Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 8.0.1 with AT&T IP Toll Free Service – Issue 1.0

Abstract

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

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Note that this document do not include the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service, which are covered on separate Application Notes.

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

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program.

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 8.0.1 with the AT&T IP Toll Free service using AT&T Virtual Private Network (AVPN) or Managed Internet Service Private Network Transport (MIS/PNT) connections[1].

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

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

The Avaya Session Border Controller for Enterprise 8.0.1 (Avaya SBCE) is the point of connection between Avaya Aura® Session Manager and the AT&T IP Toll Free service. It is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

The AT&T IP Toll Free service, referred to in the remainder of this document as IPTF, is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks utilizing AVPN or MIS/PNT transport.

Note – These Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service. That solution is described in a separate document.

[1] MIS/PNT transport does not support compressed RTP (cRTP), however AVPN transport does support cRTP.

2. General Test Approach and Test Results

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

The interoperability compliance testing focused on verifying inbound and outbound call flows between IPTF and the Customer Premises Equipment (CPE) containing Communication Manager, Session Manager, Experience Portal and the Avaya SBCE (see **Section 3.2** for call flow examples).

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

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

For the testing associated with these Application Notes, the interface between Avaya systems and the AT&T Toll Free service did not include use of any specific encryption features as requested by AT&T.

2.1. Interoperability Compliance Testing

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the IPTF network. Calls were made from the PSTN, across the IPTF network, to the CPE.

The following SIP trunking VoIP features were tested with the IPTF service:

- Inbound PSTN/IPTF calls to Communication Manager stations, Vector Directory Numbers (VDNs), Vectors, and Agents.
- Call and two-way talk path establishment between PSTN and Communication Manager telephones/Agents via IPTF.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729A and G.711Mu codecs.
- T.38 fax calls via IPTF to Communication Manager fax endpoints.
- G.711 pass-through fax calls via IPTF to Communication Manager fax endpoints.
- DTMF tone transmission using RFC 2833/4733 between Communication Manager and IPTF automated access systems.
- Verify reception of IPTF SIP Multipart/NSS headers, including SDP and XML content.
- IPTF network features such as Legacy Transfer Connect (inband) and Alternate Destination Routing (ADR).
- Long duration calls.
- Inbound caller interaction with Experience Portal applications, including prompting, caller DTMF input, wait treatment (e.g., announcements and/or music on hold) and Automatic Speech Recognition.
- Experience Portal use of SIP REFER to redirect inbound calls, via the Avaya SBCE, to the appropriate Communication Manager agent extension.
- Call and two-way talk path establishment between callers and Communication Manager agents following redirection from Experience Portal.

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

2.2. Test Results

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

1. **IP Toll Free ADR Call Redirection feature in response to a ring-no-answer condition**. There is an anomaly in the AT&T VIT lab where the Ring No Answer did not get triggered due to Lab restrictions. However, in production, if there is no answer for 20 seconds, ADR Call Redirection will be invoked.
2. **IP Toll Free ADR Call Redirection feature based on SIP error code response**. The IP Toll Free service can be configured to invoke the ADR Call Redirection feature upon receiving of an error response from the CPE.
    - The following error conditions were producible in the reference configuration and tested successfully: 480 Temporarily Unavailable, 486 Busy Here, 500 Server Internal Error and 503 Service Unavailable.
    - Even though the following error conditions were not producible in the reference configuration, the associated error codes were simulated via an Avaya SBCE signaling manipulation rule, and also tested successfully: 408 Request Timeout, 504 Server Timeout, and 600 Busy Everywhere.
3. **G.726-32 codec support**. While Communication Manager supports G.726-32, the IPTF implementation of G.726-32 results in poor audio quality. Therefore, G.726-32 codec is not supported between Communication Manager and the IPTF service.
4. **T.38/G.729 fax is limited to 9600bps when using the G4xx Media Gateways**. A G430 Media Gateway is used in the reference configuration. As a result, T.38/G.729 fax was limited to 9600 bps. Also note that the sender and receiver of a T.38 fax call may use either Group 3 or Super Group 3 fax machines, but the T.38 fax protocol carries all fax transmissions as Group 3.
5. **G.711 pass-through fax**. Inbound G.711 pass-through fax was tested in addition to T.38 fax. This was done by configuring a separate Communication Manager ip-codec-set (**Section 14**). Faxes using G.711 pass-through generally completed at better line speeds (rates of 14400 bps were observed). However, when the PSTN sender and CPE receiver both used SG3 fax devices, the results were erratic. Due to the unpredictability of pass-through techniques, which only work well on networks with very few hops and with limited end-to-end delay, G.711 fax pass-through is delivered in Communication Manager on a "best effort" basis; its success is not guaranteed, and it should be used at the customer's discretion. T.38 should be the preferred method for faxing.
6. **IP Toll Free services IP InfoPack** and **Landline/Mobility test cases could not be executed**. The AT&T supplied IP Toll Free test plan specifies test cases to verify the inbound transmission of INFOPAK and Landline/Mobility data by the IP Toll Free service. Due to network provisioning and lab support issues, these test cases could not be executed.
7. **Removal of unnecessary SIP headers**. In an effort to reduce packet size (or block a header containing private addressing), Session Manager is provisioned to remove SIP headers not required by the AT&T IPTF service (see **Section 6.4.2**). These headers are:
    - AV-Global-Session-ID, Alert-Info, Endpoint-View, P-AV-Message-Id, P-Charging-Vector, P-Location, Av-Secure-Indication

