



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, Avaya Aura® Experience Portal 7.2, and Avaya Session Border Controller for Enterprise 7.2 with Verizon Business IP Contact Center (IPCC) Services Suite – Issue 1.0

Abstract

These Application Notes describe a sample configuration of Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, Avaya Aura® Experience Portal 7.2, and Avaya Session Border Controller for Enterprise 7.2 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.

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1. Introduction

These Application Notes describe a sample configuration of Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, Avaya Aura® Experience Portal 7.2, and Avaya Session Border Controller for Enterprise 7.2 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 Experience Portal or 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 a newer version 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 <http://www.verizonenterprise.com/products/business-communications/customer-contact-solutions/>

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 10** 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.
- Inbound caller interaction with Experience Portal applications, including prompting, caller DTMF input, wait treatment (e.g., announcements and/or music on hold), Automatic Speech Recognition, and Text to Speech

- Experience Portal use of SIP REFER to redirect inbound calls, via the Avaya SBCE, to the appropriate Communication Manager agent extension
- Call and two-way talk path establishment between callers and Communication Manager agents following redirection from Experience Portal
- Inbound calls to a self-service Experience Portal application which forwards the call to 8YY or any other PSTN number over Verizon IPCC service using SIP REFER
- 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 three minutes of ring no answer conditions, 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 6.8.1**. 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 observations may be noteworthy:

- 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 Redirect.
- 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 theoretically allow a call to remain in Communication Manager vector processing upon failure of a vector-triggered REFER attempt. However, most such call scenarios could not be verified on the production Verizon circuit used for testing. On the production circuit, Verizon would send a BYE to terminate the call upon encountering REFER transfer failures, so there was no opportunity for the call to remain in Communication Manager vector processing. 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 8.3.9** for additional information.

2.3. Support

2.3.1 Avaya

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

2.3.2 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 configuration is comprised of the Avaya CPE location connected via a T1 Internet connection to the Verizon Business IPCC Services node. The Avaya CPE location simulates a customer site. 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 using destination port 5072, using UDP for transport. The PIP service defines a secure MPLS connection between the Avaya CPE T1 connection and the Verizon Business IPCC Services node.

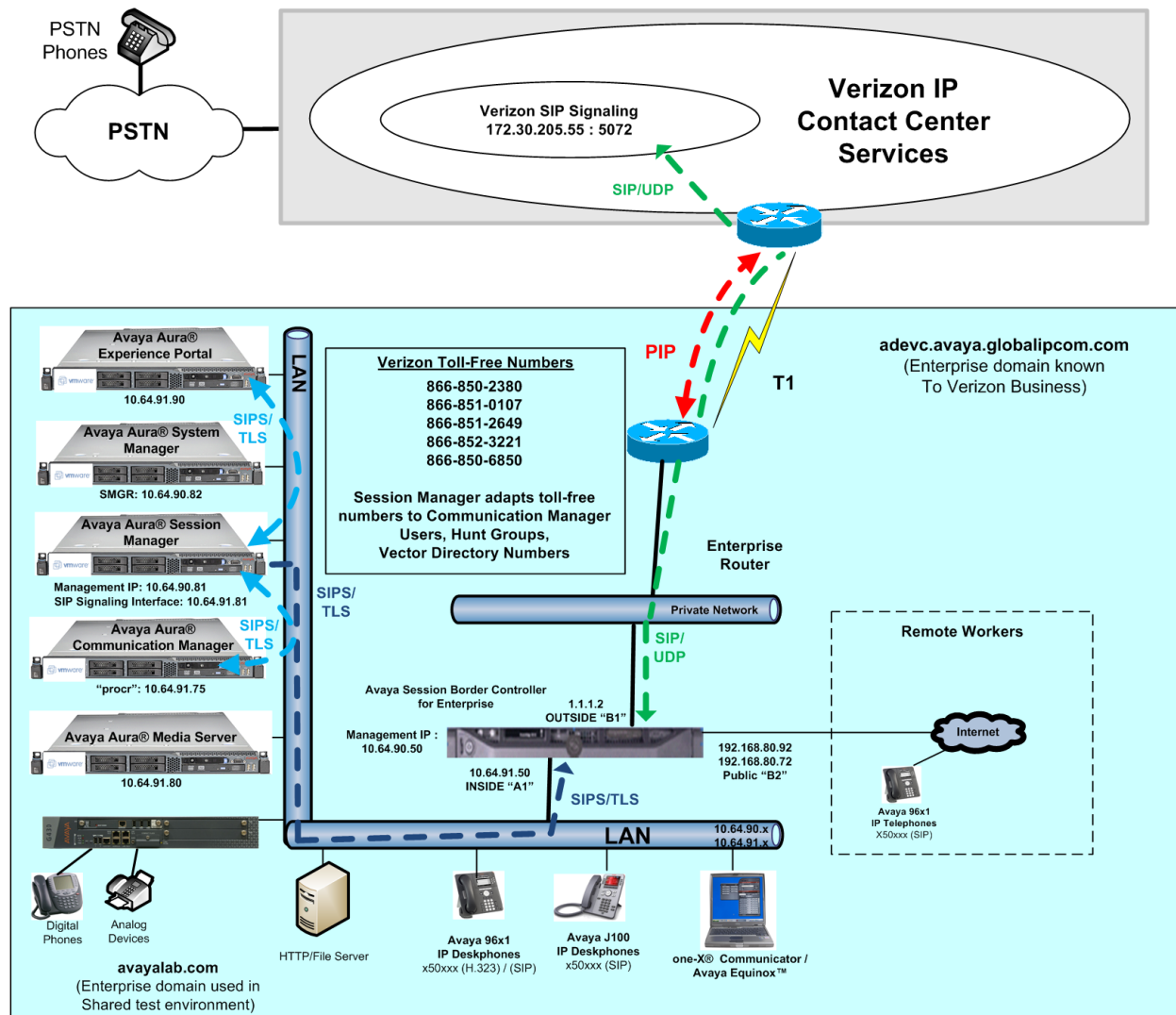


Figure 1: Avaya Interoperability Test Lab Configuration

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 8.0 and Communication Manager Release 8.0 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 5060 via TCP
- 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 – Avaya Aura® Communication Manager

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 10.1.1**.

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.

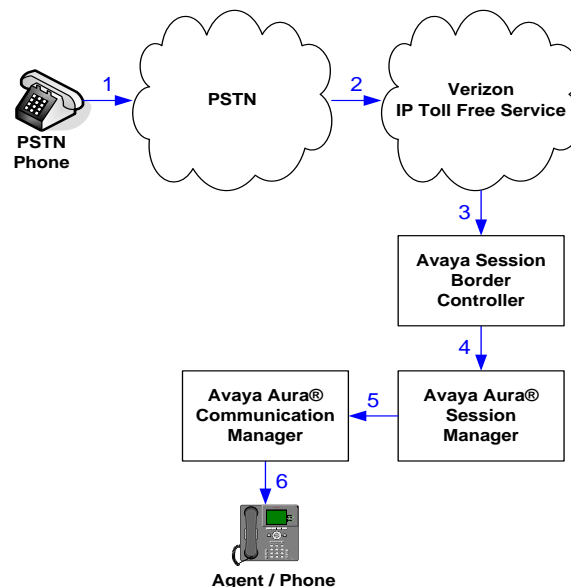


Figure 3: Inbound Verizon IP Toll Free Call – No Redirection

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 10.1.2** 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 (UI) 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.

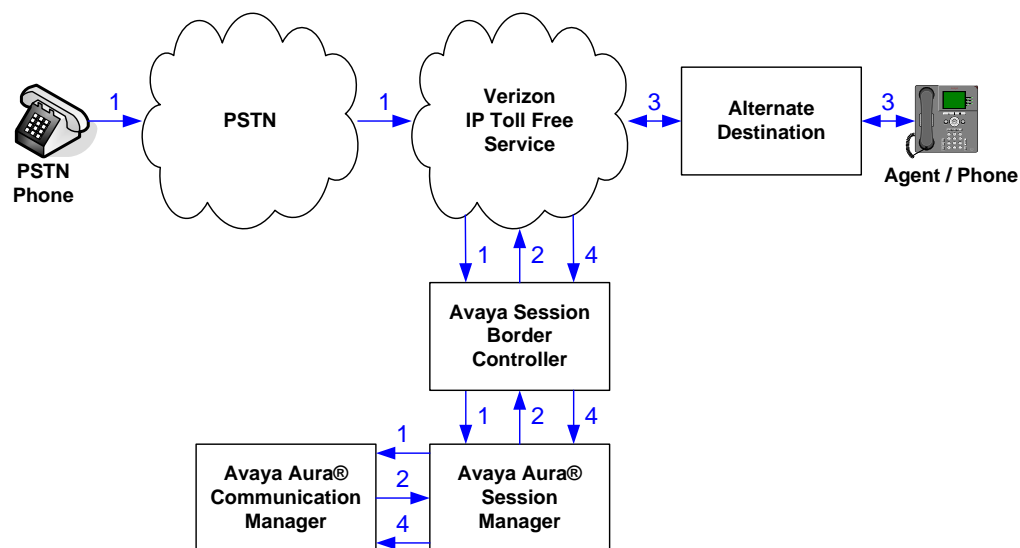


Figure 4: Inbound Verizon IP Toll Free– Post-Answer SIP REFER Redirection Successful

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 theory, if redirection is successful, 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**, except for egregious configuration errors, this “REFER error handling” scenario could not be verified on the production Verizon circuit used for testing. On the production circuit, Verizon sends a SIP BYE 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”, which precludes any further call processing by Communication Manager. As another example, in cases where misconfiguration is introduced to cause the Refer-To header to be malformed (e.g., no “+” in Refer-To), Verizon will send a BYE immediately after a “NOTIFY/sipfrag 603 Server Internal Error”. If REFER is configured in the vector, but Network Call Redirection is not enabled for the SIP trunk group, Communication Manager will not send the REFER to Verizon, and vector processing will continue at the step following the route-to step that would normally trigger the REFER.

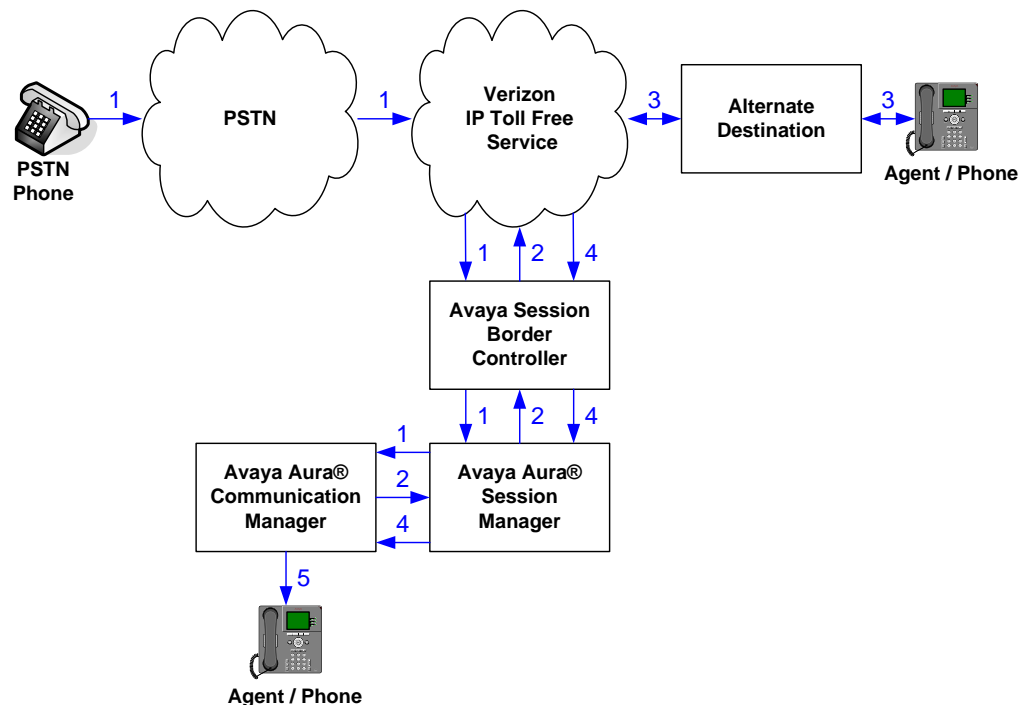


Figure 5: Inbound Verizon IP Toll Free– Post-Answer SIP REFER Redirection Unsuccessful

3.3. Call Flows – Avaya Aura® Experience Portal

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

3.3.1 Inbound IP Toll Free Call handled by Avaya Aura® Experience Portal

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

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

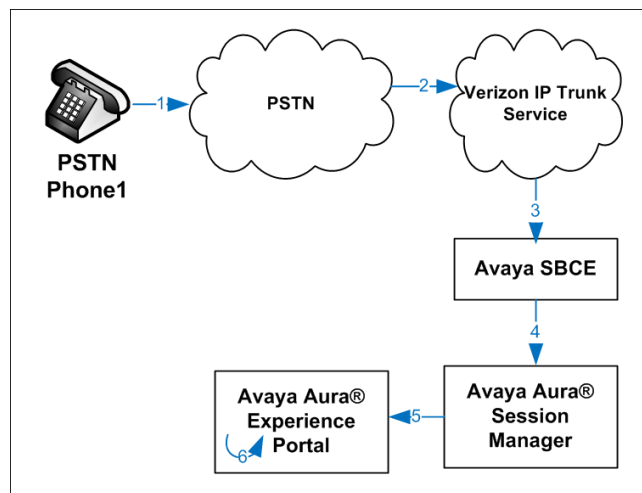


Figure 6: Inbound Call Handling Entirely by Avaya Aura® Experience Portal

3.3.2 Inbound IP Toll Free Call redirected to Avaya Aura® Communication Manager

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

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

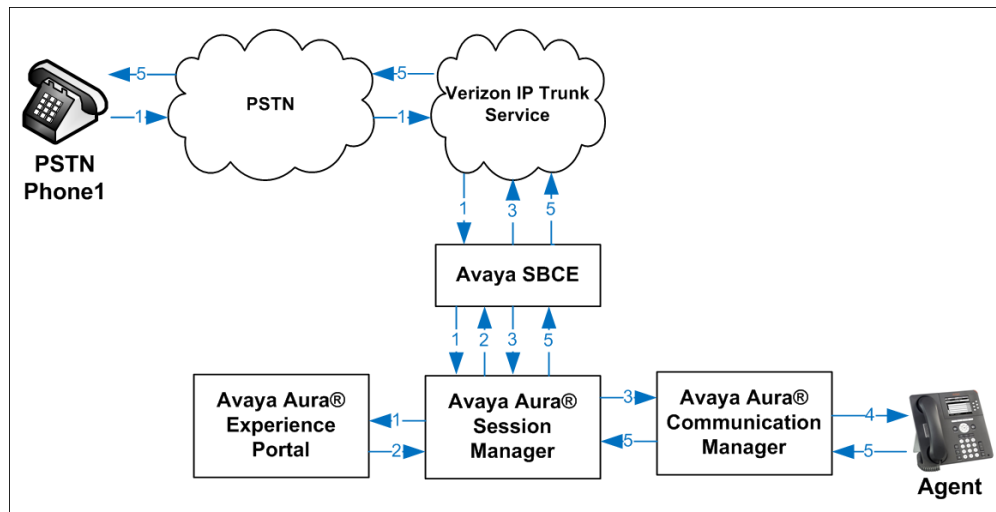


Figure 7: Avaya Aura® Experience Portal Transfers Call to Avaya Aura® Communication Manager

3.3.3 Inbound IP Toll Free Call redirected to PSTN number

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

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

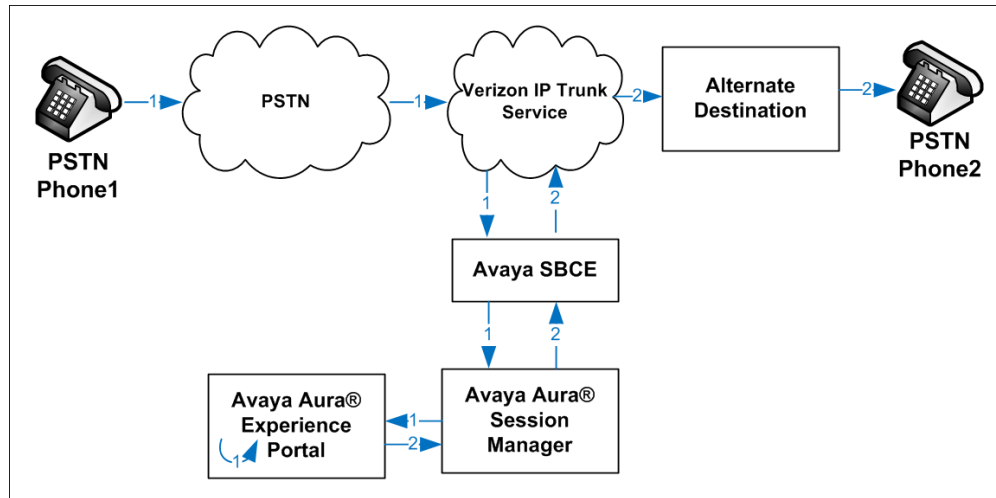


Figure 8: Inbound Call forwarded by Experience Portal to another PSTN number

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	8.0.0.0-R018x.00.0.822.0
Avaya Aura® System Manager	8.0.0.0.098174
Avaya Aura® Session Manager	8.0.0.0.800035
Avaya Session Border Controller for Enterprise	7.2.2.0-11-15522
Avaya Aura® Messaging	7.0 SP 0
Avaya Aura® Experience Portal	7.2.0.0.1117
Avaya Aura® Media Server	8.0.0.117
G450 Gateway	40.10.0
Avaya 96X1– Series Telephones (SIP)	R7.1.3.0.11
Avaya 96X1– Series Telephones (H.323)	R6.66.04
Avaya J100 – Series Telephones (SIP)	3.0.0.2.2
Avaya Equinox™ for Windows	3.4.0.152.46
Avaya 2400 – Series Digital Telephones	N/A

Table 1: Equipment and Software Used in the Sample Configuration

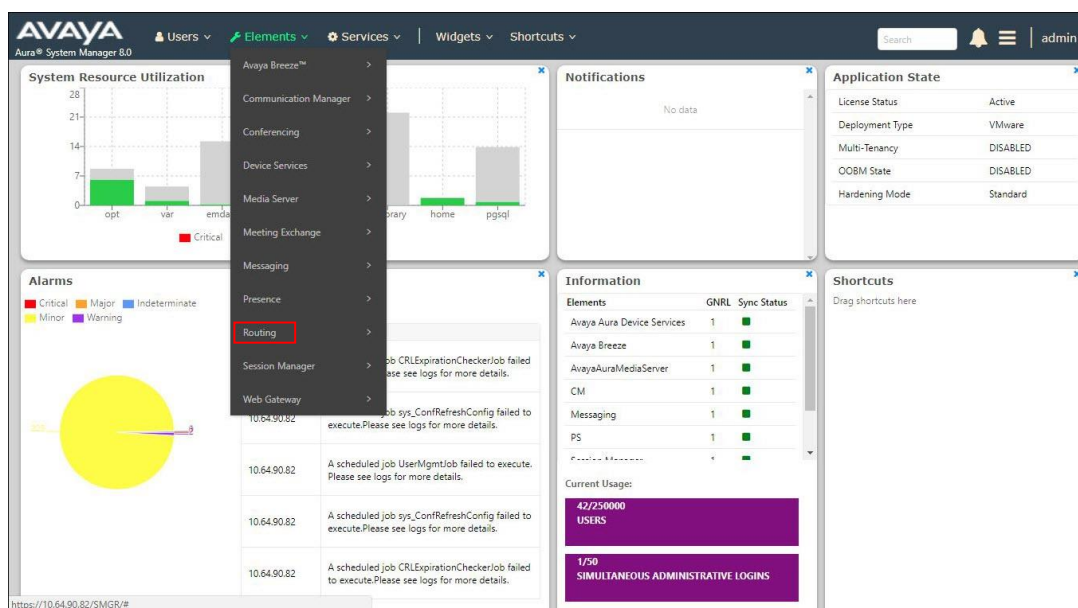
5. Configure Avaya Aura® Session Manager Release 8.0

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

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

- Define a SIP Domain.
- Define a Location for Customer Premises Equipment (CPE).
- Configure the Adaptation Modules that will be associated with the SIP Entities for Communication Manager, the Avaya SBCE, and Messaging.
- Define SIP Entities corresponding to Session Manager, Communication Manager, Experience Portal, the Avaya SBCE, and Messaging.
- Define Entity Links describing the SIP trunks between Session Manager, Communication Manager, Experience Portal, and Messaging, as well as the SIP trunks between the Session Manager and the Avaya SBCE.
- Define Routing Policies associated with the Communication Manager, Experience Portal, 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.

