



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 AVPN¹ or MIS/PNT² 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/E911 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.

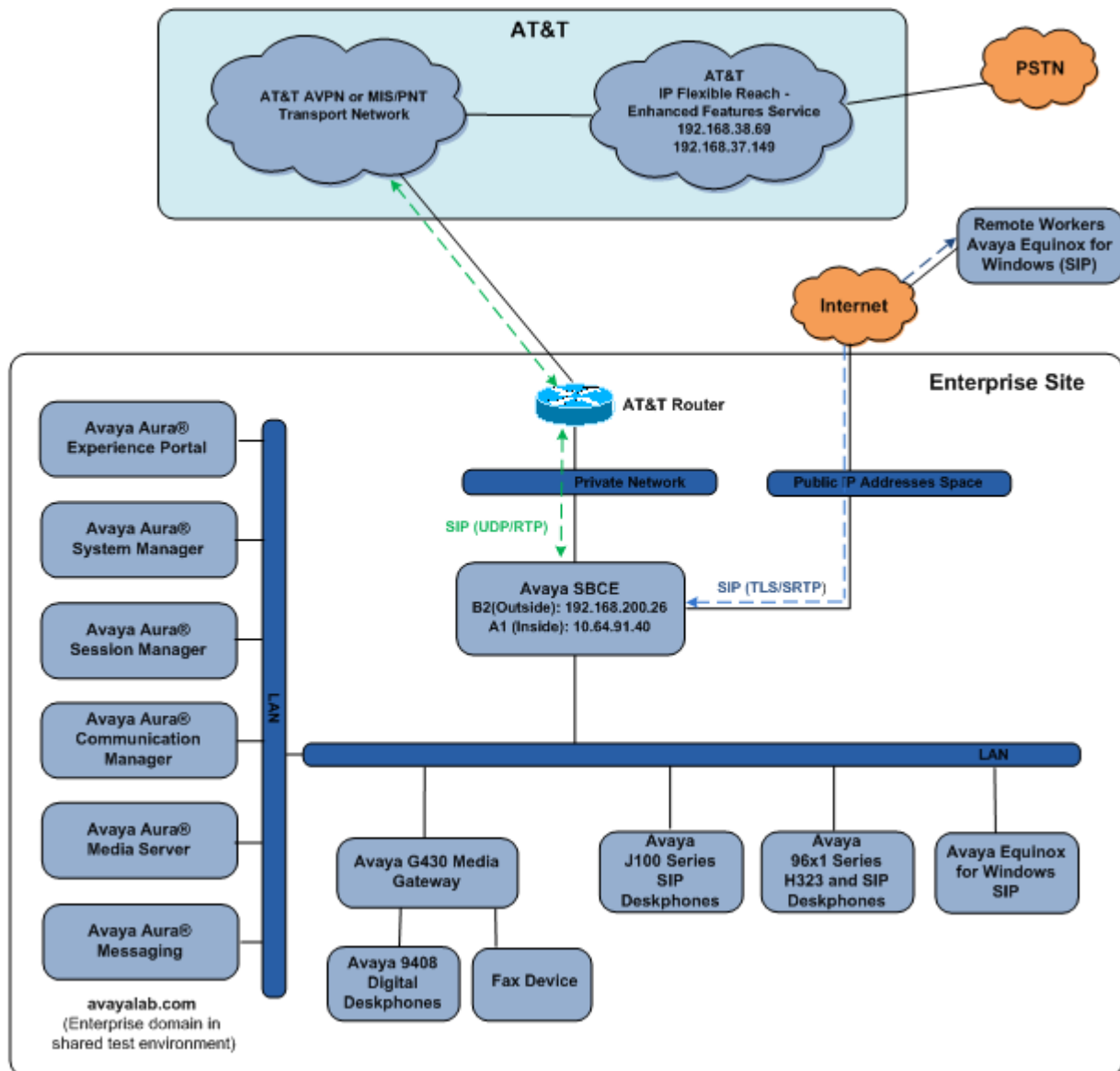


Figure 1: Reference configuration

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.

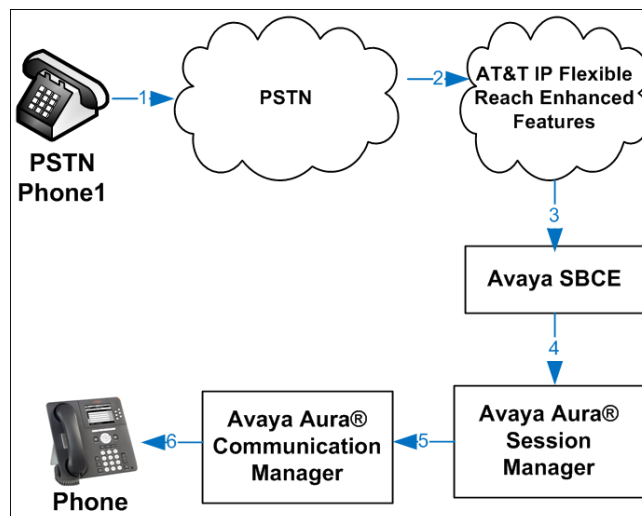


Figure 2: Inbound IPFR-EF Call

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.

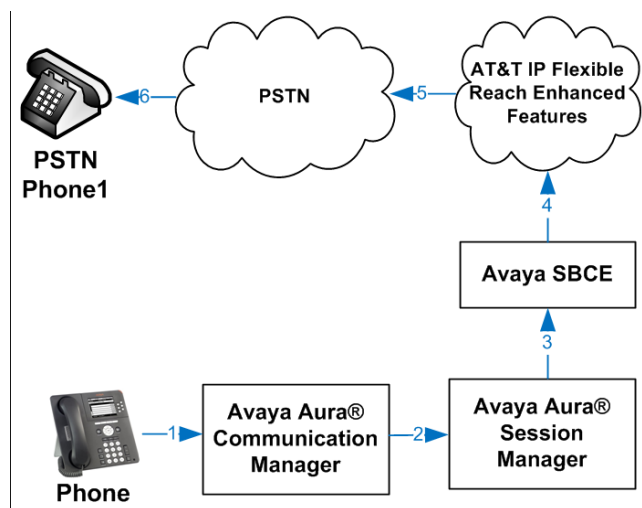


Figure 3: Outbound IPFR-EF Call

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.
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.

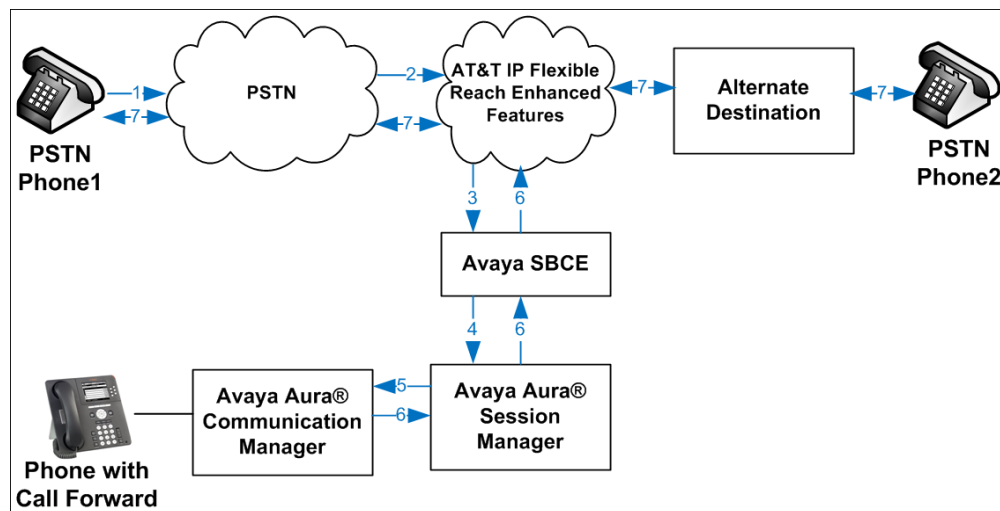


Figure 4: Station Re-directed (e.g., Call Forward) IPFR-EF Call

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).

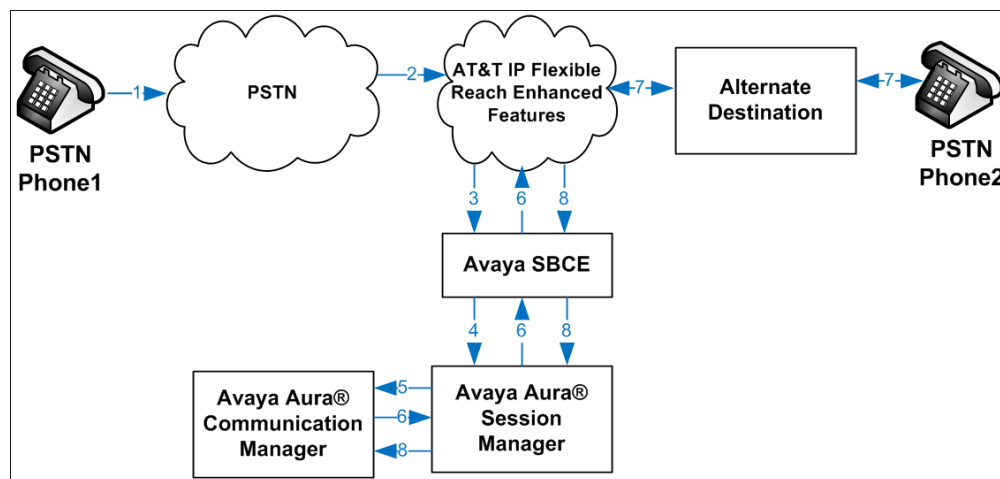


Figure 5: Network Based Blind Transfer Using REFER (Communication Manager Vector)

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.

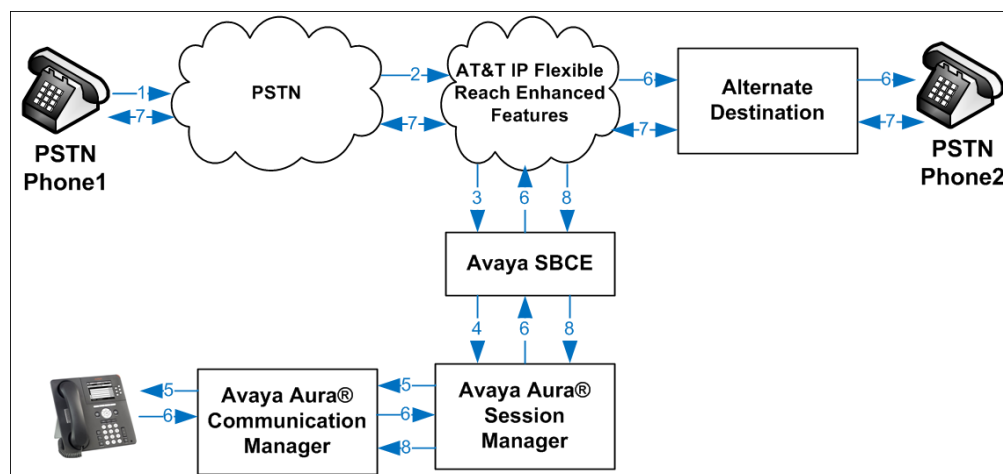


Figure 6: Attended/Unattended Transfer Using REFER (Communication Manager Station)

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.

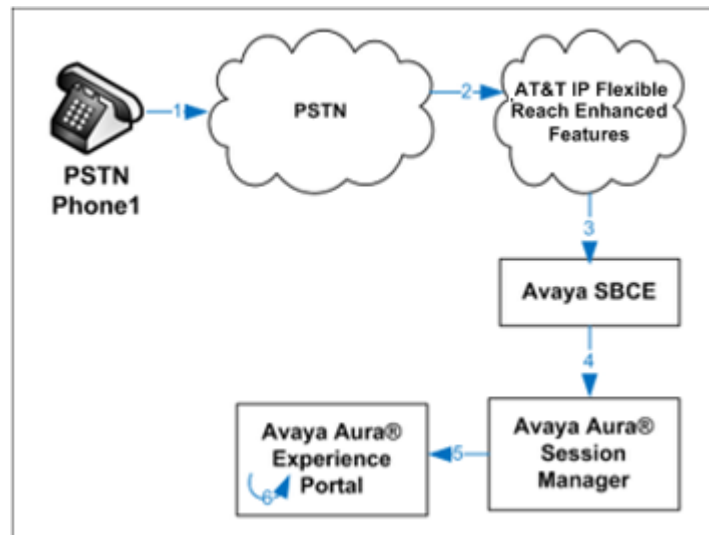


Figure 7: Inbound Call Handling Entirely by Avaya Aura® Experience Portal

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.

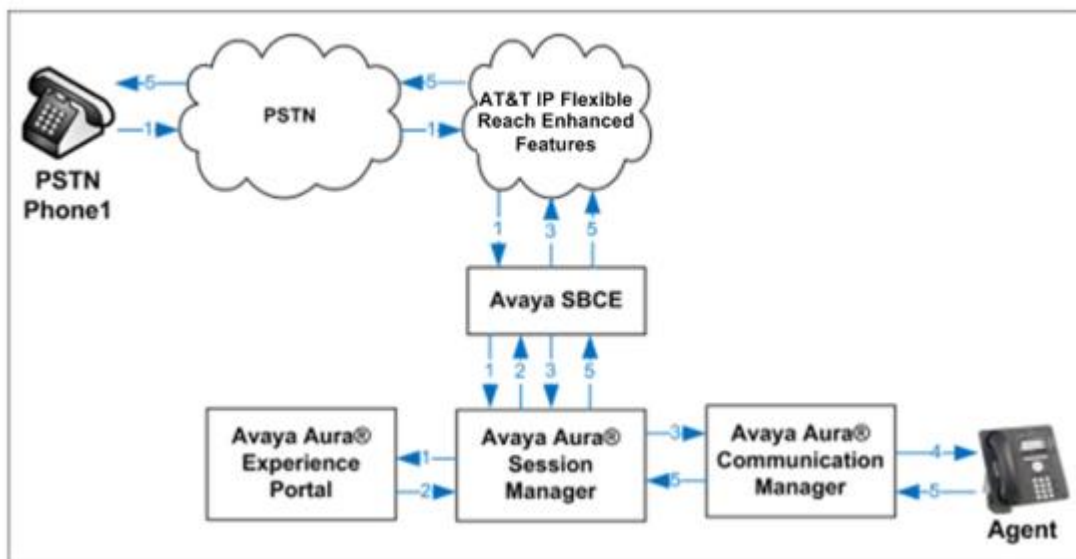


Figure 8: Avaya Aura® Experience Portal Transfers Call to Avaya Aura® Communication Manager

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.

