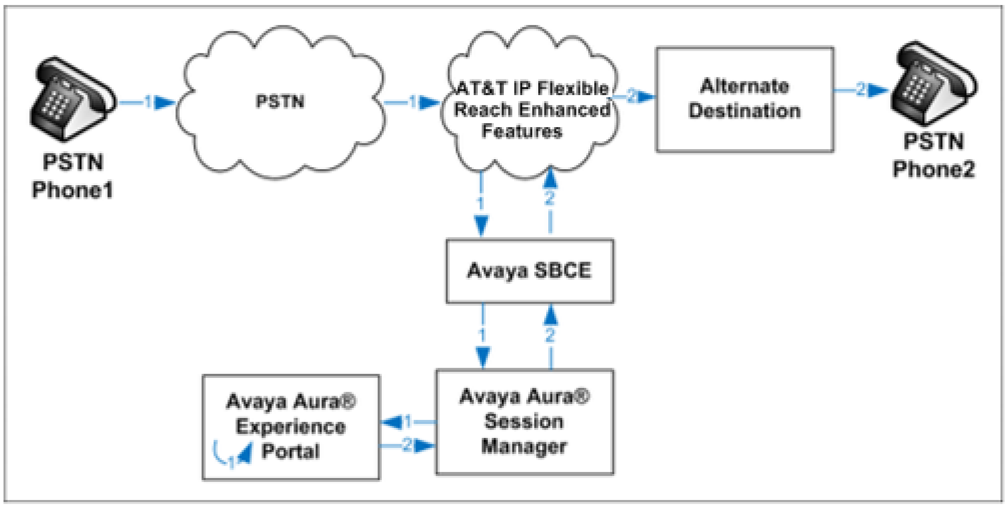


Figure 9: Inbound Call transferred by Experience Portal to another PSTN number

The Experience Portal test application used for compliance testing can perform blind or consultative call transfers of inbound calls that are to be transferred back to the PSTN. This is done in similar fashion to the network based transfer call flows shown previously for Communication Manager (**Section 3.2.2** and **Section 3.2.3**).

On a blind transfer, Experience Portal sends a REFER without the Replaces parameter back to IPFR-EF. On consultative transfers, Experience Portal places a new outbound call, sending a SIP INVITE out to the IPFR-EF network. Once the new call is answered, Experience Portal sends a REFER with Replaces parameter to the network. In both cases, IPFR-EF replaces the call with the alternate destination specified in the REFER and connects the calling party to the alternate party directly, clearing the existing calls to Experience Portal.

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® Communication Manager	8.1.0.2 (Service Pack 2)
Avaya Aura® Session Manager	8.1.0.0.810007
Avaya Aura® System Manager	8.1.0.0.079880
Avaya Aura® Experience Portal	7.2.3.0.0441
Avaya Session Border Controller for Enterprise	8.0.1.0-10-17555
Avaya Aura® Messaging	7.1 SP 1
Avaya Aura® Media Server	8.0.1.121
Avaya G430 Media Gateway	41.10.0
Avaya 96x1 Series IP Deskphone (H.323)	6.8202
Avaya 96x1 Series IP Deskphone (SIP)	7.1.6.1.3
Avaya J129 IP Deskphone (SIP)	4.0.2.1.3
Avaya 9408 Digital Deskphone	20.06
Avaya Equinox for Windows	3.6.4.31.2
Fax device	Ventafax 7.10

Table 2: Equipment and Software Versions

5. Configure 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 Error! Reference source not found. **- [9]**Error! Reference source not found. in the References section for additional information.

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.

5.1. Verify Licensed Features

Note – This section describes steps to verify Communication Manager feature settings that are required for the reference configuration described in these Application Notes. Depending on access privileges and licensing, some or all of the following settings might only be viewed, and not modified.

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:  1000      2
         Maximum Administered Remote Office Trunks:  4000      0
Max Concurrently Registered Remote Office Stations:  1000      0
          Maximum Concurrently Registered IP eCons:   68       0
     Max Concur Reg Unauthenticated H.323 Stations:   100      0
                    Maximum Video Capable Stations:   2400     0
               Maximum Video Capable IP Softphones:   1000     6
                   Maximum Administered SIP Trunks:  4000     75
  Max Administered Ad-hoc Video Conferencing Ports:  4000      0
   Max Number of DS1 Boards with Echo Cancellation:     80     0
```

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

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

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

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

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

            IP Attendant Consoles? y
```

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

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

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

5.2. System-Parameters Features

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

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

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


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

5.3. Dial Plan

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

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

- 5-digit extensions with a **Call Type** of **ext** beginning with:
  - The digits **1**, **5**, **7** and **8** for Communication Manager extensions.
- 3-digit dial access code (indicated with a **Call Type** of **dac**), e.g., access code ***xx** for SIP Trunk Access Codes (TAC). See the trunk forms in **Section 5.8**.

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

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

5.4. Node Names

Node names define IP addresses to various Avaya components in the enterprise. In the reference configuration a Processor Ethernet (procr) based Communication Manager platform is used. Note that the Communication Manager procr name and IP address are entered during installation. The procr IP address was used to define the Communication Manager SIP Entities in **Section 6.5**.

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

- Session Manager SIP signaling interface (e.g., **SM** and **10.64.91.81**).
- Media Server (e.g., **AMS** and **10.64.91.86**). 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                 10.64.91.86
SM                  10.64.91.81
default             0.0.0.0
procr               10.64.91.75
procr6              ::
```

5.5. Processor Ethernet Configuration

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

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

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

                  Type: PROCR
                                                      Target socket load: 4800

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

                                IPV4 PARAMETERS
             Node Name: procr                    IP Address: 10.64.91.75
           Subnet Mask: /24
```

5.6. IP Network Regions

Network regions provide a means to logically group resources such as codecs, UDP port ranges, and inter-region communication. In the shared Communication Manager configuration used for the testing, the Avaya G430 Media Gateway and Avaya Media Server are in region 1. To provide testing flexibility, network region 4 was associated to components used specifically for the AT&T SIP trunk access.

5.6.1. IP Network Region 1 – Local CPE Region

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

- Enter a descriptive name (e.g., **Enterprise**).
- Enter the enterprise domain (e.g., **avayalab.com**) in the **Authoritative Domain** field (see **Section 6.2**).
- Enter **1** for the **Codec Set** parameter.
- **Intra-region IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible within the same region.
- **Inter-region IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible between regions.
- **UDP Port Min**: – Set to **16384** (**AT&T requirement**).
- **UDP Port Max**: – Set to **32767** (**AT&T requirement**).

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

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avayalab.com
    Name: Enterprise            Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                      IP Audio Hairpinning? n
   UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

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

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

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

 RTCP Reporting to Monitor Server Enabled? y

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

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

- Verify that next to region **1** in the **dst rgn** column, the codec set is **1**.
- Next to region **4** in the **dst rgn** column, enter **4** for the codec set (this means region 1 is permitted to talk to region 4 and it will use codec set 4 to do so). The **direct WAN** and **Units** columns will self-populate with **y** and **No Limit** respectively.
- Let all other values default for this form.

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

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    1                                                               all
 2    2     y    NoLimit                                             n       t
 3    1     y    NoLimit                                             n       t
 4    4     y    NoLimit                                             n       t
```

5.6.2. IP Network Region 4 – AT&T Trunk Region

Repeat the steps in **Section 5.6.1** with the following changes:

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

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

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

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

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

 Source Region: 4     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    4     y    NoLimit                                             n       t
 2    4     y    NoLimit                                             n       t
 3    3     y    NoLimit                                             n       t
 4    4                                                               all
```

5.7. IP Codec Sets

Use the **change ip-codec-set** command to define a list of codecs to use for calls within the enterprise, and for calls between the enterprise and the service provider.

5.7.1. 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** and **G.729A** are included in the codec list. Note that the packet interval size will default to 20ms. Set the **Media Encryption** based on customer requirements. In the reference configuration, **1-srtp-aescm128-hmac80** was the preferred crypto suite, with **none** set as the second option.

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

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

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

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

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

                          IP MEDIA PARAMETERS

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

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

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

5.7.2. Codecs for IP Network Region 4 (calls to/from AT&T)

This IP codec set will be used for IPFR-EF calls. Repeat the steps in **Section 5.7.1** with the following changes:

- Provision the codecs in the order shown below.
- Set **Frames Per Pkt** to **3**. This will auto-populate **30** for the **Packet Size (ms)** field, and specify a PTIME value of 30 in the SDP (recommended by AT&T). See **Section 2.2** for limitations.

```
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.711MU           n           3        30

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

change ip-codec-set 4                                           Page   2 of   2
                          IP CODEC SET
                              Allow Direct-IP Multimedia? n
                                                                      Packet
                          Mode                   Redundancy           Size(ms)
    FAX                   t.38-standard           0           ECM: y
    Modem                 off                     0
    TDD/TTY               US                      3
    H.323 Clear-channel   n                       0
    SIP 64K Data          n                       0                   20
```

5.8. SIP Trunks

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

- Inbound/outbound AT&T access – SIP Trunk 5. This trunk will use TLS port 5065..
- Internal CPE access (e.g., Avaya SIP telephones, Messaging, etc.) – SIP Trunk 3. This trunk will use TLS port 5061.

Note that different ports are assigned to each trunk. This is necessary so Session Manager can distinguish the traffic on the service provider trunk, from the traffic on the trunk used for other enterprise SIP traffic.

Note – Although TLS is used as the transport protocols between the Avaya CPE components, UDP was used between the Avaya SBCE and the IPFR-EF service. See the note in **Section 6.5** regarding the use of TLS transport protocols in the CPE.

5.8.1. SIP Trunk for Inbound/Outbound AT&T calls

This section describes the steps for administering the SIP trunk to Session Manager used for IPFR-EF calls. Trunk Group 5 is defined. This trunk corresponds to the **CM-TG5** SIP Entity defined in **Section 6.5.2**.

5.8.1.1 Signaling Group 5

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 system will auto detect and set the **Peer Server** to **SM**.
- **Near-end Node Name** – Set to the node name of the **procr** noted in **Section 5.4**.
- **Far-end Node Name** – Set to the node name of Session Manager as administered in **Section 5.4** (e.g., **SM**).
- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5065**.
- **Far-end Network Region** – Set the IP network region to **4**, as set in **Section 5.6.2**.
- **Far-end Domain** – Enter **avayalab.com**. This is the domain provisioned for Session Manager in **Section 6.2**.
- **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.
- **Initial IP-IP Direct Media** is set to **n**.

- **H.323 Station Outgoing Direct Media** is set to **n**.
- Use the default parameters on **page 2** of the form (not shown).

```
add signaling-group 5                                          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                       Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: SM
 Near-end Listen Port: 5065                 Far-end Listen Port: 5065
                                         Far-end Network Region: 4

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

5.8.1.2 Trunk Group 5

Step 1 - Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g., **5**). 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 IPFR**).
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., ***05**).
- **Direction** – Set to **two-way**.
- **Service Type** – Set to **public-ntwrk**.
- **Signaling Group** – Set to the signaling group administered in **Section 5.8.1.1** (e.g., **5**).
- **Number of Members** – Enter the maximum number of simultaneous calls desired on this trunk group (based on licensing) (e.g., **10**).

```
add trunk-group 5                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 5                    Group Type: sip          CDR Reports: y
  Group Name: ATT IPFR                    COR: 5        TN: 1        TAC: *05
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 5
                                                     Number of Members: 10
```

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

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

```
add trunk-group 5                                               Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

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

 Disconnect Supervision - In? y  Out? y

             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
Caller ID for Service Link Call to H.323 1xC: station-extension
```

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

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

```
add trunk-group 5                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y
```

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

- Verify **Network Call Redirection** is set to **y**.
- Set **Send Diversion Header** to **y**. This is required for Communication Manager station Call Forward scenarios to IPFR-EF service.
- Set **Telephone Event Payload Type** to the RTP payload type recommended by the IPFR-EF service (e.g., **100**).
- Set **Identity for Calling Party Display** to **From**. Note that the display issue described in **Section 2.2**, **Item 1** may be resolved by setting the **Identity for Calling Party Display** parameter to **From**.

Note – The IPFR-EF service does not support History Info header. As shown below, by default this header is supported by Communication Manager. In the reference configuration, the History Info header is automatically removed from SIP signaling by Session Manager, as part of the AttAdapter (see **Section 6.4.2**). Alternatively, History Info may be disabled here.

```
add trunk-group 5                                                Page   4 of  21
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? y
                                   Support Request History? y
                              Telephone Event Payload Type: 100
                                       Shuffling with SDP? n

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

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

5.8.2. Local SIP Trunk (Avaya SIP Telephone, Messaging Access, etc.)

Trunk Group 3 corresponds to the **CM-TG3** SIP Entity defined in **Section 6.5.3**.

5.8.2.1 Signaling Group 3

Repeat the steps in **Section 5.8.1.1** with the following changes:

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

Step 2 - Set the following parameters on page 1:

- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5061**
- **Far-end Network Region** – Set to the IP network region **1**, as defined in **Section 5.6.1**.

5.8.2.2 Trunk Group 3

Repeat the steps in **Section 5.8.1.2** with the following changes:

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

- **Group Name** – Enter a descriptive name (e.g., **SM Enterprise**).
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., ***03**).
- **Service Type** – Set to **tie**.
- **Signaling Group** – Set to the number of the signaling group administered in **Section 5.8.2.1** (e.g., **3**).

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

- Same as **Section 5.8.1.2**.

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

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

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

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

Use default values for all other settings.

5.9. Public Numbering

In the reference configuration, the public-unknown-numbering form, (used in conjunction with the **Numbering Format: public** setting in **Section 5.8.1.2**), is used to convert Communication Manager local extensions to IPFR-EF DNIS numbers, for inclusion in any SIP headers directed to the IPFR-EF service via the public trunk.

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

Step 2 - Add each Communication Manager station extension and their corresponding IPFR-EF DNIS numbers (for the public trunk to AT&T). Communication Manager will insert these AT&T DNIS numbers in E.164 format into the From, Contact, and PAI headers as appropriate. In the reference configuration, a range of extensions were added as follows:

- **Ext Len** – Enter the total number of digits in the local extension range (e.g., **5**).
- **Ext Code –** Enter the first two digits for Communication Manager extensions (e.g., **54** for extension range 54xxx, and **59** for extension range 59xxx).
- **Trk Grp(s)** – Enter the number of the Public trunk group (e.g., **5**).
- **Private Prefix** – Enter the corresponding IPFR-EF DNIS number prefix (e.g., **146955** and **130355**).
- **Total Len** – Enter the total number of digits after the digit conversion (e.g., **11**).

```
change public-unknown-numbering 5 ext-digits 5                  Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                               Total
Ext Ext               Trk       CPN             CPN
Len Code              Grp(s)    Prefix          Len
                                                   Total Administered: 46
 5  14                5         17325552754     11    Maximum Entries: 240
 5  50                4         173255          11
 5  54                5         146955          11 Note: If an entry applies to
 5  59                5         130355          11 a SIP connection to Avaya
 5  10001             2         18665553221     11 Aura(R) Session Manager,
                                                   the resulting number must
                                                   be a complete E.164 number.

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

5.10. Private Numbering

In the reference configuration, the private-numbering form, (used in conjunction with the **Numbering Format: private** setting in **Section 5.8.2.2**), is used to send Communication Manager local extension numbers to Session Manager, for inclusion in any SIP headers directed to SIP endpoints and Messaging.

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

- **Ext Len** – Enter the total number of digits in the local extension range (e.g., **5**).
- **Ext Code –** Enter the Communication Manager extension patterns defined in the Dial Plan in **Section 5.3** (e.g., **54** and **59**).
- **Trk Grp(s)** – Enter the number of the Local trunk group (e.g., **3**).
- **Total Len** - Enter the total number of digits after the digit conversion (e.g., **5**).

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

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 5  10             3                           5     Total Administered: 6
 5  11             3                           5        Maximum Entries: 540
 5  12             3                           5
 5  54             3                           5
 5  59             3                           5
```

5.11. Route Patterns

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

5.11.1. Route Pattern for National Calls to AT&T

This form defines the public SIP trunk, based on the route-pattern selected by the ARS table in **Section 5.12**. The routing defined in this section is simply an example and not intended to be prescriptive. Other routing policies may be appropriate for different customer networks. In the reference configuration, route pattern 1 is used for national calls, route pattern 2 is used for international calls, and route pattern 4 is used for service calls and IPFR-EF Call Forward feature access codes.

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

- In the **Grp No** column, enter **5** for public trunk 5, and the **FRL** column enter **0** (zero).
- In the **Pfx Mrk** column, enter **1** to ensure 1 + 10 digits are sent to the service provider for FNPA calls.
- In the **Inserted Digits** column, enter **p** to have Communication Manager insert a plus sign (+) in front of the number dialed to convert it to an E.164 formatted number.

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

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 5    0       1                p                                    n   user
 2:                                                                    n   user
 3:                                                                    n   user

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

5.11.2. Route Pattern for International Calls to AT&T

Repeat the steps in **Section 5.11.1** to add a route pattern for international calls with the following changes:

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

- In the **Grp No** column, enter **5** for public trunk 5, and the **FRL** column enter **0** (zero).
- In the **Pfx Mrk** column, leave blank (default).
- In the **No. Del Digits** column, enter **3** to have Communication Manager remove the international 011 prefix from the number.
- In the **Inserted Digits** column, enter **p** to have Communication Manager insert a plus sign (+) in front of the number dialed to convert it to an E.164 formatted number.

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

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 5    0                    3   p                                    n   user
 2:                                                                    n   user
 3:                                                                    n   user

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

5.11.3. Route Pattern for Service Calls to AT&T

Repeat the steps in **Section 5.11.1** to add a route pattern for x11 and IPFR-EF Call Forward feature access codes calls that do not require a leading plus sign, with the following changes:

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

- In the **Grp No** column, enter **5** for public trunk 5, and the **FRL** column enter **0** (zero).
- In the **Pfx Mrk** column, leave blank (default).
- In the **Inserted Digits** column, leave blank (default).

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

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 5    0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user

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

5.11.4. Route Pattern for Calls within the CPE

This form defines the Route pattern for the local SIP trunk, based on the route-pattern selected by the AAR table in **Section 5.13** (e.g., calls to Avaya SIP telephone extensions or Messaging).

Step 1 - Repeat the steps in **Section 5.11.1** with the following changes:

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

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

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

5.12. Automatic Route Selection (ARS) Dialing

The ARS table is selected based on the caller dialing the ARS access code (e.g., **9**) as defined in **Section 5.3**. The access code is removed, and the ARS table matches the remaining outbound dialed digits and sends them to the designated route-pattern (see **Section 5.11**).

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

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

Step 2 - Repeat **Step 1** for all other outbound call strings. In addition, IPFR-EF Call Forward feature access codes ***7** and ***9** are defined here as well.

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

          Dialed            Total     Route    Call   Node  ANI
          String          Min  Max   Pattern   Type   Num   Reqd
    1720                  11   11     1        fnpa         n
    18                    11   11     1        fnpa         n
    19                    11   11     1        fnpa         n
    1900                  11   11     deny     fnpa         n
    1900555               11   11     deny     fnpa         n
    1xxx976               11   11     deny     fnpa         n
    *7                    3    16     4        svcl         n
    *9                    3    16     4        svcl         n
    311                   3    3      4        svcl         n
    011                   10   18     2        intl         n
    411                   3    3      4        svcl         n
```

5.13. Automatic Alternate Routing (AAR) Dialing

AAR is used for outbound calls within the CPE.

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

- **Dialed String -** In the reference configuration all SIP telephones used extensions in the range 54xxx, therefore enter **54**.
- **Min** & **Max** – Enter **5**
- **Route Pattern** – Enter **3**
- **Call Type** – Enter **lev0**

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

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

          Dialed            Total     Route    Call   Node  ANI
          String          Min  Max   Pattern   Type   Num   Reqd
    54                    5    5      3        lev0         n
```

5.14. Avaya G430 Media Gateway Provisioning

In the reference configuration, an Avaya G430 Media Gateway is provisioned. The G430 is used for local DSP resources, announcements, Music On Hold, etc.

Note – Only the Media Gateway provisioning associated with the G430 registration to Communication Manager is shown below. For additional information for the provisioning of the Medias Gateway see **[7]** in the References section.

Step 1 - Use SSH to connect to the G430 (not shown). Note that the Media Gateway prompt will contain "???" if the Media Gateway is not registered to Communication Manager (e.g., ***G430-???(super)#***).
Step 2 - Enter the **show system** command and copy down the G430 serial number.
Step 3 - Enter the **set mgc list x.x.x.x** command where x.x.x.x is the IP address of the Communication Manager Procr (e.g., **10.64.91.75**, see **Section 5.5**).
Step 4 - Enter the **copy run start** command to save the G430 configuration.
Step 5 - From Communication Manager SAT, enter **add media-gateway x** where x is an available Media Gateway identifier (e.g., **1**).
Step 6 – On the Media Gateway form (not shown), enter the following parameters:

- Set **Type** = **g430**.
- Set **Name** = a descriptive name (e.g., **G430-1**).
- Set **Serial Number** = enter the serial number copied from **Step 2**.
- Set the **Link Encryption Type** parameter as desired (**any-ptls/tls** was used in the reference configuration).
- Set **Network Region** = 1.

Wait a few minutes for the G430 to register to Communication Manager. When the Media Gateway registers, the G430 SSH connection prompt will change to reflect the Media Gateway Identifier assigned in **Step 5** (e.g., ***G430-001(super)#***).

Step 7 - Enter the **display media-gateway 1** command and verify that the G430 has registered.

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

                    Type: g430
                    Name: G430-1
               Serial No: 11IS31439520
    Link Encryption Type: any-ptls/tls       Enable CF? n
          Network Region: 1                   Location: 1
         Use for IP Sync? n                  Site Data:
           Recovery Rule: none

              Registered?  y
   FW Version/HW Vintage: 41 .9  .0  /1
       MGP IPV4 Address: 10.64.91.91
       MGP IPV6 Address:
   Controller IP Address: 10.64.91.75
             MAC Address: 00:1b:4f:53:37:69

   Mutual Authentication? optional
```

5.15. Avaya Aura® Media Server Provisioning

In the reference configuration, an Avaya Aura® Media Server is provisioned. The Media Server is used, along with the G430 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]** in the References section for additional information.

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

Step 2 - On the Media Server Element Manager, navigate to **Home → System Configuration → Signaling Protocols → SIP →Node and Routes** and add the Communication Manager Procr interface IP address (e.g., **10.64.91.75**, see **Section 5.4**) 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., **80**), 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 5.4**.
- **Far-end Node Name** – Set to the node name of Media Server as administered in **Section 5.4** (e.g., **AMS**).
- **Near-end Listen Port** – Set to **9061** (default).
- **Far-end Listen Port** – Set to **5061** (default).
- **Far-end Network Region** – Set the IP network region to **1**, as set in **Section 5.6.1**.
- **Far-end Domain** – Automatically populated with the IP address of the Media Server.

```
add signaling-group 80                                              Page   1 of   2
                                SIGNALING GROUP

 Group Number: 80             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: 10.64.91.86
```

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., **80**).
- **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: 80
          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
```

5.16. Save Translations

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

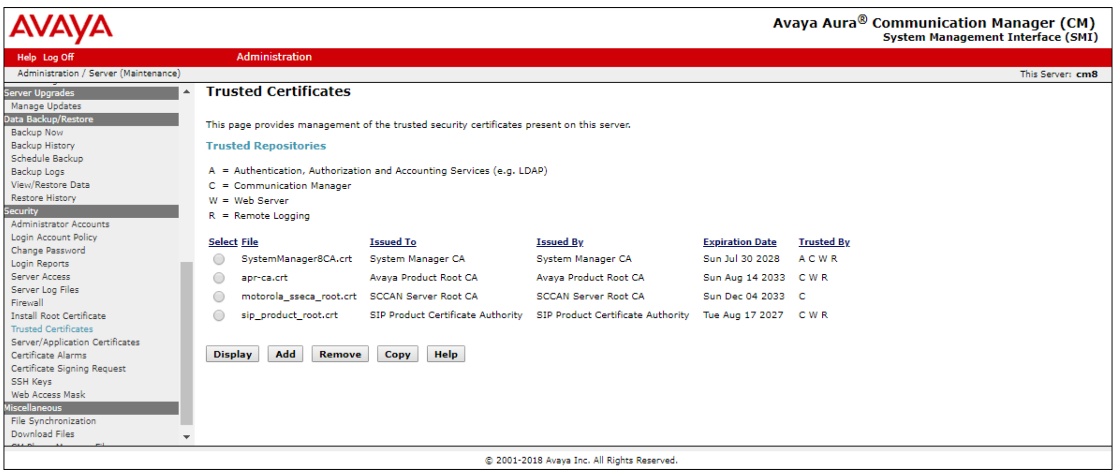
5.17. Verify TLS Certificates – Communication Manager

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

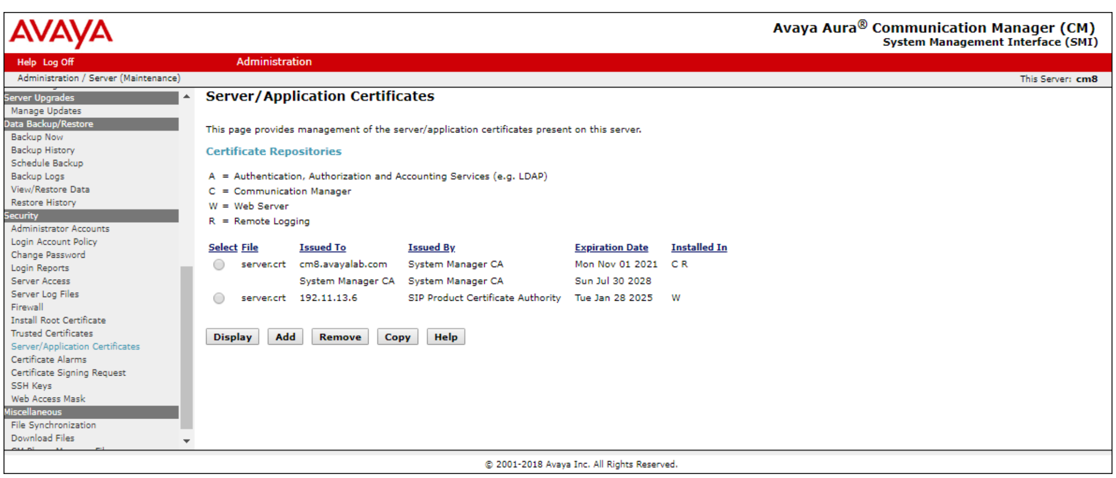
In the reference configuration, TLS transport is used for the communication between Session Manager and Communication Manager. Follow the steps below to verify the certificates used by Communication Manager.

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

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



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



6. Configure Avaya Aura® Session Manager

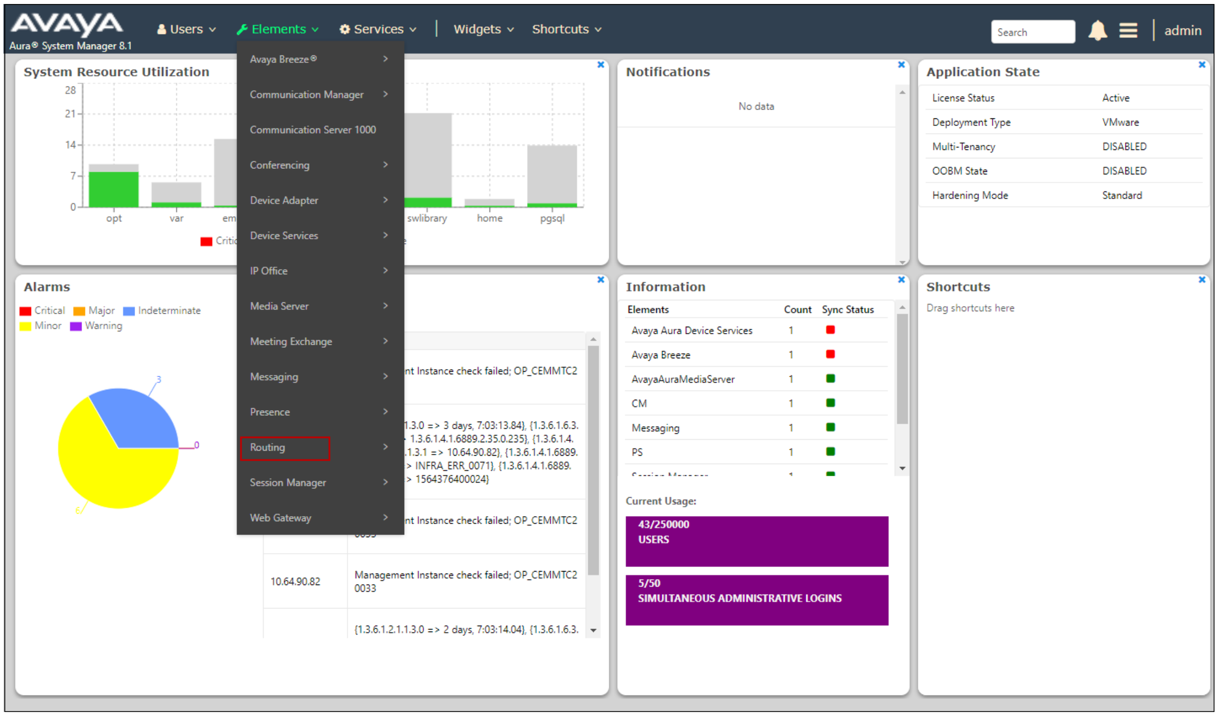
Note – These Application Notes assume that basic System Manager and Session Manager administration has already been performed. Consult **[1] - [4]** in the References section for further details.

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

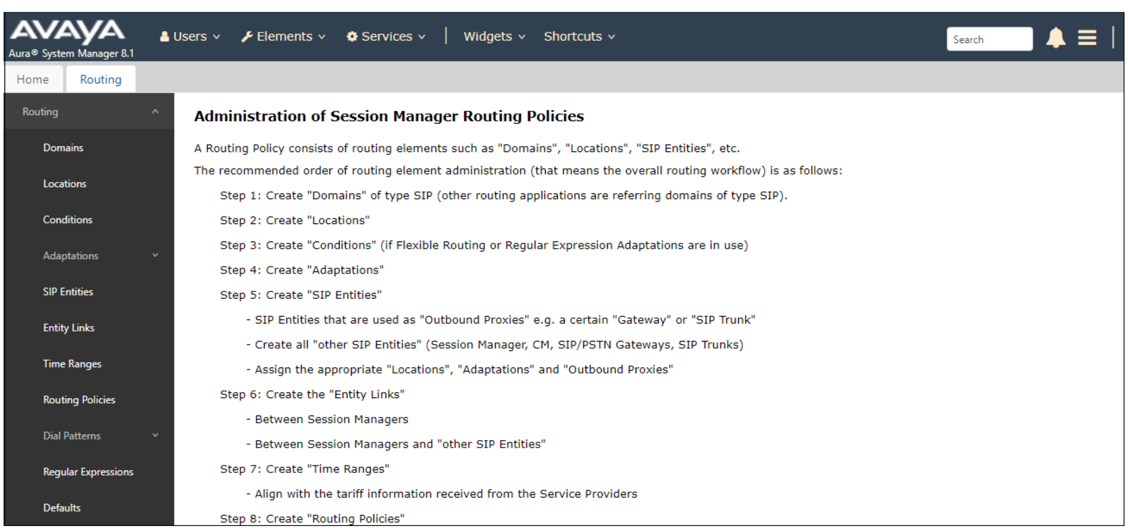
- Define a SIP Domain.
- Define a Location for Customer Premises Equipment (CPE).
- Configure the Adaptation Modules that will be associated with the SIP Entities for Communication Manager and the Avaya SBCE.
- Define SIP Entities corresponding to Session Manager, Communication Manager, the Avaya SBCE,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

6.1. System Manager Login and Navigation

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



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



6.2. 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.