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



5.1. SIP Domain

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

Step 2 - Click **New**. 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.



5.2. Locations

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

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

5.2.1 Main Location

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

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

Step 2 - In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern:** Leave blank.
- **Notes:** Add a brief description.

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

Location Details Commit Cancel

General

* Name:

Notes:

Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

Overall Managed Bandwidth

Managed Bandwidth Units:

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): Kbit/Sec

* Minimum Multimedia Bandwidth: Kbit/Sec

* Default Audio Bandwidth: Kbit/sec

Alarm Threshold

Overall Alarm Threshold: %

Multimedia Alarm Threshold: %

* Latency before Overall Alarm Trigger: Minutes

* Latency before Multimedia Alarm Trigger: Minutes

Location Pattern

0 Items Filter: Enable

IP Address Pattern	Notes

5.2.2 Common Location

To configure the Avaya SBCE Location, repeat the steps in **Section 5.2.1** with the following changes:

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

5.3. Configure Adaptations

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

- 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.

5.3.1 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.

Home / Elements / Routing / Adaptations

Adaptation Details Commit Cancel Help ?

General

* Adaptation Name:

* Module Name:

Module Parameter Type:

Name	Value
fromto	true
osrcd	avayalab.com

Select : All, None

Egress URI Parameters:

Notes:

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).

1. **Example 1 – 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

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

Select : All, None

5.3.2 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 5.3.1** 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”

The screenshot shows the 'Adaptation Details' page in the Avaya Session Manager interface. The left sidebar contains a navigation menu with options: Home, Routing, Domains, Locations, Adaptations (selected), SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Adaptation Details' and has a 'General' tab selected. At the top right of the main area are 'Commit' and 'Cancel' buttons. The form contains the following fields:

- Adaptation Name:** SBC1-Adaptation for Verizon
- Module Name:** VerizonAdapter (selected from a dropdown)
- Module Parameter Type:** Name-Value Parameter (selected from a dropdown)

Below these fields is a table for Name-Value Parameters. The table has columns for 'Name' and 'Value'. There are two entries:

Name	Value
eRHdrs	"AV-Global-Session-ID, Alert-Info, Endpoint-View, P-AV-Message-Id, P-Charging-Vector, P-Location, AV-Correlation-ID, Av-Secure-Indication"
fromto	true

Below the table is a 'Select' dropdown menu with options 'All' and 'None'. At the bottom of the form are fields for 'Egress URI Parameters' and 'Notes' (containing 'SBC - Verizon IPT').

5.4. SIP Entities

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

- Session Manager (**Section 5.4.1**).
- Communication Manager for Verizon trunk access (**Section 5.4.2**) – 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 5.4.3**) – 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 5.4.4**) – 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 5.4.5**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls to/from Messaging.
- Experience Portal (**Section 5.4.6**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls to/from Experience Portal.

Note – In the reference configuration, TLS is used as the transport protocol between Session Manager and Communication Manager (ports 5061 and 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.

5.4.1 Avaya Aura® Session Manager SIP Entity

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

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

- **Name** – Enter a descriptive name (e.g., **SessionManager**).
- **FQDN or IP Address** – Enter the IP address of Session Manager signaling interface, (*not* the management interface), provisioned during installation (e.g., **10.64.91.81**).
- **Type** – Verify **Session Manager** is selected.
- **Location** – Select location **Main** (**Section 5.2.1**).
- **Outbound Proxy** – (Optional) Leave blank or select another SIP Entity. For calls to SIP domains for which Session Manager is not authoritative, Session Manager routes those calls to this **Outbound Proxy** or to another SIP proxy discovered through DNS if **Outbound Proxy** is not specified.
- **Time Zone** – Select the time zone in which Session Manager resides.
- **Minimum TLS Version** – Select 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.

The screenshot displays the 'SIP Entity Details' configuration page. The left-hand navigation pane is expanded to 'SIP Entities'. The main content area is titled 'SIP Entity Details' and includes 'Commit' and 'Cancel' buttons. The 'General' tab is active, showing the following fields: 'Name' (Session Manager), 'IP Address' (10.64.91.81), 'SIP FQDN' (empty), 'Type' (Session Manager), 'Notes' (empty), 'Location' (Main), 'Outbound Proxy' (empty), 'Time Zone' (America/Fortaleza), 'Minimum TLS Version' (Use Global Setting), and 'Credential name' (empty). The 'Monitoring' tab is also visible, showing 'SIP Link Monitoring' and 'CRLF Keep Alive Monitoring', both set to 'Use Session Manager Configuration'.

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 5.5**. Click on **Add** and provision entries as follows:

- **Port** – Enter **5061**
- **Protocol** – Select **TLS**
- **Default Domain** – Select a SIP domain administered in **Section 5.1** (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**.

The screenshot shows a web interface titled "Listen Ports". At the top, there are "Add" and "Remove" buttons. Below them, it says "3 Items" with a refresh icon and a "Filter: Enable" link. The main part of the interface is a table with the following columns: "Listen Ports", "Protocol", "Default Domain", "Endpoint", and "Notes". There are three rows of data:

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

At the bottom left of the table, there is a "Select : All, None" option.

Note – The **Entity Links** section of the form (not shown) will be automatically populated when the Entity Links are defined in **Section 5.5**. The **SIP Responses to an OPTIONS Request** section of the form is not used in the reference configuration.

5.4.2 Avaya Aura® Communication Manager SIP Entity – Public Trunk

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

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

- **Name** – Enter a descriptive name (e.g., **CM-TG2**).
- **FQDN or IP Address** – Enter the IP address of Communication Manager Processor Ethernet (procr) described in **Section 6.4** (e.g., **10.64.91.65**).
- **Type** – Select **CM**.
- **Adaptation** – Select the Adaptation **CM-TG2-VzIPCC** administered in **Section 5.3.1**.
- **Location** – Select a Location **Main** administered in **Section 5.2.1**.
- **Time Zone** – Select the time zone in which Communication Manager resides.
- In the **SIP Link Monitoring** section of the **SIP Entity Details** page select:
 - Select **Use Session Manager Configuration** for **SIP Link Monitoring** and the **CRLF Keep Alive Monitoring** fields. Use the default values for the remaining parameters.

Step 3 - Click on **Commit**.

The screenshot displays the 'SIP Entity Details' configuration page. The breadcrumb trail at the top is 'Home / Elements / Routing / SIP Entities'. The page title is 'SIP Entity Details' with 'Commit' and 'Cancel' buttons. The 'General' section includes fields for Name (CM-TG2), FQDN or IP Address (10.64.91.65), Type (CM), Notes (Trunk Group 2 - Vz-Toll-Free inbound), Adaptation (CM-TG2-VzIPCC), Location (Main), Time Zone (America/Denver), SIP Timer B/F (4), Minimum TLS Version (Use Global Setting), Credential name, Securable (checkbox), Call Detail Recording (none), Loop Detection Mode (Off), SIP Link Monitoring (Use Session Manager Configuration), CRLF Keep Alive Monitoring (Use Session Manager Configuration), Supports Call Admission Control (checkbox), Shared Bandwidth Manager (checkbox), Primary Session Manager Bandwidth Association, and Backup Session Manager Bandwidth Association. The 'Loop Detection' and 'Monitoring' sections are also visible.

5.4.3 Avaya Aura® Communication Manager SIP Entity – Local Trunk

To configure the Communication Manager Local trunk SIP Entity, repeat the steps in **Section 5.4.2** with the following changes:

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

5.4.4 Avaya Session Border Controller for Enterprise SIP Entity

Repeat the steps in **Section 5.4.2** with the following changes:

- **Name** – Enter a descriptive name (e.g., **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 8.5.1**).
- **Type** – Select **SIP Trunk**.
- **Adaptations** – Select Adaptation **SBC1-Adaptation for Verizon** (**Section 5.3.2**).

5.4.5 Avaya Aura® Messaging SIP Entity

Repeat the steps in **Section 5.4.2** 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.

5.4.6 Avaya Aura® Experience Portal SIP Entity

Repeat the steps in **Section 5.4.2** with the following changes:

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

5.5. Entity Links

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

- Session Manager to Communication Manager Public trunk (**Section 5.5.1**).
- Session Manager to Communication Manager Local trunk (**Section 5.5.2**).
- Session Manager to Avaya SBCE (**Section 5.5.3**).
- Session Manager to Messaging (**Section 5.5.4**).
- Session Manager to Experience Portal (**Section 5.5.5**).

Note – Once the Entity Links have been committed, the link information will also appear on the associated SIP Entity pages configured in **Section 5.4**.

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

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

Step 1 - In the left pane under **Routing**, click on **Entity Links**, then click on **New** (not shown).

Step 2 - Continuing in the **Entity Links** page, provision the following:

- **Name** – Enter a descriptive name for this link to Communication Manager (e.g., **SM to CM TG2**).
- **SIP Entity 1** – Select the SIP Entity administered in **Section 5.4.1** for Session Manager (e.g., **SessionManager**).
- **Protocol** – Select **TLS** (see **Section 6.8.1**).
- **SIP Entity 1 Port** – Enter **5071**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.2** for the Communication Manager public entity (e.g., **CM-TG2**).
- **SIP Entity 2 Port** – Enter **5071** (see **Section 6.8.1**).
- **Connection Policy** – Select **trusted**.
- Leave other fields as default.

Step 3 - Click on **Commit**.

The screenshot shows the 'Entity Links' configuration page in the Avaya Aura interface. The page has a breadcrumb trail: Home / Elements / Routing / Entity Links. There are 'Commit' and 'Cancel' buttons at the top right. Below the header, there is a table with one item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, DNS Override, Connection Policy, Deny New Service, and Notes. The item in the table is 'SM to CM TG2', with SIP Entity 1 set to 'SessionManager', Protocol set to 'TLS', and SIP Entity 2 set to 'CM-TG2'. The Connection Policy is set to 'trusted'. There is a 'Filter: Enable' button at the top right of the table. At the bottom left, there is a 'Select: All, None' dropdown.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy	Deny New Service	Notes
SM to CM TG2	SessionManager	TLS	5071	CM-TG2	5071		trusted		

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

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

- **Name** – Enter a descriptive name for this link to Communication Manager (e.g., **SM to CM TG3**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.3** for the Communication Manager local entity (e.g., **CM-TG3**).
- **SIP Entity 2 Port** – Enter **5061** (see **Section 6.8.1**).

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

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

- **Name** – Enter a descriptive name for this link to the Avaya SBCE (e.g., **SM to SBC1**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.4** for the Avaya SBCE entity (e.g., **SBC1**).
- **SIP Entity 2 Port** – Enter **5061**.

5.5.4 Entity Link to Avaya Aura® Messaging

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

- **Name** – Enter a descriptive name for this link to Messaging (e.g., **SM to AAM**).

- SIP Entity 1 **Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.5** for the Aura® Messaging entity (e.g., **Aura Messaging**).
- SIP Entity 2 **Port** – Enter **5061**.

5.5.5 Entity Link to Avaya Aura® Experience Portal

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

- **Name** – Enter a descriptive name for this link to Messaging (e.g., **SM to ExperiencePortal**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.6** for the Experience Portal entity (e.g., **ExperiencePortal**).
- **SIP Entity 2 Port** – Enter **5061**

5.6. Time Ranges

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

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

Step 3 - Click on **Commit**. Repeat these steps to provision additional time ranges as required.

The screenshot shows the 'Time Ranges' configuration page. The left sidebar has a 'Routing' menu with 'Time Ranges' selected. The main area shows a table with one item named '24/7' that is active on all days of the week from 00:00 to 23:59.

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

5.7. Routing Policies

In this section, the following Routing Policies are administered:

- Inbound calls to Communication Manager extensions (**Section 5.7.1**).
- Inbound calls to Aura® Messaging (**Section 5.7.2**).
- Inbound calls to Experience Portal (**Section 5.7.3**).

5.7.1 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 TG1**), 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.

Home / Elements / Routing / Routing Policies

Routing Policy Details

CommitCancel

Help ?

General

* Name: To CM TG2

Disabled: ☐

* Retries: 0

Notes: Trunk Group 2 VzIPCC to CM

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CM-TG2	10.64.91.65	CM	Trunk Group 2 - Vz-Toll-Free inbound

Step 4 - In the **SIP Entities** list page, select the SIP Entity administered in **Section 5.4.2** for the Communication Manager public SIP Entity (**CM-TG2**), and click on **Select**.

SIP Entities				
15 Items		Filter: Enable		
	Name	FQDN or IP Address	Type	Notes
<input type="radio"/>	Aura Messaging	10.64.91.54	Messaging	Aura Messaging
<input type="radio"/>	Breeze	10.64.91.17	Avaya Breeze	
<input type="radio"/>	CM-TG1	10.64.91.65	CM	Trunk Group 1 - CM to Vz-IPT
<input checked="" type="radio"/>	CM-TG2	10.64.91.65	CM	Trunk Group 2 - Vz-Toll-Free inbound
<input type="radio"/>	CM-TG3	10.64.91.65	CM	Trunk Group 3 - CM to Enterprise
<input type="radio"/>	CM-TG4	10.64.91.65	CM	Trunk Group 4 - ATT IPTF
<input type="radio"/>	CM-TG5	10.64.91.65	CM	Trunk Group 5 - ATT IPFR
<input type="radio"/>	CS1000	10.80.140.103	Other	CS1000 7.65
<input type="radio"/>	IP500	10.64.19.70	Other	IP Office
<input type="radio"/>	Presence	10.64.91.17	Presence Services	
<input type="radio"/>	SBC1	10.64.91.50	SIP Trunk	Avaya SBC-1 to PSTN
<input type="radio"/>	SBC2	10.64.91.100	SIP Trunk	Avaya SBC-2 to PSTN
<input type="radio"/>	SBCE-ipv6	10.64.91.40	SIP Trunk	SBCE for IPv6 testing
<input type="radio"/>	SBCE-ipv6-Toll Free	10.64.91.41	SIP Trunk	SBCE for IPv6 testing
<input type="radio"/>	SessionManager	10.64.91.11	Session Manager	Session Manager

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

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

Step 7 - Returning to the **Routing Policy Details** page in the **Time of Day** section, enter a **Ranking** of 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 5.8**) they will appear in the **Dial Pattern** section of this form.

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel

Help ?

General

* Name: To CM TG2

Disabled:

* Retries: 0

Notes: Trunk Group 2 VzIPCC to CM

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CM-TG2	10.64.91.65	CM	Trunk Group 2 - Vz-Toll-Free inbound

Time of Day

Add Remove View Gaps/Overlaps

1 Item

Filter: Enable

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

Select : All, None

5.7.2 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 5.7.1** 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 5.4.5** for Aura® Messaging (e.g., **AAM**).

5.7.3 Routing Policy for Inbound Routing to Experience Portal

This routing policy is for inbound calls to Experience Portal. Repeat the steps in **Section 5.7.1** with the following differences:

- Enter a descriptive **Name** (e.g., **To Experience Portal**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.
- In the **SIP Entities** list page, select the SIP Entity administered in **Section 5.4.6** for Experience Portal (e.g., **ExperiencePortal**).

5.8. 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 5.3.1** 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**.

The screenshot shows the 'Dial Pattern Details' page with the 'General' tab selected. The form contains the following fields and values:

- Pattern:** 8668502380
- Min:** 10
- Max:** 10
- Emergency Call:** ☐
- Emergency Priority:** 1
- Emergency Type:** (empty field)
- SIP Domain:** avayalab.com (selected from a dropdown menu)
- Notes:** (empty text area)

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

Step 3 - Scrolling down to the **Originating Location and Routing Policies** section of the **Dial Pattern Details** page (not shown), click on **Add**.

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

Step 5 - In the **Routing Policies** section, check the checkbox corresponding to the Routing Policy administered for routing calls to the Communication Manager public trunk in **Section 5.7.1** (e.g., **To CM TG2**), and click on **Select** (not shown).

Originating Location

☐ Apply The Selected Routing Policies to All Originating Locations

4 Items [Filter: Enable](#)

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

Select : All, None

Routing Policies

12 Items [Filter: Enable](#)

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	To AAM	<input type="checkbox"/>	Aura Messaging	
<input type="checkbox"/>	To CM TG1	<input type="checkbox"/>	CM-TG1	Trunk Group 1 PSTN1 to CM
<input checked="" type="checkbox"/>	To CM TG2	<input type="checkbox"/>	CM-TG2	Trunk Group 2 VzIPCC to CM
<input type="checkbox"/>	To CM TG3	<input type="checkbox"/>	CM-TG3	Enterprise Traffic
<input type="checkbox"/>	To CM TG4	<input type="checkbox"/>	CM-TG4	Trunk Group 4 PSTN4 to CM
<input type="checkbox"/>	To CM-TG5	<input type="checkbox"/>	CM-TG5	Trunk Group 5 PSTN5 to CM
<input type="checkbox"/>	To CS1000	<input type="checkbox"/>	CS1000	
<input type="checkbox"/>	To IP500	<input type="checkbox"/>	IP500	
<input type="checkbox"/>	To SBC1	<input type="checkbox"/>	SBC1	
<input type="checkbox"/>	To SBC2	<input type="checkbox"/>	SBC2	
<input type="checkbox"/>	To SBCE-IPv6	<input type="checkbox"/>	SBCE-ipv6	
<input type="checkbox"/>	to SBCE-IPv6 TollFree	<input type="checkbox"/>	SBCE-ipv6-Toll Free	

Select : All, None

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.