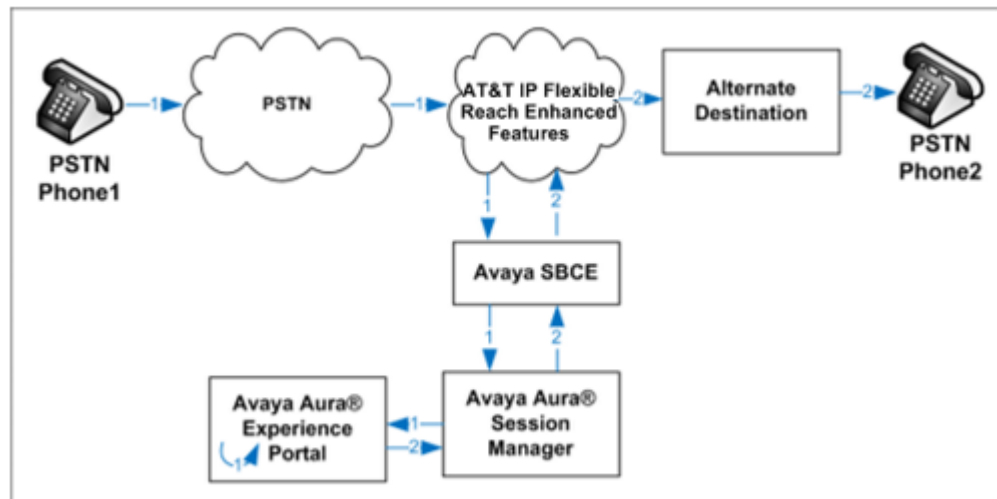


Figure 9: Inbound Call transferred by Experience Portal to another PSTN number

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	Digital Loss Plan Modification? y	
Async. Transfer Mode (ATM) Trunking? n	DS1 MSP? y	
ATM WAN Spare Processor? n	DS1 Echo Cancellation? y	
ATMS? 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		
Personal Station Access (PSA)? y	System Management Data Transfer? n	
PNC Duplication? n	Tenant Partitioning? y	
Port Network Support? y	Terminal Trans. Init. (TTI)? y	
Posted Messages? y	Time of Day Routing? 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 String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call 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	
Codec Set: 1	Intra-region IP-IP Direct Audio: yes
UDP Port Min: 16384	Inter-region IP-IP Direct Audio: yes
UDP Port Max: 32767	IP Audio Hairpinning? n
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 rgn	codec set	direct WAN	WAN-BW-limits Units	Video Total Norm	Intervening Prio Shr	Regions	Dyn CAC	A R	G L	c 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 rgn	codec set	direct WAN	WAN-BW-limits Units	Video Total Norm	Intervening Prio Shr	Regions	Dyn CAC	A R	G L	c 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 Codec	Silence Suppression	Frames Per Pkt	Packet 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						
	Mode	Redun- dancy	ECM: y	Packet Size (ms)		
FAX	t.38-standard	0				
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 Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: G.729A	n	3	30
2: G.711MU	n	3	30

Media Encryption

1: 1-srtp-aescm128-hmac80

2: none

Encrypted SRTCP: enforce-unenc-srtcp

change ip-codec-set 4

Page 2 of 2

IP CODEC SET

Allow Direct-IP Multimedia? n

	Mode	Redundancy	ECM: y	Packet Size (ms)
FAX	t.38-standard	0		
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		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? 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
Direction: two-way	Outgoing Display? n	TAC: *05
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	
	UII 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					
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len	
5	14	5	17325552754	11	Total Administered: 46
5	50	4	173255	11	Maximum Entries: 240
5	54	5	146955	11	Note: If an entry applies to a SIP connection to Avaya Aura(R) Session Manager, the resulting number must be a complete E.164 number.
5	59	5	130355	11	
5	10001	2	18665553221	11	
					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 Len	Ext Code	Trk Grp(s)	Private Prefix	Total 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
2:
3:
                                n      user
                                n      user
                                n      user

      BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature PARM Sub  Numbering LAR
      0 1 2 M 4 W      Request      Dgts Format
1: y y y y y n  n      rest      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
2:
3:
                                n      user
                                n      user
                                n      user

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


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 17xxxxxxxxxx.
- 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 String	Total		Route	Call	Node	ANI
		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**.

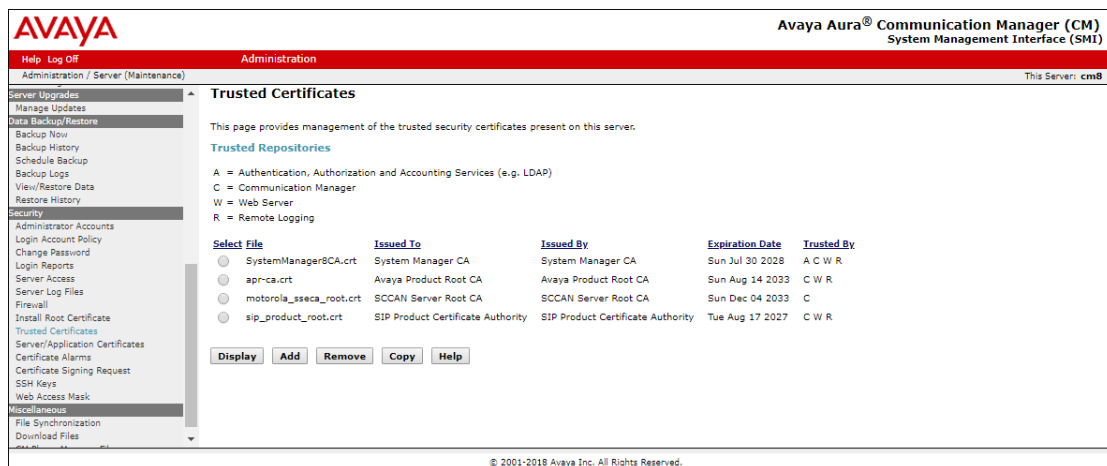
5.17. Verify TLS Certificates – Communication Manager

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.

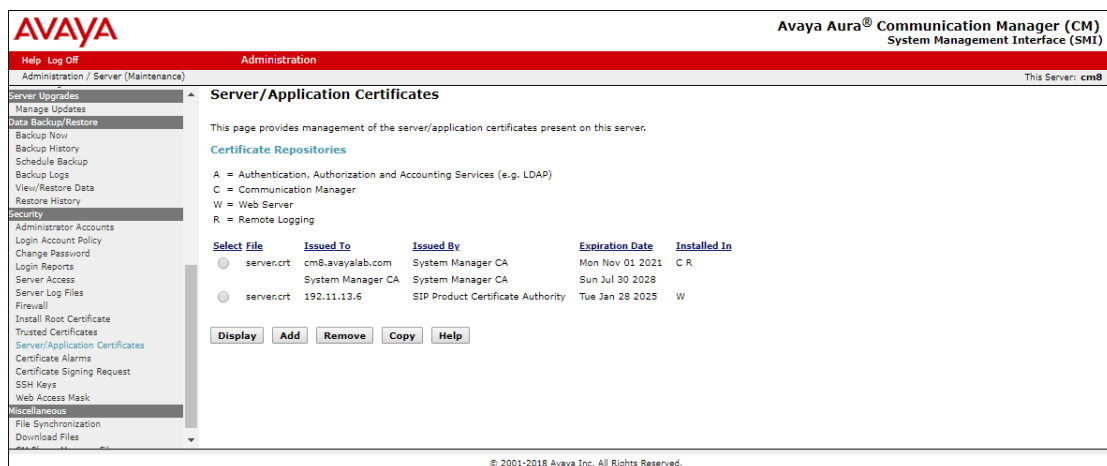
In the reference configuration, TLS transport is used for the communication between Session Manager and Communication Manager. Follow the steps below to verify the certificates used by Communication Manager.

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

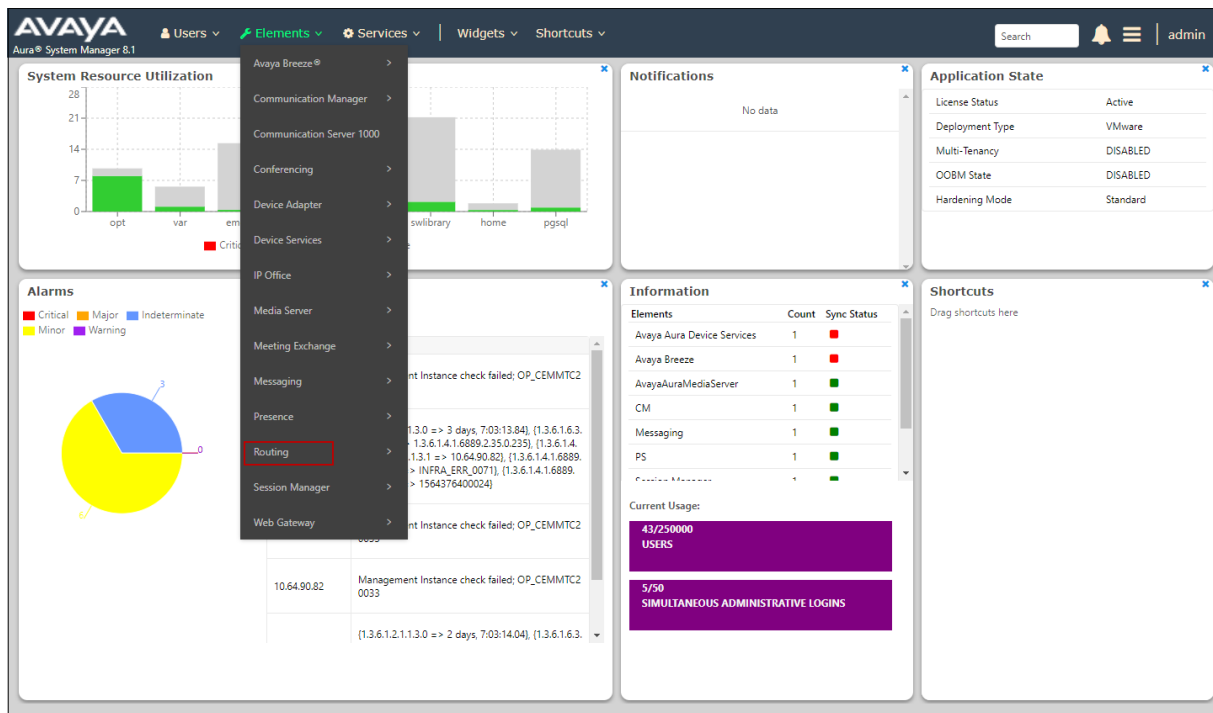
Note – These Application Notes assume that basic System Manager and Session Manager administration has already been performed. Consult [1] - [4] in the References section for further details.

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.

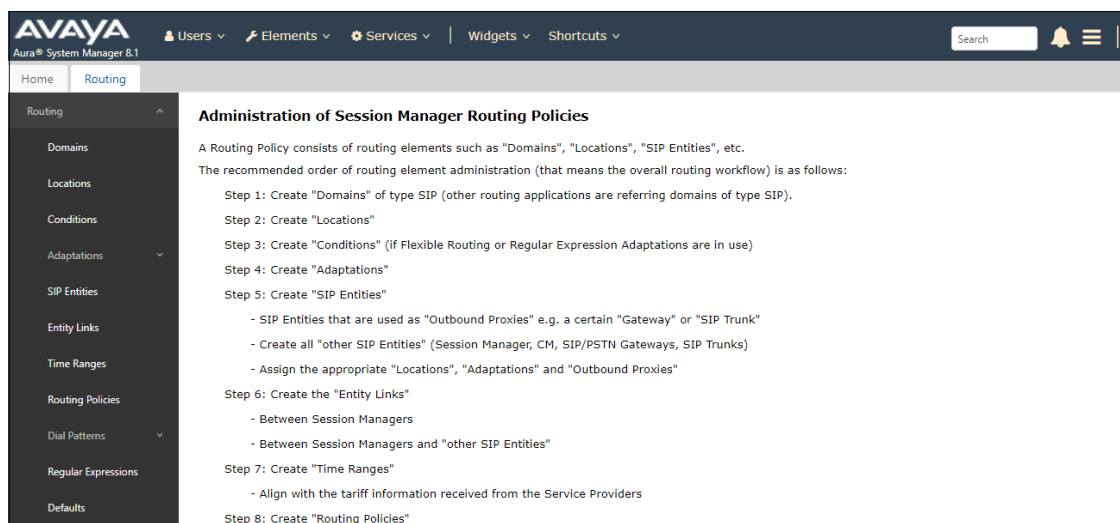
- Define a SIP Domain.
- Define a Location for Customer Premises Equipment (CPE).
- Configure the Adaptation Modules that will be associated with the SIP Entities for Communication Manager and the Avaya SBCE.
- Define SIP Entities corresponding to Session Manager, Communication Manager, the Avaya SBCE, 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.

The screenshot shows the 'Domain Management' interface. On the left is a navigation menu with 'Routing' expanded, showing 'Domains', 'Locations', 'Adaptations', 'SIP Entities', and 'Entity Links'. The main panel has a title 'Domain Management' and buttons for 'New', 'Edit', 'Delete', 'Duplicate', and 'More Actions'. Below these is a table with 1 item. The table has columns for 'Name', 'Type', and 'Notes'. The row shows 'avayalab.com' as the Name and 'sip' as the Type. At the bottom, it says 'Select : All, None'.

Name	Type	Notes
avayalab.com	sip	

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.

AVAYA
Aura® System Manager 8.1

Users | Elements | Services | Widgets | Shortcuts

Home | Routing | Search | admin

Location Details [Commit] [Cancel] Help ?

General

* Name: Main
Notes: Avaya SIL

Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:
Associated CM SIP Entity:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec
Total Bandwidth:
Multimedia Bandwidth:
Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): 2000 Kbit/Sec
Maximum Multimedia Bandwidth (Inter-Location): 2000 Kbit/Sec
* Minimum Multimedia Bandwidth: 64 Kbit/Sec
* Default Audio Bandwidth: 80 Kbit/Sec

Alarm Threshold

Overall Alarm Threshold: 80 %
Multimedia Alarm Threshold: 80 %
* Latency before Overall Alarm Trigger: 5 Minutes
* Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

Add Remove

0 Items Filter: Enable

IP Address Pattern	Notes

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**).

The screenshot shows the 'Adaptation Details' configuration page. On the left, a sidebar under 'Routing' has 'Adaptations' highlighted. The main panel has a title 'Adaptation Details' and a 'General' tab. Below the tab are several input fields: 'Adaptation Name' with the value 'CM TG5 ATT IPFR', 'Module Name' with a dropdown menu showing 'DigitConversionAdapter', 'Module Parameter Type' with a dropdown menu, 'Egress URI Parameters' with an empty text box, and 'Notes' with the value 'CM - ATT - IPFR'. At the top right of the main panel are 'Commit' and 'Cancel' buttons, and a 'Help ?' link.

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

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	*30355593	*10	*10		*5		destination ▼		10 digit DNIS to extension
<input type="checkbox"/>	*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.

The screenshot shows the 'Adaptation Details' configuration page. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Conditions, Adaptations (selected), Regular Expression..., SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Adaptation Details' and includes 'Commit' and 'Cancel' buttons. Under the 'General' tab, the 'Adaptation Name' is 'SBC1-Adaptation for ATT', the 'Module Name' is 'AttAdapter', and the 'Module Parameter Type' is 'Name-Value Parameter'. Below this is a table for Name-Value Parameters with columns 'Name' and 'Value'. One entry is visible: 'eRHdrs' with a long list of Avaya headers. There are 'Add' and 'Remove' buttons for this table. Below the table are fields for 'Egress URI Parameters' and 'Notes' (containing 'SBC - ATT IPTF'). At the bottom, there are two sections for 'Digit Conversion for Incoming Calls to SM' and 'Digit Conversion for Outgoing Calls from SM', each with an 'Add' and 'Remove' button and a table with columns: Matching Pattern, Min, Max, Phone Context, Delete Digits, Insert Digits, Address to modify, Adaptation Data, and Notes. Both digit conversion sections currently show '0 Items'.