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

8. **Avaya SIP endpoints may generate three Bandwidth headers; b=TIAS:64000, b=CT:64, and b=AS:64, causing AT&T network issues**. Certain Avaya SIP endpoints (e.g., 9641, 9621, and 9608 models) may generate various Bandwidth headers depending on the call flow. It has been observed that sending these Bandwidth headers may cause issues with AT&T services. Therefore, an Avaya SBCE Signaling Manipulation Rule is used to remove these headers (see **Section 8.8**).
9. **Enhanced CID – NSS feature**. The inbound calls to Communication Manager are not exercising the Enhanced CID feature. Although Communication Manager is accepting SIP Multipart/NSS headers, it is neither passing nor acting upon it. It is simply being ignored.
10. **Avaya SBCE inserts a=ptime:20 in the SIP SDP toward Communication Manager**. AT&T includes a=maxptime:30 in the SIP SDP to recommend a ptime value of 30ms, but does not specify a ptime value in the SDP. If no media packetization attribute (ptime) is included in the SIP Session Description Protocol (SDP), Avaya SBCE inserts "a=ptime:20", specifying 20 milliseconds. Although Communication Manager can be configured to send ptime with a value of 30ms (See **Section 5.7.2**), it will send a ptime value of 20ms when it receives "a=ptime:20" from the Avaya SBCE. This causes the media packetization to be set to 20ms. No issues were found during testing due to this behavior.

2.3. Support

AT&T customers may obtain support information for the AT&T IP Toll Free service by visiting https://www.business.att.com/products/ip-toll-free.html or by calling (800) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting the Support page: http://support.avaya.com. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers provided on the Support website to directly access specific support and consultation services based upon their Avaya support agreements.

3. Reference Configuration

The reference configuration used in these Application Notes is shown in **Figure 1** and consists of several components:

- Session Manager provides core SIP routing and integration services that enables communication between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Avaya SIP endpoints register to Session Manager.
- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager provides the voice communication services for a particular enterprise site. Avaya H.323 endpoints register to Communication Manager.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G430 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya Aura® Media Server provides additional media resources for Communication Manager.
- Experience Portal self-service applications allow callers to automatically obtain assistance or information without the need for agent interaction. Experience Portal can also redirect calls to Communication Manager agents based on the caller's selections to the prompts.
- Avaya Aura® Messaging (Messaging) was used in the reference configuration to provide voice mailbox capabilities. This solution is extensible to other Avaya messaging platforms. The provisioning of Avaya Aura® Messaging is beyond the scope of this document.
- Avaya desk telephones are represented with Avaya 96x1 Series IP Deskphones (running H.323 and SIP firmware), J100 Series IP Deskphones using the SIP software bundle Avaya 9408 Digital Deskphones, as well as Avaya Equinox™ for Windows softphones.
- The Avaya SBCE provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the IPTF service and the enterprise internal network.
- The IPTF service uses SIP over UDP to communicate with enterprise edge SIP devices, e.g., the Avaya SBCE. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Avaya SBCE (e.g., UDP, TCP, or TLS) and Communication Manager (e.g., TCP or TLS). In the reference configuration, Session Manager uses SIP over TLS to communicate with the Avaya SBCE, Experience Portal, Messaging and Communication Manager.
- Inbound calls were placed from the PSTN via the IPTF service, through the Avaya SBCE to Session Manager. Session Manager used the configured dial patterns and routing policies to determine where to send the call (e.g., Communication Manager, Experience Portal). Communication Manager terminated the calls to the appropriate Agent queue, Agent phone, or fax extension.

