Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1 and Avaya Session Border Controller for Enterprise 8.1 with Verizon Business IP Contact Center Services Suite – Issue 1.0

Abstract

These Application Notes describe a sample configuration of Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1 and Avaya Session Border Controller for Enterprise 8.1 with Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite includes the IP Toll Free VoIP Inbound and IP-IVR SIP trunk service offers. This service suite provides toll free inbound calling via standards-based SIP trunks as well as re-routing of inbound toll-free calls to alternate destinations based upon SIP messages (i.e., REFER) generated by Communication Manager. The Communication Manager Network Call Redirection (NCR) and SIP User-to-User Information (UUI) features can be utilized together to transmit UUI within SIP signaling messages to alternate destinations via the Verizon network. These Application Notes update previously published Application Notes with newer versions of Communication Manager, Session Manager, and Avaya Session Border Controller for Enterprise.

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

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

Table of Contents

1. Introduction 5
2. General Test Approach and Test Results 6
  2.1. Interoperability Compliance Testing 7
  2.2. Test Results 8
  2.3. SIP Header Removal 9
  2.4. Support 9
    2.4.1 Avaya 9
    2.4.2 Verizon 9
3. Reference Configuration 10
  3.1. History Info and Diversion Headers 11
  3.2. Call Flows 12
    3.2.1 Inbound IP Toll Free Call with no Network Call Redirection 12
    3.2.2 Inbound IP Toll Free Call with Post-Answer Network Call Redirection 13
    3.2.3 Inbound IP Toll Free Call with Unsuccessful Network Call Redirection 14
4. Equipment and Software Validated 15
5. Configure Avaya Aura® Communication Manager 16
  5.1. Verify Licensed Features 16
  5.2. System-Parameters Features 18
  5.3. Dial Plan 19
  5.4. Node Names 19
  5.5. Processor Ethernet Configuration 20
  5.6. IP Codec Sets 21
    5.6.1 Codecs for IP Network Region 1 (calls within the CPE) 21
    5.6.2 Codecs for IP Network Region 2 (calls from Verizon) 22
  5.7. Network Regions 23
    5.7.1 IP Network Region 1 – Local CPE Region 23
    5.7.2 IP Network Region 2 – Verizon Trunk Region 24
  5.8. SIP Trunks 25
    5.8.1 SIP Trunk for Inbound Verizon calls 25
    5.8.2 Local SIP Trunk (Avaya SIP Telephone and Messaging Access) 29
  5.9. Contact Center Configuration 30
    5.9.1 Announcements 30
    5.9.2 Post-Answer Redirection to a PSTN Destination 30
    5.9.3 Post-Answer Redirection With UUI to a SIP Destination 31
    5.9.4 ACD Configuration for Call Queued for Handling by Agent 33
  5.10. Public Numbering 36
  5.11. Private Numbering 37
  5.12. Route Patterns 38
    5.12.1 Route Pattern for Calls within the CPE 38
  5.13. Automatic Alternate Routing (AAR) Dialing 38
  5.14. Avaya G430 Media Gateway Provisioning 39
  5.15. Avaya Aura® Media Server Provisioning 40
  5.16. Save Translations 41
  5.17. Verify TLS Certificates – Communication Manager 42
6. Configure Avaya Aura® Session Manager 43

6.1. System Manager Login and Navigation 44
6.2. SIP Domain 45
6.3. Locations 45
6.3.1 Main Location 45
6.3.2 Common Location 46
6.4. Configure Adaptations 47
6.4.1 Adaptation for Avaya Aura® Communication Manager Extensions 47
6.4.2 Adaptation for the Verizon Business IPCC Services 49
6.5. SIP Entities 49
6.5.1 Avaya Aura® Session Manager SIP Entity 50
6.5.2 Avaya Aura® Communication Manager SIP Entity – Public Trunk 52
6.5.3 Avaya Aura® Communication Manager SIP Entity – Local Trunk 53
6.5.4 Avaya Session Border Controller for Enterprise SIP Entity 53
6.5.5 Avaya Aura® Messaging SIP Entity 53
6.6. Entity Links 54
6.6.1 Entity Link to Avaya Aura® Communication Manager – Public Trunk 54
6.6.2 Entity Link to Avaya Aura® Communication Manager – Local Trunk 55
6.6.3 Entity Link for the Verizon Business IPCC Services via the Avaya SBCE 55
6.6.4 Entity Link to Avaya Aura® Messaging 55
6.7. Time Ranges 56
6.8. Routing Policies 56
6.8.1 Routing Policy for Verizon Routing to Avaya Aura® Communication Manager 56
6.8.2 Routing Policy for Inbound Routing to Avaya Aura® Messaging 57
6.9. Dial Patterns 58
6.10. Verify TLS Certificates – Session Manager 60
7. Configure Avaya Session Border Controller for Enterprise 62
7.1. Device Management – Status 63
7.2. TLS Management 65
7.2.1 Verify TLS Certificates – Avaya Session Border Controller for Enterprise 65
7.2.2 Server Profiles 66
7.2.3 Client Profiles 67
7.3. Network Management 68
7.4. Media Interfaces 69
7.5. Signaling Interfaces 70
7.6. Server Interworking Profiles 71
7.6.1 Server Interworking Profile – Enterprise 71
7.6.2 Server Interworking Profile – Verizon 72
7.7. Signaling Manipulation 73
7.8. SIP Server Profiles 74
7.8.1 SIP Server Profile – Session Manager 74
7.8.2 SIP Server Profile – Verizon 76
7.9. Routing Profiles 77
7.9.1 Routing Profile – Session Manager 78
7.9.2 Routing Profile – Verizon 79
7.10. Topology Hiding Profiles 80
7.10.1 Topology Hiding – Enterprise 80

7.10.2 Topology Hiding – Verizon 81
7.11. Application Rules 82
7.12. Media Rules 83
7.12.1 Enterprise – Media Rule 83
7.12.2 Verizon – Media Rule 84
7.13. Signaling Rules 85
7.13.1 Signaling Rule - Enterprise 85
7.13.2 Signaling Rule - Verizon 86
7.14. Endpoint Policy Groups 86
7.14.1 Endpoint Policy Group – Enterprise 86
7.14.2 Endpoint Policy Groups – Verizon 87
7.15. Endpoint Flows – Server Flows 88
7.15.1 Server Flow – Enterprise 88
7.15.2 Server Flow – Verizon 89
8. Verizon Business IPCC Services Suite Configuration 90
8.1. Service Access Information 90
9. Verification Steps 91
9.1. Avaya Aura® Communication Manager Verifications 91
9.1.1 Example Incoming Call from PSTN via Verizon IPCC to Agent 91
9.1.2 Example Incoming Call Referred via Call Vector to PSTN Destination 94
9.2. Avaya Aura® Session Manager Verification 96
9.3. Avaya Session Border Controller for Enterprise Verification 98
9.3.1 Incidents 98
9.3.2 Server Status 99
9.3.3 Diagnostics 100
9.3.4 Tracing 101
10. Conclusion 102
11. Additional References 103
11.1. Avaya 103
11.2. Verizon Business 103

1. Introduction

These Application Notes describe a sample configuration of Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1and Avaya Session Border Controller for Enterprise 8.1 with Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite includes the IP Toll Free VoIP Inbound and IP-IVR SIP trunk service offers. This service suite provides toll free inbound calling via standards-based SIP trunks as well as re-routing of inbound toll-free calls to alternate destinations based upon SIP messages (i.e., REFER) generated by Communication Manager. The Communication Manager Network Call Redirection (NCR) and SIP User-to-User Information (UUI) features can be utilized together to transmit UUI within SIP signaling messages to alternate destinations via the Verizon network. These Application Notes update previously published Application Notes [VZ-IPCC] with newer versions of Session Manager, Communication Manager, and Avaya Session Border Controller for Enterprise.

In the sample configuration, an Avaya Session Border Controller for Enterprise (Avaya SBCE) is used as an edge device between the Avaya CPE and Verizon Business. The Avaya SBCE performs SIP header manipulation and provides topology hiding to convert the private Avaya CPE IP addressing to IP addressing or domains appropriate for the Verizon access method. Session Manager is used as the Avaya SIP trunking "hub" connecting to Communication Manager, the Avaya SBCE, and other applications.

The Verizon Business IPCC Services suite described in these Application Notes is designed for business customers. The suite provides inbound toll-free service via standards-based SIP trunks. Using SIP Network Call Redirection (NCR), trunk-to-trunk connections of certain inbound calls at Communication Manager can be avoided by requesting that the Verizon network transfer the inbound caller to an alternate destination. In addition, the Communication Manager SIP User-to-User Information (UUI) feature can be utilized with the SIP NCR feature to transmit UUI within SIP signaling messages to alternate destinations. This capability allows the service to transmit a limited amount of call-related data between call centers to enhance customer service and increase call center efficiency. Examples of UUI data might include a customer account number obtained during a database query or the best service routing data exchanged between sites using Communication Manager.

Verizon Business IPCC Services suite is a portfolio of IP Contact Center (IPCC) interaction services that includes VoIP Inbound and IP Interactive Voice Response (IP-IVR). Access to these features may use Internet Dedicated Access (IDA) or Private IP (PIP). PIP was used for the sample configuration described in these Application Notes. VoIP Inbound is the base service offering that offers core call routing and termination features. IP-IVR is an enhanced service offering that includes features such as menu-routing, custom transfer, and additional media capabilities.

For more information on the Verizon Business IP Contact Center service, visit https://enterprise.verizon.com/products/customer-experience-services/transport-and-intelligent-routing/ip-contact-center/

2. General Test Approach and Test Results

The test approach was manual testing of inbound and referred calls using the Verizon Business IPCC Services on a production Verizon PIP access circuit, as shown in **Figure 1**. Testing was successful. Test observations or limitations are described in **Section 2.2**.

See **Section 3.2** for an overview of key call flows and **Section 9** for detailed verifications and traces illustrating key call flows.

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

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

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

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

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

2.1. Interoperability Compliance Testing

The interoperability compliance testing included the execution of test cases details in the Verizon-authored interoperability test plan.

- SIP OPTIONS monitoring of the health of the SIP trunks was verified. Both the Avaya enterprise equipment and Verizon Business can monitor health using SIP OPTIONS.
- Incoming calls from the PSTN were routed to the toll-free numbers assigned by Verizon Business to the Avaya location. Configuration was varied such that these incoming toll-free calls were directed to Communication Manager telephone extensions and Communication Manager VDNs containing call routing logic to exercise SIP Network Call Redirection.
- Proper disconnect when either party hangs up an active call.
- Proper disconnect when the PSTN caller abandons (i.e., hangs up) a toll-free call before the call has been answered.
- Proper SIP 486 response and busy tone heard by the caller when a PSTN user calls a toll-free number directed to a busy user or resource when no redirection on busy conditions was configured (which would be unusual in a contact center).
- Proper termination of an inbound IP Toll Free call left in a ringing state for a relatively long duration, which again would be unusual in a contact center. In the sample configuration, Verizon sent a SIP CANCEL to cancel the call after approximately 35 seconds of ring no answer condition, returning busy tone to the PSTN caller.
- Privacy requests for inbound toll-free calls from the PSTN were verified. That is, when privacy is requested by a PSTN caller (e.g., dialing *67 from a mobile phone), the inbound toll-free call can be successfully completed while withholding presentation of the PSTN caller ID to user displays. (When the caller requests privacy, Verizon IPCC sends the caller ID in the P-Asserted-Identity header and includes “Privacy: id” which is honored by Communication Manager).
- Inbound toll-free call long holding time call stability. The Avaya CPE sends a re-INVITE with SDP to refresh the session at the configured session refresh interval specified on the Communication Manager trunk group handling the call. In the sample configuration, the session refresh re-INVITE was sent after 900 seconds (15 minutes), the interval configured for the trunk group in **Section 5.8.1**. The call continued with proper talk path.
- Telephony features such as hold and resume. When a Communication Manager user holds a call in the sample configuration, Communication Manager will send a re-INVITE to Verizon IP Toll Free service with a media attribute “sendonly”. The Verizon 200 OK to this re-INVITE will include media attribute “recvonly”. While the call remains on hold, RTP will flow from the Avaya CPE to Verizon, but no RTP will flow from Verizon to the Avaya CPE (i.e., as intended). When the user resumes the call from hold, bi-directional media path resumes. Although it would be unexpected in a contact center, calls on hold for longer than the session refresh interval were tested, and such calls could be resumed after the session refresh re-asserted the “sendonly” state.
- Transfer of toll-free calls between Communication Manager users.
- Incoming voice calls using the G.729A and G.711 ULAW codecs, and proper protocol procedures related to media.
- DTMF transmission using RFC2833. For inbound toll-free calls, PSTN users dialing post-answer DTMF digits are recognized properly by the Avaya CPE.

- Proper DiffServ markings for SIP signaling and RTP media flowing from the Avaya CPE to Verizon.
- Incoming fax calls using T.38.
- Remote Avaya SIP endpoints connected through Avaya SBCE were used along with local Avaya endpoints in the verification of these Application Notes.

2.2. Test Results

The interoperability compliance testing of the sample configuration was completed with successful results as described in **Section 2.1**. The following limitations are noted for the sample configuration described in these Application Notes.

- Verizon Business IPCC Services suite does not support History Info or Diversion Headers. The Avaya CPE will not send History-Info or Diversion header to Verizon IPCC in the sample configuration.
- Verizon Business IPCC Services suite does not support SIP 302 Moved Temporarily redirection messages.
- Verizon Business IPCC Services suite does not support G.729 Annex B. When using G729, the Avaya CPE will always include "annexb=no" in SDP in the sample configuration.
- **Section 3.2.3** summarizes a call flow that would allow Communication Manager to continue the processing of a call upon failure of a vector-triggered REFER attempt to the PSTN. However, such call scenario could not be verified on the production Verizon circuit used for testing. On the production circuit, Verizon sent a BYE to terminate the call immediately upon encountering REFER transfer failures, so there was no opportunity for the call to continue being processed by the Communication Manager. See **Section 3.2.3** for additional information.
- During testing, Verizon's IP Interactive Voice Response (IP-IVR) service did not accept the SIP REFER method unless the URI in the Refer-To header included the IP address presented in the From header within the original SIP INVITE. This IP address was different from the IP address included in the Contact header. The Avaya SBCE Topology Hiding profile was used to populate the From header IP address in the Refer-To header for both the IP-IVR and IP Toll Free services. Calls were successfully diverted using REFER for both Verizon services with this Topology Hiding profile in place. See **Section 7.10.2**.

2.3. SIP Header Removal

To support advanced SIP telephony features in the Avaya Aura® enterprise environment, certain proprietary headers may be included in the SIP message sent toward Verizon. These extra headers can cause the SIP message to become larger than the specified Maximum Transmission Unit (MTU) and create fragmented UDP packets. These fragmented packets may not be re-assembled properly on the far-end by Verizon's equipment, for instance, when packets arrive out of order. To prevent fragmented packets, any unnecessary or proprietary headers should be removed from the SIP message before being sent to Verizon. Session Manager can remove these headers by specifying the "*eRHdrs*" parameter within the "*VerizonAdapter*" adaptation. See **Section 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 "*gsid*" and "*epv*" parameters that may be included within the Contact header by applying a Sigma script to the Verizon server configuration. See **Section 7.7**.

2.4. Support

2.4.1 Avaya

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

2.4.2 Verizon

For technical support on Verizon Business IPCC Services offer, visit online support at http://www.verizonenterprise.com/support/

3. Reference Configuration

Figure 1 illustrates the sample configuration used for the DevConnect compliance testing. The Avaya CPE location simulates a customer site. The PIP service defines a secure MPLS connection between the Avaya CPE and the Verizon Business IPCC Services node. At the edge of the Avaya CPE location is an Avaya Session Border Controller for Enterprise. The Avaya SBCE receives traffic from the Verizon Business IPCC Services on port 5060 and sends traffic to the Verizon Business IPCC Services on destination port 5072, using UDP for transport.

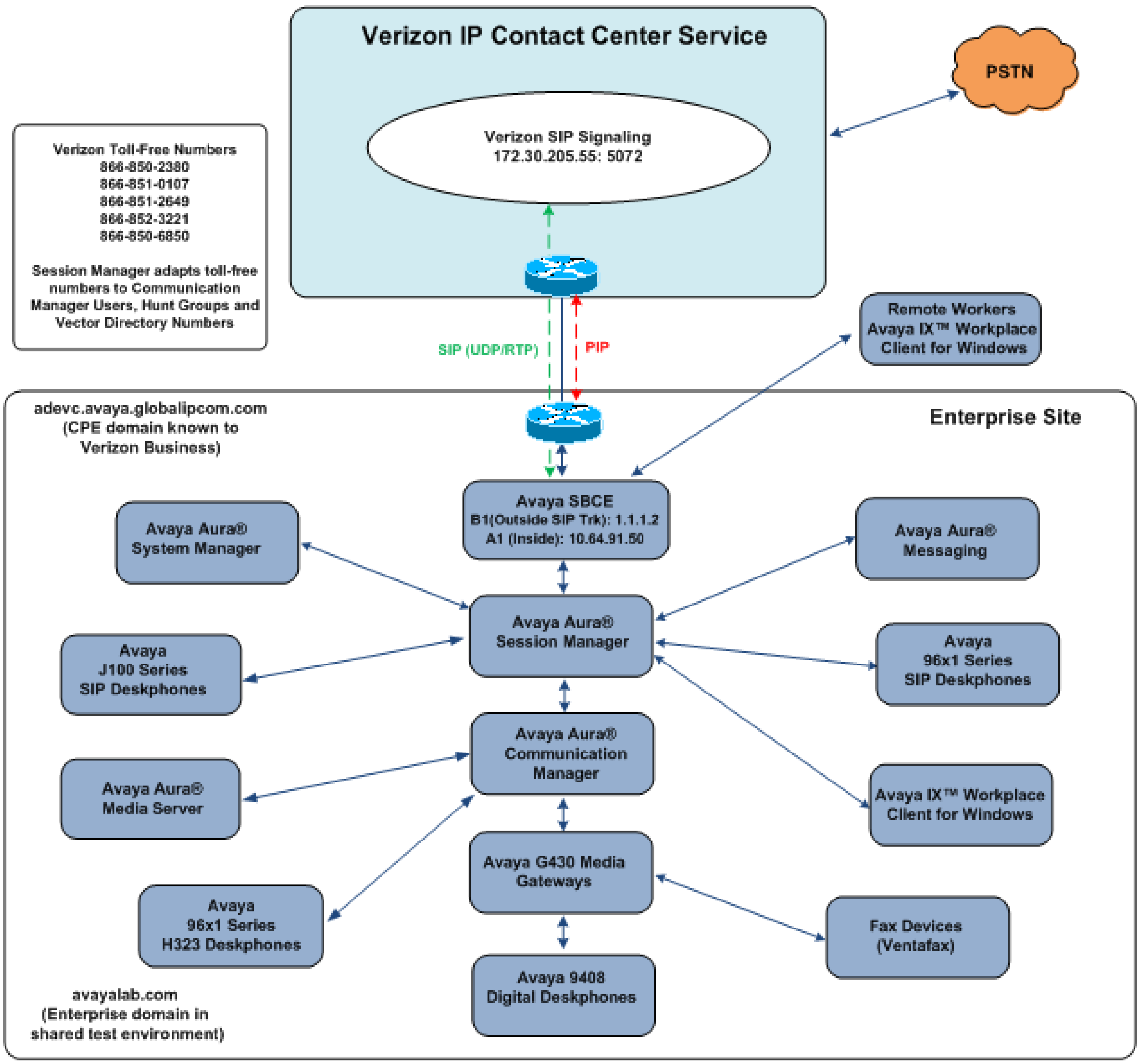


Figure 1: Avaya Interoperability Test Lab Configuration

The Verizon toll-free numbers were mapped by Session Manager to various Communication Manager extensions. The extension mappings were varied during the testing to allow inbound toll-free calls to terminate directly on user extensions or indirectly through hunt groups, vector directory numbers (VDNs) and vectors to user extensions and contact center agents.

The Avaya CPE environment was known to Verizon Business IPCC Services as FQDN "*adevc.avaya.globalipcom.com*". For efficiency, the Avaya CPE environment utilizing Session Manager Release 8.1 and Communication Manager Release 8.1 was shared among other ongoing test efforts at the Avaya Solutions and Interoperability Test lab. Access to the Verizon Business IPCC Services was added to a configuration that already used domain "*avayalab.com*" at the enterprise. As such, the Avaya SBCE is used to adapt the domains as needed. These Application Notes indicate the configuration that would not be required in cases where the CPE domain in Communication Manager and Session Manager match the CPE domain known to Verizon.

The following summarizes various header contents and manipulations for toll-free calls in the sample configuration:

- Verizon Business IPCC Services node sends the following in the initial INVITE to the CPE:
  - The CPE FQDN of ***adevc.avaya.globalipcom.com*** in the Request URI.
  - The Verizon Business IPCC Services gateway IP address in the From header.
  - The enterprise SBC outside IP address (e.g., 1.1.1.2) in the To header.
  - Sends the INVITE to Avaya CPE using destination port 5060 via UDP
- Avaya Session Border Controller for Enterprise sends Session Manager:
  - The Request URI contains ***avayalab.com.***
  - The host portion of the From header and PAI header contains ***avayalab.com***
  - The host portion of the To header contains ***avayalab.com***
  - Sends the packet to Session Manager using destination port 5061 via TLS
- Session Manager sends Communication Manager
  - The Request URI contains ***avayalab.com***, to match the shared Avaya SIL test environment.
  - Sends the packet to Communication Manager using destination port 5071 via TLS to allow Communication Manager to distinguish Verizon traffic from other traffic arriving from the same instance of Session Manager.

Note – The Fully Qualified Domain Names and IP addressing specified in these Application Notes apply only to the reference configuration shown in **Figure 1**. Verizon Business customers will use FQDNs and IP addressing appropriate for the unique customer environment.

3.1. History Info and Diversion Headers

The Verizon Business IPCC Services suite does not support SIP History Info headers or Diversion headers. Therefore, Communication Manager was provisioned not to send History Info headers or Diversion headers.

3.2. Call Flows

To understand how inbound Verizon toll-free calls are handled by Session Manager and Communication Manager, key call flows are summarized in this section.

3.2.1 Inbound IP Toll Free Call with no Network Call Redirection

The first call scenario illustrated in **Figure 3** is an inbound Verizon IP Toll Free call that is routed to Communication Manager, which in turn routes the call to a vector, agent, or phone. No redirection is performed in this simple scenario. A detailed verification of such a call with Communication Manager traces can be found in **Section 9.1.1**.

1. A PSTN phone originates a call to a Verizon IP Toll Free number.
2. The PSTN routes the call to the Verizon IP Toll Free service network.
3. The Verizon IP Toll Free service routes the call to the Avaya Session Border Controller for Enterprise.
4. The Avaya Session Border Controller for Enterprise performs any configured SIP header modifications, and routes the call to Session Manager.
5. Session Manager applies any configured SIP header adaptations and digit conversions, and based on configured Routing Policies, determines where the call should be routed. In this case, Session Manager routes the call to Communication Manager using a unique port so that Communication Manager can distinguish this call as having arrived from Verizon IPCC.
6. Depending on the called number, Communication Manager routes the call to:
   a) a hunt group or vector, which in turn routes the call to an agent or phone, or
   b) directly to a phone.

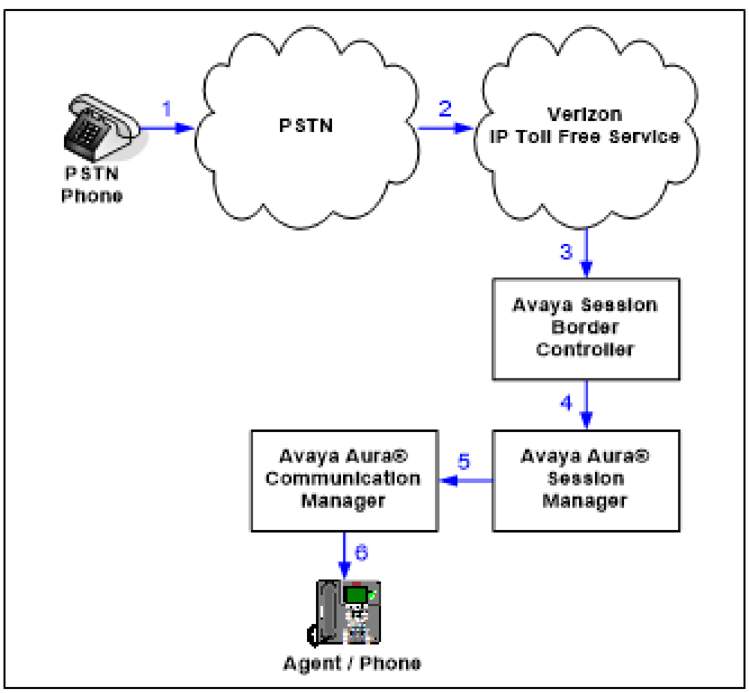


Figure 3: Inbound Verizon IP Toll Free Call – No Redirection

3.2.2 Inbound IP Toll Free Call with Post-Answer Network Call Redirection

The second call scenario illustrated in **Figure 4** is an inbound Verizon IP Toll Free call that is routed to a Communication Manager Vector Directory Number (VDN) to invoke call handling logic in a vector. The vector answers the call and then redirects the call back to the Verizon IP Toll Free service for routing to an alternate destination. Note that Verizon IP Toll Free service does not support redirecting a call before it is answered (using a SIP 302), and therefore the vector must include a step that results in answering the call, such as playing an announcement, prior to redirecting the call using REFER.

A detailed verification of such call with Communication Manager traces can be found in **Section 9.1.2** for a Verizon IP Toll Free SIP-connected alternate destination. In this example, the Verizon IP Toll Free service can be used to pass User to User Information (UUI) from the redirecting site to the alternate destination.

1. Same as the first five steps in **Figure 3**.
2. Communication Manager routes the call to a vector, which answers the call, plays an announcement, and attempts to redirect the call by sending a SIP REFER message out the SIP trunk from which the inbound call arrived. The SIP REFER message specifies the alternate destination in the Refer-To header. The SIP REFER message passes back through Session Manager and the Avaya SBCE to the Verizon IP Toll Free service network.
3. The Verizon IP Toll Free service places a call to the target party contained in the Refer-To header. Upon answer, the calling party is connected to the target party.
4. The Verizon IP Toll Free service notifies the Avaya CPE that the referred call has been answered (NOTIFY/sipfrag 200 OK). Communication Manager sends a BYE. The calling party and the target party can talk. The trunk upon which the call arrived in Step 1 is idle.

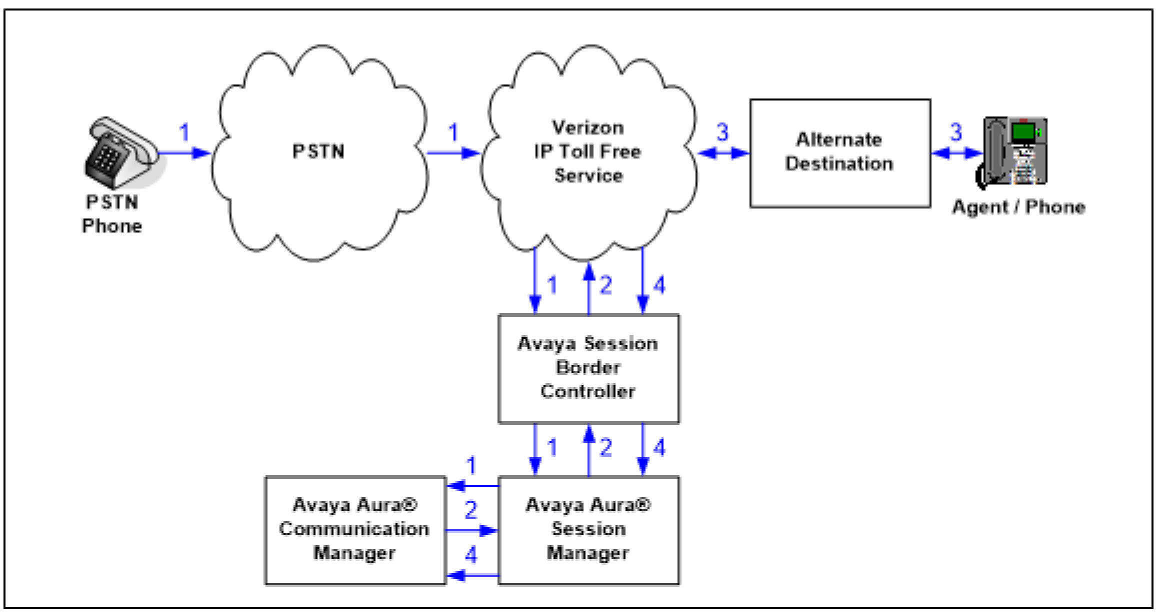


Figure 4: Inbound Verizon IP Toll Free– Post-Answer SIP REFER Redirection Successful

3.2.3 Inbound IP Toll Free Call with Unsuccessful Network Call Redirection

The next call scenario illustrated in **Figure 5** is similar to the previous call scenario, except that the redirection is unsuccessful. In this case, Communication Manager can “take the call back” and continue vector processing. For example, the call may route to an alternative agent, phone, or announcement after unsuccessful NCR.