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.

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

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**.

SIP Entity Details Commit Cancel

General

* Name: CM-TG5

* FQDN or IP Address: 10.64.91.75

Type: CM

Notes: Trunk Group 5 - ATT IPFR

Adaptation: CM TG5 ATT IPFR

Location: CM-TG-5

Time Zone: America/Denver

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

Minimum TLS Version: Use Global Setting

Credential name:

Securable: ☐

Call Detail Recording: none

Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

Monitoring

SIP Link Monitoring: Use Session Manager Configuration

CRLF Keep Alive Monitoring: Use Session Manager Configuration

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**.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy	Deny New Service	Notes
* SM to CM TGS	* Session Manager	TLS	* 5065	* CM-TG5	* 5065	<input type="checkbox"/>	trusted	<input type="checkbox"/>	

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.

Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	

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.

Name	FQDN or IP Address	Type	Notes
CM-TG5	10.64.91.75	CM	Trunk Group 5 - ATT IPFR

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**.

SIP Entities				
13 Items				
	Name	FQDN or IP Address	Type	Notes
<input type="radio"/>	Aura Messaging	10.64.91.84	Messaging	Aura Messaging
<input type="radio"/>	Breeze	10.64.91.18	Avaya Breeze	
<input type="radio"/>	CM-TG1	10.64.91.75	CM	Trunk Group 1 - CM to Vz-IPT
<input type="radio"/>	CM-TG2	10.64.91.75	CM	Trunk Group 2 - Vz-Toll-Free inbound
<input type="radio"/>	CM-TG3	10.64.91.75	CM	Trunk Group 3 - CM to Enterprise
<input type="radio"/>	CM-TG4	10.64.91.75	CM	Trunk Group 4 - ATT IPTF
<input type="radio"/>	CM-TG5	10.64.91.75	CM	Trunk Group 5 - ATT IPFR
<input type="radio"/>	IP500	10.64.19.70	Other	IP Office
<input type="radio"/>	Presence	10.64.91.18	Presence Services	
<input type="radio"/>	SBC1	10.64.91.50	SIP Trunk	Avaya SBC-1 to PSTN
<input type="radio"/>	SBC2	10.64.91.100	SIP Trunk	Avaya SBC-2 to PSTN
<input type="radio"/>	SBCE-ATT	10.64.91.40	SIP Trunk	SBCE for AT&T testing
<input type="radio"/>	SBCE-Toll Free	10.64.91.41	SIP Trunk	SBCE for IPTF testing
Select : None				

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

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

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

Routing Policy Details
Commit
Cancel

General

Name: To CM-TG5
Disabled:
Retries: 0
Notes: Trunk Group 5 PSTN to CM

SIP Entity as Destination
Select

Name	FQDN or IP Address	Type	Notes
CM-TG5	10.64.91.75	CM	Trunk Group 5 - ATT IPFR

Time of Day
Add
Remove
View Gaps/Overlaps

1 Item
Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	

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**.

The screenshot shows the 'Global Settings' page in the Session Manager interface. The left sidebar contains navigation links: Session Manager, Dashboard, Session Manager Admin..., Global Settings (selected), Communication Profile..., Network Configuration, Device and Location..., Application Configur..., System Status, System Tools, and Performance. The main content area is titled 'Global Settings' and includes a 'Help ?' link. Below the title is a subtitle 'Administer settings that apply to all Session Managers' and buttons for 'Commit', 'Cancel', and 'View Defaults'. The settings are organized into two columns. The left column includes: 'Failback Policy' (Auto), 'Allow Unauthenticated Emergency Calls' (checked), 'ELIN SIP Entity' (None), 'Ignore SDP for Call Admission Control' (unchecked), 'Disable Call Admission Control Threshold Alarms' (unchecked), 'Disable Loop Detection Alarms' (unchecked), '*Loop Detection Alarms Threshold (hours)' (24), 'Enable Dial Plan Ranges' (unchecked), 'Enable Regular Expression Adaptations' (unchecked), 'Enable Flexible Routing' (checked and highlighted with a red box), 'Set Precedence for Routing' (Dial Patterns), 'Set Dial Patterns Precedence' (a table with 'Precedence Order' and 'Dial Patterns' columns, showing 'Destination', 'Location', and 'Origination' in order), and 'Enable Load Balancer' (unchecked). The right column includes: 'Enable IPv6' (unchecked), 'Allow Unsecured PPM Traffic' (checked), 'Minimum SIP Entity TLS Version' (1.2), 'Minimum Endpoint TLS Version' (1.0), 'TLS Endpoint Certificate Validation' (None), 'Enable End to End Secure Call Indication' (checked), 'Enable Military Support' (unchecked), 'Enable Application Sequence for Emergency Calls' (checked), 'Emergency Call Resource-Priority Headers' (empty), 'Enable Implicit Users Applications for SIP users' (checked), and 'Enable SIP Resiliency' (unchecked).

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 303555xxxx). 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**.

Dial Pattern Details [Commit] [Cancel] [Help ?](#)

General

* Pattern: 303555

* Min: 10

* Max: 10

Emergency Call: ☐

SIP Domain: avayalab.com

Notes: AT&T DIDs

Originating Locations, Origination Dial Pattern Sets, and Routing Policies

Add Remove

1 Item

Originating Location Name	Originating Location Notes	Origination Dial Pattern Set Name	Origination Dial Pattern Set Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/> Common-SBCs	SBC to PSTN			To CM-TG5	0	<input type="checkbox"/>	CM-TG5	Trunk Group 5 PSTN to CM

Select : All, None

Denied Originating Locations and Origination Dial Pattern Sets

Add Remove

0 Items

Originating Location	Notes	Origination Dial Pattern Set Name	Origination Dial Pattern Set Notes
----------------------	-------	-----------------------------------	------------------------------------

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**.

Originating Location

SelectCancel

Originating Location

☐ Apply The Selected Routing Policies to All Originating Locations

5 Items

Filter: Enable

<input type="checkbox"/>	Name	Notes
<input type="checkbox"/>	CM-TG-5	CM-TG-5
<input checked="" type="checkbox"/>	Common-SBCs	SBC to PSTN
<input type="checkbox"/>	Main	Avaya SIL
<input type="checkbox"/>	RemoteAccess	Remote Access from SBCE1

Select : All, None

Origination Dial Pattern Sets

1 Item

Filter: Enable

Name	Notes
------	-------

Routing Policies

14 Items

Filter: Enable

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	To AAM	<input type="checkbox"/>	Aura Messaging	
<input type="checkbox"/>	To CM TG1	<input type="checkbox"/>	CM-TG1	Trunk Group 1 PSTN1 to CM
<input type="checkbox"/>	To CM TG2	<input type="checkbox"/>	CM-TG2	Trunk Group 2 VzIPCC to CM
<input type="checkbox"/>	To CM TG3	<input type="checkbox"/>	CM-TG3	Enterprise Traffic
<input type="checkbox"/>	To CM TG4	<input type="checkbox"/>	CM-TG4	Trunk Group 4 PSTN4 to CM
<input checked="" type="checkbox"/>	To CM-TG5	<input type="checkbox"/>	CM-TG5	Trunk Group 5 PSTN to CM
<input type="checkbox"/>	To CM TG7	<input type="checkbox"/>	CM-TG7	Incoming calls from Masergy
<input type="checkbox"/>	To Experience Portal	<input type="checkbox"/>	ExperiencePortal	
<input type="checkbox"/>	To IP500	<input type="checkbox"/>	IP500	

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.

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SPOC 11/27/2019

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Au81SBC8EP-IPFR

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 1xxxxyyxxxx, 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**).

Dial Pattern Details Commit Cancel Help ?

General

* **Pattern:** +

* **Min:** 10

* **Max:** 36

Emergency Call: ☐

SIP Domain: avayalab.com

Notes: E.164 Public Numbers

Originating Locations, Origination Dial Pattern Sets, and Routing Policies

Add Remove

8 Items Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Origination Dial Pattern Set Name	Origination Dial Pattern Set Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	CM-TG-5	CM-TG-5			To SBCE-ATT	0	<input type="checkbox"/>	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.

Dial Patterns

NewEditDeleteDuplicateMore Actions

43 Items

Filter: Enable

<input type="checkbox"/>	Pattern	Min	▲	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
<input type="checkbox"/>	0	1		36	<input type="checkbox"/>			avayalab.com	0+ NANPA
<input type="checkbox"/>	*7	2		36	<input type="checkbox"/>			-ALL-	ATT -IPflex feature code
<input type="checkbox"/>	*9	2		36	<input type="checkbox"/>			-ALL-	ATT -IPflex feature code
<input type="checkbox"/>	x11	3		3	<input type="checkbox"/>			avayalab.com	Outbound Services
<input type="checkbox"/>	911	3		3	<input checked="" type="checkbox"/>	All Emergency	1	-ALL-	
<input type="checkbox"/>	9999	4		36	<input type="checkbox"/>			-ALL-	
<input type="checkbox"/>	1411	4		4	<input type="checkbox"/>			avayalab.com	Outbound PSTN Information
<input type="checkbox"/>	15555	5		5	<input checked="" type="checkbox"/>	test EMERG	1	-ALL-	Test emergency outbound
<input type="checkbox"/>	12xxx	5		5	<input type="checkbox"/>			-ALL-	Enterprise Extensions
<input type="checkbox"/>	11000	5		5	<input type="checkbox"/>			-ALL-	Messaging Pilot number
<input type="checkbox"/>	7	5		5	<input type="checkbox"/>			-ALL-	CM VDNs
<input type="checkbox"/>	89	5		5	<input type="checkbox"/>			-ALL-	Enterprise Extensions
<input type="checkbox"/>	50	5		5	<input type="checkbox"/>			-ALL-	Enterprise Extensions
<input type="checkbox"/>	14xxx	5		5	<input type="checkbox"/>			-ALL-	Enterprise Extensions
<input type="checkbox"/>	5551212	7		7	<input type="checkbox"/>			avayalab.com	Outbound Directory Service

Select : All, None

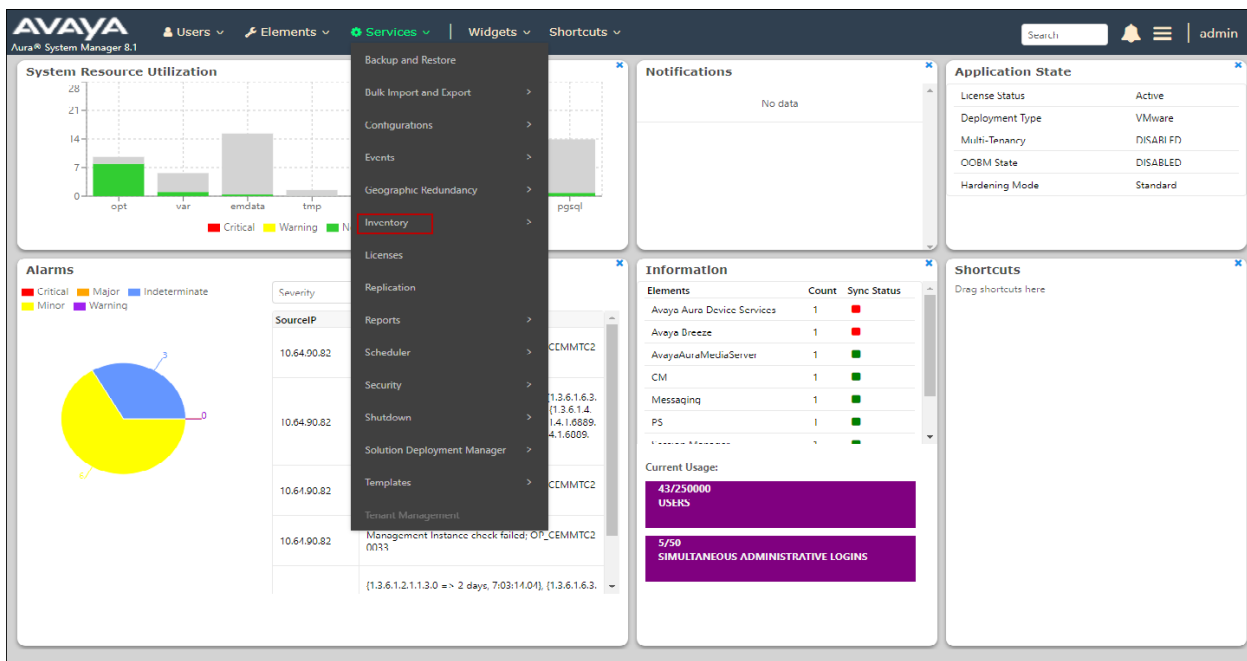
Page 1 of 3

6.10. Verify TLS Certificates – Session Manager

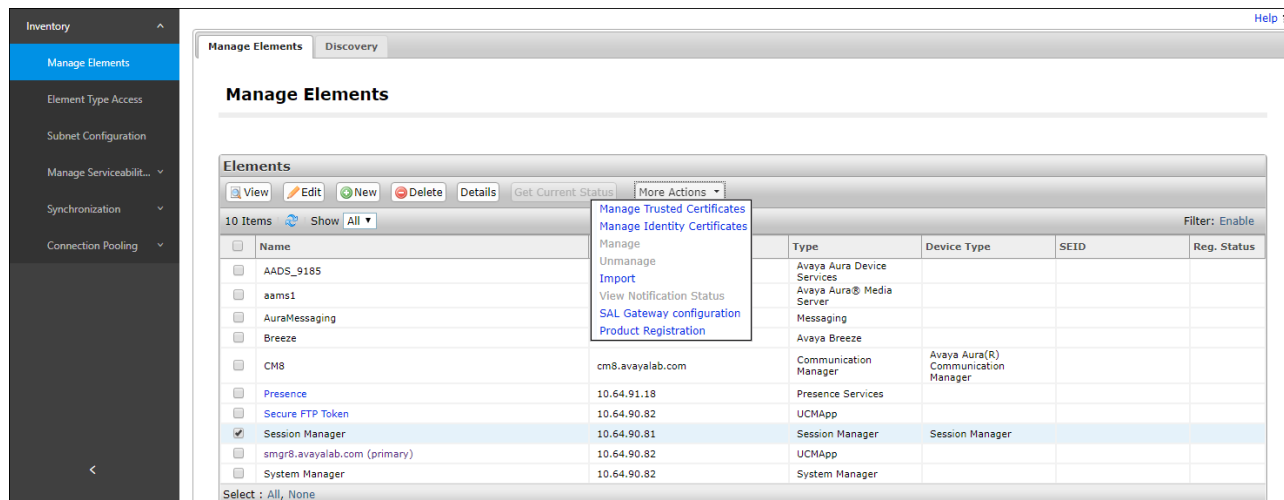
Note – Testing was done with System Manager signed identity certificates. The procedure to obtain and install certificates is outside the scope of these Application Notes.