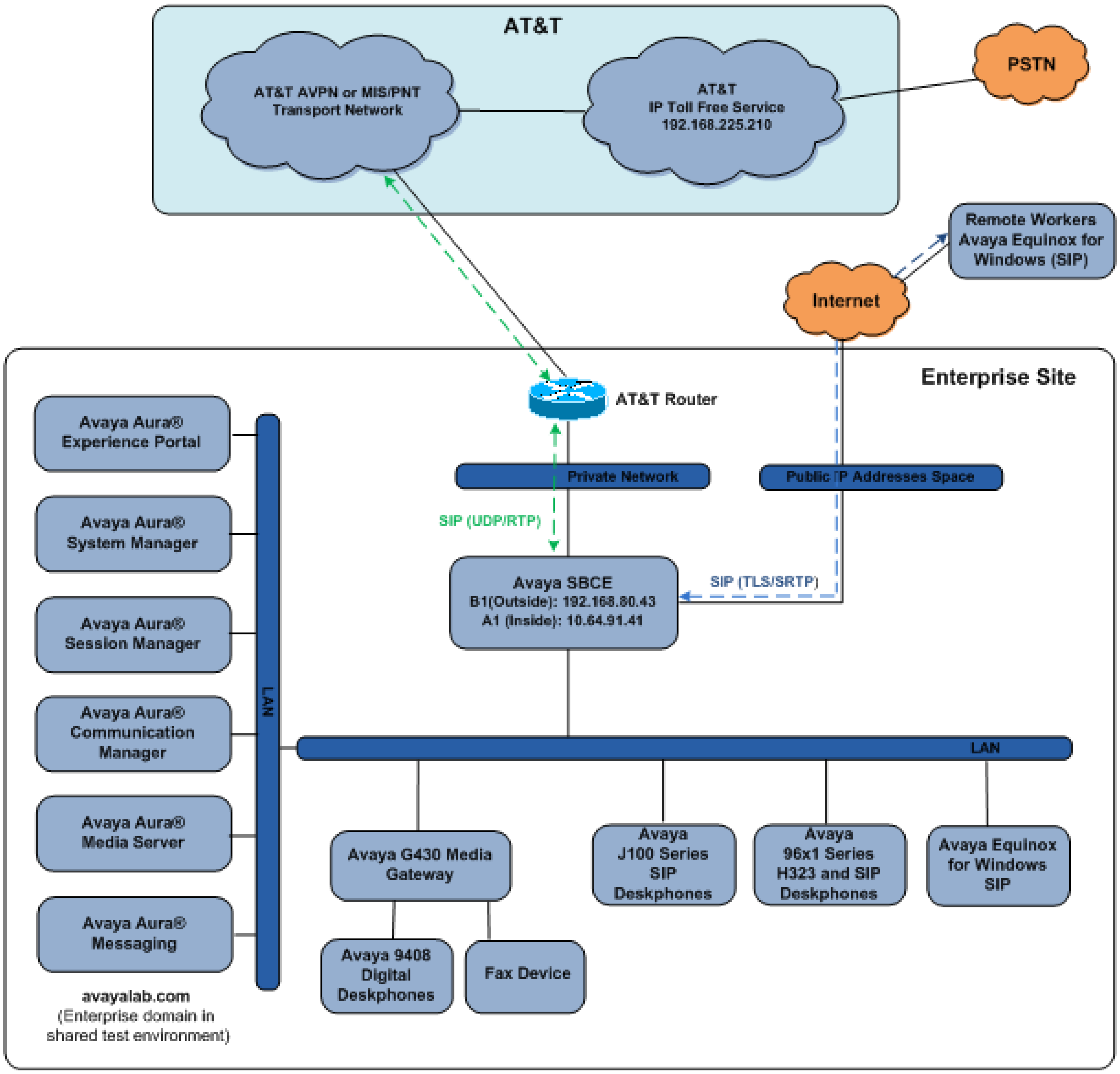


Figure 1: Reference configuration

Note – In the reference configuration, the IPTF service delivered 10 DNIS digits, with the format *00000xxxxx*. These DNIS digits are used in the provisioning defined in the following sections, not the dialed digits. The DNIS digit length can vary depending on the customer's needs. Although during testing 10 digits were used, the total length supported by the IPTF service is 21 digits, including the five leading zeroes.

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

Note - The AT&T IP Toll Free service Border Element IP address and DNIS digits, (destination digits specified in the SIP Request URIs sent by the AT&T Toll Free service) are shown in this document as examples. AT&T Customer Care will provide the actual IP addresses and DNIS digits as part of the IP Toll Free provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura® System Manager	
IP Address	10.64.90.82
Avaya Aura® Session Manager	
IP Address	10.64.91.81
Avaya Aura® Communication Manager	
IP Address	10.64.91.75
Communication Manager dialplan	89xxx = Stations 2xxxx = Agents 71xxx = Agent skill queue VDNs
Avaya Aura® Messaging	
IP Address	10.64.91.84
Avaya Aura® Experience Portal	
IP Address	10.64.91.90
Avaya Session Border Controller for Enterprise (SBCE)	
IP Address of Inside (Private) Interface	10.64.91.41
IP Address of Outside (Public) Interface	192.168.80.43 (see note below)
AT&T IP Toll Free Border Element	
IP Address	192.168.225.210

Table 1: Illustrative Values Used in these Application Notes

Note – For security reasons, the actual IP addresses of the Avaya SBCE and AT&T BE are not included in this document. However, as placeholders in the following configuration sections, the IP address of **192.168.80.43** (Avaya SBCE public interface) and **192.168.225.210** (AT&T BE IP address) are specified.

3.2. Call Flows

To understand how inbound AT&T IP Toll Free service calls are handled in the Avaya CPE environment, three basic call flows are described in this section.

3.2.1. Communication Manager Call Flow

In the general call flow shown on **Figure 2** below, an inbound IPTF service call arrives at the Avaya SBCE and is subsequently routed to Session Manager and to Communication Manager.

1. A PSTN telephone originates a call to an IPTF service number.
2. The PSTN routes the call to the IPTF service network.
3. The IPTF service routes the call to the Avaya SBCE.
4. The Avaya SBCE performs SIP Network Address Translation (NAT) and any necessary SIP header modifications and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to Communication Manager.
6. Depending on the called number, Communication Manager routes the call to an Agent queue or telephone.

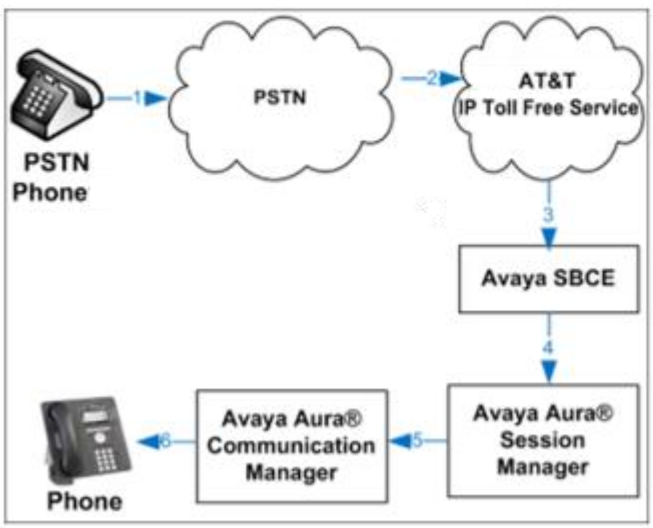


Figure 2: Inbound AT&T IP Toll Free Service Call to an Agent queue/telephone

Note: The IPTF service features such as Legacy Transfer Connect and Alternate Destination Routing utilize this call flow as well.