6.3. Locations

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

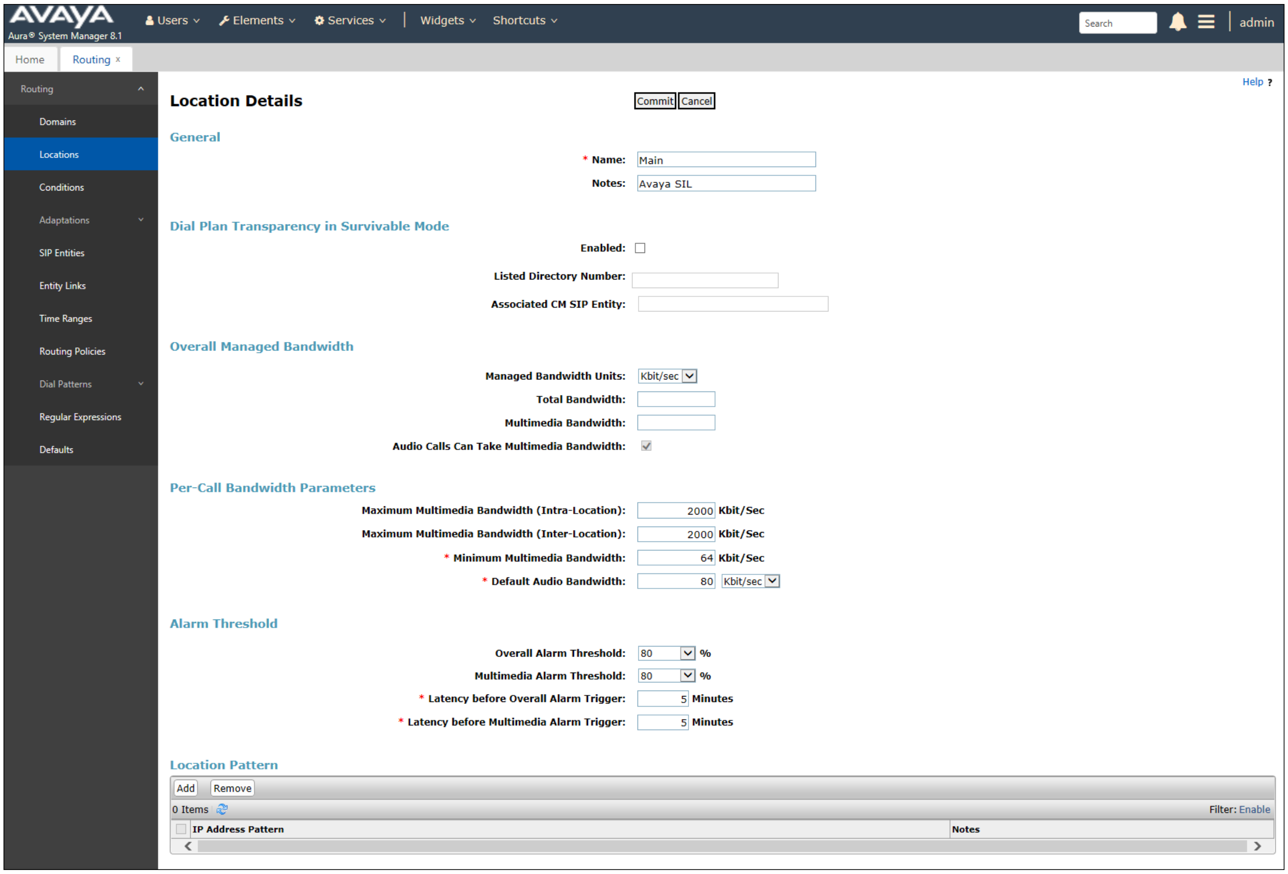
- **Main** – The customer site containing System Manager, Session Manager, and local SIP endpoints.
- **CM-TG-5** – Communication Manager trunk group 5 designated for AT&T.
- **Common SBCs** – Avaya SBCE

6.3.1. Main Location

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

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

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



6.3.2. CM-TG-5 Location

To configure the Communication Manager Trunk Group 5 Location, repeat the steps in **Section 6.3.1** with the following changes (not shown):

- **Name** – Enter a descriptive name (e.g., **CM-TG-5**).

6.3.3. Common-SBCs Location

To configure the Avaya SBCE Location, repeat the steps in **Section 6.3.1** with the following changes (not shown):

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

6.4. Configure Adaptations

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

- Calls from AT&T (**Section 6.4.1**) - Modification of SIP messages sent to Communication Manager extensions.
  - The AT&T DNIS number digit string in the Request URI is replaced with the associated Communication Manager extensions/VDN.
- Calls to AT&T (**Section 6.4.2**) - Modification of SIP messages sent by Communication Manager extensions.
  - The History-Info header is removed automatically by the **AttAdapter**.
  - Avaya SIP headers not required by AT&T are removed (see **Section 2.4**).

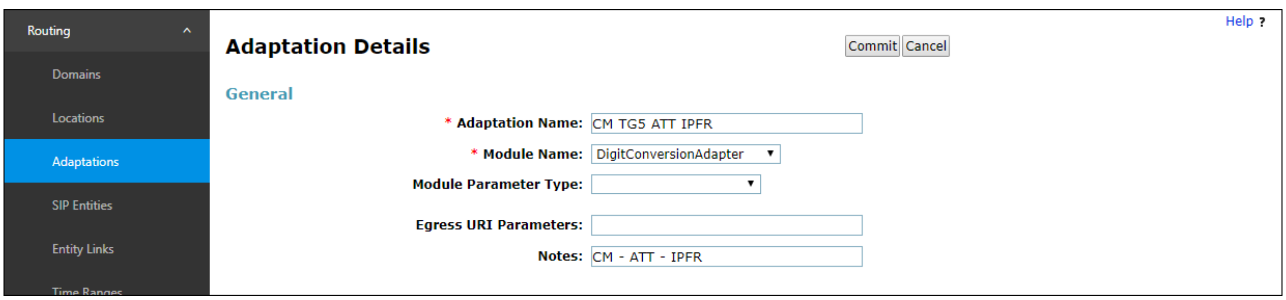
6.4.1. Adaptation for Avaya Aura® Communication Manager Extensions

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

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

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

1. A descriptive **Name**, (e.g., **CM TG5 ATT IPFR**).
2. Select **DigitConversionAdapter** from the **Module Name** drop-down menu (if no module name is present, select **<click to add module>** and enter **DigitConversionAdapter**).



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

1. **Example 1 – destination extension range**: 30355593xx is a range of DNIS digits sent in the Request URI by the IPFR-EF service that is associated with Communication Manager extension range 59300 thru 59399.
   - Enter **30355593** in the **Matching Pattern** column.
   - Enter **10** in the **Min/Max** columns.
   - Enter **5** in the **Delete 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 AT&T DNIS numbers/Communication manager extensions.

Step 5 - Click on **Commit**.

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

Note – In the reference configuration, the AT&T IPFR-EF service delivered 10-digit DNIS numbers.

Digit Conversion for Outgoing Calls from SM

Add Remove

2 Items Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* 30355593	* 10	* 10		* 5		destination		10 digit DNIS to extension
	* 46955548	* 10	* 10		* 5		destination		10 digit DNIS to extension

Select : All, None

Commit Cancel

6.4.2. Adaptation for the AT&T IP Flexible Reach – Enhanced Features 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 6.4.1** with the following changes.

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

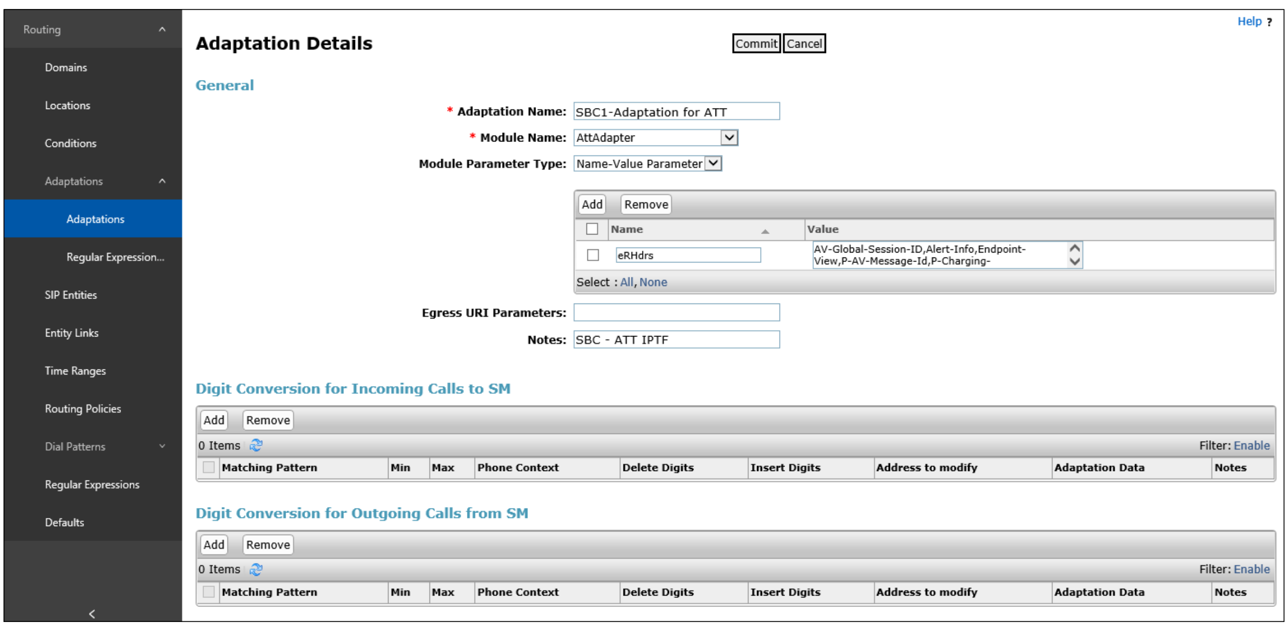
1. A descriptive **Name**, (e.g., **SBC1-Adaptation for ATT**).
2. Select **AttAdapter** from the **Module Name** drop-down menu (if no module name is present, select **<click to add module>** and enter **AttAdapter**). The AttAdapter will automatically remove History-Info headers, (which the IPFR-EF service does not support), sent by Communication Manager (see **Section 5.8.1**).

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.
   - **AV-Global-Session-ID,Alert-Info,Endpoint-View,P-AV-Message-Id,P-Charging-Vector,P-Location,AV-Correlation-ID,Av-Secure-Indication**

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



6.5. SIP Entities

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

- Session Manager (**Section 6.5.1**).
- Communication Manager for AT&T trunk access (**Section 6.5.2**) – This entity, and its associated Entity Link (using TLS with port 5065), is for calls to/from AT&T and Communication Manager via the Avaya SBCE.
- Communication Manager for local trunk access (**Section 6.5.3**) – This entity, and its associated Entity Link (using TLS with port 5061), is primarily for traffic between Avaya SIP telephones and Communication Manager, as well as calls to Messaging.
- Avaya SBCE (**Section 6.5.4**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls to/from the IPFR-EF service via the Avaya SBCE.
- Messaging (**Section 6.5.5**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls to/from Messaging.
- Experience Portal (**Section 6.5.6**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls to/from Experience Portal.

Note – In the reference configuration, TLS is used as the transport protocol between Session Manager and Communication Manager (ports 5061 and 5065), and to the Avaya SBCE (port 5061). The connection between the Avaya SBCE and the AT&T IPFR-EF service uses UDP/5060 per AT&T requirements.

6.5.1. Avaya Aura® Session Manager SIP Entity

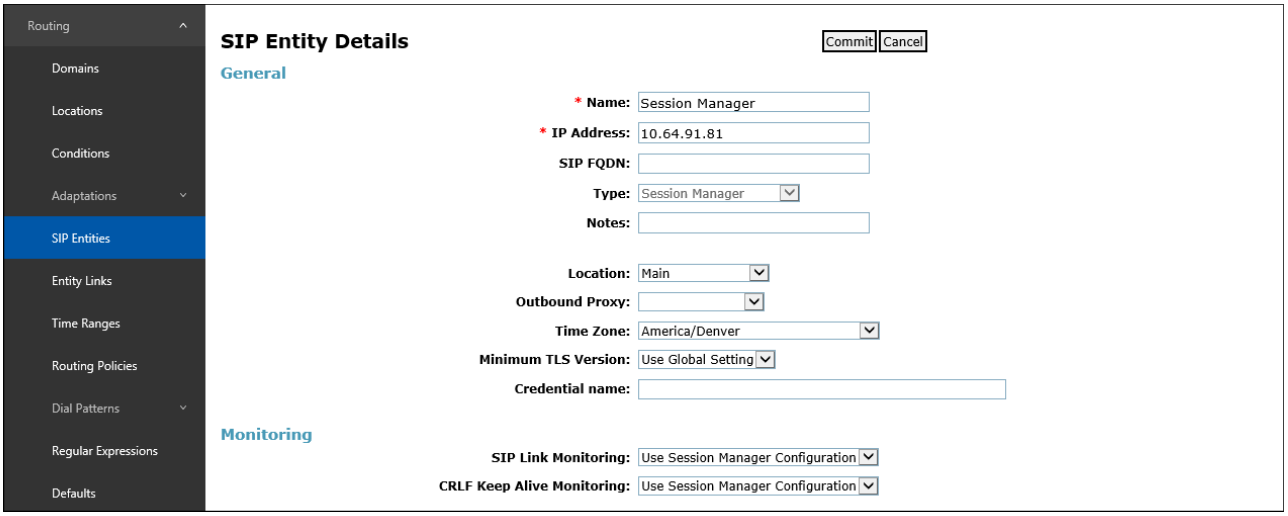
Step 1- In the left pane under **Routing**, click on **SIP Entities**. In the **SIP Entities** page click on **New** (not shown).

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

- **Name –** Enter a descriptive name (e.g., **SessionManager**).
- **FQDN or IP Address –** Enter the IP address of Session Manager signaling interface, (*not* the management interface), provisioned during installation (e.g., **10.64.91.81**).
- **Type –** Verify **Session Manager** is selected.
- **Location** – Select location **Main** (**Section 6.3.1**).
- **Outbound Proxy** – (Optional) Leave blank or select another SIP Entity. For calls to SIP domains for which Session Manager is not authoritative, Session Manager routes those calls to this **Outbound Proxy** or to another SIP proxy discovered through DNS if **Outbound Proxy** is not specified.
- **Time Zone** – Select the time zone in which Session Manager resides.
- **Minimum TLS Version** – Select the TLS version, or select **Use Global Settings** to use the default TLS version, configurable at the global level (**Elements→Session Manager→Global Settings**).

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

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



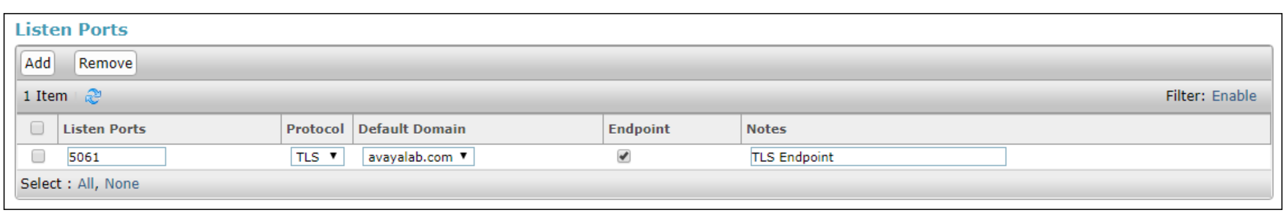
Step 4 - Scrolling down to the **Listen Port** section of the **SIP Entity Details** page. This section defines a default set of ports that Session Manager will use to listen for SIP requests, typically from registered SIP endpoints. Session Manager can also listen on additional ports defined elsewhere such as the ports specified in the SIP Entity Link definition in **Section 6.6**. Click on **Add** and provision entries as follows:

- **Port** – Enter **5061**
- **Protocol** – Select **TLS**
- **Default Domain** – Select a SIP domain administered in **Section 6.2** (e.g., **avayalab.com**)
- **Endpoint** – Check the checkbox to have this port be used for SIP endpoint registration.

Step 5 - Enter any notes as desired and leave all other fields on the page blank/default.

Step 6 - Click on **Commit**.

Note – The **Entity Links** section of the form (not shown) will be automatically populated when the Entity Links are defined in **Section 6.6**. The **SIP Responses to an OPTIONS Request** section of the form is not used in the reference configuration.



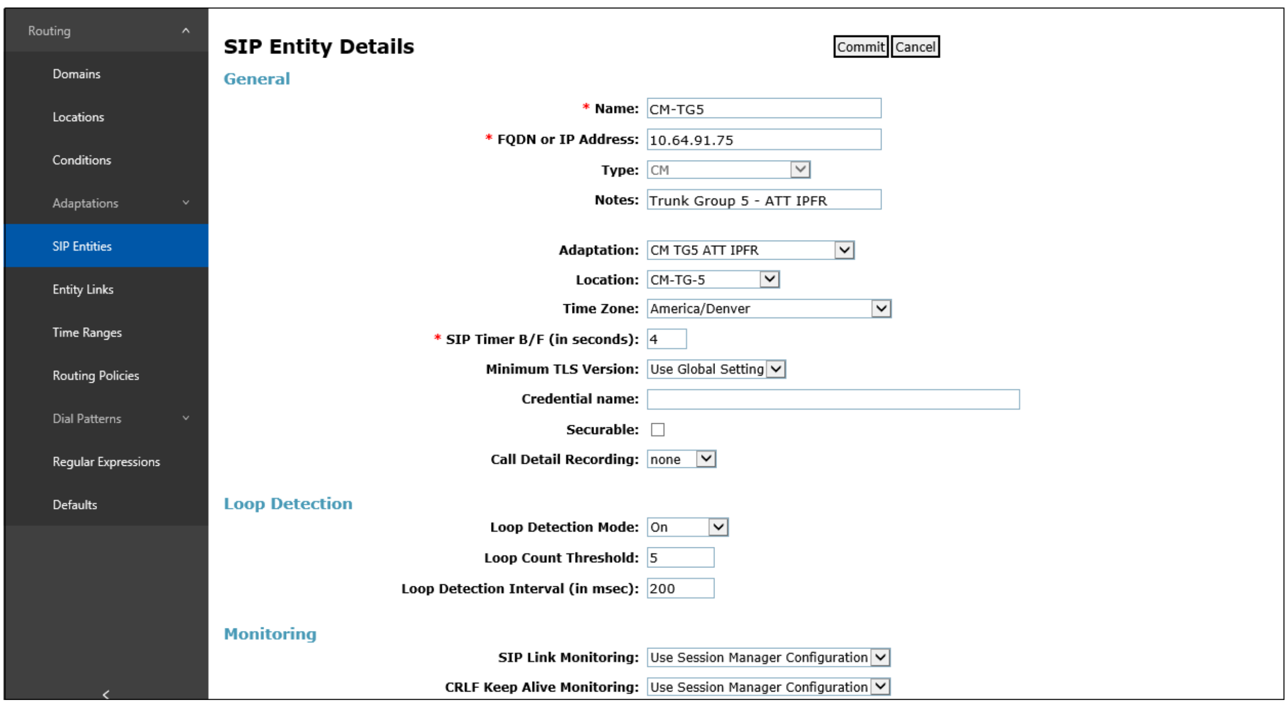
6.5.2. Avaya Aura® Communication Manager SIP Entity – Public Trunk

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-TG5**).
- **FQDN or IP Address –** Enter the IP address of Communication Manager Processor Ethernet (procr) described in **Section 5.5** (e.g., **10.64.91.75**).
- **Type –** Select **CM**.
- **Adaptation –** Select the Adaptation **CM TG5 ATT IPFR** administered in **Section 6.4.1**.
- **Location** – Select Location **CM-TG-5** administered in **Section 6.3.2**.
- **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**.



6.5.3. Avaya Aura® Communication Manager SIP Entity – Local Trunk

To configure the Communication Manager Local trunk SIP Entity, repeat the steps in **Section 6.5.2** with the following changes:

- **Name –** Enter a descriptive name (e.g., **CM-TG3**).
- **Adaptations –** Leave this field blank.
- **Location –** Select Location **Main** administered in **Section 6.3.1**.

6.5.4. Avaya Session Border Controller for Enterprise SIP Entity

Repeat the steps in **Section 6.5.2** with the following changes:

- **Name –** Enter a descriptive name (e.g., **SBCE-ATT**).
- **FQDN or IP Address –** Enter the IP address of the A1 (private) interface of the Avaya SBCE (e.g., **10.64.91.40**, see **Section 8.3**.
- **Type –** Select **SIP Trunk**.
- **Adaptations –** Select Adaptation **SBC1-Adaptation for ATT** (**Section 6.4.2**).
- **Location –** Select Location **Common-SBCs** administered in **Section 6.3.3**.

6.5.5. Avaya Aura® Messaging SIP Entity

Repeat the steps in **Section 6.5.2** with the following changes:

- **Name –** Enter a descriptive name (e.g., **Aura Messaging**).
- **FQDN or IP Address –** Enter the IP address of Messaging (e.g., **10.64.91.54**, see **Section 3.1**).
- **Type –** Select **Messaging**.
- **Adaptations –** Leave this field blank.
- **Location –** Select Location **Main** administered in **Section 6.3.1**.

6.5.6. Avaya Aura® Experience Portal SIP Entity

Repeat the steps in **Section 6.5.2** with the following changes:

- **Name –** Enter a descriptive name (e.g., **ExperiencePortal**).
- **FQDN or IP Address –** Enter the IP address of Experience Portal (e.g., **10.64.91.90,** see **Section 3.1**).
- **Type –** Select **Voice Portal**.
- **Adaptations –** Leave this field blank.
- **Location –** Select Location **Main** administered in **Section 6.3.1**.

6.6. Entity Links

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

- Session Manager to Communication Manager Public trunk (**Section 6.6.1**).
- Session Manager to Communication Manager Local trunk (**Section 6.6.2**).
- Session Manager to Avaya SBCE (**Section 6.6.3**).
- Session Manager to Messaging (**Section 6.6.4**).
- Session Manager to Experience Portal (**Section 6.6.5**).

Note – Once the Entity Links have been committed, the link information will also appear on the associated SIP Entity pages configured in **Section 6.5**.

Note – See the information in **Section 6.5** regarding the transport protocols and ports used in the reference configuration.

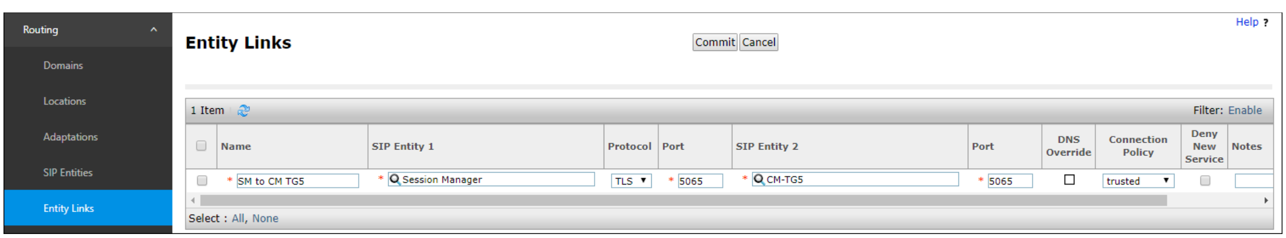
6.6.1. Entity Link to Avaya Aura® Communication Manager – Public Trunk

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 TG5**).
- **SIP Entity 1** – Select the SIP Entity administered in **Section 6.5.1** for Session Manager (e.g., **SessionManager**).
- **Protocol** – Select **TLS** (see **Section 5.8.1**).
- SIP Entity 1 **Port** – Enter **5065**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.5.2** for the Communication Manager public entity (e.g., **CM-TG5**).
- SIP Entity 2 **Port** – Enter **5065** (see **Section 5.8.1**).
- **Connection Policy** – Select **trusted**.
- Leave other fields as default.

Step 3 - Click on **Commit**.



6.6.2. Entity Link to Avaya Aura® Communication Manager – Local Trunk

To configure this Entity Link, repeat the steps in **Section 6.6.1**, with the following changes:

- **Name** – Enter a descriptive name for this link to Communication Manager (e.g., **SM to CM TG3**).
- SIP Entity 1 **Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.5.3** for the Communication Manager local entity (e.g., **CM-TG3**).
- SIP Entity 2 **Port** – Enter **5061** (see **Section 5.8.2**).

6.6.3. Entity Link for the AT&T IP Flexible Reach – Enhanced Features Service via the Avaya SBCE

To configure this Entity Link, repeat the steps in **Section 6.6.1**, with the following changes:

- **Name** – Enter a descriptive name for this link to the Avaya SBCE (e.g., **SM to SBCE**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.5.4** for the Avaya SBCE entity (e.g., **SBCE-ATT**).
- **SIP Entity 2 Port** – Enter **5061**.

6.6.4. Entity Link to Avaya Aura® Messaging

To configure this Entity Link, repeat the steps in **Section 6.6.1**, with the following changes:

- **Name** – Enter a descriptive name for this link to Messaging (e.g., **SM to AAM**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.5.5** for the Aura® Messaging entity (e.g., **Aura Messaging**).
- **SIP Entity 2 Port** – Enter **5061**.

6.6.5. Entity Link to Avaya Aura® Experience Portal

To configure this Entity Link, repeat the steps in **Section 6.6.1**, with the following changes:

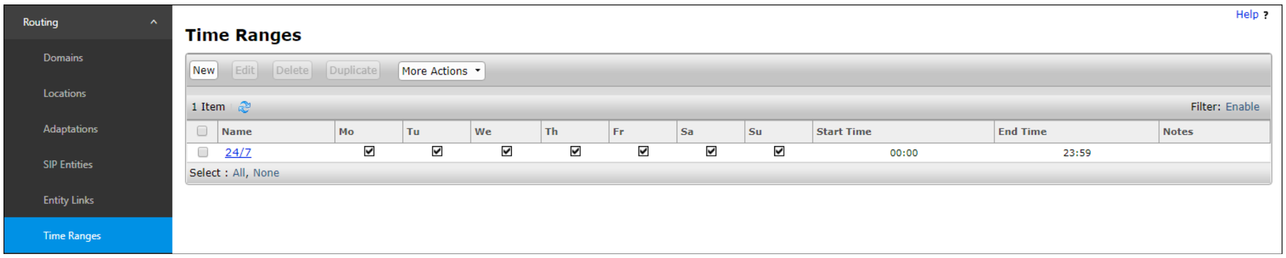
- **Name** – Enter a descriptive name for this link to Messaging (e.g., **SM to ExperiencePortal**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.5.6** for the Experience Portal entity (e.g., **ExperiencePortal**).
- **SIP Entity 2 Port** – Enter **5061**.

6.7. Time Ranges – (Optional)

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.



6.8. Routing Policies

In this section, the following Routing Policies are administered:

- Inbound calls to Communication Manager extensions (**Section 6.8.1**).
- Inbound calls to Aura® Messaging (**Section 6.8.2**).
- Inbound calls to Experience Portal (**Section 6.8.3**).
- Outbound calls to AT&T/PSTN (**Section 6.8.4**).

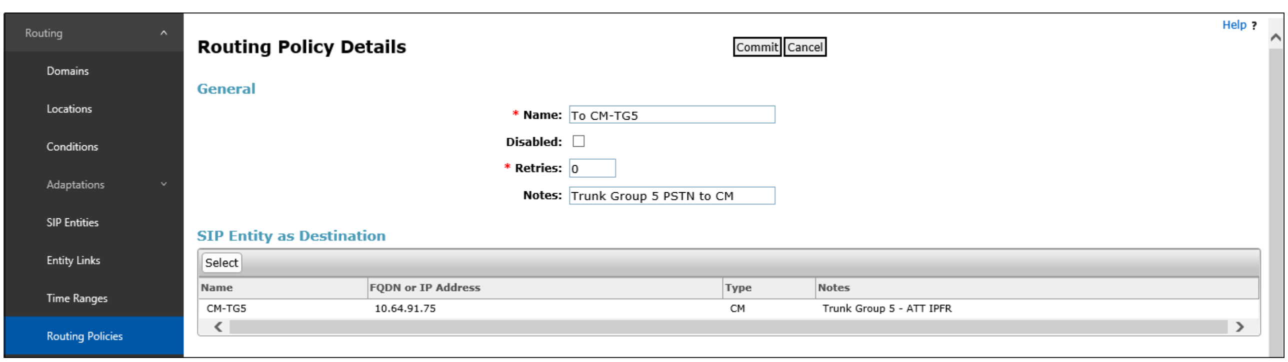
6.8.1. Routing Policy for AT&T Inbound Calls to Avaya Aura® Communication Manager

This Routing Policy is used for inbound calls from AT&T.

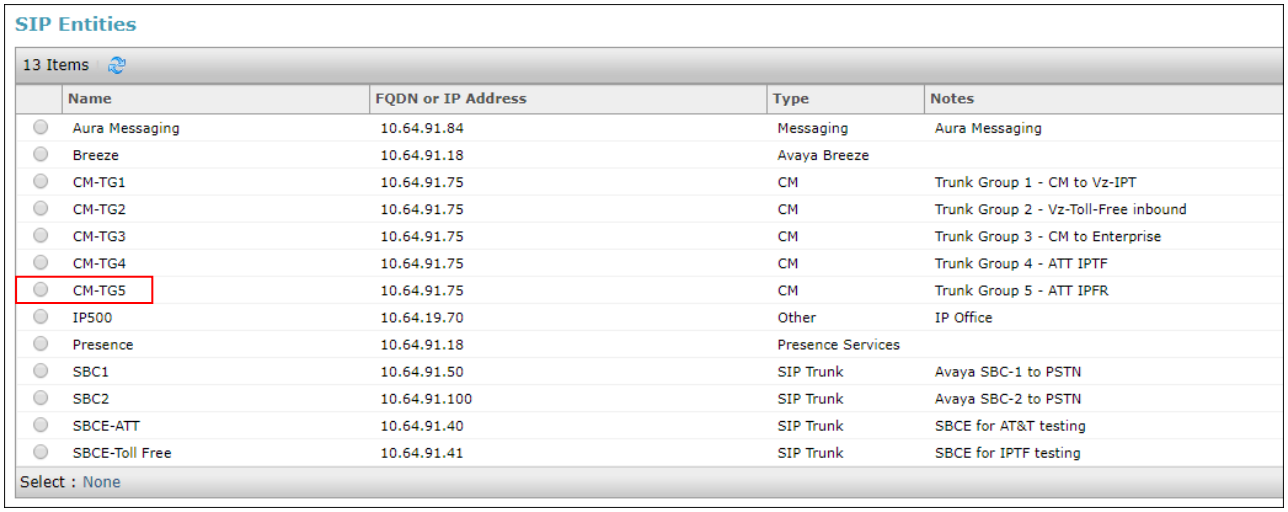
Step 1 - In the left pane under **Routing**, click on **Routing Policies**. In the **Routing Policies** page click on **New** (not shown).

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-TG5**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.

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

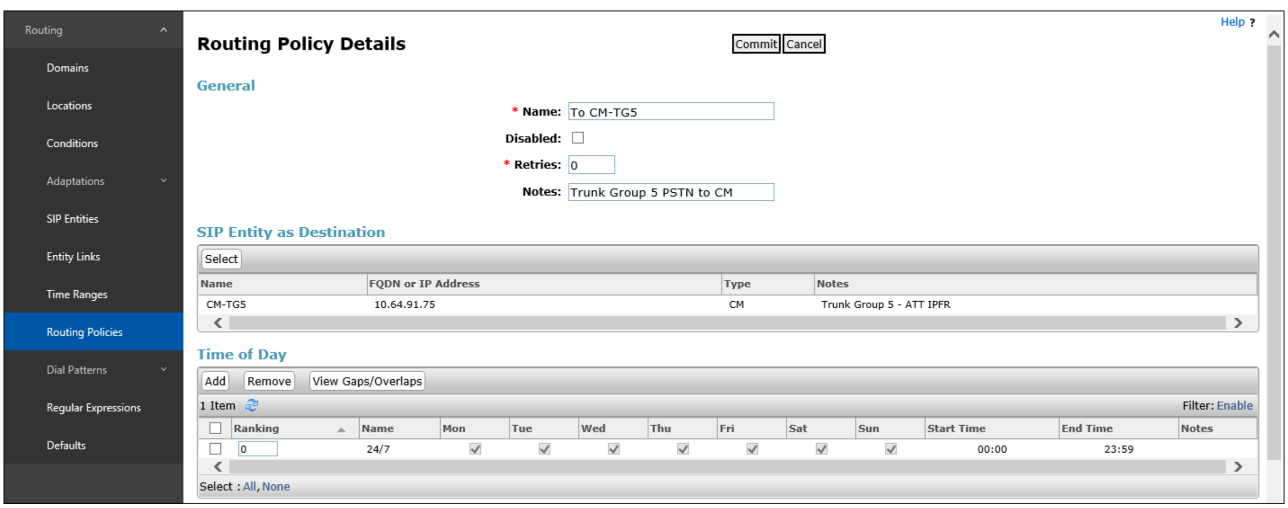


Step 4 - In the **SIP Entities** list page, select the SIP Entity administered in **Section 6.5.2** for the Communication Manager public SIP Entity (**CM-TG5**), and click on **Select**.



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 6.7.7**, and click on **Select**.
Step 7 - Returning to the **Routing Policy Details** page in the **Time of Day** section, enter a **Ranking** of **0**.
Step 8 - No **Regular Expressions** were used in the reference configuration.
Step 9 - Click on **Commit**.