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

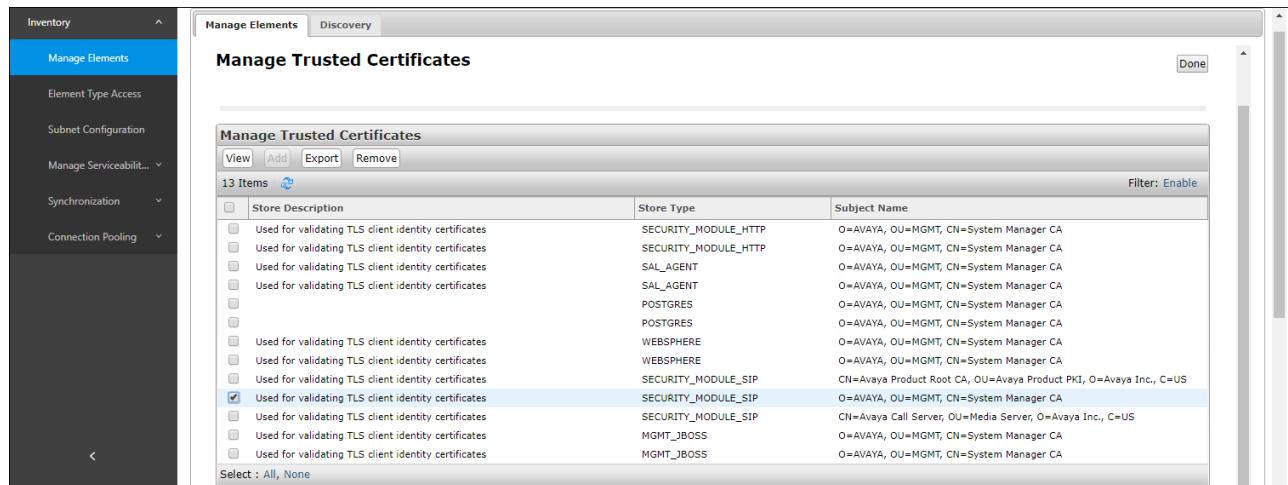
Step 1 - From the **Home** screen, under the **Services** heading, select **Inventory**.



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**.

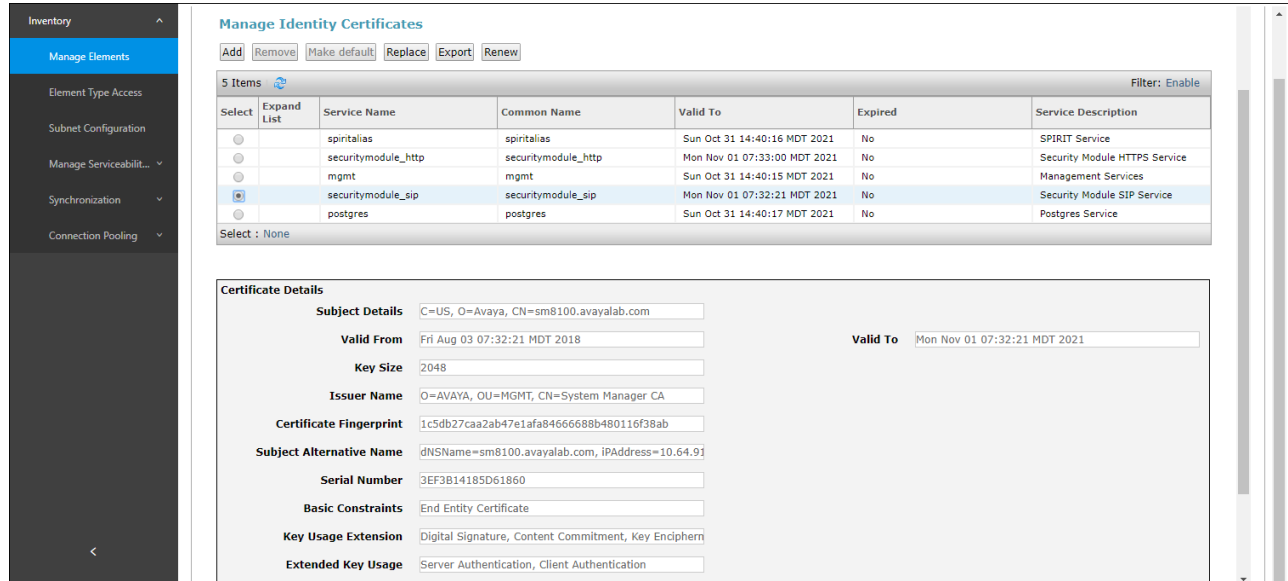


Step 3 - Verify the System Manager Certificate Authority certificate is listed in the trusted store, **SECURITY_MODULE_SIP**. Click **Done** to return to the previous screen.



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 call³.

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.

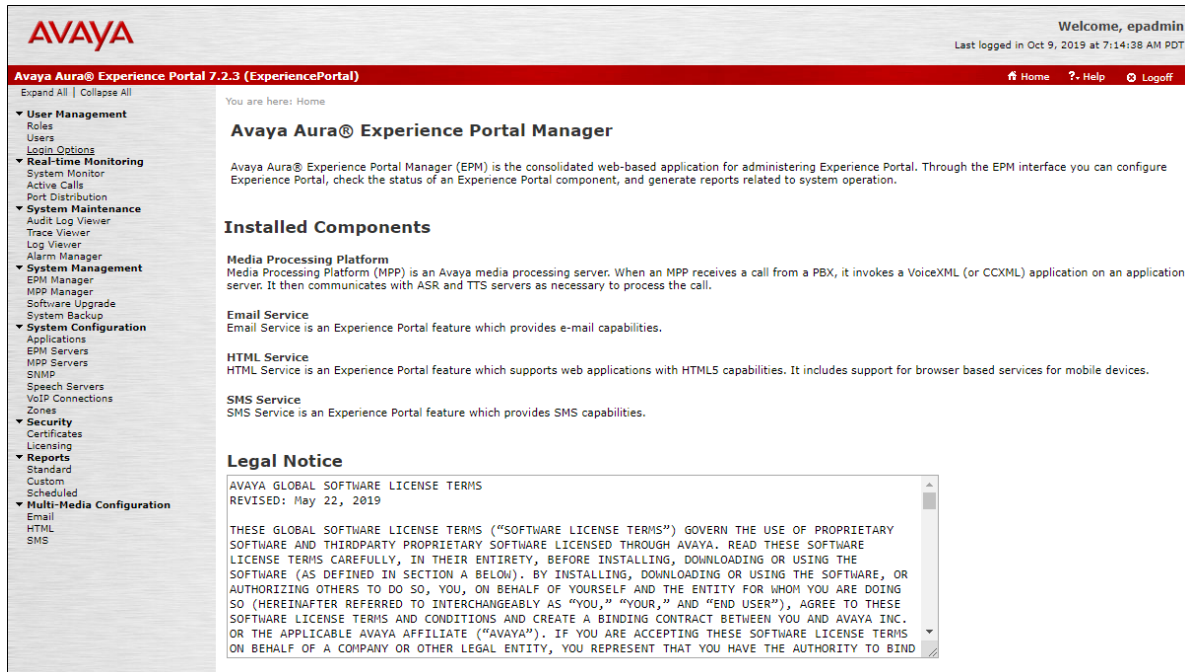
³ An application may be configured with “inbound default” as the called number, to process all inbound calls that do not match any other application references.

7.2. Logging In and Licensing

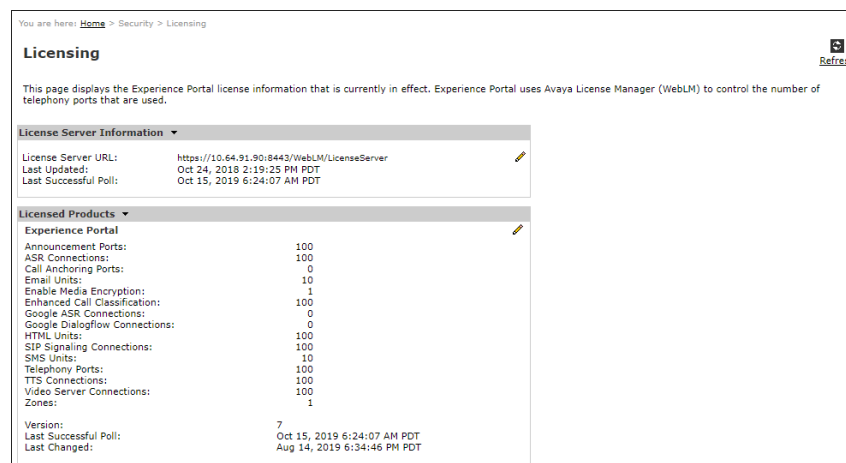
This section describes the steps on Experience Portal for administering a SIP connection to the Session Manager.

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.

Expand All | Collapse All

You are here: [Home](#) > System Configuration > VoIP Connections

VoIP Connections

This page displays a list of Voice over Internet Protocol (VoIP) servers that Experience Portal communicates with. You can configure multiple SIP connections, but only one SIP connection can be enabled at any one given time.

H.323 SIP

<input type="checkbox"/>	Name	Enable	Proxy Transport	Proxy/DNS Server Address	Proxy Server Port	Listener Port	SIP Domain	Maximum Simultaneous Calls
<input type="checkbox"/>	SM8	Yes	TLS	10.64.91.81	5061	5061	avayalab.com	10

Add **Delete** **Help**

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**.

Expand All | Collapse All

- ▼ User Management
 - Roles
 - Users
 - Login Options
- ▼ Real-time Monitoring
 - System Monitor
 - Active Calls
 - Port Distribution
- ▼ System Maintenance
 - Audit Log Viewer
 - Trace Viewer
 - Log Viewer
 - Alarm Manager
- ▼ System Management
 - EPM Manager
 - MPP Manager
 - Software Upgrade
 - System Backup
- ▼ System Configuration
 - Applications
 - EPM Servers
 - MPP Servers
 - SNMP
 - Speech Servers
 - VoIP Connections
 - Zones
- ▼ Security
 - Certificates
 - Licensing
- ▼ Reports
 - Standard
 - Custom
 - Scheduled
- ▼ Multi-Media Configuration
 - Email
 - HTML
 - SMS

You are here: [Home](#) > System Configuration > [VoIP Connections](#) > Change SIP Connection

Change SIP Connection

Use this page to change the configuration of a SIP connection.

Name: SM8

Enable: ☒ Yes ☐ No

Proxy Transport: TLS ▼

☒ Proxy Servers ☐ DNS SRV Domain

Address	Port	Priority	Weight	
10.64.91.81	5061	0	0	Remove

[Additional Proxy Server](#)

Listener Port: 5061

SIP Domain: avayalab.com

P-Asserted-Identity:

Maximum Redirection Attempts: 2

Consultative Transfer: ☒ INVITE with REPLACES ☐ REFER

SIP Reject Response Code: ☒ ASM (503) ☐ SES (480) ☐ Custom 503

SIP Timers

T1: 250 milliseconds

T2: 2000 milliseconds

B and F: 4000 milliseconds

Call Capacity

Maximum Simultaneous Calls: 10

☒ All Calls can be either inbound or outbound

☐ Configure number of inbound and outbound calls allowed

SRTP

Enable: ☒ Yes ☐ No

Encryption Algorithm: ☒ AES_CM_128 ☐ NONE

Authentication Algorithm: ☒ HMAC_SHA1_80 ☐ HMAC_SHA1_32

RTCP Encryption Enabled: ☐ Yes ☒ No

RTP Authentication Enabled: ☒ Yes ☐ No

[Add](#)

Configured SRTP List

<No SRTP List>

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.

Expand All | Collapse All

You are here: [Home](#) > System Configuration > [Speech Servers](#)

Speech Servers

This page displays the list of Automated Speech Recognition (ASR) and Text-to-Speech (TTS) servers that Experience Portal communicates with.

[ASR](#) [TTS](#)

Name	Enable	Network Address	Engine Type	MRCP	Base Port	Total Number of Licensed ASR Resources	Languages
LVASR	Yes	10.64.101.83	LumenVox	MRCP V2 TCP	5060	10	en-US

[Add](#) [Delete](#) [Customize](#) [Help](#)

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.

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [Applications](#) > Change Application

Change Application

Use this page to change the configuration of an application.

Name: Test-ccxml

Enable: ☒ Yes ☐ No

Type:

Reserved SIP Calls: ☒ None ☐ Minimum ☐ Maximum

Requested:

URI

☒ Single ☐ Fail Over ☐ Load Balance

CCXML URL: [Verify](#)

Mutual Certificate Authentication: ☐ Yes ☒ No

Basic Authentication: ☐ Yes ☒ No

ASR Speech Servers [▶](#)

TTS Speech Servers [▶](#)

Application Launch [▼](#)

☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number: [Add](#)

[Remove](#)

SIP Header Source:

Speech Parameters [▶](#)

Reporting Parameters [▶](#)

Advanced Parameters [▶](#)

[Save](#) [Apply](#) [Cancel](#) [Help](#)

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**.

Expand All | Collapse All

You are here: [Home](#) > System Configuration > MPP Servers

MPP Servers

This page displays the list of Media Processing Platform (MPP) servers in the Experience Portal system. When an MPP receives a call from a PBX, it invokes a VoiceXML application on an application server and communicates with ASR and TTS servers as necessary to process the call.

<input type="checkbox"/>	Name	Host Address	Network Address (VoIP)	Network Address (MRCP)	Network Address (AppSvr)	Maximum Simultaneous Calls	Trace Level
<input type="checkbox"/>	mpp1	10.64.91.90	<Default>	<Default>	<Default>	11	Use MPP Settings

Add **Delete**

MPP Settings **Browser Settings** **Video Settings** **VoIP Settings** **Help**

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**.

Expand All | Collapse All

You are here: [Home](#) > System Configuration > [MPP Servers](#) > Change MPP Server

Change MPP Server

Use this page to change the configuration of an MPP. Take care when changing the MPP Trace Logging Thresholds. Do not set Trace Levels to Finest if your Experience Portal system has heavy call traffic. The system might experience performance issues if Trace Levels are set to Finest. Set Trace Levels to Finest only when you are troubleshooting the system.

Name: mpp1
Host Address: 10.64.91.90
Network Address (VoIP): <Default>
Network Address (MRCP): <Default>
Network Address (AppSvr): <Default>
Maximum Simultaneous Calls: 11
Restart Automatically: ☒ Yes ☐ No

MPP Certificate

Owner: CN=ep.avayalab.com,O=Avaya,OU=EPM
Issuer: CN=ep.avayalab.com,O=Avaya,OU=EPM
Serial Number: 89f44cd176674542
Signature Algorithm: SHA256withRSA
Valid from: October 17, 2018 11:03:28 AM PDT until October 14, 2028 11:03:28 AM PDT
Certificate Fingerprints
MD5: dd:26:1a:d3:d1:62:d3:04:55:40:1b:98:0b:38:44:46
SHA: 4d:26:ba:2f:55:8d:3b:5f:8e:d0:6f:ee:7f:48:49:22:38:79:ae:bf
SHA-256: 17:6d:d2:9a:9b:ee:e3:35:da:67:c2:99:38:e6:14:03:c7:84:1d:94:a9:a0:f9:ac:66:57:da:28:43:59:ae:c7
Subject Alternative Names
DNS Name: ep
DNS Name: ep.avayalab.com
IP Address: 10.64.91.90

Categories and Trace Levels ▶

Save **Apply** **Cancel** **Help**

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.