3.2.2. Experience Portal Call Flows

The call scenario illustrated on **Figure 3** below shows an inbound call arriving and remaining on Experience Portal.

1. A PSTN phone originates a call to an IPTF number.
2. The PSTN routes the call to the IPTF network.
3. IPTF 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.

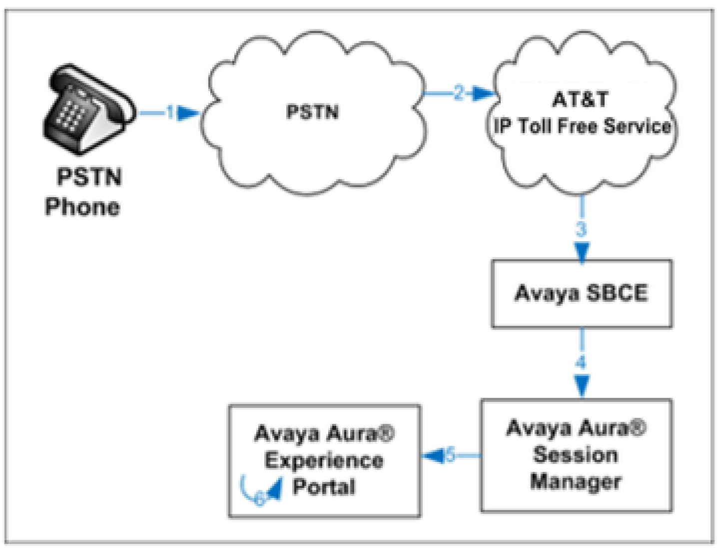


Figure 3: Inbound Call Handling Entirely by Avaya Aura® Experience Portal

The next call scenario illustrated on **Figure 4** below shows an inbound call arriving on Experience Portal, and transferred to an agent in Communication Manager.

1. Same as the first five steps from the previous 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.

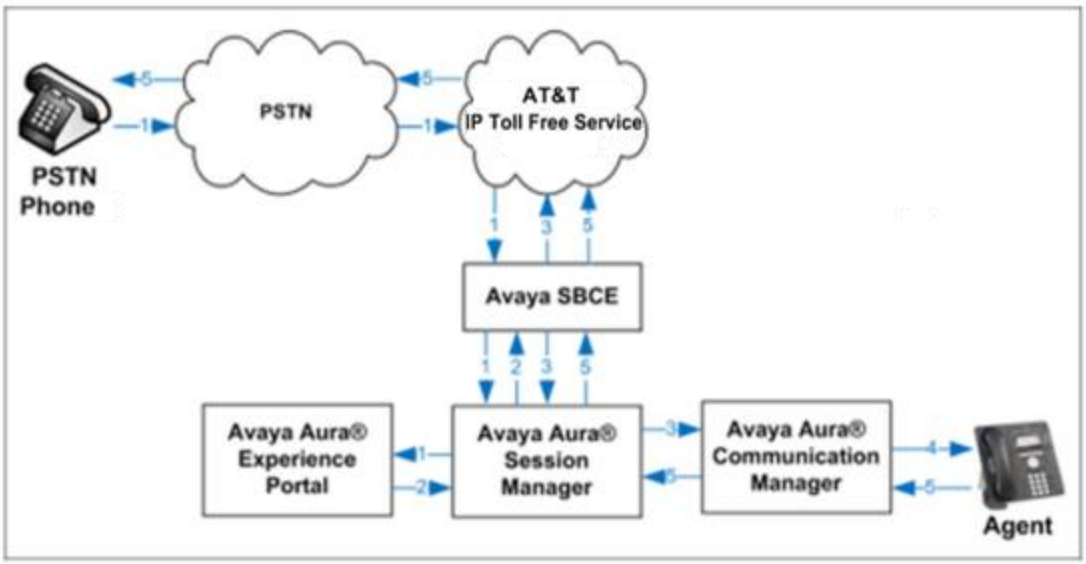


Figure 4: Avaya Aura® Experience Portal Transfers Call to Avaya Aura® Communication Manager

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

4. Equipment and Software Validated

The following equipment and software was used for the reference configuration described in these Application Notes.

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

Table 2: Equipment and Software Versions

5. Configure Avaya Aura® Communication Manager

This section describes the administration steps for Communication Manager in support of the reference configuration described in these Application Notes. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager administration has already been performed. Consult **[5]** and **[6]** in the References section for further details if necessary.

Note – In the following sections, only the parameters that are highlighted in **bold** text are applicable to these application notes. Other parameter values may or may not match based on local configurations.

5.1. System-Parameters Customer-Options

This section reviews the Communication Manager licenses and features that are required for the reference configuration described in these Application Notes.

NOTE - For any required features that cannot be enabled in the steps that follow, contact an authorized Avaya account representative to obtain the necessary licenses.

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

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

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

Step 2 - On **Page 5** of the form, verify that the **Media Encryption Over IP** field is set to **y**.

```
display system-parameters customer-options                      Page   5 of  12
                                OPTIONAL FEATURES
   Emergency Access to Attendant? y                              IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                        ISDN Feature Plus? n
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                 ISDN-PRI? y
              ESS Administration? y               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
     External Device Alarm Admin? y                 Media Encryption Over IP? y
   Five Port Networks Max Per MCC? n    Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                 Multifrequency Signaling? y
      Global Call Classification? y         Multimedia Call Handling (Basic)? y
             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 3 - On **Page 6** of the form, verify that the **Processor Ethernet** field is set to **y**.

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

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

5.2. System-Parameters Features

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

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

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


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

5.3. Dial Plan

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

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

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

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

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

5.4. Node Names

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

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

- Session Manager SIP signaling interface (e.g., **SM** and **10.64.91.81**).
- Media Server (e.g., **AMS** and **10.64.91.86**). The Media Server node name is only needed if a Media Server is present.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name                 IP Address
AMS                   10.64.91.86
SM                    10.64.91.81
default               0.0.0.0
procr                 10.64.91.75
procr6                ::
```