Home / Elements / Routing / Dial Patterns [Help ?](#)

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

General

* Pattern: 8668502380

* Min: 10

* Max: 10

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: avayalab.com

Notes:

Originating Locations and Routing Policies

[Add](#) [Remove](#)

1 Item [Filter: Enable](#)

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Common	SBC to PSTN	To CM TG2	0	<input type="checkbox"/>	CM-TG2	Trunk Group 2 VzIPCC to CM

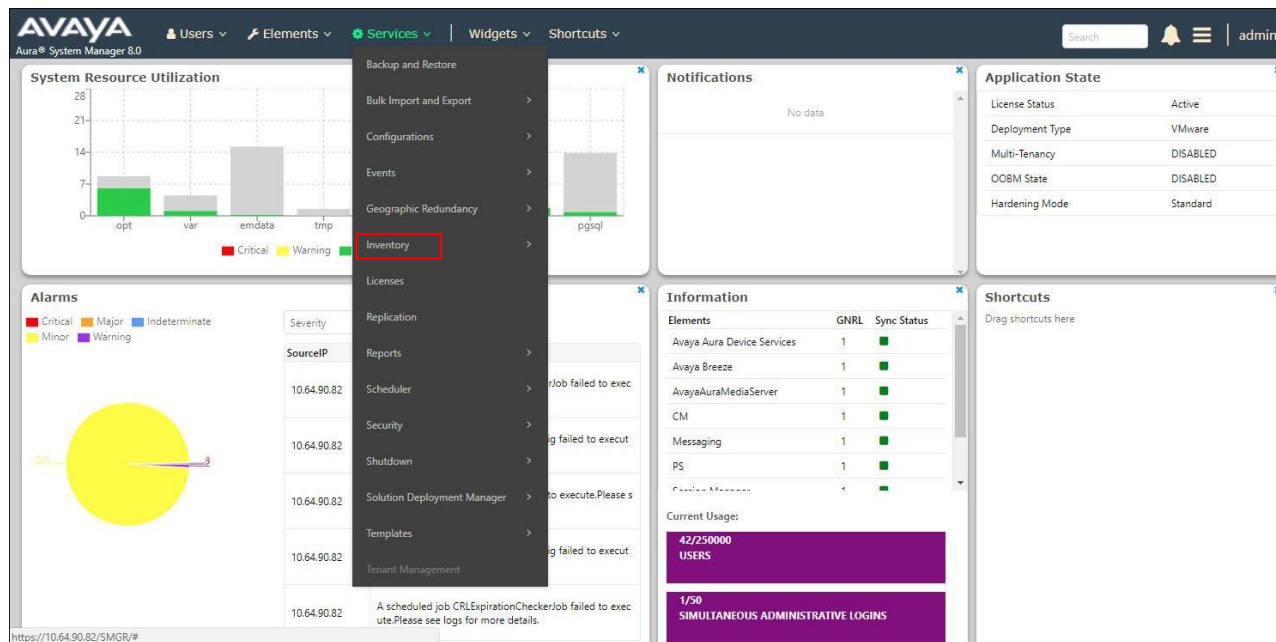
Select : All, None

5.9. Verify TLS Certificates – Session Manager

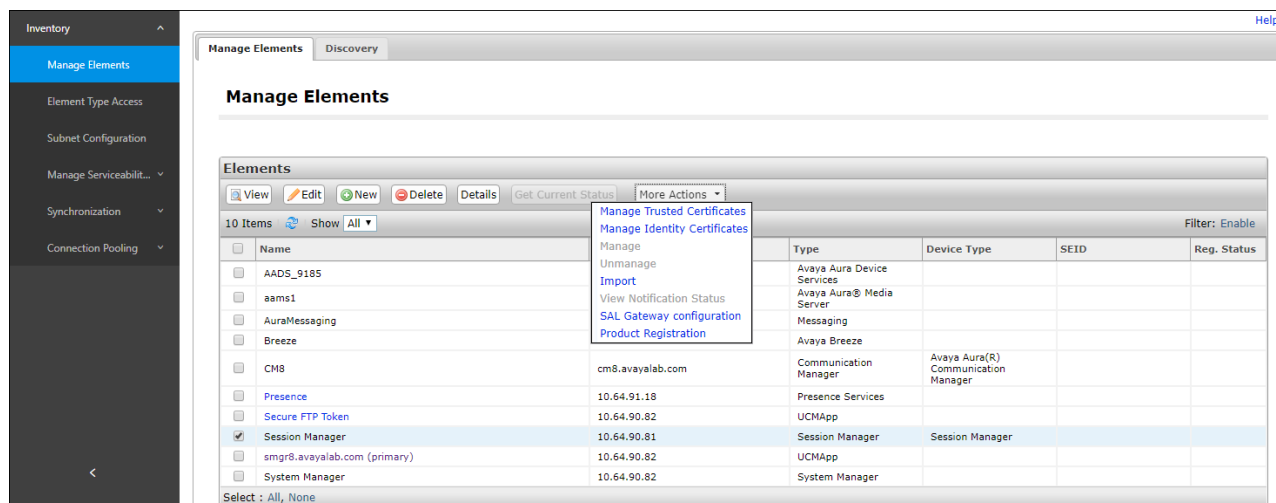
Note – Testing was done with System Manager signed identity certificates. The procedure to obtain and install certificates is outside the scope of these Application Notes.

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

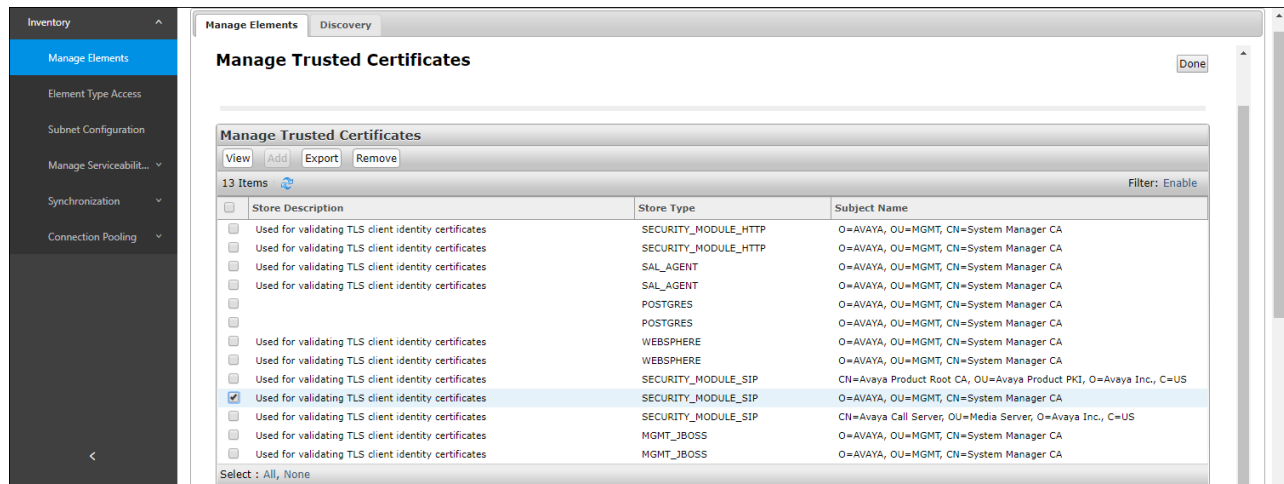
Step 1 - From the **Home** screen, under the **Services** heading, select **Inventory**.



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

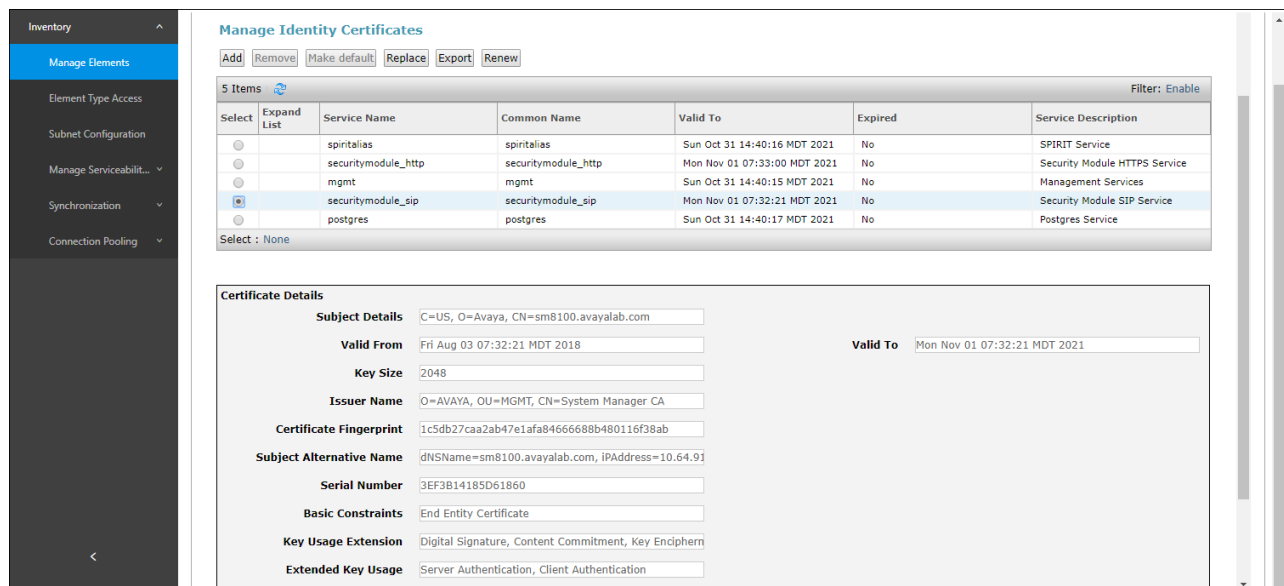


Step 3 - Verify the **System Manager Certificate Authority** certificate is listed in the trusted store, **SECURITY_MODULE_SIP**. Click **Done** to return to the previous screen.



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).



6. Configure Avaya Aura® Communication Manager Release 8.0

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.

6.1. 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: 2400		1	
Maximum Administered Remote Office Trunks: 4000		0	
Maximum Concurrently Registered Remote Office Stations: 2400		0	
Maximum Concurrently Registered IP eCons: 68		0	
Max Concur Registered Unauthenticated H.323 Stations: 100		0	
Maximum Video Capable Stations: 2400		3	
Maximum Video Capable IP Softphones: 2400		10	
Maximum Administered SIP Trunks: 4000		60	
Maximum Administered Ad-hoc Video Conferencing Ports: 4000		0	
Maximum Number of DS1 Boards with Echo Cancellation: 80		0	

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

display system-parameters customer-options		Page 4 of 12
OPTIONAL FEATURES		
Abbreviated Dialing Enhanced List? y	Audible Message Waiting? y	
Access Security Gateway (ASG)? n	Authorization Codes? y	
Analog Trunk Incoming Call ID? y	CAS Branch? n	
A/D Grp/Sys List Dialing Start at 01? y	CAS Main? n	
Answer Supervision by Call Classifier? y	Change COR by FAC? n	
ARS? y	Computer Telephony Adjunct Links? y	
ARS/AAR Partitioning? y	Cvg Of Calls Redirected Off-net? y	
ARS/AAR Dialing without FAC? n	DCS (Basic)? y	
ASAI Link Core Capabilities? n	DCS Call Coverage? y	
ASAI Link Plus Capabilities? n	DCS with Rerouting? y	
Async. Transfer Mode (ATM) PNC? n	Digital Loss Plan Modification? y	
Async. Transfer Mode (ATM) Trunking? n	DS1 MSP? y	
ATM WAN Spare Processor? n	DS1 Echo Cancellation? y	
ATMS? y		
Attendant Vectoring? y		

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

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

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

display system-parameters customer-options		Page 6 of 12
OPTIONAL FEATURES		
Multinational Locations? n	Station and Trunk MSP? y	
Multiple Level Precedence & Preemption? n	Station as Virtual Extension? y	
Multiple Locations? n		
Personal Station Access (PSA)? y	System Management Data Transfer? n	
PNC Duplication? n	Tenant Partitioning? y	
Port Network Support? y	Terminal Trans. Init. (TTI)? y	
Posted Messages? y	Time of Day Routing? y	
	TN2501 VAL Maximum Capacity? y	
	Uniform Dialing Plan? y	
Private Networking? y	Usage Allocation Enhancements? y	
Processor and System MSP? y		
Processor Ethernet? y	Wideband Switching? y	
	Wireless? n	
Remote Office? y		
Restrict Call Forward Off Net? y		
Secondary Data Module? y		

6.2. System-Parameters Features

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

change system-parameters features		Page 1 of 19
FEATURE-RELATED SYSTEM PARAMETERS		
Self Station Display Enabled? y		
Trunk-to-Trunk Transfer: all		
Automatic Callback with Called Party Queuing? n		
Automatic Callback - No Answer Timeout Interval (rings): 3		
Call Park Timeout Interval (minutes): 10		
Off-Premises Tone Detect Timeout Interval (seconds): 20		
AAR/ARS Dial Tone Required? y		
Music (or Silence) on Transferred Trunk Calls? all		
DID/Tie/ISDN/SIP Intercept Treatment: attendant		
Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred		
Automatic Circuit Assurance (ACA) Enabled? n		
Abbreviated Dial Programming by Assigned Lists? n		
Auto Abbreviated/Delayed Transition Interval (rings): 2		
Protocol for Caller ID Analog Terminals: Bellcore		
Display Calling Number for Room to Room Caller ID Calls? n		

6.3. Dial Plan

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

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

- 5-digit extensions with a **Call Type** of **ext** beginning with:
 - The digits **1** and **2** for Communication Manager extensions.
- 3-digit dial access code (indicated with a **Call Type** of **dac**), e.g., access code ***xx** for SIP Trunk Access Codes (TAC). See the trunk forms in **Section 6.8**.

change dialplan analysis						Page 1 of 12					
DIAL PLAN ANALYSIS TABLE											
Location: all											
Percent Full: 1											
	Dialed	Total	Call	Dialed	Total	Call	Dialed	Total	Call		
	String	Length	Type	String	Length	Type	String	Length	Type		
1		5	ext								
2		5	ext								
3		5	ext								
4		5	ext								
5		5	ext								
60		3	ext								
66		2	fac								
7		5	ext								
8		5	ext								
9		1	fac								
*		3	dac								

6.4. Node Names

Node names define IP addresses to various Avaya components in the enterprise. In the reference configuration a Processor Ethernet (procr) based Communication Manager platform is used. Note that the Communication Manager procr name and IP address are entered during installation. The procr IP address was used to define the Communication Manager SIP Entities in **Section 5.4**.

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

- Session Manager SIP signaling interface (e.g., **SM** and **10.64.91.81**).
- Media Server (e.g., **AMS** and **10.64.91.80**). The Media Server node name is only needed if a Media Server is present.

change node-names ip

Page 1 of 2

IP NODE NAMES

Name	IP Address
AMS	10.64.91.80
SM	10.64.91.81
default	0.0.0.0
procr	10.64.91.75
procr6	::

6.5. 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
                                                           Gatekeeper Priority: 5

Network Region: 1

                                                           IPV4 PARAMETERS
Node Name: procr                                           IP Address: 10.64.91.75

Subnet Mask: /24
```

6.6. IP Codec Sets

6.6.1 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          20
3: G.729A             n          20

Media Encryption                                           Encrypted SRTCP: enforce-unenc-srtcp
1: 1-srtp-aescml28-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	
	Mode	Redun- dancy	Packet 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:			

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

This IP codec set will be used for Verizon Business IPCC calls. Repeat the steps in **Section 6.6.1** with the following changes:

- Provision the codecs in the order shown below.
- 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 1 of 2

IP CODEC SET

Codec Set: 2

Audio Codec	Silence Suppression	Frames Per Pkt	Packet 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

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

	Mode	Redun- dancy	Packet 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

2:

6.7. Network Regions

Network regions provide a means to logically group resources. In the shared Communication Manager configuration used for the testing, the Avaya G450 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.

6.7.1 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 (see **Section 5.1**).
- Enter **1** for the **Codec Set** parameter.
- **Intra-region IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible within the same region.
- **Inter-region IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible between regions.

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

Step 2 - On page 2 of the form:

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

change ip-network-region 1		Page 2 of 20
IP NETWORK REGION		
RTCP Reporting to Monitor Server Enabled? y		
RTCP MONITOR SERVER PARAMETERS		
Use Default Server Parameters? y		

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

- Verify that next to region **1** in the **dst rgn** column, the codec set is **1**.
- Next to region **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 rgn	codec set	direct WAN	WAN-BW-limits Units	Video Total Norm	Intervening Prio Shr	Regions	Dyn CAC	A R	G L	c e		
1	1								all			
2	2	y	NoLimit					n		t		

6.7.2 IP Network Region 4 – Verizon Trunk Region

Repeat the steps in **Section 6.7.1** 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 rgn	codec set	direct WAN	WAN-BW-limits Units	Video Total Norm	Intervening Prio Shr	Regions	Dyn CAC	A R	G L	c e		
1	2	y	NoLimit					n		t		
2	2								all			
3												

6.8. SIP Trunks

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

- Inbound Verizon IPCC access – SIP Trunk 2
 - Note that this trunk will use TLS port 5071 as described in **Section 5.5.1**.
- Internal CPE access (e.g., Avaya SIP telephones, Messaging, etc.) – SIP Trunk 3
 - Note that this trunk will use TLS port 5061 as described in **Section 5.5.2**.

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 5.4** regarding the use of TLS transport protocols in the CPE.

6.8.1 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 **CM-TG2** SIP Entity defined in **Section 5.4.2**.

6.8.1.1 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 6.4**.
- **Far-end Node Name** – Set to the node name of Session Manager as administered in **Section 6.4** (e.g., **SM**).
- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5071**.
- **Far-end Network Region** – Set the IP network region to **2**, as set in **Section 6.6.2**.
- **Far-end Domain** – Enter **avayalab.com**. This is the domain provisioned for Session Manager in **Section 5.1**.
- **DTMF over IP** – Set to **rtp-payload** to enable Communication Manager to use DTMF according to RFC 2833.
- **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized directly to the associated stations, to reduce the use of media resources on the Avaya Media Gateway when possible (known as shuffling).

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

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

6.8.1.2 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 6.8.1.1** (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	Night Service:	
Dial Access? n			
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	
	UI 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	

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

Trunk 3 corresponds to the CM-TG3 SIP Entity defined in **Section 5.4.3**.

6.8.2.1 Signaling Group 3

Repeat the steps in **Section 6.8.1.1** 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 6.6.1**.

6.8.2.2 Trunk Group 3

Repeat the steps in **Section 6.8.1.2** 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 6.8.2.1** (e.g., **3**).

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

- Same as **Section 6.8.1.2**

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.

6.9. 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 UII 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 UII. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes.

6.9.1 Announcements

Various announcements will be used within the vectors. In the sample configuration, these announcements were sourced by the Avaya G450 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>**.

```
list announcement
```

ANNOUNCEMENTS/AUDIO SOURCES				
Announcement Extension	Type	Name	Source Pt/Bd/Grp	Num of Files
11001	integrated	callcenter-main	001V9	1
11002	integ-mus	holdmusic	001V9	1
11003	integrated	disconnect	001V9	1
11004	integrated	no_agents	001V9	1
11005	integrated	dtmf_test	001V9	1
11006	integrated	please_wait	001V9	1
11007	integrated	REFER_Test	001V9	1

6.9.2 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 10.1.2**. 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	VECTOR DIRECTORY NUMBER	Page 1 of 3
Extension: 10001		
Name*: Refer-to-PSTN		
Destination: Vector Number	1	
Attendant Vectoring? n		
Meet-me Conferencing? n		
Allow VDN Override? n		
COR: 1		
TN*: 1		
Measured: none		

VDN 10001 is associated with vector 1, which is shown below. Vector 1 plays an announcement (step 03) to answer the call. After the announcement, the **route-to number** (step 05) includes **~r+13035387024** where the number 303-538-7024 is a PSTN destination. This step causes a REFER message to be sent where the Refer-To header includes “+13035387024” 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 1	Page 1 of 6
CALL VECTOR	
Number: 1	Name: Refer-to-PSTN
Multimedia? n	Attendant Vectoring? n
Basic? y	EAS? y
Prompting? y	LAI? 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+13035387024 with cov n if unconditionally
06 #	If Refer fails queue to skill 1
07 queue-to	skill 1 pri m
08	

6.9.3 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 UII 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-UII
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

```

6.9.4 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 one-X® 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:	46:
2:	17:	47:
3:	18:	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		

6.10. Public Numbering

In the reference configuration, the public-unknown-numbering form, (used in conjunction with the **Numbering Format: public** setting in **Section 6.8.1.2**), 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					
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len	
5	10001	2	18668523221	11	Total Administered: 16
5	10003	2	18668510107	11	Maximum Entries: 240
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.