1. Same as **Figure 4**.
2. Same as **Figure 4**.
3. The Verizon IP Toll Free service places a call to the target party (alternate destination), but the target party is busy or otherwise unavailable.
4. The Verizon IP Toll Free service notifies the redirecting/referring party (Communication Manager) of the error condition.
5. Communication Manager routes the call to a local agent, phone, or announcement.

However, as noted in **Section 2.2**, this “unsuccessful transfer” scenario could not be verified on the production Verizon circuit used for testing. On the production circuit, Verizon sends a SIP BYE message which terminates Communication Manager vector processing for failure scenarios. For example, if a 486 Busy is received from the target of the REFER, Verizon will send a BYE immediately after a “NOTIFY/sipfrag 486 Busy Here”, which precludes any further call processing by Communication Manager. As another example, in cases where mis-configuration is introduced to cause the Refer-To header to be malformed (e.g., no “+” in Refer-To), Verizon will similarly send a BYE immediately after a “NOTIFY/sipfrag 603 Server Internal Error”.

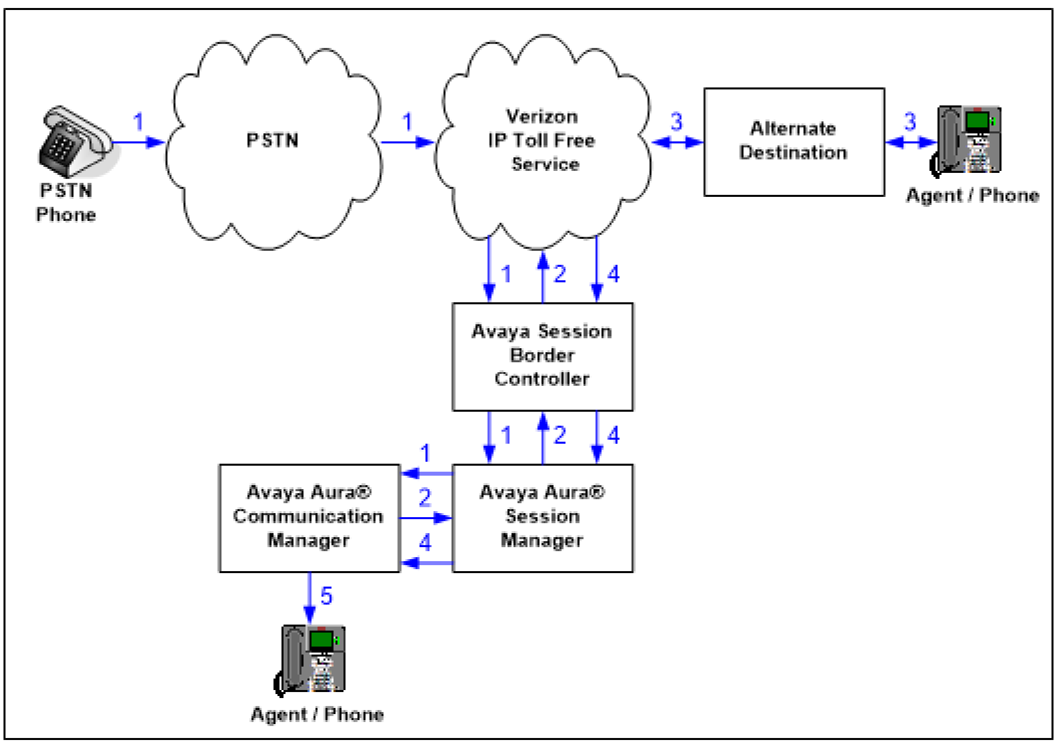


Figure 5: Inbound Verizon IP Toll Free– Post-Answer SIP REFER Redirection Unsuccessful

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® System Manager	8.1.2.0.0611097 (Feature Pack 2)
Avaya Aura® Session Manager	8.1.2.0.812039
Avaya Aura® Communication Manager	8.1.2.0.0-FP2 (patch 26095)
Avaya Session Border Controller for Enterprise	8.1.0.0.14-18490
Avaya Aura® Messaging	7.1.Service Pack 2
Avaya Aura® Media Server	8.0.2.93
G430 Gateway	41.24.0
Avaya 96X1 Series IP Deskphone (SIP)	6.8304
Avaya 96X1 Series IP Deskphone (H.323)	7.1.8.0.9
Avaya J100 Series IP Deskphone(SIP)	4.0.4.0.10
Avaya IX™ Workplace Client for Windows	3.8.2.20.6
Avaya 9408 Digital Deskphone	20.06
Fax device	Ventafax 7.10

Table 1: Equipment and Software Used in the Sample Configuration

5. Configure Avaya Aura® Communication Manager

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

Note – The initial installation, configuration, and licensing of the Avaya servers and media gateways for Communication Manager are assumed to have been previously completed and are not discussed in these Application Notes. Consult **[5] - [6]** for further details.

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. If any of the required features are not set, and cannot be configured, contact an authorized Avaya account representative to obtain the necessary licenses/access.

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

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

IP PORT CAPACITIES                                          USED
                 Maximum Administered H.323 Trunks:  4000      0
       Maximum Concurrently Registered IP Stations:  1000      0
         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     5
                   Maximum Administered SIP Trunks:  4000     95
  Max Administered Ad-hoc Video Conferencing Ports:  4000      0
   Max Number of DS1 Boards with Echo Cancellation:    80      0
```

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

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

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

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

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

           IP Attendant Consoles? y
```

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

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

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

5.2. System-Parameters Features

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

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

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


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

5.3. Dial Plan

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

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

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

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

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

5.4. Node Names

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

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

- Session Manager SIP signaling interface (e.g., **SM** and **10.64.91.81**).
- Media Server (e.g., **AMS801** 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
AMS801              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.

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


                  Type: PROCR
                                                         Target socket load: 4800

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


                                 IPV4 PARAMETERS
             Node Name: procr                     IP Address: 10.64.91.75


           Subnet Mask: /24
```

5.6. 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.

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

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

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

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

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

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

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

                          IP MEDIA PARAMETERS

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

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

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

5.6.2 Codecs for IP Network Region 2 (calls from Verizon)

This IP codec set will be used for Verizon Business IP Trunking calls. Repeat the steps in **Section 5.6.1** with the following changes:

On **Page 1**, provision the codecs in the order shown below.

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

                          IP MEDIA PARAMETERS
    Codec Set: 2

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


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

On **Page 2**, set **FAX Mode** to **t.38-G711-fallback**, **ECM** to **y**, and **FB-Timer** to **4**

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

                          IP MEDIA PARAMETERS

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

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


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

5.7. Network Regions

Network regions provide a means to logically group resources. 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 2 was associated with other components used specifically for the Verizon testing.

5.7.1 IP Network Region 1 – Local CPE Region

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

- Enter a descriptive name (e.g., **Enterprise**).
- Enter the enterprise domain (e.g., **avayalab.com**) in the **Authoritative Domain** field.
- Enter **1** for the **Codec Set** parameter.
- **Intra-region IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible within the same region.
- **Inter-region IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible between regions.

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

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

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

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

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

5.7.2 IP Network Region 2 – Verizon Trunk Region

Repeat the steps in **Section 5.7.1** with the following changes:

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

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

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

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

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

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

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 Verizon IPCC access – SIP Trunk 2. This trunk will use TLS port 5071.
- Internal CPE access (e.g., Avaya SIP telephones, Messaging, etc.) – SIP Trunk 3. This trunk will use TLS port 5061.

Note – Although TLS is used as the transport protocols between the Avaya CPE components, UDP was used between the Avaya SBCE and the Verizon Business IPCC Services. See the note in **Section 6.5** regarding the use of TLS transport protocols in the CPE.

5.8.1 SIP Trunk for Inbound Verizon calls

This section describes the steps for administering the SIP trunk to Session Manager used for Verizon IPCC service calls. Trunk 1 is defined. This trunk corresponds to the Session Manager **CM-TG2** SIP Entity defined later in **Section 6.5.2**.

5.8.1.1 Signaling Group 2

Step 1 - Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **2**), 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 **5071**.
- **Far-end Network Region** – Set the IP network region to **2**, as set in **Section 5.7.2**.
- **Far-end Domain** – Enter **avayalab.com**.
- **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).

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

 Group Number: 2             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                   Far-end Node Name: SM
 Near-end Listen Port: 5071                  Far-end Listen Port: 5071
                                          Far-end Network Region: 2

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

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

5.8.1.2 Trunk Group 2

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

- **Group Type** – Set to **sip**.
- **Group Name** – Enter a descriptive name (e.g., **Verizon IPCC**).
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., ***02**).
- **Direction** – Set to **incoming**.
- **Service Type** – Set to **public-ntwrk**.
- **Signaling Group** – Set to the signaling group administered in **Section 5.8.1.1** (e.g., **2**).
- **Number of Members** – Enter the maximum number of simultaneous calls desired on this trunk group (based on licensing) (e.g., **10**).

```
add trunk-group 2                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 2                    Group Type: sip          CDR Reports: y
  Group Name: Verizon IPCC                COR: 1        TN: 1        TAC: *02
   Direction: incoming       Outgoing Display? n
 Dial Access? n                                   Night Service:

Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     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 2                                              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 2                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                     Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y

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

 Show ANSWERED BY on Display? y
```

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

- Verify **Network Call Redirection** is set to **y**.
- Set **Telephone Event Payload Type** to the RTP payload type recommended by Verizon (e.g., **101**).
- Set **Convert 180 to 183 for Early Media** to **y**. Verizon recommends that inbound calls to the enterprise result in a 183 with SDP rather than a 180 with SDP.

Note – The Verizon Business IPCC Services do not support the Diversion header or the History-Info header, and therefore both **Support Request History** and **Send Diversion Header** are set to "**n**".

```
add trunk-group 2                                              Page   4 of  21
                              PROTOCOL VARIATIONS

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


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

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

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

Trunk 3 corresponds to the Session Manager **CM-TG3** SIP Entity defined later 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 5.7.1**.

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**).
- **Direction** – Set to **two-way**.
- **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 **Send Diversion Header** to **n**.
- Verify **Identity for Calling Party Display** is set to **P-Asserted-Identity** (default).

Use default values for all other settings.

5.9. Contact Center Configuration

This section describes the basic commands used to configure Vector Directory Numbers (VDNs) and corresponding vectors. These vectors contain steps that invoke the Communication Manager SIP Network Call Redirection (NCR) functionality. These Application Notes provide rudimentary vector definitions to demonstrate and test the SIP NCR and UUI functionalities. In general, call centers will use vector functionality that is more complex and tailored to individual needs. Call centers may also use customer hosts running applications used in conjunction with Application Enablement Services (AES) to define call routing and provide associated UUI. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes.

5.9.1 Announcements

Various announcements will be used within the vectors. In the sample configuration, these announcements were sourced by the Avaya G430 Media Gateway. The following abridged list command summarizes the announcements used in conjunction with the vectors in this section. To add an announcement extension, use the command **add announcement <extension>** (not shown).

```
list announcement

                           ANNOUNCEMENTS/AUDIO SOURCES
Announcement                                                      Source      Num of
Extension             Type       Name                            Pt/Bd/Grp    Files
11001                 integrated callcenter-main                  005V9       1
11002                 integ-mus  holdmusic                        005V9       1
11003                 integrated disconnect                       005V9       1
11004                 integrated no_agents                        005V9       1
11005                 integrated dtmf_test                        005V9       1
11006                 integrated please_wait                      005V9       1
11007                 integrated REFER_Test                       005V9       1
```

5.9.2 Post-Answer Redirection to a PSTN Destination

This section provides an example configuration of a vector that will use post-answer redirection to a PSTN destination. A corresponding detailed verification is provided in **Section 9.1.2**. In this example, the inbound toll-free call is routed to VDN 10001 shown in the following screen. The originally dialed Verizon IP Toll Free number may be mapped to VDN 10001 by Session Manager digit conversion, or via the incoming call handling treatment for the Communication Manager trunk group handling the call.

```
display vdn 10001                                               Page   1 of   3
                              VECTOR DIRECTORY NUMBER

                               Extension: 10001
                                   Name*: Refer-to-PSTN
                             Destination: Vector Number        2
                     Attendant Vectoring? n
                    Meet-me Conferencing? n
                      Allow VDN Override? n
                                     COR: 1
                                     TN*: 1
                                Measured: none
```

VDN 10001 is associated with vector 2, which is shown below. Vector 2 plays an announcement (step 03) to answer the call. After the announcement, the **route-to number** (step 05) includes **~r+17863310799** where the number 786-331-0799 is a PSTN destination. This step causes a REFER message to be sent where the Refer-To header includes "+17863310799" as the user portion. Note that Verizon Business IPCC Services require the "+" in the Refer-To header for this type of call redirection.

```
display vector 2                                                Page   1 of   6
                                    CALL VECTOR

    Number: 2                 Name: Refer-to-PSTN
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 #    Play announcement to caller in step 3. This answers the call.
03 announcement 11006
04 #    Refer the call to PSTN Destination in step 5 below.
05 route-to     number ~r+17863310799   with cov n if unconditionally
06 #    If Refer fails queue to skill 1
07 queue-to     skill 1    pri m
08
```

5.9.3 Post-Answer Redirection With UUI to a SIP Destination

This section provides an example of post-answer redirection with UUI passed to a SIP destination. In this example, the inbound call is routed to VDN 10003 shown in the following screen. The originally dialed Verizon toll-free number may be mapped to VDN 10003 by Session Manager digit conversion, or via the incoming call handling treatment for the Communication Manager trunk group handling the call.

```
display vdn 10003                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 10003
                                 Name*: REFER with UUI
                           Destination: Vector Number        3
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

To facilitate testing of NCR with UUI, the following vector variables were defined.

```
change variables                                                Page   1 of  39
                              VARIABLES FOR VECTORS

Var Description                     Type    Scope Length Start Assignment       VAC
A   uui                             asaiuui L     16     1
B   uui                             asaiuui L     16     17
C
```

VDN 10003 is associated with vector 3, which is shown below. Vector 3 sets data in the vector variables A and B (steps 03 and 04) and plays an announcement to answer the call (step 05). After the announcement, the **route-to** number step includes **~r+18668512649**. This step causes a REFER message to be sent where the Refer-To header includes **+18668512649** as the user portion. The Refer-To header will also contain the UUI set in variables A and B. Verizon will include this UUI in the INVITE ultimately sent to the SIP-connected target of the REFER, which is toll-free number “18668512649”. In the sample configuration, where only one location was used, 866-851-2649 is another toll-free number assigned to the same circuit as the original call. In practice, NCR with UUI would allow Communication Manager to send call or customer-related data along with the call to another contact center.

```
display vector 3                                                Page   1 of   6
                                  CALL VECTOR

    Number: 3                Name: Refer-with-UUI
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 set          A      = none   CATR  1234567890123456
03 set          B      = none   CATR  7890123456789012
04 #    Play announcement to answer call and route to ~r to cause Refer
05 announcement 11007
06 route-to     number ~r+18668512649   with cov n if unconditionally
07 #    If Refer fails play announcement and disconnect
08 disconnect   after announcement 11003
```

5.9.4 ACD Configuration for Call Queued for Handling by Agent

This section provides a simple example configuration for VDN, vector, hunt group, and agent logins used to queue inbound Verizon IPCC calls for handling by an agent.

The following screens show an example ACD hunt group. On page 1, note the bolded values.

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

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

             Queue Limit: unlimited
```

The following screens show an example ACD hunt group. On the abbreviated page 2 shown below, note Skill is set to **y**.

```
display hunt-group 1                                            Page   2 of   4
                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n        Service Level Target (% in sec): 80 in 20
```

VDN 10004, shown below, is associated with vector 4.

```
display vdn 10004                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 10004
                                 Name*: Sales
                           Destination: Vector Number        4
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
```

In this simple example, vector 4 briefly plays ring back, then queues the call to skill 1. Announcement 11004 is a simple recurring announcement. If an agent is immediately available to handle the call, the call will be delivered to the agent. If an agent is not immediately available, the call will be queued, and the caller will hear the announcement. Once an agent becomes available, the call will be delivered to the agent.

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

    Number: 4                Name: Sales
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 #    Wait hearing ringback
02 wait-time    2   secs hearing ringback
03 #    Simple queue to skill with recurring announcement until available
04 queue-to     skill 1    pri m
05 announcement 11004
06 wait-time    30  secs hearing music
07 goto step    5             if unconditionally
08 stop
```

The following screen illustrates an example agent-loginID 20001. In the sample configuration, an Avaya IP Deskphone logged in using agent-loginID 20001 and the configured Password to staff and take calls for skill 1.

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

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

                                           LoginID for ISDN/SIP Display? n
                                                               Password:
                                                Password (enter again):
                                                            Auto Answer: station
                                                      MIA Across Skills: system
                                              ACW Agent Considered Idle: system
                                              Aux Work Reason Code Type: system
                                                Logout Reason Code Type: system
```

The following abridged screen shows Page 2 for agent-loginID 20001. Note that the Skill Number (**SN**) has been set to 1.

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

    SN   RL SL          SN   RL SL
 1: 1       1       16:                 31:                 46:
 2:                 17:                 32:                 47:
 3:                 18:                 33:                 48:
```

To enable a telephone or one-X® Agent client to log in with the agent-loginID shown above, ensure that **Expert Agent Selection (EAS) Enabled** is set to **y** as shown in the screen below.

```
change system-parameters features                               Page  11 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
  EAS
         Expert Agent Selection (EAS) Enabled? y
        Minimum Agent-LoginID Password Length: 4
```

5.10. 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 Verizon public numbers, for inclusion in any SIP headers directed to the Verizon Business IPCC Services 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 Vector Directory Numbers (VDN) and their corresponding Verizon DNIS numbers (for the public trunk to Verizon). Communication Manager will insert these Verizon DNIS numbers in E.164 format into the From, Contact, and PAI headers as appropriate:

- **Ext Len** – Enter the total number of digits in the local extension range (e.g., **5**).
- **Ext Code –** Enter a Communication Manager extension (e.g., **10001**).
- **Trk Grp(s)** – Enter the number of the Public trunk group (e.g., **2**).
- **Private Prefix** – Enter the corresponding Verizon DNIS number (e.g., **18668523221**).
- **Total Len** – Enter the total number of digits after the digit conversion (e.g., **11**).

```
change public-unknown-numbering 0                               Page   1 of   2
                            NUMBERING - PUBLIC/UNKNOWN FORMAT
                                                Total
Ext Ext             Trk       CPN                CPN
Len Code            Grp(s)    Prefix             Len
                                                     Total Administered: 16
 5  10001           2         18668523221       11      Maximum Entries: 240
 5  10003           2         18668510107       11
 5  10004           2         18668502380       11   Note: If an entry applies to
```

Note – Without this configuration, calls to the VDNs would result in a 5-digit user portion of the Contact header in the 183 with SDP and 200 OK returned to Verizon. Although this did not present any user-perceivable problem in the sample configuration, the configuration in the bolded rows above illustrate how to cause Communication Manager to populate the Contact header with user portions that correspond with a Verizon Business IPCC number. In the course of the testing, multiple Verizon toll-free numbers were associated with different Communication Manager extensions and functions.

5.11. 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 6.3** (e.g., **14** and **20**).
- **Trk Grp(s)** – Enter the number of the Local trunk group (e.g., **3**).
- **Total Len** - Enter the total number of digits after the digit conversion (e.g., **5**).

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

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  10               3                           5     Total Administered: 6
 5  11               3                           5        Maximum Entries: 540
 5  12               3                           5
 5  14               3                           5
 5  20               3                           5
```

5.12. Route Patterns

Route Patterns are used to direct outbound calls via the public or local CPE SIP trunks. Since Verizon Business IPCC is an inbound only service, no route pattern is defined for outbound calls on the public trunk.

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

- In the **Grp No** column enter **3** for SIP trunk 3 (local trunk).
- In the **FRL** column enter **0** (zero).
- In the **Numbering Format** column, across from line **1**, enter **lev0-pvt**.

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

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

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 50xxx, therefore enter **50**.
- **Min** & **Max** – Enter **5**
- **Route Pattern** – Enter **3**
- **Call Type** – Enter **lev0**

Step 2 - Repeat **Step 1**, and create entries for other different SIP extension ranges, Messaging access extension**,** etc. as needed.

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

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

5.14. Avaya G430 Media Gateway Provisioning

In the reference configuration, a G430 Media Gateway is provisioned. The G430 is located in the Main site and is used for local DSP resources, announcements, Music On Hold, etc.

Note – Only the Media Gateway provisioning associated with the G430 registration to Communication Manager is shown below. For additional information on G450 provisioning, see Error! Reference source not found..

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 Processor Ethernet (e.g., **10.64.91.65**, see **Section 5.4**).
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 .24  .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 located in the Main site and 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 Error! Reference source not found. and **[10]** for additional information.

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

Step 2 - On the Media Server Element Manager, navigate to **Home → System Configuration → Signaling Protocols → SIP →Node and Routes** and add the Communication Manager Procr interface IP address (e.g., **10.64.91.65**, see **Section 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., **AMS801**).
- **Near-end Listen Port** and **Far-end Listen Port** – The default ports **9061** and **5061** are used. These ports may be changed to other values if desired.
- **Far-end Network Region** – Set the IP network region to **1**, as set in **Section 5.7.1**.
- **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: AMS801
 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: AMS801
                     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**.

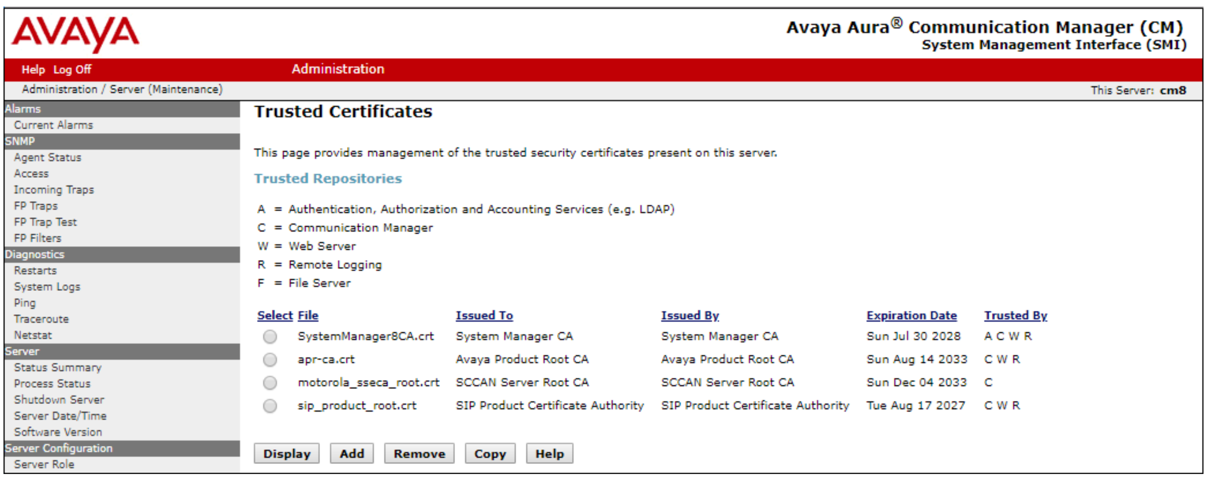
5.17. Verify TLS Certificates – Communication Manager

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.

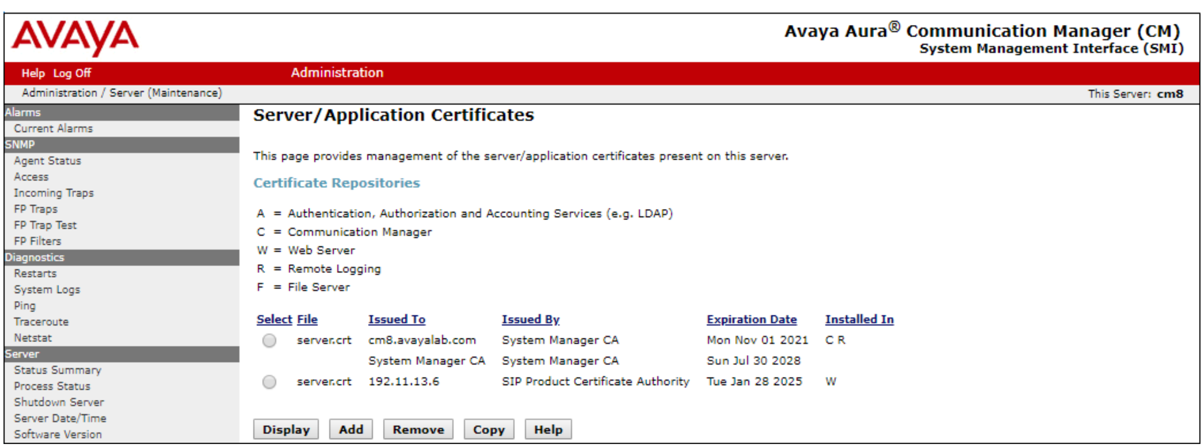
In the reference configuration, TLS transport is used for the communication between Session Manager and Communication Manager. The following procedures show how to verify the certificates used by Communication Manager.

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

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



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



6. Configure Avaya Aura® Session Manager

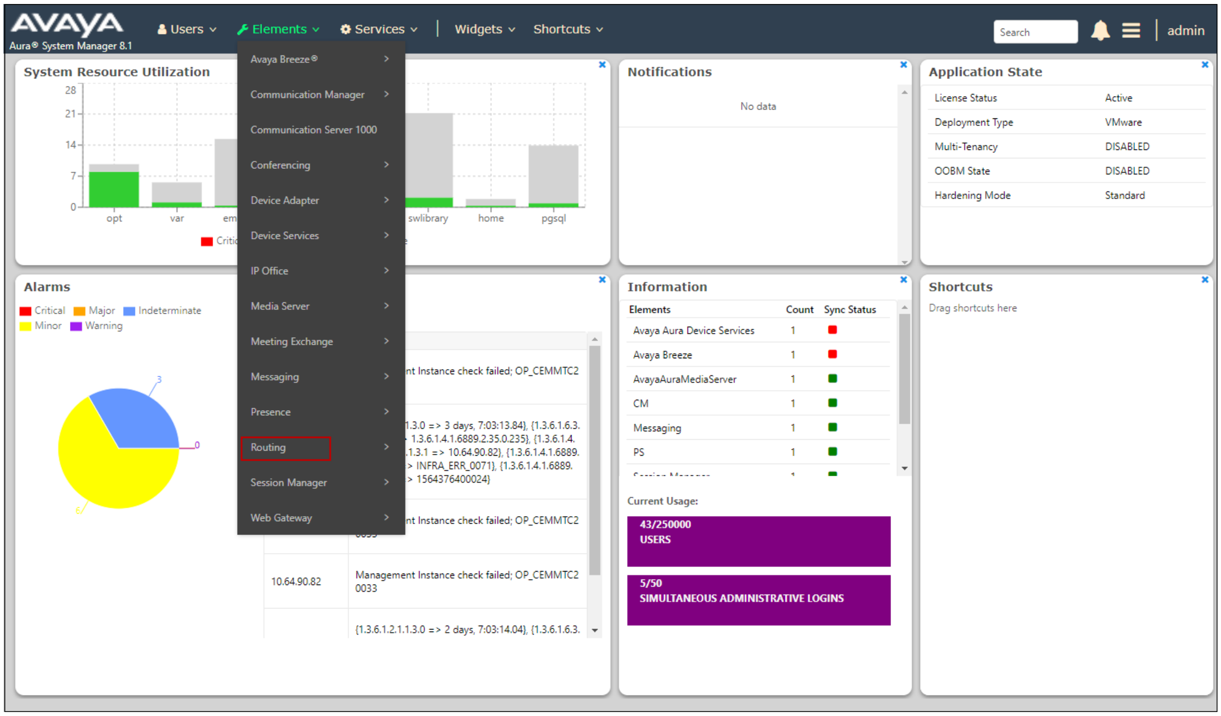
This section provides the procedures for configuring Session Manager to process inbound and outbound calls between Communication Manager and the Avaya SBCE. In the reference configuration, all Session Manager provisioning is performed via System Manager.

- 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 and Messaging.
- Define Entity Links describing the SIP trunks between Session Manager, Communication Manager and Messaging, as well as the SIP trunks between the Session Manager and the Avaya SBCE.
- Define Routing Policies associated with Communication Manager, Messaging and the Avaya SBCE.
- Define Dial Patterns, which govern which Routing Policy will be selected for inbound and outbound call routing.
- Verify TLS Certificates.

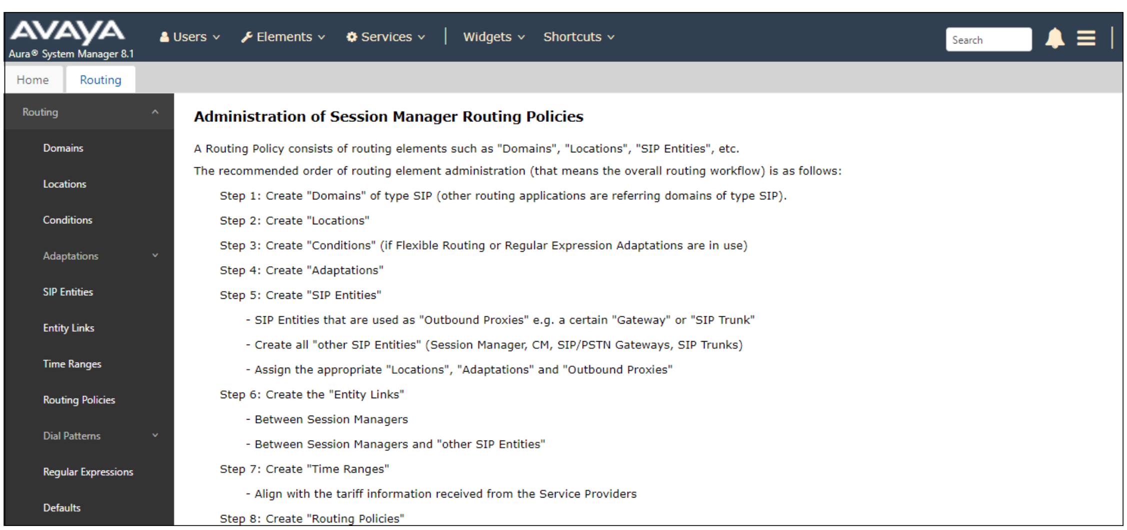
Note – These Application Notes assume that basic System Manager and Session Manager administration has already been performed. Consult **[1]- [4]** in the Additional References section for further details.