5.5. Processor Ethernet

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

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

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

                  Type: PROCR
                                                   Target socket load: 4800

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

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

5.6. IP Network Regions

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

5.6.1. IP Network Region 1 – Local CPE Region

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

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

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

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

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

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

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

 RTCP Reporting to Monitor Server Enabled? y

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

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

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

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

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

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

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

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

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

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

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

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

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

Note: An additional IP Network Region and IP Codec Set were created in the reference configuration, used to test G.711 pass-through fax. Details of this optional configuration can be found in **Section 14**.

5.7. IP Codec Sets

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

Note – The IPTF service offers G.729A, G.726-32, and G.711MU codecs in their Invite SDP. G.726-32 codec is supported by Communication Manager, but testing found issues when G.726-32 codec is used (see **Section 2.2**, **item 2**). In addition, some calls could require support of G.729B (silence suppression). Therefore G.729B is also included in the codec lists.

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

Step 1 - Enter the **change ip-codec-set x** command, where **x** is the number of an IP codec set used for internal calls (e.g., **1**). On **Page 1** of the **ip-codec-set** form, ensure that **G.711MU**, **G.729A**, and **G.729B** are included in the codec list. Note that the packet interval size will default to 20ms. Under **Media Encryption**, ensure **1-srtp-aescm128-hmac80** is included to support Secure Real-time Transport Protocol (SRTP).

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

                          IP CODEC SET
    Codec Set: 1

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

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

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

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

                          IP CODEC SET

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

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

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

Step 1 - Repeat the steps in **Section 5.7.1** with the following changes.

- Provision the codecs in the order shown below. Note that the order of G.729A and G.729B codecs may be reversed as required.
- Set **Frames Per Pkt** to **3**. This will auto-populate **30** for the **Packet Size (ms)** field, and specify a PTIME value of 30 in the SDP (recommended by AT&T). See **Section 2.2**, **Item 9** for limitations with the packet size.

```
change ip-codec-set 4                                           Page   1 of   2

                          IP CODEC SET
    Codec Set: 4

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729A           n          3        30
 2: G.729B           n          3        30
 3: G.711MU          n          3        30

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

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

Note: An additional IP Network Region and IP Codec Set were created in the reference configuration, used to test G.711 pass-through fax. Details of this optional configuration can be found in **Section 14**.

5.8. SIP Trunks

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

- Inbound IPTF access – SIP Trunk 4. This trunk will use TLS port 5064
- Internal CPE access (e.g., Avaya SIP telephones, etc.) – SIP Trunk 3. This trunk will use TLS port 5061.

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

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

5.8.1. SIP Trunk for Inbound AT&T calls

This section describes the steps for administering the SIP trunk to Session Manager used for inbound IPTF calls. This trunk corresponds to the **CM-TG4** SIP Entity defined in **Section 6.5.2**.

5.8.1.1 Signaling Group 4

Step 1 - Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **4**), and provision the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tls**.
- Verify that **IMS Enabled?** is set to **n**.
- Verify that **Peer Detection Enabled?** is set to **y**. The systems will auto detect and set the **Peer Server** to **SM**.
- **Near-end Node Name** – Set to the node name of the **procr** noted in **Section 5.4**.
- **Far-end Node Name** – Set to the node name of Session Manager as administered in **Section 5.4** (e.g., **SM**).
- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5064**.
- **Far-end Network Region** – Set the IP network region to **4**, as set in **Section 5.6.2**.
- **Far-end Domain** – Enter **avayalab.com**. This is the domain provisioned for Session Manager in **Section 6.2**.
- **DTMF over IP** – Set to **rtp-payload** to enable Communication Manager to use DTMF according to RFC 2833.
- **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized directly to the associated stations, to reduce the use of media resources on the Avaya Media Gateway when possible (known as shuffling).
- **Enable Layer 3 Test** – Set to **y**. This directs Communication Manager to send SIP OPTIONS messages to Session Manager to check link status.

```
add signaling-group 4                                           Page   1 of   2
                                SIGNALING GROUP

 Group Number: 4              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM                      Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: SM
 Near-end Listen Port: 5064                 Far-end Listen Port: 5064
                                         Far-end Network Region: 4

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

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

5.8.1.2 Trunk Group 4

Step 1 - Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g., **4**). On **Page 1** of the **trunk-group** form, provision the following:

- **Group Type** – Set to **sip**.
- **Group Name** – Enter a descriptive name (e.g., **ATT IPTF**).
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., ***04**).
- **Direction** – Set to **incoming**.
- **Service Type** – Set to **public-ntwrk**.
- **Signaling Group** – Set to the signaling group administered in **Step 1** (e.g., **4**).
- **Number of Members** – Enter the maximum number of simultaneous calls desired on this trunk group (based on licensing) (e.g., **20**).

```
add trunk-group 4                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 4                    Group Type: sip           CDR Reports: y
  Group Name: ATT IPTF                    COR: 1        TN: 1        TAC: *04
   Direction: incoming       Outgoing Display? n
 Dial Access? n                                    Night Service:

Service Type: public-ntwrk         Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 4
                                                     Number of Members: 20
```

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

- Set the **Preferred Minimum Session Refresh Interval (sec):** to **900**.

```
add trunk-group 4                                               Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                          Redirect On OPTIM Failure: 5000

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

 Disconnect Supervision - In? y  Out? y

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

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

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