Note – Once the **Dial Patterns** are defined (**Section 6.9**) they will appear in the **Dial Pattern** section of this form.



6.8.2. Routing Policy for Inbound Routing to Avaya Aura® Messaging

This routing policy is for inbound calls to Aura® Messaging for message retrieval. Repeat the steps in **Section 6.8.1** with the following differences:

- Enter a descriptive **Name** (e.g., **To AAM**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.
- In the **SIP Entities** list page, select the SIP Entity administered in **Section 6.5.5** for Aura® Messaging (e.g., **AAM**).

6.8.3. Routing Policy for Inbound Calls to Experience Portal

This routing policy is for inbound calls to Experience Portal. Repeat the steps in **Section 6.8.1** with the following differences:

- Enter a descriptive **Name** (e.g., **To Experience Portal**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.
- In the **SIP Entities** list page, select the SIP Entity administered in **Section 6.5.6** for Experience Portal (e.g., **ExperiencePortal**).

6.8.4. Routing Policy for Outbound Calls to AT&T

This Routing Policy is used for Outbound calls to AT&T. Repeat the steps in **Section 6.8.1** with the following differences:

- Enter a descriptive **Name** for routing calls to the AT&T IPFR-EF service via the Avaya SBCE (e.g., **To SBCE-ATT**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.
- In the **SIP Entities** list page, select the SIP Entity administered in **Section 6.5.4** for the Avaya SBCE SIP Entity (e.g., **SBCE-ATT**).

6.9. Dial Patterns

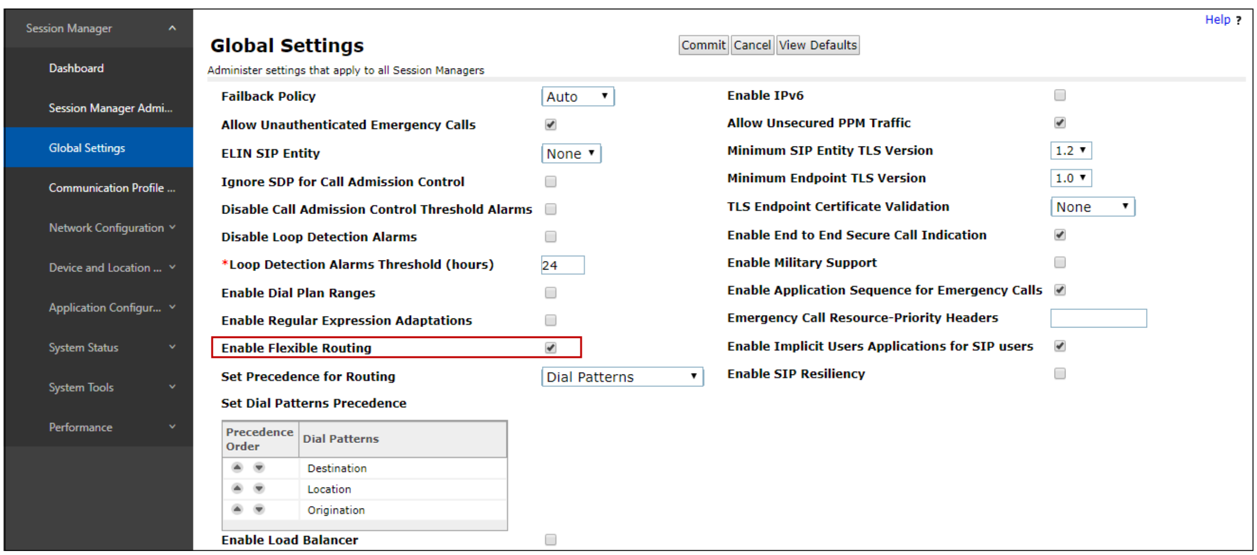
In this section, Dial Patterns are administered matching the following calls:

- Inbound PSTN calls via the IPFR-EF service to Communication Manager (**Section 6.9.1**).
- Outbound calls to AT&T (**Section 6.9.2**).

Note: One of the routing enhancements in Session Manager release 8.1 is the addition of Origination Dial Pattern sets functionality. This configuration is optional. Origination Dial Pattern sets can be created to include digits patterns, that can be matched by Session Manager to make more granular routing decisions, like the use of alternate routes or call restriction for calls arriving to Session Manager from different users out of the the same Originating Location. This is done by matching the number present in the From header of the call. More information can be found on [**2**] on the References section.

Origination Dial Patterns were not used in the reference configuration.

If Origination Dial Patterns are to be used in the customer configuration, **Enable Flexible Routing** needs to be checked under **Elements → Session Manager → Global Settings**.



6.9.1. Matching Inbound PSTN Calls to Avaya Aura® Communication Manager

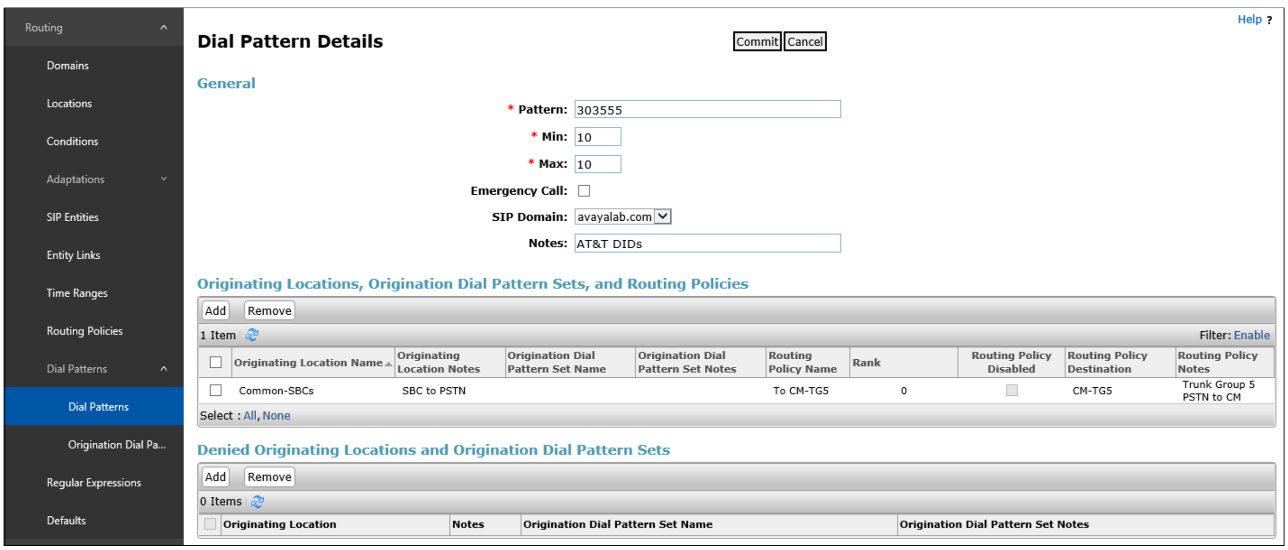
In the reference configuration inbound calls from the IPFR-EF service sent 10 DNIS digits in the SIP Request URI (for security purposes, these digits are represented in this document as 303**555**xxxx). The DNIS pattern must be matched for further call processing. Depending on customer deployments, the IPFR-EF service may send different DNIS digit lengths.

Note – Be sure to match on the DNIS digits specified in the AT&T Request URI, not the DID dialed digits. 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:

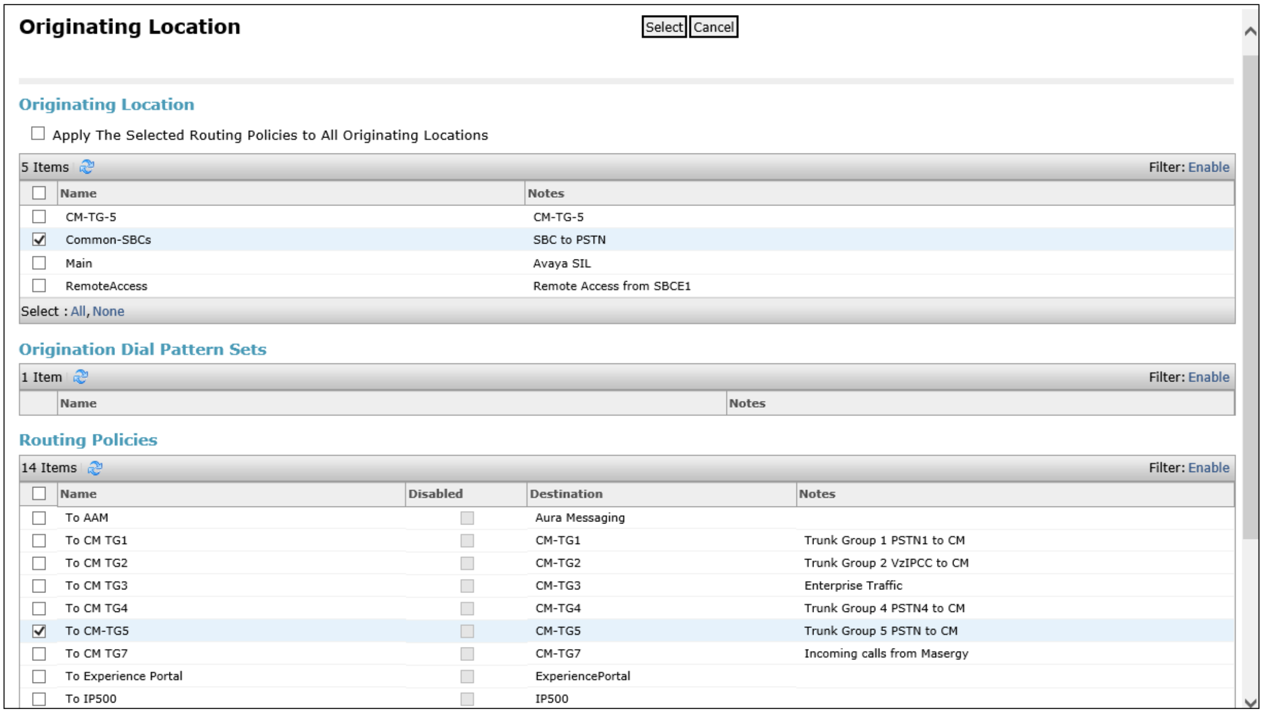
- **Pattern** – Enter **303555**. Note – The Adaptation defined for Communication Manager in **Section 6.4.1** will convert the various 303-555-xxxx numbers into their corresponding Communication Manager extensions.
- **Min** and **Max** – Enter **10**.
- **SIP Domain** – Select the enterprise SIP domain, e.g., **avayalab.com**.



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

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

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 6.8.1** (e.g., **To CM-TG5**), and click on **Select**.



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

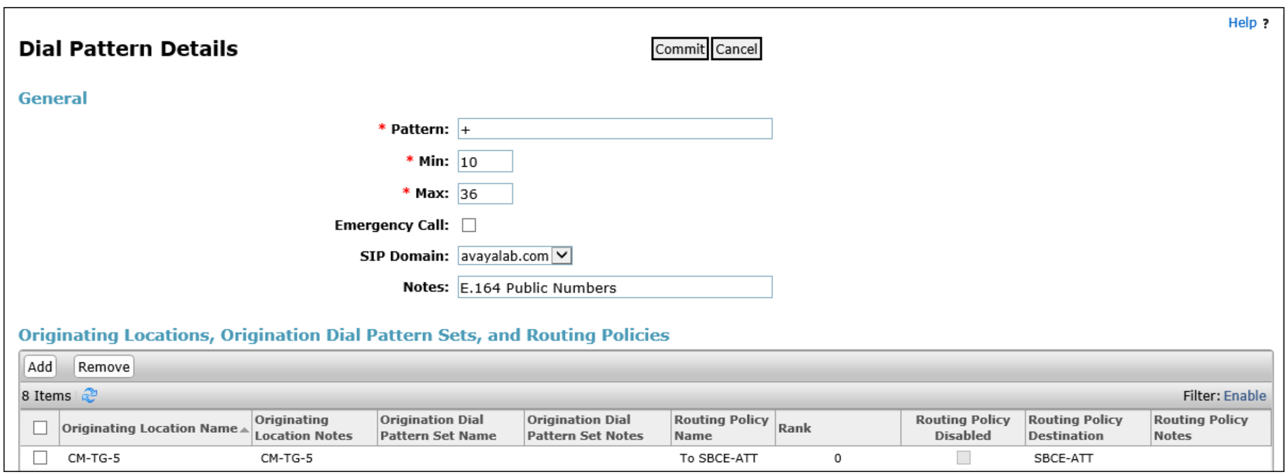
Step 7 - Repeat **Steps 1-6** for any additional inbound dial patterns from AT&T.

6.9.2. Matching Outbound Calls to AT&T

In this section, Dial Patterns are administered for all outbound calls to AT&T. In the reference configuration 1xxxyyyxxxx, x11, and 011 international calls were verified. In addition, IPFR-EF Call Forward feature access codes ***7** and ***9** (e.g., *71yyyzzzxxxx & *91yyyzzzxxxx) are specified.

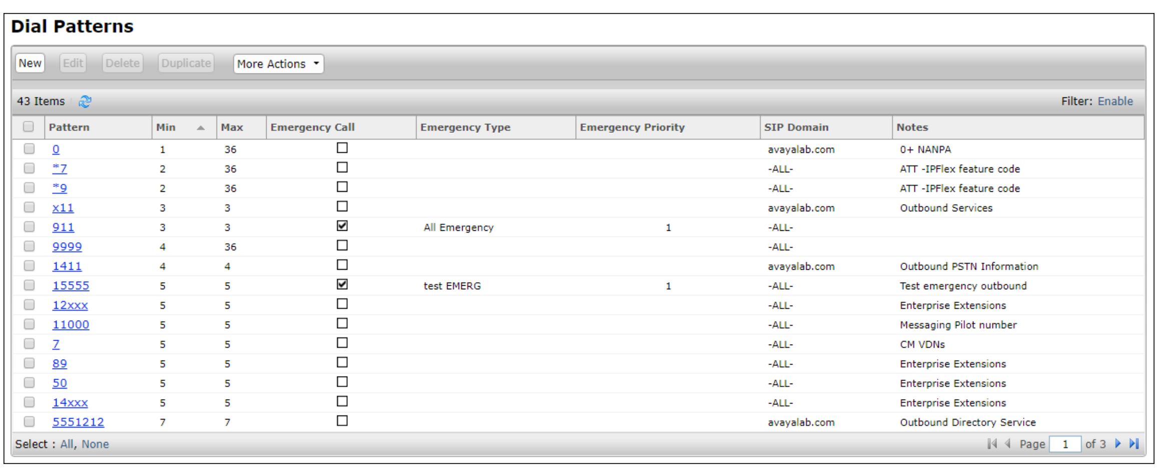
Step 1 - Repeat the steps shown in **Section 6.9.1**, with the following changes:

- In the **General** section of the **Dial Pattern Details** page, enter a dial pattern for routing calls to AT&T/PSTN (e.g., +). This will match any outbound call prefixed with a plus sign (+), such as an E.164 formatted number.
- Enter a **Min** pattern of **10**.
- Enter a **Max** pattern of **36**.
- In the **Routing Policies** section of the **Originating Locations, Origination Dial Patterns and Routing Policies** page, check the checkbox for the Originating Location corresponding to the Communication Manager Trunk Group 5 (e.g., **CM-TG-5**) and the Routing Policy administered for routing calls to AT&T in **Section 6.8.4** (e.g., **To SBCE-ATT**).



Step 2 - Repeat **Step 1** to add patterns for IPFR-EF Call Forward access codes with patterns ***7** and ***9**, and **Min=2/Max=36**.

Step 3 - Repeat **Step 1** to add any additional outbound patterns as required.

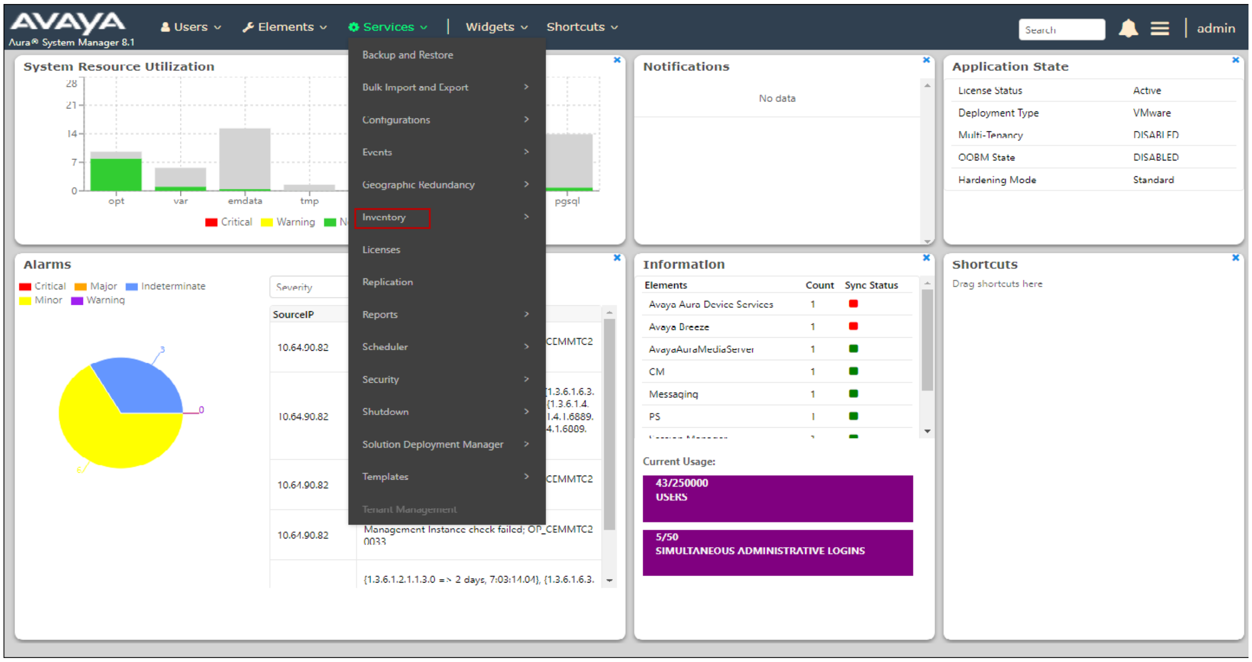


6.10. Verify TLS Certificates – Session Manager

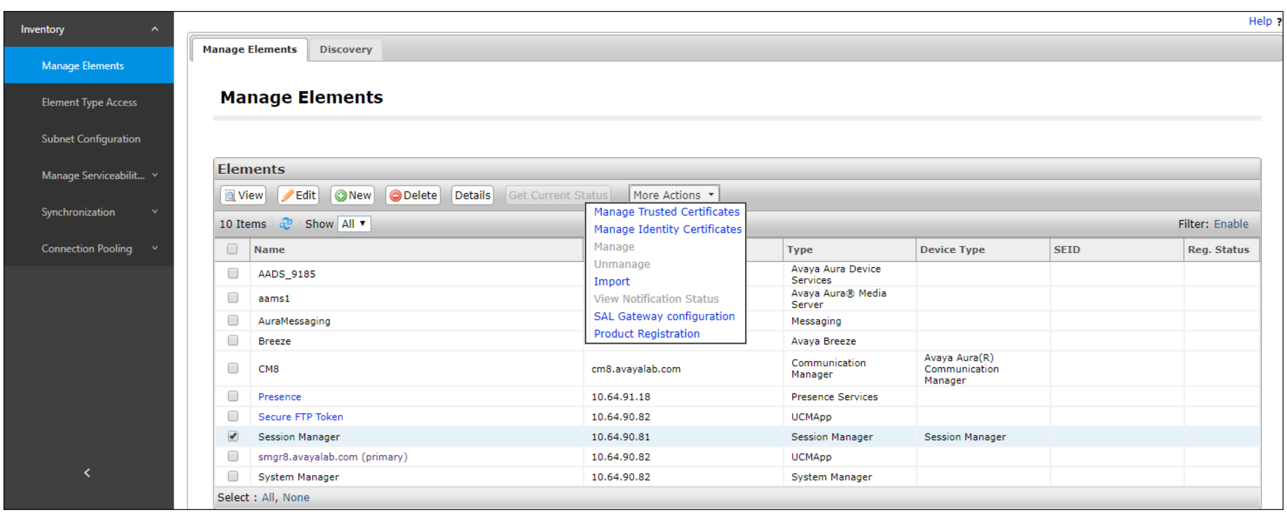
Note – Testing was done with System Manager signed identity certificates. The procedure to obtain and install certificates is outside the scope of these Application Notes.

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

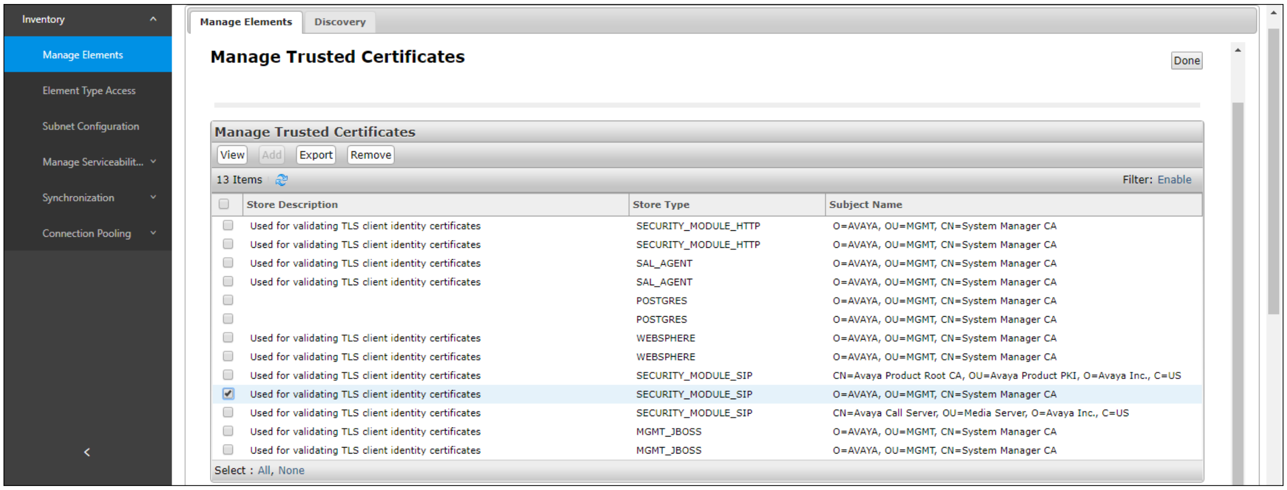
Step 1 - From the **Home** screen, under the **Services** heading, select **Inventory**.



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

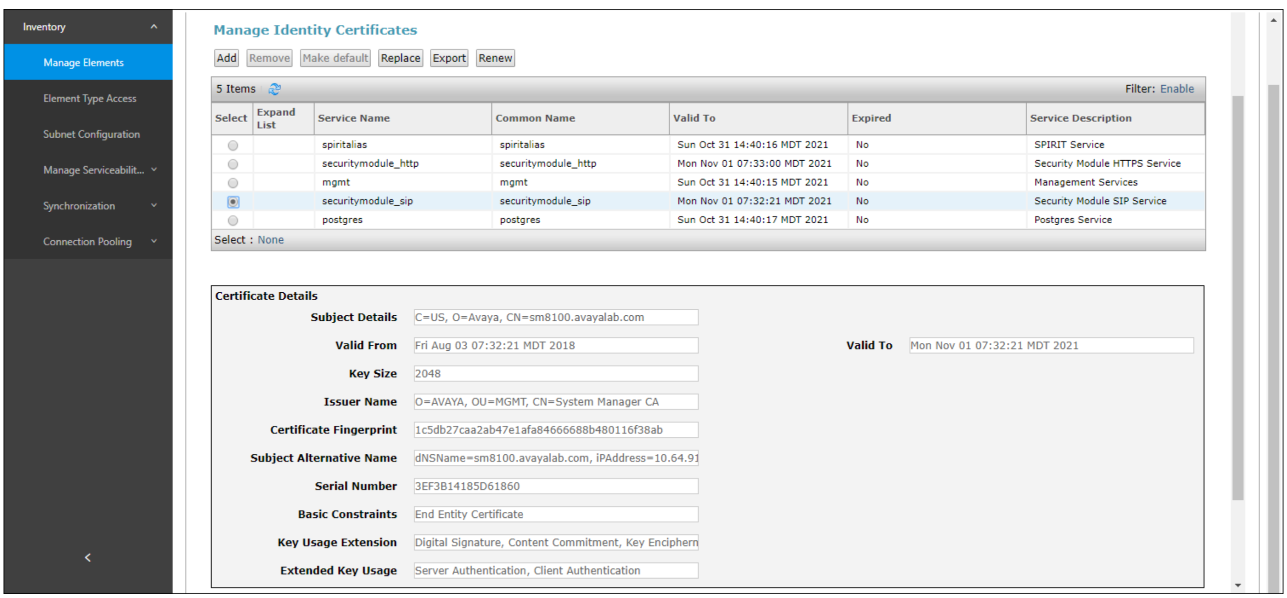


Step 3 - Verify the System Manager Certificate Authority certificate is listed in the trusted store, **SECURITY_MODULE_SIP**. Click **Done** to return to the previous screen.



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

Step 5 - Verify the **Security Module SIP** service has a valid identity certificate signed by System Manager. If the **Subject Details** and **Subject Alternative Name** fields of the System Manager signed certificate need to be updated, click **Replace**, otherwise click **Done**.



7. Avaya Aura® Experience Portal

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

7.1. Background

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

References to the Voice XML and/or Call Control XML applications are administered on Experience Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Experience Portal, the called party DNIS number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match is found, Experience Portal informs the caller that the call cannot be handled, and disconnects the call[3].

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

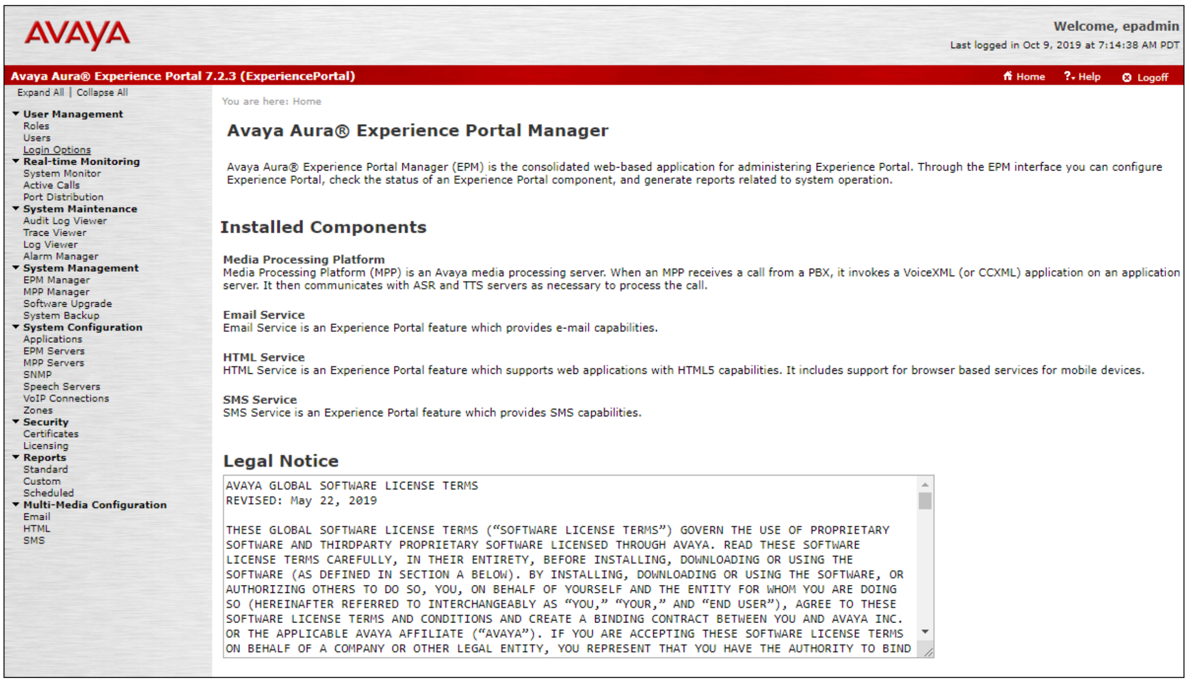
[3] An application may be configured with “inbound default” as the called number, to process all inbound calls that do not match any other application references.

7.2. Logging In and Licensing

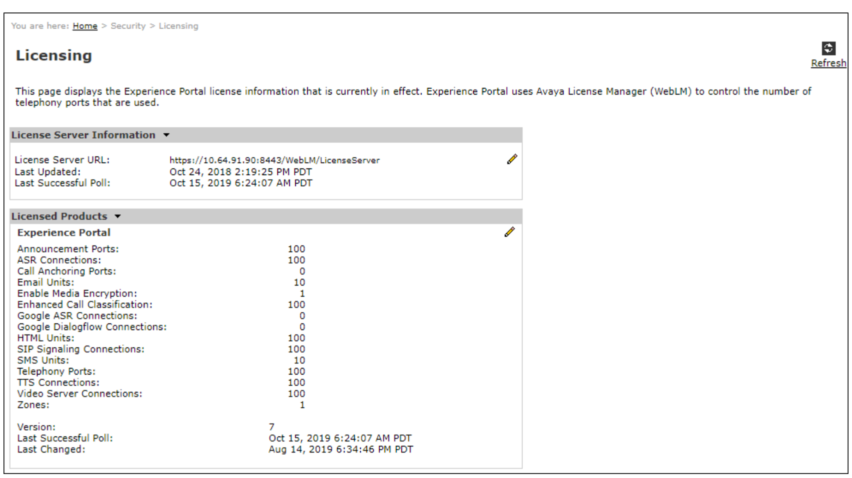
This section describes the steps on Experience Portal for administering a SIP connection to the Session Manager.

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

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



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

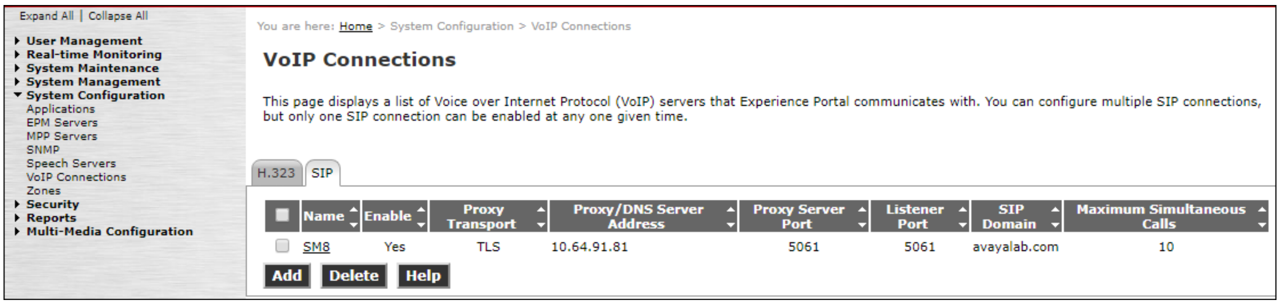


7.3. VoIP Connection

This section defines a SIP trunk between Experience Portal and Session Manager.