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.



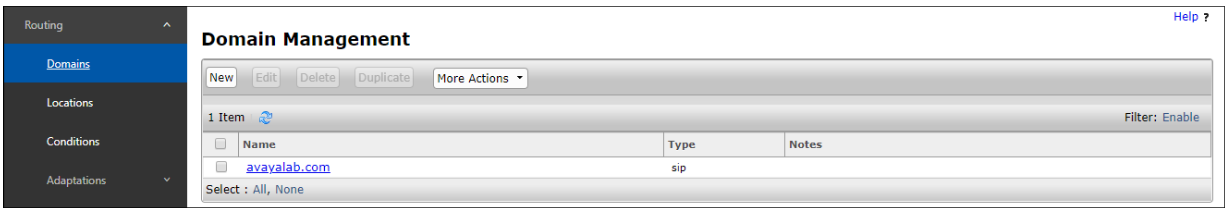
6.2. SIP Domain

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

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

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

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



6.3. Locations

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

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

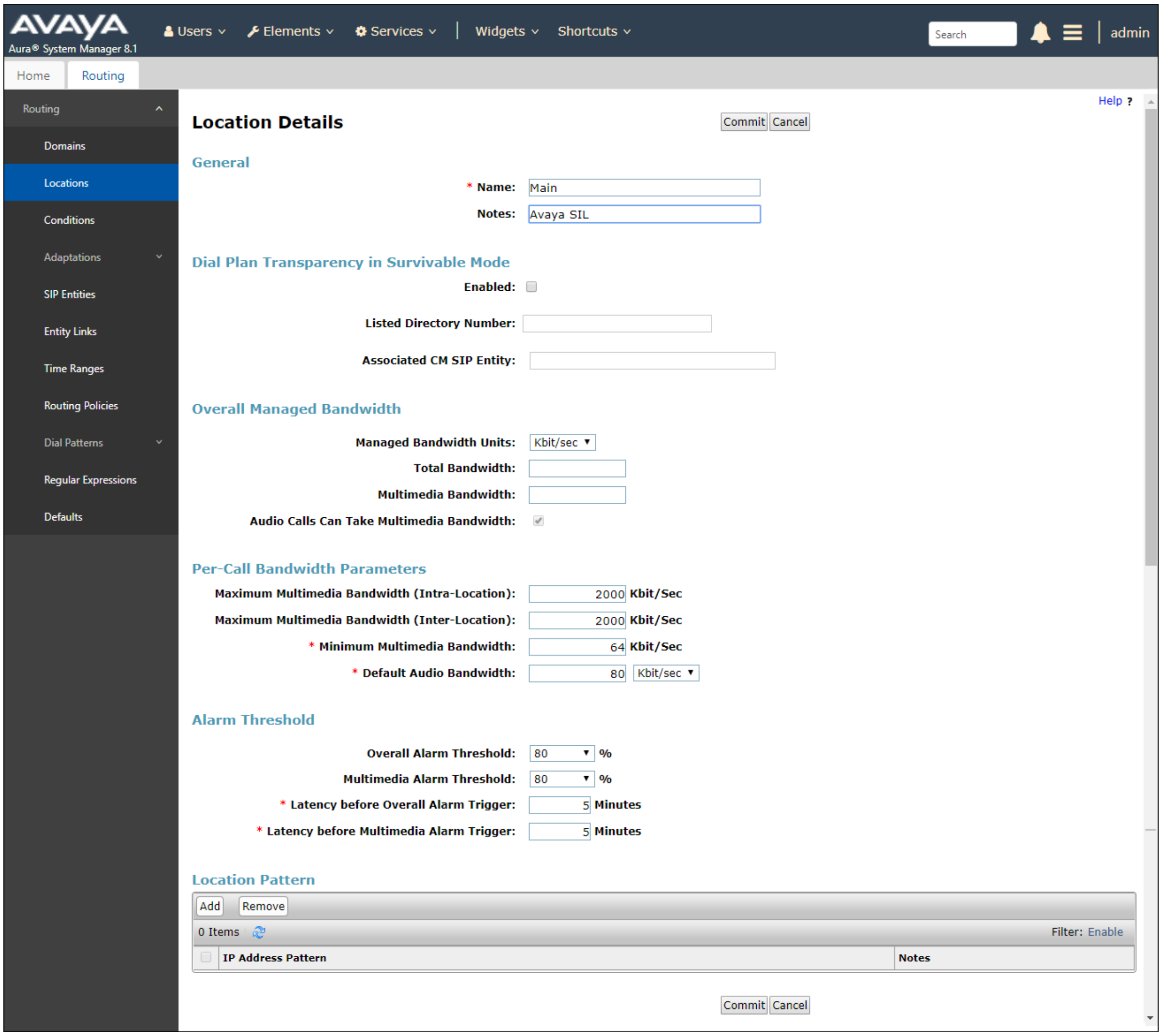
6.3.1 Main Location

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 – Default values are used on all the remaining fields.

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



6.3.2 Common Location

To configure the Avaya SBCE Location, repeat the steps in **Section 6.3.1** with the following changes:

- **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 from Verizon to Communication Manger.

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

6.4.1 Adaptation for Avaya Aura® Communication Manager Extensions

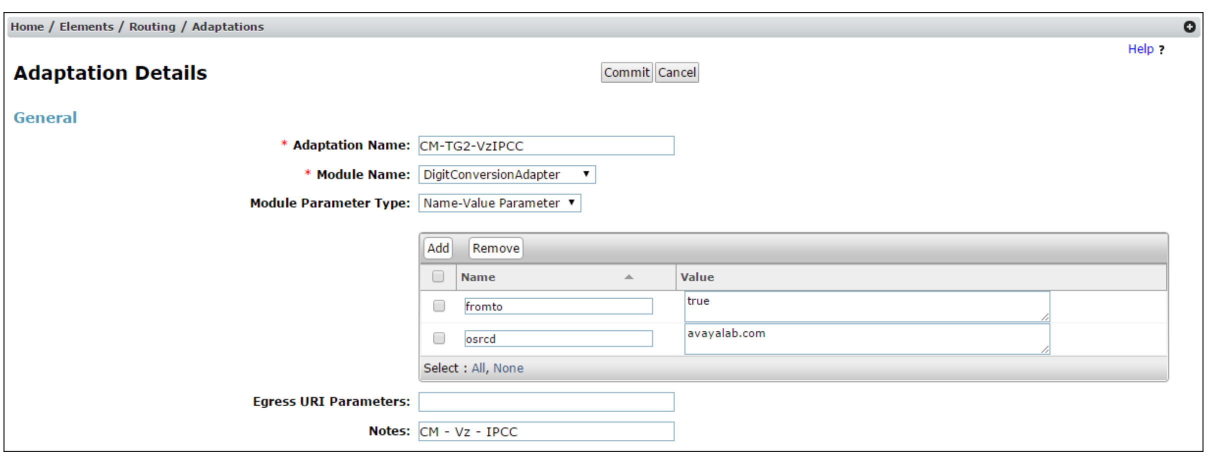
The Adaptation administered in this section is used for modification of SIP messages to Communication Manager extensions from Verizon.

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

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

1. A descriptive **Name**, (e.g., **CM-TG2-VzIPCC**).
2. Select **DigitConversionAdapter** from the **Module Name** drop down.
3. Select **Name-Value Parameter** from the **Module Parameter Type** drop down:
   - **Name**: **fromto** **Value**: **true**
     - This adapts the From and To headers along with the Request-Line and PAI headers.
   - **Name**: **osrcd** **Value**: **avayalab.com**
     - This enables the source domain to be overwritten with "avayalab.com". For example, for inbound PSTN calls from Verizon to Communication Manager, the PAI header will contain "avayalab.com".

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



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

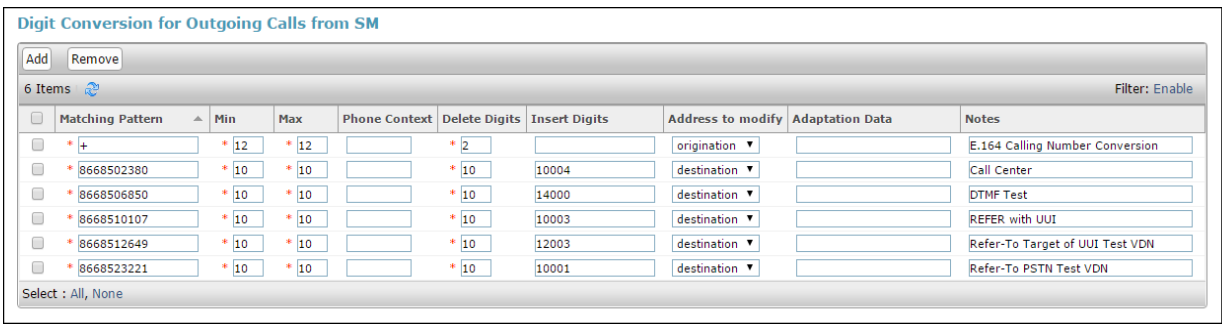
Example – destination extension: 8668502380 is a DNIS string sent in the Request URI by the Verizon Business IPCC Services that is associated with Communication Manager VDN 10004.

- Enter **8668502380** in the **Matching Pattern** column.
- Enter **10** in the **Min/Max** columns.
- Enter **10** in the **Delete Digits** column.
- Enter **10004** in the **Insert Digits** column.
- Specify that this should be applied to the SIP **destination** headers in the **Address to modify** column.
- Enter any desired notes.

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

Step 5 - Click on **Commit**.

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



Digit Conversion for Outgoing Calls from SM

Add Remove

6 Items Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* +	* 12	* 12		* 2		origination		E.164 Calling Number Conversion
	* 8668502380	* 10	* 10		* 10	10004	destination		Call Center
	* 8668506850	* 10	* 10		* 10	14000	destination		DTMF Test
	* 8668510107	* 10	* 10		* 10	10003	destination		REFER with UUI
	* 8668512649	* 10	* 10		* 10	12003	destination		Refer-To Target of UUI Test VDN
	* 8668523221	* 10	* 10		* 10	10001	destination		Refer-To PSTN Test VDN

Select : All, None

6.4.2 Adaptation for the Verizon Business IPCC Services

The Adaptation administered in this section is used for modification of SIP messages from Communication Manager to Verizon. Repeat the steps in **Section 6.4.1** with the following changes.

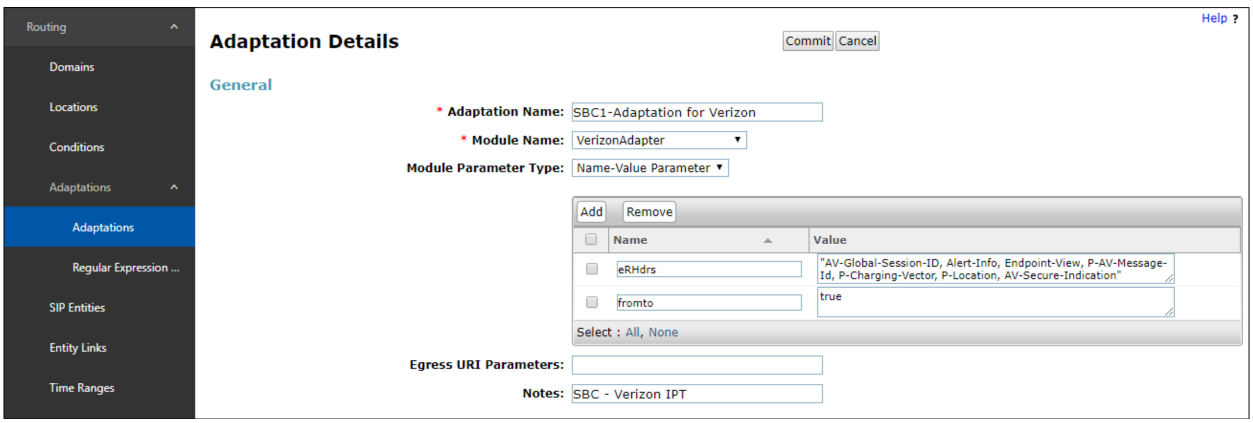
Step 1 - In the **Adaptation Details** page, enter:

1. A descriptive **Name**, (e.g., **SBC1-Adaptation for Verizon**).
2. Select **VerizonAdapter** from the **Module Name** drop down menu.

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

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

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



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 Verizon trunk access (**Section 6.5.2**) – This entity, and its associated Entity Link (using TLS with port 5071), is for calls from Verizon 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 from the Verizon Business IPCC Services 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.

Note – In the reference configuration, TLS is used as the transport protocol between Session Manager and Communication Manager (ports 5061 and 5071), and to the Avaya SBCE (port 5061). The connection between the Avaya SBCE and the Verizon Business IPCC Services uses UDP/5072 per Verizon 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:

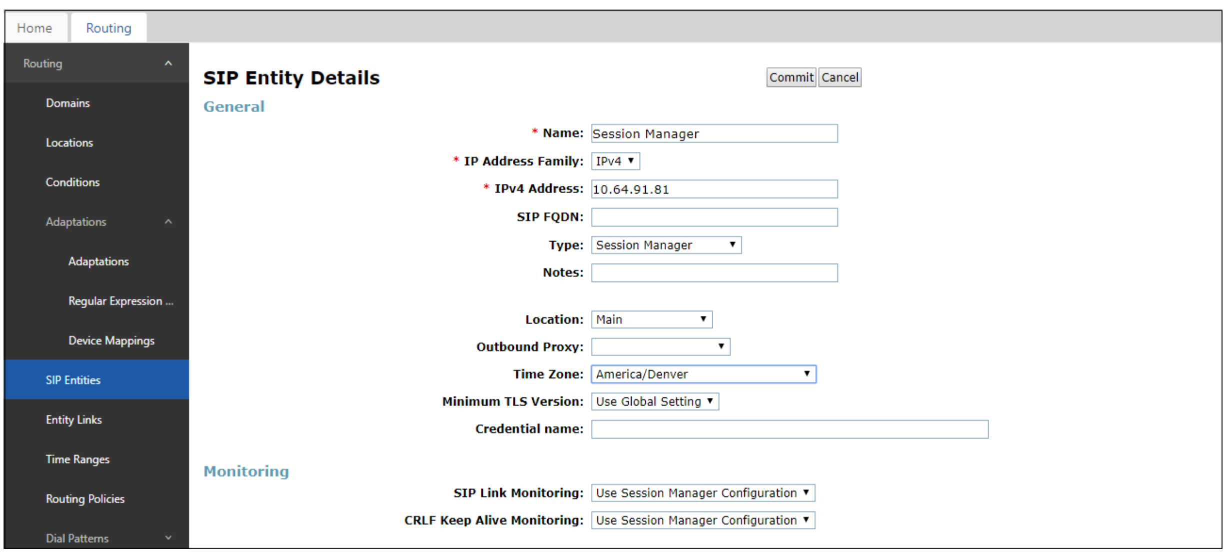
- **Name –** Enter a descriptive name (e.g., **Session Manager**).

Note – the **IP Address Family** in the SIP Entity form only needs to be specified if both IPv4 and IPv6 addresses have been enabled in Session Manager. If only IPv4 is enabled, the **IP Address Family** field is not present, and the **IPv4 Address** field is named **IP Address**.

- **IP Address Family** – Select **IPv4**.
- **IPv4 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 **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:

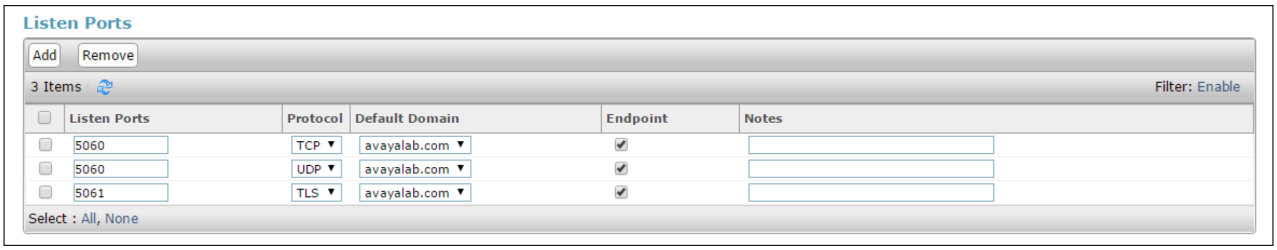
- **Port –** Enter **5061**
- **Protocol –** Select **TLS**
- **Default Domain –** Select a SIP domain administered in **Section 6.2** (e.g., **avayalab.com**)
- Check **Endpoint**.

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

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

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

Step 7 - Click on **Commit**.


Listen Ports

Add Remove

3 Items Filter: Enable

	Listen Ports	Protocol	Default Domain	Endpoint	Notes
	5060	TCP	avayalab.com	✔	
	5060	UDP	avayalab.com	✔	
	5061	TLS	avayalab.com	✔	

Select : All, None

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

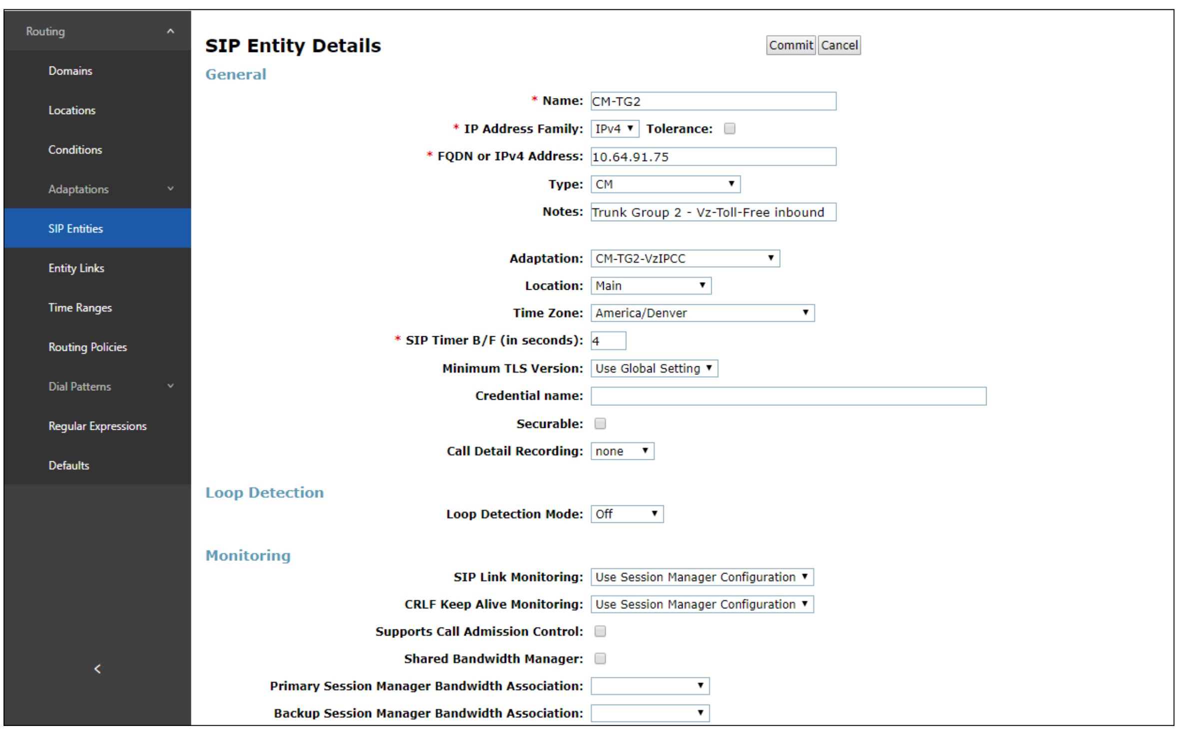
6.5.2 Avaya Aura® Communication Manager SIP Entity – Public Trunk

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

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

- **Name –** Enter a descriptive name (e.g., **CM-TG2**).
- **IP Address Family** – Select **IPv4**.
- **FQDN or IPv4 Address –** Enter the IP address of Communication Manager Processor Ethernet (procr) described in **Section 5.4** (e.g., **10.64.91.75**).
- **Type** – Select **CM**.
- **Adaptation** – Select the Adaptation **CM-TG2-VzIPCC** administered in **Section 6.4.1**.
- **Location** – Select a Location **Main** administered in **Section 6.3.1**.
- **Time Zone** – Select the time zone in which Communication Manager resides.
- In the **SIP Link Monitoring** section of the **SIP Entity Details** page select:
    - Select **Use Session Manager Configuration** for **SIP Link Monitoring** and the **CRLF Keep Alive Monitoring** fields. Use the default values for the remaining parameters.

Step 3 - Click on **Commit**.



6.5.3 Avaya Aura® Communication Manager SIP Entity – Local Trunk

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

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

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., **SBC1**).
- **FQDN or IP Address –** Enter the IP address of the A1 (private) interface of the Avaya SBCE (e.g., **10.64.91.50**, see **Section 7.5**).
- **Type –** Select **SIP Trunk**.
- **Adaptations –** Select Adaptation **SBC1-Adaptation for Verizon** (**Section 6.4.2**).
- **Location –** Select Location **Common-SBCs** administered in **Section** Error! Reference source not found..

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**).
- **Type –** Select **Messaging**.
- **Adaptations –** Leave this field blank.

6.6. Entity Links

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

- Session Manager to Communication Manager Public trunk (**Section** Error! Reference source not found.).
- 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**).

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**.

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

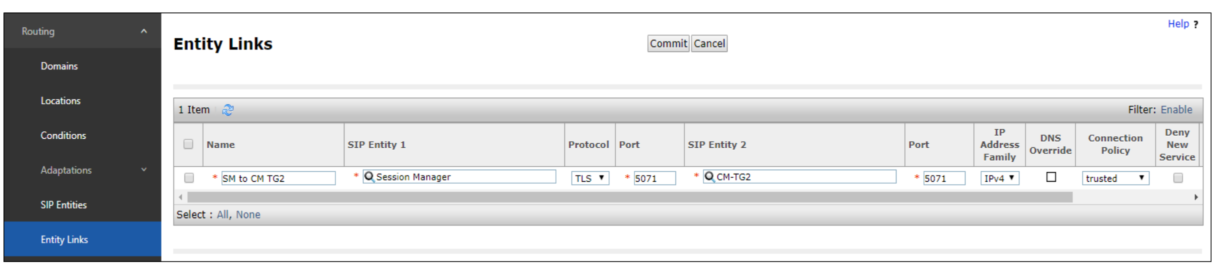
Step 1 - In the left pane under **Routing**, click on **Entity Links**, then click on **New** (not shown).
Step 2 - Continuing in the **Entity Links** page, provision the following:

- **Name** – Enter a descriptive name for this link to Communication Manager (e.g., **SM to CM TG2**).
- **SIP Entity 1** – Select the SIP Entity administered in **Section 6.5.1** for Session Manager (e.g., **Session Manager**).
- **Protocol** – Select **TLS** (see **Section 5.8.1**).
- SIP Entity 1 **Port** – Enter **5071**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.5.2** for the Communication Manager public entity (e.g., **CM-TG2**).
- SIP Entity 2 **Port** – Enter **5071** (see **Section 5.8.1**).
- **IP Address Family –** Select **IPv4**.

Note – the **IP Address Family field** is only present in the Entity Link form if both IPv4 and IPv6 addresses have been enabled in Session Manager.

- **Connection Policy** – Select **trusted**.
- Leave other fields as default.

Step 3 - Click on **Commit**.



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

To configure this Entity Link, repeat the steps in **Section** Error! Reference source not found., 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 5.8.12**).

6.6.3 Entity Link for the Verizon Business IPCC Services via the Avaya SBCE

To configure this Entity Link, repeat the steps in **Section** Error! Reference source not found., with the following changes:

- **Name** – Enter a descriptive name for this link to the Avaya SBCE (e.g., **SM to SBC1**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.5.4** for the Avaya SBCE entity (e.g., **SBC1**).
- **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** Error! Reference source not found., with the following changes:

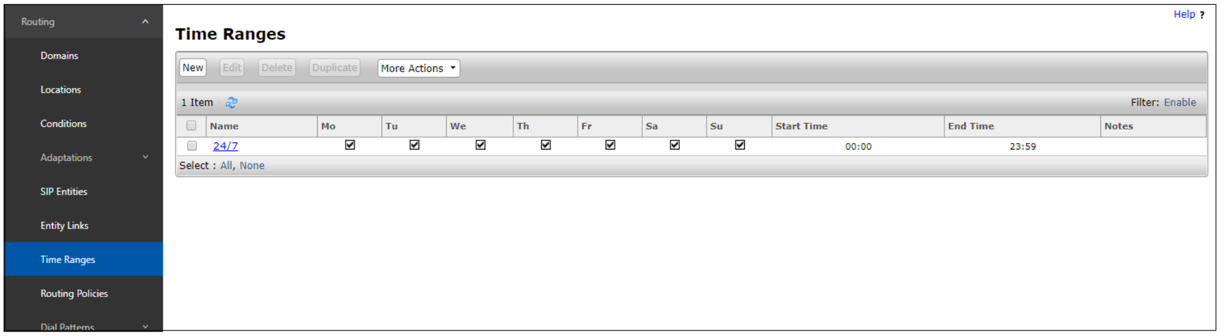
- **Name** – Enter a descriptive name for this link to Messaging (e.g., **SM to AAM**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.5.5** for the Aura® Messaging entity (e.g., **Aura Messaging**).
- **SIP Entity 2 Port** – Enter **5061**.

6.7. Time Ranges

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

Step 2 - Continuing in the **Time Ranges** page, enter a descriptive **Name**, check the checkbox(s) for the desired day(s) of the week, and enter the desired **Start Time** and **End Time**.

Step 3 - Click on **Commit** (not shown). Repeat these steps to provision additional time ranges as required.



6.8. Routing Policies

In this section, the following Routing Policies are administered:

- Inbound calls to Communication Manager extensions (**Section 6.8.1**).
- Inbound calls to Aura® Messaging (**Section 6.8.2**).

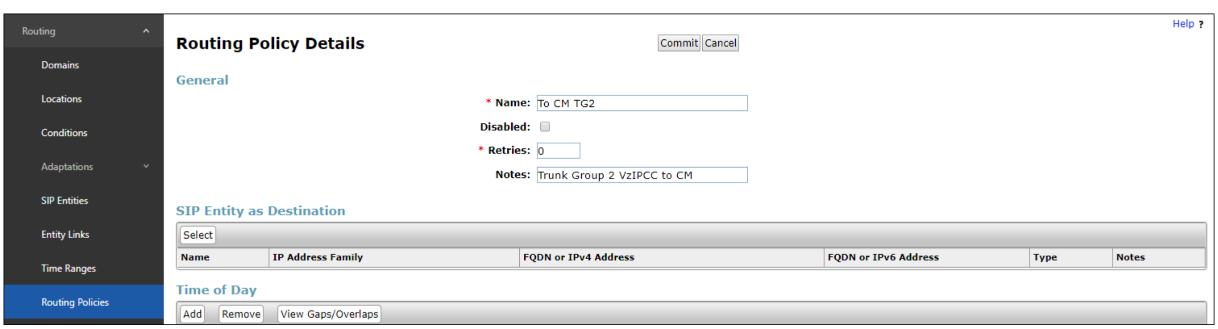
6.8.1 Routing Policy for Verizon Routing to Avaya Aura® Communication Manager

This Routing Policy is used for inbound calls from Verizon.

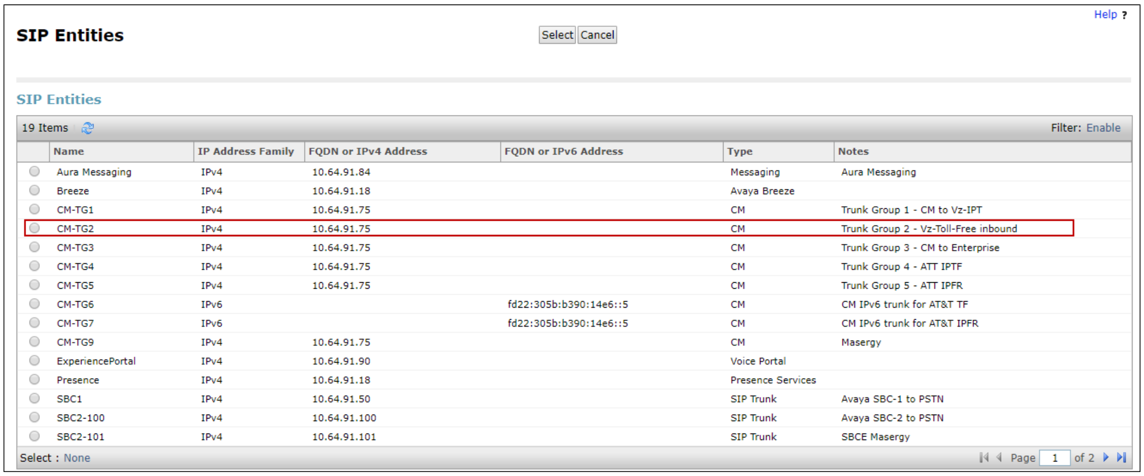
Step 1 - In the left pane under **Routing**, click on **Routing Policies**. In the **Routing Policies** page click on **New** (not shown).