```
add trunk-group 4                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y

                     Numbering Format: public
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? y
                                               Replace Unavailable Numbers? y

                                                  Hold/Unhold Notifications? y


 Show ANSWERED BY on Display? y
```

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

- Set **Telephone Event Payload Type** to the RTP payload type recommended by the IPTF service (e.g., **100**).

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

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

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? n

                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 100
                                       Shuffling with SDP? n

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

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

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

This trunk corresponds to the **CM-TG3** SIP Entity defined in **Section 6.5.3**.

5.8.2.1 Signaling Group 3

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

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

Step 2 - Set the following parameters on page 1:

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

5.8.2.2 Trunk Group 3

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

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

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

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

- Same as **Section 5.8.1.2**.

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

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

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

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

Use default values for all other settings.

5.9. Public Numbering

In the reference configuration, the public-unknown-numbering form, (used in conjunction with the **Numbering Format: public** setting in **Section 5.8.1**), is used to convert Communication Manager local extensions to IPTF DNIS numbers, for inclusion in any SIP headers directed to the IPTF service via the public trunk.

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

Step 2 - Add any Communication Manager station extensions and their corresponding IPTF DNIS number (for the public trunk):

- **Ext Len** – Enter the total number of digits in the local extension range (e.g., **5**).
- **Ext Code –** Enter the Communication Manager station extension (e.g., SIP phone **89324**). (Not shown).
- **Trk Grp(s)** – Enter the number of the Public trunk group (e.g., **4**).
- **CPN Prefix** – Enter the corresponding IPTF DNIS number (e.g., **0000011041**).
- **CPN Len** – Enter the total number of digits after the digit conversion (e.g., **10**).

Step 3 - Add any Communication Manager Agent skill VDN extensions and their corresponding IPTF DNIS number (for the public trunk):

- **Ext Len** – Enter the total number of digits in the local extension range (e.g., **5**).
- **Ext Code –** Enter the Communication Manager extension (e.g., Skill VDN **71041**).
- **Trk Grp(s)** – Enter the number of the Public trunk group (e.g., **4**).
- **CPN Prefix** – Enter the corresponding IPTF DNIS number (e.g., **0000011041**).
- **CPN Len** – Enter the total number of digits after the digit conversion (e.g., **10**).

Step 4 - Repeat **Steps 2** and **3** for all IPTF DNIS numbers and their corresponding Communication Manager station, Skill, or Agent extensions.

```
change public-unknown-numbering 5 ext-digits 71041             Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                            Total
Ext Ext              Trk       CPN            CPN
Len Code             Grp(s)    Prefix         Len
                                                  Total Administered: 20
 5  71041            4         0000011041     15     Maximum Entries: 240
 5  71042            4         0000021042     15
 5  71043            4         0000031043     15  Note: If an entry applies to
 5  71044            4         0000041044     15  a SIP connection to Avaya
                                                  Aura(R) Session Manager,
                                                  the resulting number must
                                                  be a complete E.164 number.

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

5.10. Private Numbering

In the reference configuration, the private-numbering form, (used in conjunction with the **Numbering Format: private** setting in **Section 5.8.2**), 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 Communication Manager extension patterns defined in the Dial Plan in **Section 5.3** (e.g., **20, 71, 89**).
- **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 1                                 Page   1 of   2
                            NUMBERING - PRIVATE FORMAT

Ext Ext              Trk         Private          Total
Len Code             Grp(s)      Prefix           Len
 5  12               3                            5     Total Administered: 6
 5  14               3                            5        Maximum Entries: 540
 5  20               3                            5
 5  71               3                            5
 5  89               3                            5
```

5.11. Route Pattern for Local SIP Trunk

Route Patterns are used to direct calls to the Local SIP trunk for access to SIP phones or other destinations in the CPE. This form specifies the local SIP trunk (e.g., 3), based on the route-pattern selected by the AAR table in **Section 5.12** (e.g., calls SIP phone extensions).

Note – As IPTF is an inbound only service, no outbound route patterns are defined for the public SIP trunk.

Step 1 - Enter the **change route-pattern 3** command and enter the following:

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

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

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

5.12. Automatic Alternate Routing (AAR) Dialing

AAR is used to direct calls to the local SIP trunk for Avaya SIP telephones, using the route pattern defined in **Section 5.11**.

Step 1 - Enter the following:

- **Dialed String -** In the reference configuration all SIP telephones used extensions in the range 89xxx, therefore enter **89**.
- **Min** & **Max** – Enter **5**.
- **Route Pattern** – Enter **3**.
- **Call Type** – Enter **lev0**.

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

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    20                    5    5     3         lev0         n
    89                    5    5     3         lev0         n
```

5.13. Provisioning for Simulated Call Center Functionality

In the reference configuration, a Call Center environment (skill queues and Agents) was simulated on Communication Manager. The administration of Communication Manager Call Center type elements – Agents, skills (hunt groups), vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Consult **[6]** and **[10]** in the References section for further details. The samples that follow are provided for reference purposes only.