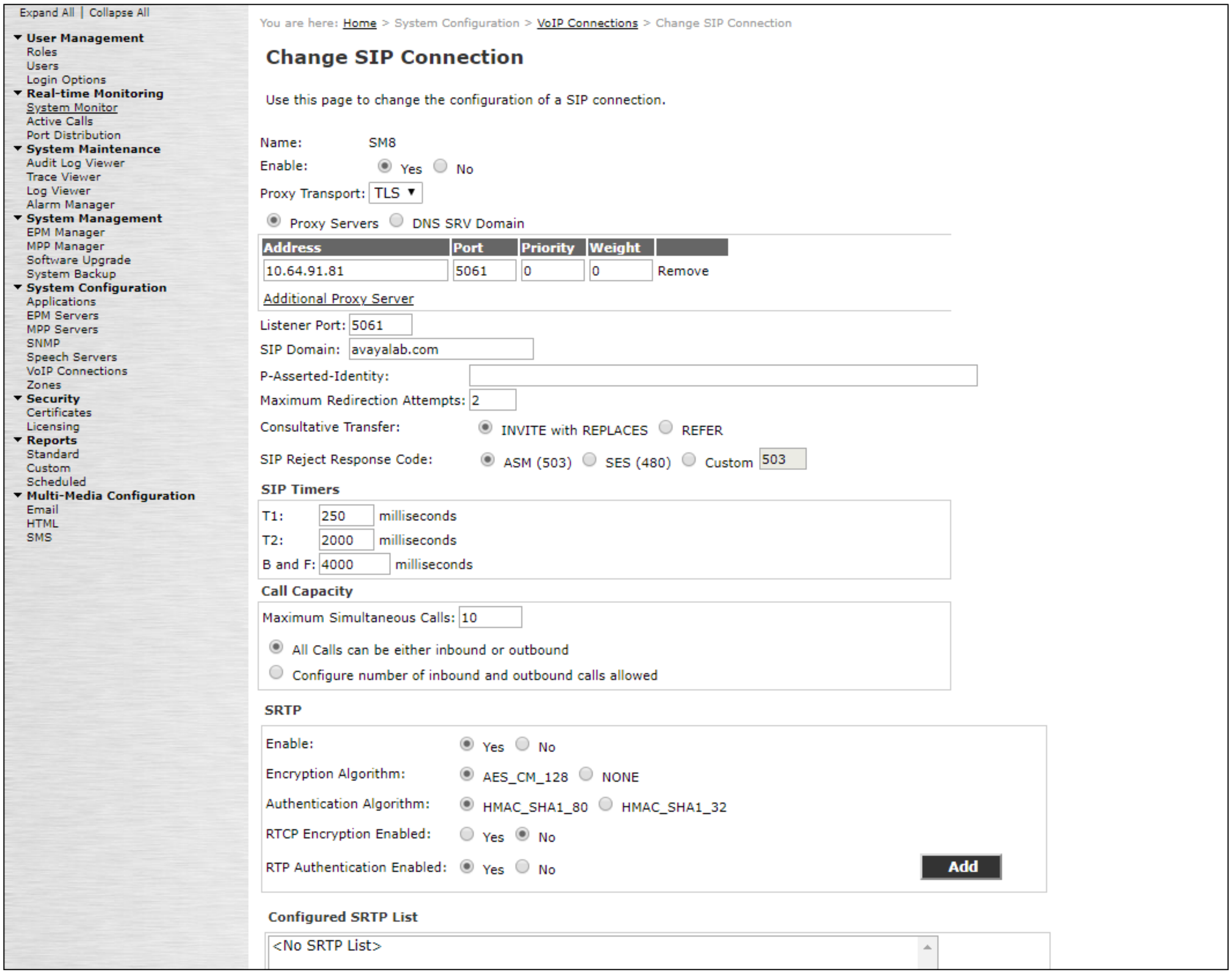
Step 1 - In the left pane, navigate to **System Configuration→VoIP Connections**. On the **VoIP Connections** page, select the **SIP** tab and click **Add** to add a SIP trunk.

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



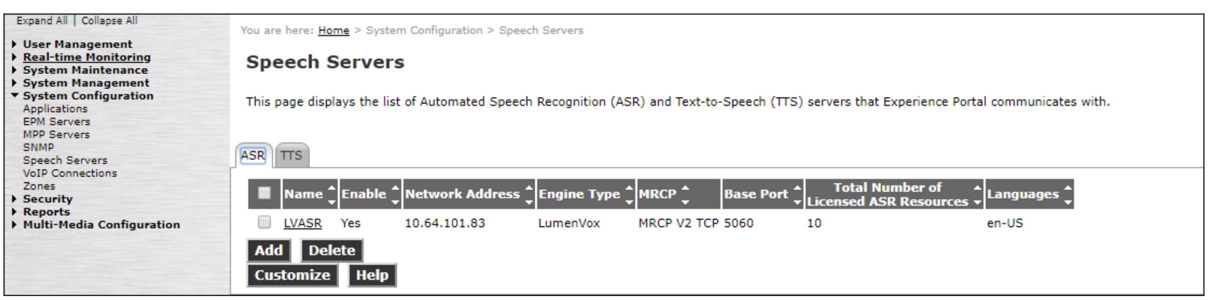
Step 2 - Configure a SIP connection as follows:

- **Name** – Set to a descriptive name (e.g., **SM8**).
- **Enable** – Set to **Yes**.
- **Proxy Server Transport** – Set to **TLS**.
- Select **Proxy Servers**, and enter:
  - **Proxy Server Address** = **10.64.91.81** (the IP address of the Session Manager signaling interface defined in **Section 6.5.1**).
  - **Port** = **5061**
  - **Priority** = **0** (default)
  - **Weight** = **0** (default)
- **Listener Port** – Set to **5061**.
- **SIP Domain** – Set to **avayalab.com** (**Section** Error! Reference source not found.).
- **Consultative Transfer** – Select **INVITE with REPLACES**.
- **SIP Reject Response Code** – Select **ASM (503)**.
- **Maximum Simultaneous Calls** – Set to a number in accordance with licensed capacity. In the reference configuration a value of **10** was used.
- Select **All Calls can be either inbound or outbound**.
- **SRTP Enable** = **Yes**
- **Encryption Algorithm** = **AES_CM_128**
- **Authentication Algorithm** = **HMAC_SHA1_80**
- **RTCP Encryption Enabled** = **No**
- **RTP Authentication Enabled** = **Yes**
- Use default values for all other fields.
- Click **Save**.



7.4. Speech Servers

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

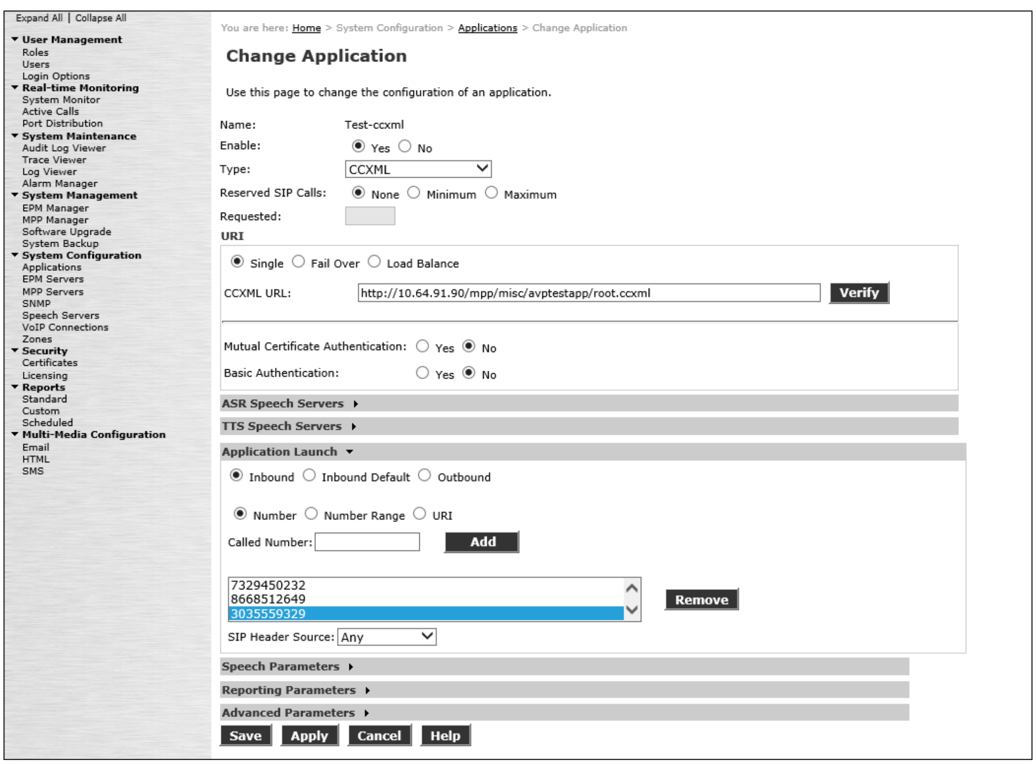


7.5. Application References

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

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

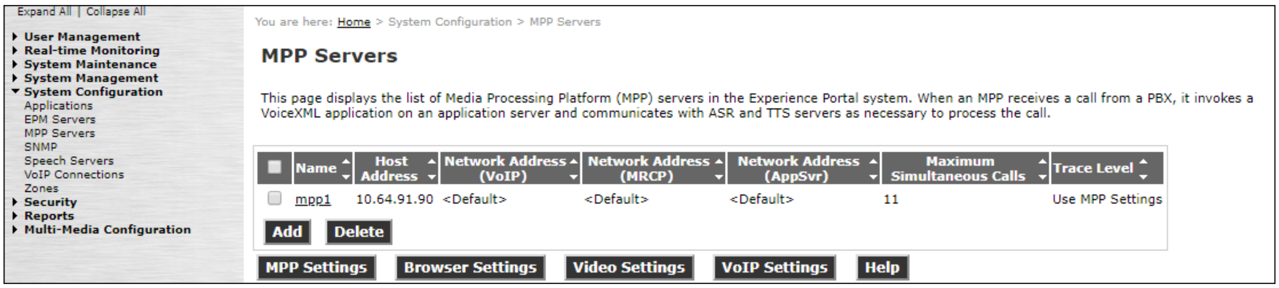
- **Name** – Set to a descriptive name (e.g., **Test-ccxml**).
- **Enable** – Set to **Yes**. This field determines which application(s) will be executed based on their defined criteria.
- **Type** – Select **VoiceXML**, **CCXML**, or **CCXML/VoiceXML** according to the application type.
- **VoiceXML** and/or **CCXML URL** – Enter the necessary URL(s) to access the VXML and/or CCXML application(s) on the application server. In the sample screen below, the Experience Portal test application on a single server is referenced.
- **ASR and TTS Speech Servers** – Select the appropriate ASR and/or TTS servers as necessary.
- **Application Launch** – Set to **Inbound**.
- **Called Number** – Enter the number to match against an inbound SIP INVITE message, and click **Add**. In the sample configuration illustrated in these Application Notes, the dialed IPFR-EP DID number 303-555-9329 was used. Repeat to define additional called party numbers as needed. Inbound AT&T IPFR-EP calls with these called party numbers will be handled by the application defined in this section.



7.6. MPP Servers and VoIP Settings

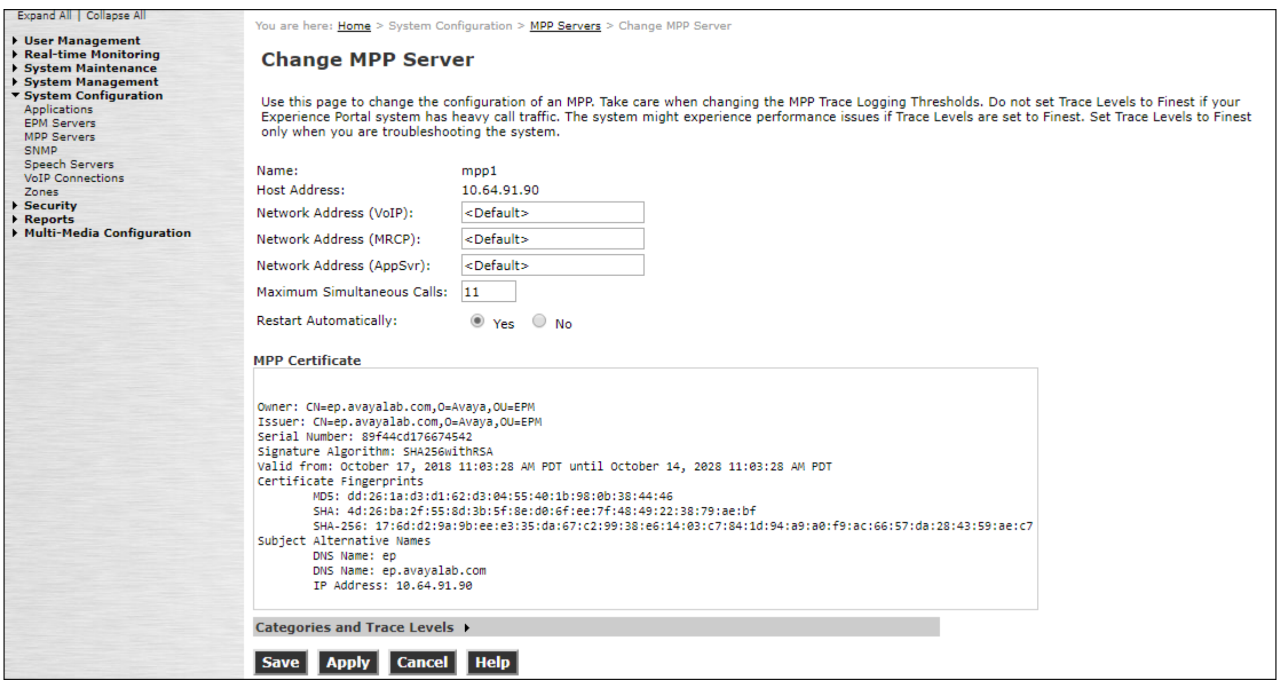
This section illustrates the procedure for viewing or changing the MPP Settings. In the sample configuration, the MPP Server is co-resident on a single server with the Experience Portal Management server (EPM).

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



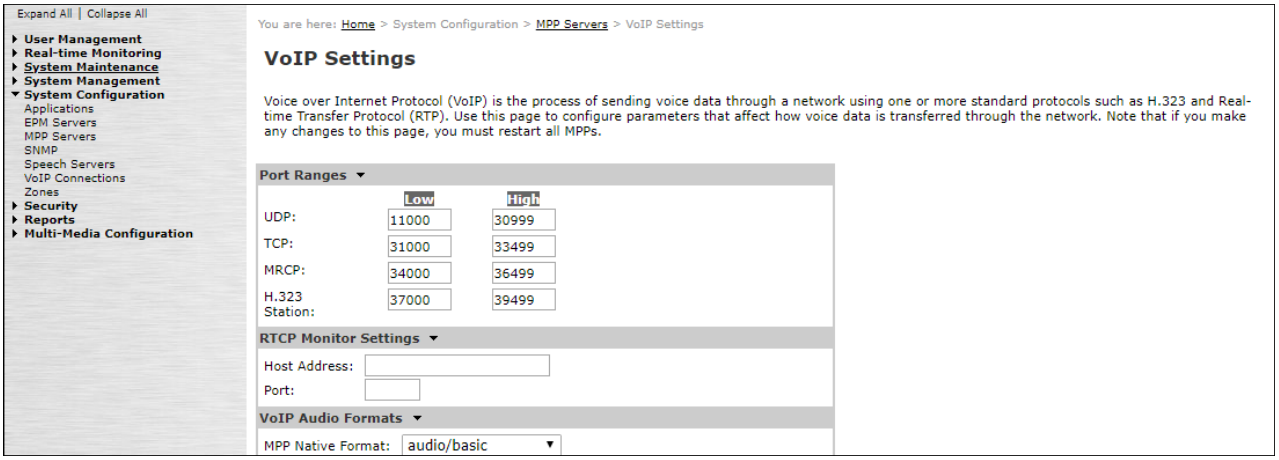
Step 2 - Enter any descriptive name in the **Name** field (e.g., **mpp1**) and the IP address of the MPP server in the **Host Address** field and click **Continue** (not shown).

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



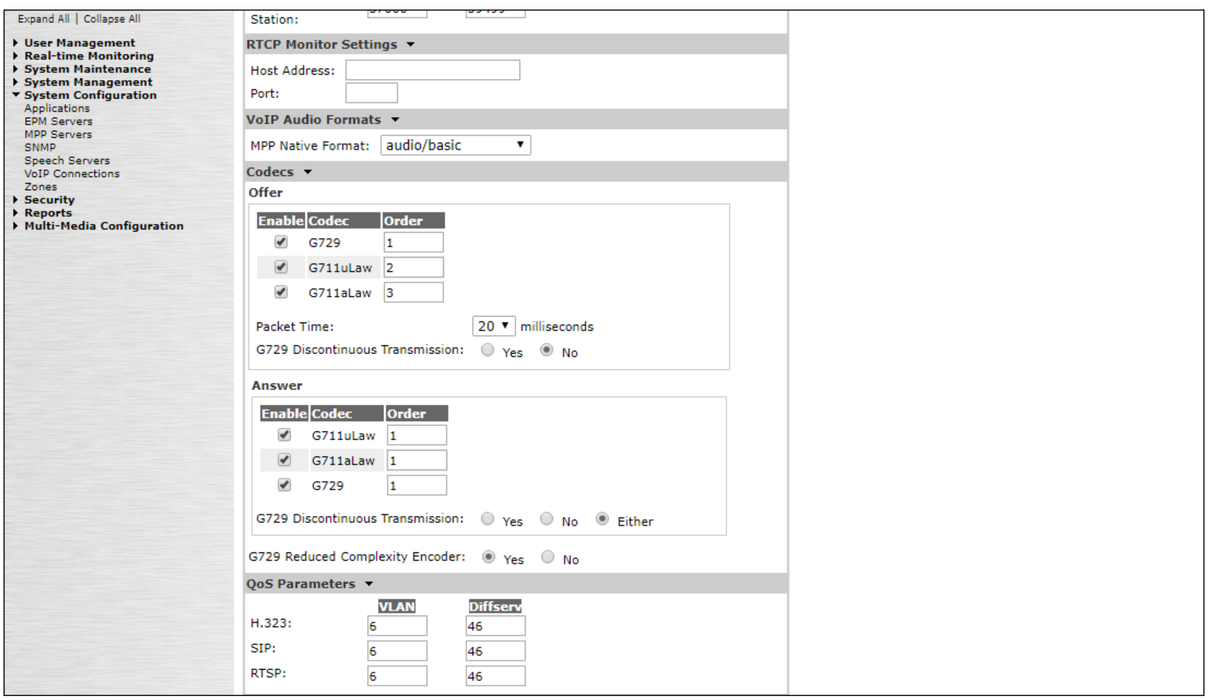
Step 4 - Click **VoIP Settings** tab on the screen displayed in **Step 1**, and the following screen is displayed.

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



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

Step 5 - Click on **Save**.



7.7. Configuring RFC2833 Event Value Offered by Experience Portal

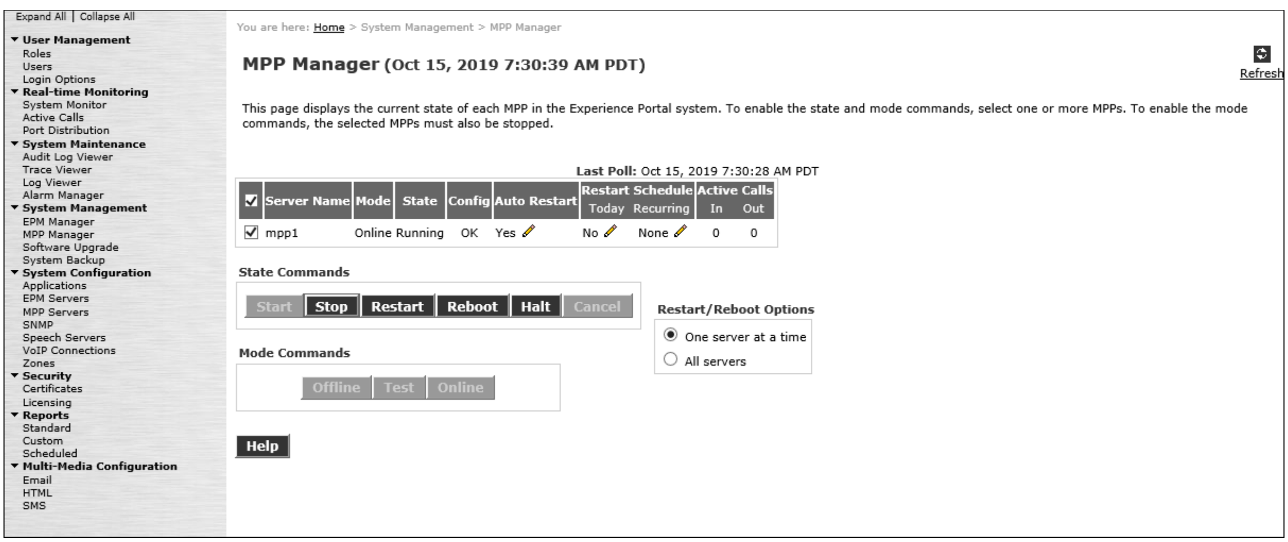
For incoming calls from AT&T IPFR-EP services to Experience Portal, AT&T specifies the value 100 for the RFC2833 telephone-events that signal DTMF digits entered by the user. When Experience Portal answers, the SDP from Experience Portal matches this offered value.

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

- Access Experience Portal via the command line interface.
- Navigate to the following directory: /opt/Avaya/ ExperiencePortal /MPP/config
- Edit the file mppconfig.xml.
- Search for the parameter “mpp.sip.rfc2833.payload”. If there is no such parameter specified, add a line such as the following to the file, where the value 100 is the value to be used for the RFC2833 events. If the parameter is already specified in the file, simply edit the value assigned to the parameter.
  ```
  <parameter name="mpp.sip.rfc2833.payload">100</parameter>
  ```
- In the verification of these Application Notes, the line was added directly above the line where the “sip.session.expires” parameter is configured.

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

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



8. Configure Avaya Session Border Controller for Enterprise

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

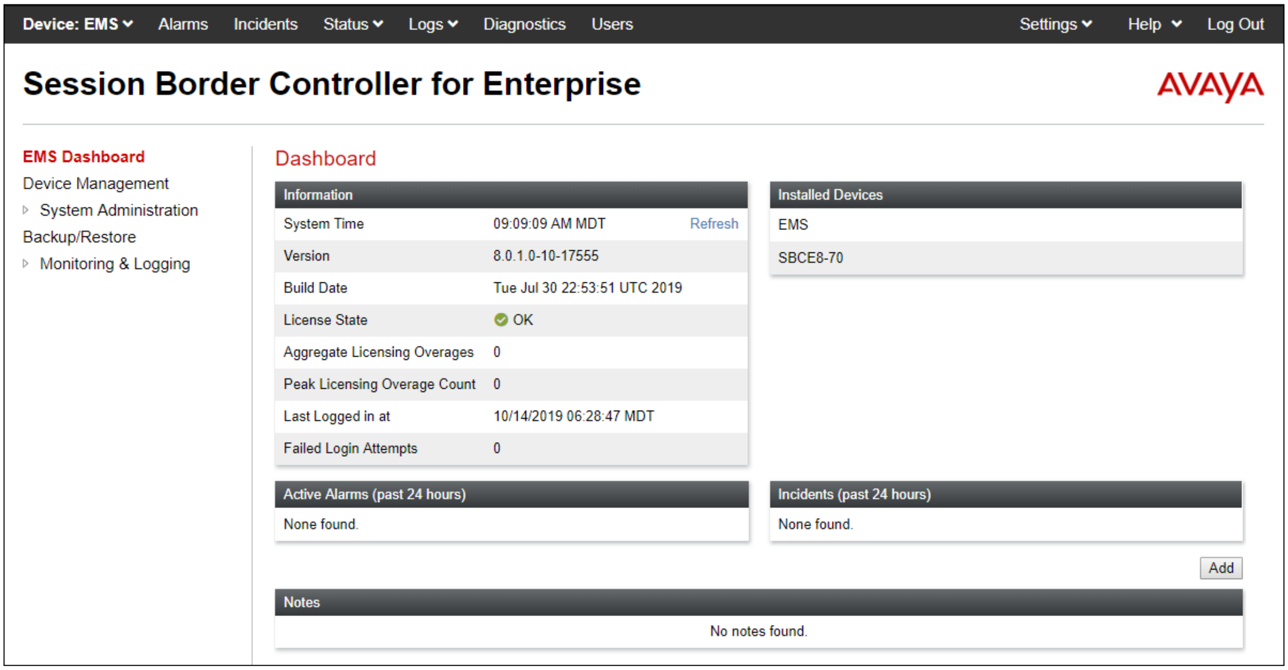
Note: The installation and initial provisioning of the Avaya SBCE is beyond the scope of this document. Refer to **[10]** and **[11]** in the References section for additional information.

Note: The Avaya SBCE supports a Remote Worker configuration whereby Communication Manager SIP endpoints residing on the public side of the Avaya SBCE, can securely register/operate as a "local" Communication Manager station in the private CPE. While Remote Worker functionality was tested in the reference configuration, Remote Worker provisioning is beyond the scope of this document.

Use a WEB browser to access the Element Management Server (EMS) web interface, and enter https://*ipaddress*/sbc in the address field of the web browser, where *ipaddress* is the management LAN IP address of the Avaya SBCE. Log in using the appropriate credentials.

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

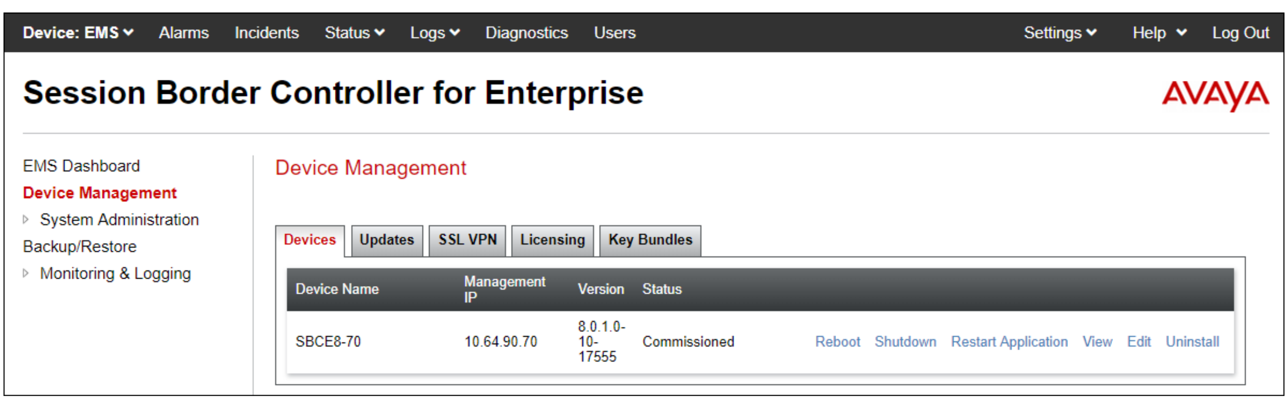
Note – The provisioning described in the following sections use the menu options listed in the left-hand column shown below.



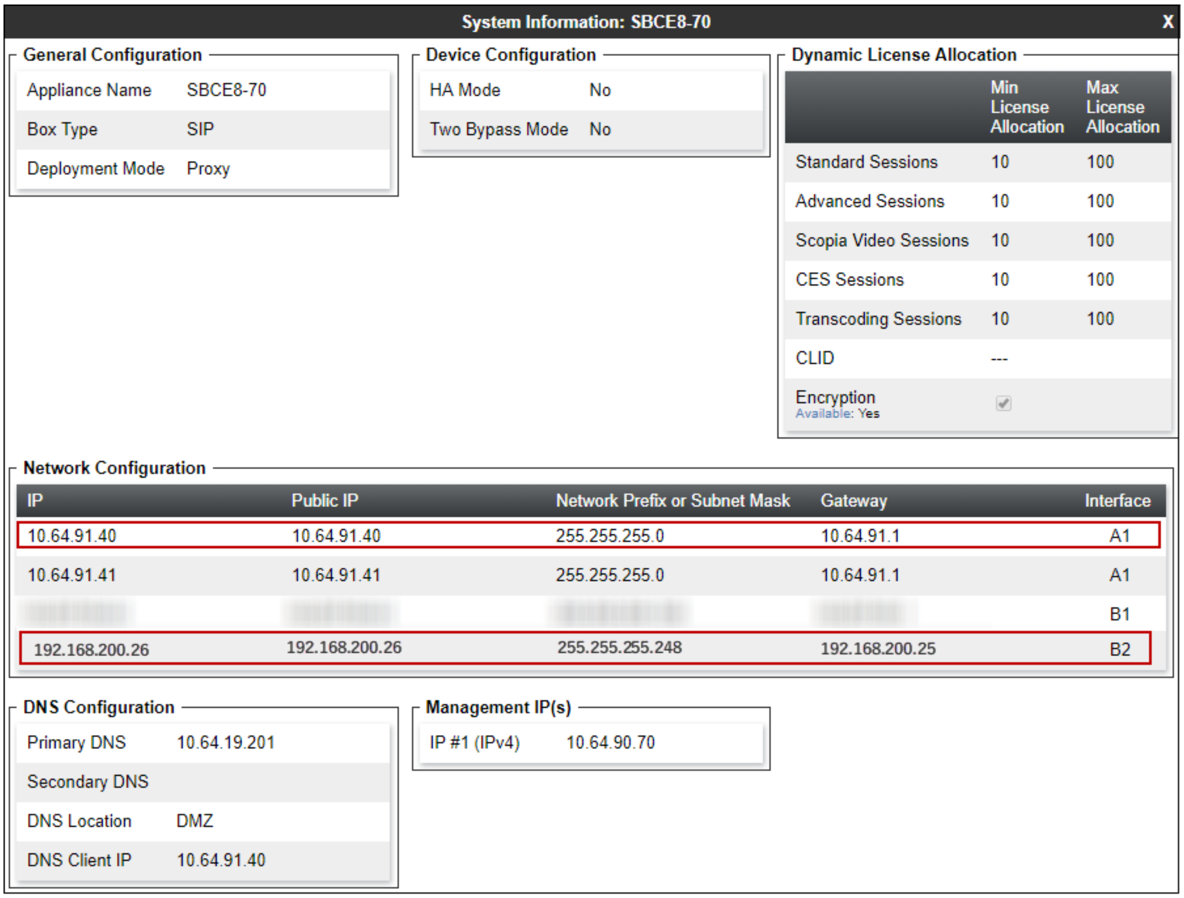
8.1. Device Management – Status

Step 1 - Select **Device Management** on the left-hand menu. A list of installed devices is shown on the **Devices** tab on the right pane. In the case of the sample configuration, a single device named **SBCE8-70** is shown. Verify that the **Status** column shows **Commissioned**. If not, contact your Avaya representative.

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



Step 2 - Click on **View** to display the **System Information** screen. The screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**. In the shared test environment, the highlighted **A1** and **B2** IP addresses are the ones relevant to the configuration of the SIP trunk to AT&T.



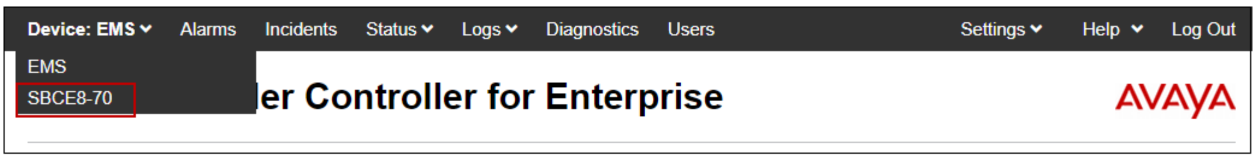
8.2. TLS Management

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

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

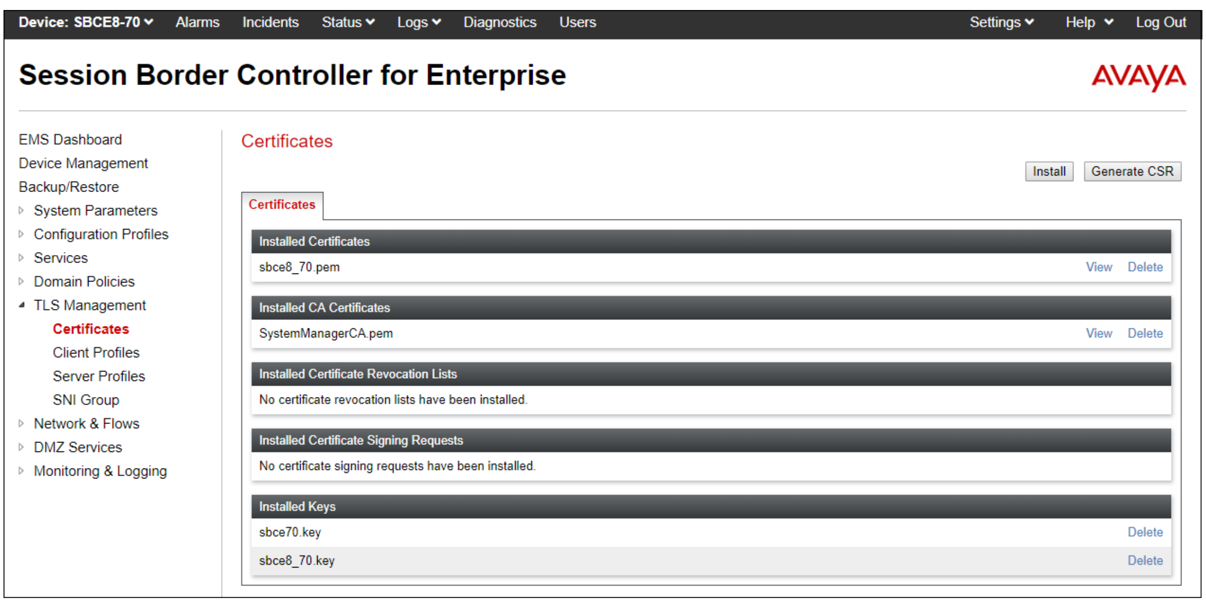
8.2.1. Verify TLS Certificates – Avaya Session Border Controller for Enterprise

To access the SBCE configuration menus, select the SBCE device from the top navigation menu.