Expand All | Collapse All

▶ User Management
▶ Real-time Monitoring
▶ System Maintenance
▶ System Management
▼ System Configuration
Applications
EPM Servers
MPP Servers
SNMP
Speech Servers
VoIP Connections
Zones
▶ Security
▶ Reports
▶ Multi-Media Configuration

You are here: [Home](#) > [System Configuration](#) > [MPP Servers](#) > VoIP Settings

VoIP Settings

Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols such as H.323 and Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network. Note that if you make any changes to this page, you must restart all MPPs.

Port Ranges ▼

	Low	High
UDP:	11000	30999
TCP:	31000	33499
MRCP:	34000	36499
H.323 Station:	37000	39499

RTCP Monitor Settings ▼

Host Address:

Port:

VoIP Audio Formats ▼

MPP Native Format:

- 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**.

Station:

RTCP Monitor Settings ▼

Host Address:

Port:

VoIP Audio Formats ▼

MPP Native Format:

Codecs ▼

Offer

Enable	Codec	Order
<input checked="" type="checkbox"/>	G729	1
<input checked="" type="checkbox"/>	G711uLaw	2
<input checked="" type="checkbox"/>	G711aLaw	3

Packet Time: milliseconds

G729 Discontinuous Transmission: ☐ Yes ☒ No

Answer

Enable	Codec	Order
<input checked="" type="checkbox"/>	G711uLaw	1
<input checked="" type="checkbox"/>	G711aLaw	1
<input checked="" type="checkbox"/>	G729	1

G729 Discontinuous Transmission: ☐ Yes ☐ No ☒ Either

G729 Reduced Complexity Encoder: ☒ Yes ☐ No

QoS Parameters ▼

	VLAN	Diffserv
H.323:	6	46
SIP:	6	46
RTSP:	6	46

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.

The screenshot shows the Experience Portal MPP Manager GUI. The left sidebar contains a navigation menu with categories like User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration, Security, Reports, and Multi-Media Configuration. The main content area is titled "MPP Manager (Oct 15, 2019 7:30:39 AM PDT)" and includes a "Refresh" button. Below the title, there is a table with columns: Server Name, Mode, State, Config, Auto Restart, Restart Schedule, and Active Calls. The table shows one MPP, mpp1, which is Online Running. Below the table, there are sections for State Commands (Start, Stop, Restart, Reboot, Halt, Cancel) and Mode Commands (Offline, Test, Online). A "Help" button is also present.

Server Name	Mode	State	Config	Auto Restart	Restart Schedule	Active Calls		
					Today	Recurring	In	Out
mpp1	Online	Running	OK	Yes	No	None	0	0

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 screenshot shows the Avaya Session Border Controller for Enterprise login interface. On the left, the Avaya logo is displayed in red, with the text "Session Border Controller for Enterprise" below it. On the right, there is a "Log In" section with fields for "Username:" (containing "ucsec") and "Password:" (containing masked characters). A "Log In" button is positioned below the password field. Below the login fields, a "WELCOME TO AVAYA SBC" message is displayed, followed by a disclaimer: "Unauthorized access to this machine is prohibited. This system is for the use authorized users only. Usage of this system may be monitored and recorded by system personnel." Below this, a consent statement reads: "Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible evidence of criminal activity, system personnel may provide the evidence from such monitoring to law enforcement officials." At the bottom, the copyright notice "© 2011 - 2019 Avaya Inc. All rights reserved." is visible.

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.

The screenshot shows the Avaya Session Border Controller for Enterprise EMS Dashboard. The top navigation bar includes links for Device: EMS, Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" with the AVAYA logo. The left sidebar lists "EMS Dashboard" with sub-items: Device Management, System Administration, Backup/Restore, and Monitoring & Logging. The main content area is titled "Dashboard" and contains several sections:

- Information:** A table showing system details.

System Time	09:09:09 AM MDT	Refresh
Version	8.0.1.0-10-17555	
Build Date	Tue Jul 30 22:53:51 UTC 2019	
License State	OK	
Aggregate Licensing Overages	0	
Peak Licensing Overage Count	0	
Last Logged in at	10/14/2019 06:28:47 MDT	
Failed Login Attempts	0	
- Installed Devices:** A list showing "EMS" and "SBCE8-70".
- Active Alarms (past 24 hours):** "None found."
- Incidents (past 24 hours):** "None found."
- Notes:** "No notes found."

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.

The screenshot shows the Avaya Session Border Controller for Enterprise EMS Dashboard with the "Device Management" section selected in the left sidebar. The main content area is titled "Device Management" and contains a tabbed interface with "Devices", "Updates", "SSL VPN", "Licensing", and "Key Bundles". The "Devices" tab is active, displaying a table of installed devices:

Device Name	Management IP	Version	Status						
SBCE8-70	10.64.90.70	8.0.1.0-10-17555	Commissioned	Reboot	Shutdown	Restart Application	View	Edit	Uninstall

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.

System Information: SBCE8-70

General Configuration

Appliance Name SBCE8-70
Box Type SIP
Deployment Mode Proxy

Device Configuration

HA Mode No
Two Bypass Mode No

Dynamic License Allocation

	Min License Allocation	Max License Allocation
Standard Sessions	10	100
Advanced Sessions	10	100
Scopia Video Sessions	10	100
CES Sessions	10	100
Transcoding Sessions	10	100
CLID	---	
Encryption	Available: Yes <input checked="" type="checkbox"/>	

Network Configuration

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.64.91.40	10.64.91.40	255.255.255.0	10.64.91.1	A1
10.64.91.41	10.64.91.41	255.255.255.0	10.64.91.1	A1
				B1
192.168.200.26	192.168.200.26	255.255.255.248	192.168.200.25	B2

DNS Configuration

Primary DNS 10.64.19.201
Secondary DNS
DNS Location DMZ
DNS Client IP 10.64.91.40

Management IP(s)

IP #1 (IPv4) 10.64.90.70

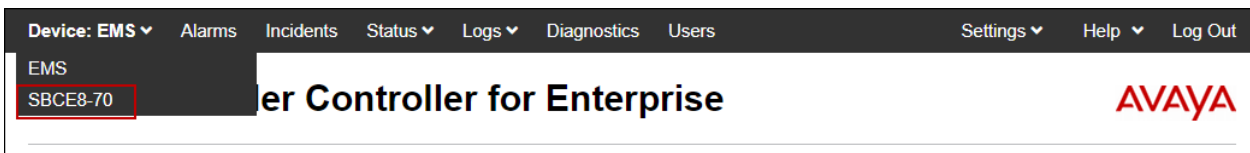
8.2. TLS Management

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.

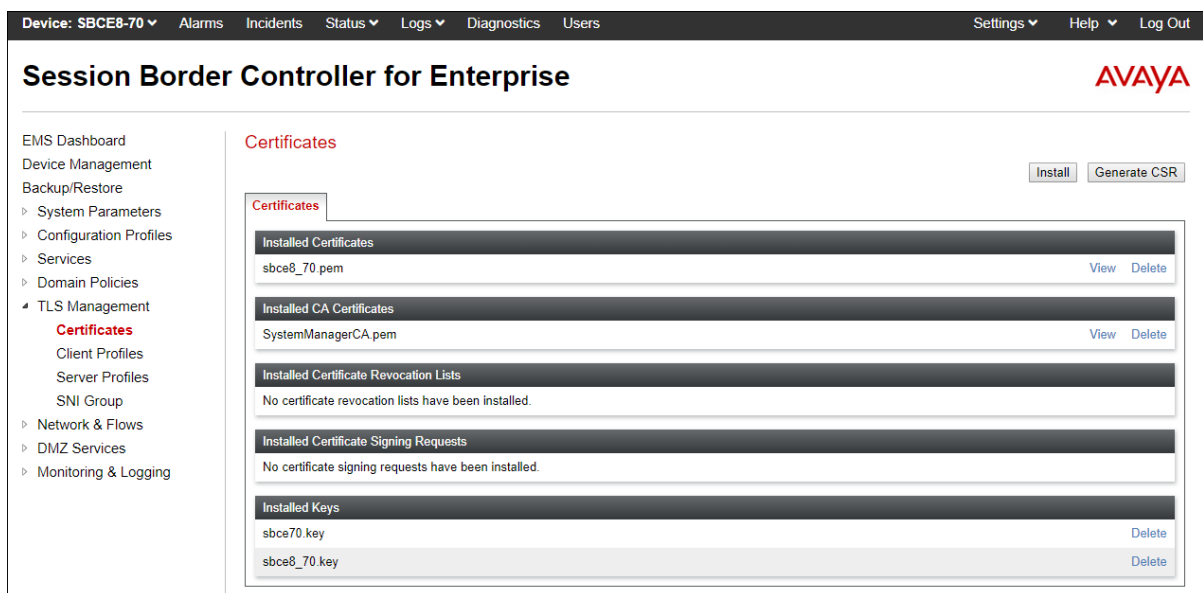
8.2.1. Verify TLS Certificates – Avaya Session Border Controller for Enterprise

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 'Edit Profile' dialog box contains a warning message at the top: "WARNING: Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems." Below the warning, the 'TLS Profile' section includes fields for 'Profile Name' (sbce8_70Server), 'Certificate' (sbce8_70.pem), 'SNI Options' (None), and 'SNI Group' (None). The 'Certificate Verification' section includes 'Peer Verification' (None), 'Peer Certificate Authorities' (SystemManagerCA.pem), 'Peer Certificate Revocation Lists' (empty), and 'Verification Depth' (0). A 'Next' button is at the bottom right.

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

The 'Session Border Controller for Enterprise' interface shows the 'Server Profiles: sbce8_70Server' section. The 'Server Profile' form is displayed with the following details: 'Profile Name' (sbce8_70Server), 'Certificate' (sbce8_70.pem), 'SNI Options' (None), 'Peer Verification' (None), 'Extended Hostname Verification' (unchecked), 'Renegotiation Time' (0), 'Renegotiation Byte Count' (0), 'Version' (TLS 1.2 selected), 'Ciphers' (Default selected), and 'Value' (HIGH:IDH:ADH:IMD5:1aNULL:1eNULL:@STRENGTH). The 'Edit' button is at the bottom right of the 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 'Edit Profile' dialog box shows the configuration for a TLS Client Profile. At the top, a warning message states: 'WARNING: Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems.' The 'TLS Profile' section includes fields for 'Profile Name' (sbce8_70Client), 'Certificate' (sbce8_70.pem), and 'SNI' (Enabled). The 'Certificate Verification' section includes 'Peer Verification' (Required), 'Peer Certificate Authorities' (SystemManagerCA.pem), 'Peer Certificate Revocation Lists' (empty), 'Verification Depth' (1), 'Extended Hostname Verification' (disabled), and 'Server Hostname' (empty). A 'Next' button is at the bottom.

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

The 'Session Border Controller for Enterprise' interface shows the 'Client Profiles' section. The profile 'sbce8_70Client' is selected. The 'Client Profile' form is displayed with the following details: 'TLS Profile' (Profile Name: sbce8_70Client, Certificate: sbce8_70.pem, SNI: Enabled), 'Certificate Verification' (Peer Verification: Required, Peer Certificate Authorities: SystemManagerCA.pem, Peer Certificate Revocation Lists: ---, Verification Depth: 1, Extended Hostname Verification: disabled), 'Renegotiation Parameters' (Renegotiation Time: 0, Renegotiation Byte Count: 0), and 'Handshake Options' (Version: TLS 1.2, TLS 1.1, TLS 1.0; Ciphers: Default, FIPS, Custom; Value: HIGH IDH IADH IMD5 IaNULL IaNULL @STRENGTH). An 'Edit' button is at the bottom.

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**.

The screenshot shows the 'Session Border Controller for Enterprise' interface. On the left is a navigation menu with 'Network Management' highlighted. The main area is titled 'Network Management' and has two tabs: 'Interfaces' (selected) and 'Networks'. Below the tabs is a table with columns 'Interface Name', 'VLAN Tag', and 'Status'. The table lists four interfaces: A1 (Enabled), A2 (Disabled), B1 (Enabled), and B2 (Enabled). An 'Add VLAN' button is in the top right corner.

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

The screenshot shows the 'Session Border Controller for Enterprise' interface with the 'Networks' tab selected. The table displays network configurations with columns: Name, Gateway, Subnet Mask / Prefix Length, Interface, and IP Address. It lists three networks: 'Inside-A1', 'Inside-B1', and 'Outside-B2'. Each row has 'Edit' and 'Delete' links. An 'Add' button is in the top right corner.

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	
Inside-A1	10.64.91.1	255.255.255.0	A1	10.64.91.40, 10.64.91.41	Edit Delete
Inside-B1	192.168.200.1	255.255.255.248	B1	192.168.200.25	Edit Delete
Outside-B2	192.168.200.25	255.255.255.248	B2	192.168.200.26	Edit Delete

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.**).

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left-hand navigation menu is expanded to 'Network & Flows', and 'Advanced Options' is selected. The main content area is titled 'Advanced Options' and contains several tabs: 'Periodic Statistics', 'Feature Control', 'SIP Options', 'Network Options', 'Port Ranges' (which is active), 'RTCP Monitoring', and 'Load Monitoring'. Below the tabs, a warning message states: 'Changes to the settings below require an application restart before taking effect. Application restarts can be issued from Device Management.' The 'Port Range Configuration' section contains four rows, each with a label and a range input field:

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

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

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.

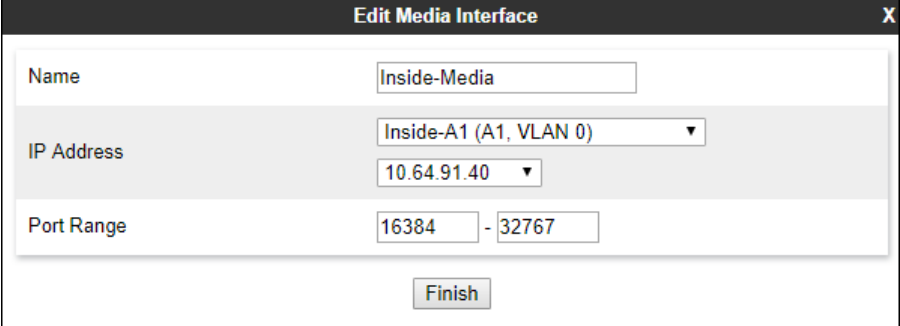
Some ports in the range required by AT&T were already allocated by the Avaya SBCE for internal use, by default. **Section 8.4** shows the steps required to reallocate the port ranges used by the Avaya SBCE, so the range required by AT&T could be accommodated.

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**.