- Agent form – **Page 1**

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

                Login ID: 20001                                 AAS? n
                    Name: Agent 1                             AUDIX? n
                      TN: 1       Check skill TNs to match agent TN? n
                     COR: 2
           Coverage Path: 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: acd
 AUX Agent Remains in LOA Queue: system            MIA Across Skills: system
AUX Agent Considered Idle (MIA): system    ACW Agent Considered Idle: system
             Work Mode on Login: system    Aux Work Reason Code Type: system
                                             Logout Reason Code Type: system
                        Maximum time agent in ACW before logout (sec): system
                                            Forced Agent Logout Time:   :
    WARNING:  Agent must log in again before changes take effect
```

- Agent form – **Page 2**

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

    SN   RL SL          SN   RL SL
 1: 1       1       16:
```

- Skill 1 Hunt Group form – **Page 1**

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

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

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

- Skill 1 VDN form – **Page 1**

```
display vdn 71041                                             Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 71041
                                 Name*: ATT Toll-Free 1
                           Destination: Vector Number        4
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

- Skill 1 Vector form – **Page 1**

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

    Number: 4                Name: Call Center
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 #    Wait hearing ringback
02 wait-time    2   secs hearing ringback
03 #    Play greeting and collect 1 digit
04 collect      1    digits after announcement 11001    for none
05 goto step    7             if digits           =      1
06 stop
07 #    Simple queue to skill with recurring announcement until available
08 queue-to     skill 1    pri m
09 announcement 11004
10 wait-time    30  secs hearing music
11 goto step    8             if unconditionally
12 stop
```

5.14. Avaya G430 Media Gateway Provisioning

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

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

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

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

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

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

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

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

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

   Mutual Authentication? optional
```

5.15. Avaya Aura® Media Server Provisioning

In the reference configuration, an Avaya Aura® Media Server is provisioned. The Media Server is used, along with the G430 Media Gateway, for local DSP resources, announcements, and Music On Hold.

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

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

Step 2 - On the Media Server Element Manager, navigate to **Home → System Configuration → Signaling Protocols → SIP →Node and Routes** and add the Communication Manager Procr interface IP address (e.g., **10.64.91.75**, see **Section 5.4**) as a trusted node (not shown).

Step 3 - On Communication Manager, enter the **add signaling-group x** command where x is an unused signaling group (e.g., **80**), and provision the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tls**
- Verify that **Peer Detection Enabled?** – Set to **n**.
- **Peer Server** to **AMS**.
- **Near-end Node Name** – Set to the node name of the **procr** noted in **Section 5.4**.
- **Far-end Node Name** – Set to the node name of Media Server as administered in **Section 5.4** (e.g., **AMS**).
- **Near-end Listen Port** – Set to **9061** (default).
- **Far-end Listen Port** – Set to **5061** (default).
- **Far-end Network Region** – Set the IP network region to **1**, as set in **Section 5.6.1**.
- **Far-end Domain** – Automatically populated with the IP address of the Media Server.

```
add signaling-group 60                                          Page   1 of   2
                                SIGNALING GROUP

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


  Peer Detection Enabled? n  Peer Server: AMS



   Near-end Node Name: procr                Far-end Node Name: AMS
 Near-end Listen Port: 9061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1

Far-end Domain: 10.64.91.86
```

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

- Signaling **Group** – Enter the signaling group previously configured for Media Server (e.g., **80**).
- **Voip Channel License Limit** – Enter the number of VoIP channels for this Media Server (based on licensing) (e.g., **300**).
- **Dedicated Voip Channel Licenses** – Enter the number of VoIP channels licensed to this Media Server (e.g., **300**)
- Remaining fields are automatically populated based on the signaling group provisioning for the Media Server.

```
add media-server 1                                              Page   1 of   1
                                  MEDIA SERVER

                    Media Server ID: 1

                    Signaling Group: 80
         Voip Channel License Limit: 300
    Dedicated Voip Channel Licenses: 300

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

5.16. Save Translations

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

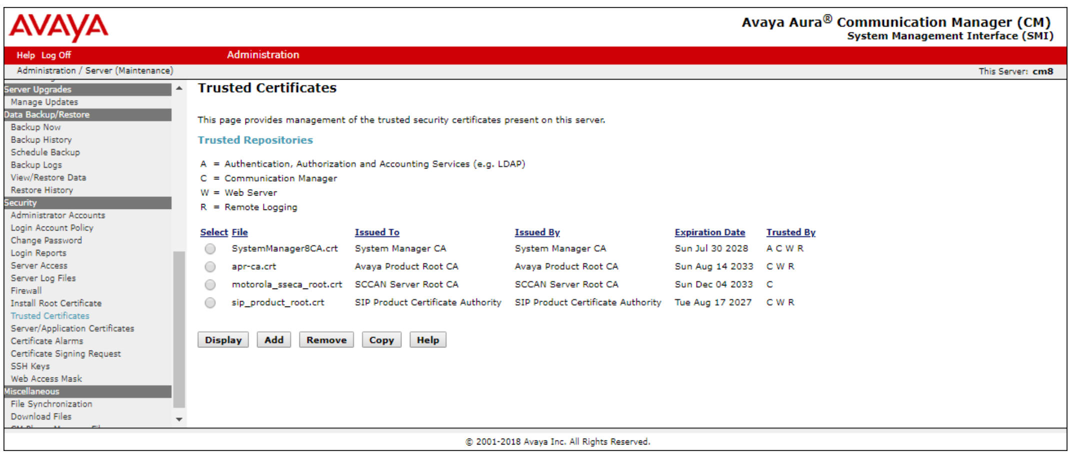
5.17. Verify TLS Certificates – Communication Manager