Step 2 - In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing Verizon calls to Communication Manager (e.g., **To CM TG2**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.

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

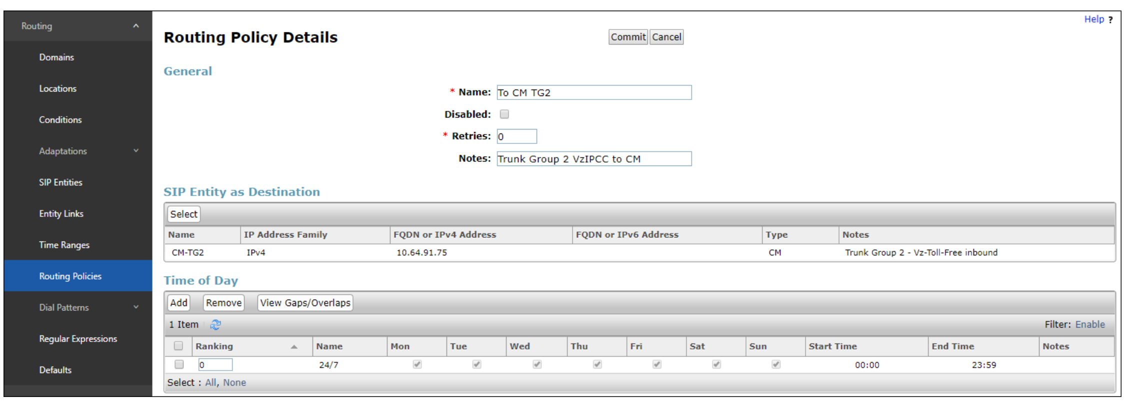


Step 4 - In the **SIP Entities** list page, select the SIP Entity administered in **Section 6.5.2** for the Communication Manager public SIP Entity (**CM-TG2**), and click on **Select**.



Step 5 - Returning to the **Routing Policy Details** page in the **Time of Day** section, click on **Add**.
Step 6 - In the **Time Range List** page (not shown), check the checkbox(s) corresponding to one or more Time Ranges administered in **Section 6.7**, and click on **Select**.
Step 7 - Returning to the **Routing Policy Details** page in the **Time of Day** section, enter a **Ranking** of **0**.
Step 8 - No **Regular Expressions** were used in the reference configuration.
Step 9 - Click on **Commit**.

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



6.8.2 Routing Policy for Inbound Routing to Avaya Aura® Messaging

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

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

6.9. Dial Patterns

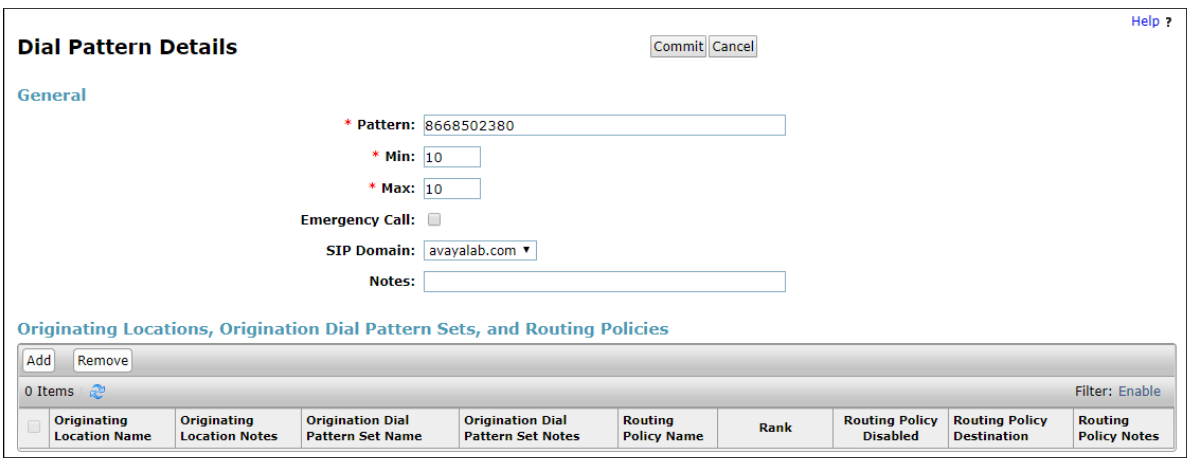
In this section, Dial Patterns are administered matching Inbound PSTN calls via the Verizon Business IPCC Services to Communication Manager.

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

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

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

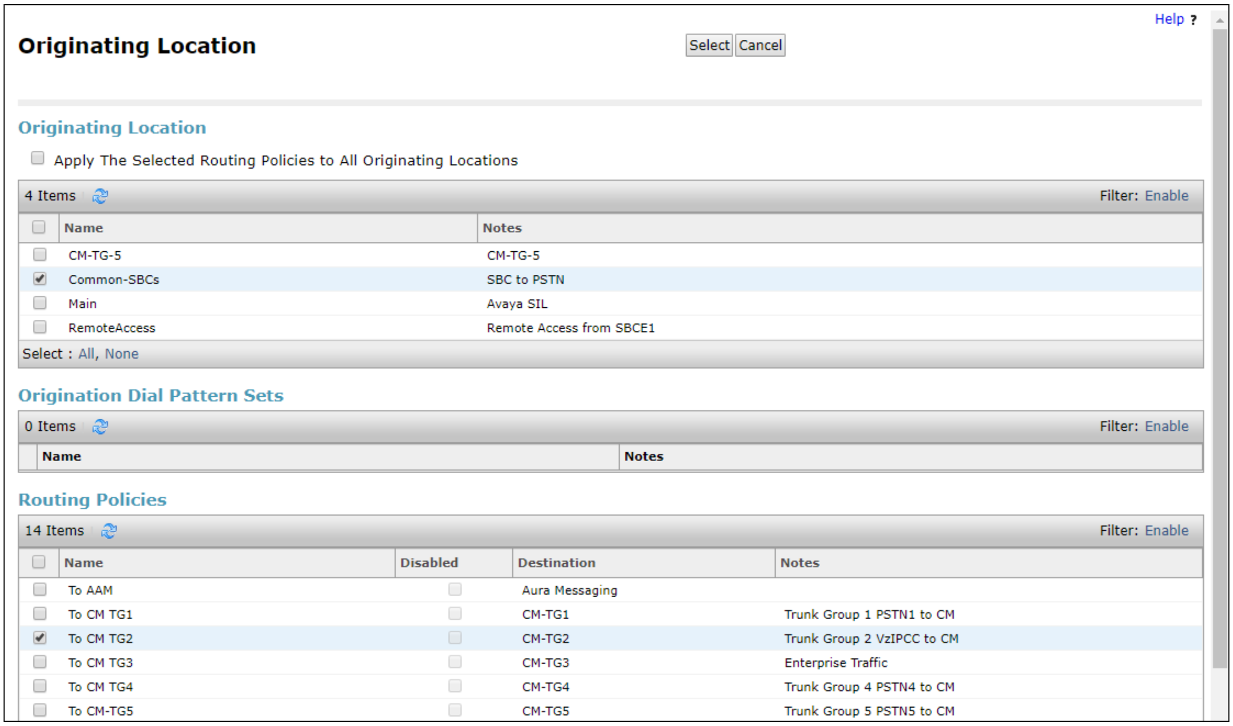
- **Pattern** – Enter **8668502380**. Note – The Adaptation defined for Communication Manager in **Section 6.4.1** will convert the various 866-xxx-xxxx toll-free numbers into their corresponding Communication Manager extensions.
- **Min** and **Max** – Enter **10**.
- **SIP Domain** – Select the enterprise SIP domain, e.g., **avayalab.com**.



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

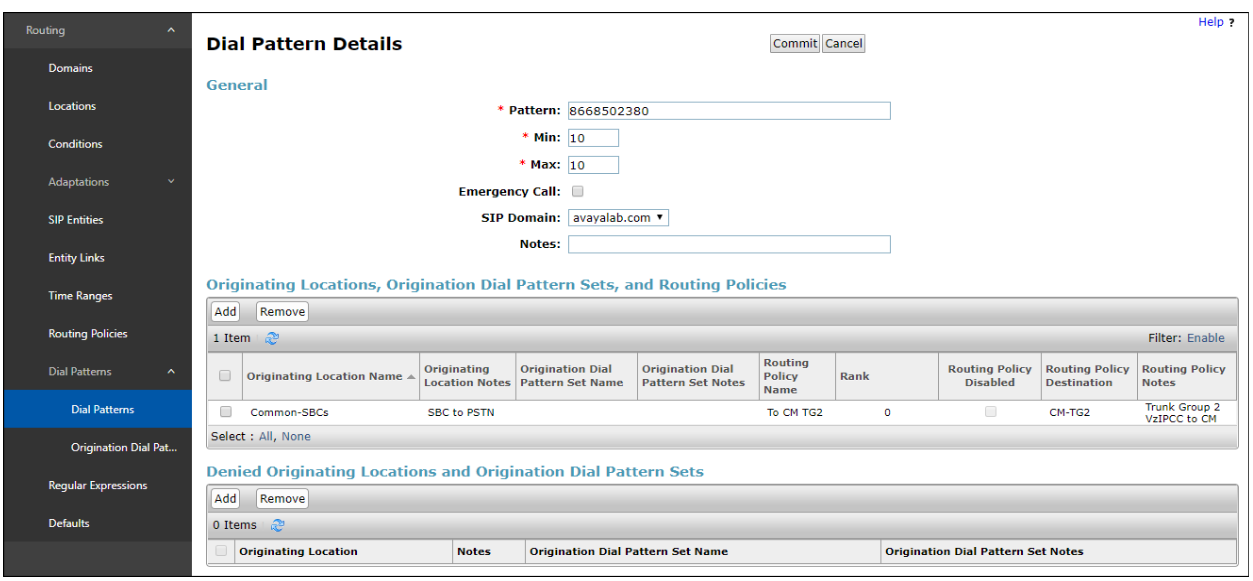
Step 4 - In the **Originating Location**, check the checkbox corresponding to the Avaya SBCE location, e.g., **Common-SBCs**.

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 TG2**), and click on **Select**.



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

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

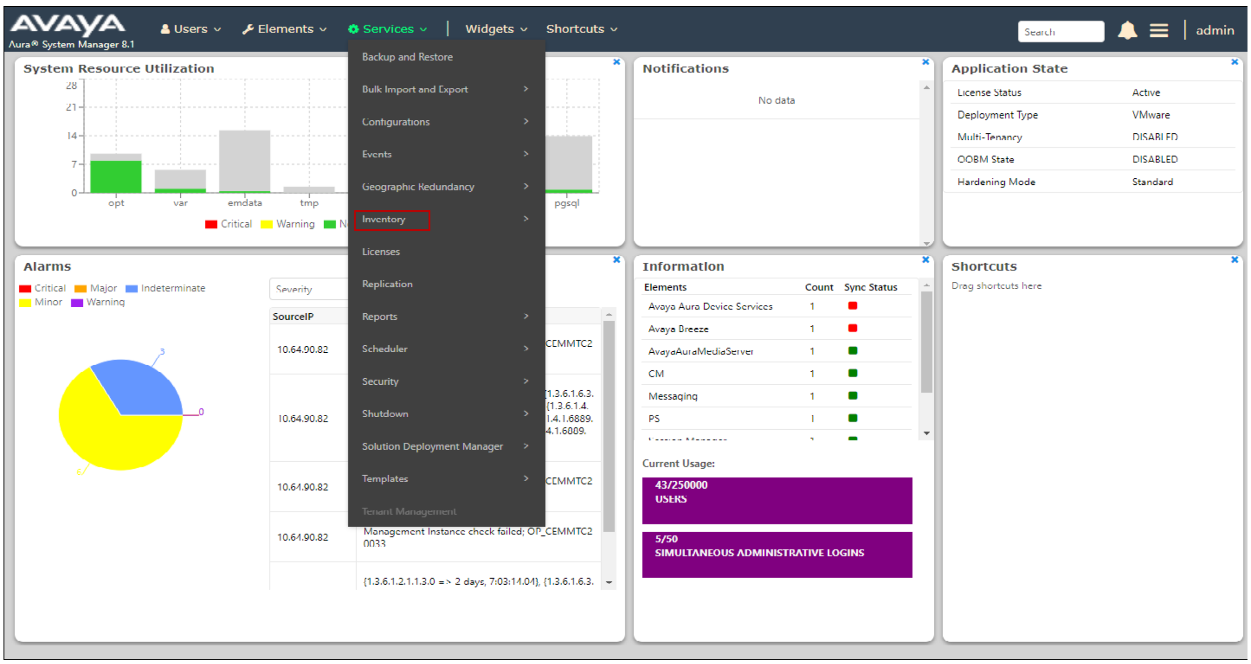


6.10. Verify TLS Certificates – Session Manager

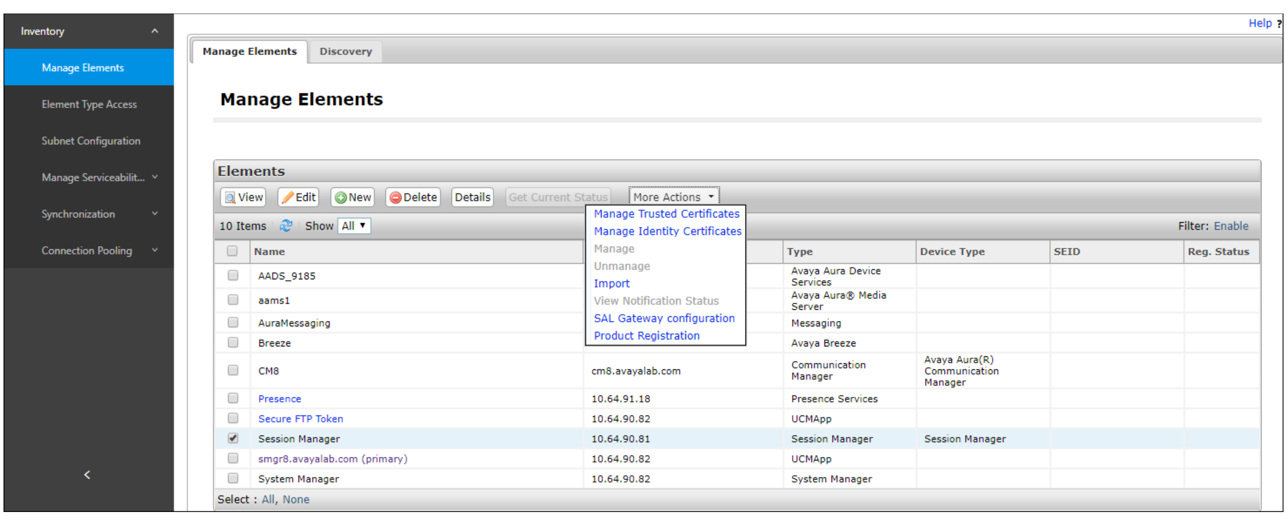
Note – Testing was done with System Manager signed identity certificates. The procedure to obtain and install certificates is outside the scope of these Application Notes.

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

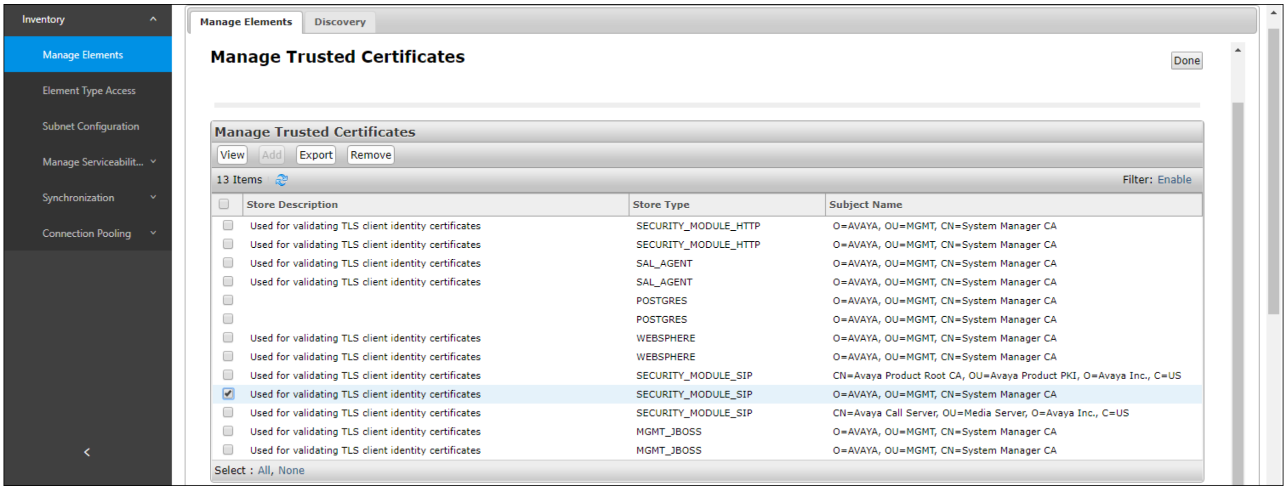
Step 1 - From the **Home** screen, under the **Services** heading, select **Inventory**.



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

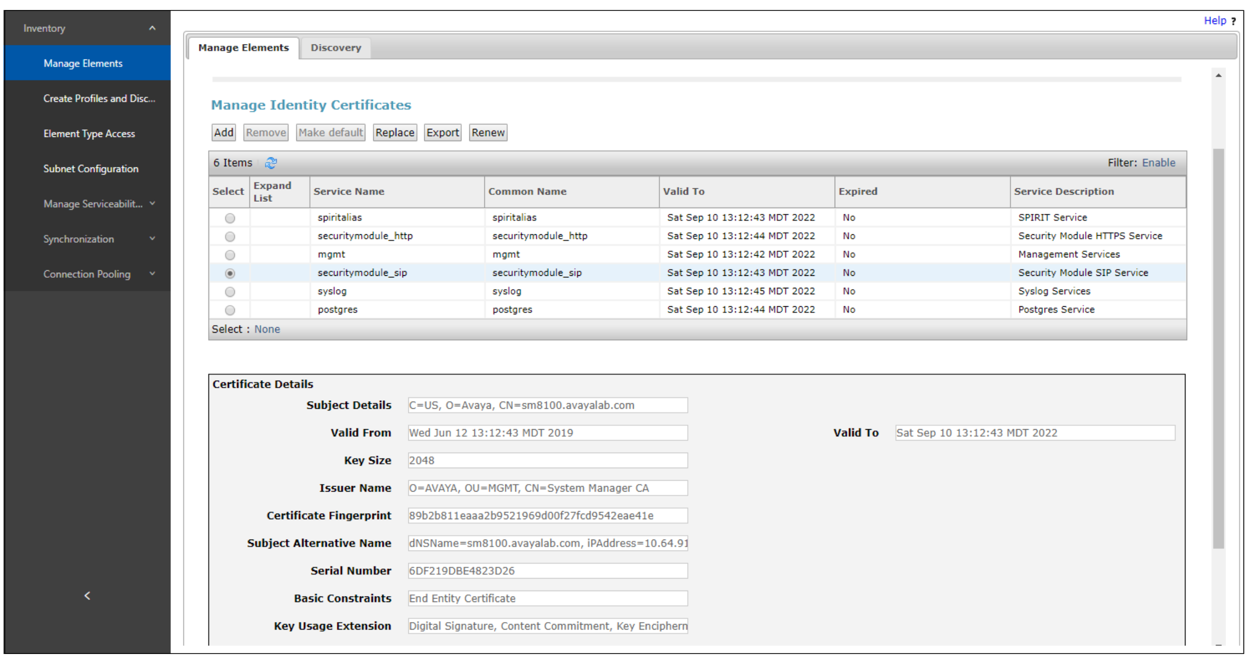


Step 3 - Verify the **System** Manager Certificate Authority certificate is listed in the trusted store, **SECURITY_MODULE_SIP**. Click **Done** to return to the previous screen.



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

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



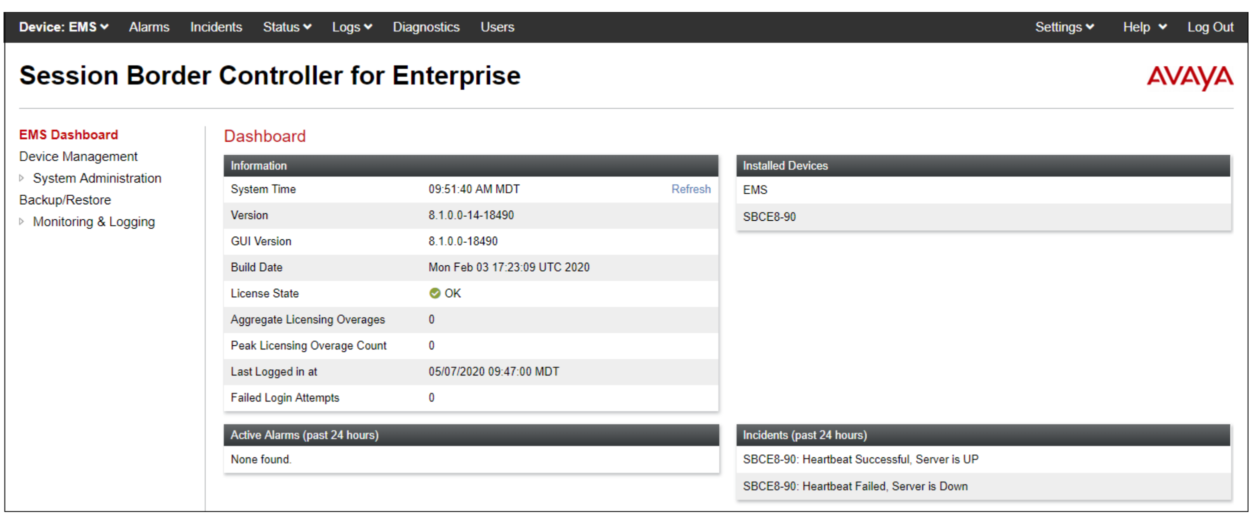
7. Configure Avaya Session Border Controller for Enterprise

This section covers the configuration of the Avaya SBCE. It is assumed that the initial provisioning of the Avaya SBCE, including the assignment of the management interface IP Address and license installation have already been completed; hence these tasks are not covered in these Application Notes. For more information on the installation and provisioning of the Avaya SBCE consult the Avaya SBCE documentation in the **Additional References** section.

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

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

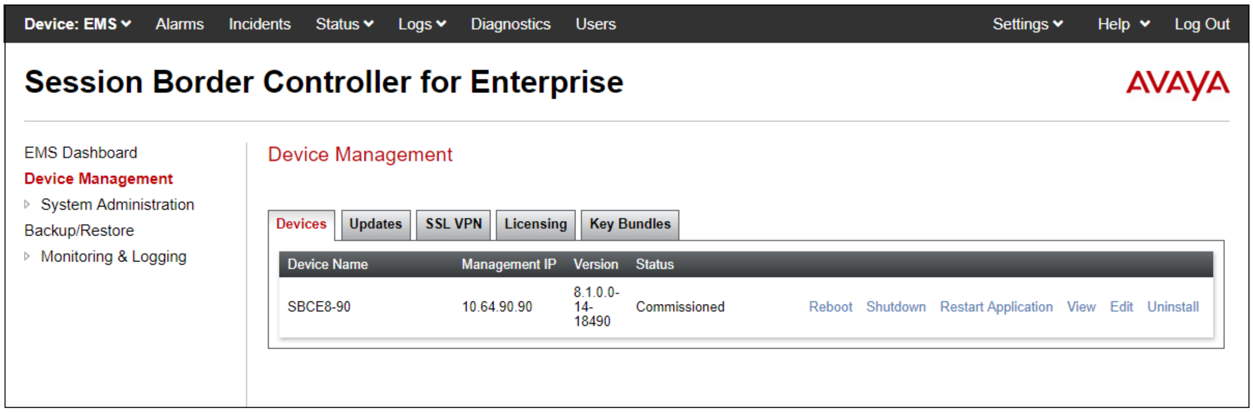
Note – The provisioning described in the following sections use the menu options listed in the left-hand column shown below.



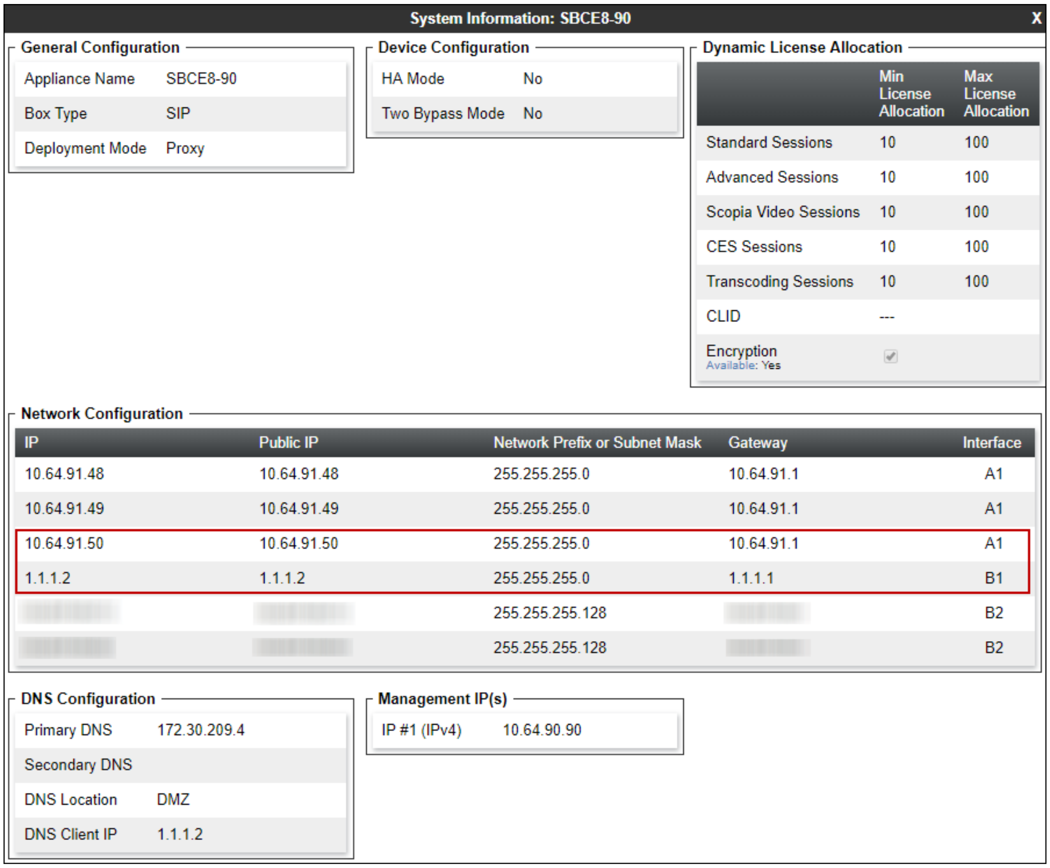
7.1. Device Management – Status

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 **SBCE8-90** is shown. Verify that the **Status** column shows **Commissioned**. If not, contact your Avaya representative. To view the configuration of this device, click **View** on the screen below.

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



The **System Information** screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation, corresponding to **Figure 1**. In the shared test environment, the highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to Verizon. Other IP addresses assigned to interfaces **A1** and **B2** on the screen below are used to support remote workers and are not the focus of these Application Notes. Note that the **Management IP** must be on a separate subnet from the IP interfaces designated for SIP traffic.