The screenshot shows the 'Edit Media Interface' window with the following configuration:

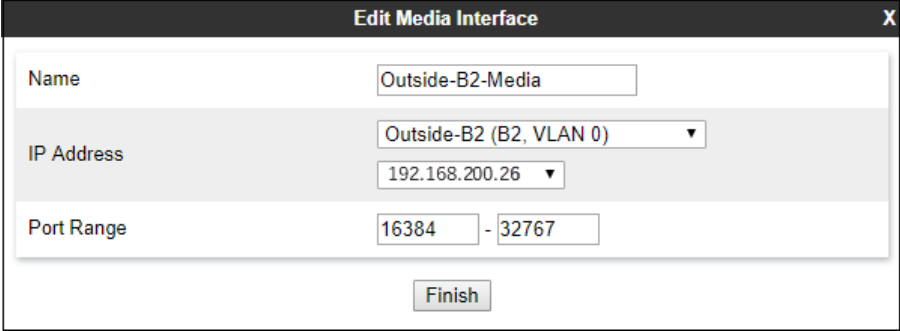
Field	Value
Name	Inside-Media
IP Address	Inside-A1 (A1, VLAN 0) 10.64.91.40
Port Range	16384 - 32767

A 'Finish' button is located at the bottom right of the form.

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**.



The screenshot shows the 'Edit Media Interface' window with the following configuration:

Field	Value
Name	Outside-B2-Media
IP Address	Outside-B2 (B2, VLAN 0) 192.168.200.26
Port Range	16384 - 32767

A 'Finish' button is located at the bottom right of the form.

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**.

The screenshot shows the 'Edit Signaling Interface' window with the following configuration:

Field	Value
Name	Inside-Sig-40
IP Address	Inside-A1 (A1, VLAN 0) 10.64.91.40
TCP Port	Leave blank to disable
UDP Port	Leave blank to disable
TLS Port	5061
TLS Profile	sbce8_70Server
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

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**.

The screenshot shows the 'Edit Signaling Interface' window with the following configuration:

Field	Value
Name	Outside-B2-Signaling
IP Address	Outside-B2 (B2, VLAN 0) 192.168.200.26
TCP Port	Leave blank to disable
UDP Port	5060
TLS Port	Leave blank to disable
TLS Profile	None
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

Finish

8.7. Server Interworking Profiles

The Server Interworking 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.

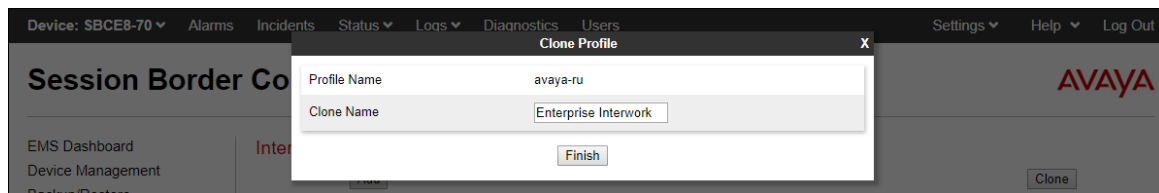
8.7.1. Server Interworking Profile – Enterprise

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.

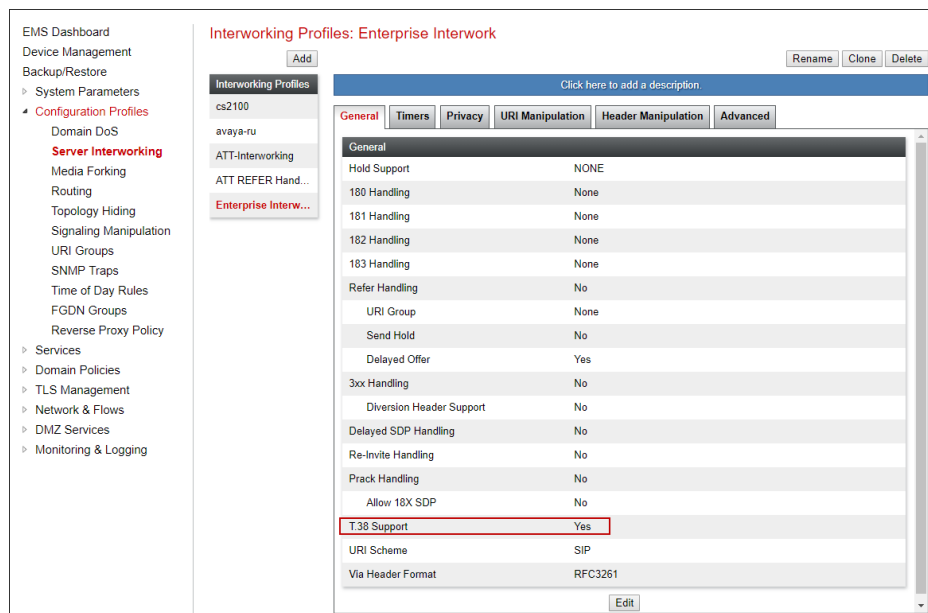
Step 3 - Enter profile name: (e.g., **Enterprise Interwork**), and click **Finish** to continue.



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.

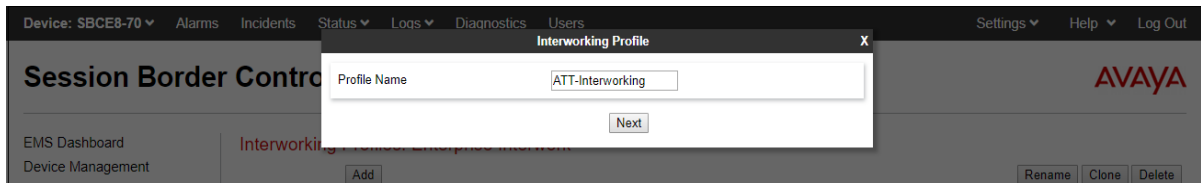
- Check **T38 Support**.
- All other options can be left with default values. Click **Finish** (not shown).



8.7.2. Server Interworking – AT&T

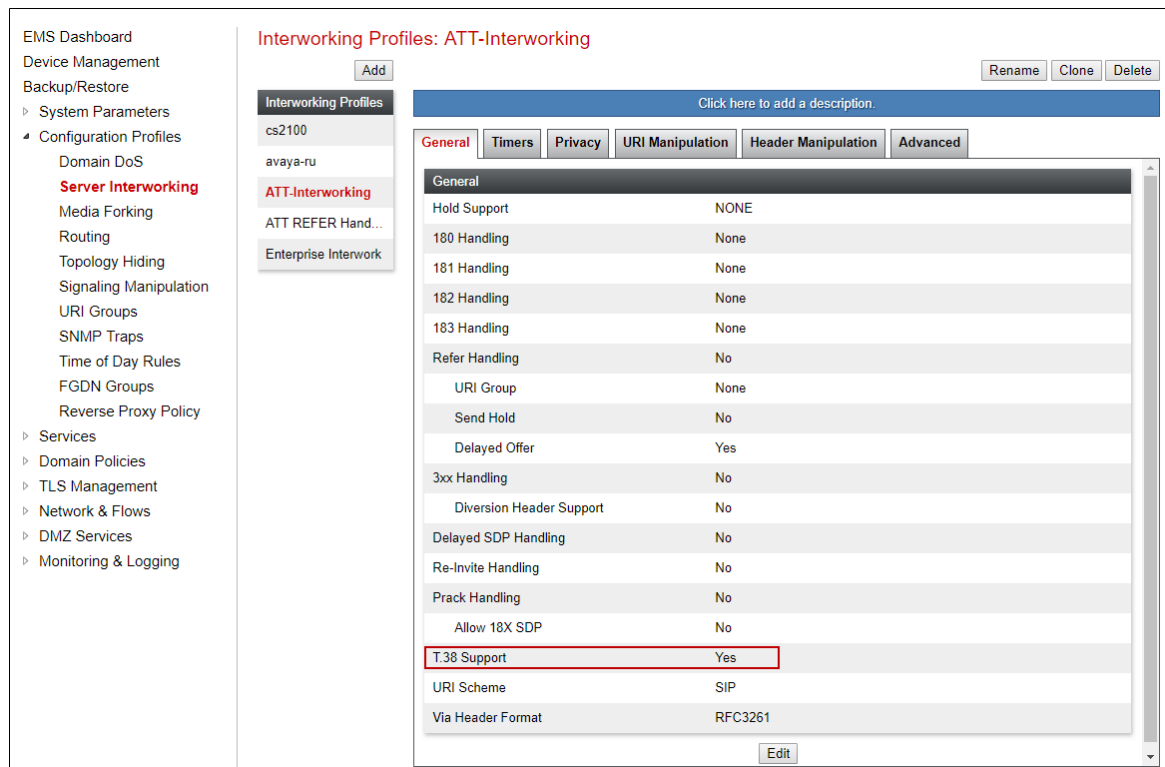
Repeat the steps shown in **Section Error! Reference source not found.** to add an Interworking Profile for the connection to AT&T via the public network, with the following changes:

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.

The screenshot shows the 'Interworking Profiles: ATT-Interworking' configuration page. On the left, a sidebar lists 'Interworking Profiles' with options: 'cs2100', 'avaya-ru', 'ATT-Interworking' (highlighted in red), 'ATT REFER Handl...', and 'Enterprise Interwork'. The main area has tabs for 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'Timers' tab is active, showing a table of SIP Timers:

SIP Timers	
Min-SE	---
Init Timer	---
Max Timer	---
Trans Expire	4 seconds
Invite Expire	---
Retry After	---

Buttons for 'Add', 'Rename', 'Clone', 'Delete', and 'Edit' are visible.

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).

The screenshot shows the 'Interworking Profiles: ATT-Interworking' configuration page, now on the 'Advanced' tab. The sidebar is the same. The 'Advanced' tab is active, showing a table of advanced settings:

Record Routes	Both Sides
Include End Point IP for Context Lookup	No
Extensions	None
Diversion Manipulation	No
Has Remote SBC	Yes
Route Response on Via Port	No
Relay INVITE Replace for SIPREC	No
MOBX Re-INVITE Handling	No

Below this table is a section for 'DTMF' settings:

DTMF	
DTMF Support	None

Buttons for 'Add', 'Rename', 'Clone', 'Delete', and 'Edit' are visible.

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.

```

Title Script for IPFR-CM
1 within session "ALL"
2 {
3   act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
4   {
5
6     //Remove gsid and epv parameters from Contact header to hide internal topology
7     remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
8     remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);
9
10    //Remove Bandwidth from SDP
11    %BODY(1).regex_replace("b=(TAS[ASCT]):(\d+)\r\n","");
12
13    // fix call-fwd
14    %HEADERS["Diversion"][1].regex_replace("sips","sip");
15  }
16 }
17
18 //OPTIONAL - Change AT&T Max-Forwards value from 0 to 30
19 within session "OPTIONS"
20 {
21   act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="AFTER_NETWORK"
22   {
23     %HEADERS["Max-Forwards"][1] = "30";
24   }
25 }
26
27 // OPTIONAL Experience Portal - modify PAI Header
28 within session "INVITE"
29 {
30   act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
31   {
32     if (%INITIAL_REQUEST = "true") then
33

```

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

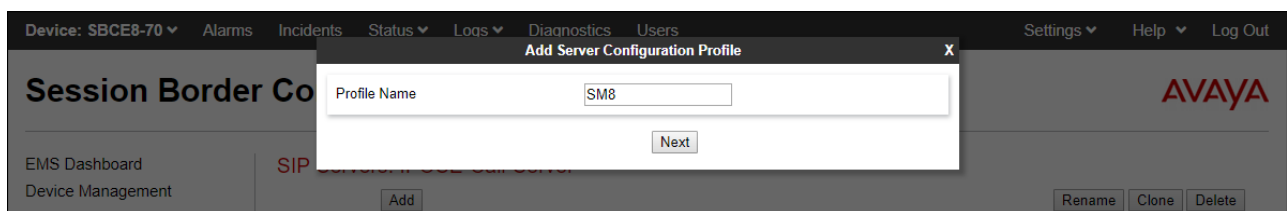
The **SIP Server Profile** contains parameters to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics, and trusted domains.

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 - GeneralX

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.

The screenshot shows a window titled "Edit SIP Server Profile - Heartbeat". It contains the following fields and controls:

- Enable Heartbeat**: A checkbox that is checked.
- Method**: A dropdown menu with "OPTIONS" selected.
- Frequency**: A text input field containing "60", followed by the label "seconds".
- From URI**: A text input field containing "sbce70@avayalab.com".
- To URI**: A text input field containing "sm@avayalab.com".
- Finish**: A button at the bottom right.

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**.

The screenshot shows a window titled "Edit SIP Server Profile - Advanced". It contains the following fields and controls:

- Enable DoS Protection**: A checkbox that is unchecked.
- Enable Grooming**: A checkbox that is checked.
- Interworking Profile**: A dropdown menu with "Enterprise Interwork" selected.
- Signaling Manipulation Script**: A dropdown menu with "None" selected.
- Securable**: A checkbox that is unchecked.
- Enable FGDN**: A checkbox that is unchecked.
- TCP Failover Port**: A text input field.
- TLS Failover Port**: A text input field.
- Tolerant**: A checkbox that is unchecked.
- URI Group**: A dropdown menu with "None" selected.
- Finish**: A button at the bottom right.

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.

Edit SIP Server Profile - General

Server Type can not be changed while this SIP Server Profile is associated to a Server Flow.

Server Type: Trunk Server

SIP Domain:

DNS Query Type: NONE/A

TLS Client Profile: None

Add

IP Address / FQDN	Port	Transport	
192.168.38.69	5060	UDP	Delete
192.168.37.149	5060	UDP	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 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.

The screenshot shows the 'Edit SIP Server Profile - Heartbeat' window. It contains the following fields and values:

Field	Value
Enable Heartbeat	<input checked="" type="checkbox"/>
Method	OPTIONS
Frequency	300 seconds
From URI	SBCE@avaya.com
To URI	IPFR@att.com

A 'Finish' button is located at the bottom right of the form.

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**.

The screenshot shows the 'Edit SIP Server Profile - Advanced' window. It contains the following fields and values:

Field	Value
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	ATT-Interworking
Signaling Manipulation Script	Script for IPFR-CM
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	
TLS Failover Port	
Tolerant	<input type="checkbox"/>
URI Group	None

A 'Finish' button is located at the bottom right of the form.

8.10. Routing Profiles

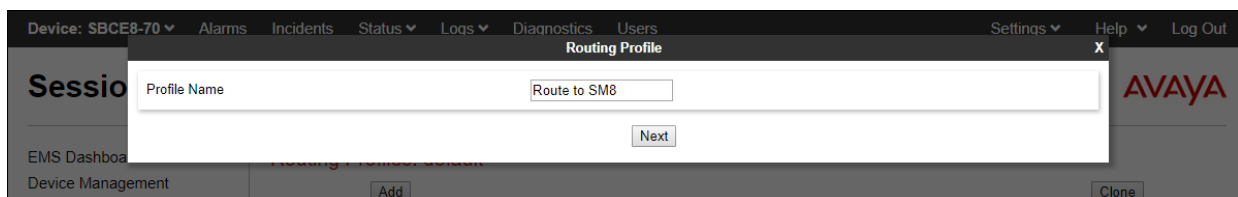
Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and determine which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport 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**.