6.11. Private Numbering

In the reference configuration, the private-numbering form, (used in conjunction with the **Numbering Format: private** setting in **Section 6.8.2.2**), 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 Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len		
5	10	3		5	Total Administered: 6	
5	11	3		5	Maximum Entries: 540	
5	12	3		5		
5	14	3		5		
5	20	3		5		

6.12. 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 6.13** (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			

6.13. 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 14xxx, therefore enter **14**.
- **Min & Max** - Enter **5**
- **Route Pattern** - Enter **3**
- **Call Type** - Enter **lev0**

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

change aar analysis 0										Page 1 of 2	
AAR DIGIT ANALYSIS TABLE											
Location: all										Percent Full: 1	
Dialed		Total		Route	Call	Node	ANI				
String		Min	Max	Pattern	Type	Num	Reqd				
14		5	5	3	lev0		n				

6.14. Avaya G450 Media Gateway Provisioning

In the reference configuration, a G450 Media Gateway is provisioned. The G450 is located in the Main site and is used for local DSP resources, announcements, Music On Hold, etc.

Note – Only the Media Gateway provisioning associated with the G450 registration to Communication Manager is shown below. For additional information on G450 provisioning, see [7].

Step 1 - Use SSH to connect to the G450 (not shown). Note that the Media Gateway prompt will contain “???” if the Media Gateway is not registered to Communication Manager (e.g., *G450-???(super)#*).

Step 2 - Enter the **show system** command and copy down the G450 serial number (e.g., **08IS38199678**).

Step 3 - Enter the **set mgc list x.x.x.x** command where x.x.x.x is the IP address of the Communication Manager Processor Ethernet (e.g., **10.64.91.65**, see **Section 6.4**).

Step 4 - Enter the **copy run start** command to save the G450 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** = **g450**
- Set **Name** = a descriptive name (e.g., **G450-1**)
- Set **Serial Number** = the serial number copied from **Step 2** (e.g., **08IS38199678**)
- Set the **Link Encryption Type** parameter as desired (**any-ptls/tls** was used in the reference configuration)
- Set **Network Region** = **1**

Wait a few minutes for the G450 to register to Communication Manager. When the Media Gateway registers, the G450 SSH connection prompt will change to reflect the Media Gateway Identifier assigned in **Step 5** (e.g., *G450-001(super)#*).

Step 7 - Enter the **display media-gateway 1** command and verify that the G450 has registered.

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

Type: g450
Name: G450-1
Serial No: 08IS38199678
Link Encryption Type: any-ptls/tls      Enable CF? n
Network Region: 1                      Location: 1
Use for IP Sync? y                     Site Data:
Recovery Rule: 1

Registered? y
FW Version/HW Vintage: 38 .18 .0 /1
MGP IPV4 Address: 10.64.19.61
MGP IPV6 Address:
Controller IP Address: 10.64.91.65
MAC Address: 00:1b:4f:03:52:18

Mutual Authentication? optional
```

6.15. Avaya Aura® Media Server Provisioning

In the reference configuration, an Avaya Aura® Media Server is provisioned. The Media Server is located in the Main site and is used, along with the G450 Media Gateway, for local DSP resources, announcements, and Music On Hold.

Note – Only the Media Server provisioning associated with Communication Manager is shown below. See [8] and [9] for additional information.

Step 1 - Access the Media Server Element Manager web interface by typing

“**https://x.x.x.x:8443**” (where x.x.x.x is the IP address of the Media Server) (not shown).

Step 2 - On the Media Server Element Manager, navigate to **Home → System Configuration → Signaling Protocols → SIP → Node and Routes** and add the Communication Manager Procr interface IP address (e.g., **10.64.91.65**, see **Section 6.4**) as a trusted node (not shown).

Step 3 - On Communication Manager, enter the **add signaling-group x** command where x is an unused signaling group (e.g., **60**), and provision the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tls**
- Verify that **Peer Detection Enabled?** – Set to **n**.
- **Peer Server** to **AMS**.
- **Near-end Node Name** – Set to the node name of the **procr** noted in **Section 6.4**.
- **Far-end Node Name** – Set to the node name of Media Server as administered in **Section 6.4** (e.g., **AMS**).
- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5060**.
- **Far-end Network Region** – Set the IP network region to **1**, as set in **Section 6.6.1**.
- **Far-end Domain** – Automatically populated with the IP address of the Media Server.

```

add signaling-group 60
                                SIGNALING GROUP
                                Page 1 of 2

Group Number: 60                Group Type: sip
                                Transport Method: tls

Peer Detection Enabled? n      Peer Server: AMS

Near-end Node Name: procr       Far-end Node Name: AMS
Near-end Listen Port: 5060      Far-end Listen Port: 5060
                                Far-end Network Region: 1

Far-end Domain: 10.64.91.60

```

Step 4 - On Communication Manager, enter the **add media-server x** command where x is an available Media Server identifier (e.g., 1). Enter the following parameters:

- **Signaling Group** – Enter the signaling group previously configured for Media Server (e.g., 60).
- **Voip Channel License Limit** – Enter the number of VoIP channels for this Media Server (based on licensing) (e.g., 300).
- **Dedicated Voip Channel Licenses** – Enter the number of VoIP channels licensed to this Media Server (e.g., 300)
- Remaining fields are automatically populated based on the signaling group provisioning for the Media Server.

```

add media-server 1
                                MEDIA SERVER
                                Page 1 of 1

Media Server ID: 1

Signaling Group: 60
Voip Channel License Limit: 300
Dedicated Voip Channel Licenses: 300