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.

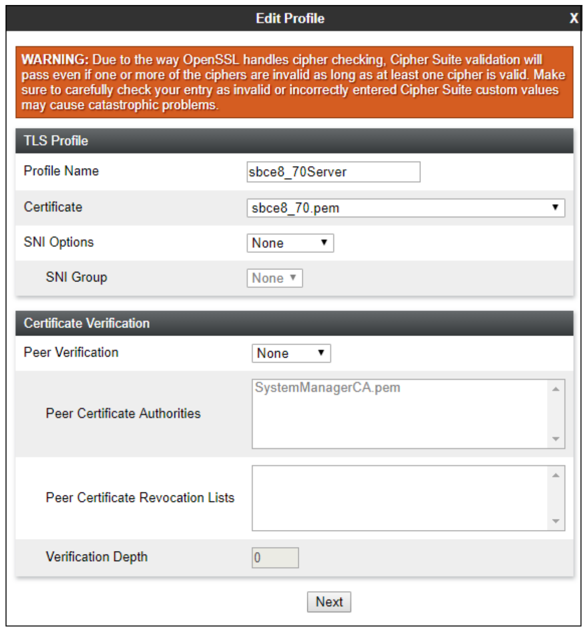


8.2.2. Server Profiles

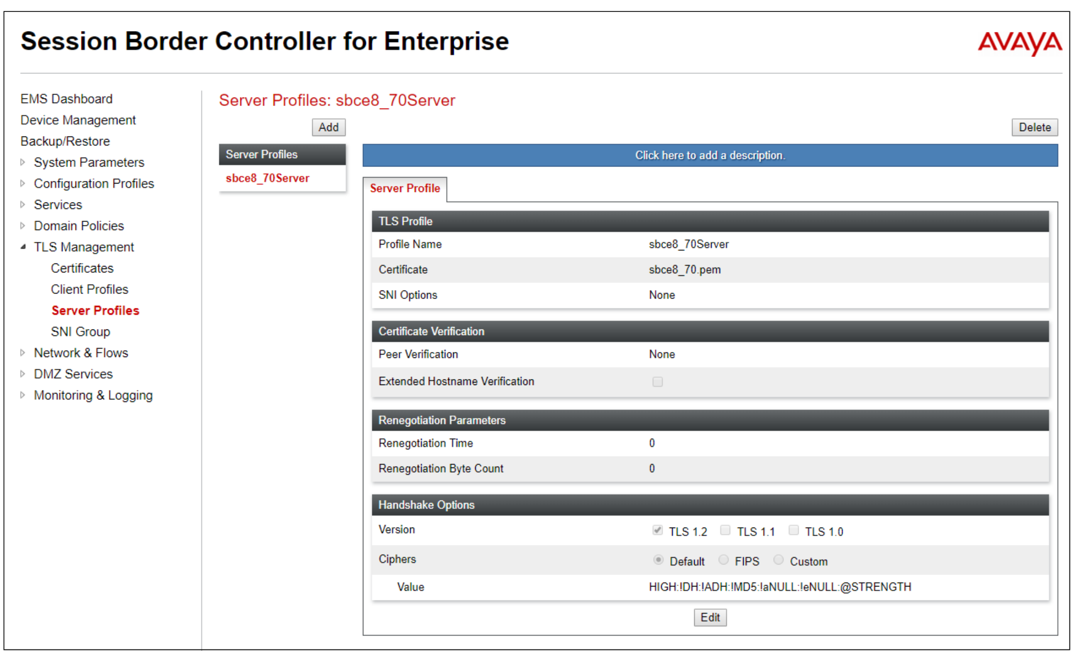
Step 1 - Select **TLS Management → Server Profiles** and click on **Add**. Enter the following:

- **Profile Name:** enter a descriptive name. (e.g., **sbce8_70Server**).
- **Certificate:** select the identity certificate, e.g., **sbce8_70.pem**, from pull down menu.
- **Peer Verification** = **None**.
- Click **Next**.

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



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

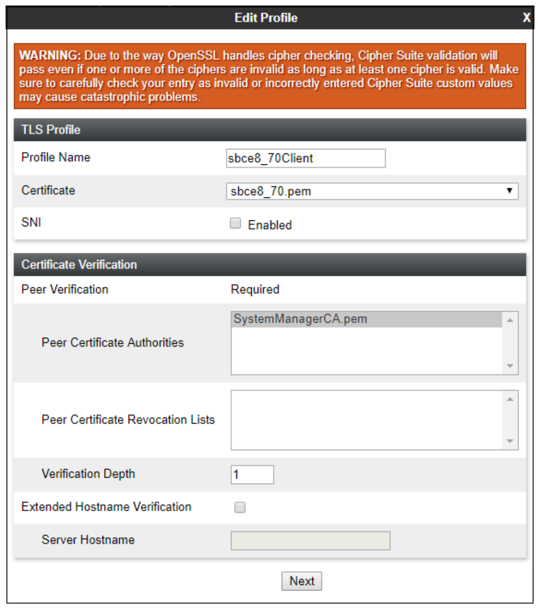


8.2.3. Client Profiles

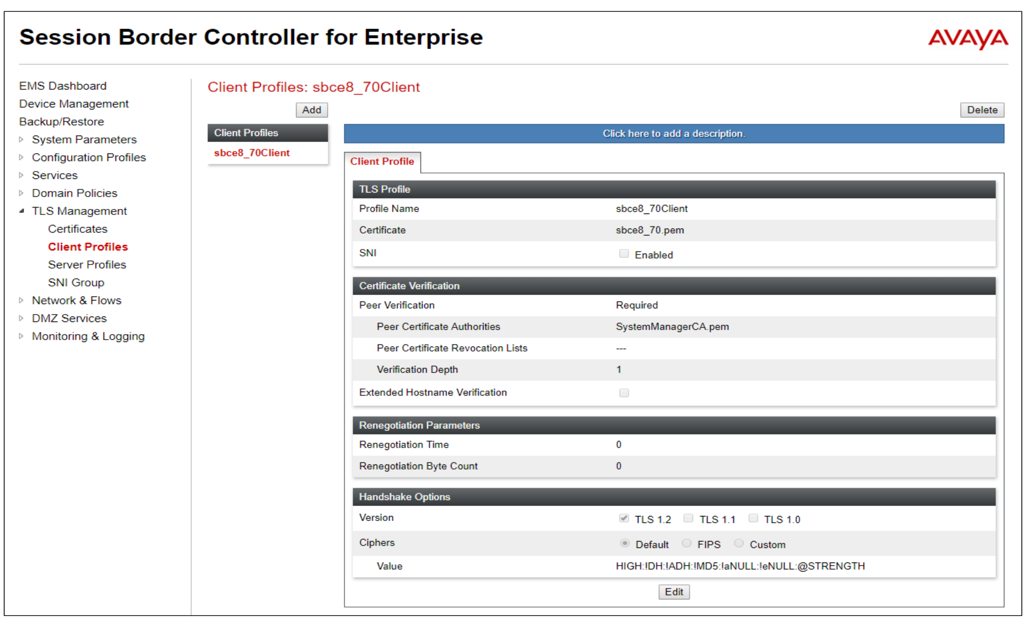
Step 1 - Select **TLS Management → Server Profiles**, and click on **Add**. Enter the following:

- **Profile Name:** enter a descriptive name (e.g., **sbce8_70Client**)
- **Certificate:** select the identity certificate, e.g., **sbce8_70.pem**, from pull down menu.
- **Peer Verification** = **Required**.
- **Peer Certificate Authorities:** select the CA certificate used to verify the certificate received from Session Manager, e.g., **SystemManagerCA.pem**.
- Enter 1 under **Verification Depth**. Click **Next**.

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



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



8.3. Network Management

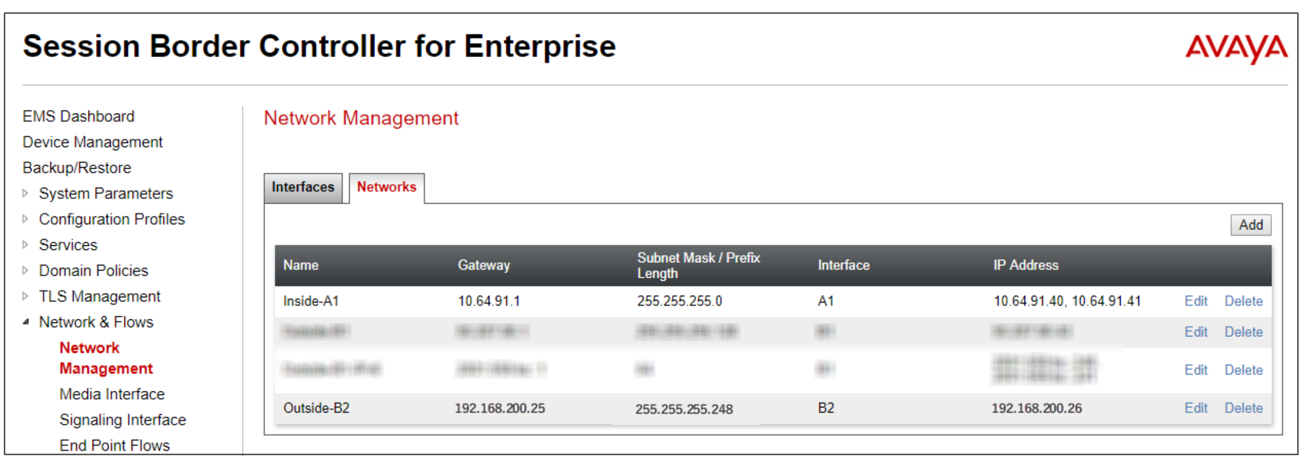
The Network Management screen is where the network interface settings are configured and enabled. During the installation process of Avaya SBCE, certain network-specific information is defined such as device IP address(es), public IP address(es), netmask, gateway, etc., to interface the device to the network. It is this information that populates the various Network Management tab displays, which can be edited as needed to optimize device performance and network efficiency. Navigate to **Networks & Flows → Network Management**. On the **Networks** tab, verify the IP addresses assigned to the interfaces. The following screen shows the enterprise interface is assigned to **A1** and the interface towards AT&T is assigned to **B2**.

Step 1 - Select **Networks & Flows → 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 B2 (public) interfaces are used. To enable an interface, click the corresponding **Disabled** link under the Status column to change it to **Enabled**.

Step 3 - Select the **Networks** tab to display the IP provisioning for the A1 and B2 interfaces. The following Avaya SBCE IP addresses and associated interfaces were used in the sample configuration:

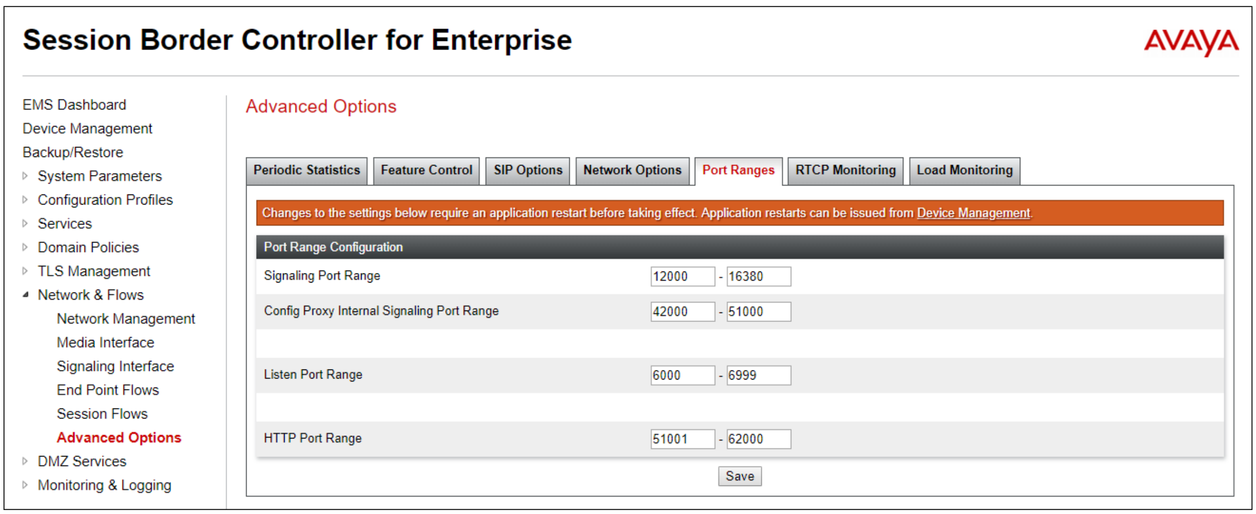
- **B2: 192.168.200.26** – IP address configured for the AT&T IPFR-EF service. This address is known to AT&T. See **Section 3**.
- **A1**: **10.64.91.40** – IP address configured for AT&T IPFR-EF service to Session Manager.



8.4. Advanced Options

AT&T required the UDP port ranges of the media to be configured in the **16384** – **32767** range. However, by default ranges 12000 to 21000 and 22000 to 31000 are already allocated by the Avaya SBCE for internal use. The following steps reallocate the port ranges used by the Avaya SBCE, so the range required by AT&T can be defined on the Avaya SBCE Media Interfaces (**Section 8.5**).

Step 1 - Select **Network & Flows → Advanced Options** from the menu on the left-hand side.
Step 2 - Select the **Port Ranges** tab.
Step 3 - In the **Signaling Port Range** row, change the range to **12000** – **16380**
Step 4 - In the **Config Proxy Internal Signaling Port Range** row, change the range to **42000** – **51000**.
Step 5 – In the **Listen Port Range** row, change the range to **6000** – **6999**.
Step 6 – In the **HTTP Port Range** row, change the range to **51001** – **62000**.
Step 7 - Select **Save**. Note that changes to these values require an application restart (see **Section** Error! Reference source not found.).



8.5. Media Interfaces

Media Interfaces are created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. As mentioned in **Section 8.4**, the AT&T IPFR-EF service specifies that customers use RTP ports in the range of **16384 – 32767**. Both inside and outside ports have been changed to this range, though only the outside port range is required by the AT&T IPFR-EF service.

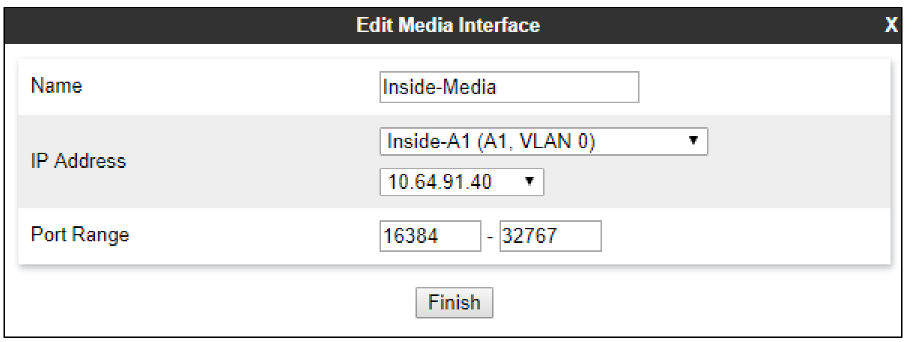
Some ports in the range required by AT&T were already allocated by the Avaya SBCE for internal use, by default. **Section 8.4** shows the steps required to reallocate the port ranges used by the Avaya SBCE, so the range required by AT&T could be accommodated.

Step 1 - Select **Network & Flows → Media Interface** on the left-hand side menu,

Step 2 - Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:

- **Name**: **Inside-Media**
- **IP Address**: Select **Inside-A1 (A1, VLAN0)** and **10.64.91.40**
- **Port Range**: **16384 – 32767**

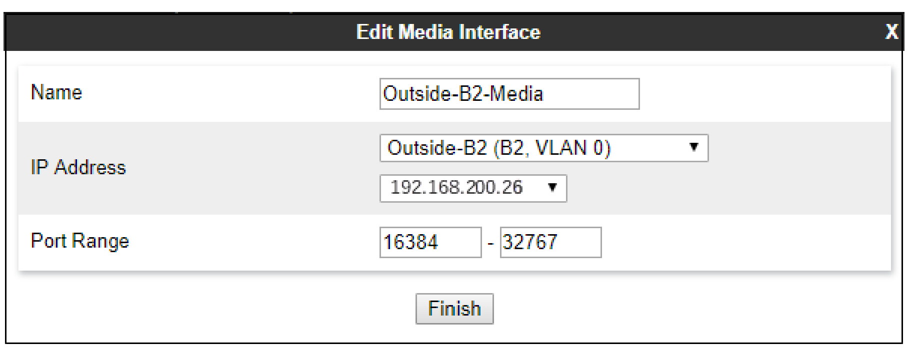
Step 3 - Click **Finish**.



Step 4 - Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:

- **Name**: **Outside-B2-Media**
- **IP Address**: Select **Outside-B2 (B2, VLAN0)** and **192.168.200.26**
- **Port Range**: **16384 – 32767**

Step 5 - Click **Finish**.



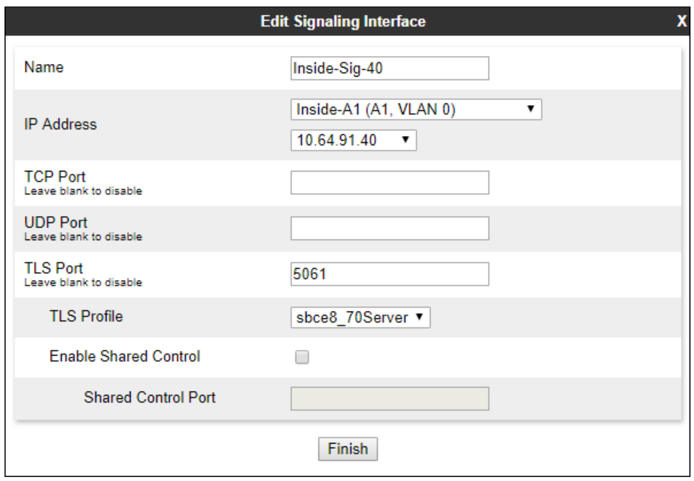
8.6. Signaling Interfaces

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

Step 1 - Select **Network & Flows → Signaling Interface** from the menu on the left-hand side
Step 2 - Select **Add** (not shown) and enter the following:

- **Name**: **Inside-Sig-40**
- **IP Address**: Select **Inside-A1 (A1, VLAN0)** and **10.64.91.40**
- **TLS Port**: **5061**
- **TLS Profile**: Select the TLS server profile created in **Section 8.2.2**

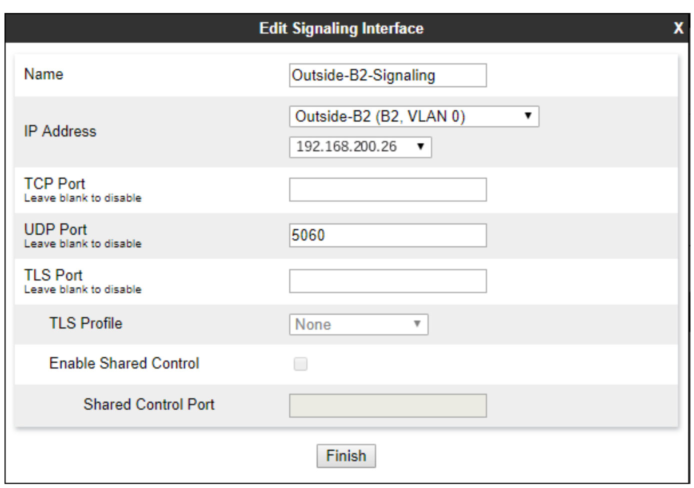
Step 3 - Click **Finish**.



Step 4 - Select **Add** again, and enter the following:

- **Name**: **Outside-B2-Signaling**
- **IP Address**: Select **Outside-B2 (B2, VLAN0)** and **192.168.200.26**
- **UDP Port**: **5060**

Step 5 - Click **Finish.**



8.7. Server Interworking Profiles

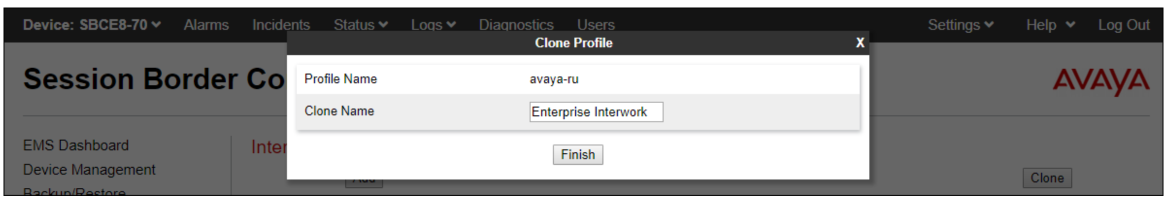
The Server Internetworking profiles include parameters to make the Avaya SBCE function in an enterprise VoIP network using different implementations of the SIP protocol. There are default profiles available that may be used as is, or modified, or new profiles can be configured as described below.

In the sample configuration, separate Server Interworking Profiles were created for the enterprise and AT&T IPFR-EF service.

8.7.1. Server Interworking Profile – Enterprise

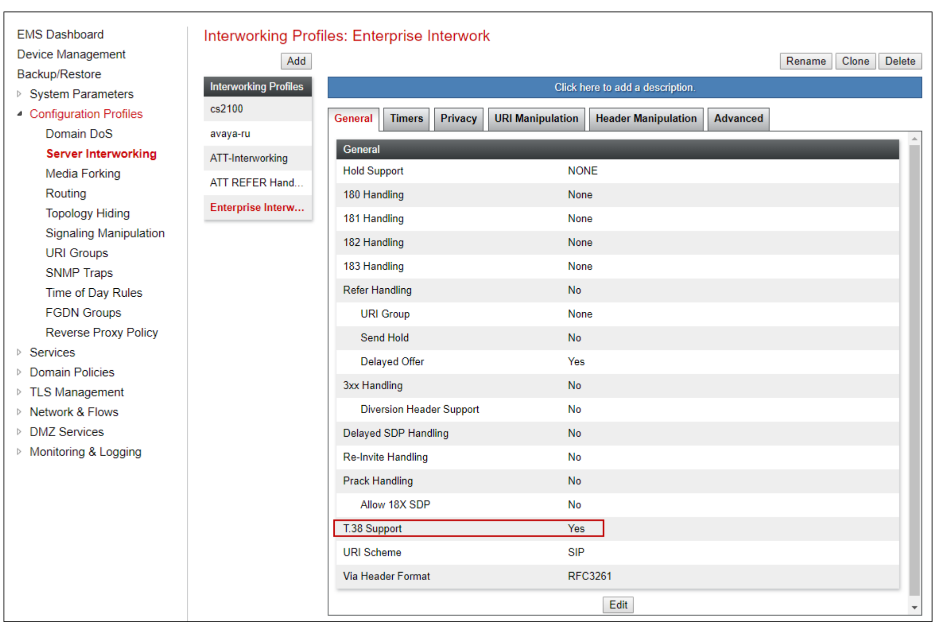
In the sample configuration, the enterprise Server Interworking profile was cloned from the default **avaya-ru** profile and then modified.

Step 1 - Select **Configuration 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** to continue.



Step 4 - The new Enterprise Interwork profile will be listed. Select it, scroll to the bottom of the Profile screen, and click on **Edit**.
Step 5 - The **General** screen will open.

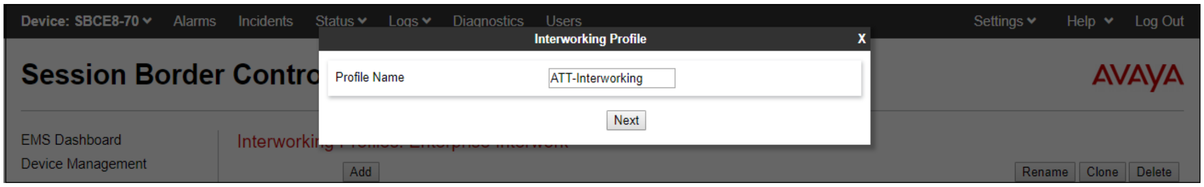
- Check **T38 Support**.
- All other options can be left with default values. Click **Finish** (not shown).



8.7.2. Server Interworking – AT&T

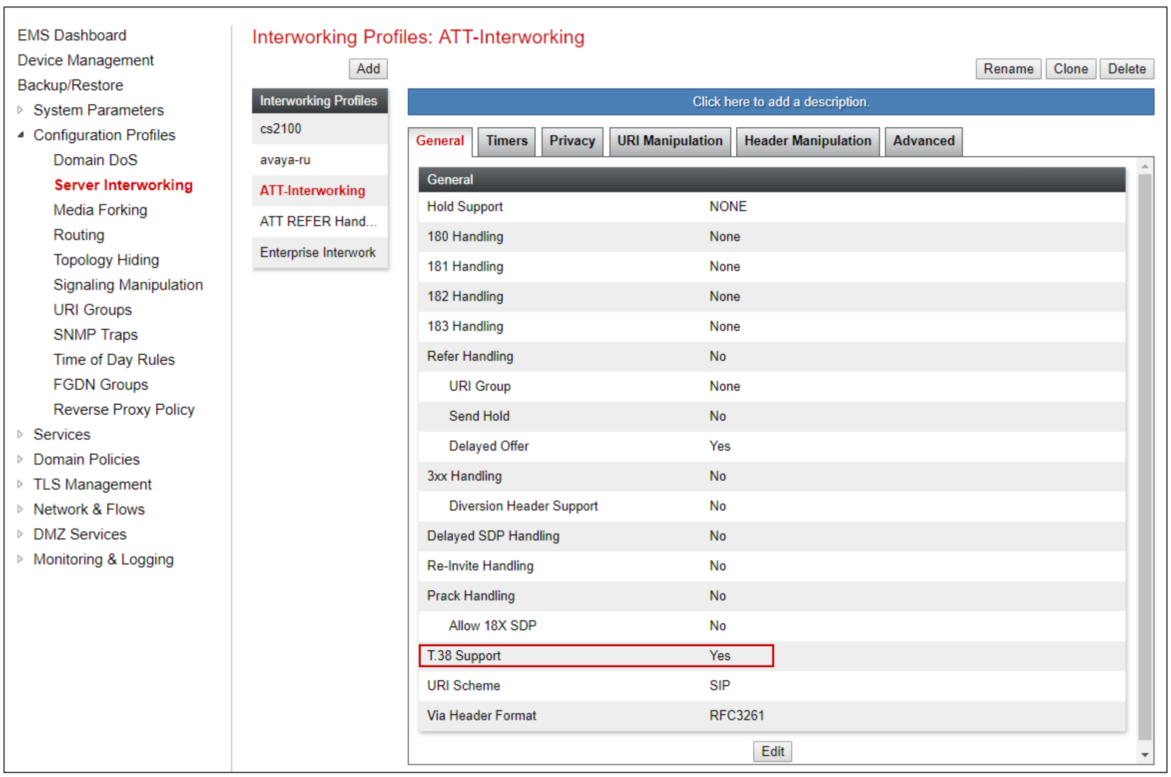
Repeat the steps shown in **Section** Error! Reference source not found. to add an Interworking Profile for the connection to AT&T via the public network, with the following changes:

Step 1 - Select **Add Profile** and enter a profile name: (e.g., **ATT-Interworking**) and click **Next**.

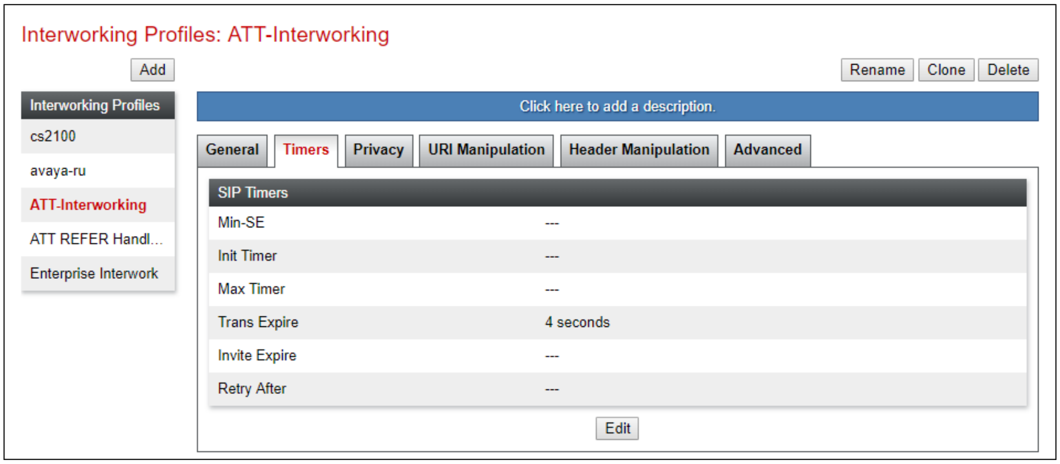


Step 2 - The **General** screen will open:

- Default values are used with the exception of **T.38 Support** set to **Yes**



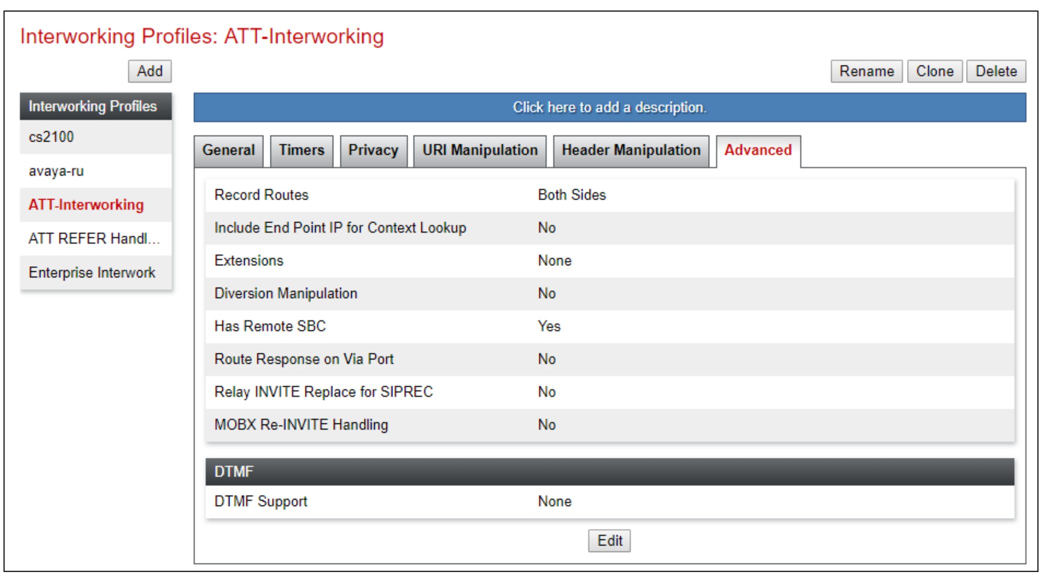
Step 3 – On the **Timers** tab, the **Trans Expire** timer is set to the allotted time the Avaya SBCE will try the first primary server before trying the secondary server, if one exists. The screen shows the value used during the compliance testing. See **Sections 8.9.2** and **8.10.2** for multiple AT&T border elements configuration.



Step 4 - Click **Next** to accept default parameters for the **Privacy**, **URI Manipulation**, and **Header Manipulation** tabs (not shown).

Step 5 – On the **Advanced/DTMF** tab:

- In the **Record Routes** field, check **Both Sides**.
- All other options can be left as default. Click **Finish** (not shown).



8.8. Signaling Manipulation

Signaling Manipulations are SigMa scripts the Avaya SBCE can use to manipulate SIP headers/messages. In the reference configuration, a 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 Server Interworking Profiles (**Section 8.7**) or Signaling Rules (**Section 8.14**) does not meet the desired result. Refer to References **[10]** for information on the Avaya SBCE scripting language.

The script can be created externally as a regular text file and imported in the Signaling Manipulation screen, or they can be written directly in the page using the embedded Sigma Editor.