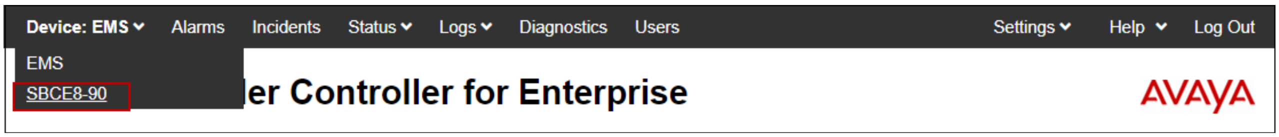
7.2. TLS Management

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

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

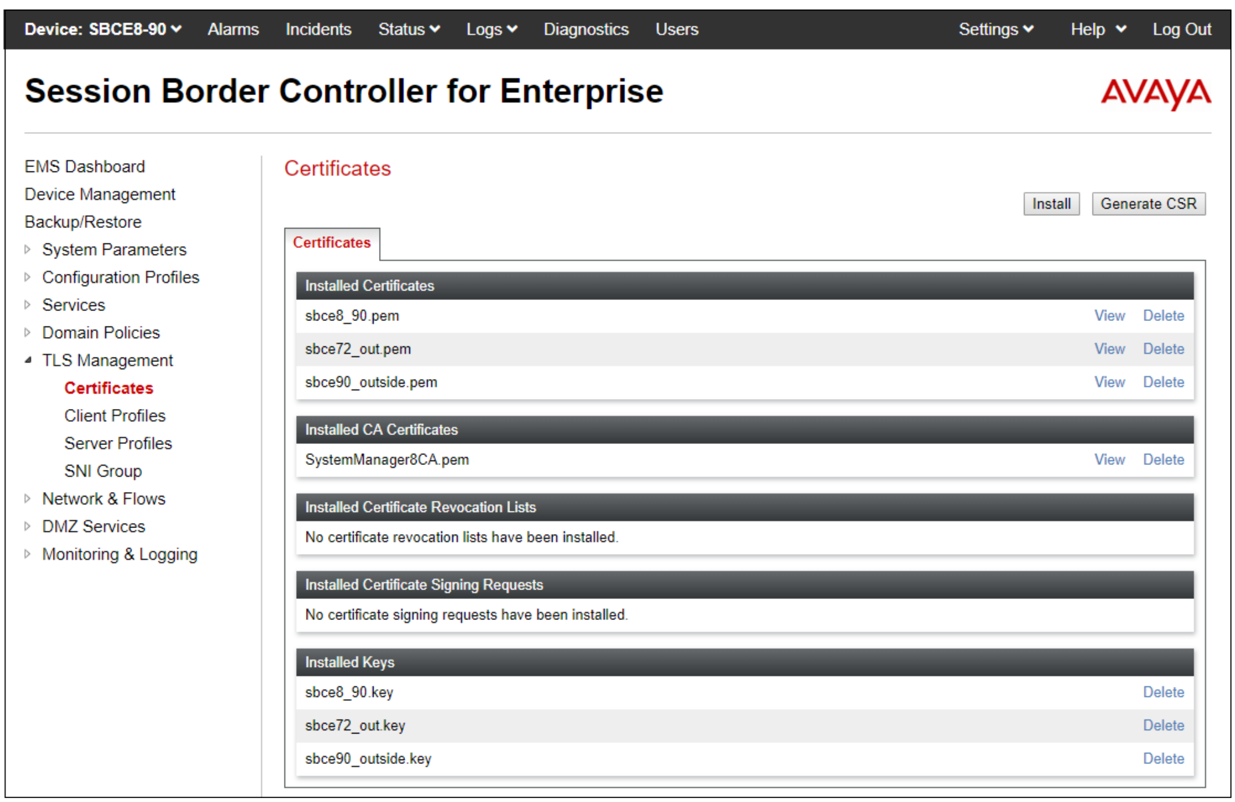
7.2.1 Verify TLS Certificates – Avaya Session Border Controller for Enterprise

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.

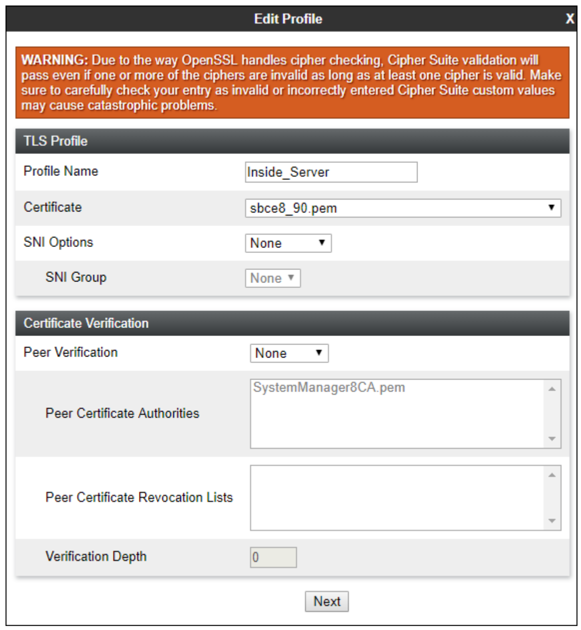


7.2.2 Server Profiles

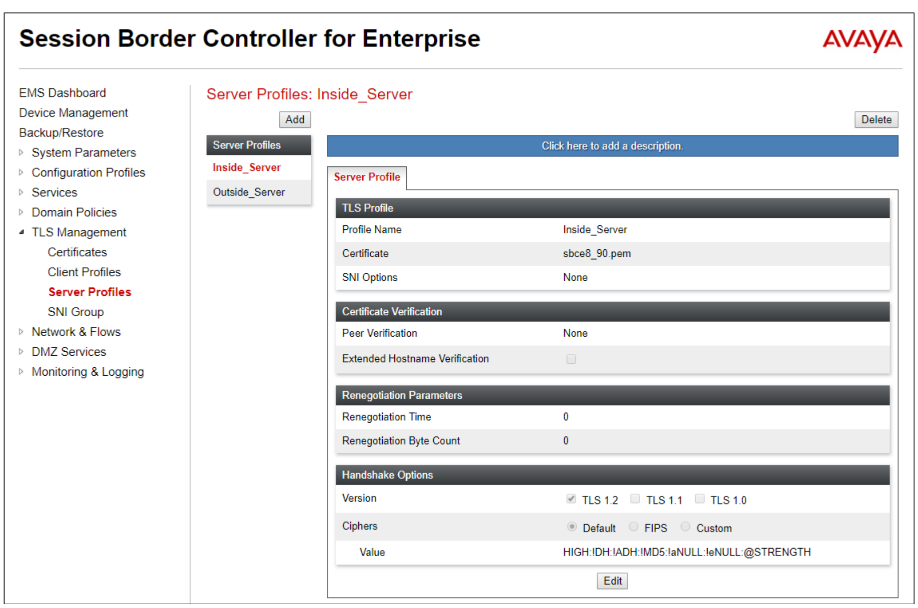
Step 1 - Select **TLS Management → Server Profiles** and click on **Add**. Enter the following:

- **Profile Name:** enter descriptive name.
- **Certificate:** select the identity certificate, e.g., **sbce8_90.pem**, from pull down menu.
- **Peer Verification** = **None**.
- Click **Next**.

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



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

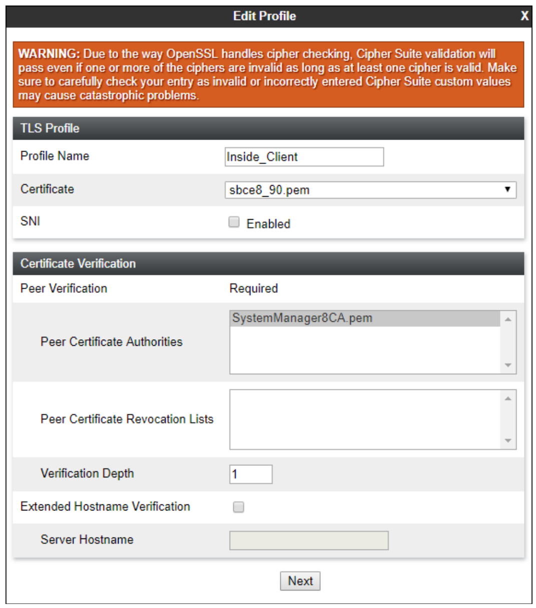


7.2.3 Client Profiles

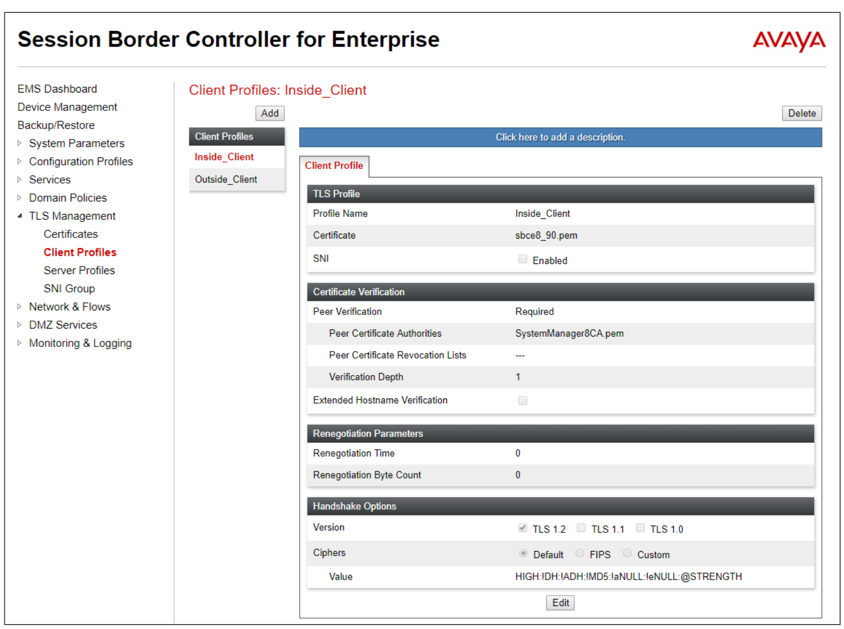
Step 1 - Select **TLS Management → Server Profiles** and click on **Add**. Enter the following:

- **Profile Name:** enter descriptive name.
- **Certificate:** select the identity certificate, e.g., **sbce8_90.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., **SystemManager8CA.pem**.
- **Verification Depth:** enter **1**. Click **Next**.

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



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

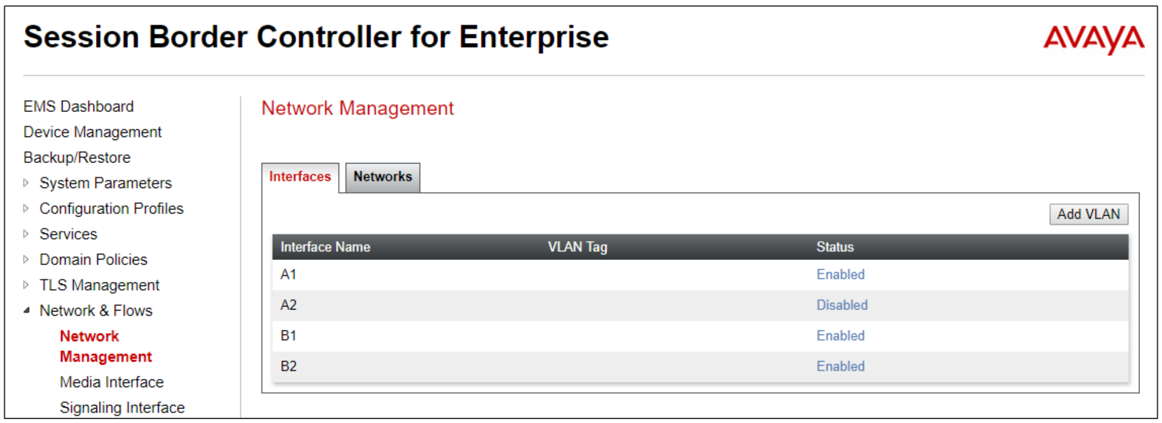


7.3. Network Management

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.

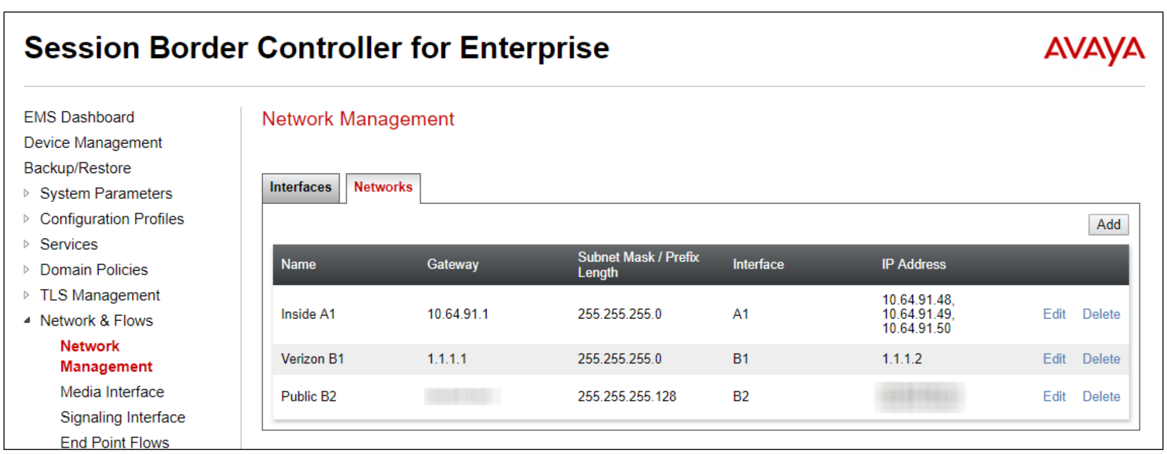
Step 1 - Select **Networks & Flows → Network Management** from the menu on the left-hand side.

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



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

- **A1**: **10.64.91.50** – "Inside" IP address, toward Session Manager.
- **B1: 1.1.1.2** – "Outside" IP address toward the Verizon SIP trunk. This address is known to Verizon.



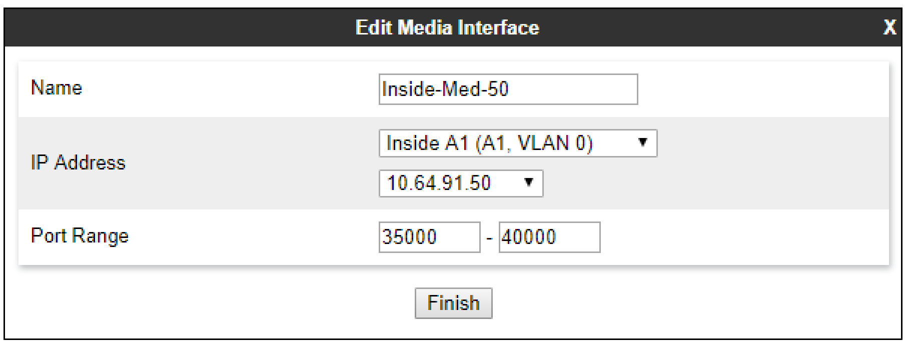
7.4. Media Interfaces

Media Interfaces are created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address, and one of the ports in this range as the listening IP address and port in which the SBCE will accept media from the connected server. Create a SIP Media Interface for both the inside and outside IP interfaces.

Step 1 - Select **Network & Flows → Media Interface** from the menu on the left-hand side.
Step 2 - Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:

- **Name**: Enter an appropriate name (e.g., **Inside-Med-50**).
- **IP Address**: Select **Inside-A1 (A1,VLAN0)** and **10.64.91.50** from the drop-down menus.
- **Port Range**: **35000 – 40000**.

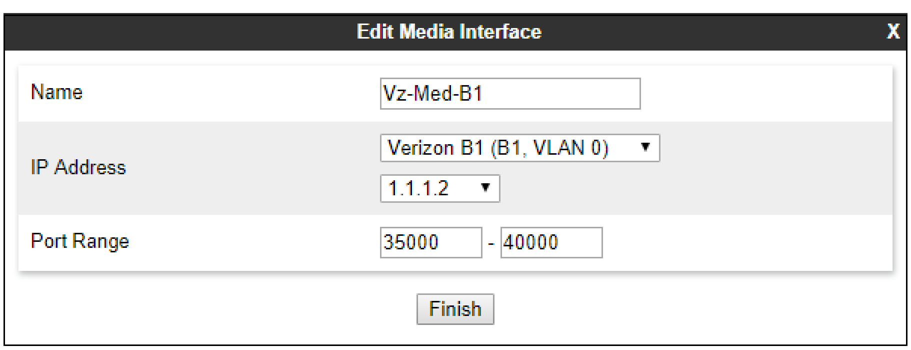
Step 3 - Click **Finish**.



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

- **Name**: Enter an appropriate name (e.g., **Vz-Med-B1**).
- **IP Address**: Select **Verizon-B1 (B1,VLAN0)** and **1.1.1.2** from the drop-down menus.
- **Port Range**: **35000 – 40000**.

Step 5 - Click **Finish**.



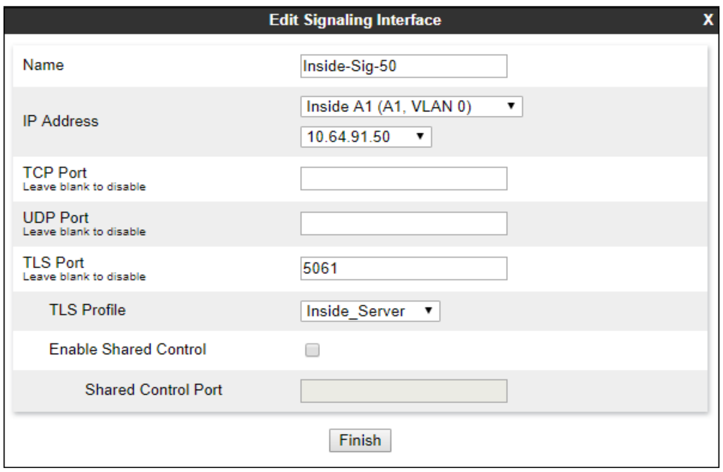
7.5. Signaling Interfaces

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 both 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**: Enter an appropriate name (e.g., **Inside-Sig-50**).
- **IP Address**: Select **Inside A1 (A1,VLAN0)** and **10.64.91.50**.
- **TLS Port**: **5061**.
- **TLS Profile**: Select the TLS server profile created in **Section** Error! Reference source not found. (e.g., **Inside_Server**)

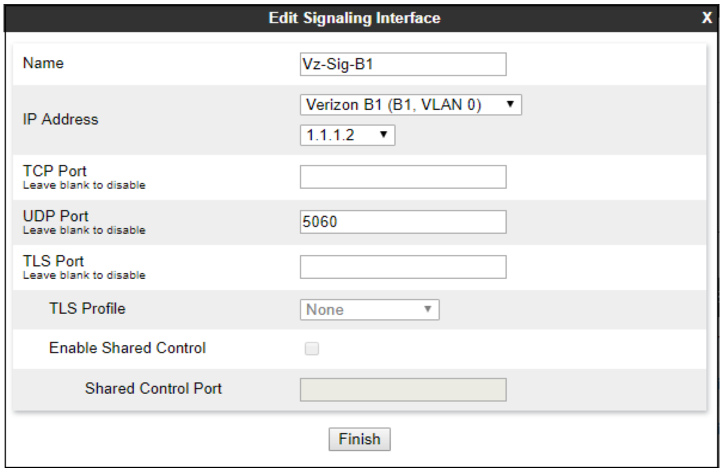
Step 3 - Click **Finish**.



Step 4 - Select **Add** (not shown), and enter the following:

- **Name**: Enter an appropriate name (e.g., **Vz-Sig-B1**).
- **IP Address**: Select **Verizon B1 (B1,VLAN0)** and **1.1.1.2**.
- **UDP Port**: **5060**.

Step 5 - Click **Finish**.



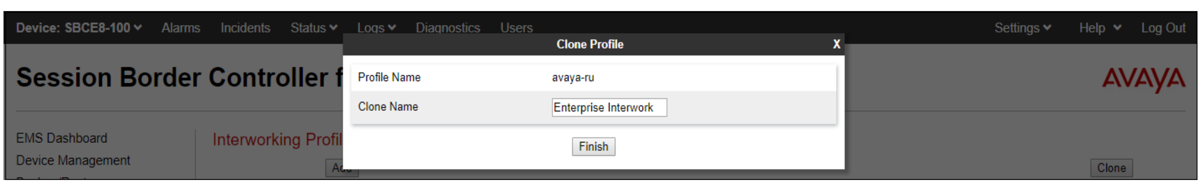
7.6. Server Interworking Profiles

The Server Interworking Profile includes 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. Create separate Server Interworking Profiles for the enterprise and the service provider.

7.6.1 Server Interworking Profile – Enterprise

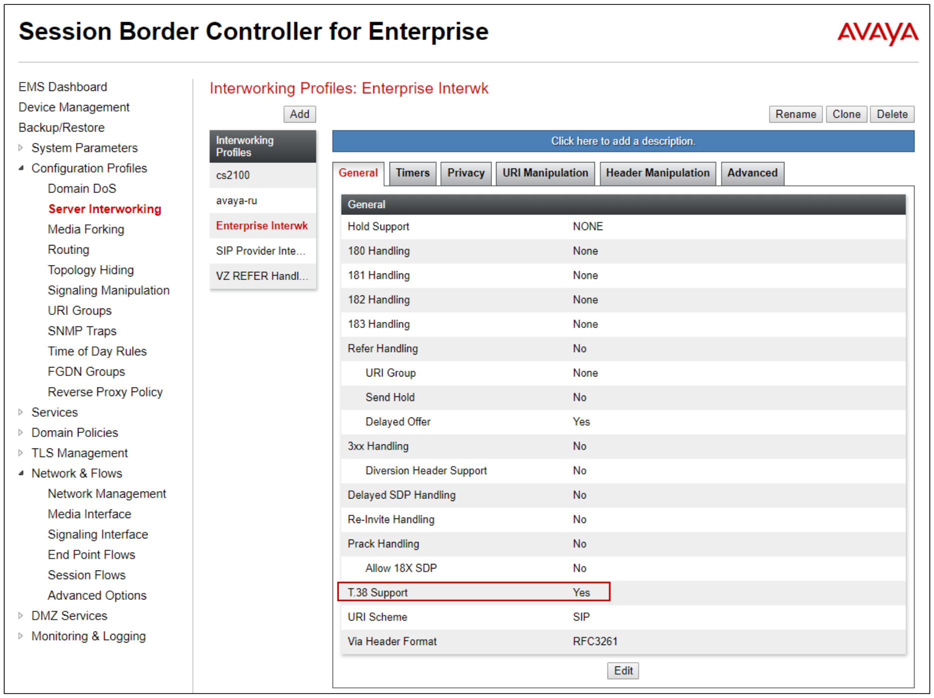
In the sample configuration, the enterprise Server Interworking profile was cloned from the default **avaya-ru** profile and then modified.

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**.



7.6.2 Server Interworking Profile – Verizon

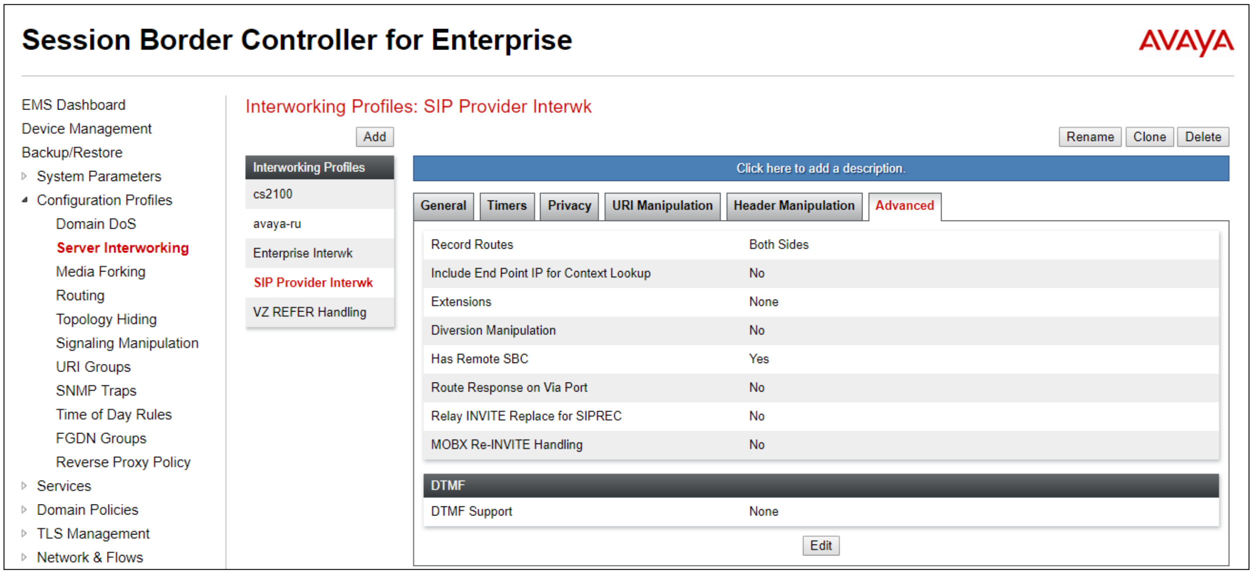
In the sample configuration, the Server Interworking profile for Verizon was created by adding a new profile.

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

Step 2 - The **General** screen will open (not shown):
- Check **T38 Support**.
- All other options can be left as default.
- Click **Next**.

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

Step 4 - The **Advanced/DTMF** screen will open:
- In the **Record Routes** field, check **Both Sides**.
- All other options can be left as default.
- Click **Finish** (not shown).


Session Border Controller for Enterprise
AVAYA
EMS Dashboard
Device Management
Backup/Restore
System Parameters
Configuration Profiles
Domain DoS
Server Interworking
Media Forking
Routing
Topology Hiding
Signaling Manipulation
URI Groups
SNMP Traps
Time of Day Rules
FGDN Groups
Reverse Proxy Policy
Services
Domain Policies
TLS Management
Network & Flows
Interworking Profiles: SIP Provider Interwk
Add
Rename
Clone
Delete
Interworking Profiles
cs2100
avaya-ru
Enterprise Interwk
SIP Provider Interwk
VZ REFER Handling
Click here to add a description.
General
Timers
Privacy
URI Manipulation
Header Manipulation
Advanced
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
DTMF
DTMF Support
None
Edit

7.7. Signaling Manipulation

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

Note – Use of the Signaling Manipulation scripts require higher processing requirements on the Avaya SBCE. Therefore, this method of header manipulation should only be used in cases where the use of Server Interworking Profiles (**Section 7.6**) or Signaling Rules (**Section 7.13**) does not meet the desired result. Refer to **[11]** in the Additional References section for information on the Avaya SBCE scripting language.

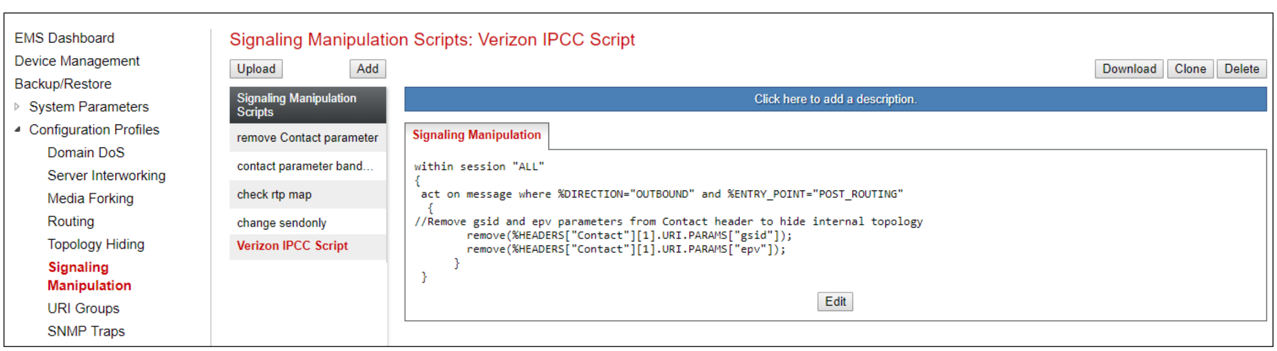
The script can be created externally as a regular text file and pasted in the Signaling Manipulation screen, or they can be written directly in the page using the embedded Sigma Editor.

A Sigma script was created during the compliance test to remove the gsid and epv parameters from the outbound Contact header. See **Section** □.

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.
Step 3 - Enter a name for the script in the **Title** box (e.g., **Verizon IPCC Script**). The following script is defined:

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

Step 4 - **Click** on **Save**. The script editor will test for any errors, and the window will close. This script will be applied to the Verizon Server Configuration later in **Section** Error! Reference source not found..



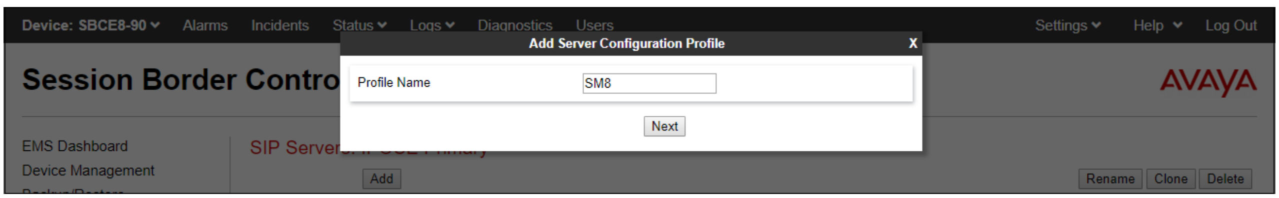
7.8. SIP Server Profiles

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.

7.8.1 SIP Server Profile – Session Manager

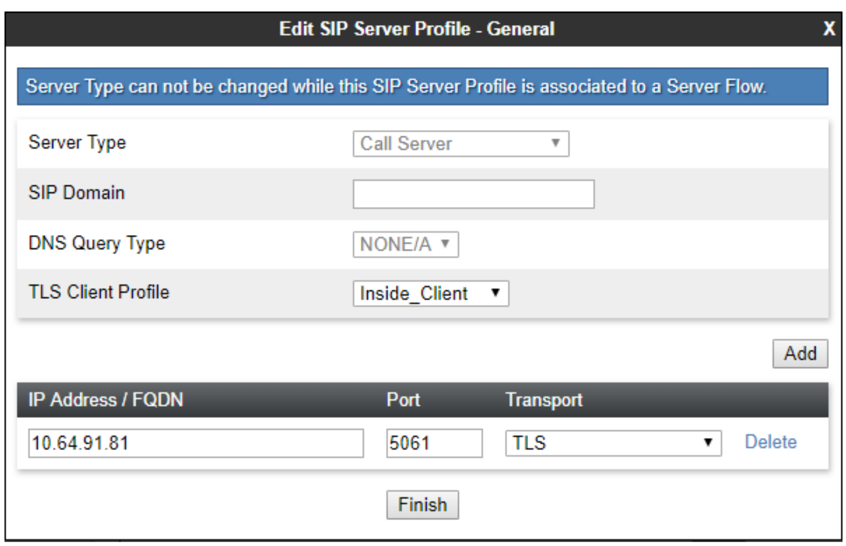
This section defines the SIP Server Profile for the Avaya SBCE connection to Session Manager.