Node Name: AMS
Network Region: 1
Location: 1
Announcement Storage Area: ANNC-be99ad1a-1f39-41e5-ba04-000c29f8f3f3

```

6.16. Save Translations

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

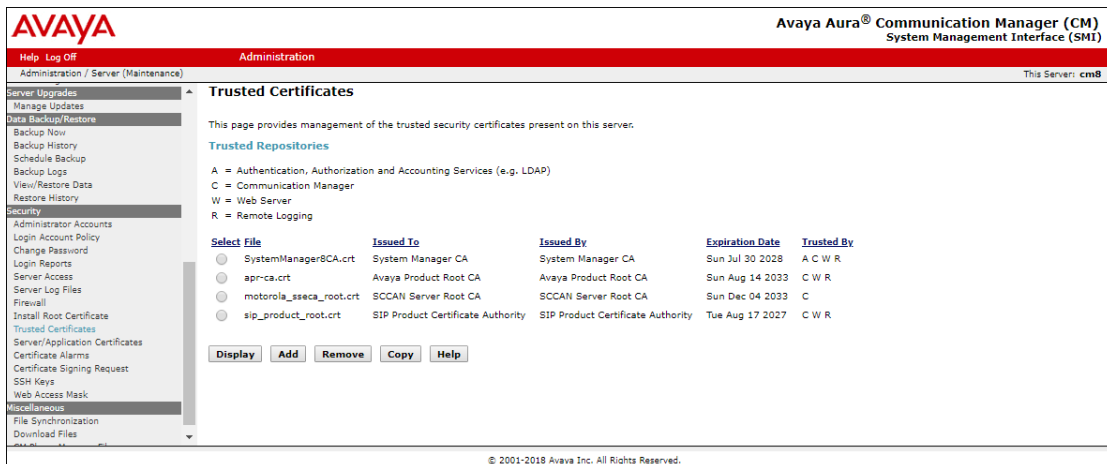
6.17. Verify TLS Certificates – Communication Manager

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.



Avaya Aura® Communication Manager (CM) System Management Interface (SMI)

Help Log Off Administration

Administration / Server (Maintenance) This Server: cm8

Trusted Certificates

This page provides management of the trusted security certificates present on this server.

Trusted Repositories

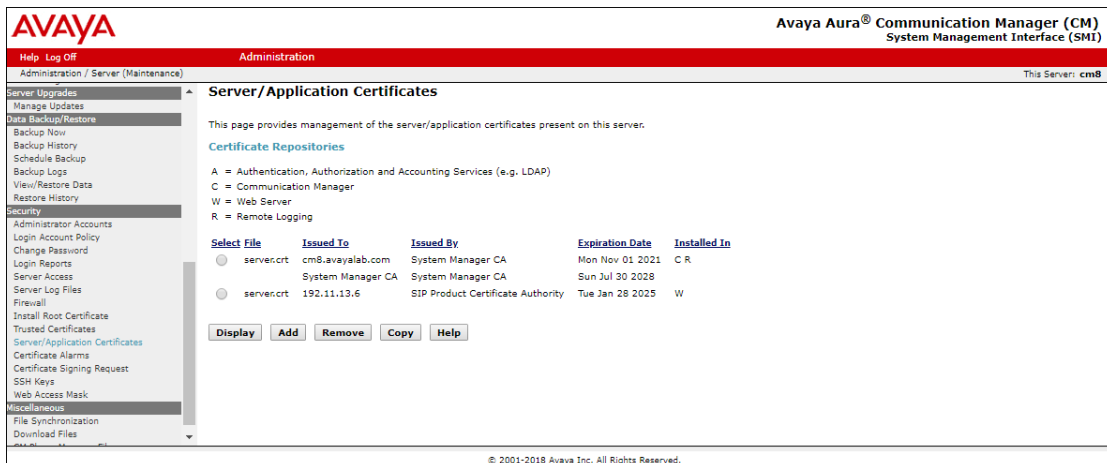
A = Authentication, Authorization and Accounting Services (e.g. LDAP)
C = Communication Manager
W = Web Server
R = Remote Logging

Select File	Issued To	Issued By	Expiration Date	Trusted By
<input type="radio"/> SystemManager8CA.crt	System Manager CA	System Manager CA	Sun Jul 30 2028	A C W R
<input type="radio"/> apr-ca.crt	Avaya Product Root CA	Avaya Product Root CA	Sun Aug 14 2033	C W R
<input type="radio"/> motorola_sscca_root.crt	SCCAN Server Root CA	SCCAN Server Root CA	Sun Dec 04 2033	C
<input type="radio"/> sip_product_root.crt	SIP Product Certificate Authority	SIP Product Certificate Authority	Tue Aug 17 2027	C W R

Display Add Remove Copy Help

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



Avaya Aura® Communication Manager (CM) System Management Interface (SMI)

Help Log Off Administration

Administration / Server (Maintenance) This Server: cm8

Server/Application Certificates

This page provides management of the server/application certificates present on this server.

Certificate Repositories

A = Authentication, Authorization and Accounting Services (e.g. LDAP)
C = Communication Manager
W = Web Server
R = Remote Logging

Select File	Issued To	Issued By	Expiration Date	Installed In
<input type="radio"/> server.crt	cm8.avayalab.com	System Manager CA	Mon Nov 01 2021	C R
<input type="radio"/> server.crt	192.11.13.6	SIP Product Certificate Authority	Tue Jan 28 2025	W

Display Add Remove Copy Help

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7. Avaya Aura ® Experience Portal

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

7.1. Background

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

References to the Voice XML and/or Call Control XML applications are administered on Experience Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Experience Portal, the called party DNIS number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match is found, Experience Portal informs the caller that the call cannot be handled, and disconnects the call¹.

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

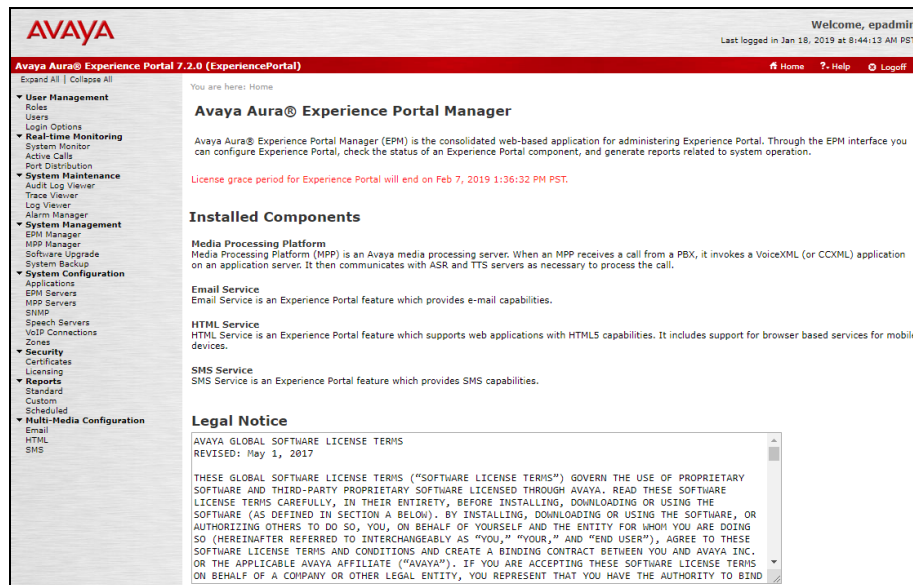
¹ An application may be configured with “inbound default” as the called number, to process all inbound calls that do not match any other application references.

7.2. Logging In and Licensing

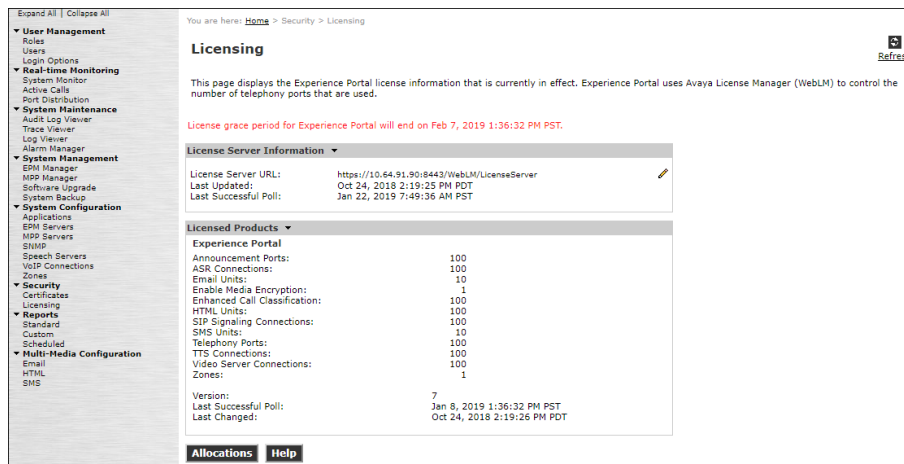
This section describes the steps on Experience Portal for administering a SIP connection to the Session Manager.

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

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



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



7.3. VoIP Connection

This section defines a SIP trunk between Experience Portal and Session Manager (**Section 5.5.5**).

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

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

Name	Enable	Proxy Transport	Proxy/DNS Server Address	Proxy Server Port	Listener Port	SIP Domain	Maximum Simultaneous Calls
SM8	Yes	TLS	10.64.91.81	5061	5061	avayalab.com	10

Step 2 - Configure a SIP connection as follows:

- **Name** – Set to a descriptive name (e.g., **SM8**).
- **Enable** – Set to **Yes**.
- **Proxy Server Transport** – Set to **TLS**.
- Select **Proxy Servers**, and enter:
 - **Proxy Server Address** = **10.64.91.81** (the IP address of the Session Manager signaling interface defined in **Section 5.4.1**).
 - **Port** = **5061**
 - **Priority** = **0** (default)
 - **Weight** = **0** (default)
- **Listener Port** – Set to **5061**.
- **SIP Domain** – Set to **avayalab.com** (see **Section 5.1**).
- **Consultative Transfer** – Select **INVITE with REPLACES**.
- **SIP Reject Response Code** – Select **ASM (503)**.
- **Maximum Simultaneous Calls** – Set to a number in accordance with licensed capacity. In the reference configuration a value of **10** was used.
- Select **All Calls can be either inbound or outbound**.
- **SRTP Enable** = **Yes**
- **Encryption Algorithm** = **AES_CM_128**
- **Authentication Algorithm** = **HMAC_SHA1_80**
- **RTCP Encryption Enabled** = **No**
- **RTP Authentication Enabled** = **Yes**
- Use default values for all other fields.
- Click **Save**.

Expand All | Collapse All

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

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

Change SIP Connection

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

Name: SM8

Enable: ☒ Yes ☐ No

Proxy Transport: **TLS**

☒ Proxy Servers ☐ DNS SRV Domain

Address	Port	Priority	Weight	
10.64.91.81	5061	0	0	Remove

[Additional Proxy Server](#)

Listener Port: 5061

SIP Domain: avayalab.com

P-Asserted-Identity:

Maximum Redirection Attempts: 2

Consultative Transfer: ☒ INVITE with REPLACES ☐ REFER

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

SIP Timers

T1: 250 milliseconds

T2: 2000 milliseconds

B and F: 4000 milliseconds

Call Capacity

Maximum Simultaneous Calls: 10

☒ All Calls can be either inbound or outbound

☐ Configure number of inbound and outbound calls allowed

SRTP

Enable: ☒ Yes ☐ No

Encryption Algorithm: ☒ AES_CM_128 ☐ NONE

Authentication Algorithm: ☒ HMAC_SHA1_80 ☐ HMAC_SHA1_32

RTCP Encryption Enabled: ☐ Yes ☒ No

RTP Authentication Enabled: ☒ Yes ☐ No

Add

Configured SRTP List

<No SRTP List>

7.4. Speech Servers

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

Expand All | Collapse All

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

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

Speech Servers

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

ASR **TTS**

	Name	Enable	Network Address	Engine Type	MRCP	Base Port	Total Number of Licensed ASR Resources	Languages
<input type="checkbox"/>	LVASR	Yes	10.64.101.83	LumenVox	MRCP V2 TCP	5060	10	en-US

Add **Delete** **Customize** **Help**

7.5. Application References

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

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

- **Name** – Set to a descriptive name (e.g., **Test-ccxml**).
- **Enable** – Set to **Yes**. This field determines which application(s) will be executed based on their defined criteria.
- **Type** – Select **VoiceXML**, **CCXML**, or **CCXML/VoiceXML** according to the application type.
- **VoiceXML** and/or **CCXML URL** – Enter the necessary URL(s) to access the VXML and/or CCXML application(s) on the application server. In the sample screen below, the Experience Portal test application on a single server is referenced.
- **Speech Servers ASR** and **TTS** – Select the appropriate ASR and/or TTS servers as necessary.
- **Application Launch** – Set to **Inbound**.
- **Called Number** – Enter the number to match against an inbound SIP INVITE message and click **Add**. In the sample configuration illustrated in these Application Notes, the dialed Verizon IPCC toll-free number 866-851-2649 was used. Repeat to define additional called party numbers as needed. Inbound Verizon Business calls with these called party numbers will be handled by the application defined in this section.

Change Application

Use this page to change the configuration of an application.

Name: Test-ccxml

Enable: ☒ Yes ☐ No

Type: CCXML

Reserved SIP Calls: ☒ None ☐ Minimum ☐ Maximum

Requested:

URI

☒ Single ☐ Fail Over ☐ Load Balance

CCXML URL: **Verify**

Mutual Certificate Authentication: ☐ Yes ☒ No

Basic Authentication: ☐ Yes ☒ No

Speech Servers

ASR: LumenVox

Languages: <None>

Selected Languages: en-US

TTS: LumenVox

Voices: <None>

Selected Voices: en-US Chris M

Application Launch

☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number:

Add

55556
7329450232
8668512649

Remove

7.6. MPP Servers and VoIP Settings

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

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

Expand All | Collapse All

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

MPP Servers

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

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

Add **Delete**

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

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

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

Expand All | Collapse All

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

Change MPP Server

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

Name: mpp1

Host Address: 10.64.91.90

Network Address (VoIP): <Default>

Network Address (MRCP): <Default>

Network Address (AppSvr): <Default>

Maximum Simultaneous Calls: 11

Restart Automatically: ☒ Yes ☐ No

MPP Certificate

Owner: CN=ep.avaya1ab.com,O=Avaya,OU=EPM
Issuer: CN=ep.avaya1ab.com,O=Avaya,OU=EPM
Serial Number: 89f44cd176674542
Signature Algorithm: SHA256withRSA
Valid from: October 17, 2018 11:03:28 AM PDT until October 14, 2028 11:03:28 AM PDT
Certificate Fingerprints
MD5: dd:26:1a:d3:d1:62:d3:04:55:40:1b:98:0b:38:44:46
SHA: 4d:26:ba:2f:55:8d:3b:5f:8e:d8:6f:ee:7f:48:49:22:38:79:ae:bf
SHA-256: 17:6d:d2:9a:9b:ee:e3:35:da:67:c2:99:38:e6:14:03:c7:84:1d:94:a9:a0:f9:ac:66:57:da:28:43:59:ae:c7
Subject Alternative Names
DNS Name: ep
DNS Name: ep.avaya1ab.com
IP Address: 10.64.91.90

Categories and Trace Levels

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

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

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

Expand All | Collapse All

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

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

VoIP Settings

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

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

RTCP Monitor Settings

Host Address:

Port:

VoIP Audio Formats

MPP Native Format:

- In the Codecs section set:
 - Set **Packet Time** to **20**.
 - Verify the **G729 Codec** is enabled.
 - Set **G729 Discontinuous Transmission** to **No** (G.729A).
 - Set the **Offer Order** to the preferred codec. In the sample configuration, **G729** is the first codec, followed by **G711uLaw**, then **G711aLaw**.
- Use default values for all other fields.

Step 5 - Click on **Save**.

Expand All | Collapse All

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

Station:

RTCP Monitor Settings

Host Address:

Port:

VoIP Audio Formats

MPP Native Format:

Codecs

Offer

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

Packet Time: milliseconds

G729 Discontinuous Transmission: ☐ Yes ☒ No

Answer

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

G729 Discontinuous Transmission: ☐ Yes ☐ No ☒ Either

G729 Reduced Complexity Encoder: ☒ Yes ☐ No

QoS Parameters

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

7.7. Configuring RFC2833 Event Value Offered by Experience Portal

The configuration change example noted in this section is not required for any of the call flows illustrated in these Application Notes. For incoming calls from Verizon services to Experience Portal, Verizon specifies the value 101 for the RFC2833 telephone-events that signal DTMF digits entered by the user. When Experience Portal answers, the SDP from Experience Portal matches this Verizon offered value.

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

- Access Experience Portal via the command line interface.
- Navigate to the following directory: /opt/Avaya/ ExperiencePortal /MPP/config
- Edit the file mppconfig.xml.
- Search for the parameter “mpp.sip.rfc2833.payload”. If there is no such parameter specified, add a line such as the following to the file, where the value 101 is the value to be used for the RFC2833 events. If the parameter is already specified in the file, simply edit the value assigned to the parameter.
`<parameter name="mpp.sip.rfc2833.payload">101</parameter>`
- In the verification of these Application Notes, the line was added directly above the line where the sip.session.expires parameter is configured.

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

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

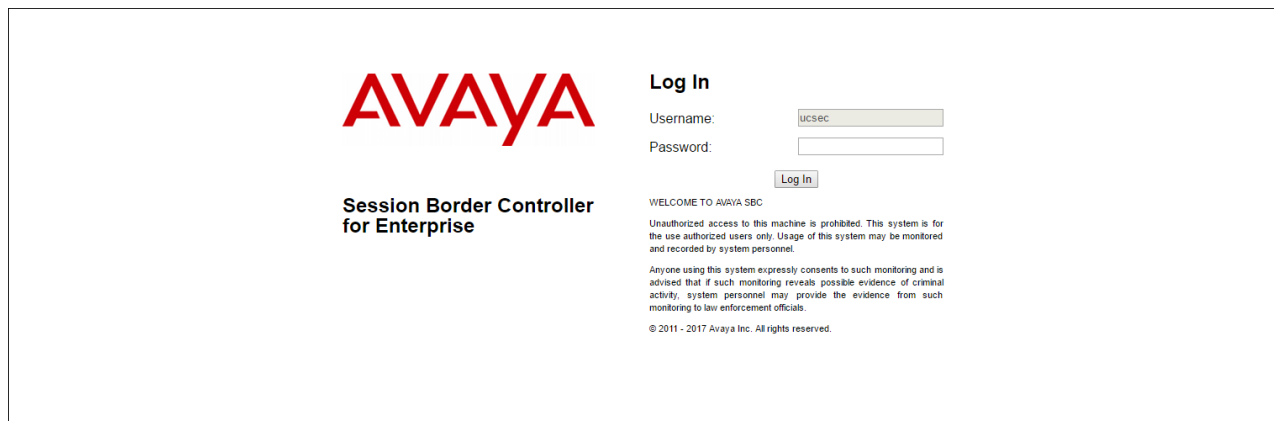
The screenshot shows the MPP Manager GUI. The left sidebar contains a navigation menu with options like User Management, Real-time Monitoring, System Maintenance, System Management, EPM Manager, MPP Manager, Software Upgrade, System Backup, System Configuration, Security, Reports, and Multi-Media Configuration. The main content area is titled 'MPP Manager (Jan 22, 2019 9:07:05 AM PST)' and includes a 'Refresh' button. Below the title, there is a table with columns: Server Name, Mode, State, Config, Auto Restart, Restart Schedule (Today, Recurring), and Active Calls (In, Out). The 'State' column is highlighted with a red box. The table shows one entry for 'mpp1' with 'Online' mode and 'Running' state. Below the table, there are 'State Commands' (Start, Stop, Restart, Reboot, Halt, Cancel) and 'Mode Commands' (Offline, Test, Online). A 'Restart/Reboot Options' section on the right allows selecting 'One server at a time' or 'All servers'.

Server Name	Mode	State	Config	Auto Restart	Restart Schedule	Active Calls		
					Today	Recurring	In	Out
<input checked="" type="checkbox"/> mpp1	Online	Running	OK	Yes	No	None	0	0

8. Configure Avaya Session Border Controller for Enterprise Release 7.2

These Application Notes assume that the installation of the Avaya SBCE and the assignment of all IP addresses have already been completed, including the management IP address. Consult [10] and [11] for additional information.

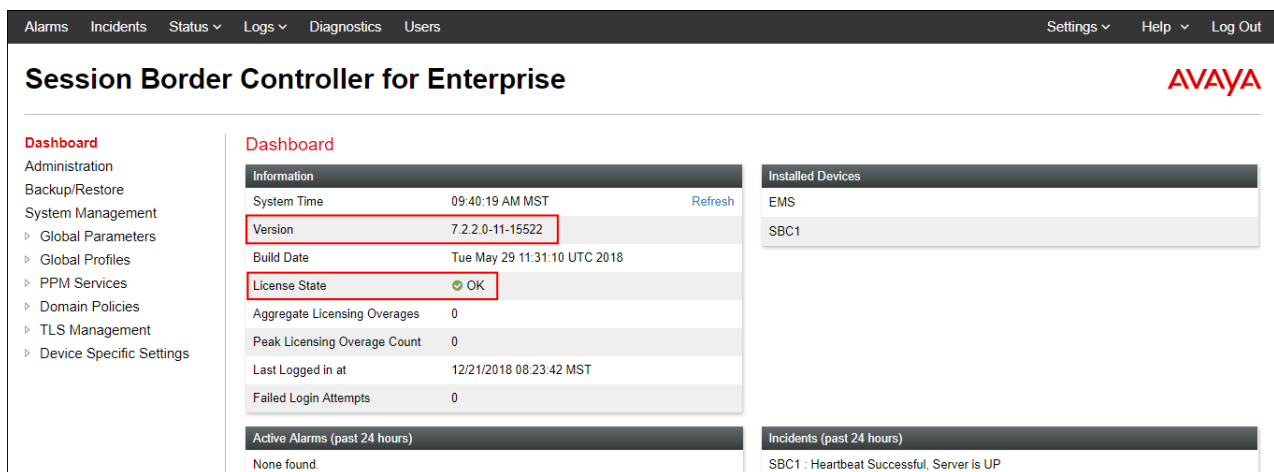
In the sample configuration, the management IP is 10.64.90.50. Access the web management interface by entering `https://<ip-address>` where `<ip-address>` is the management IP address assigned during installation. Log in with the appropriate credentials. Click **Log In**.



The login page for the Avaya Session Border Controller for Enterprise. It features the Avaya logo and the product name. On the right, there is a 'Log In' section with fields for 'Username' (containing 'ucsec') and 'Password', and a 'Log In' button. Below the login fields, there is a 'WELCOME TO AVAYA SBC' message and a disclaimer: 'Unauthorized access to this machine is prohibited. This system is for the use authorized users only. Usage of this system may be monitored and recorded by system personnel.' A small text block below the disclaimer states: 'Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible evidence of criminal activity, system personnel may provide the evidence from such monitoring to law enforcement officials.' At the bottom, it says '© 2011 - 2017 Avaya Inc. All rights reserved.'

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

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

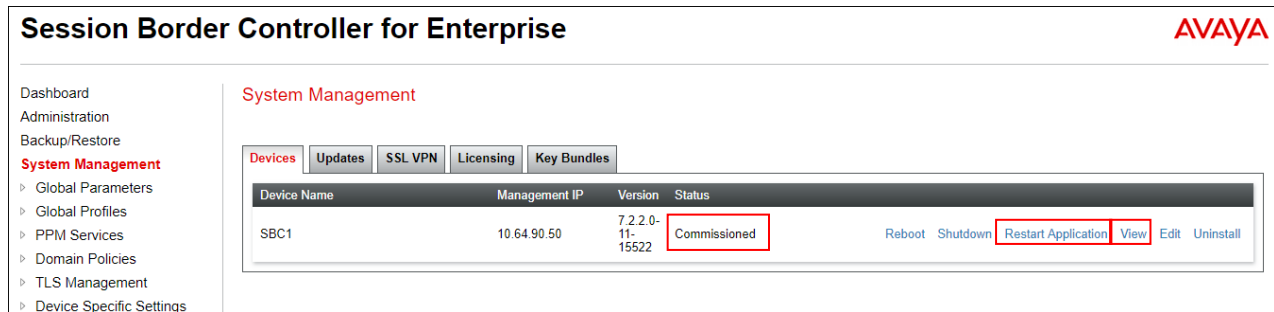


The dashboard of the Avaya Session Border Controller for Enterprise. It has a top navigation bar with 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', and 'Users'. On the right of the top bar are 'Settings', 'Help', and 'Log Out'. The main header shows 'Session Border Controller for Enterprise' and the Avaya logo. A left-hand menu lists various options: 'Dashboard', 'Administration', 'Backup/Restore', 'System Management', 'Global Parameters', 'Global Profiles', 'PPM Services', 'Domain Policies', 'TLS Management', and 'Device Specific Settings'. The main content area is titled 'Dashboard' and contains several sections: 'Information' (with fields for System Time, Version, Build Date, License State, Aggregate Licensing Overages, Peak Licensing Overage Count, Last Logged in at, and Failed Login Attempts), 'Installed Devices' (listing EMS and SBC1), 'Active Alarms (past 24 hours)' (showing 'None found'), and 'Incidents (past 24 hours)' (showing 'SBC1 : Heartbeat Successful, Server is UP').

8.1. System Management – Status

Step 1 - Select **System Management** and verify that the **Status** column says **Commissioned**. If not, contact your Avaya representative.

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



Session Border Controller for Enterprise AVAYA

System Management

Devices Updates SSL VPN Licensing Key Bundles

Device Name	Management IP	Version	Status						
SBC1	10.64.90.50	7.2.2.0-11-15522	Commissioned	Reboot	Shutdown	Restart Application	View	Edit	Uninstall

Step 2 - Click on **View** (shown above) to display the **System Information** screen. The following shows the relevant IP information highlighted 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 these interfaces and interface **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.

System Information: SBC1																																							
General Configuration Appliance Name SBC1 Box Type SIP Deployment Mode Proxy		Device Configuration HA Mode No Two Bypass Mode No		Dynamic License Allocation <table border="1"> <thead> <tr> <th></th> <th>Min License Allocation</th> <th>Max License Allocation</th> </tr> </thead> <tbody> <tr> <td>Standard Sessions</td> <td>10</td> <td>500</td> </tr> <tr> <td>Advanced Sessions</td> <td>10</td> <td>500</td> </tr> <tr> <td>Scopia Video Sessions</td> <td>10</td> <td>500</td> </tr> <tr> <td>CES Sessions</td> <td>10</td> <td>500</td> </tr> <tr> <td>Transcoding Sessions</td> <td>10</td> <td>500</td> </tr> <tr> <td>CLID</td> <td colspan="2">---</td> </tr> <tr> <td>Encryption Available: Yes</td> <td colspan="2"><input checked="" type="checkbox"/></td> </tr> </tbody> </table>		Min License Allocation	Max License Allocation	Standard Sessions	10	500	Advanced Sessions	10	500	Scopia Video Sessions	10	500	CES Sessions	10	500	Transcoding Sessions	10	500	CLID	---		Encryption Available: Yes	<input checked="" type="checkbox"/>												
	Min License Allocation	Max License Allocation																																					
Standard Sessions	10	500																																					
Advanced Sessions	10	500																																					
Scopia Video Sessions	10	500																																					
CES Sessions	10	500																																					
Transcoding Sessions	10	500																																					
CLID	---																																						
Encryption Available: Yes	<input checked="" type="checkbox"/>																																						
Network Configuration <table border="1"> <thead> <tr> <th>IP</th> <th>Public IP</th> <th>Network Prefix or Subnet Mask</th> <th>Gateway</th> <th>Interface</th> </tr> </thead> <tbody> <tr> <td>1.1.1.2</td> <td>1.1.1.2</td> <td>255.255.255.0</td> <td>1.1.1.1</td> <td>B1</td> </tr> <tr> <td>10.64.91.48</td> <td>10.64.91.48</td> <td>255.255.255.0</td> <td>10.64.91.1</td> <td>A1</td> </tr> <tr> <td>10.64.91.49</td> <td>10.64.91.49</td> <td>255.255.255.0</td> <td>10.64.91.1</td> <td>A1</td> </tr> <tr> <td>10.64.91.50</td> <td>10.64.91.50</td> <td>255.255.255.0</td> <td>10.64.91.1</td> <td>A1</td> </tr> <tr> <td>192.168.80.44</td> <td>192.168.80.44</td> <td>255.255.255.128</td> <td>192.168.80.1</td> <td>B2</td> </tr> <tr> <td>192.168.80.92</td> <td>192.168.80.92</td> <td>255.255.255.128</td> <td>192.168.80.1</td> <td>B2</td> </tr> </tbody> </table>					IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface	1.1.1.2	1.1.1.2	255.255.255.0	1.1.1.1	B1	10.64.91.48	10.64.91.48	255.255.255.0	10.64.91.1	A1	10.64.91.49	10.64.91.49	255.255.255.0	10.64.91.1	A1	10.64.91.50	10.64.91.50	255.255.255.0	10.64.91.1	A1	192.168.80.44	192.168.80.44	255.255.255.128	192.168.80.1	B2	192.168.80.92	192.168.80.92	255.255.255.128	192.168.80.1	B2
IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface																																			
1.1.1.2	1.1.1.2	255.255.255.0	1.1.1.1	B1																																			
10.64.91.48	10.64.91.48	255.255.255.0	10.64.91.1	A1																																			
10.64.91.49	10.64.91.49	255.255.255.0	10.64.91.1	A1																																			
10.64.91.50	10.64.91.50	255.255.255.0	10.64.91.1	A1																																			
192.168.80.44	192.168.80.44	255.255.255.128	192.168.80.1	B2																																			
192.168.80.92	192.168.80.92	255.255.255.128	192.168.80.1	B2																																			
DNS Configuration Primary DNS 172.30.209.4 Secondary DNS DNS Location DMZ DNS Client IP 1.1.1.2		Management IP(s) IP #1 (IPv4) 10.64.90.50																																					

8.2. TLS Management

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

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

8.2.1 Verify TLS Certificates – Avaya Session Border Controller for Enterprise

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.

- Dashboard
- Administration
- Backup/Restore
- System Management
 - Global Parameters
 - Global Profiles
 - PPM Services
 - Domain Policies
 - TLS Management
 - Certificates**
 - Client Profiles
 - Server Profiles
 - Device Specific Settings

Certificates

[Install](#) [Generate CSR](#)

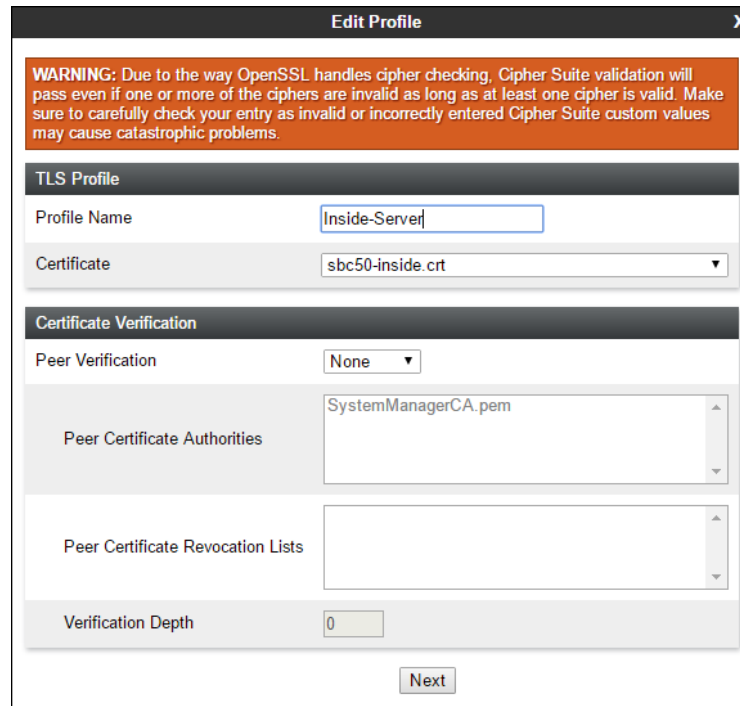
Certificates	
Installed Certificates	
sbcb50-inside.crt	View Delete
sbcb50-outside.crt	View Delete
sbcb92-out.crt	View Delete
sbcb92-outside.crt	View Delete
Installed CA Certificates	
SystemManagerCA.pem	View Delete
Installed Certificate Revocation Lists	
No certificate revocation lists have been installed.	
Installed Keys	
avaya.com.key	Delete
sbcb50-inside.key	Delete
sbcb50-outside.key	Delete
sbcb92-out.key	Delete
sbcb92-outside.key	Delete

8.2.2 Server Profiles

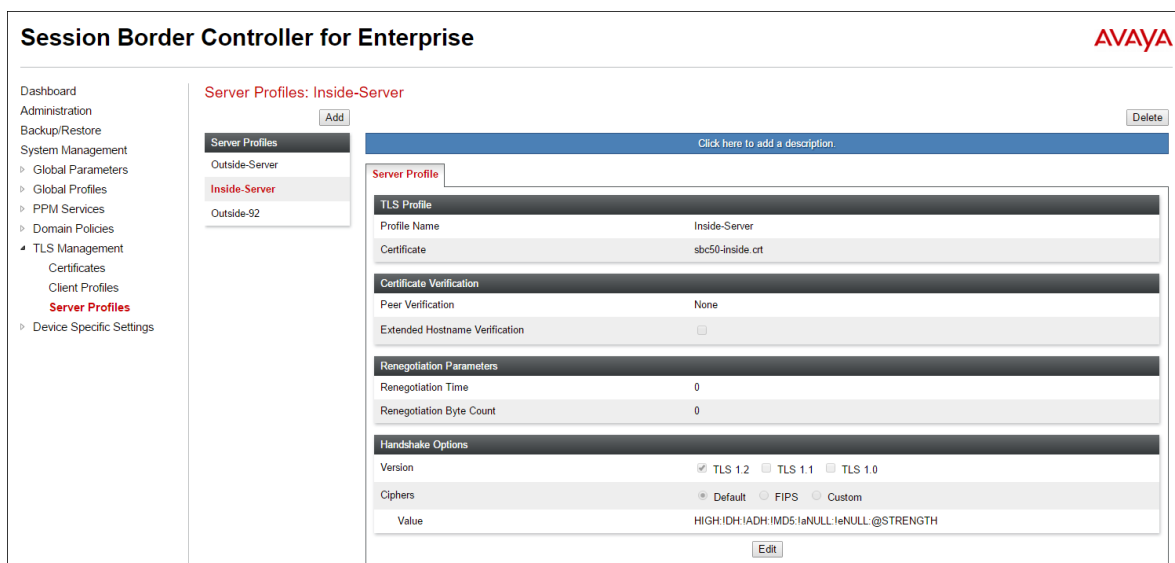
Step 1 - Select **TLS Management** → **Server Profiles**, and click on **Add**. Enter the following:

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

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



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



8.2.3 Client Profiles

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

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

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

The screenshot shows a window titled "Edit Profile" with a close button (X) in the top right corner. At the top, there is a red warning box with the following text: "WARNING: Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems." Below the warning, the "TLS Profile" section contains a "Profile Name" text field with "Inside-Client" entered and a "Certificate" dropdown menu showing "sbc50-inside.crt". The "Certificate Verification" section includes a "Peer Verification" label with the value "Required", a "Peer Certificate Authorities" dropdown menu showing "SystemManagerCA.pem", and a "Peer Certificate Revocation Lists" dropdown menu. Below these, the "Verification Depth" is set to "1" in a text field. There are checkboxes for "Extended Hostname Verification" (unchecked) and "Custom Hostname Override" (unchecked). At the bottom right, there is a "Next" button.

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

The screenshot shows the 'Session Border Controller for Enterprise' web interface. On the left is a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, PPM Services, Domain Policies, TLS Management, Certificates, Client Profiles (highlighted), Server Profiles, and Device Specific Settings. The main area is titled 'Client Profiles: Inside-Client'. It features a list of profiles with 'Inside-Client' selected. The 'Client Profile' form is displayed, showing the following configuration:

- TLS Profile:** Profile Name: Inside-Client, Certificate: sbc50-inside.crt
- Certificate Verification:** Peer Verification: Required, Peer Certificate Authorities: SystemManagerCA.pem, Peer Certificate Revocation Lists: ---, Verification Depth: 1, Extended Hostname Verification: ☐
- Renegotiation Parameters:** Renegotiation Time: 0, Renegotiation Byte Count: 0
- Handshake Options:** Version: ☒ TLS 1.2, ☐ TLS 1.1, ☐ TLS 1.0; Ciphers: ☒ Default, ☐ FIPS, ☐ Custom; Value: HIGH:DH:1ADH:1MD5:1aNULL:1eNULL:@STRENGTH

8.3. Global Profiles

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

8.3.1 Server Interworking – Avaya

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

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

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

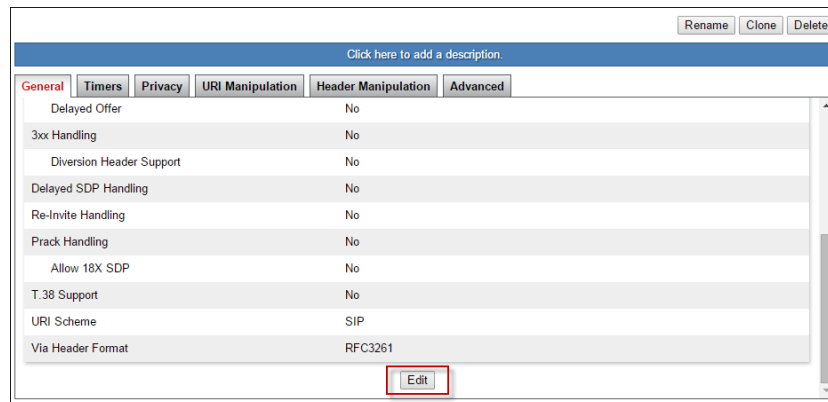
The screenshot shows the 'Interworking Profiles: avaya-ru' section. The left-hand menu has 'Global Profiles' expanded, and 'Server Interworking' is selected. The 'Interworking Profiles' list shows 'cs2100', 'avaya-ru' (highlighted), 'OCS-Edge-Server', 'disco-ccm', and 'cups'. The 'Clone' button is highlighted in the top right corner. The main area displays a warning: 'It is not recommended to edit the defaults. Try cloning or adding a new profile instead.' Below this are tabs for 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'General' tab is active, showing the following configuration:

- Hold Support:** NONE
- 180 Handling:** None
- 181 Handling:** None

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

The screenshot shows the 'Clone Profile' dialog box. It has a title bar with 'Clone Profile' and a close button. The dialog contains two input fields: 'Profile Name' with the value 'avaya-ru' and 'Clone Name' with the value 'Enterprise Interwork'. A 'Finish' button is located at the bottom right.

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



Click here to add a description.

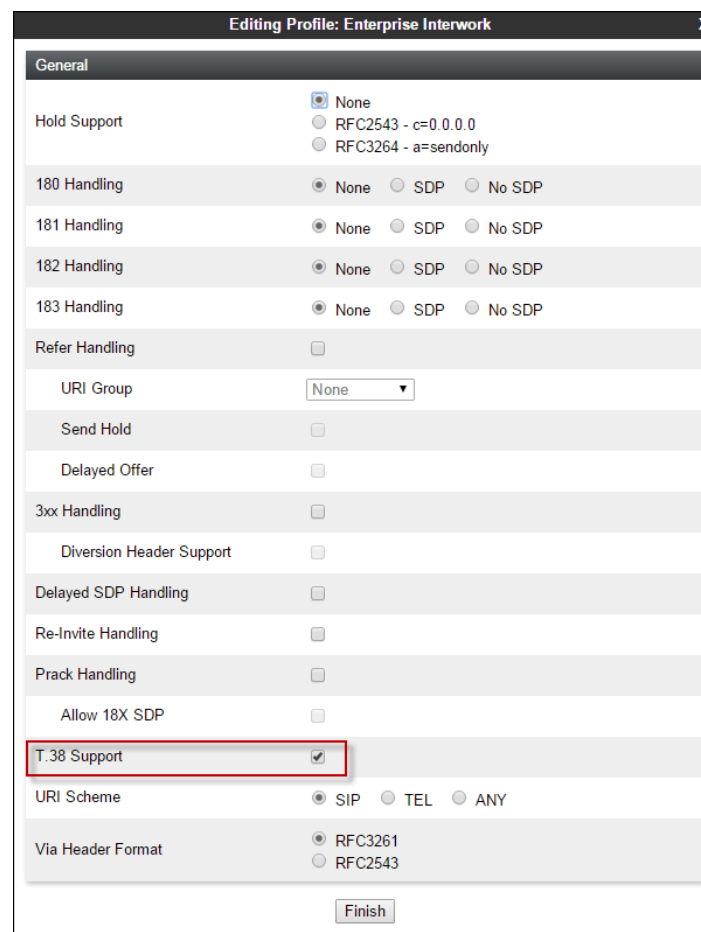
General Timers Privacy URI Manipulation Header Manipulation Advanced

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

Edit

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

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



Editing Profile: Enterprise Interwork

General

Hold Support ☒ None ☐ RFC2543 - c=0.0.0.0 ☐ RFC3264 - a=sendonly

180 Handling ☒ None ☐ SDP ☐ No SDP

181 Handling ☒ None ☐ SDP ☐ No SDP

182 Handling ☒ None ☐ SDP ☐ No SDP

183 Handling ☒ None ☐ SDP ☐ No SDP

Refer Handling ☐

URI Group

Send Hold ☐

Delayed Offer ☐

3xx Handling ☐

Diversion Header Support ☐

Delayed SDP Handling ☐

Re-Invite Handling ☐

Prack Handling ☐

Allow 18X SDP ☐

T.38 Support ☒

URI Scheme ☒ SIP ☐ TEL ☐ ANY

Via Header Format ☒ RFC3261 ☐ RFC2543

Finish

Step 6 - Returning to the Interworking Profile screen, select the **Advanced** tab, accept the default values, and click **Finish**.

Editing Profile: Enterprise Interwork

Record Routes

- ☒ None
- ☐ Single Side
- ☒ Both Sides
- ☐ Dialog-Initiate Only (Single Side)
- ☐ Dialog-Initiate Only (Both Sides)

Include End Point IP for Context Lookup ☒

Extensions Avaya ▼

Diversion Manipulation ☐

Diversion Condition None ▼

Diversion Header URI

Has Remote SBC ☒

Route Response on Via Port ☐

Relay INVITE Replace for SIPREC ☐

MOBX Re-INVITE Handling ☐

DTMF

DTMF Support

- ☒ None
- ☐ SIP Notify
- ☐ RFC 2833 Relay & SIP Notify
- ☐ SIP Info
- ☐ RFC 2833 Relay & SIP Info
- ☐ Inband

Finish

8.3.2 Server Interworking – Verizon

Repeat the steps shown in **Section 8.3.1** to add an Interworking Profile for the connection to Verizon via the public network, with the following changes:

Note – See **Section 13** for additional steps necessary for Experience Portal to redirect calls to Communication Manager using SIP REFER.

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

Editing Profile: SIP Provider Interwk X

Record Routes: ☐ None, ☐ Single Side, ☒ Both Sides, ☐ Dialog-Initiate Only (Single Side), ☐ Dialog-Initiate Only (Both Sides)

Include End Point IP for Context Lookup: ☐

Extensions: None ▼

Diversion Manipulation: ☐

Diversion Condition: None ▼

Diversion Header URI:

Has Remote SBC: ☒

Route Response on Via Port: ☐

Relay INVITE Replace for SIPREC: ☐

MOBX Re-INVITE Handling: ☐

DTMF

DTMF Support: ☒ None, ☐ SIP Notify, ☐ RFC 2833 Relay & SIP Notify, ☐ SIP Info, ☐ RFC 2833 Relay & SIP Info, ☐ Inband

Finish

8.3.3 Signaling Manipulation

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

Note – Use of the Signaling Manipulation scripts require higher processing requirements on the Avaya SBCE. Therefore, this method of header manipulation should only be used in cases where the use of Signaling Rules or Interworking Profiles does not meet the desired result. Refer to [10] for information on the Avaya SBCE scripting language.

Step 1 - Select **Global Profiles → Signaling Manipulation** from the left-hand menu (not shown).

Step 2 - Select **Add**.

Step 3 - Enter a name for the script in the **Title** box (e.g., **remove Contact parameter**). 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 is applied to the Verizon Server Configuration in **Section 8.3.5, Step 3**.

The screenshot shows a web-based interface for managing signaling manipulation scripts. At the top, the title is "Signaling Manipulation Scripts: remove Contact parameters". Below the title, there are buttons for "Upload", "Add", "Download", "Clone", and "Delete". A sidebar on the left contains a "Signaling Manipulation Scripts" menu item, a "remove Contact para..." script entry, and a "test script" button. The main content area has a blue header with the text "Click here to add a description." Below this, there is a tab labeled "Signaling Manipulation" and a text area containing the same script code as shown in the previous block. An "Edit" button is located at the bottom right of the script text area.

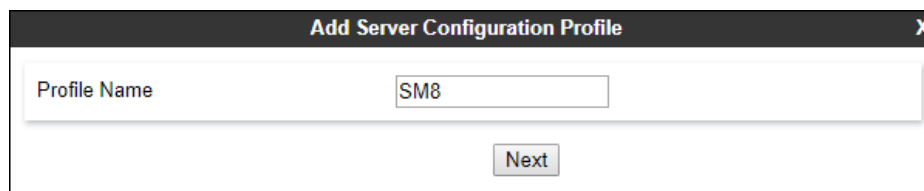
Note – These parameters contain unnecessary information for Verizon, including the internal domain. Removing these parameters helps to mask the internal topology of the enterprise and reduces the size of the SIP packet sent to Verizon. The Endpoint-View header and other proprietary headers are removed using an adaptation as illustrated in **Section 5.3**.

8.3.4 Server Configuration – Session Manager

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

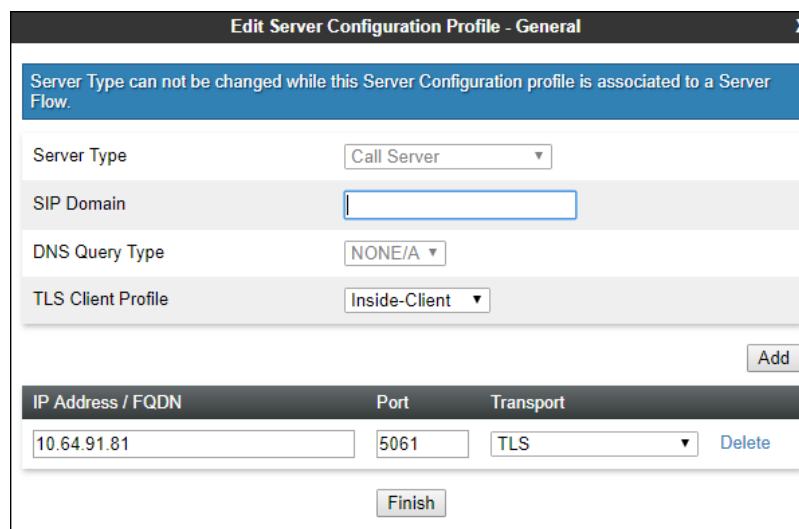
Step 1 - Select **Global Profiles → Server Configuration** from the left-hand menu.

Step 2 - Select **Add Profile** and the **Profile Name** window will open. Enter a Profile Name (e.g., SM8) and click **Next**.



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

- Select **Server Type: Call Server**
- **SIP Domain:** Leave blank (default)
- **DNS Query Type:** Select **NONE/A** (default)
- **TLS Client Profile:** Select the profile create in **Section 8.2.3** (e.g., **Inside-Client**)
- **IP Address:** **10.64.91.81** (Session Manager network IP address)
- **Transport:** Select **TLS**
- **Port:** **5061**
- Select **Next**



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

- Select **Next** to accept default values

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

- Select **Enterprise Interwork** (created in **Section 8.3.1**), for **Interworking Profile**
- Check **Enable Grooming**
- In the **Signaling Manipulation Script** field select **none**

- Select **Finish**

Note – Since TLS transport is specified in **Step 3**, then the **Enable Grooming** option should be enabled.

8.3.5 Server Configuration – Verizon

Repeat the steps in **Section 8.3.4**, with the following changes, to create a Server Configuration 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)
- **Transport:** Select **UDP**
- **Port:** **5072**
- Select **Next** until the Advanced tab is reached

Edit Server Configuration Profile - GeneralX

Server Type can not be changed while this Server Configuration profile is associated to a Server Flow.

Server TypeTrunk Server

SIP Domain

TLS Client ProfileNone

Add

IP Address / FQDN	Port	Transport	
172.30.205.55	5072	UDP	Delete

Finish

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

- Select **SIP Provider Interwk** (created in **Section 8.3.2**), for **Interworking Profile**.
- Select **remove Contact parameter** (created in **Section 8.3.3**) for **Signaling Manipulation Script**.
- Select **Finish** (not shown).

The screenshot shows a configuration window with five tabs: General, Authentication, Heartbeat, Ping, and Advanced. The Advanced tab is selected. It contains a list of settings with checkboxes and dropdown menus. The settings are: Enable DoS Protection (checkbox), Enable Grooming (checkbox), Interworking Profile (dropdown set to SIP Provider Interwk), Signaling Manipulation Script (dropdown set to remove Contact parameter), Securable (checkbox), Enable FGDN (checkbox), Tolerant (checkbox), and URI Group (dropdown set to None). An Edit button is located at the bottom right of the settings list.

Setting	Value
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SIP Provider Interwk
Signaling Manipulation Script	remove Contact parameter
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
Tolerant	<input type="checkbox"/>
URI Group	None

8.3.6 Routing – To Session Manager

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

Step 1 - Select **Global Profiles → Routing** from the left-hand menu, and select **Add** (not shown)

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

The screenshot shows a window titled "Routing Profile" with a close button (X) in the top right corner. It contains a text input field labeled "Profile Name" with the value "route to SM8" entered. Below the input field is a "Next" button.

Step 3 - The Routing Profile window will open. Using the default values shown, click on **Add**.

Step 4 - The **Next-Hop Address** window will open. Populate the following fields:

- **Priority/Weight = 1**
- **Server Configuration = SM8** (from **Section 8.3.4**).
- **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**.

Profile : route to SM8 - Edit Rule

URI Group: * Time of Day: default

Load Balancing: Priority NAPTR: ☐

Transport: None Next Hop Priority: ☒

Next Hop In-Dialog: ☐ Ignore Route Header: ☐

ENUM: ☐ ENUM Suffix:

Add

Priority / Weight	Server Configuration	Next Hop Address	Transport	
1	SM8	10.64.91.81:5061 (TLS)	None	Delete

Finish

8.3.7 Routing – To Verizon

Repeat the steps in **Section 8.3.6**, 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 8.3.5**).
- **Next Hop Address:** select **172.30.205.55:5072 (UDP)**.

Step 3 - Click **Finish**.

Profile : route to Vz IPCC - Edit Rule

URI Group: * Time of Day: default

Load Balancing: Priority NAPTR: ☐

Transport: None Next Hop Priority: ☒

Next Hop In-Dialog: ☐ Ignore Route Header: ☐

ENUM: ☐ ENUM Suffix:

Add

Priority / Weight	Server Configuration	Next Hop Address	Transport	
1	Verizon IPCC	172.30.205.55:5072 (UDP)	None	Delete

Finish

8.3.8 Topology Hiding – Enterprise Side

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

Step 1 - Select **Global Profiles → Topology Hiding** from the left-hand side menu.

Step 2 - Select the **Add** button, enter Profile Name: (e.g., **Enterprise-Topology**), and click **Next**.

Topology Hiding Profile

Profile Name: Enterprise-Topology

Next

Step 3 - The **Topology Hiding Profile** window will open. Click on the **Add Header** button repeatedly until no new headers are added to the list, and the **Add Header** button is no longer displayed.

Topology Hiding Profile

Add Header

Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Auto	

Back Finish

Topology Hiding Profile

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

Back Finish

Step 4 - Populate the fields as shown below, and click **Finish**. Note that **avayalab.com** is the domain used by the CPE (see **Sections 5.1, 6.7, and 6.8**).

Edit Topology Hiding Profile

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

Finish

8.3.9 Topology Hiding – Verizon Side

Repeat the steps in **Section 8.3.8**, 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 th profile**).
- 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. If the IP-IVR service is not used, the Refer-To header can retain the default **Replace Action** of “Auto”.

Topology Hiding Profiles: Vz IPCC th profile

Add Rename Clone Delete

Topology Hiding Profiles

- default
- disco_th_profile
- Vz th profile
- Enterprise-Topology
- Vz IPCC th profile**
- IP500v2-Topology
- IPOSE-Topology

Click here to add a description.

Topology Hiding

Header	Criteria	Replace Action	Overwrite Value
Referred-By	IP/Domain	Overwrite	adevc.avaya.globalipcom.com
SDP	IP/Domain	Auto	---
From	IP/Domain	Overwrite	adevc.avaya.globalipcom.com
Record-Route	IP/Domain	Auto	---
Refer-To	IP/Domain	Overwrite	199.173.94.24
Via	IP/Domain	Auto	---
Request-Line	IP/Domain	Auto	---
To	IP/Domain	Auto	---

Edit

8.4. Domain Policies

The Domain Policies feature allows users to configure, apply, and manage various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating in the enterprise.

8.4.1 Application Rules

Step 1 - Select **Domain Policies** → **Application Rules** from the left-hand side menu (not shown).

Step 2 - Select the **default-trunk** rule (not shown).

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

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

Session Border Controller for Enterprise AVAYA

Dashboard
Administration
Backup/Restore
System Management
‣ Global Parameters
‣ Global Profiles
‣ PPM Services
‣ **Domain Policies**
 Application Rules
 Border Rules
 Media Rules
 Security Rules
 Signaling Rules
 End Point Policy Groups
 Session Policies

Application Rules: sip-trunk

Add Filter By Device... Rename Clone Delete

Application Rules

- default
- default-trunk
- default-subscriber-low
- default-subscriber-high
- default-server-low
- default-server-high
- sip-trunk**
- rw app rule

Click here to add a description.

Application Rule

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Audio	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	200	200
Video	<input type="checkbox"/>	<input type="checkbox"/>		

Miscellaneous

CDR Support	Off
RTCP Keep-Alive	No

Edit

8.4.2 Media Rules

Media Rules are used to define QoS parameters. Separate media rules are created for Verizon and Session Manager.

8.4.2.1 Enterprise – Media Rule

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 (not shown), and the **Clone Rule** window will open.

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

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

- Select the **Encryption** tab (not shown).
- 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**.

Audio Encryption	
Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80
Preferred Format #2	RTP
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime <small>Leave blank to match any value.</small>	2^N <input type="text"/>
Interworking	<input checked="" type="checkbox"/>

Video Encryption	
Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80
Preferred Format #2	RTP
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime <small>Leave blank to match any value.</small>	2^N <input type="text"/>
Interworking	<input checked="" type="checkbox"/>

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

Finish

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

8.4.2.2 Verizon – Media Rule

Repeat the steps in **Section 8.4.2.1**, 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 **Vz SIPTrk Med Rule**

The completed **Vz SIPTrk Med Rule** screen is shown below.

8.4.3 Signaling Rules

In the reference configuration, Signaling Rules are used to define QoS parameters.

8.4.3.1 Enterprise – Signaling Rules

Step 1 - Select **Domain Policies** → **Signaling Rules** from the left-hand side menu (not shown).

Step 2 - The **Signaling Rules** window will open (not shown). From the Signaling Rules menu, select the **default** rule.

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

- In the **Rule Name** field enter **enterprise sig rule**

- Click **Finish**. The newly created rule will be displayed (not shown).

Step 4 - Highlight the **enterprise sig rule**, select the **Signaling QoS** tab and enter the following:

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

Step 5 - Click **Finish**.

Signaling QoS

Enabled ☒

☐ ToS

Precedence Routine 000

ToS Minimize Delay 1000

☒ DSCP

Value EF 101110

Finish

8.4.3.2 Verizon – Signaling Rule

Step 1 - Select **Domain Policies** from the menu on the left-hand side menu (not shown).

Step 2 - Select **Signaling Rules** (not shown).

Step 3 - From the Signaling Rules menu, select the **default** rule.

Step 4 - Select **Clone Rule** button

- Enter a name: **Vz SIPTrk Sig Rule**

Step 5 - Click **Finish**

Step 6 - Highlight the **Vz SIPTrk Sig Rule**, select the **Signaling QoS** tab and enter the following:

- Click the **Edit** button and the **Signaling QoS** window will open.
- Verify that **Enabled** is selected.
- Select **DCSP**
- Select **Value** = **AF32**

Step 5 - Click **Finish**.

Signaling QoS

Enabled ☒

☐ ToS

Precedence Routine 000

ToS Minimize Delay 1000

☒ DSCP

Value AF32 011100

Finish

8.4.4 Endpoint Policy Groups – Enterprise Connection

Step 1 - Select **Domain Policies** from the menu on the left-hand side.

Step 2 - Select **End Point Policy Groups**.

Step 3 - Select **Add**.

- **Name:** enterprise-sip-trunk
- **Application Rule:** sip-trunk (created in Section 8.4.1)
- **Border Rule:** default
- **Media Rule:** enterprise med rule (created in Section 8.4.2)
- **Security Rule:** default-low
- **Signaling Rule:** enterprise sig rule (created in Section 8.4.3.1)

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

The screenshot shows the 'Policy Groups: enterprise-sip-trunk' configuration page. On the left is a navigation menu with 'End Point Policy Groups' highlighted. The main area has a list of policy groups on the left, with 'enterprise-sip-trunk' selected. The right side shows the configuration details for this group, including a table with the following data:

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

8.4.5 Endpoint Policy Groups – Verizon Connection

Step 1 - Repeat steps 1 through 4 from Section 8.4.4 with the following changes:

- **Group Name:** Vz-policy-group
- **Media Rule:** Vz SIPTrk Med Rule (created in Section 8.4.2.2)
- **Signaling Rule:** Vz SIPTrk Sig Rule (created in Section 8.4.3.2)

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

The screenshot shows the 'Policy Groups: Vz-policy-group' configuration page. The navigation menu on the left is the same, with 'End Point Policy Groups' highlighted. The main area shows the configuration for 'Vz-policy-group'. The table of rules is as follows:

Order	Application	Border	Media	Security	Signaling
1	default-server-high	default	Vz SIPTrk Med Rule	default-low	Vz SIPTrk Sig Rule

8.5. Device Specific Settings

Device Specific Settings allows aggregate system information to be viewed and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network. Specifically, it gives the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality and protocol scrubber rules, end-point and session call flows, as well as the ability to manage system logs and control security features.

8.5.1 Network Management

Step 1 - Select **Device Specific Settings** → **Network Management** from the menu on the left-hand side.

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

The screenshot shows the 'Network Management: SBC1' page. On the left is a sidebar menu with 'Device Specific Settings' expanded and 'Network Management' selected. The main content area has two tabs: 'Interfaces' (active) and 'Networks'. Below the tabs is a table with columns 'Interface Name', 'VLAN Tag', and 'Status'. The table lists four interfaces: A1 (Enabled), A2 (Disabled), B1 (Enabled), and B2 (Enabled). An 'Add VLAN' button is in the top right corner.

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

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

The screenshot shows the 'Network Management: SBC1' page with the 'Networks' tab selected. The table displays IP provisioning details for three networks: Verizon B1, Inside A1, and Public B2. Each row includes columns for Name, Gateway, Subnet Mask / Prefix Length, Interface, and IP Address, along with 'Edit' and 'Delete' links.

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	
Verizon B1	1.1.1.1	255.255.255.0	B1	1.1.1.2	Edit Delete
Inside A1	10.64.91.1	255.255.255.0	A1	10.64.91.48, 10.64.91.49, 10.64.91.50	Edit Delete
Public B2	192.168.0.1	255.255.255.128	B2	192.168.0.44 192.168.0.92	Edit Delete

8.5.2 Media Interfaces

The Media Interface screen is where the SIP media ports are defined. Avaya SBCE will send SIP media on the defined ports. Create a SIP Media Interface for both the inside and outside IP interfaces.

Step 1 - Select **Device Specific Settings** from the menu on the left-hand side.

Step 2 - Select **Media Interface**.

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

- **Name:** Inside-Med-50
- **IP Address:** Select **Inside-A1 (A1,VLAN0)** and **10.64.91.50**
- **Port Range:** 35000 – 40000

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

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

- **Name:** Vz-Med-B1
- **IP Address:** Select **Verizon-B1 (B1,VLAN0)** and **1.1.1.2**
- **Port Range:** 35000 – 40000

Step 6 - Click **Finish** (not shown). Note that changes to these values require an application restart (see **Section 8.1**).

The completed **Media Interface** screen in the shared test environment is shown below.

Media Interface: SBC1

Devices: SBC1

Media Interface

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

Name	Media IP Network	Port Range	TLS Profile	
Inside-Med-50	10.64.91.50 Inside A1 (A1, VLAN 0)	35000 - 40000	None	Edit Delete
Vz-Med-B1	1.1.1.2 Verizon B1 (B1, VLAN 0)	35000 - 40000	None	Edit Delete

Add

8.5.3 Signaling Interface

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 **Device Specific Settings** from the menu on the left-hand side.

Step 2 - Select **Signaling Interface**.

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

- **Name: 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 8.2.2** (e.g., **Inside-Server**)

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

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

- **Name: Vz-sig**
- **IP Address:** Select **Verizon B1 (B1,VLAN0)** and **1.1.1.2**
- **UDP Port: 5060**

Step 6 - Click **Finish** (not shown). Note that changes to these values require an application restart (see **Section 8.1**).

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile	
Vz-sig	1.1.1.2 Verizon B1 (B1, VLAN 0)	---	5060	---	None	Edit Delete
Inside-sig-50	10.64.91.50 Inside A1 (A1, VLAN 0)	---	---	5061	Inside-Server	Edit Delete

8.5.4 Server Flows – For Session Manager

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:** Vz enterprise side.
- **Server Configuration:** EnterpriseCallServer (Section 8.3.4).
- **URI Group:** *
- **Transport:** *
- **Remote Subnet:** *
- **Received Interface:** Vz-sig (Section 8.5.3).
- **Signaling Interface:** Inside-sig-50 (Section 8.5.3).
- **Media Interface:** Inside-Med-50 (Section 8.5.2).
- **End Point Policy Group:** enterprise-sip-trunk (Section 8.4.4).
- **Routing Profile:** route to Vz IPCC (Section 8.3.7).
- **Topology Hiding Profile:** Enterprise-Topology (Section 8.3.8).
- Let other values default.

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

View Flow: Vz enterprise side

X

Criteria	
Flow Name	Vz enterprise side
Server Configuration	EnterpriseCallServer
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Vz-sig

Profile	
Signaling Interface	Inside-sig-50
Media Interface	Inside-Med-50
Secondary Media Interface	None
End Point Policy Group	enterprise-sip-trunk
Routing Profile	route to Vz IPCC
Topology Hiding Profile	Enterprise-Topology
Signaling Manipulation Script	None
Remote Branch Office	Any

8.5.5 Server Flows – For Verizon

Step 1 - Repeat steps 1 through 4 from Section 8.5.4, with the following changes:

- **Flow Name:** Verizon IPCC to CM Flow.
- **Server Configuration:** Verizon IPCC (Section 8.3.5).
- **URI Group:** *
- **Transport:** *
- **Remote Subnet:** *
- **Received Interface:** Inside-sig-50 (Section 8.5.3).
- **Signaling Interface:** Vz-sig (Section 8.5.3).
- **Media Interface:** Vz-Med-B1 (Section 8.5.2).
- **End Point Policy Group:** Vz-policy-group (Section 8.4.5).
- **Routing Profile:** route to SM (Section 8.3.6).
- **Topology Hiding Profile:** Vz IPCC th profile (Section 8.3.9).

View Flow: Verizon IPCC to CM Flow		Profile	
Flow Name	Verizon IPCC to CM Flow	Signaling Interface	Vz-sig
Server Configuration	Verizon IPCC	Media Interface	Vz-Med-B1
URI Group	*	Secondary Media Interface	None
Transport	*	End Point Policy Group	Vz-policy-group
Remote Subnet	*	Routing Profile	route to SM
Received Interface	Inside-sig-50	Topology Hiding Profile	Vz IPCC th profile
		Signaling Manipulation Script	None
		Remote Branch Office	Any

9. Verizon Business IPCC Services Suite Configuration

Information regarding Verizon Business IPCC Services suite offer can be found at <http://www.verizonbusiness.com/products/contactcenter/ip/> 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.

9.1. 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
<i>adevc.avaya.globalipcom.com</i> <i>UDP port 5060</i>	<i>172.30.205.55</i> <i>UDP Port 5072</i>

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

10. Verification Steps

This section provides example verifications of the Avaya configuration with Verizon Business Private IP (PIP) Trunk service.

10.1. Avaya Aura® Communication Manager Verifications

This section illustrates verifications from Communication Manager.

10.1.1 Example Incoming Call from PSTN via Verizon IPCC to Telephone

Incoming PSTN calls arrive from Verizon at Avaya SBCE, which sends the call to Session Manager. Session Manager sends the call to Communication Manager. On Communication Manager, the incoming call arrives via signaling group 2 and trunk group 2.

The following edited Communication Manager **list trace tac** trace output shows a call incoming on trunk group 2. The PSTN telephone dialed 866-850-6850. Session Manager mapped the number received from Verizon to the extension of a Communication Manager telephone (x12003). Extension 12003 is an IP Telephone with IP address 10.64.91.157 in Region 1. Initially, the G450 Media Gateway (10.64.91.81) is used, but as can be seen in the final trace output, once the call is answered, the final RTP media path is “ip-direct” from the IP Telephone (10.64.91.157) to the “inside” of the Avaya SBCE (10.64.91.50) in Region 2.