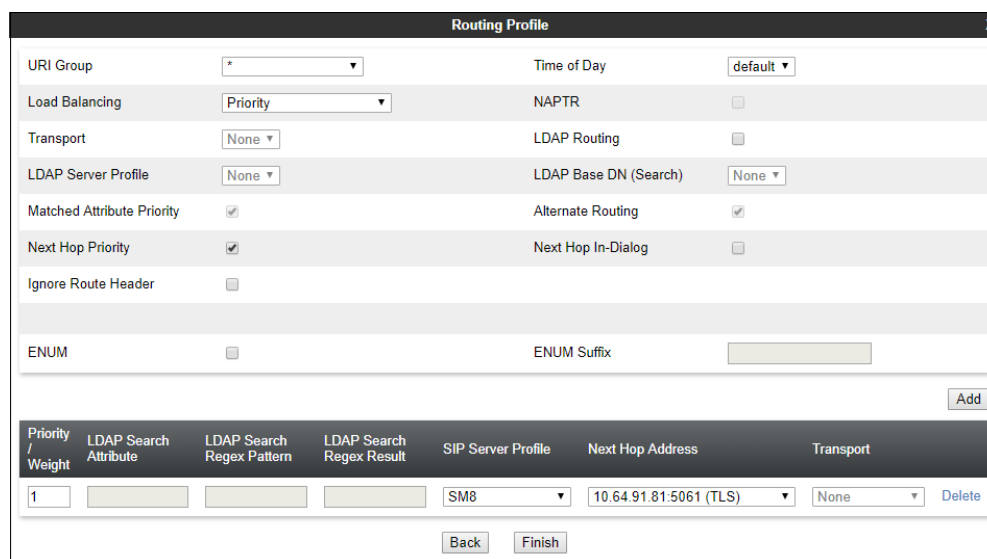
Step 2 - Enter a **Profile Name**: (e.g., **Route to SM8**) and click **Next**.

The screenshot shows the 'Routing Profile' configuration window. At the top, there's a navigation bar with 'Device: SBCE8-70' and various menu items like 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. Below this, the 'Routing Profile' window is open, showing a 'Profile Name' field with the text 'Route to SM8' and a 'Next' button. The background shows a partial view of the 'Session Manager' configuration page with 'EMS Dashboard' and 'Device Management' sections.

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**.

The screenshot shows the 'Routing Profile' configuration window with the 'Next-Hop Address' section expanded. The top section contains various settings: 'URI Group' (dropdown), 'Time of Day' (dropdown), 'Load Balancing' (dropdown), 'NAPTR' (checkbox), 'Transport' (dropdown), 'LDAP Routing' (checkbox), 'LDAP Server Profile' (dropdown), 'LDAP Base DN (Search)' (dropdown), 'Matched Attribute Priority' (checkbox), 'Alternate Routing' (checkbox), 'Next Hop Priority' (checkbox), 'Next Hop In-Dialog' (checkbox), 'Ignore Route Header' (checkbox), 'ENUM' (checkbox), and 'ENUM Suffix' (text field). Below this is a table with columns: 'Priority / Weight', 'LDAP Search Attribute', 'LDAP Search Regex Pattern', 'LDAP Search Regex Result', 'SIP Server Profile', 'Next Hop Address', and 'Transport'. The first row shows '1', empty fields, 'SM8', '10.64.91.81:5061 (TLS)', and 'None'. At the bottom are 'Back' and 'Finish' buttons.

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**.

URI Group	Time of Day	Load Balancing	NAPTR	Transport	LDAP Routing	LDAP Server Profile	LDAP Base DN (Search)	Matched Attribute Priority	Alternate Routing	Next Hop Priority	Next Hop In-Dialog	Ignore Route Header	ENUM	ENUM Suffix
*	default	Priority	<input type="checkbox"/>	None	<input type="checkbox"/>	None	None	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	

Add

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport	Delete
1				ATT-trk-svr	192.168.38.69:5060 (UDP)	None	Delete
2				ATT-trk-svr	192.168.37.149:5060 (UDP)	None	Delete

Back 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.

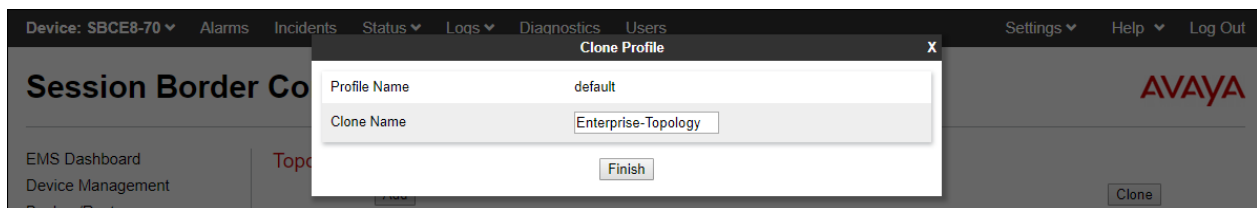
8.11.1. Topology Hiding – Enterprise

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**.

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

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**.

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

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.

Session Border Controller for Enterprise

Application Rules: sip-trunk

Application Rules

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

Miscellaneous

CDR Support: Off

RTCP Keep-Alive: No

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

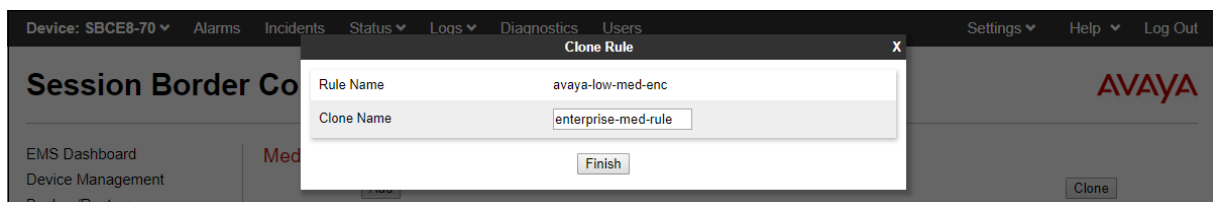
In the sample configuration, the default Media Rule **avaya-low-med-enc** was cloned to create the enterprise Media Rule, and modified as shown below:

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.

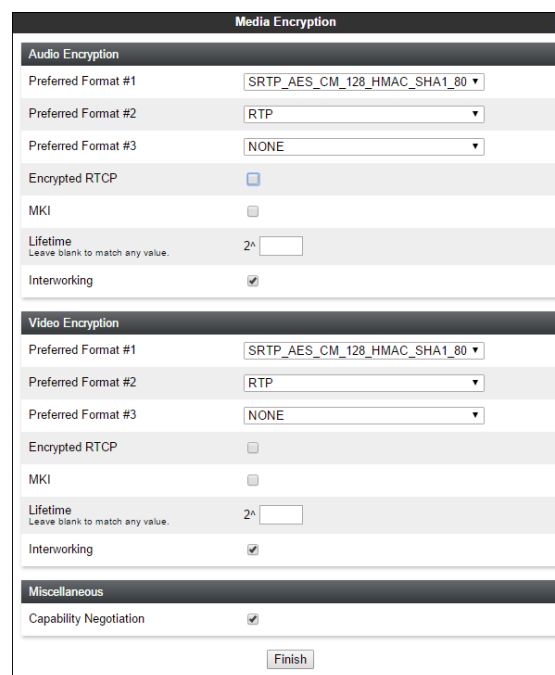
- In the **Clone Name** field enter the new Media Rule name (e.g., **enterprise-med-rule**)
- Click **Finish**. The newly created rule will be displayed.



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.

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

Device Management

Backup/Restore

System Parameters

Configuration Profiles

Services

Domain Policies

- Application Rules
- Border Rules
- Media Rules**
- Security Rules
- Signaling Rules
- Charging Rules
- End Point Policy Groups
- Session Policies

TLS Management

Network & Flows

DMZ Services

Monitoring & Logging

Media Rules: enterprise-med-rule

Add

RenameCloneDelete

Click here to add a description.

EncryptionCodec PrioritizationAdvancedQoS

Audio Encryption

Preferred Formats	SRTP_AES_CM_128_HMAC_SHA1_80 RTP
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime	Any
Interworking	<input checked="" type="checkbox"/>

Video Encryption

Preferred Formats	SRTP_AES_CM_128_HMAC_SHA1_80 RTP
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime	Any
Interworking	<input checked="" type="checkbox"/>

Miscellaneous

Capability Negotiation	<input checked="" type="checkbox"/>
------------------------	-------------------------------------

Edit

MAA:Reviewed
SPOC 11/27/2019

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Au81SBC8EP-IPFR

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.

The screenshot shows the 'Media Rules: att-med-rule' configuration page. On the left is a navigation menu with categories like EMS Dashboard, Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, Application Rules, Border Rules, Media Rules (highlighted), Security Rules, Signaling Rules, Charging Rules, End Point Policy Groups, Session Policies, TLS Management, Network & Flows, and DMZ Services. The main content area has a title 'Media Rules: att-med-rule' and an 'Add' button. Below the title is a list of media rules: default-low-med, default-low-med-enc, default-high, default-high-enc, avaya-low-med-enc, att-med-rule (highlighted), and enterprise-med-rule. The configuration area for 'att-med-rule' has tabs for Encryption, Codec Prioritization, Advanced, and QoS. The 'Encryption' tab is active, showing sections for Audio Encryption, Video Encryption, and Miscellaneous. Audio Encryption has Preferred Formats set to RTP and Interworking checked. Video Encryption also has Preferred Formats set to RTP and Interworking checked. Miscellaneous has Capability Negotiation unchecked. There are 'Rename', 'Clone', and 'Delete' buttons at the top right, and an 'Edit' button at the bottom right.

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

This screenshot shows the 'QoS' tab of the 'Media Rules: att-med-rule' configuration page. The 'QoS' tab is active, showing sections for Media QoS Marking, Audio QoS, and Video QoS. Media QoS Marking has 'Enabled' checked and 'QoS Type' set to DSCP. Audio QoS has 'Audio DSCP' set to EF. Video QoS has 'Video DSCP' set to EF. The 'Edit' button is at the bottom right. The left navigation menu is the same as in the previous screenshot, with 'Media Rules' highlighted.

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.

The screenshot displays the 'Session Border Controller for Enterprise' web interface. The left-hand navigation menu is expanded to 'Domain Policies' > 'Signaling Rules'. The main content area shows the configuration for the 'enterprise-sig-rule'. The 'General' tab is selected, showing a table of inbound and outbound signaling actions. The 'Content-Type Policy' section is also visible, showing 'Enable Content-Type Checks' is checked, and 'Action' is set to 'Allow'.

Category	Request	Response
Inbound		
Requests	Allow	
Non-2XX Final Responses		Allow
Optional Request Headers	Allow	
Optional Response Headers		Allow
Outbound		
Requests	Allow	
Non-2XX Final Responses		Allow
Optional Request Headers	Allow	
Optional Response Headers		Allow

Content-Type Policy

Field	Value
Enable Content-Type Checks	<input checked="" type="checkbox"/>
Action	Allow
Multipart Action	Allow
Exception List	Exception List

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**.

The screenshot displays the 'Session Border Controller for Enterprise' web interface. The left-hand navigation menu is expanded to 'Domain Policies' > 'Signaling Rules'. The main content area shows the configuration for the 'att-sig-rule'. The 'Signaling QoS' tab is selected, showing a table of QoS parameters. The 'QoS Type' is set to 'DSCP' and the 'DSCP' value is set to 'AF41'.

Field	Value
Signaling QoS	<input checked="" type="checkbox"/>
QoS Type	DSCP
DSCP	AF41

8.15. Endpoint Policy Groups

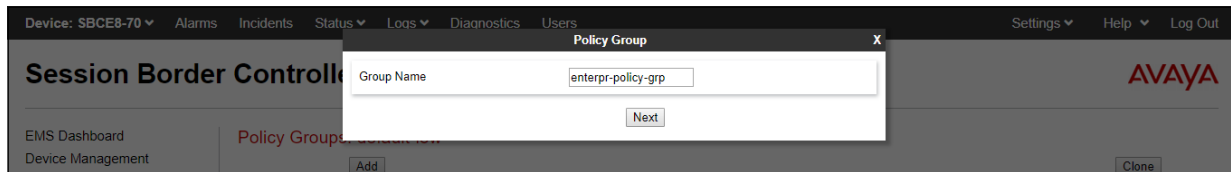
The rules created within the Domain Policies are assigned to an End Point Policy Group. The End Point Policy Group is then applied to a Server Flow in **Section 8.16**.

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**.

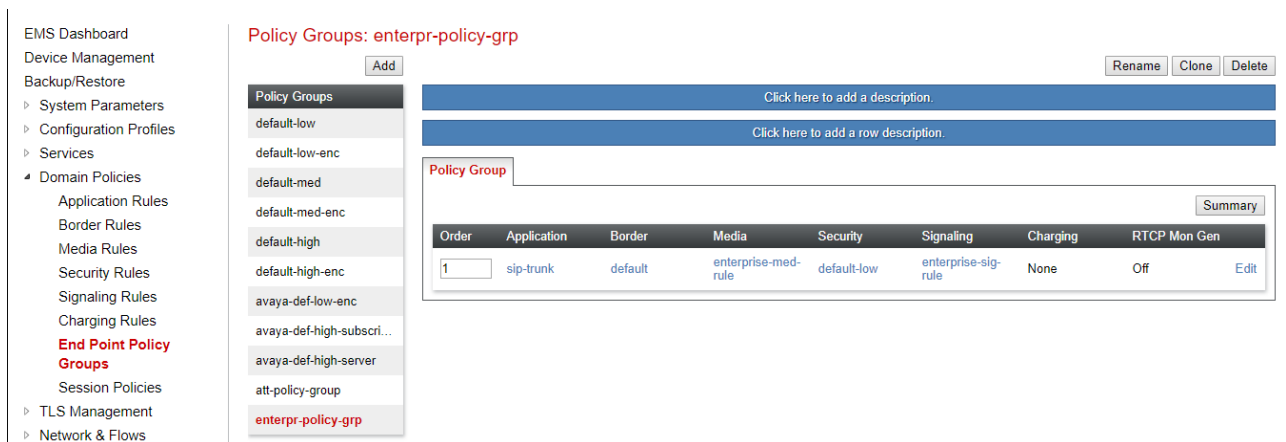


Step 3 – On the **Policy Group** window (not shown), select the following.

- **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.

The screenshot displays the EMS Dashboard interface. On the left is a navigation menu with categories like EMS Dashboard, Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, Application Rules, Border Rules, Media Rules, Security Rules, Signaling Rules, Charging Rules, End Point Policy Groups (highlighted in red), and Session Policies. The main area is titled 'Policy Groups: att-policy-group' and includes an 'Add' button. Below this is a list of policy groups: default-low, default-low-enc, default-med, default-med-enc, default-high, default-high-enc, avaya-def-low-enc, avaya-def-high-subscri..., and avaya-def-high-server. The 'att-policy-group' is at the bottom. To the right of the list are buttons for 'Rename', 'Clone', and 'Delete'. Below the list is a 'Policy Group' configuration table with columns: Order, Application, Border, Media, Security, Signaling, Charging, and RTP Mon Gen. The table contains one row with the following values: Order 1, Application sip-trunk, Border default, Media att-med-rule, Security default-low, Signaling att-sig-rule, Charging None, and RTP Mon Gen Off. There are also buttons for 'Click here to add a description.', 'Click here to add a row description.', 'Summary', and 'Edit'.