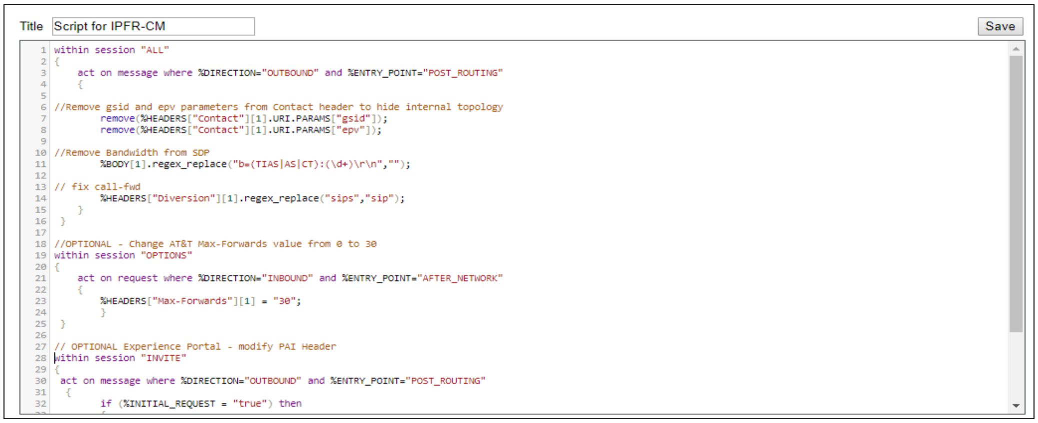
A Sigma script was created during the compliance test to correct the following interoperability issues:

- Remove the gsid and epv parameters from outbound Contact headers. See **Section** Error! Reference source not found.
- Remove the Bandwidth headers sent by some Avaya SIP endpoints. See **Section** Error! Reference source not found.
- Change the Diversion header scheme from SIPS to SIP towards AT&T. See **Section** Error! Reference source not found.
- Change the value of the Max-Forwards header on the SIP OPTIONS messages sent by AT&T from “0” to “30”, to be able to reach Communication Manager. See **Section** Error! Reference source not found.
- Modify the P-Asserted-Identity header of outbound INVITEs from Experience Portal to the PSTN, with a DID number known to AT&T. See **Section** Error! Reference source not found.

The details of the complete script appear on **Section 14**.

Step 1 - Select **Configuration Profiles → Signaling Manipulation** from the menu on the left.
Step 2 - Click **Add Script** (not shown) and the script editor window will open.

- Enter a name for the script in the **Title** box (e.g., **Script for IPFR-CM**).
- Copy and paste the script from **Section 14** in this document.



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 SIP Server profile in **Section 8.9.2**.

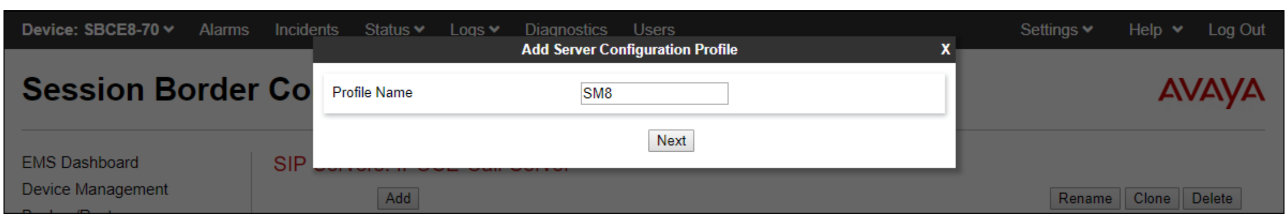
8.9. SIP Server Profiles

The **SIP Server Profile** contains parameters to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics, and trusted domains.

8.9.1. SIP Server Profile – Session Manager

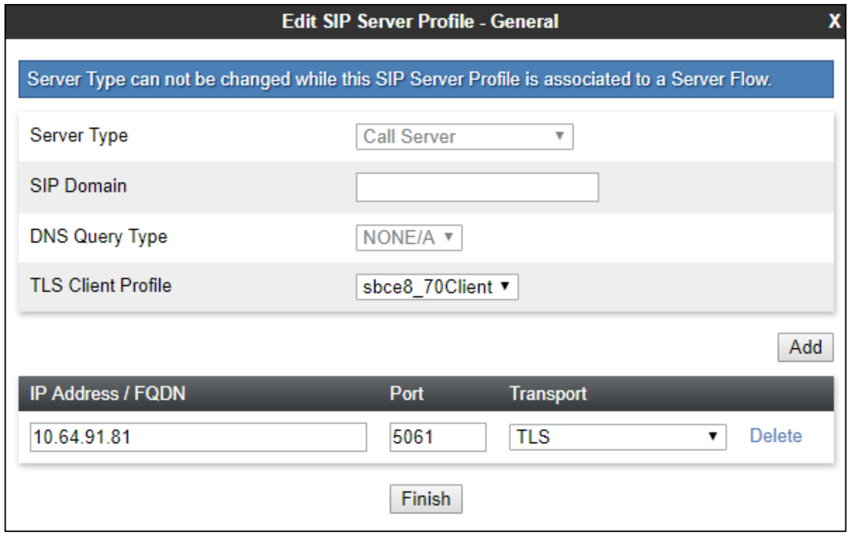
This section defines the SIP Server Profile for the Avaya SBCE connection to Session Manager.

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



Step 3 - The **Edit SIP Server Profile** window will open.

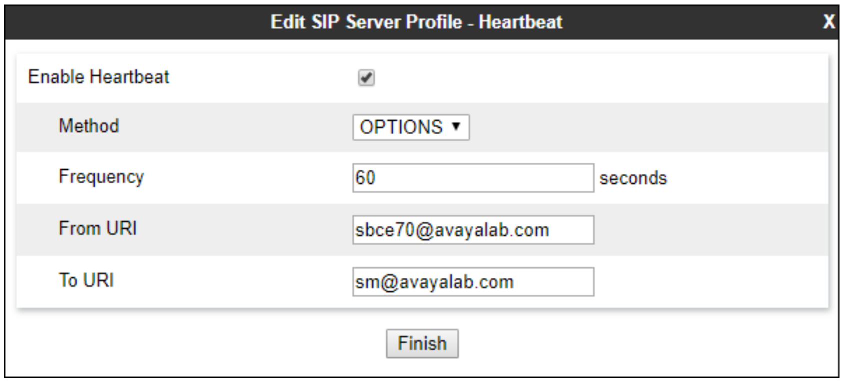
- Select **Server Type**: **Call Server**
- **SIP Domain**: Leave blank (default)
- **DNS Query Type**: Select **NONE/A** (default)
- **TLS Client Profile**: Select the profile create in **Section 8.2.3** (e.g., **sbce8_70Client**)
- **IP Address/FQDN**: **10.64.91.81** (Session Manager Security Module IP address)
- Select **Port**: **5061**, **Transport**: **TLS**.
- If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish** and proceed to the next tab.


Edit SIP Server Profile - General
X
Server Type can not be changed while this SIP Server Profile is associated to a Server Flow.
Server Type
Call Server
SIP Domain
DNS Query Type
NONE/A
TLS Client Profile
sbce8_70Client
Add
IP Address / FQDN
Port
Transport
10.64.91.81
5061
TLS
Delete
Finish

Step 4 – Default values can be used on the **Authentication** tab.

Step 5 – On the **Heartbeat** tab, check the **Enable Heartbeat** box to have the Avaya SBCE source "heartbeats" toward Session Manager. This configuration is optional.

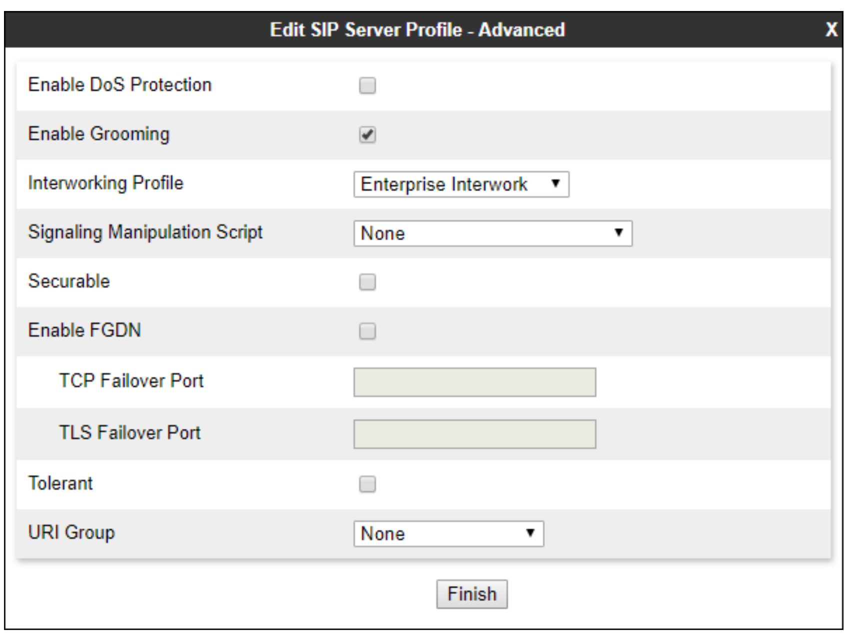
- Select **OPTIONS** from the **Method** drop-down menu.
- Select the desired frequency that the SBCE will source OPTIONS toward Session Manager.
- Make logical entries in the **From URI** and **To URI** fields that will be used in the OPTIONS headers.



Step 6 – Default values are used on the **Registration** and **Ping** tabs.

Step 7 – On the **Advanced** tab:

- Select the **Enterprise Interwork** (created in **Section 8.7.1**), for **Interworking Profile**.
- Since TLS transport is specified in **Step 3**, then the **Enable Grooming** option should be enabled.
- In the **Signaling Manipulation Script** field select **none**.
- Select **Finish**.



8.9.2. SIP Server Profile – AT&T

Note – The AT&T IPFR-EF service may provide a Primary and Secondary Border Element. This section describes the Avaya SBCE provisioning to support this redundant configuration.

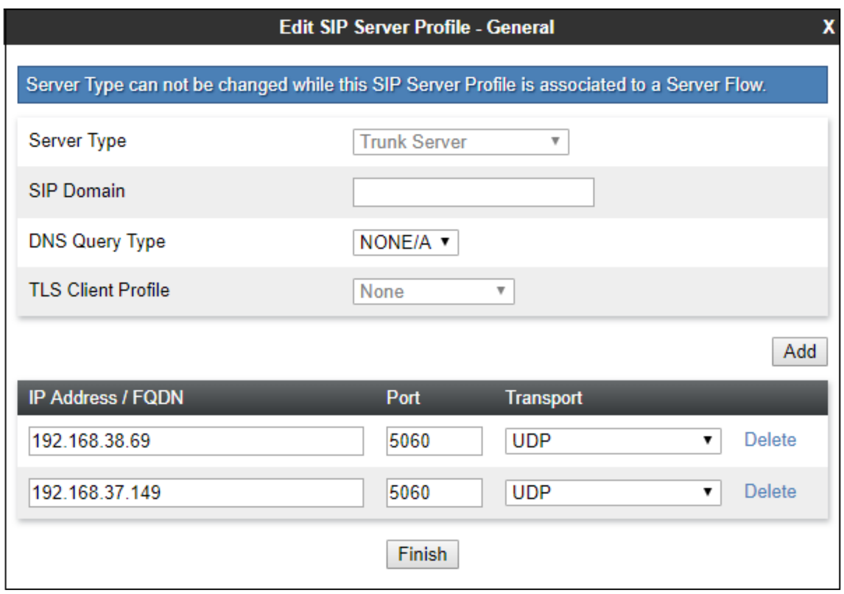
Repeat the steps in **Section 8.9.1**, with the following changes, to create a SIP Server Profile for the Avaya SBCE connection to AT&T.

Step 1 - Select **Add** and enter a Profile Name (e.g., **ATT-trk-svr**) and select **Next** (not shown).
Step 2 - On the **General** window, enter the following:

- **Server Type:** Select **Trunk Server**
- **IP Address/FQDN: 192.168.38.69** (AT&T Border Element IP address)
- **Transport**: Select **UDP**
- **Port: 5060**

Step 3 – For the additional AT&T Border Element IP addresses, click **Add** and enter the following:

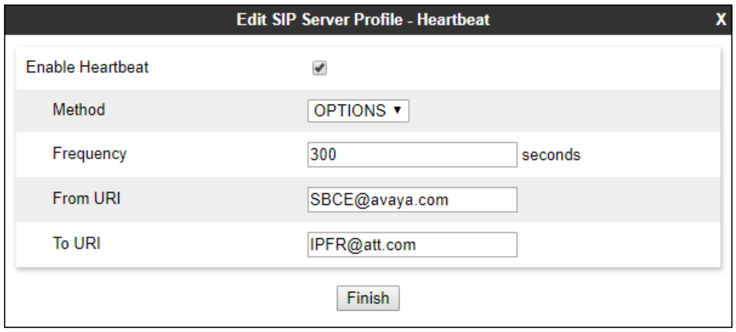
- **IP Address/FQDN: 192.168.37.149** (AT&T Border Element IP address)
- **Transport**: Select **UDP**
- **Port: 5060**
- If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish** and proceed to the next tab.



Step 4 – Default values can be used on the **Authentication** tab.

Step 5 – On the **Heartbeat** tab, check the **Enable Heartbeat** box to have the Avaya SBCE source "heartbeats" toward AT&T. This configuration is optional.

- Select **OPTIONS** from the **Method** drop-down menu.
- Select the desired frequency that the SBCE will source OPTIONS toward AT&T.
- Make logical entries in the **From URI** and **To URI** fields that will be used in the OPTIONS headers.

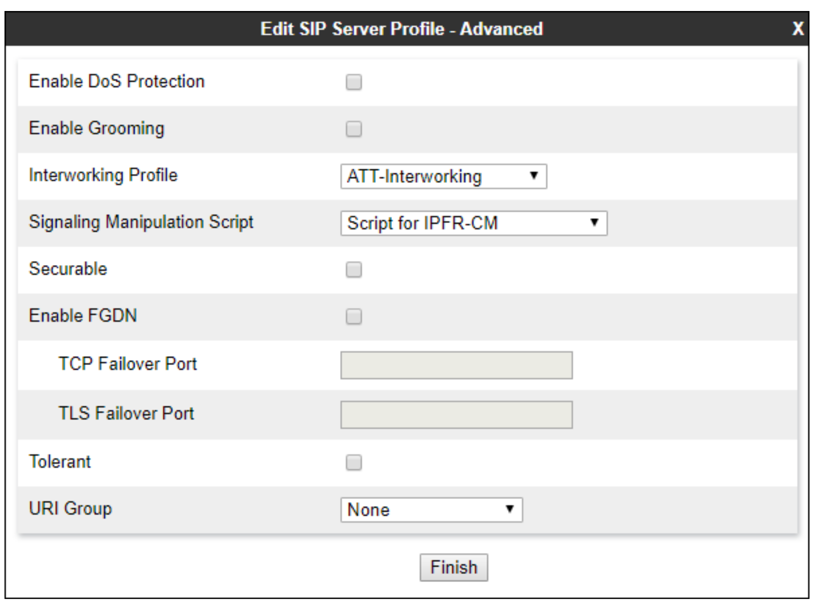


Note - Avaya SBCE will issue OPTIONS messages to the primary (192.168.38.69) and secondary (192.168.37.149) border elements. If the SBCE fails to get a response to the OPTIONS sent to 192.168.38.69, the SBCE will redirect outbound calls to 192.168.37.149.

Step 6 – Default values are used on the **Registration** and **Ping** tabs.

Step 7 – On the **Advanced** window, enter the following:

- **Enable Grooming** is not used for UDP connections and is left unchecked.
- Select **ATT-Interworking** (created in **Section 8.7.2**), for **Interworking Profile**.
- Select the **Script for IPFR-CM** (created in **Section 8.8**) for **Signaling Manipulation Script**.
- Select **Finish**.



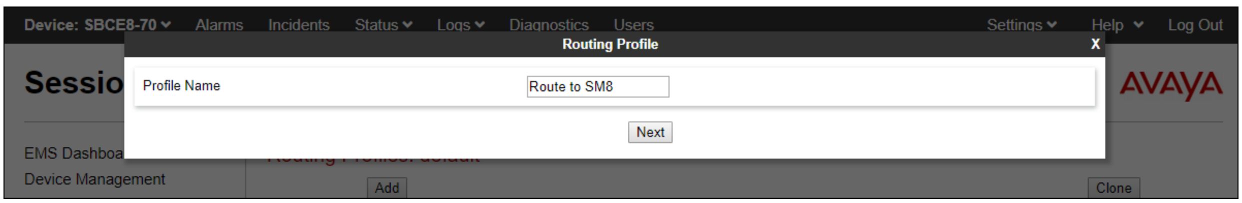
8.10. Routing Profiles

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and determine which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types. Separate Routing Profiles were created in the reference configuration for Session Manager and AT&T.

8.10.1. Routing Profile – Session Manager

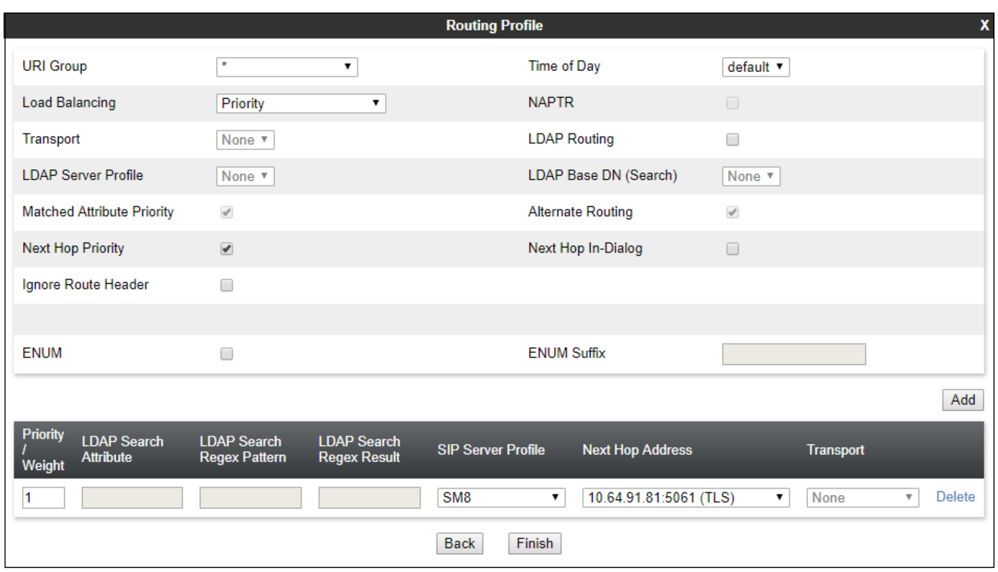
This provisioning defines the Routing Profile for the connection to Session Manager.

Step 1 - Select **Configuration Profiles → Routing** from the left-hand menu, and select **Add**.
Step 2 - Enter a **Profile Name**: (e.g., **Route to SM8**) and click **Next**.



Step 3 - The Routing Profile window will open. The parameters in the top portion of the profile are left at their default settings. Click the **Add** button.
Step 4 - The **Next-Hop Address** section will open at the bottom of the profile. Populate the following fields:

- **Priority/Weight** = **1**
- **SIP Server Profile** = **SM8** (from **Section 8.9.1**).
- **Next Hop Address:** Verify that the **10.64.91.81:5061 (TLS)** entry from the drop-down menu is selected (Session Manager IP address). Also note that the **Transport** field is grayed out. Click on **Finish**.



8.10.2. Routing Profile – AT&T

Repeat the steps in **Section 8.10.1**, with the following changes, to add a Routing Profile for the Avaya SBCE connection to AT&T.

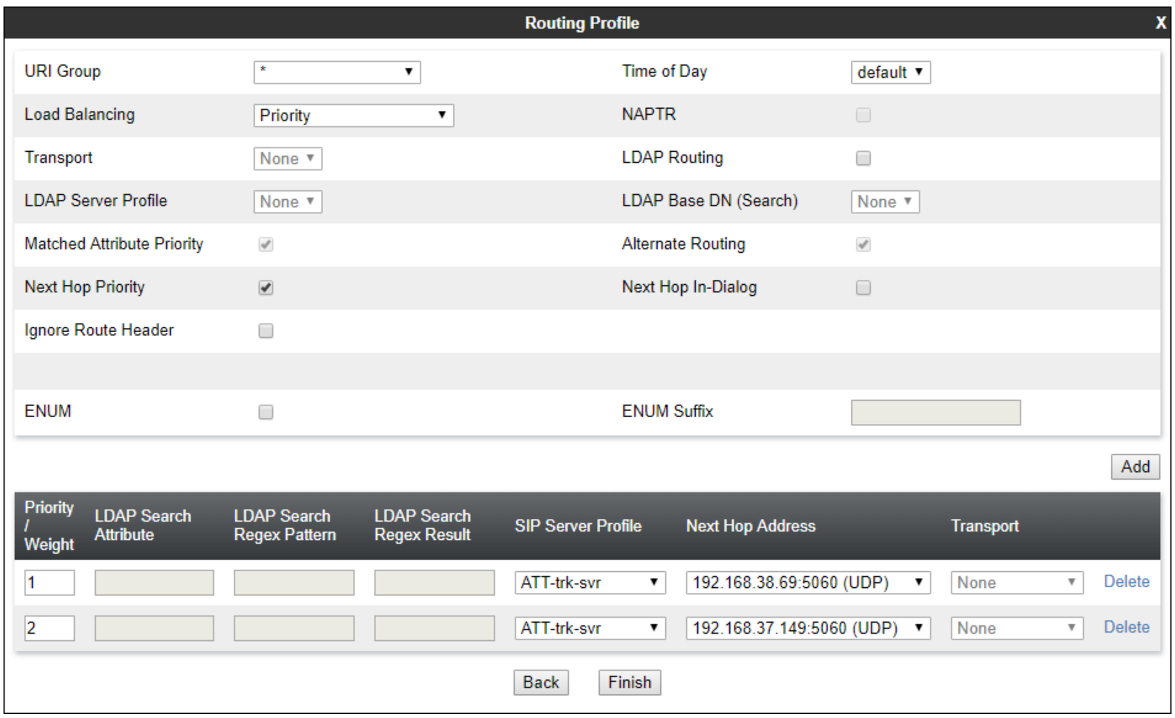
Step 1 - Enter a Profile Name: (e.g., **Route to ATT IPFR**).
Step 2 - On the **Next-Hop Address** window, for the first routing rule (AT&T Primary Border Element), populate the following fields:

- **Priority/Weight** = **1**
- **Server Configuration** = **ATT-trk-svr** (**from Section 8.9.2**).
- **Next Hop Address:** select **192.168.38.69:5060 (UDP)**.

Step 3 - For the second routing rule (AT&T Secondary Border Element) click **Add** and enter the following:

- **Priority/Weight** = **2**
- **Server Configuration** = **ATT-trk-svr** (**from Section 8.9.2**).
- **Next Hop Address:** select **192.168.37.149:5060 (UDP)**.

Step 4 - Click **Finish**.



Note – If desired, the **Load Balancing** parameter may be used to modify how the two defined AT&T Border Elements are accessed. **Priority** was used in the Reference Configuration.

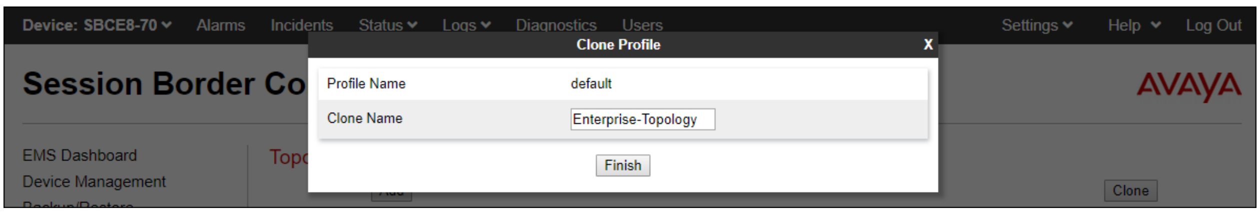
8.11. Topology Hiding Profiles

The Topology Hiding profile manages how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks.

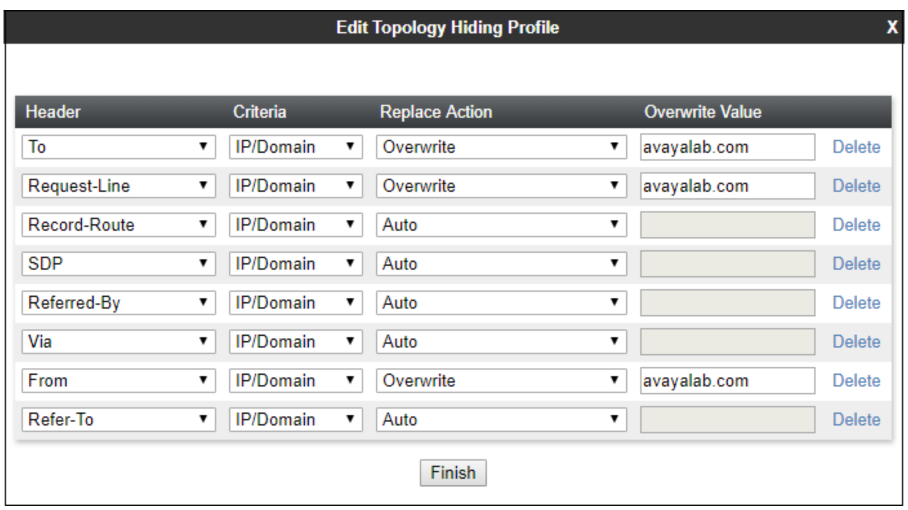
8.11.1. Topology Hiding – Enterprise

In the sample configuration, the enterprise Topology Hiding Profile was cloned from the **default** profile and then modified.

Step 1 - Select **Configuration Profiles → Topology Hiding** from the left-hand menu.
Step 2 - Select the pre-defined **default** profile and click the **Clone** button.
Step 3 - Enter profile name: (e.g., **Enterprise-Topology**), and click **Finish** to continue.



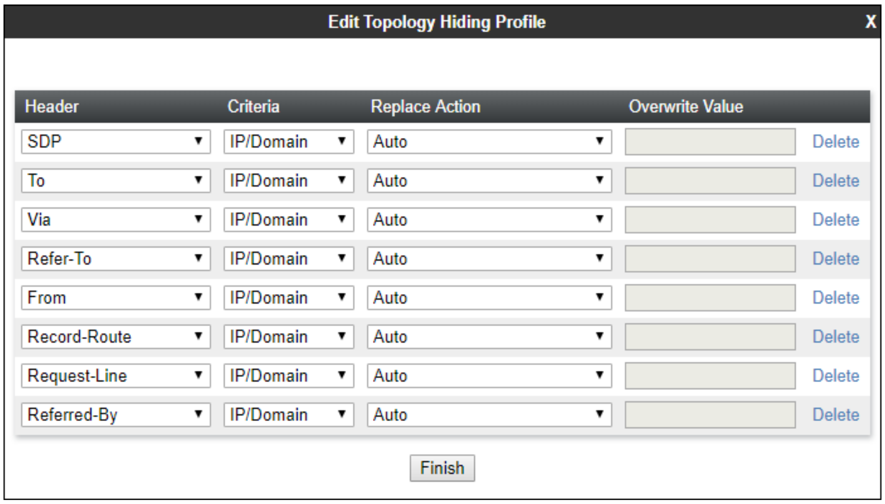
Step 4 - Edit the newly created **Enterprise-Topology** profile.
Step 5 - For the **Request-Line**, **To** and **From** headers select **Overwrite** under the **Replace Action** column. Enter the domain of the enterprise (e.g., **avayalab.com**) on the **Overwrite Value** field.
Step 6 - Click **Finish**.



8.11.2. Topology Hiding – AT&T

Repeat the steps in **Section 8.11.1**, with the following changes, to create a Topology Hiding Profile for the Avaya SBCE connection to AT&T.

Step 1 - Enter a Profile Name (e.g., **SIP-Trunk-Topology**).
Step 2 - Use the default values for all fields.
Step 3 - Click **Finish**.

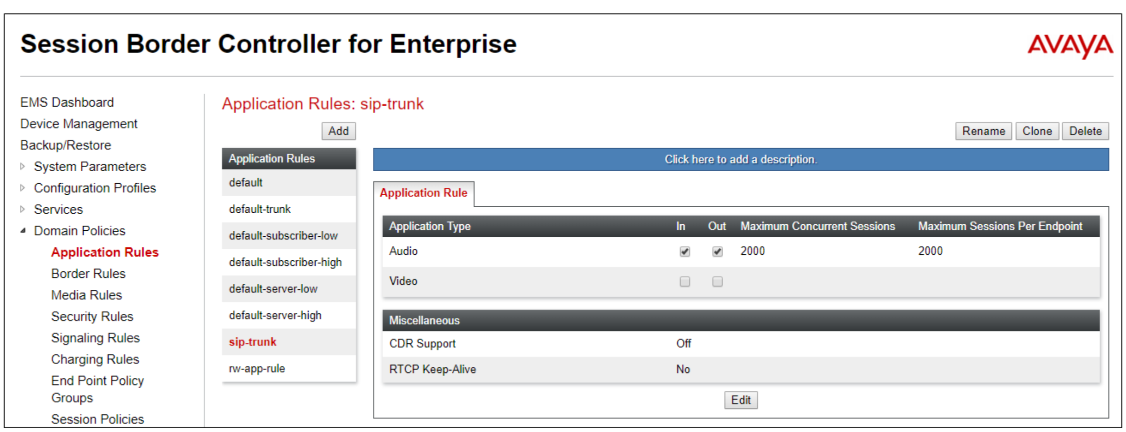


8.12. Application Rules

Application Rules define which types of SIP-based Unified Communications (UC) applications the Avaya SBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, you can determine the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion.

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

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



8.13. Media Rules

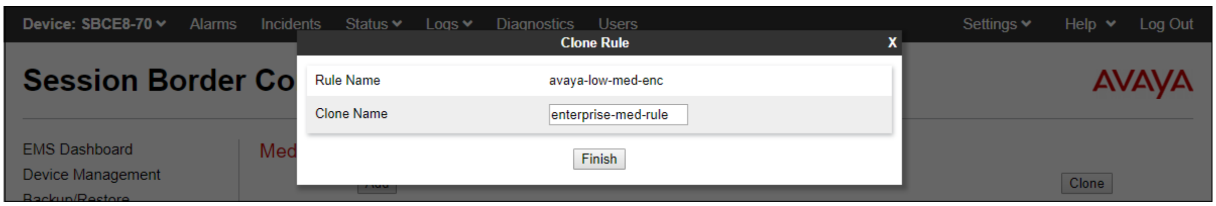
Media Rules are used to define media encryption and QoS parameters. Separate media rules are created for the enterprise and AT&T.

8.13.1. Enterprise – Media Rule

In the sample configuration, the default Media Rule **avaya-low-med-enc** was cloned to create the enterprise Media Rule, and modified as shown below:

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

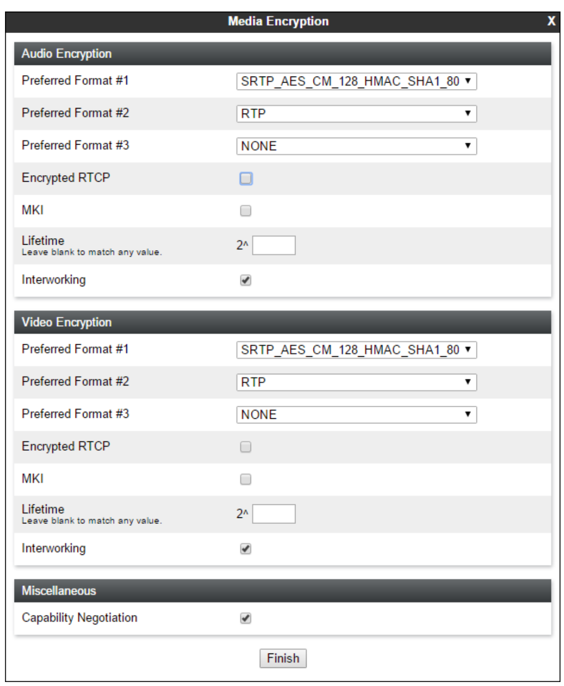
- In the **Clone Name** field enter the new Media Rule name (e.g., **enterprise-med-rule**)
- Click **Finish.** The newly created rule will be displayed.