```
LIST TRACE
time      data
/* Incoming call arrives to Communication Manager for x12003 */
11:58:07 TRACE STARTED 09/14/2017 CM Release String cold-01.0.532.0-23985
11:58:11 SIP<INVITE sips:12003@avayalab.com SIP/2.0
11:58:11      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
11:58:11      active trunk-group 2 member 1      cid 0x5ea
/* Communication Manager sends 183 with SDP as a result of TG 2 configuration */
11:58:11 SIP>SIP/2.0 183 Session Progress
11:58:11      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
11:58:11      dial 12003
11:58:11      ring station      12003 cid 0x5ea
/* Media Server at 10.64.91.60, ringback tone heard by caller */
11:58:11      G729 ss:off ps:20
          rgn:2 [10.64.91.50]:35056
          rgn:1 [10.64.91.60]:6090
11:58:11      G72264K ss:off ps:20
          rgn:1 [10.64.91.157]:25426
          rgn:1 [10.64.91.60]:6092
/* User Answers call, Communication Manager sends 200 OK */
11:58:11 SIP>SIP/2.0 200 OK
11:58:11      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
11:58:11      active station      12003 cid 0x5ea
11:58:12 SIP<ACK sips:10.64.91.65:5071;transport=tls SIP/2.0
11:58:12      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
11:58:12 SIP>INVITE sips:+13035382177@10.64.91.50:5061;transport=tls
11:58:12 SIP>;gsid=45b06bf0-9976-11e7-9cb3-000c29e8354a;sipappsessio
11:58:12 SIP>nid=app-ybxwefkfgh8k SIP/2.0
11:58:12      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
11:58:12 SIP<SIP/2.0 100 Trying
11:58:12      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
11:58:12 SIP<SIP/2.0 200 OK