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).

View Flow: SM Flow IPFR	
Criteria	
Flow Name	SM Flow IPFR
Server Configuration	SM8
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Outside-B2-Signaling
Profile	
Signaling Interface	Inside-Sig-40
Media Interface	Inside-Media
Secondary Media Interface	None
End Point Policy Group	enterpr-policy-grp
Routing Profile	Route to ATT IPFR
Topology Hiding Profile	Enterprise-Topology
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>

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).

View Flow: IPFR Flow	
Criteria	
Flow Name	IPFR Flow
Server Configuration	ATT-trk-svr
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Inside-Sig-40
Profile	
Signaling Interface	Outside-B2-Signaling
Media Interface	Outside-B2-Media
Secondary Media Interface	None
End Point Policy Group	att-policy-group
Routing Profile	Route to SM8
Topology Hiding Profile	SIP Trunk-Topology
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>

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.729 and 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
14:07:32      rgn:4 [10.64.91.40]:17704
14:07:32      rgn:1 [10.64.91.86]:6052
14:07:32      G72264K ss:off ps:20
14:07:32      rgn:1 [10.5.5.211]:24706
14:07:32      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** in 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

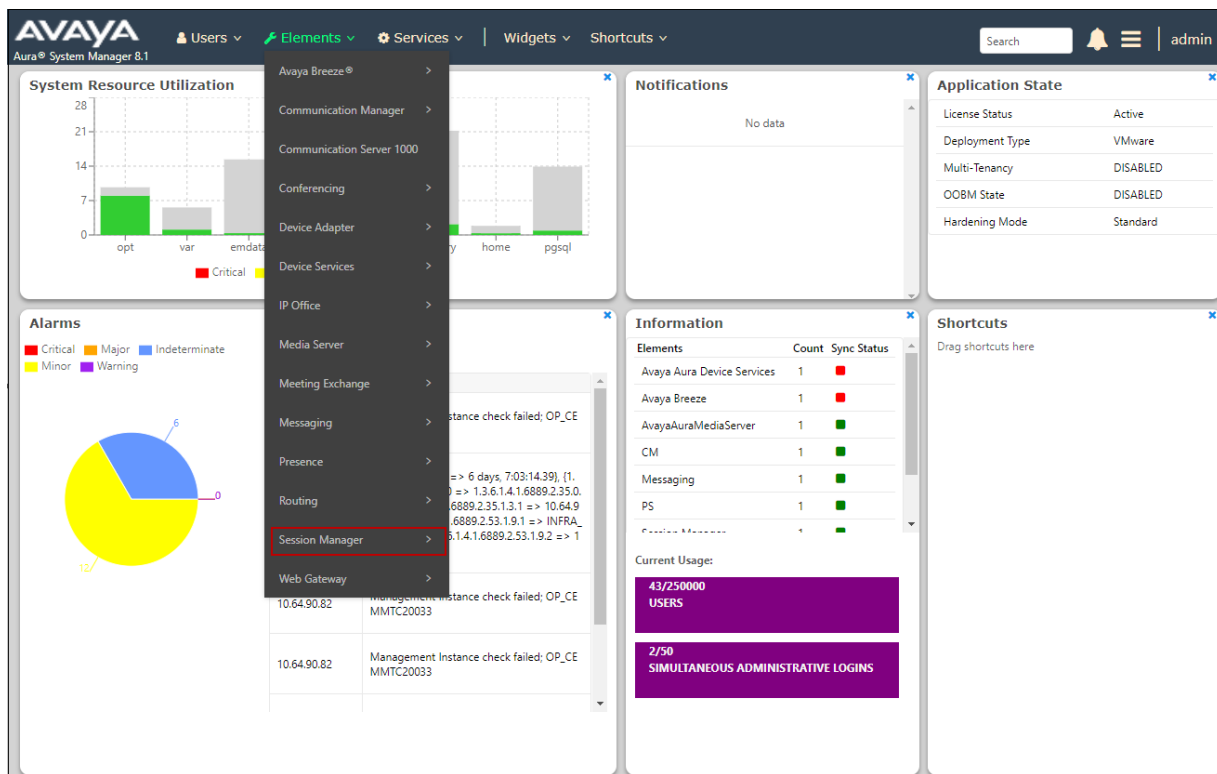
src port: T000051
T000051:TX:10.64.91.40:17704/g729/20ms/1-srtp-aescm128-hmac80
S000009:RX:10.5.5.211:24706/g729/20ms/1-srtp-aescm128-hmac80

```

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.

Session Manager

Dashboard

Session Manager Admin...

Global Settings

Communication Profile ...

Network Configuration

Device and Location ...

Application Configur...

System Status

System Tools

Help ?

Session Manager Dashboard

This page provides the overall status and health summary of each administered Session Manager.

Session Manager Instances

Service State

Shutdown System

EASG

As of 8:00 AM

1 Item

Show

All

Filter: Enable

	Session Manager	Type	Tests Pass	Alarms	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Data Replication	User Data Storage Status	License Mode	EASG	Version
<input type="checkbox"/>	Session Manager	Core	✓	0/0/0	Up	Accept New Service	2/15	0	6/6		✓	Normal	Enabled	8.1.0.0.810007

Select : All, None

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.

ID	Device	Date & Time	Category	Type	Cause
785619498994851	SBCE8-70	Oct 16, 2019 9:16:37 AM	Media Anomaly Detection	Media Inactivity Detected From Both Parties	Call Audit Cleanup
785616619198423	SBCE8-70	Oct 16, 2019 7:40:38 AM	Policy	Server Heartbeat	Heartbeat Successful, Server is UP
785616619190094	SBCE8-70	Oct 16, 2019 7:40:38 AM	Policy	Server Heartbeat	Heartbeat Successful, Server is UP
785616503469695	SBCE8-70	Oct 16, 2019 7:36:46 AM	Policy	Server Heartbeat	Heartbeat Failed, Server is Down
785616503469585	SBCE8-70	Oct 16, 2019 7:36:46 AM	Policy	Server Heartbeat	Heartbeat Failed, Server is Down
785616501493307	SBCE8-70	Oct 16, 2019 7:36:42 AM	Policy	Server Heartbeat	Heartbeat Successful, Server is UP
785616501482423	SBCE8-70	Oct 16, 2019 7:36:42 AM	Policy	Server Heartbeat	Heartbeat Successful, Server is UP
785317561958470	SBCE8-70	Oct 9, 2019 9:32:03 AM	Policy	Routing Failure	Max forwards Exceeded
785317555809802	SBCE8-70	Oct 9, 2019 9:31:51 AM	Policy	Routing Failure	Max forwards Exceeded

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 screenshot shows the Avaya SBCE Enterprise interface. The top navigation bar includes 'Device: SBCE8-70', 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', and 'Users'. The 'Status' menu is expanded, showing 'SIP Statistics', 'Periodic Statistics', 'User Registrations', and 'Server Status' (highlighted with a red box). The main dashboard area is titled 'Session Border Controller Enterprise' and features the Avaya logo. On the left, there is an 'EMS Dashboard' sidebar with links to 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Services', 'Domain Policies', 'TLS Management', 'Network & Flows', and 'DMZ Services'. The central 'Dashboard' section displays system information: System Time (09:39:07 AM MDT), Version (8.0.1.0-10-17555), Build Date (Tue Jul 30 22:53:51 UTC 2019), License State (OK), Aggregate Licensing Overages (0), and Peak Licensing Overage Count (0). On the right, the 'Installed Devices' section lists 'EMS' and 'SBCE8-70'.

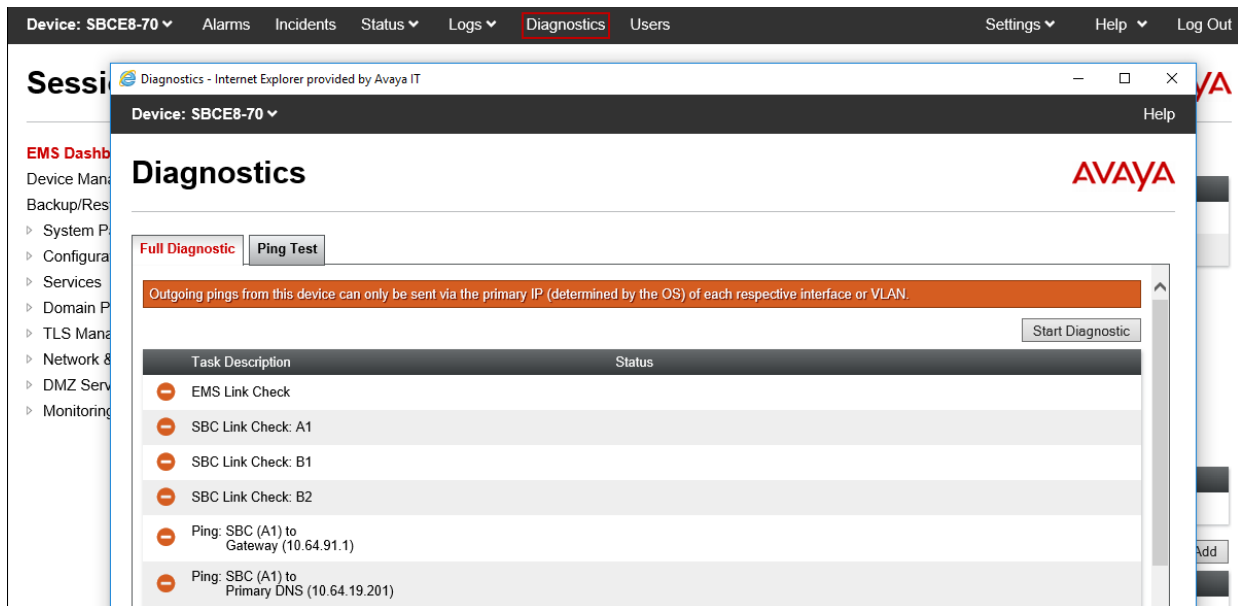
The **Server Status** screen provides information about the condition of the connection to the connected SIP Servers. This functionality requires Heartbeat to be enabled on the SIP Server Configuration profiles, as configured in **Section 8.99**.

The screenshot shows the 'Status' page in the Avaya SBCE Enterprise interface. The 'Server Status' tab is selected. The page displays a table of connected SIP servers with the following columns: Server Profile, Server FQDN, Server IP, Server Port, Server Transport, Heartbeat Status, Registration Status, and TimeStamp.

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

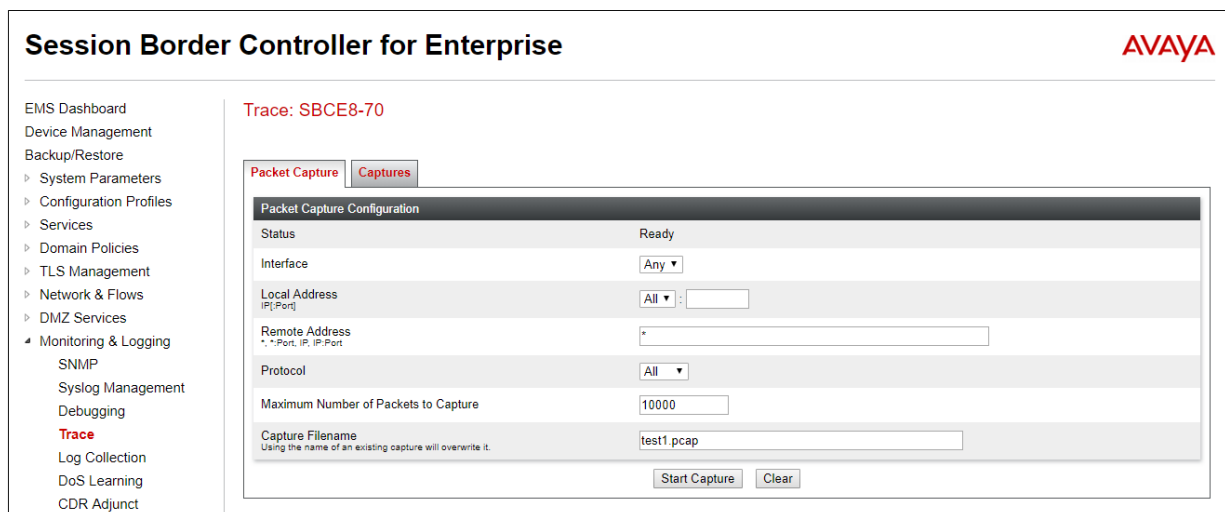
10.4.3. Diagnostic

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.

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

Device Management

Backup/Restore

System Parameters

Configuration Profiles

Services

Domain Policies

TLS Management

Network & Flows

DMZ Services

Monitoring & Logging

SNMP

Syslog Management

Debugging

Trace

Log Collection

DoS Learning

CDR Adjunct

Trace: SBCE8-70

Packet Capture

Captures

A packet capture is currently in progress. This page will automatically refresh until the capture completes.

Packet Capture Configuration

Status	In Progress
Interface	Any
Local Address <small>(IP,Port)</small>	All
Remote Address <small>* -Port, IP, IP-Port</small>	
Protocol	All
Maximum Number of Packets to Capture	10000
Capture Filename <small>Using the name of an existing capture will overwrite it.</small>	test1.pcap

Stop Capture

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

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

Device Management

Backup/Restore

System Parameters

Configuration Profiles

Services

Domain Policies

TLS Management

Network & Flows

DMZ Services

Monitoring & Logging

SNMP

Syslog Management

Debugging

Trace

Trace: SBCE8-70

Packet Capture

Captures

Refresh

File Name	File Size (bytes)	Last Modified	
test1_20190724082944.pcap	696,320	July 24, 2019 8:30:26 AM MDT	Delete

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**.

Edit URI X

Each entry should match a valid SIP URI.

WARNING: Invalid or incorrectly entered regular expressions may cause unexpected results.

Note: This regular expression is case-insensitive.

Ex: [0-9]{3,5}\.user@domain\.com, (simple|advanced)\-user[A-Z]{3}@.*

Scheme

☒ sip:/sips:
☐ tel:

Type

☐ Plain
☐ Dial Plan
☒ Regular Expression

URI 59[0-9]{3}@.*

Finish

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

Session Border Controller for Enterprise

URI Groups: internal-extensions

URI Listing

URI Group	Edit	Delete
59[0-9]{3}@.*	Edit	Delete
54[0-9]{3}@.*	Edit	Delete

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**.

Session Border Controller for Enterprise

Interworking Profiles: ATT-Interworking

General

Setting	Value
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	Yes
URI Group	internal-extensions
Send Hold	No
Delayed Offer	Yes
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
Prack Handling	No
Allow 18X SDP	No
T.38 Support	Yes
URI Scheme	SIP
Via Header Format	RFC3261

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";
            }
        }
    }
}
```

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