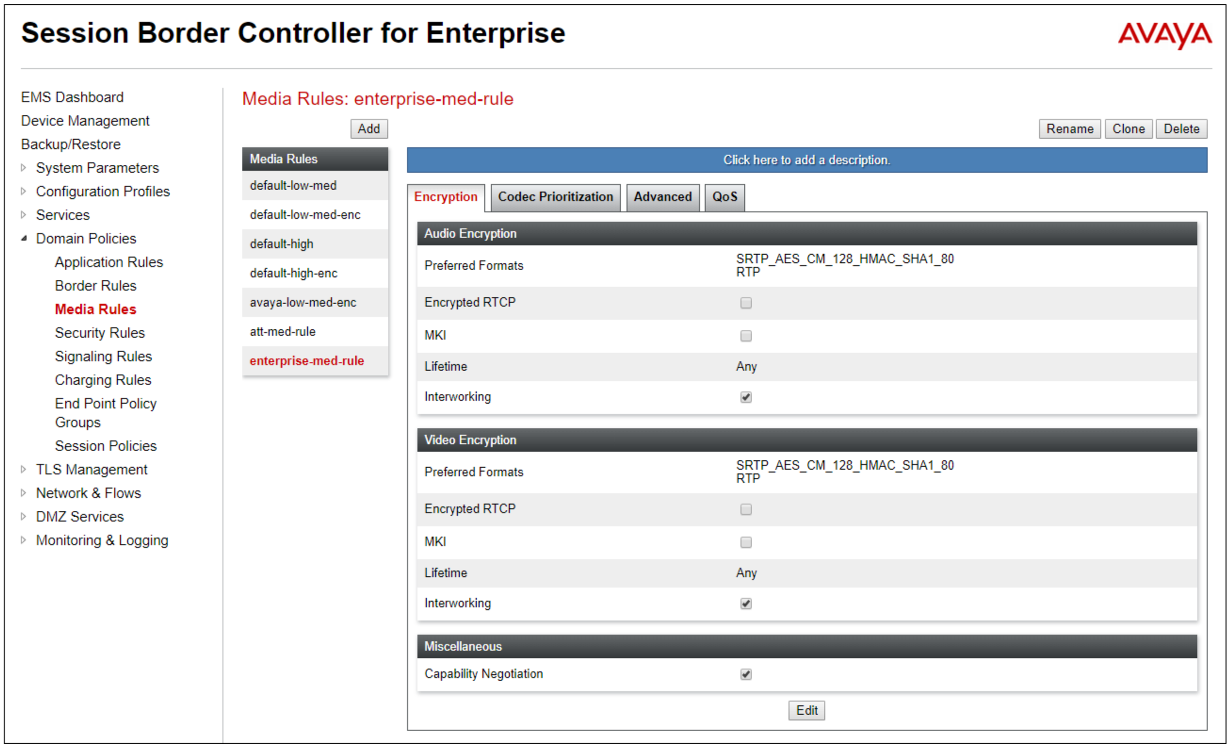
Step 4 - On the **enterprise med rule** just created, select the **Encryption** tab.

- Click the **Edit** button and the **Media Encryption** window will open.
- In the **Audio Encryption** section, select **RTP** for **Preferred Format #2**.
- In the **Video Encryption** section, select **RTP** for **Preferred Format #2**.
- In the **Miscellaneous** section, select **Capability Negotiation**.

Step 5 - Click **Finish**.



The completed **enterprise-med-rule** is shown on the screen below.

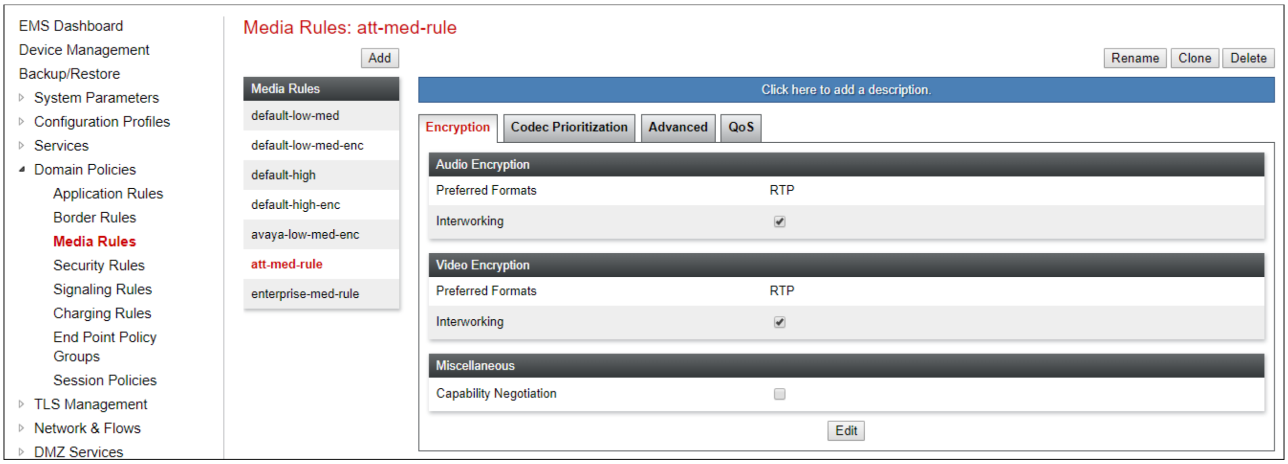


8.13.2. AT&T – Media Rule

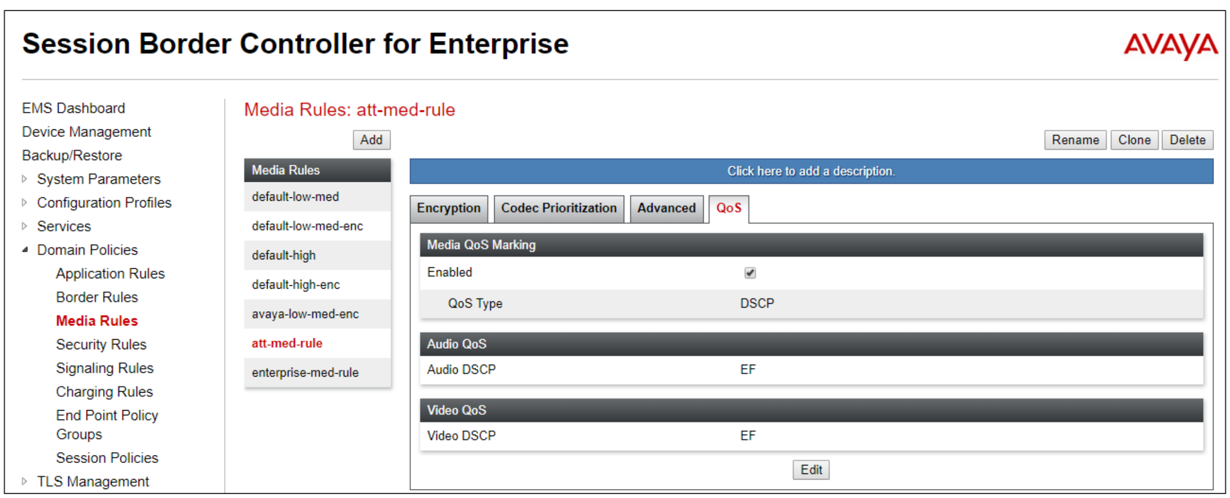
Repeat the steps in **Section 8.13.1**, with the following changes, to create a Media Rule for AT&T.

1. Clone the **default-low-med** rule
2. In the **Clone Name** field enter the new Media Rule name (e.g., **att-med-rule**)

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



DSCP values **EF** for expedited forwarding (default value) are used for Media **QoS**.



8.14. Signaling Rules

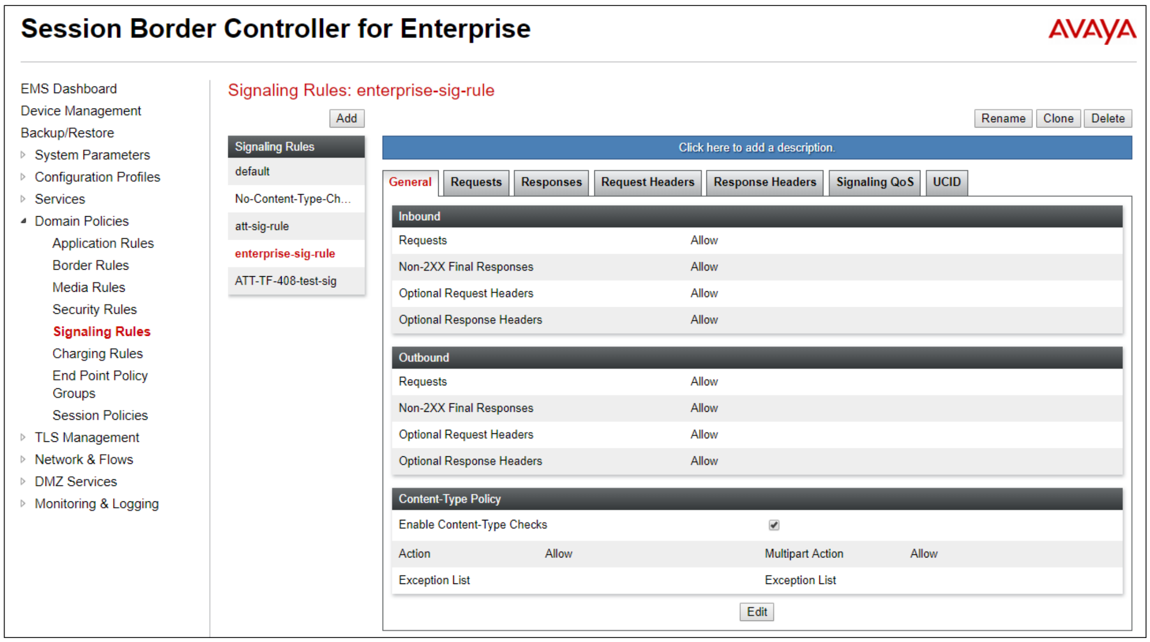
Signaling Rules are used to define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message, and to specify QoS parameters for the SIP signaling packets.

8.14.1. Signaling Rule – Enterprise

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

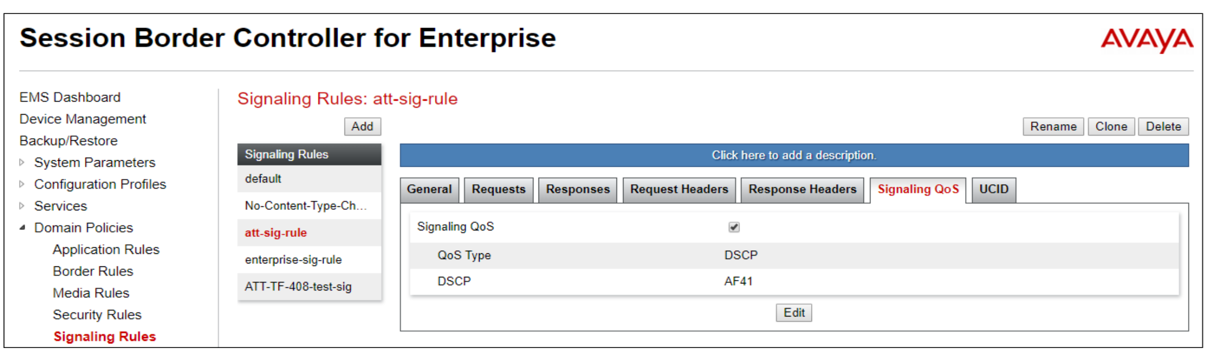
- In the **Rule Name** field enter the new Signaling Rule name (e.g., **enterprise-sig-rule)**
- Click **Finish**.

Signaling Rule **enterprise-sig-rule** show below was left unchanged from the default rule.



8.14.2. Signaling Rule – AT&T

Signaling Rule **att-sig-rule** was similarly cloned from the **default** rule and used for AT&T. Note that the DSCP value **AF41** for assured forwarding (default value) is set for **Signaling QoS**.



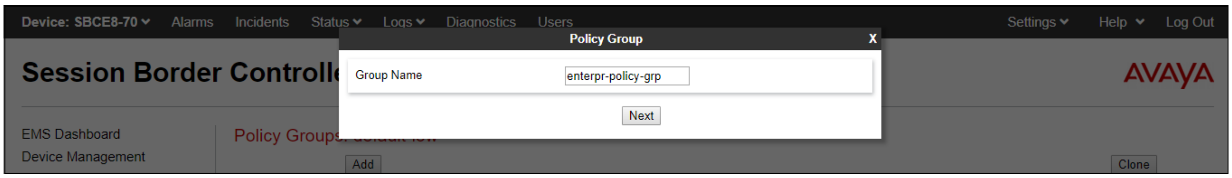
8.15. Endpoint Policy Groups

The rules created within the Domain Policies are assigned to an End Point Policy Group. The End Point Policy Group is then applied to a Server Flow in **Section 8.16**.

8.15.1. End Point Policy Group – Enterprise

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

- Enter a name for the Policy Group (e.g., **enterpr-policy-grp)**
- Click **Next**.

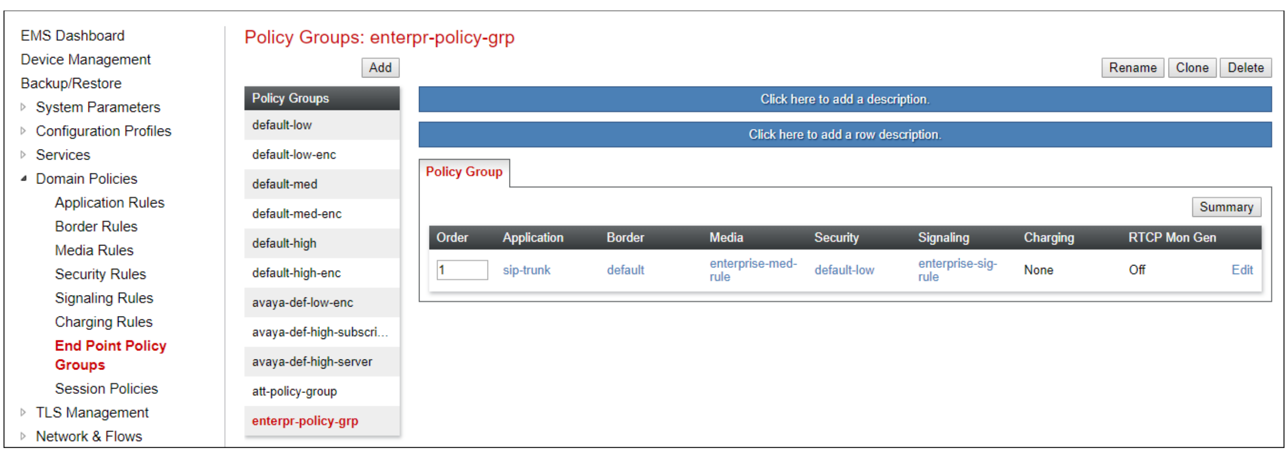


Step 3 – On the **Policy Group** window (not shown), select the following.

- **Application Rule**: **sip-trunk** (created in **Section 8.12**).
- **Border Rule**: **default**.
- **Media Rule**: **enterprise-med-rule** (created in **Section 8.13.1**).
- **Security Rule**: **default-low**.
- **Signaling Rule**: **enterprise-sig-rule** (created in **Section 8.14.1**).

Step 4 - Select **Finish**.

The completed Policy Group **enterpr-policy-grp** is shown on the screen below.



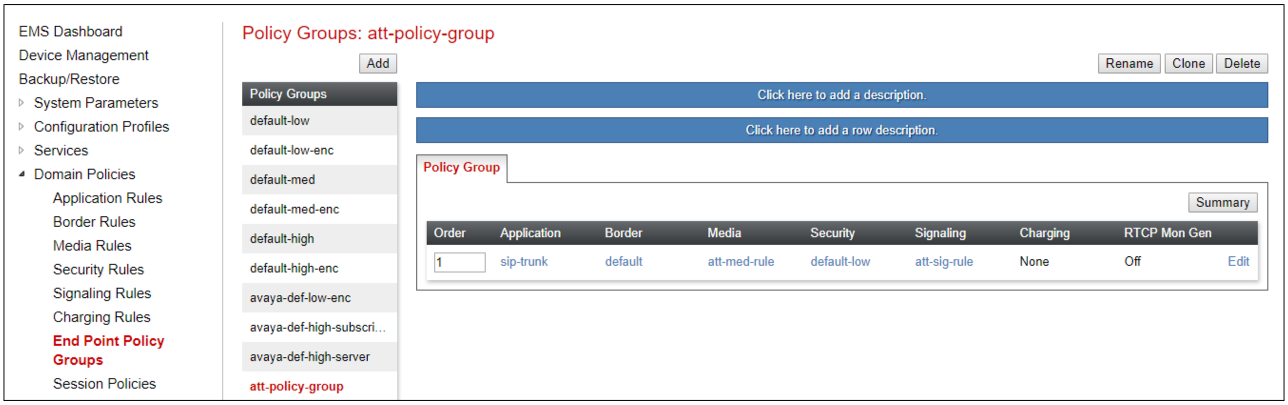
8.15.2. Endpoint Policy Group – AT&T

Step 1 - Repeat steps **1** through **4** from **Section 8.15.1** with the following changes:

- **Group Name**: **att-policy-group**
- **Media Rule**: **att-med-rule** (created in **Section 8.13.2**)
- **Signaling Rule**: **att-sig-rule** (created in **Section 8.14.2**)

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

The completed Policy Group **att-policy-grp** is shown on the screen below.



8.16. Endpoint Flows – Server Flows

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

Create separate Server Flows for the enterprise and AT&T IPFR-EF service. These flows use the interfaces, polices, and profiles defined in previous sections.

8.16.1. Server Flows – Enterprise

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

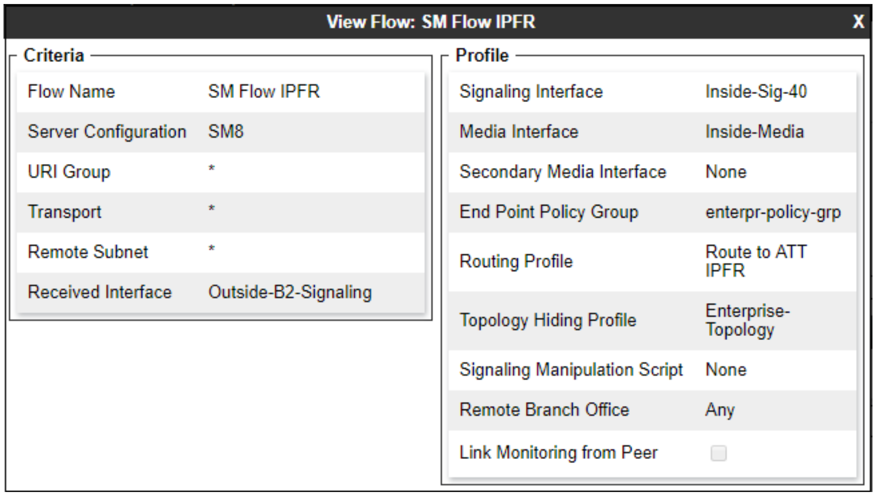
Step 2 - Select the **Server Flows** tab (not shown).

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

- **Flow Name**: Enter a name for the flow, e.g., **SM Flow IPFR**
- **Server Configuration**: **SM8** (**Section 8.9.1**).
- **URI Group**: *
- **Transport**: *
- **Remote Subnet**: *
- **Received Interface**: **Outside-B2-Signaling** (**Section 8.6**).
- **Signaling Interface**: **Inside-Sig-40** (**Section 8.6**).
- **Media Interface**: **Inside-Media** (**Section 8.5**).

- **End Point Policy Group**: **enterpr-policy-grp** (**Section 8.15.1**).
- **Routing Profile**: **Route to ATT IPFR** (**Section 8.10.2**).
- **Topology Hiding Profile**: **Enterprise-Topology** (**Section 8.11.1**).
- Let other fields at the default values.

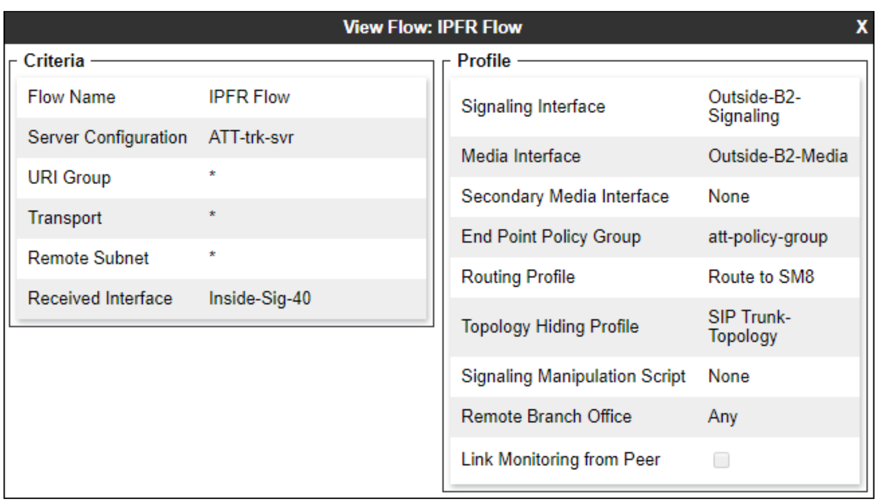
Step 4 - Click **Finish** (not shown).



8.16.2. Server Flow – AT&T

Step 1 - Repeat steps **1** through **4** from **Section 8.16.1**, with the following changes:

- **Flow Name**: Enter a name for the flow, e.g., **IPFR flow**.
- **Server Configuration**: **ATT-trk-svr** (**Section 8.9.2**).
- **Received Interface**: **Inside-Sig-40** (**Section 8.6**).
- **Signaling Interface**: **Outside-B2-Signaling (Section 8.6**).
- **Media Interface**: **Outside-B2-Media** (**Section 8.5**).
- **End Point Policy Group**: **att-policy-group** (**Section 8.15.2**).
- **Routing Profile**: **Route to SM8** (**Section 8.10.1**).
- **Topology Hiding Profile**: **SIP-Trunk-Topology** (**Section 8.11.2**).



9. AT&T IP Flexible Reach – Enhanced Features Configuration

Information regarding the AT&T IPFR-EF service offer can be obtained at https://www.business.att.com/products/sip-trunking.html or by contacting an AT&T sales representative.

The reference configuration described in these Application Notes is located in the Avaya Solutions and Interoperability Test Lab. AT&T provided the IPFR-EF service border element IP addresses, the access DID numbers, and the associated DNIS digits used in the reference configuration. In addition, the AT&T IPFR-EF features, and their associated access numbers are also provisioned and assigned by AT&T.

10. Verification Steps

The following steps may be used to verify the configuration.

10.1. AT&T IP Flexible Reach – Enhanced Features

The following scenarios may be executed to verify Communication Manager, Session Manager, Avaya SBCE, and the AT&T IPFR-EF service interoperability:

- Place inbound and outbound calls, answer the calls, and verify that two-way talk path exists.
- Verify that calls remain stable and disconnect properly.
- Verify basic call functions such as hold, transfer, and conference.
- Verify the use of DTMF signaling.
- Place an inbound call to a telephone, but do not answer the call. Verify that the call covers to voicemail (e.g., Aura® Messaging). Retrieve voicemail messages either locally or from PSTN.
- Using the appropriate IPFR-EF access numbers and codes, verify that the following features are successful:
    - Network based Simultaneous Ring – The "primary" and "secondary" endpoints ring, and either may be answered.
    - Network based Sequential Ring (Locate Me) – Verify that after the "primary" endpoint rings for the designated time, the "secondary" endpoint rings and may be answered.
    - Network based Call Forwarding Always (CFA/CFU), Network based Call Forwarding Ring No Answer (CF-RNA), Network based Call Forwarding Busy (CF-Busy), Network based Call Forwarding Not Reachable (CF-NR) – Verify that based on each feature criteria, calls are successfully redirected and may be answered.
- Inbound / Outbound T.38 fax.
- SIP OPTIONS monitoring of the health of the SIP trunk.
- Incoming and outgoing calls using the G.729and G.711 ULAW codecs.

10.2. Avaya Aura® Communication Manager Verification

This section illustrates verifications examples in Communication Manager.

The following edited Communication Manager ***list trace tac*** trace output shows an incoming call received on trunk group 5, member 1. The adaptation in Session Manager has previously converted the IPFR-EF DNIS number included in the Request URI, to the Communication Manager extension 59321, before sending the INVITE to Communication Manager.

Note that initially the Avaya Media Server (**10.64.91.86**) is included on the media path.

```
list trace tac *05                                                   Page   1

                                LIST TRACE

time            data

14:07:26 TRACE STARTED 10/16/2019 CM Release String cold-01.0.890.0-25578
14:07:32 SIP<INVITE sips:59321@avayalab.com SIP/2.0
14:07:32     Call-ID: 96740d473ca8542072835e318071083a
14:07:32     active trunk-group 5 member 1    cid 0xa59
14:07:32 SIP>SIP/2.0 180 Ringing
14:07:32     Call-ID: 96740d473ca8542072835e318071083a
14:07:32     dial 59321
14:07:32     ring station      59321 cid 0xa59
14:07:32     Alerting party  uses public-unknown-numbering
14:07:32     G729 ss:off ps:20
             rgn:4 [10.64.91.40]:17704
             rgn:1 [10.64.91.86]:6052
14:07:32     G72264K ss:off ps:20
             rgn:1 [10.5.5.211]:24706
             rgn:1 [10.64.91.86]:6054
14:07:40 SIP>SIP/2.0 200 OK
14:07:40     Call-ID: 96740d473ca8542072835e318071083a
14:07:40     active station      59321 cid 0xa59
14:07:40     Connected party  uses public-unknown-numbering
```

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

The following screen shows **Page 2** of the output of the **status trunk 5/*x*** command (where x is the trunk group member active on the call, **1 i**n the example) pertaining to this same call. Note the signaling using port **5065** between Communication Manager and Session Manager. Note that after "shuffling" is completed, the media is "ip-direct" from the IP Telephone (**10.5.5.211**) to the inside IP address of Avaya SBCE (**10.64.91.40**), releasing the media resources in the Media Server.

```
status trunk 5/1                                              Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling   IP Address                              Port
   Near-end:  10.64.91.75                           : 5065
    Far-end:  10.64.91.81                           : 5065
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:          H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.729
   Audio      IP Address                              Port
   Near-end:  10.5.5.211                            : 24706
    Far-end:  10.64.91.40                           : 17704
```

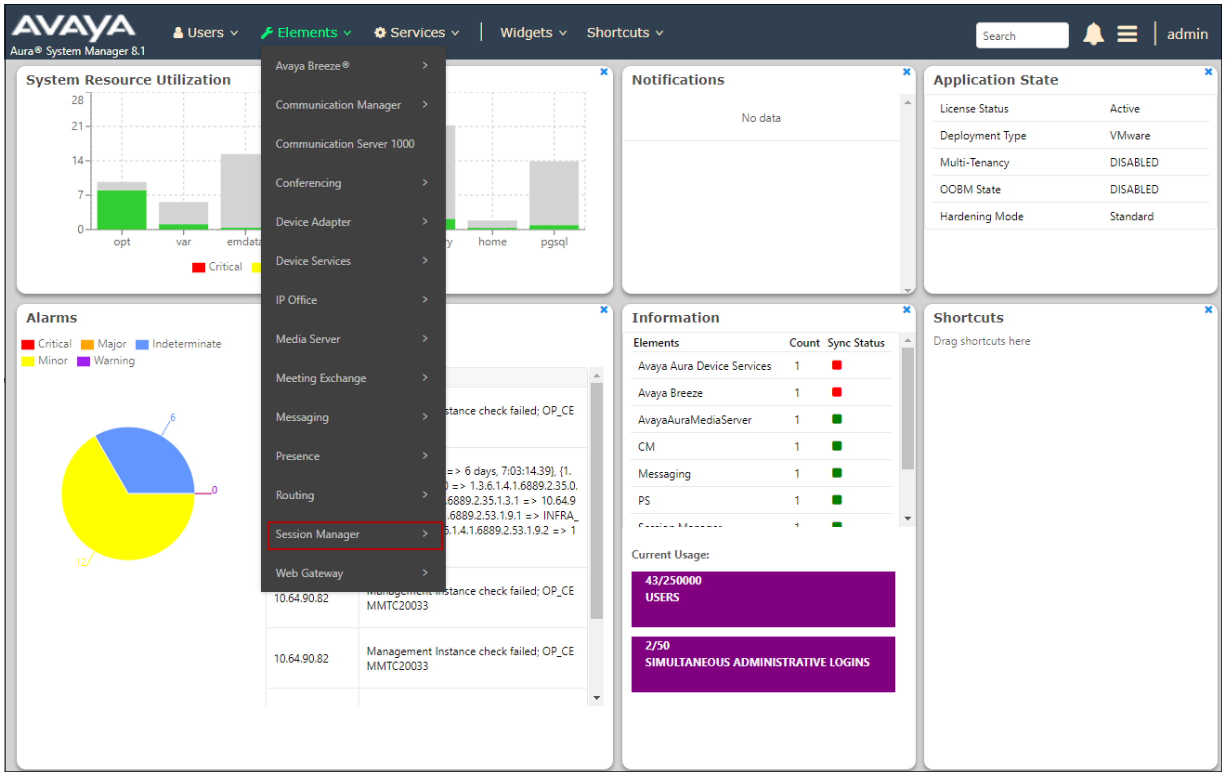
The screen below shows **Page 3** of the output of the **status trunk 5/1** command pertaining to this same call. Note that G729 and SRTP are used.

```
status trunk 5/1                                              Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T000051
T000051:TX:10.64.91.40:17704/g729/20ms/1-srtp-aescm128-hmac80
S000009:RX:10.5.5.211:24706/g729/20ms/1-srtp-aescm128-hmac80
```

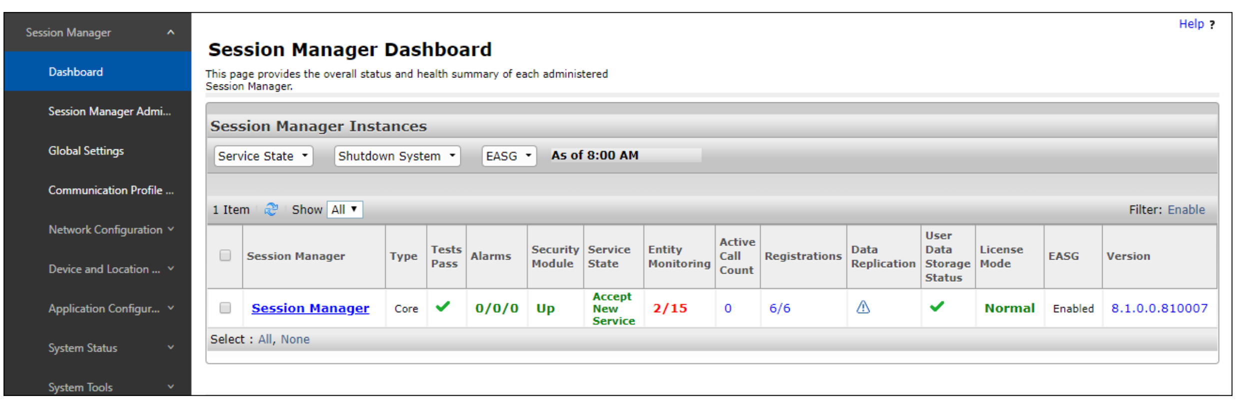
10.3. Avaya Aura® Session Manager Verification

The Session Manager configuration may be verified via System Manager.

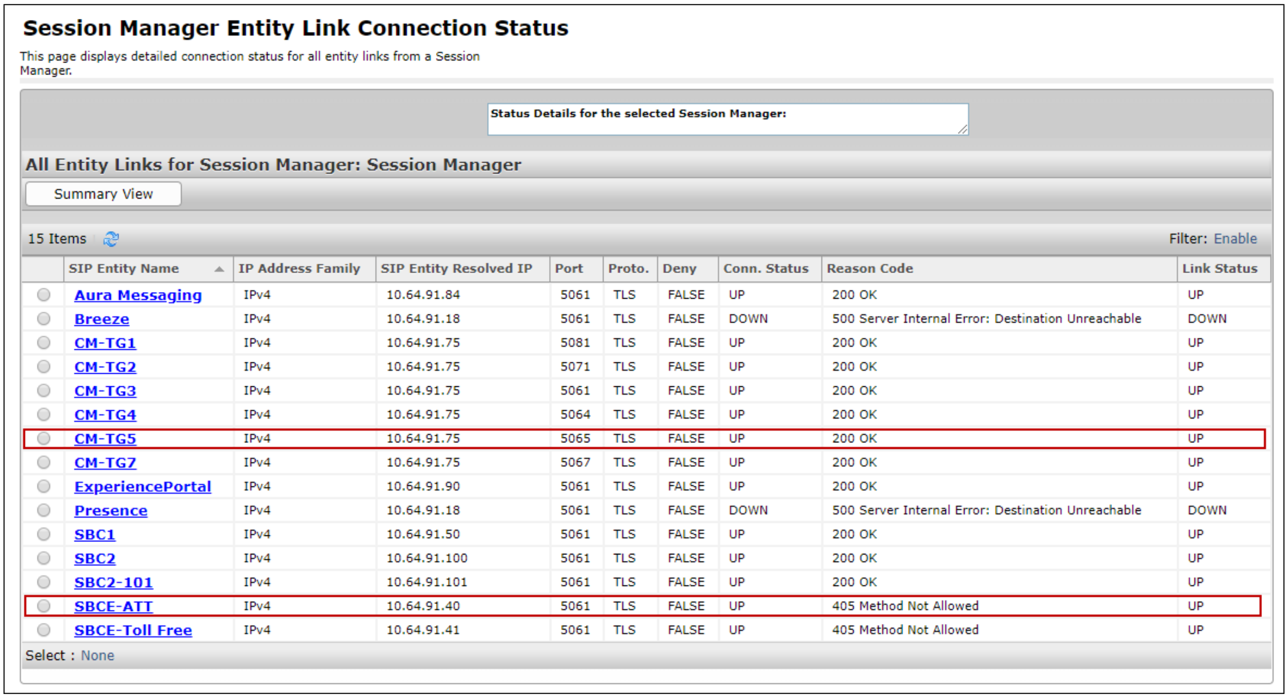
Using the procedures described in **Section 6**, access the System Manager GUI. From the **Home** screen, under the **Elements** heading, select **Session Manager**.