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 **Add Server Configuration Profile** window will open.

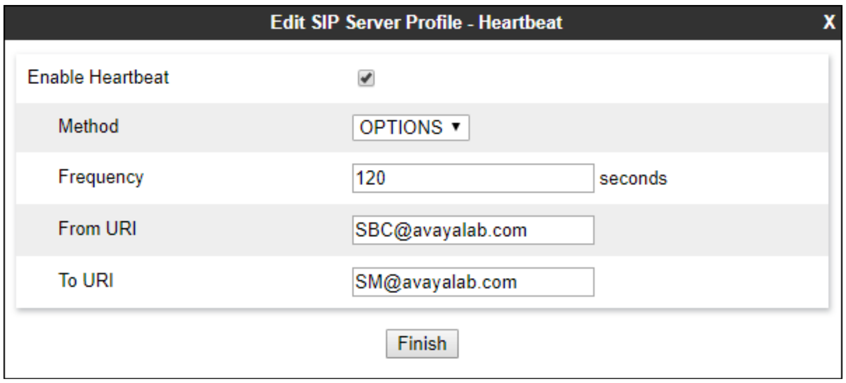
- Select **Server Type**: **Call Server**.
- **SIP Domain**: Leave blank (default).
- **DNS Query Type**: Select **NONE/A** (default).
- **TLS Client Profile**: Select the profile create in **Section** Error! Reference source not found. (e.g., **Inside-Client**).
- **IP Address**: **10.64.91.81** (Session 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.



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

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

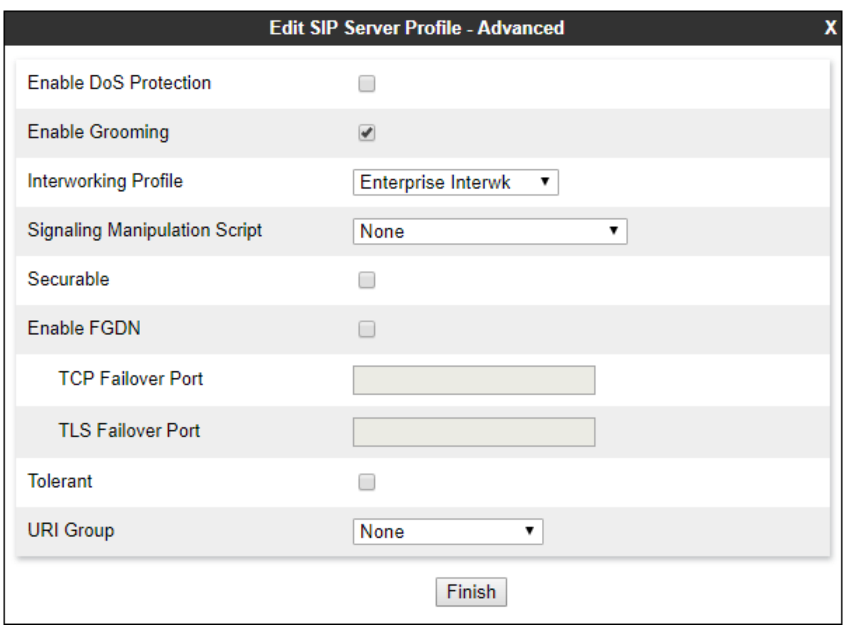
- Select **OPTIONS** from the **Method** drop-down menu.
- Select the desired frequency that the SBCE will source OPTIONS toward Session Manager.
- Make logical entries in the **From URI** and **To URI** fields that will be used in the OPTIONS headers.



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

Step 7 – On the **Advanced** tab:

- Select the **Enterprise Interwork** (created in **Section 7.6.1**), 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**.

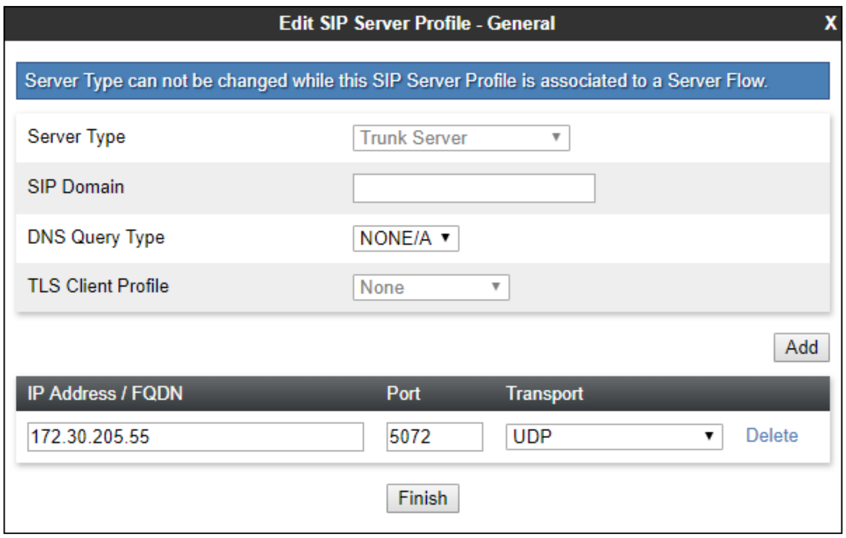


7.8.2 SIP Server Profile – Verizon

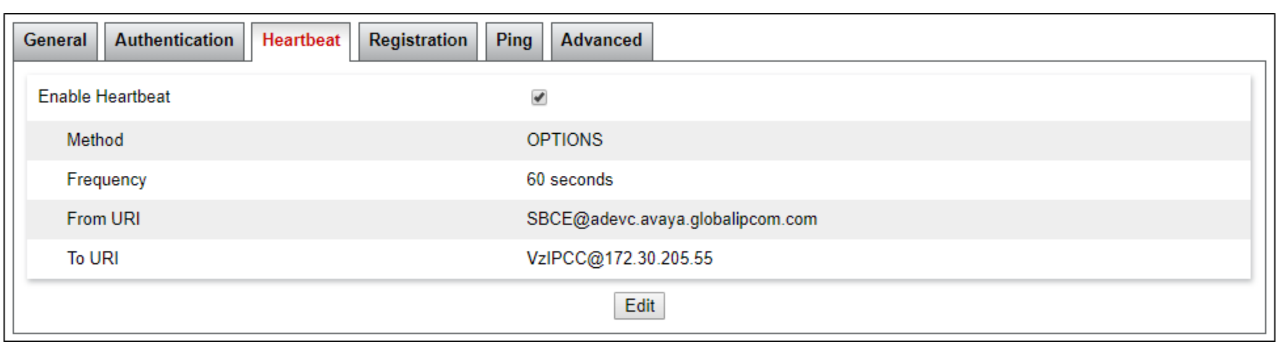
Repeat the steps in **Section 7.8.1**, with the following changes, to create a SIP Server Profile for the Avaya SBCE connection to Verizon.

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

- **Server Type:** Select **Trunk Server**.
- **IP Address: 172.30.205.55** (Verizon-provided IP address).
- Select **Port: 5072**, **Transport**: **UDP**, as specified by Verizon.
- If adding the profile, click **Next** (not shown). If editing an existing profile, click **Finish** and proceed to the next tab.

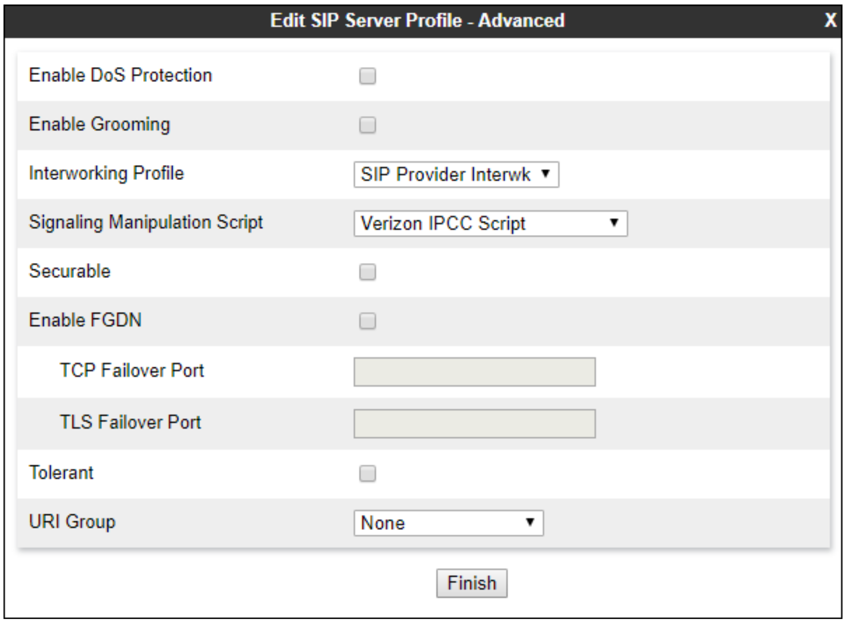


Step 4 – Default values are used on the **Authentication** tab.
Step 5 – On the **Heartbeat** tab, check the **Enable Heartbeat** box to optionally have the Avaya SBCE source "heartbeats" toward Verizon. The screen below shows the values used in the reference configuration.



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 the **SIP Provider Interwk** (created in **Section 7.6.2**), for **Interworking Profile**.
- Select the **Vz IPCC Script** (created in **Section 7.7**) for **Signaling Manipulation Script**.
- Select **Finish**.



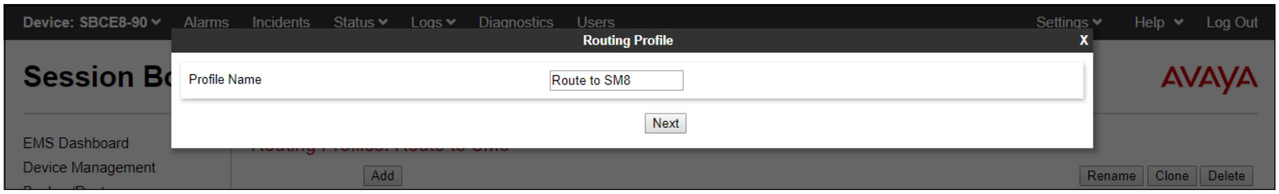
7.9. Routing Profiles

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 thereby ascertain 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 Verizon.

7.9.1 Routing Profile – Session Manager

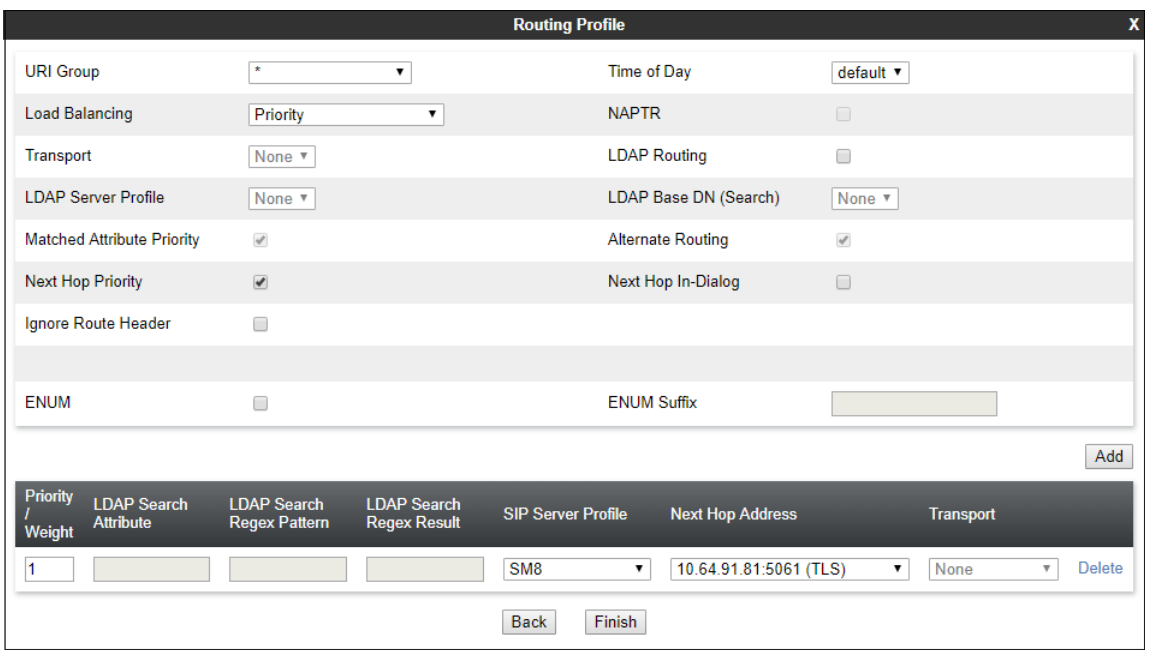
This provisioning defines the Routing Profile for the connection to Session Manager.

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



Step 3 – The Routing Profile window will open. 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** window will open. Populate the following fields:

- **Priority/Weight** = **1**
- **Server Configuration** = **SM8** (from **Section 7.8.1**).
- **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**.



7.9.2 Routing Profile – Verizon

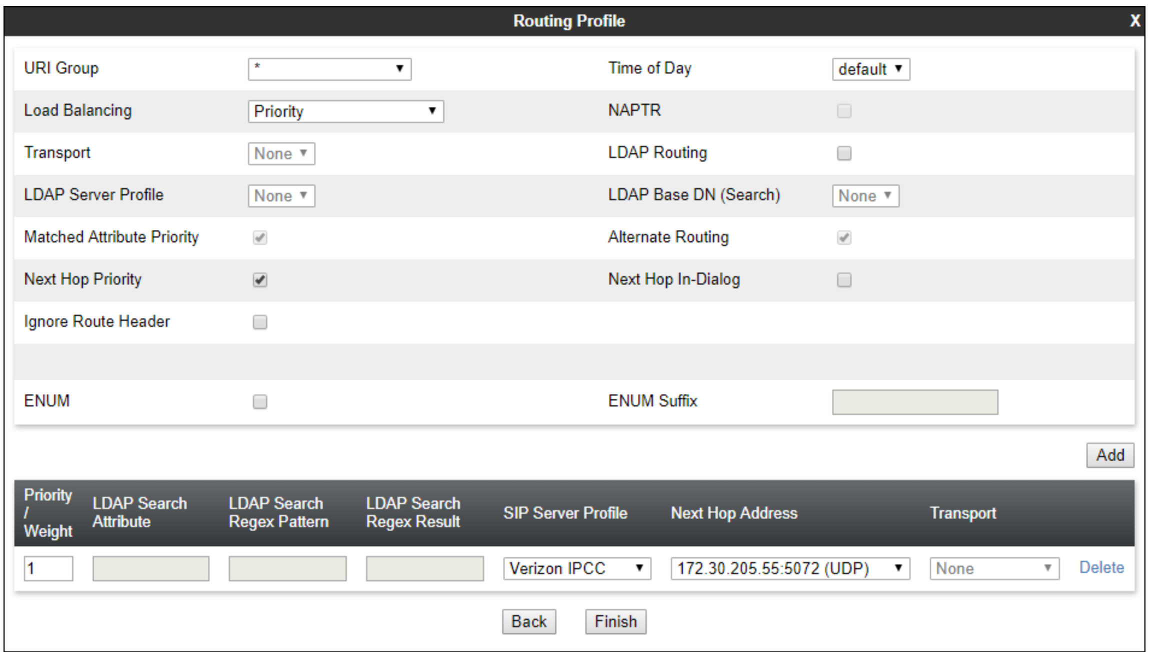
Repeat the steps in **Section 7.9.1**, with the following changes, to add a Routing Profile for the Avaya SBCE connection to Verizon.

Step 1 - On the **Global Profiles → Routing Profile** window, enter a Profile Name: (e.g., **Route to Vz IPCC**).

Step 2 - On the **Next-Hop Address** window, populate the following fields:

- **Priority/Weight** = **1**
- **Server Configuration** = **Verizon IPCC** (**from Section 7.8.2**).
- **Next Hop Address:** verify that **172.30.205.55:5072 (UDP)**.

Step 3 - Click **Finish**.



7.10. Topology Hiding Profiles

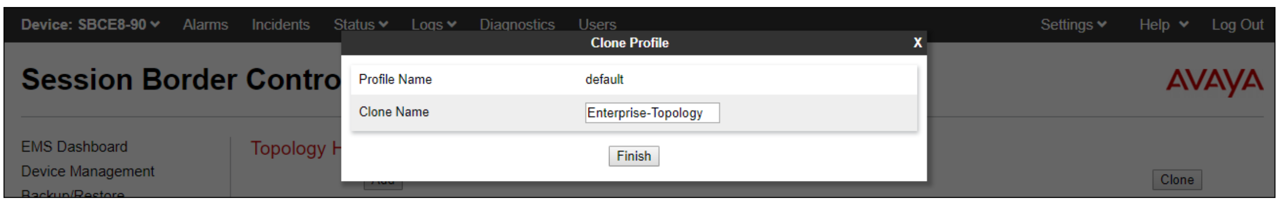
The Topology Hiding profile manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the security of the network. It hides the topology of the enterprise network from external networks.

Topology Hiding can also be used as an interoperability tool to adapt the host portion of the SIP headers, to the IP addresses or domains expected on the service provider and the enterprise networks.

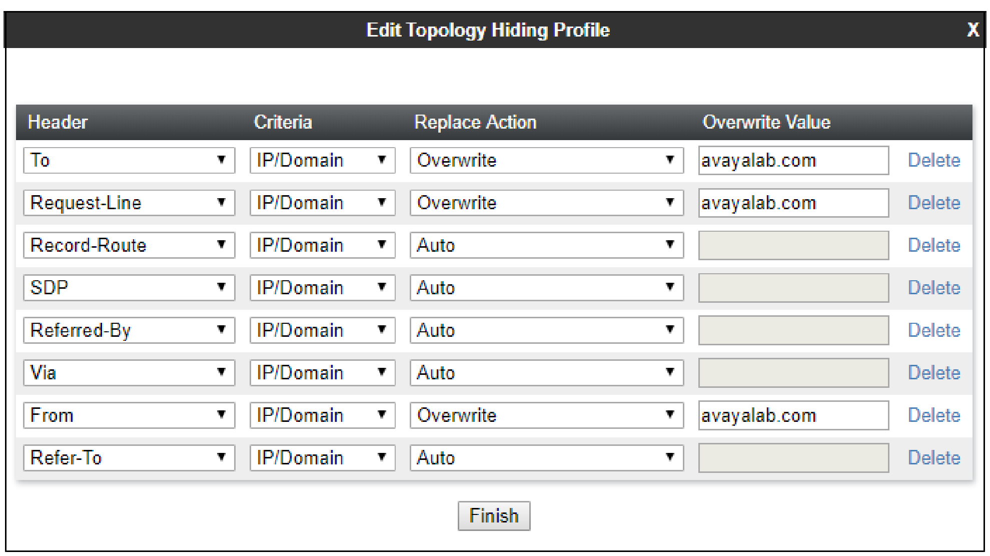
7.10.1 Topology Hiding – Enterprise

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**.



Edit Topology Hiding Profile

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

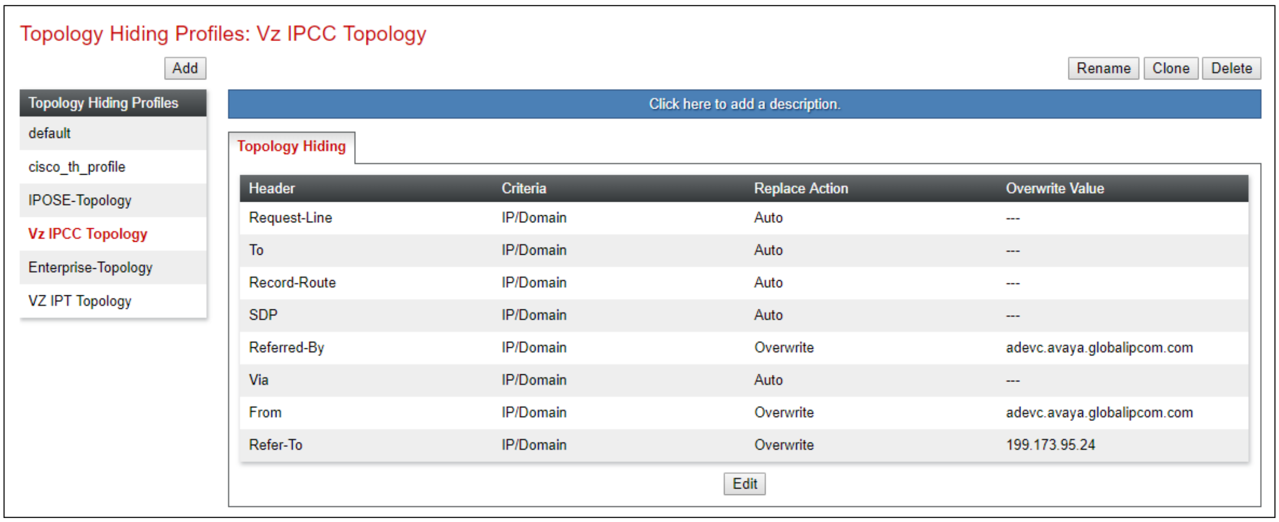
Finish

7.10.2 Topology Hiding – Verizon

Repeat the steps in **Section 7.10.1**, with the following changes, to create a Topology Hiding Profile for the Avaya SBCE connection to Verizon.

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

Note – The Refer-To header's domain is overwritten with the IP address presented in the original INVITE from Verizon's IP-IVR service. See **Section 2.2**. If the IP-IVR service is not used, the Refer-To header can retain the default **Replace Action** of "**Auto**".



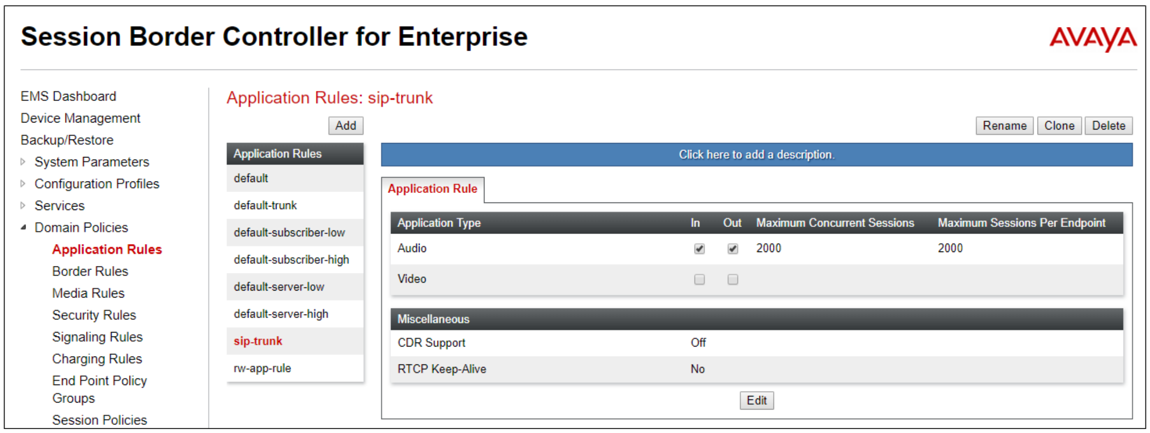
Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Auto	---
To	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Referred-By	IP/Domain	Overwrite	adevc.avaya.globalipcom.com
Via	IP/Domain	Auto	---
From	IP/Domain	Overwrite	adevc.avaya.globalipcom.com
Refer-To	IP/Domain	Overwrite	199.173.95.24

7.11. 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.



7.12. Media Rules

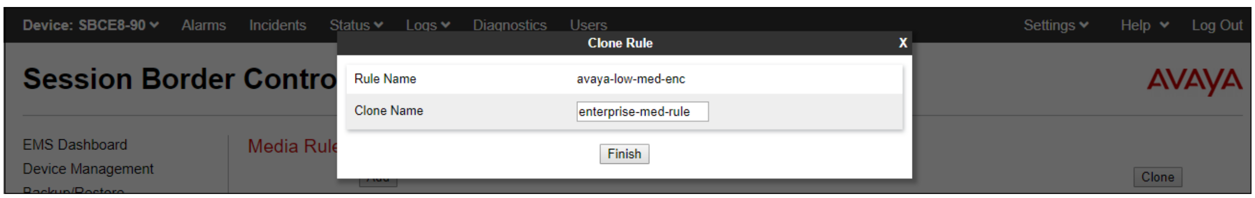
Media Rules define packet parameters for the RTP media, such as encryption techniques and QoS settings. Separate media rules are created for Verizon and Session Manager.

7.12.1 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.

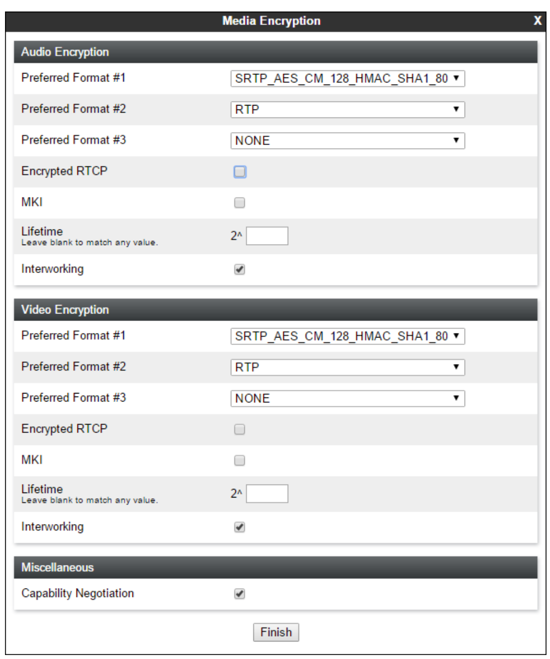
- In the **Clone Name** field enter the new Media Rule name (e.g., **enterprise-med-rule**)
- Click **Finish.** The newly created rule will be displayed.



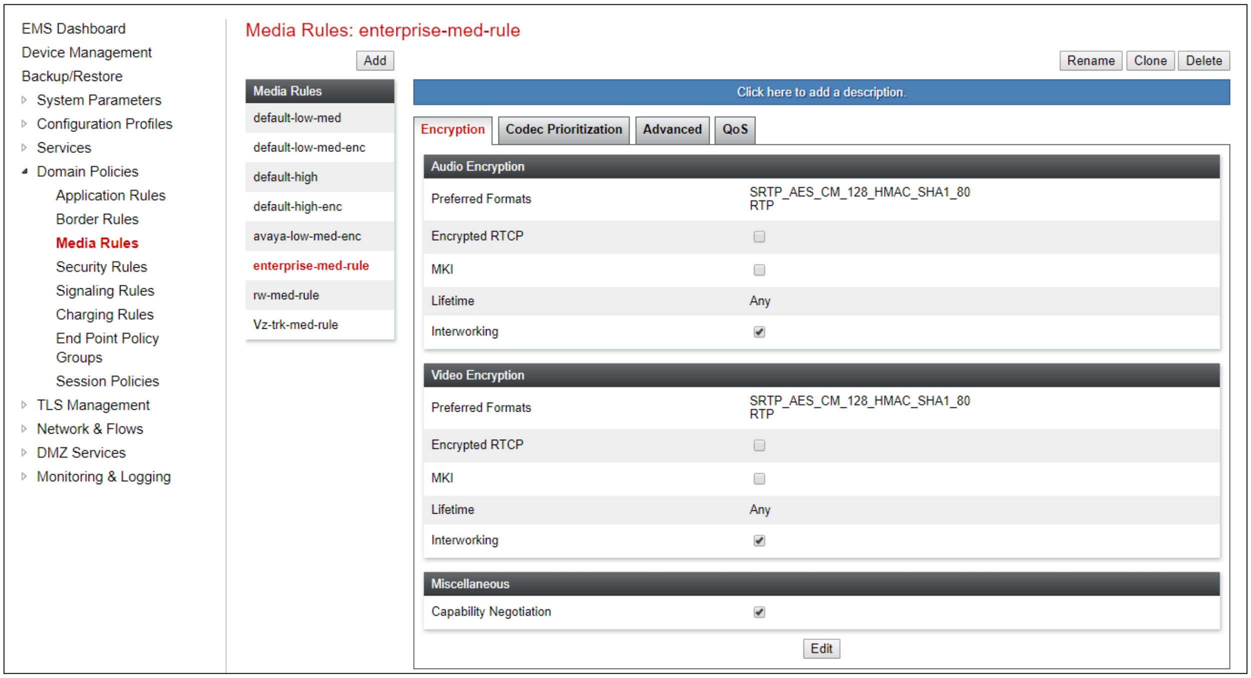
Step 4 - On the **enterprise med rule** just created, select the **Encryption** tab.

- 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.