<continued on next page>
```



```

11:58:12      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
/* Communication Manager sends re-INVITE to begin shuffle to ip-direct */
11:58:12 SIP>INVITE sips:+13035382177@10.64.91.50:5061;transport=tl
11:58:12 SIP>;gsid=45b06bf0-9976-11e7-9cb3-000c29e8354a;wlsscid=-270
11:58:12 SIP>f46a416660660;sipappsessionid=app-ybxwefkfg8k SIP/2.0
11:58:12      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
/* Final media path is ip-direct from answering IP (10.64.91.157) to inside of
SBC (10.64.91.50) */
11:58:12      G729A ss:off ps:20
                rgn:2 [10.64.91.50]:35056
                rgn:1 [10.64.91.157]:25426
11:58:12      G729 ss:off ps:20
                rgn:1 [10.64.91.157]:25426
                rgn:2 [10.64.91.50]:35056
11:58:12 SIP<SIP/2.0 100 Trying
11:58:12      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
/* Communication Manager receives 200 OK with SDP, sends ACK with SDP */
11:58:12 SIP<SIP/2.0 200 OK
11:58:12      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69
11:58:12 SIP>ACK sips:+13035382177@10.64.91.50:5061;transport=tl;gs
11:58:12 SIP>id=45b06bf0-9976-11e7-9cb3-000c29e8354a;wlsscid=-270f46
11:58:12 SIP>a416660660;sipappsessionid=app-ybxwefkfg8k SIP/2.0
11:58:12      Call-ID: a4b93elfefa7ee2746dfc32772f2cf69

```

The following screen shows **Page 2** of the output of the **status trunk** command 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 (10.64.91.157) 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:      10.64.91.65      : 5071
  Far-end:       10.64.91.11      : 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:   10.64.91.157    : 25426
  Far-end:    10.64.91.50     : 35056

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 this same call. Here it can be observed that G.729 codec is used.