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



In the example, the entry **2/15** under the **Entity Monitoring** column shows that there are alarms on 2 out of the 15 Entities being monitored by Session Manager. Clicking the entry under the **Entity Monitoring** column brings up the **Session Manager Entity Link Connection Status** page. Verify that the state of the Session Manager links of interest, to Communication Manager and the Avaya SBCE under the **Conn. Status** and **Link Status** columns is **UP**, like shown on the screen below.



Session Manager Entity Link Connection Status

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

Status Details for the selected Session Manager:

All Entity Links for Session Manager: Session Manager

Summary View

15 Items

Filter: Enable

	SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
	Aura Messaging	IPv4	10.64.91.84	5061	TLS	FALSE	UP	200 OK	UP
	Breeze	IPv4	10.64.91.18	5061	TLS	FALSE	DOWN	500 Server Internal Error: Destination Unreachable	DOWN
	CM-TG1	IPv4	10.64.91.75	5081	TLS	FALSE	UP	200 OK	UP
	CM-TG2	IPv4	10.64.91.75	5071	TLS	FALSE	UP	200 OK	UP
	CM-TG3	IPv4	10.64.91.75	5061	TLS	FALSE	UP	200 OK	UP
	CM-TG4	IPv4	10.64.91.75	5064	TLS	FALSE	UP	200 OK	UP
	CM-TG5	IPv4	10.64.91.75	5065	TLS	FALSE	UP	200 OK	UP
	CM-TG7	IPv4	10.64.91.75	5067	TLS	FALSE	UP	200 OK	UP
	ExperiencePortal	IPv4	10.64.91.90	5061	TLS	FALSE	UP	200 OK	UP
	Presence	IPv4	10.64.91.18	5061	TLS	FALSE	DOWN	500 Server Internal Error: Destination Unreachable	DOWN
	SBC1	IPv4	10.64.91.50	5061	TLS	FALSE	UP	200 OK	UP
	SBC2	IPv4	10.64.91.100	5061	TLS	FALSE	UP	200 OK	UP
	SBC2-101	IPv4	10.64.91.101	5061	TLS	FALSE	UP	200 OK	UP
	SBCE-ATT	IPv4	10.64.91.40	5061	TLS	FALSE	UP	405 Method Not Allowed	UP
	SBCE-Toll Free	IPv4	10.64.91.41	5061	TLS	FALSE	UP	405 Method Not Allowed	UP

Select : None

Note – On the **SBCE-ATT** 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 forwards the Session Manager generated OPTIONS on to the AT&T IPFR-EF Border Element, it is the AT&T Border Element that is generating the 405, and the Avaya SBCE sends the response back to Session Manager.

Other Session Manager useful verification and troubleshooting tools include:

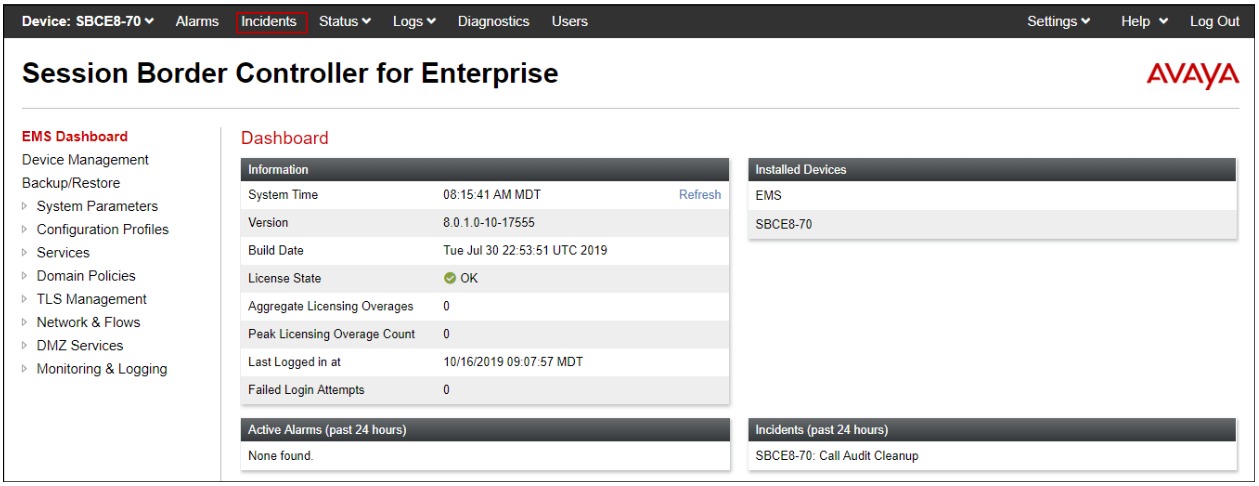
- **traceSM** – Session Manager command line tool for traffic analysis. Login to the Session Manager command line management interface to run this command.
- **Call Routing Test** - The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, from the System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

10.4. Avaya Session Border Controller for Enterprise Verification

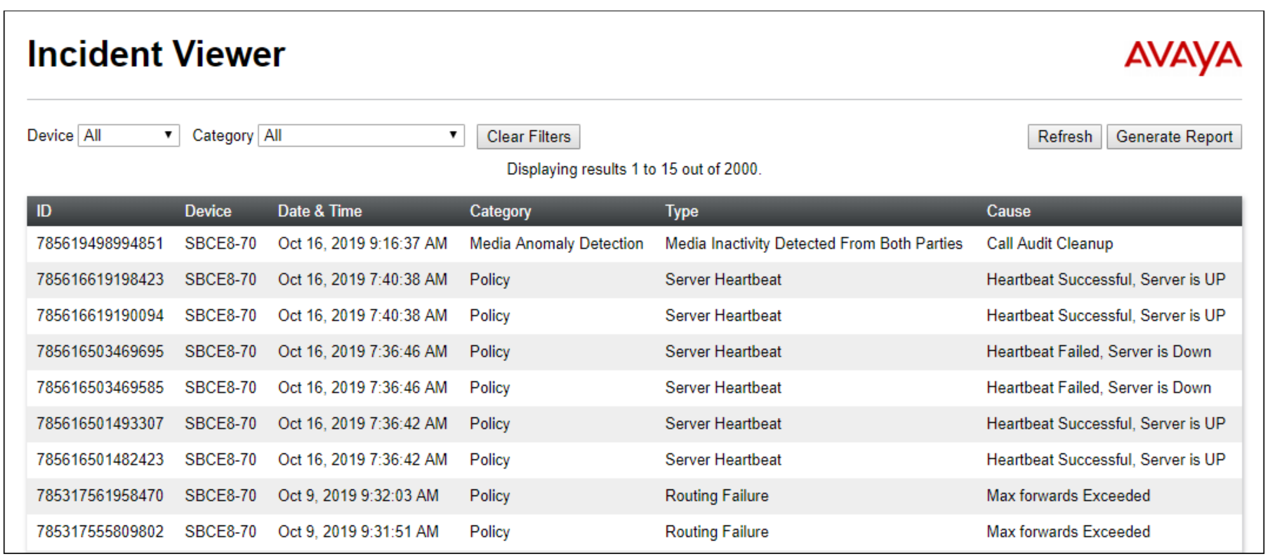
This section provides verification steps that may be performed with the Avaya SBCE.

10.4.1. Incidents

The Incident Viewer can be accessed from the Avaya top navigation menu as highlighted in the screenshot below.

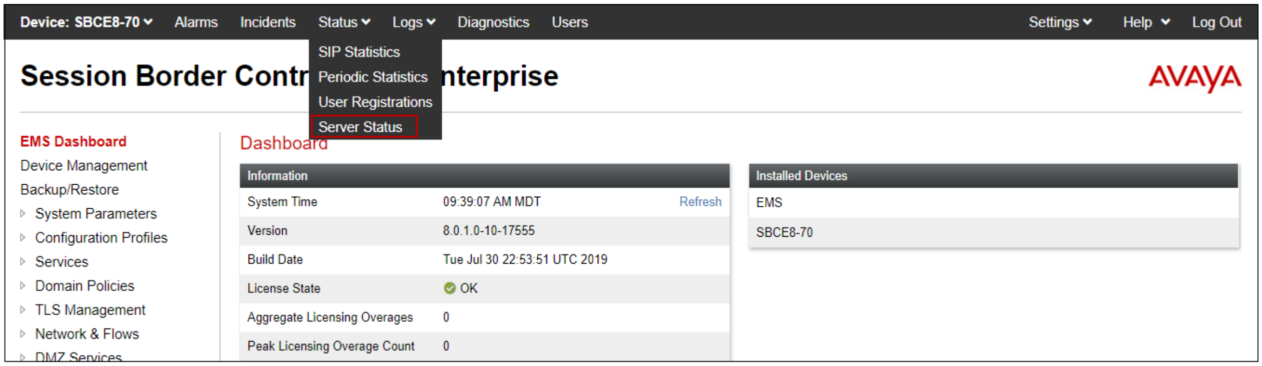


Use the Incident Viewer to verify Server Heartbeat and to troubleshoot routing failures. Further Information can be obtained by clicking on an incident in the incident viewer.

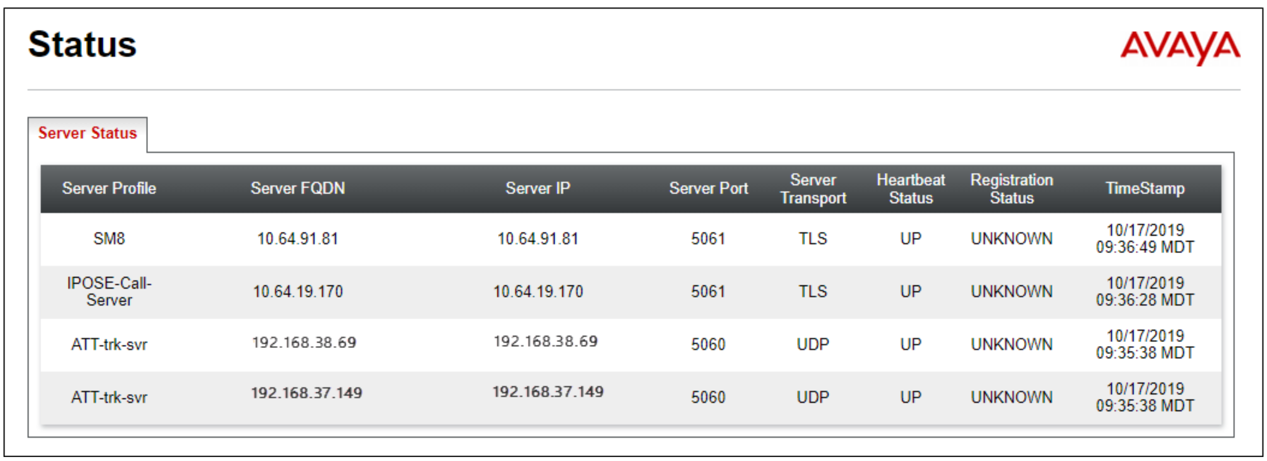


10.4.2. Server Status

The **Server Status** can be accessed from the Avaya SBCE top navigation menu by selecting the **Status** menu, and then **Server Status**.



The **Server Status** screen provides information about the condition of the connection to the connected SIP Servers. This functionality requires Heartbeat to be enabled on the SIP Server Configuration profiles, as configured in **Section 8.99**.

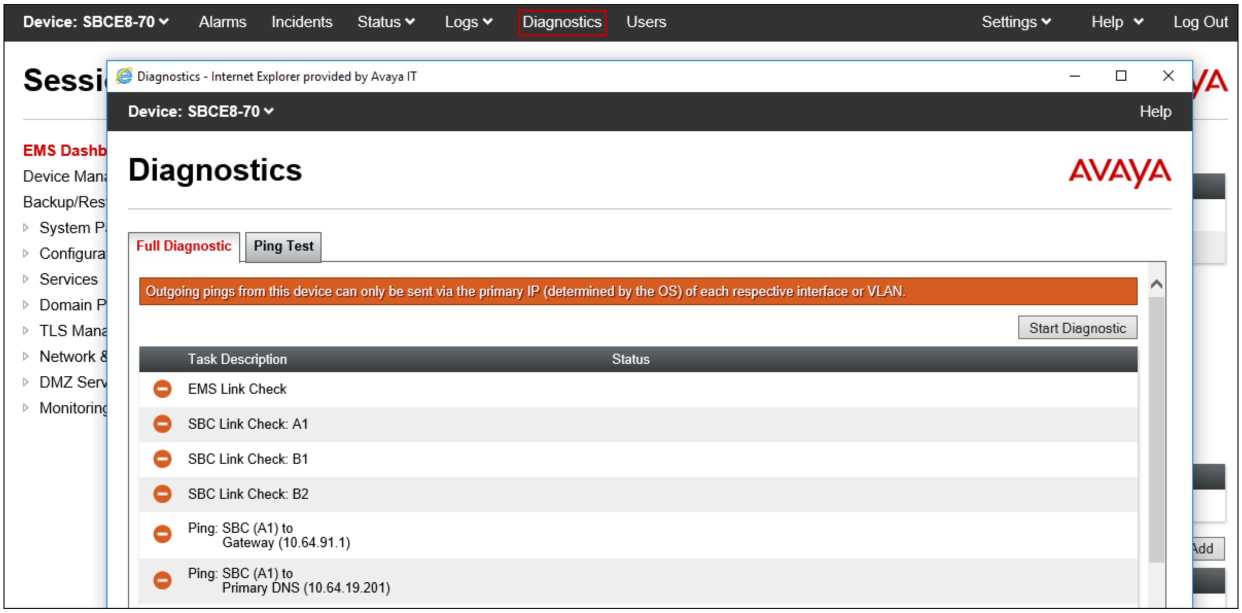

Status

Server Status

Server Profile	Server FQDN	Server IP	Server Port	Server Transport	Heartbeat Status	Registration Status	TimeStamp
SM8	10.64.91.81	10.64.91.81	5061	TLS	UP	UNKNOWN	10/17/2019 09:36:49 MDT
IPOSE-Call-Server	10.64.19.170	10.64.19.170	5061	TLS	UP	UNKNOWN	10/17/2019 09:36:28 MDT
ATT-trk-svr	192.168.38.69	192.168.38.69	5060	UDP	UP	UNKNOWN	10/17/2019 09:35:38 MDT
ATT-trk-svr	192.168.37.149	192.168.37.149	5060	UDP	UP	UNKNOWN	10/17/2019 09:35:38 MDT

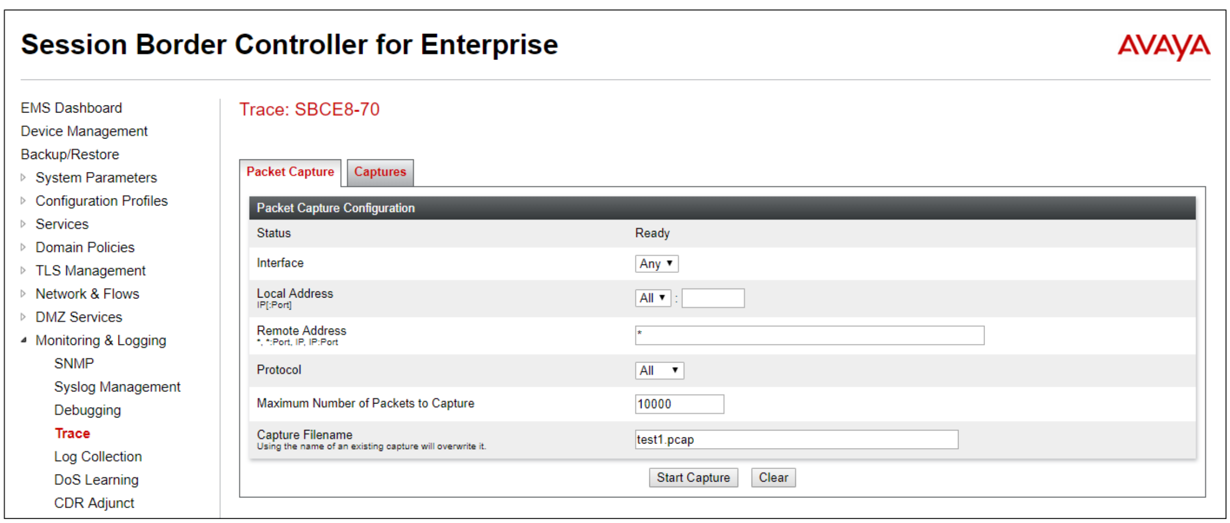
10.4.3. Diagnostic

This screen provides a **Full Diagnostics** tool to verify the link of each interface and ping the configured next-hop gateways and DNS servers. The **Ping Test** tool can be used to ping specific devices from any Avaya SBCE interface.

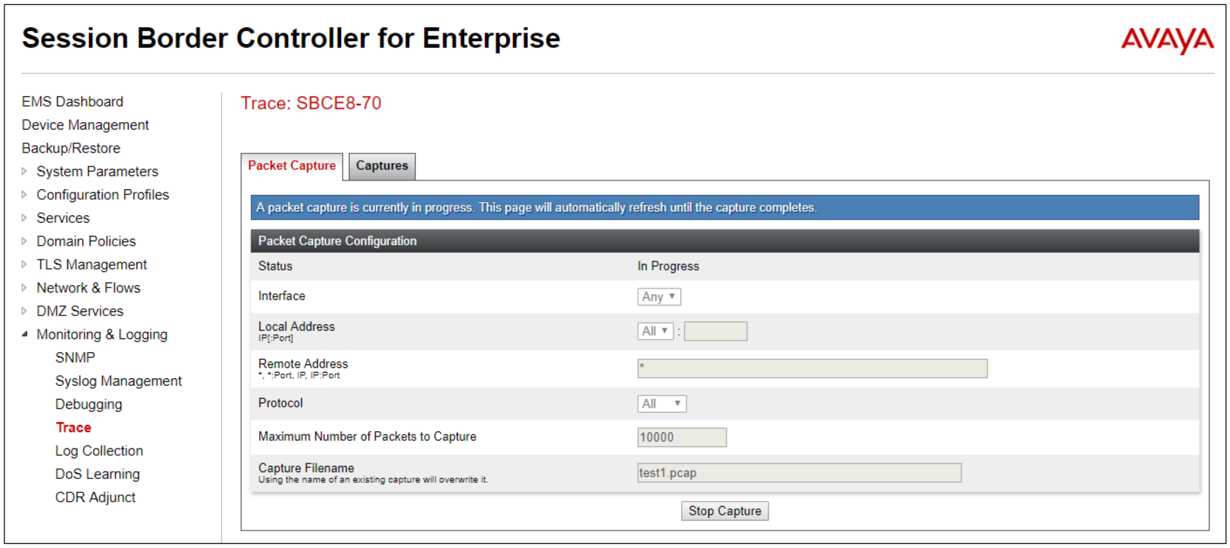


10.4.4. Tracing

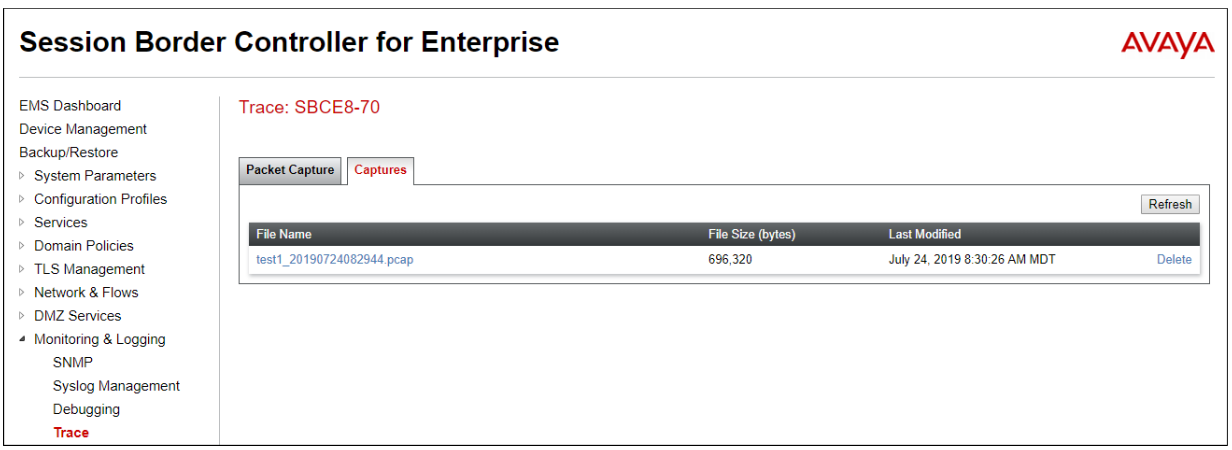
To take a call trace, navigate to **Monitoring & Logging → Trace** and select the **Packet Capture** tab. Populate the fields for the capture parameters and click **Start Capture** as shown below.



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



Select the **Captures** tab to view the files created during the packet capture.



The packet capture file can be downloaded and then viewed using a Network Protocol Analyzer like WireShark.

11. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, Avaya Aura® Experience Portal 7.2 and the Avaya Session Border Controller for Enterprise 8.0.1 can be configured to interoperate successfully with the AT&T IP Flexible Reach – Enhanced Features service, within the constraints described in **Section 2.2**.

Testing was performed on a production AT&T IP Flexible Reach – Enhanced Features 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.

12. References

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

Avaya Aura® Session Manager/System Manager

[1] *Deploying Avaya Aura® Session Manager and Branch Session Manager in Virtualized Environment*, Release 8.1, Issue 1, June 2019

[2] *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 1, June 2019

[3] *Deploying Avaya Aura® System Manager in Virtualized Environment*, Release 8.1.x, Issue 2, July 2019

[4] *Administering Avaya Aura® System Manager for Release 8.1*, Release 8.1.x, Issue 3, July 2019

Avaya Aura® Communication Manager

[5] *Deploying Avaya Aura® Communication Manager in Virtualized Environment*, Release 8.1.x, Issue 2, August 2019

[6] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 3, August 2019

[7] *Administering Avaya G430 Branch Gateway*, Release 8.1.x, Issue 1, June 2019

[8] *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 8.0.x, Issue 7, June 2019

[9] *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager*, Issue 1.1, June 2018

Avaya Session Border Controller for Enterprise

[10] *Administering Avaya Session Border Controller for Enterprise*, Release 8.0.x, Issue 4, August 2019

[11] *Deploying Avaya Session Border Controller for Enterprise in Virtualized Environment* Release 8.0.x, Issue 3, August 2019

Avaya Aura® Messaging

[12] *Administering Avaya Aura® Messaging*, Release 7.1.0, Issue 8, October 2019

Avaya Aura® Experience Portal

[13] *Administering Avaya Aura® Experience Portal*, Release 7.2.3, Issue 1, September 2019

[14] *Implementing Avaya Aura® Experience Portal on a single server*, Release 7.2.3, Issue 1, September 2019

AT&T IP Flexible Reach - Enhanced Features Service:

[15] *AT&T IP Flexible Reach – Product Description*
https://www.business.att.com/content/dam/attbusiness/briefs/voice-and-collaboration-ip-flex-reach-product-brief.pdf

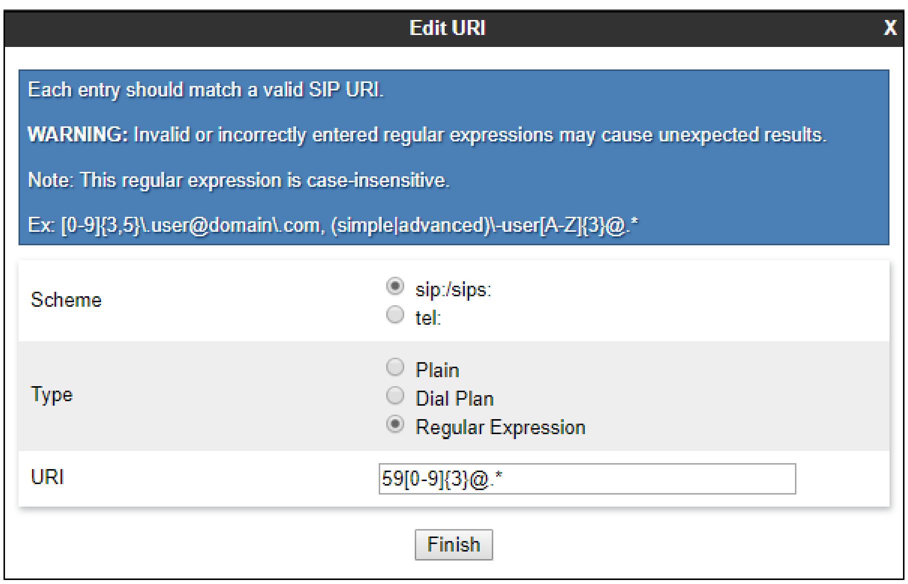
13. Appendix A – Avaya SBCE – Refer Handling

One of the important capabilities to the Experience Portal environment is the Avaya SBCE Refer Handling option. As described in **Section 3.2.4**, Experience Portal inbound call processing may include call redirection to Communication Manager agents, or other CPE destinations. This redirection is accomplished by having Experience Portal send SIP REFER messaging to the Avaya SBCE. Enabling the Refer Handling option causes the Avaya SBCE to intercept and process the REFER and generate a new SIP INVITE messages back to the CPE (e.g., Communication Manager).

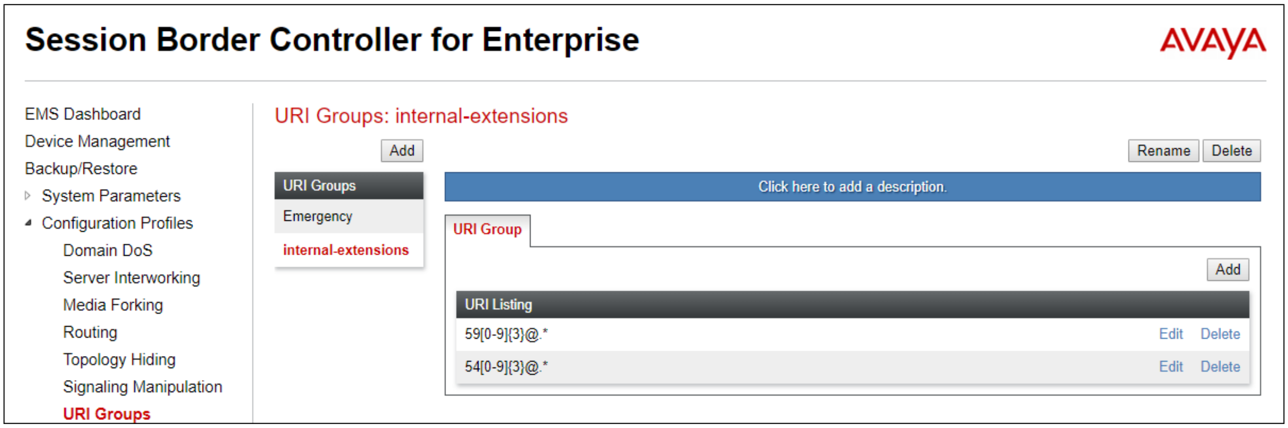
As an additional option, the Refer Handling feature can also specify *URI Group* criteria as a discriminator, whereby SIP REFER messages matching the URI Group criteria are processed by the Avaya SBCE, while SIP REFER messages that do not match the URI Group criteria, are passed through to AT&T.

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

- **Scheme**: **sip:/sips:**
- **Type: Regular Expression**
- **URI: 59[0-9]{3}@.*** This will match 5-digit local extensions starting with 59, e.g., 59001.
- Select **Finish**.



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

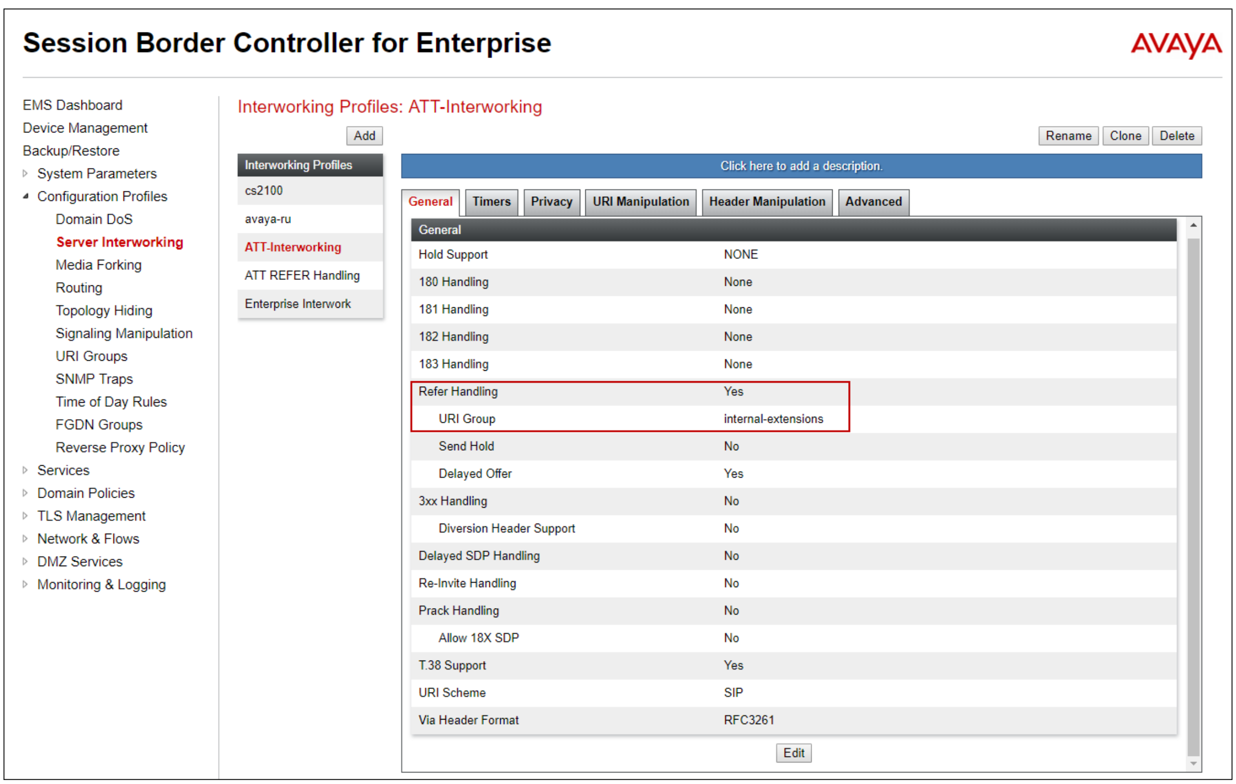


Edit the existing AT&T Server Interworking Profile to enable Refer Handling and assign the newly created URI Group.

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

Step 2 - Select the ATT-Interworking Profile created in **Section 8.7.2** and click **Edit**

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



14. Appendix B – Avaya SBCE – SigMa Script File

Details of the Signaling Manipulation script used in the configuration of the Avaya SBCE, in **Section** Error! Reference source not found.**8**.

```
within session "ALL"
{
    act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
    {

//Remove gsid and epv parameters from Contact header to hide internal topology
        remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
        remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);

//Remove Bandwidth from SDP
        %BODY[1].regex_replace("b=(TIAS|AS|CT):(\d+)\r\n","");

// fix call-fwd
        %HEADERS["Diversion"][1].regex_replace("sips","sip");
    }
 }

//OPTIONAL - Change AT&T Max-Forwards value from 0 to 30
within session "OPTIONS"
{
    act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="AFTER_NETWORK"
    {
        %HEADERS["Max-Forwards"][1] = "30";
        }
 }

// OPTIONAL Experience Portal - modify PAI Header

within session "INVITE"
{
 act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
  {
        if (%INITIAL_REQUEST = "true") then
        {
          if (%HEADERS["User-Agent"][1].regex_match("Avaya\-VoicePortal")) then
          {
            %HEADERS["P-Asserted-Identity"][1].URI.USER = "3035559329";
            }
        }
    }
 }
```

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