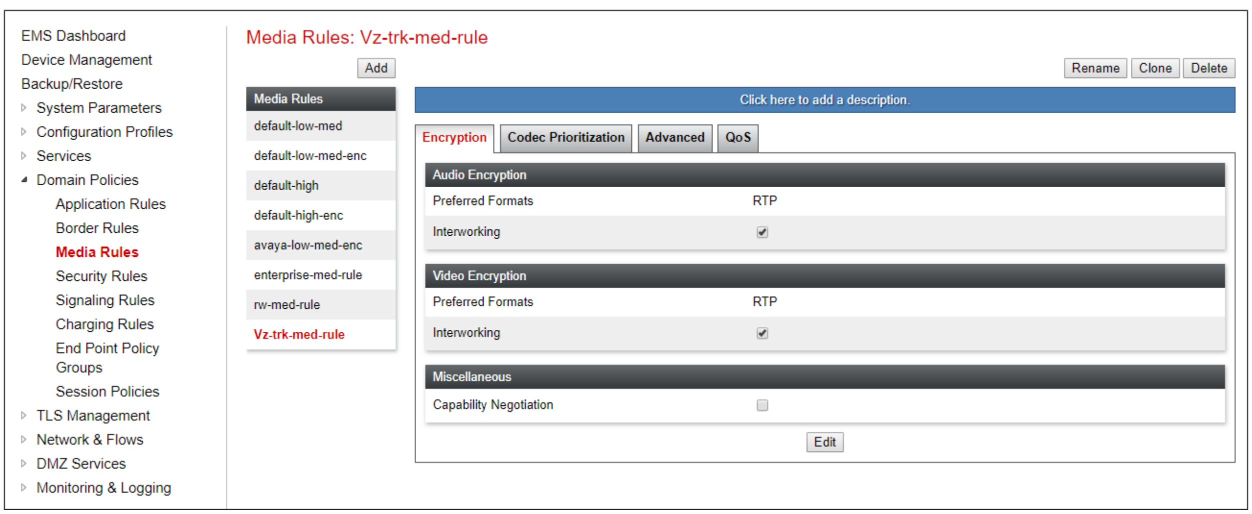


7.12.2 Verizon – Media Rule

Repeat the steps in **Section 7.12.1**, with the following changes, to create a Media Rule for Verizon.

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

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



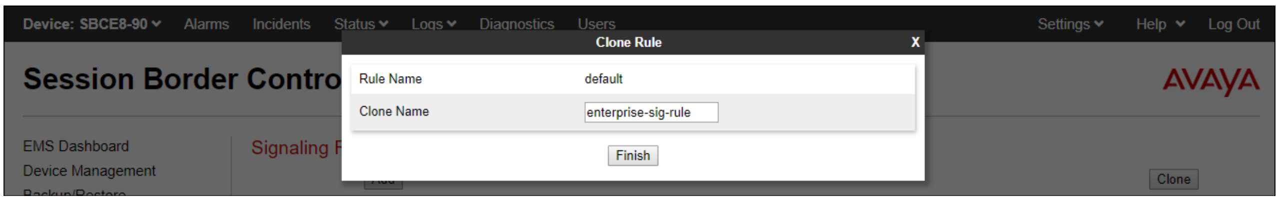
7.13. Signaling Rules

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

7.13.1 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.

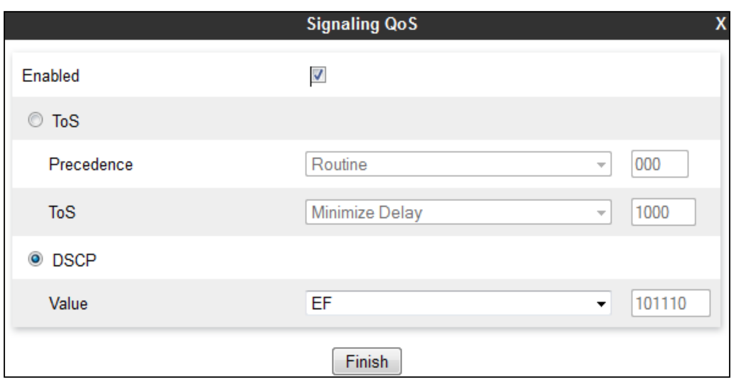
- In the **Rule Name** field enter the new Signaling Rule name (e.g., **enterprise-sig-rule)**
- Click **Finish**. The newly created rule will be displayed.



Step 4 – On the **enterprise-sig-rule** newly created, select the **Signaling QoS** tab and enter the following:

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

Step 5 - Click **Finish**.

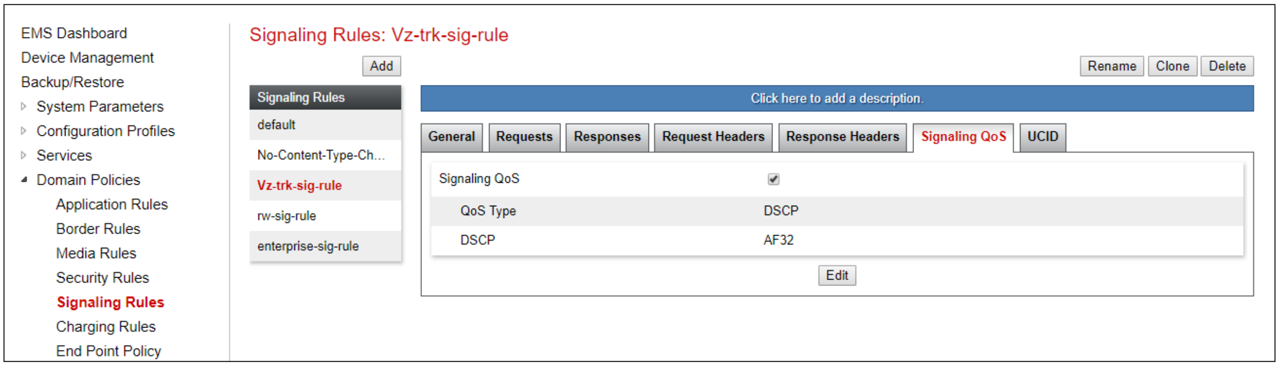


7.13.2 Signaling Rule - Verizon

Repeat the steps in **Section 7.13.1**, with the following changes, to create a Media Rule for Verizon.

- Clone the **default** rule.
- In the **Clone Name** field enter the new Media Rule name (e.g., **Vz-trk-sig-rule**).
- On the **Signaling QoS tab** select **Value** = **AF32**.

The completed **Vz-trk-sig-rule** is shown on the screen below.



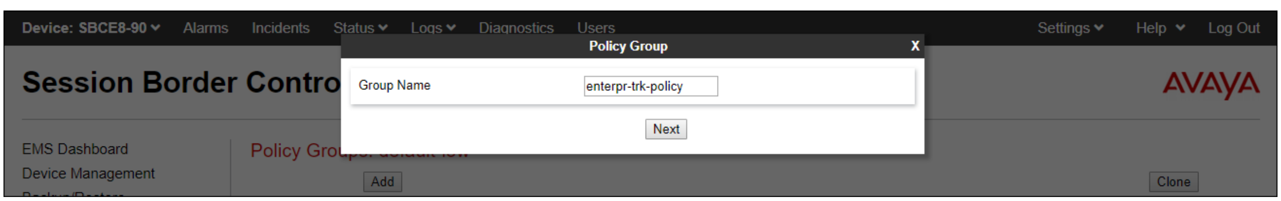
7.14. Endpoint Policy Groups

The rules created under the Domain Policy are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to a Server Flow in **Section 7.15**.

7.14.1 Endpoint Policy Group – Enterprise

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

- **Name**: **enterprise-trk-policy**.
- Click **Next**.

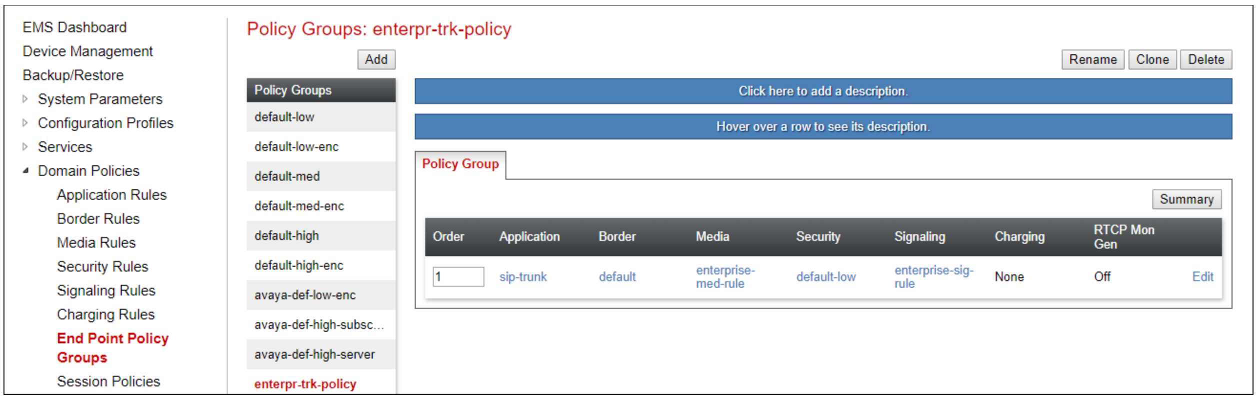


Step 3 – On the **Policy Group** window (not shown), select the following.

- **Application Rule**: **sip-trunk** (created in **Section 7.11**).
- **Border Rule**: **default**.
- **Media Rule**: **enterprise-med-rule** (created in **Section 7.12.1**).
- **Security Rule**: **default-low**.
- **Signaling Rule**: **enterprise-sig-rule** (created in **Section 7.13.1**).

Step 4 - Select **Finish**.

The completed Policy Group **enterprise-trk-policy** is shown on the screen below.

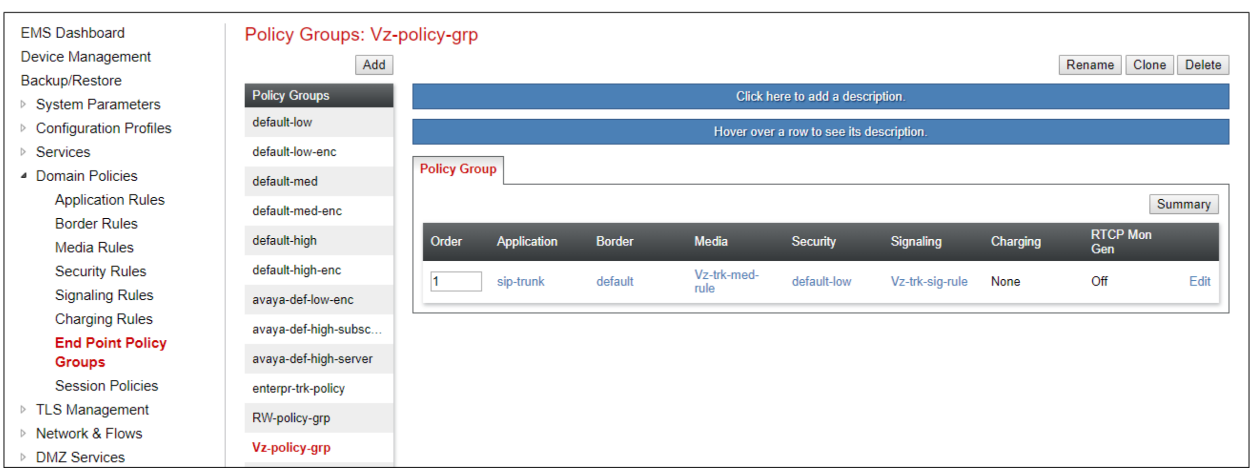


7.14.2 Endpoint Policy Groups – Verizon

Step 1 - Repeat steps **1** through **4** from **Section** Error! Reference source not found. with the following changes:

- **Group Name**: **Vz-policy-grp**.
- **Media Rule**: **Vz-trk-med-rule** (created in **Section 7.12.2**).
- **Signaling Rule**: **Vz-trk-sig-rule** (created in **Section 7.13.2**).

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



7.15. Endpoint Flows – Server Flows

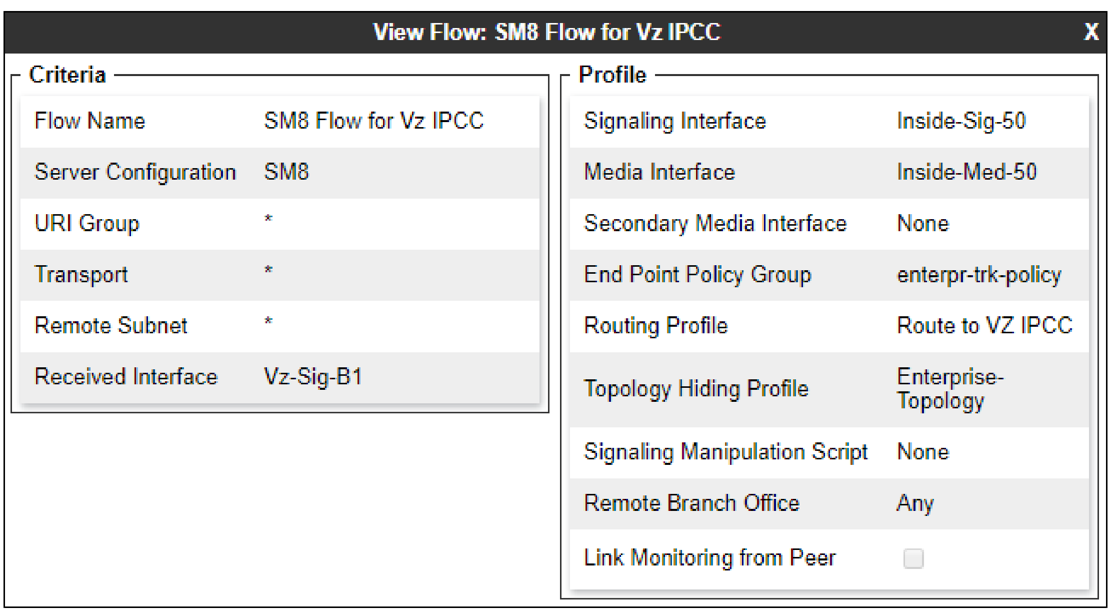
Server Flows combine the interfaces, polices, and profiles defined in the previous sections into inbound and outbound flows. When a packet is received by Avaya SBCE, the content of the packet (IP addresses, SIP URIs, etc.) is used to determine which flow it matches, so that the appropriate policies can be applied. Create separate Server Flows for the enterprise and the Verizon IP Contact Center Service.

7.15.1 Server Flow – Enterprise

Step 1 - Select **Device Specific Settings → Endpoint Flows** from the menu on the left-hand side (not shown).
Step 2 - Select the **Server Flows** tab (not shown).
Step 3 - Select **Add**, (not shown) and enter the following:

- **Flow Name**: Enter a name for the flow, e.g., **SM8 Flow for Verizon IPCC**.
- **Server Configuration**: **SM8** (**Section 7.8.1**).
- **URI Group**: *
- **Transport**: *
- **Remote Subnet**: *
- **Received Interface**: **Vz-Sig-B1** (**Section 7.5**).
- **Signaling Interface**: **Inside-Sig-50** (**Section 7.5**).
- **Media Interface**: **Inside-Med-50** (**Section 7.4**).
- **End Point Policy Group**: **enterprise-trk-policy** (**Section 7.14.1**).
- **Routing Profile**: **Route to Vz IPCC** (**Section 7.9.2**).
- **Topology Hiding Profile**: **Enterprise-Topology** (**Section 7.10.1**).
- Let other fields at their default values.

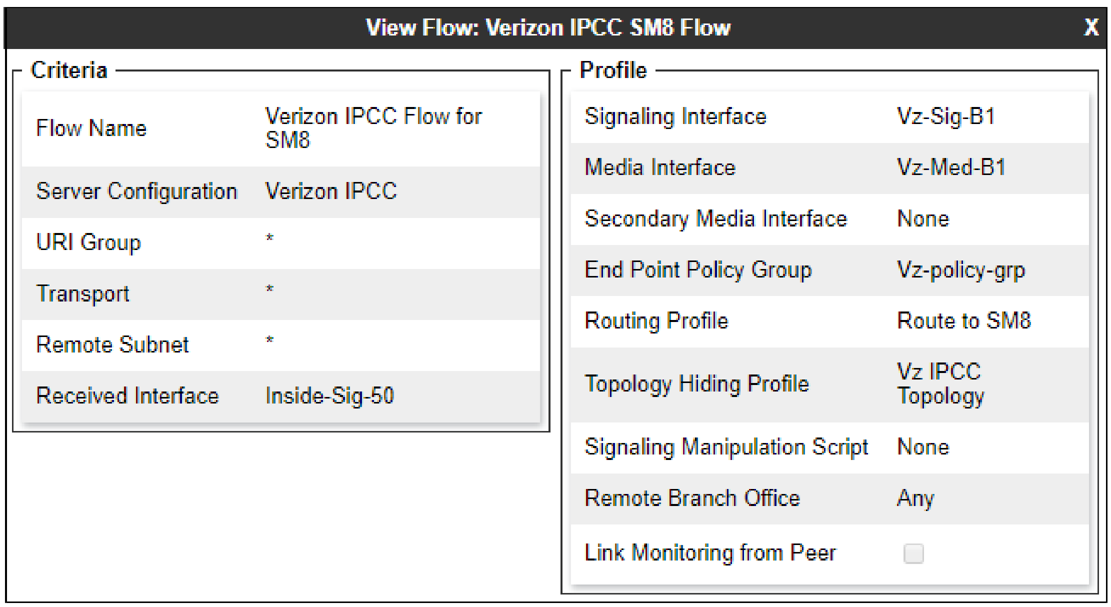
Step 4 - Click **Finish** (not shown).



7.15.2 Server Flow – Verizon

Step 1 - Repeat steps **1** through **4** from **Section 7.15.1**, with the following changes:

- **Flow Name**: **Verizon IPCC Flow for SM8**.
- **Server Configuration**: **Verizon IPCC** (**Section 7.8.2**).
- **URI Group**: *****
- **Transport**: *****
- **Remote Subnet**: *****
- **Received Interface**: **Inside-Sig-50** (**Section 7.5**).
- **Signaling Interface**: **Vz-Sig-B1 (Section 7.5**).
- **Media Interface**: **Vz-Med-B1** (**Section 7.4**).
- **End Point Policy Group**: **Vz-policy-grp** (**Section 7.14.2**).
- **Routing Profile**: **Route to SM8** (**Section 7.9.1**).
- **Topology Hiding Profile**: **Vz IPCC Topology** (**Section 7.10.2**).



View Flow: Verizon IPCC SM8 Flow

Criteria

Flow Name	Verizon IPCC Flow for SM8
Server Configuration	Verizon IPCC
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Inside-Sig-50

Profile

Signaling Interface	Vz-Sig-B1
Media Interface	Vz-Med-B1
Secondary Media Interface	None
End Point Policy Group	Vz-policy-grp
Routing Profile	Route to SM8
Topology Hiding Profile	Vz IPCC Topology
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	

8. Verizon Business IPCC Services Suite Configuration

Information regarding Verizon Business IPCC Services suite offer can be found at https://enterprise.verizon.com/products/customer-experience-services/transport-and-intelligent-routing/ip-contact-center/ or by contacting a Verizon Business sales representative.

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

8.1. Service Access Information

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

CPE (Avaya)	Verizon Network
adevc.avaya.globalipcom.com ***UDP port 5060***	***172.30.205.55*** ***UDP Port 5072***

Toll Free Numbers
866-850-2380
866-851-0107
866-851-2649
866-852-3221
866-850-6850

9. Verification Steps

This section provides example verifications of the Avaya configuration with Verizon Business IP Contact Center service.

9.1. Avaya Aura® Communication Manager Verifications

This section illustrates verifications from Communication Manager.

9.1.1 Example Incoming Call from PSTN via Verizon IPCC to Agent

The following edited Communication Manager **list trace tac** trace output shows a call incoming on trunk group 2. The PSTN telephone dialed is 866-850-2380. Session Manager mapped the number received from Verizon to the extension of Communication Manager VDN 10004. This VDN is associated with call vector 4 (see **Section 5.9.4**). Extension 50231 is an IP Telephone with IP address 10.5.5.211, in Region 1. This telephone is logged in to ACD hunt group 1, using agent-loginID 20001. Initially, the Media Server (10.64.91.86) is used. The annotations in the edited trace highlight key behaviors.

```
list trace tac *02                                               Page   1

                                LIST TRACE
time            data
-----Incoming call arrives to Communication Manager----------------------
13:20:47 TRACE STARTED 05/08/2020 CM Release String cold-01.0.890.0-26095
13:20:52 SIP<INVITE sips:10004@avayalab.com SIP/2.0
13:20:52     Call-ID: 0a14e6698fb19f2e9886542d5595e95a
13:20:52     active trunk-group 2 member 1    cid 0x10bd
----Vector step plays ringback. 183 with SDP is sent. Media Server at
10.64.91.86 on media path --------------------------------------------
13:20:52   23  1 vdn e10004 bsr appl   0 strategy 1st-found override n
13:20:52   23  1 AVDN: 10004 AVRD:
13:20:52   23  1 # Wait hearing ringback...
13:20:52   23  2 wait 2 secs hearing ringback
13:20:52 SIP>SIP/2.0 183 Session Progress
13:20:52     Call-ID: 0a14e6698fb19f2e9886542d5595e95a
13:20:52     dial 10004
13:20:52     ring vector 4      cid 0x10bd
13:20:52     G729 ss:off ps:20
             rgn:2 [10.64.91.50]:35174
             rgn:1 [10.64.91.86]:6008
---Call is routed to ACD hunt group 1. Agent 20001, logged in at extension
50231 answers the call, Communication Manager sends 200 OK--------------
13:20:54   23  8      queuing to skill   1 pri m
13:20:54   23  8 Local Agent Preference=n
13:20:54   23  8 Agent Login ID: 20001  Logged in at station: 50231
13:20:54 SIP>SIP/2.0 183 Session Progress
13:20:54     Call-ID: 0a14e6698fb19f2e9886542d5595e95a
13:20:54   23  8 LEAVING VECTOR PROCESSING cid 4285
13:20:54     G72264K ss:off ps:20
             rgn:1 [10.5.5.211]:21950
             rgn:1 [10.64.91.86]:6010
13:21:03 SIP>SIP/2.0 200 OK
13:21:03     Call-ID: 0a14e6698fb19f2e9886542d5595e95a
13:21:03     active station      50231 cid 0x10bd
13:21:03     Connected party  uses public-unknown-numbering
13:21:03 SIP<ACK sips:+18668502380@10.64.91.75:5071;transport=tls;as
13:21:03 SIP<m=1 SIP/2.0

<continued on next page>
```

Once the call is answered, the final RTP media path is “ip-direct” from the IP Telephone (10.5.5.211) to the “inside” of the Avaya SBCE (10.64.91.50) in Region 2. The Media Server is no longer involved in the media path.

```
list trace tac *02                                                    Page   2

                                LIST TRACE
time            data
---Communication Manager sends re-INVITE for direct IP-IP media (shuffling)--
13:21:03 SIP>INVITE sips:+17863310799@10.64.91.50:5061;transport=tls
13:21:03 SIP>;gsid=084cfd60-9161-11ea-9ee8-000c294b7db6;sipappsessio
13:21:03 SIP>nid=app-1r0bbcj1tpwed;wlssfcid=sip-qtvg1taqog67;asm=1 S
13:21:03 SIP>IP/2.0
13:21:03     Call-ID: 0a14e6698fb19f2e9886542d5595e95a
13:21:03 SIP<SIP/2.0 100 Trying
13:21:03     Call-ID: 0a14e6698fb19f2e9886542d5595e95a
----Communication Manager receives 200 OK with SDP to the re-INVITE---------
13:21:03 SIP<SIP/2.0 200 OK
13:21:03     Call-ID: 0a14e6698fb19f2e9886542d5595e95a
----Communication Manager sends ACK with SDP------------------------------
13:21:03 SIP>ACK sips:+17863310799@10.64.91.50:5061;transport=tls;gs
13:21:03 SIP>id=084cfd60-9161-11ea-9ee8-000c294b7db6;sipappsessionid
13:21:03 SIP>=app-1r0bbcj1tpwed;wlssfcid=sip-qtvg1taqog67;asm=1 SIP/
13:21:03 SIP>2.0
13:21:03     Call-ID: 0a14e6698fb19f2e9886542d5595e95a
---Final media path is IP-direct, from telephone (10.5.5.211) to the
SBCE A1 interface (10.64.91.50)------------------------------------------
13:21:03     G729A ss:off ps:20
             rgn:2 [10.64.91.50]:35174
             rgn:1 [10.5.5.211]:21950
13:21:03     G729 ss:off ps:20
             rgn:1 [10.5.5.211]:21950
             rgn:2 [10.64.91.50]:35174
---Extension hangs up, Communication Manager sends BYE------------------
13:24:01 SIP>BYE sips:+17863310799@10.64.91.50:5061;transport=tls;gs
13:24:01 SIP>id=084cfd60-9161-11ea-9ee8-000c294b7db6;sipappsessionid
13:24:01 SIP>=app-1r0bbcj1tpwed;wlssfcid=sip-qtvg1taqog67;asm=1 SIP/
13:24:01 SIP>2.0
13:24:01     Call-ID: 0a14e6698fb19f2e9886542d5595e95a
13:24:01     idle station      50231 cid 0x10bd
```

The following screen shows **Page 2** of the output of the **status trunk 2/*x*** command (where *x* is the trunk group member active on the call, **1 i**n the example) pertaining to this same call. Note the signaling using port 5071 between Communication Manager and Session Manager. Note the media is "**ip-direct**" from the IP Telephone (10.5.5.211) to the inside IP address of Avaya SBCE (10.64.91.50) using codec G.729.

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

Near-end Signaling Loc: PROCR
  Signaling   IP Address                              Port
   Near-end:  10.64.91.75                           : 5071
    Far-end:  10.64.91.81                           : 5071
 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                            : 21950
    Far-end:  10.64.91.50                           : 35174

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:               Video Far-end Codec:
```

The following screen shows **Page 3** of the output of the **status trunk** command pertaining to the same call. Note that G.729 codec is used.

```
status trunk 2/1                                              Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T000011
T000011:TX:10.64.91.50:35174/g729/20ms/1-srtp-aescm128-hmac80
S000598:RX:10.5.5.211:21950/g729a/20ms/1-srtp-aescm128-hmac80
```

9.1.2 Example Incoming Call Referred via Call Vector to PSTN Destination

The following edited and annotated Communication Manager **list trace tac** trace output shows a call incoming on trunk group 2. The PSTN telephone dialed was 866-852-3221. Session Manager mapped the number received from Verizon to the VDN 10001. This VDN is associated with call vector 2 (see **Section 5.9.42**). The vector answers the call, plays an announcement to the caller, and then uses a "route-to" step to cause a REFER message to be sent with a Refer-To header containing the number configured in the vector. The annotations in the edited trace highlight key behaviors. At the conclusion, the PSTN caller that dialed the Verizon toll-free number is connected to the Referred-to PSTN destination, and no trunks (i.e., from trunk 2 handling the call) are in use.