```
status trunk 2/1                                     Page 3 of 3
SRC PORT TO DEST PORT TALKPATH
src port: T00011
T00011:TX:10.64.91.50:35056/g729/20ms/1-srtp-aescm128-hmac80
S00003:RX:10.64.91.157:25426/g729a/20ms/1-srtp-aescm128-hmac80
```

10.1.2 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 call was routed to a Communication Manager vector directory number (VDN 10001) associated with a call vector (call vector 1). 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 “route-to” step. The PSTN telephone dialed 866-852-3221. Session Manager can map the number received from Verizon to the VDN extension (x10001), or the incoming call handling table for trunk group 1 can do the same. In the trace below, Session Manager had already mapped the Verizon number to the Communication Manager VDN extension. The annotations in the edited trace highlight key behaviors. At the conclusion, the PSTN caller that dialed the Verizon toll-free number is talking to the Referred-to PSTN destination, and no trunks (i.e., from trunk 1 handling the call) are in use.

Page 1

```

list trace tac *02
/* Session Manager has adapted the dialed number 8668523221 to VDN 10001 */
12:28:09 TRACE STARTED 09/14/2017 CM Release String cold-01.0.532.0-23985
12:28:15 SIP<INVITE sips:10001@avayalab.com SIP/2.0
12:28:15     Call-ID: 9990ed3160377b0d0c7d822a3d541b55
12:28:15     active trunk-group 2 member 1      cid 0x62d
12:28:15     0 0 ENTERING TRACE cid 1581
12:28:15     2 1 vdn e10001 bsr appl      0 strategy 1st-found override n
12:28:15     2 1 AVDN: 10001 AVRDN:
12:28:15     2 1 wait 2 secs hearing ringback
/* Vector step plays ringback. 183 with SDP is sent*/
12:28:15 SIP>SIP/2.0 183 Session Progress
12:28:15     Call-ID: 9990ed3160377b0d0c7d822a3d541b55
12:28:15     dial 10001
12:28:15     ring vector 2      cid 0x62d
12:28:15     G729 ss:off ps:20
12:28:15     rgn:2 [10.64.91.50]:35082
12:28:15     rgn:1 [10.64.91.60]:6134
12:28:17     2 2 # Play announcement to caller i...
12:28:17     2 3 announcement 11006
12:28:17 SIP>SIP/2.0 183 Session Progress
12:28:17     Call-ID: 9990ed3160377b0d0c7d822a3d541b55
12:28:17     2 3      announcement: board 001V9 ann ext: 11006
/* Vector step answers call with announcement. 200 OK is sent */
12:28:17 SIP>SIP/2.0 200 OK
12:28:17     Call-ID: 9990ed3160377b0d0c7d822a3d541b55
12:28:17     active announcement      11006 cid 0x62d
12:28:17     hear ann source 001V9 ext 11006 cid 0x62d
12:28:17     Connected party uses public-unknown-numbering
12:28:17 SIP<ACK sips:+18668523221@10.64.91.65:5071;transport=tls SI
12:28:17 SIP<P/2.0

```

<continued on next page>

```

12:28:17      Call-ID: 9990ed3160377b0d0c7d822a3d541b55
12:28:18      idle announcement          cid 0x62d
12:28:18      2  4 # Refer the call to PSTN Destin...
12:28:18      2  5 route-to number ~r+13035387024 cov n if unconditionally
/* Caller hears pre-REFER announcement, announcement completes, REFER sent */
12:28:18      SIP>REFER sips:+13035382177@10.64.91.50:5061;transport=tls;
12:28:18      SIP>gsid=78db7e80-997a-11e7-9cb3-000c29e8354a;sipappsession
12:28:18      SIP>id=app-qr3kvtjmaso5 SIP/2.0
12:28:18      Call-ID: 9990ed3160377b0d0c7d822a3d541b55
/* Communication Manager receives 202 Accepted sent by Verizon IPCC */
12:28:19      SIP<SIP/2.0 202 Accepted
12:28:19      Call-ID: 9990ed3160377b0d0c7d822a3d541b55
/* Verizon IPCC sends NOTIFY with sipfrag 100 Trying and CM sends 200 OK */
12:28:19      SIP<NOTIFY sips:+18668523221@10.64.91.65:5071;transport=tls
12:28:19      SIP< SIP/2.0
12:28:19      Call-ID: 9990ed3160377b0d0c7d822a3d541b55
12:28:19      SIP>SIP/2.0 200 OK
12:28:19      Call-ID: 9990ed3160377b0d0c7d822a3d541b55
* Note that caller does not hear ringback or any audible feedback until answer
*/
/* Verizon IPCC sends NOTIFY with sipfrag 200 OK and CM sends 200 OK and BYE */
12:28:28      SIP<NOTIFY sips:+18668523221@10.64.91.65:5071;transport=tls
12:28:28      SIP< SIP/2.0
12:28:28      Call-ID: 9990ed3160377b0d0c7d822a3d541b55
12:28:28      SIP>SIP/2.0 200 OK
12:28:28      Call-ID: 9990ed3160377b0d0c7d822a3d541b55
12:28:28      2  5 LEAVING VECTOR PROCESSING cid 1581
12:28:28      SIP>BYE sips:+13035382177@10.64.91.50:5061;transport=tls;gs
12:28:28      SIP>id=78db7e80-997a-11e7-9cb3-000c29e8354a;sipappsessionid
12:28:28      SIP>=app-qr3kvtjmaso5 SIP/2.0
12:28:28      Call-ID: 9990ed3160377b0d0c7d822a3d541b55
12:28:28      idle vector 0          cid 0x62d
/* Trunks are now idle. Caller and refer-to target are connected by Verizon */

```

When the initial call arrived from Verizon, it used trunk member 1 from trunk group 2. In the final state when the PSTN caller is speaking with the answering agent at the Refer-To target, trunk member 1 is idle, reflecting the successful REFER.

```

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

10.2. Avaya Aura® System Manager and Avaya Aura® Session Manager Verifications

This section contains verification steps that may be performed using System Manager for Session Manager. Log in to System Manager. Expand **Elements** → **Session Manager** → **System Status** → **SIP Entity Monitoring**. A screen such as the following is displayed.

Home / Elements / Session Manager / System Status / SIP Entity Monitoring [Help ?](#)

SIP Entity Link Monitoring Status Summary

This page provides a summary of Session Manager SIP entity link monitoring status.

SIP Entities Status for All Monitoring Session Manager Instances

[Run Monitor](#)

1 Items | Refresh Filter: Enable

<input type="checkbox"/>	Session Manager	Type	Monitored Entities					Total
			Down	Partially Up	Up	Not Monitored	Deny	
<input type="checkbox"/>	SessionManager	Core	3	0	11	0	0	14

Select: All, None

All Monitored SIP Entities

[Run Monitor](#)

14 Items | Refresh Filter: Enable

<input type="checkbox"/>	SIP Entity Name
<input type="checkbox"/>	CM-TG3
<input type="checkbox"/>	SBC1
<input type="checkbox"/>	CM-TG1
<input type="checkbox"/>	Breeze
<input type="checkbox"/>	Presence
<input type="checkbox"/>	CM-TG2

Select: All, None < Previous | Page 2 of 2 | Next >

From the list of monitored entities, select an entity of interest, such as **SBC1**. Under normal operating conditions, the **Link Status** should be “UP” as shown in the example screen below.

All Entity Links to SIP Entity: SBC1

[Summary View](#)

Status Details for the selected Session Manager:

1 Items | Refresh Filter: Enable

Session Manager Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/> SessionManager	IPv4	10.64.91.50	5061	TLS	FALSE	UP	200 OK	UP

10.3. Avaya Session Border Controller for Enterprise Verification

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

10.3.1 Welcome Screen

The welcome screen shows alarms, incidents, and the status of all managed Avaya SBCs at a glance.

The screenshot shows the 'Session Border Controller for Enterprise' dashboard. The top navigation bar includes 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main content area is divided into three sections: 'Information', 'Installed Devices', and 'Active Alarms (past 24 hours)'. The 'Information' section displays system details like time, version, build date, license state, and licensing overages. The 'Installed Devices' section lists 'EMS' and 'SBC1'. The 'Active Alarms' section shows 'None found'. The 'Incidents (past 24 hours)' section shows 'SBC1 : Heartbeat Successful, Server is UP'.

Information	
System Time	09:40:19 AM MST Refresh
Version	7.2.2.0-11-15522
Build Date	Tue May 29 11:31:10 UTC 2018
License State	OK
Aggregate Licensing Overages	0
Peak Licensing Overage Count	0
Last Logged in at	12/21/2018 08:23:42 MST
Failed Login Attempts	0

Installed Devices	
EMS	
SBC1	

Active Alarms (past 24 hours)	
None found.	

Incidents (past 24 hours)	
SBC1 : Heartbeat Successful, Server is UP	

10.3.2 Alarms

A list of the most recent alarms can be found under the Alarm tab on the top left bar.

The screenshot shows the 'Alarms' tab selected in the top navigation bar. The main content area displays the 'Session Border Controller for Enterprise' title.

Alarm Viewer:

The screenshot shows the 'Alarm Viewer' interface. The left sidebar lists 'Devices' with 'EMS' and 'SBC1'. The main content area shows the 'Alarms' tab selected, displaying a table with columns: ID, Details, State, Time, and Device. The table is empty, showing 'No alarms found for this device.' Below the table are 'Clear Selected' and 'Clear All' buttons.


ID	Details	State	Time	Device
No alarms found for this device.				

10.3.3 Incidents

A list of all recent incidents can be found under the incidents tab at the top left next to the Alarms.

Incident Viewer:

Incident Viewer



Device All Category All Clear Filters Refresh Generate Report

Displaying results 1 to 15 out of 2000.

Type	ID	Date	Time	Category	Device	Cause
Message Dropped	751976454033216	8/28/17	2:41 PM	Policy	SBC1	No Subscriber Flow Matched
Message Dropped	751976451992077	8/28/17	2:41 PM	Policy	SBC1	No Subscriber Flow Matched
Message Dropped	751976304032669	8/28/17	2:36 PM	Policy	SBC1	No Subscriber Flow Matched
Message Dropped	751976301994346	8/28/17	2:36 PM	Policy	SBC1	No Subscriber Flow Matched

Further Information can be obtained by clicking on an incident in the incident viewer.

Incident Information

X

General Information

Incident Type	Message Dropped	Category	Policy
Timestamp	September 14, 2017 12:49:31 PM MDT	Device	SBC1
Cause	No Subscriber Flow Matched		

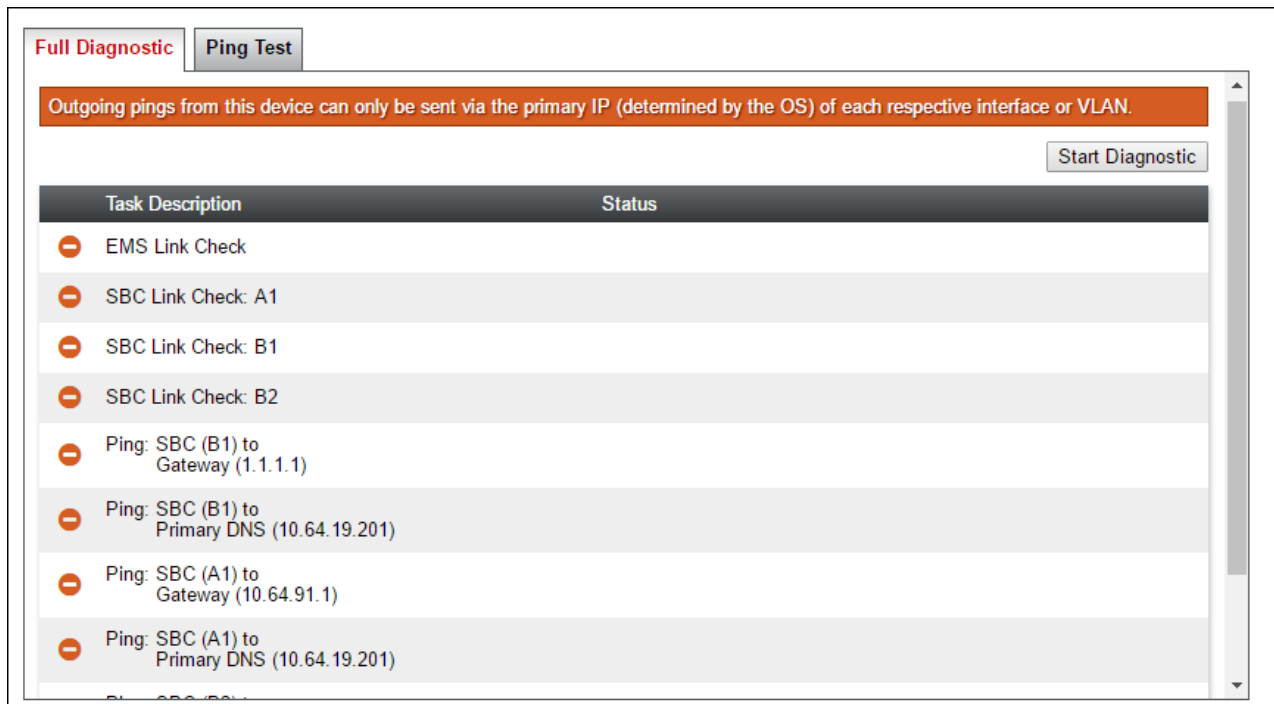
Message Data

Method Name	OPTIONS		
Call ID	74576f0995920a4c8dc0fa62a4ccf899	From	10.64.19.70
To	10.64.91.50	Source IP	10.64.19.70
Destination IP	10.64.91.50		

10.3.4 Diagnostics

The full diagnostics check will verify the link of each interface, and ping the configured next-hop gateways and DNS servers.

Click on Diagnostics on the top bar, select the Avaya SBCE from the list of devices and then click “Start Diagnostics”.



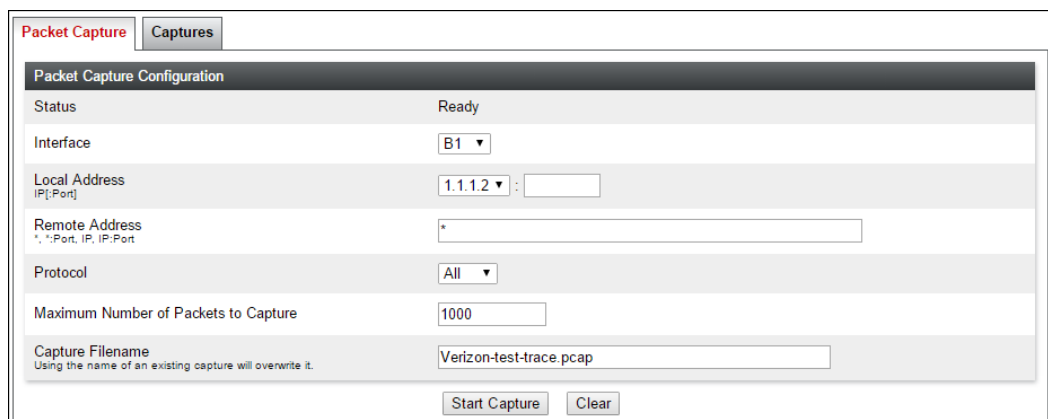
The screenshot shows the 'Full Diagnostic' tab selected. A warning message states: 'Outgoing pings from this device can only be sent via the primary IP (determined by the OS) of each respective interface or VLAN.' A 'Start Diagnostic' button is visible. Below is a table of tasks:

Task Description	Status
EMS Link Check	
SBC Link Check: A1	
SBC Link Check: B1	
SBC Link Check: B2	
Ping: SBC (B1) to Gateway (1.1.1.1)	
Ping: SBC (B1) to Primary DNS (10.64.19.201)	
Ping: SBC (A1) to Gateway (10.64.91.1)	
Ping: SBC (A1) to Primary DNS (10.64.19.201)	

10.3.5 Tracing

To take a call trace, select **Device Specific Settings** → **Troubleshooting** → **Tracie** from the left-side menu (not shown).

Select the Packet Capture tab and set the desired configuration for a call trace and click **Start Capture**.



The screenshot shows the 'Packet Capture' tab. The 'Captures' sub-tab is active. The configuration is as follows:

Packet Capture Configuration	
Status	Ready
Interface	B1
Local Address <small>IP[:Port]</small>	1.1.1.2 :
Remote Address <small>*, *:Port, IP, IP:Port</small>	*
Protocol	All
Maximum Number of Packets to Capture	1000
Capture Filename <small>Using the name of an existing capture will overwrite it.</small>	Verizon-test-trace.pcap
<div>Start Capture Clear</div>	

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

Packet Capture

Captures

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

Packet Capture Configuration

Status

In Progress

Interface

B1

Local Address

IP[:Port]

1.1.1.2

:

Remote Address

*: *:Port, IP, IP:Port

*

Protocol

All

Maximum Number of Packets to Capture

1000

Capture Filename

Using the name of an existing capture will overwrite it.

Verizon-test-trace.pcap

Stop Capture

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.

Packet Capture

Captures

Refresh

File Name	File Size (bytes)	Last Modified	
Verizon-test-trace_20170914125700.pcap	99,108	September 14, 2017 12:57:26 PM MDT	Delete

11. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 8.0, Avaya Aura® Session Manager 8.0, Avaya Aura® Experience Portal 7.2, and Avaya Session Border Controller for Enterprise 7.2 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 (UII).

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

12. References

12.1. 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 8.0, Issue 2, August 2018
- [2] *Administering Avaya Aura® Session Manager*, Release 8.0, Issue 2, August 2018
- [3] *Deploying Avaya Aura® System Manager in Virtualized Environment*, Release 8.0, Issue 2, September 2018
- [4] *Administering Avaya Aura® System Manager for Release 8.0*, Issue 4, September 2018

Avaya Aura® Communication Manager

- [5] *Deploying Avaya Aura® Communication Manager in Virtualized Environment*, Release 8.0, Issue 4, September 2018
- [6] *Administering Avaya Aura® Communication Manager*, Release 8.0, Issue 1, July 2018
- [7] *Administering Avaya G450 Branch Gateway*, Release 8.0, Issue 1, July 2018
- [8] *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 8.0, Issue 2, July 2018
- [9] *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager*, August 2015

Avaya Session Border Controller for Enterprise

- [10] *Administering Avaya Session Border Controller for Enterprise*, Release 7.2.2, Issue 9, April 2018
- [11] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.2.2, Issue 7, April 2018

Avaya Aura® Messaging

- [12] *Administering Avaya Aura® Messaging*, Release 7.0.0, Issue 4, April 2018

Avaya Aura® Experience Portal

- [13] *Administering Avaya Aura® Experience Portal*, Release 7.2.1, Issue 1, March 2018
- [14] *Implementing Avaya Aura® Experience Portal on a single server*, Release 7.2, Issue 1, July 2017

Avaya Application Notes, including the following, are also available at <http://support.avaya.com>

The following Application Notes cover Session Manager 7.1 with Verizon Business IP Contact Center Services.

[VZ-IPCC] – Application Notes for Avaya Aura® Communication Manager 7.1, Avaya Aura® Session Manager 7.1, and Avaya Session Border Controller for Enterprise 7.2 with Verizon Business IP Contact Center (IPCC) Services Suite – Issue 1.0

12.2. 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)*

13. Appendix A – Avaya Session Border Controller for Enterprise – Refer Handling

One of the capabilities important to the Experience Portal environment is the Avaya SBCE Refer Handling option. As described in **Section 3.3**, Experience Portal inbound call processing may include call redirection to Communication Manager agents, or other CPE destinations. This redirection is accomplished by having Experience Portal send SIP REFER messaging to the Avaya SBCE. Enabling the Refer Handling option causes the Avaya SBCE to intercept and process the REFER and generate a new SIP INVITE messages back to the CPE (e.g., Communication Manager).

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

Create a URI Group for numbers intended for Communication Manager.

Step 1 - Select **Global Profiles → URI Groups** from the left-hand menu.

Step 2 - Select **Add** and enter a descriptive **Group Name**, e.g., **internal-extension**, and select **Next** (not shown).

Step 3 - Enter the following:

- **Scheme:** **sip:/sips:**
- **Type:** **Regular Expression**
- **URI:** **12[0-9]{3}@.*** This will match 5-digit local extensions starting with 12, e.g., 12001.
- Select **Finish**.

Edit URIX

Each entry should match a valid SIP URI.

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

Note: This regular expression is case-insensitive.

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

Scheme

☒ sip:/sips:

☐ tel:

Type

☐ Plain

☐ Dial Plan

☒ Regular Expression

URI

12[0-9]{3}@.*

Finish

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

Session Border Controller for Enterprise

URI Groups: internal-extensions

URI Groups

Emergency

IPOSE Users

IP500v2 Users

un-reg-emerg-only

internal-extensions

Click here to add a description.

URI Group

URI Listing

12[0-9]{3}@*	Edit	Delete
50[0-9]{3}@*	Edit	Delete
89[0-9]{3}@*	Edit	Delete

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

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

Step 2 - Select the Verizon Server Interworking Profile created in **Section 8.3.2** and click **Edit**

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

Session Border Controller for Enterprise

Interworking Profiles: SIP Provider Interwk

Interworking Profiles

cs2100

avaya-ru

Enterprise Interwork

Vz REFER Handling

SIP Provider Interwk

Click here to add a description.

General Timers Privacy URI Manipulation Header Manipulation Advanced

General

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

Edit

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