```
list trace tac *02                                                   Page   1
                                LIST TRACE
time            data
--------Incoming call arrives to Communication Manager-----------------
14:33:52 TRACE STARTED 05/08/2020 CM Release String cold-01.0.890.0-26095
14:34:05 SIP<INVITE sips:10001@avayalab.com SIP/2.0
14:34:05     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
14:34:05     active trunk-group 2 member 1    cid 0x10c1
14:34:05    0  0 ENTERING TRACE cid 4289
14:34:05    2  1 vdn e10001 bsr appl   0 strategy 1st-found override n
14:34:05    2  1 AVDN: 10001 AVRD:
14:34:05    2  1 wait 2 secs hearing ringback
--------Vector step plays ringback. 183 with SDP is sent-----------------
14:34:05 SIP>SIP/2.0 183 Session Progress
14:34:05     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
14:34:05     dial 10001
14:34:05     ring vector 2      cid 0x10c1
14:34:05     G729 ss:off ps:20
             rgn:2 [10.64.91.50]:35180
             rgn:1 [10.64.91.86]:6020
14:34:07    2  2 # Play announcement to caller i...
14:34:07    2  3 announcement 11006
14:34:07 SIP>SIP/2.0 183 Session Progress
14:34:07     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
------Vector plays announcement to caller. 200 OK is sent-------------------
14:34:07    2  3      announcement: board M2:00 ann ext: 11006
14:34:07 SIP>SIP/2.0 200 OK
14:34:07     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
10:07:13 SIP>SIP/2.0 200 OK
10:07:13     Call-ID: 5b02842140d48d4d34f85e2137fda505
10:07:13     active announcement      11006 cid 0x386
10:07:13     hear audio-group 1 board M1 ext 11006 cid 0x386
10:07:13     Connected party  uses public-unknown-numbering
10:07:13 SIP<ACK sips:+18668523221@10.64.91.75:5071;transport=tls;as
10:07:13 SIP<m=1 SIP/2.0
14:34:08     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
14:34:09     idle announcement       cid 0x10c1

<continued on next page>
```

```
list trace tac *02                                                  Page   2
                                LIST TRACE
time            data
----------Communication Manager sends REFER--------------------------------
14:34:09    2  4 # Refer the call to PSTN Destin...
14:34:09    2  5 route-to number ~r+17863310799 cov y if unconditionally
14:34:09 SIP>REFER sips:+19543613200@10.64.91.50:5061;transport=tls;
14:34:09 SIP>gsid=42e4f1d0-916b-11ea-9ee8-000c294b7db6;sipappsession
14:34:09 SIP>id=app-14orue9j6ugad;wlssfcid=sip-1fv45odoj81wu;asm=1 S
14:34:09 SIP>IP/2.0
14:34:09     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
------Communication Manager receives 202 Accepted sent by Verizon IPCC-----
14:34:09 SIP<SIP/2.0 202 Accepted
14:34:09     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
------Verizon IPCC sends NOTIFY with sipfrag 100 Trying--------------
14:34:09 SIP<NOTIFY sips:+18668523221@10.64.91.75:5071;transport=tls
14:34:09 SIP<;asm=1 SIP/2.0
14:34:09     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
14:34:09 SIP>SIP/2.0 200 OK
14:34:09     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
-------Verizon IPCC sends NOTIFY with sipfrag 200 OK-------------
14:34:18 SIP<NOTIFY sips:+18668523221@10.64.91.75:5071;transport=tls
14:34:18 SIP<;asm=1 SIP/2.0
14:34:18     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
14:34:18 SIP>SIP/2.0 200 OK
14:34:18     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
14:34:18    2  5 LEAVING VECTOR PROCESSING cid 4289
------Communication Manager sends a BYE-----------------------------
14:34:18 SIP>BYE sips:+19543613200@10.64.91.50:5061;transport=tls;gs
14:34:18 SIP>id=42e4f1d0-916b-11ea-9ee8-000c294b7db6;sipappsessionid
14:34:18 SIP>=app-14orue9j6ugad;wlssfcid=sip-1fv45odoj81wu;asm=1 SIP
14:34:18 SIP>/2.0
14:34:18     Call-ID: 5c796537daa80d295e55ec7d2c68bc13
14:34:18     idle trunk-group 2 member 1    cid 0x10c1
----Trunk is now idle. Caller and Refer-To target are now connected by Verizon--
```

When the initial call arrived from Verizon, it used trunk member 1 in trunk group 2. After the successful transfer with REFER back to Verizon, trunk member 1 is now idle.

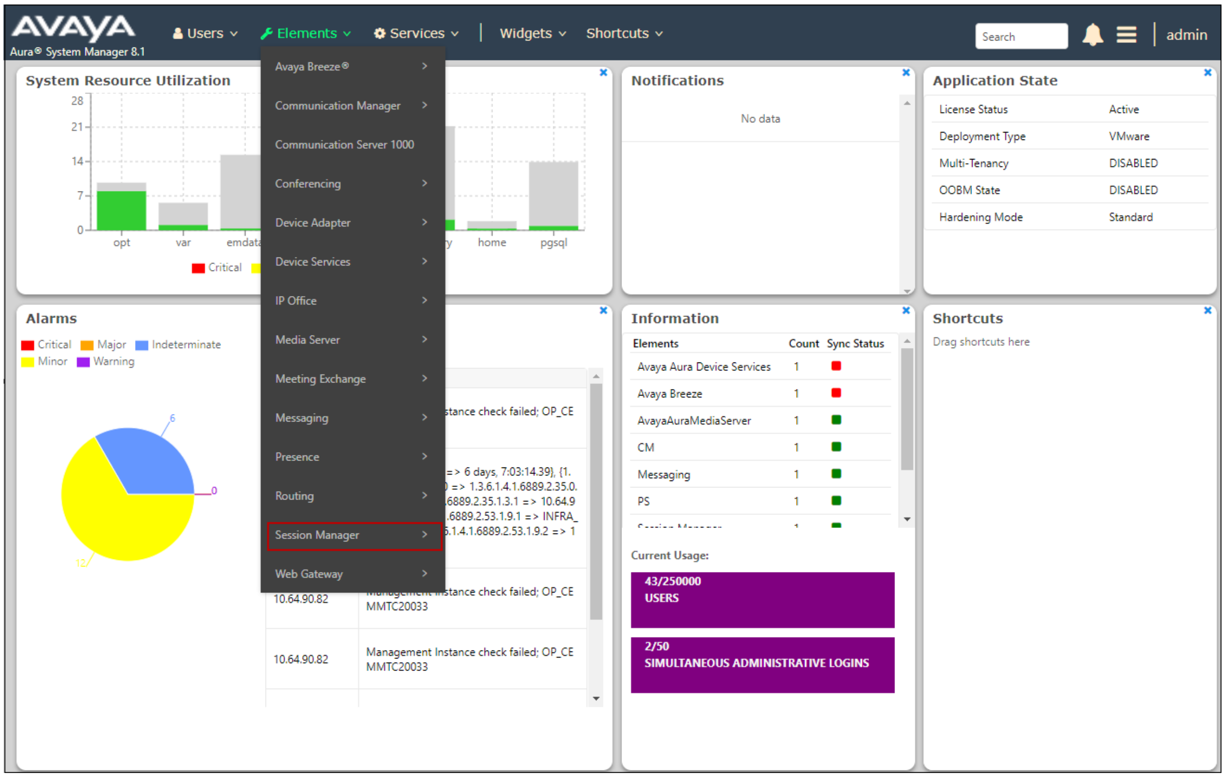
```
status trunk 2
                             TRUNK GROUP STATUS
Member   Port     Service State      Mtce Connected Ports
                                     Busy

0002/001 T00011   in-service/idle    no
0002/002 T00012   in-service/idle    no
0002/003 T00013   in-service/idle    no
0002/004 T00014   in-service/idle    no
0002/005 T00015   in-service/idle    no
0002/006 T00016   in-service/idle    no
0002/007 T00017   in-service/idle    no
0002/008 T00018   in-service/idle    no
0002/009 T00019   in-service/idle    no
0002/010 T00020   in-service/idle    no
```

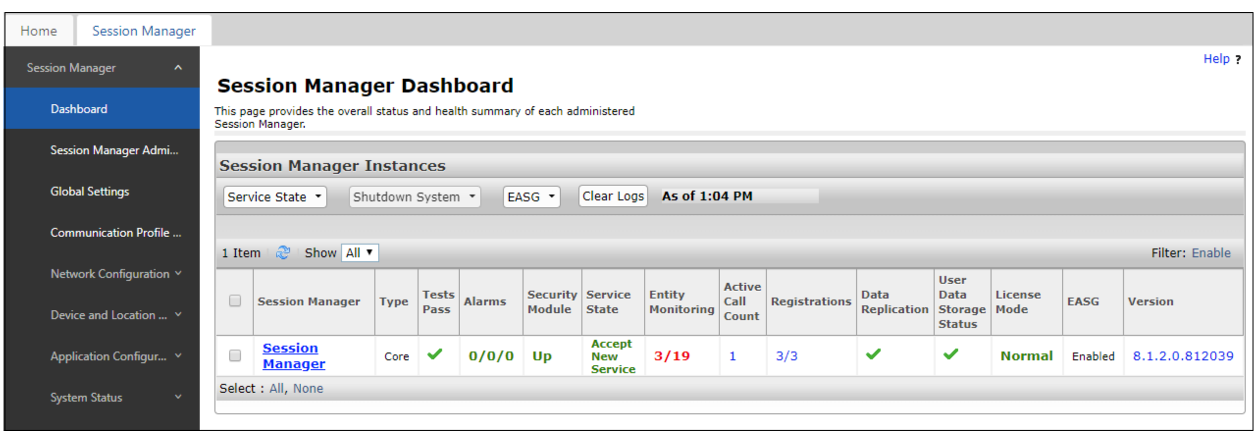
9.2. Avaya Aura® Session Manager Verification

The Session Manager configuration may be verified via System Manager.

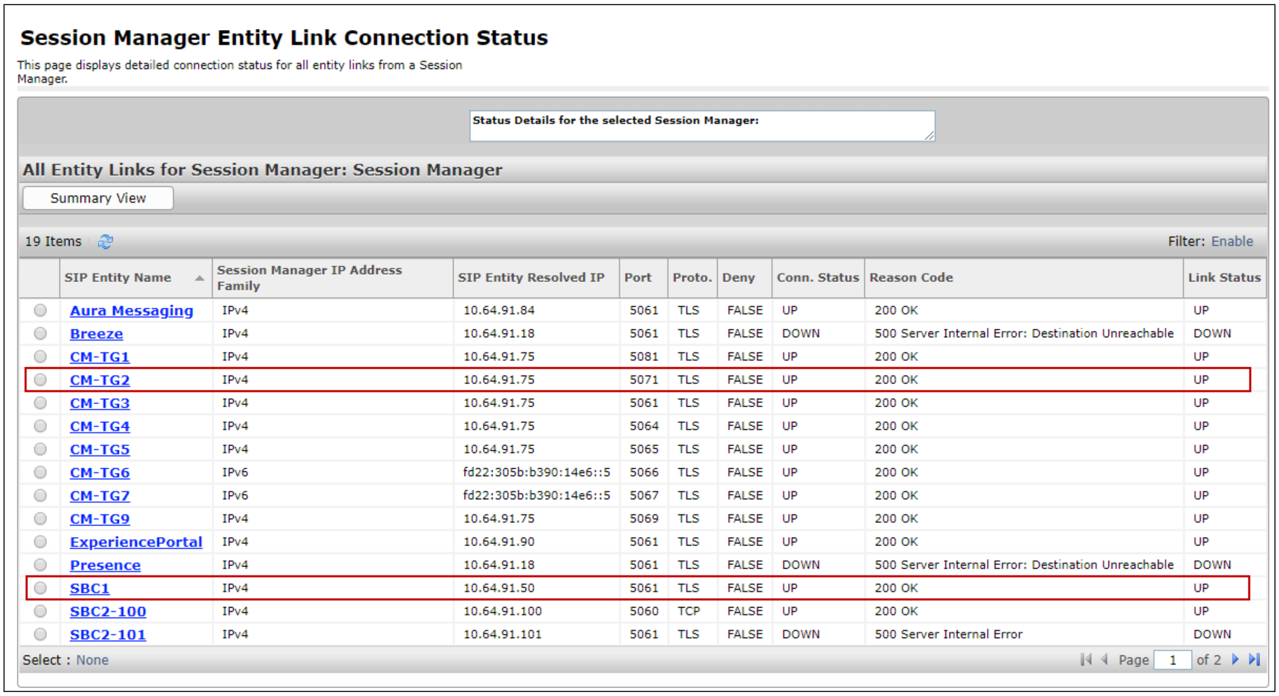
Using the procedures described in **Section 6.1**, access the System Manager GUI. From the **Home** screen, under the **Elements** heading, select **Session Manager**.



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



In the example, the entry **3/19** under the **Entity Monitoring** column shows that there are alarms on 3 out of the 19 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

19 Items | Filter: Enable

SIP Entity Name	Session Manager 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-TG6	IPv6	fd22:305b:b390:14e6::5	5066	TLS	FALSE	UP	200 OK	UP
CM-TG7	IPv6	fd22:305b:b390:14e6::5	5067	TLS	FALSE	UP	200 OK	UP
CM-TG9	IPv4	10.64.91.75	5069	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
SBC2-100	IPv4	10.64.91.100	5060	TCP	FALSE	UP	200 OK	UP
SBC2-101	IPv4	10.64.91.101	5061	TLS	FALSE	DOWN	500 Server Internal Error	DOWN

Select : None | Page 1 of 2

Other Session Manager useful verification and troubleshooting tools include:

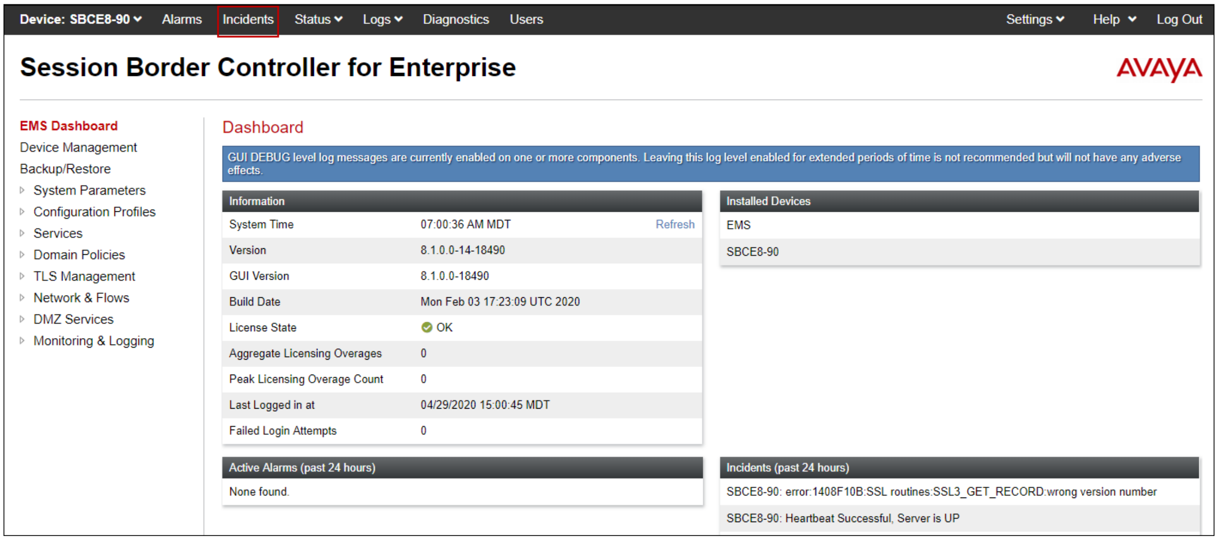
- **traceSM** – Session Manager command line tool for traffic analysis. Login to the Session Manager command line management interface to run this command.
- **Call Routing Test** - The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, from the System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

9.3. Avaya Session Border Controller for Enterprise Verification

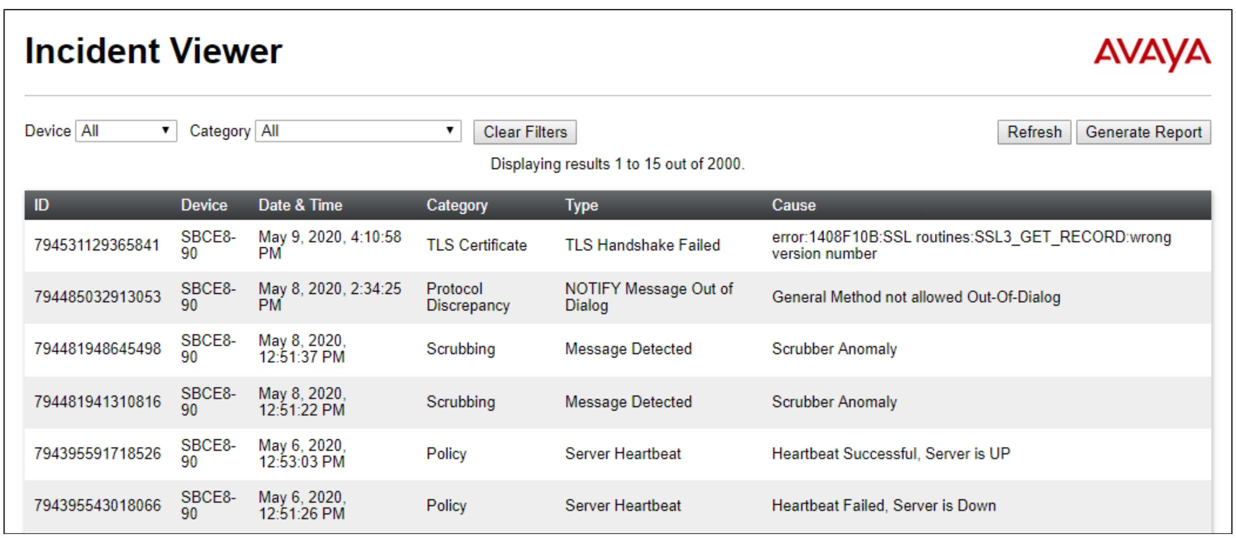
This section illustrates verifications from Avaya Session Border Controller for Enterprise.

9.3.1 Incidents

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

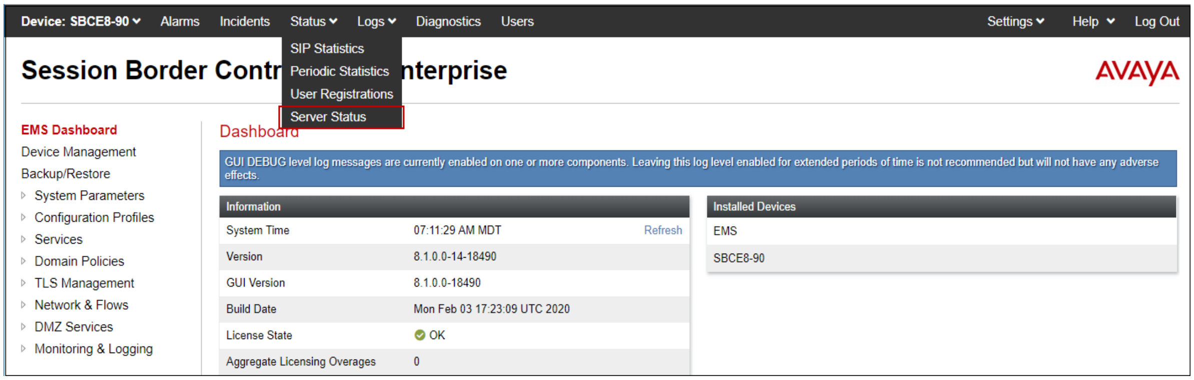


Use the Incident Viewer to verify Server Heartbeat and to troubleshoot routing failures.

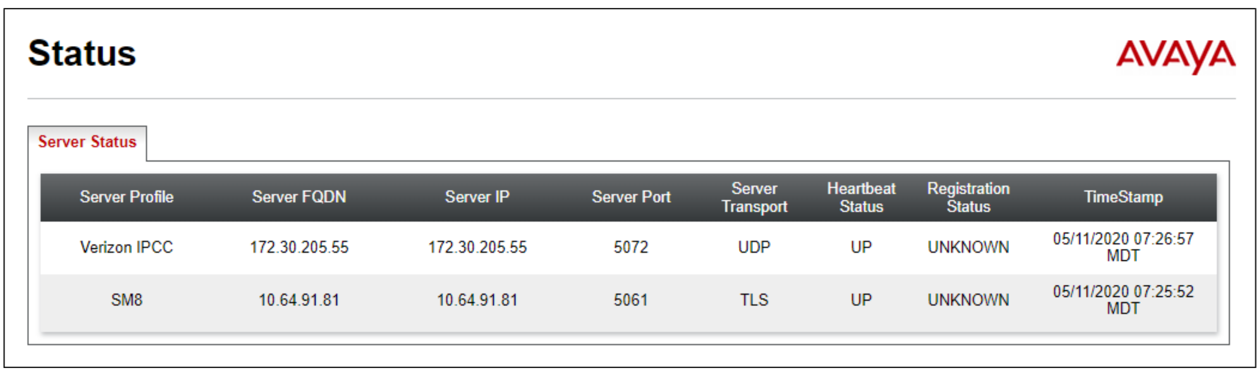


9.3.2 Server Status

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



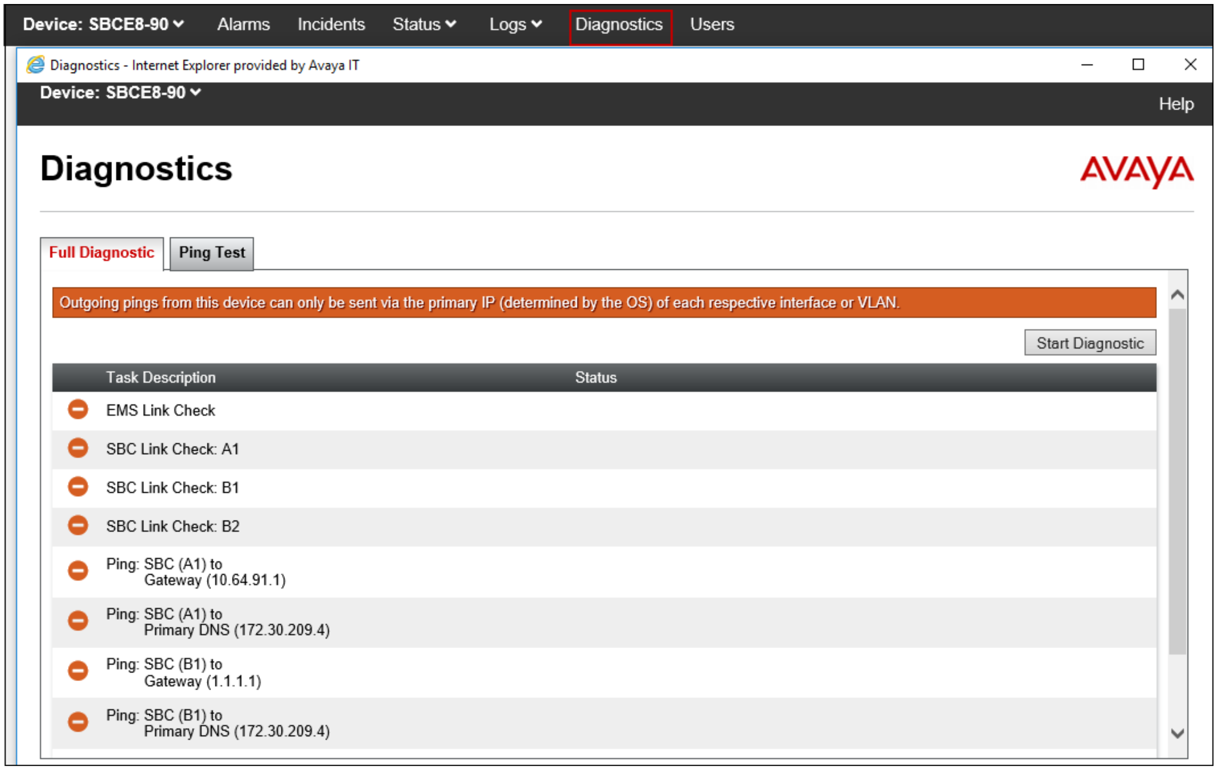
The **Server Status** screen provides information about the condition of the connection to the connected SIP Servers. This functionality requires Heartbeat to be enabled on the SIP Server Configuration profiles, as configured in **Section 7.8**.



Server Profile	Server FQDN	Server IP	Server Port	Server Transport	Heartbeat Status	Registration Status	TimeStamp
Verizon IPCC	172.30.205.55	172.30.205.55	5072	UDP	UP	UNKNOWN	05/11/2020 07:26:57 MDT
SM8	10.64.91.81	10.64.91.81	5061	TLS	UP	UNKNOWN	05/11/2020 07:25:52 MDT

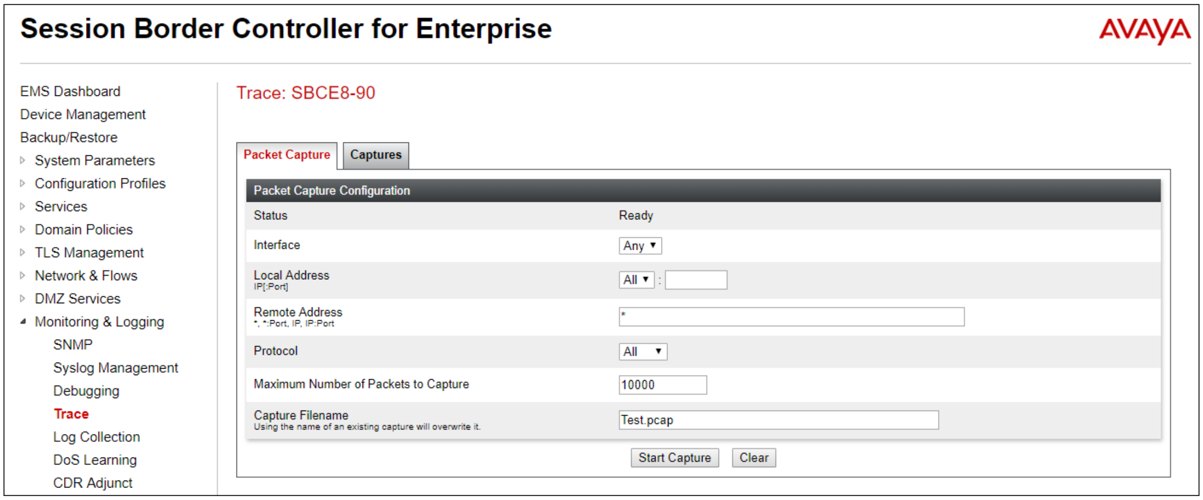
9.3.3 Diagnostics

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.

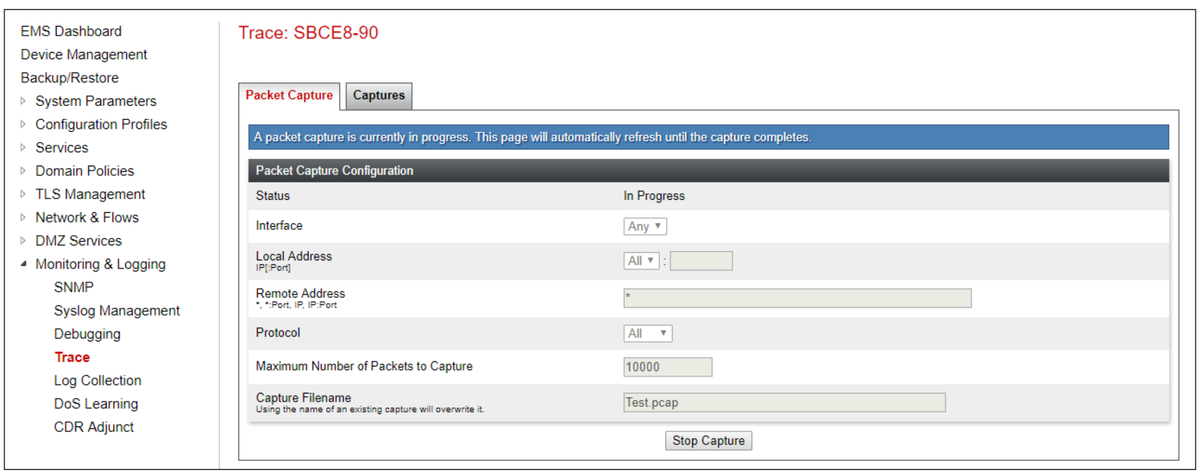


9.3.4 Tracing

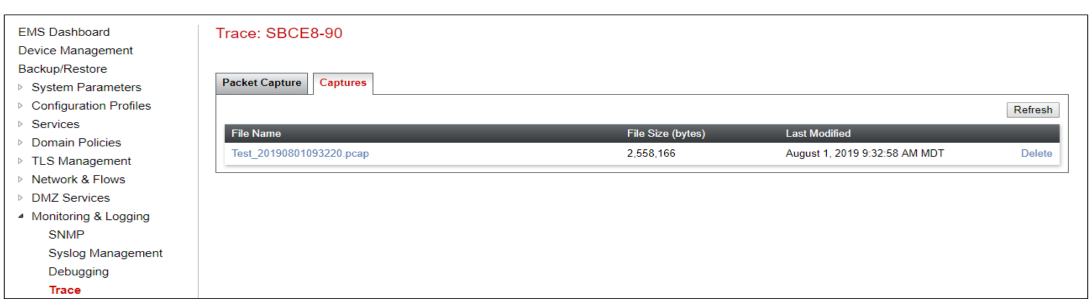
To take a call trace, navigate to **Monitoring & Logging → Trace** and select the **Packet Capture** tab. Populate the fields for the capture parameters and click **Start Capture** as shown below.



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



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



10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1 and Avaya Session Border Controller for Enterprise 8.1 can be configured to interoperate successfully with Verizon Business IP Contact Center Services suite. This solution enables inbound toll free calls over a Verizon Business VoIP Inbound SIP trunk service connection. In addition, these Application Notes further demonstrate that the Avaya Aura® Communication Manager implementation of SIP Network Call Redirection (SIP-NCR) can work in conjunction with Verizon's Business IP Contact Center services implementation of SIP-NCR to support call redirection over SIP trunks inclusive of passing User-User Information (UUI).

Please note that the sample configurations shown in these Application Notes are intended to provide configuration guidance to supplement other Avaya product documentation.

11. Additional References

11.1. Avaya

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

Avaya Aura® Session Manager/System Manager

[1] *Deploying Avaya Aura® Session Manager and Branch Session Manager in Virtualized Environment*, Release 8.1, Issue 3, March 2020
[2] *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 3, March 2020
[3] *Deploying Avaya Aura® System Manager in Virtualized Environment*, Release 8.1.x, Issue 4, March 2020
[4] *Administering Avaya Aura® System Manager for Release 8.1*, Release 8.1.x, Issue 5, March 2020

Avaya Aura® Communication Manager

[5] *Deploying Avaya Aura® Communication Manager in Virtualized Environment*, Release 8.1.x, Issue 4, March 2020
[6] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 6, March 2020
[7] *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 8.1.x, Issue 6, March 2020
[8] *Administering Avaya G430 Branch Gateway*, Release 8.1.x, Issue 3, March 2020
[9] *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 8.0.2, Issue 9, December 2019
[10] *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

[11] *Administering Avaya Session Border Controller for Enterprise*, Release 8.1, Issue 1, February 2020
[12] *Deploying Avaya Session Border Controller for Enterprise in Virtualized Environment* Release 8.1, Issue 1, February 2020
[13] *Avaya Session Border Controller for Enterprise Overview and Specification*, Release 8.1, Issue 1, February 2020

Avaya Aura® Messaging

[14] *Administering Avaya Aura® Messaging*, Release 7.1.0, Issue 9, April 2020

11.2. Verizon Business

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

- *Retail VoIP Interoperability Test Plan*
- *Network Interface Specification Retail VoIP Trunk Interface (for non-registering devices)*

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