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

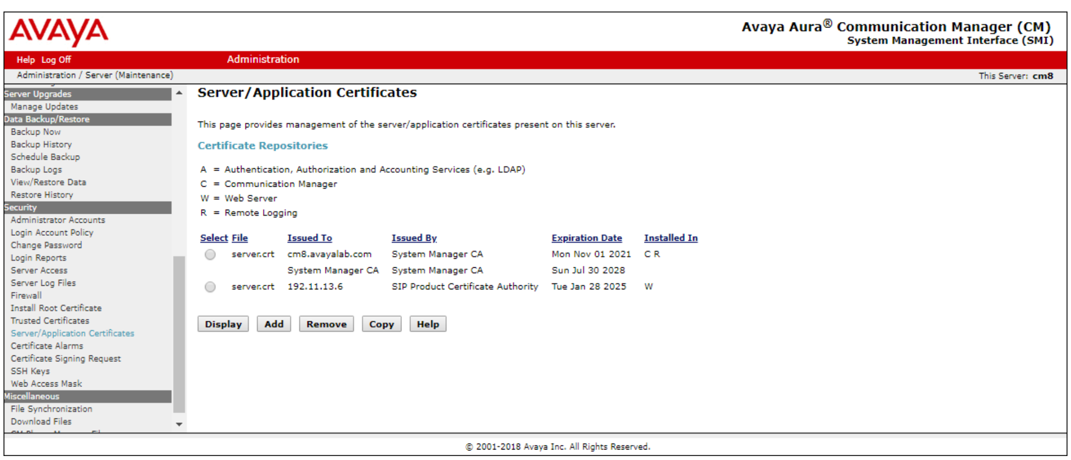
In the reference configuration, TLS transport is used for the communication between Session Manager and Communication Manager. The following procedures show how to verify the certificates used by Communication Manager.

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

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



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



6. Configure Avaya Aura® Session Manager

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

This section provides the procedures for configuring Session Manager to receive calls from and route calls to the SIP trunk between Communication Manager and Session Manager, and the SIP trunk between Session Manager and the Avaya SBCE. In addition, provisioning for calls to Avaya Experience Portal and Messaging are described.

Session Manager serves as a central point for supporting SIP-based communication services in an enterprise. Session Manager connects and normalizes disparate SIP network components and provides a central point for external SIP trunking to the PSTN. The various SIP network components are represented as SIP Entities and the connections/trunks between Session Manager and those components are represented as Entity Links.

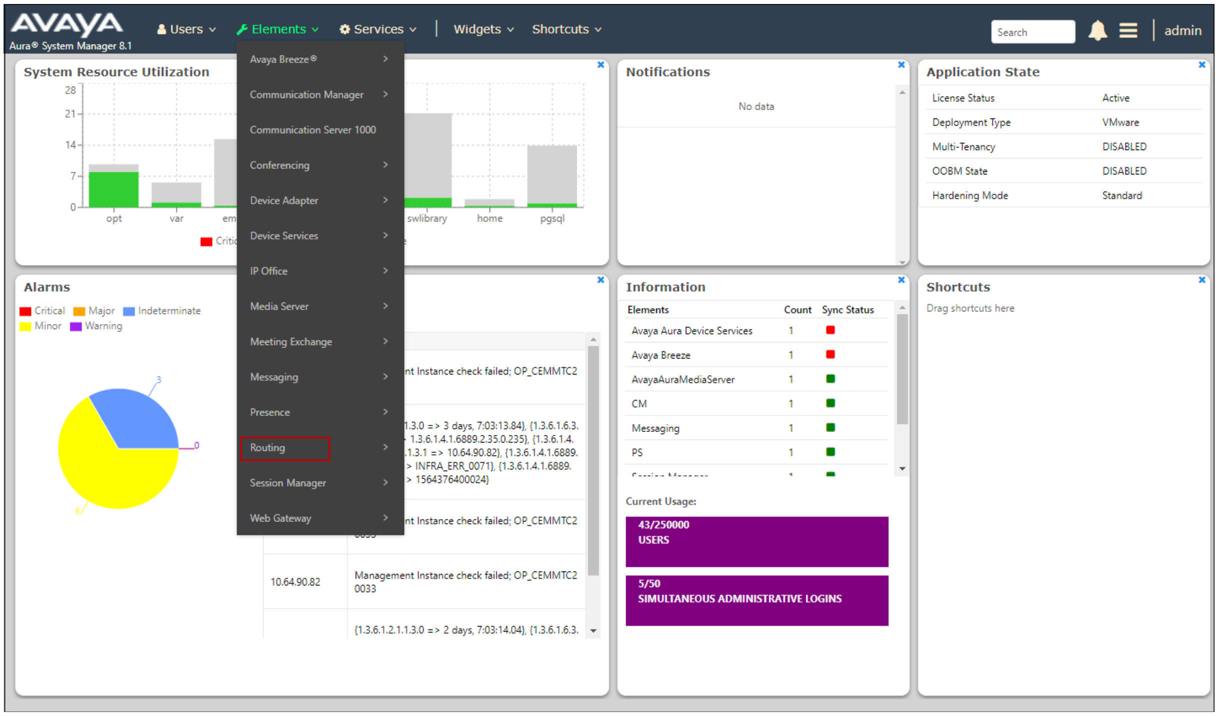
When calls arrive at Session Manager from a SIP Entity, Session Manager applies SIP protocol and numbering modifications to the calls. These modifications, referred to as Adaptations, are sometimes necessary to resolve SIP protocol differences between disparate SIP Entities, and also serve the purpose of normalizing the calls to a common or uniform numbering format, which allows for simpler administration of routing rules in Session Manager. Session Manager then matches the calls against certain criteria embodied in profiles termed Dial Patterns and determines the destination SIP Entities based on Routing Policies specified in the matching Dial Patterns. Lastly, before the calls are routed to the respective destinations, Session Manager again applies Adaptations in order to bring the calls into conformance with the SIP protocol interpretation and numbering formats expected by the destination SIP Entities.

The following administration activities will be described:

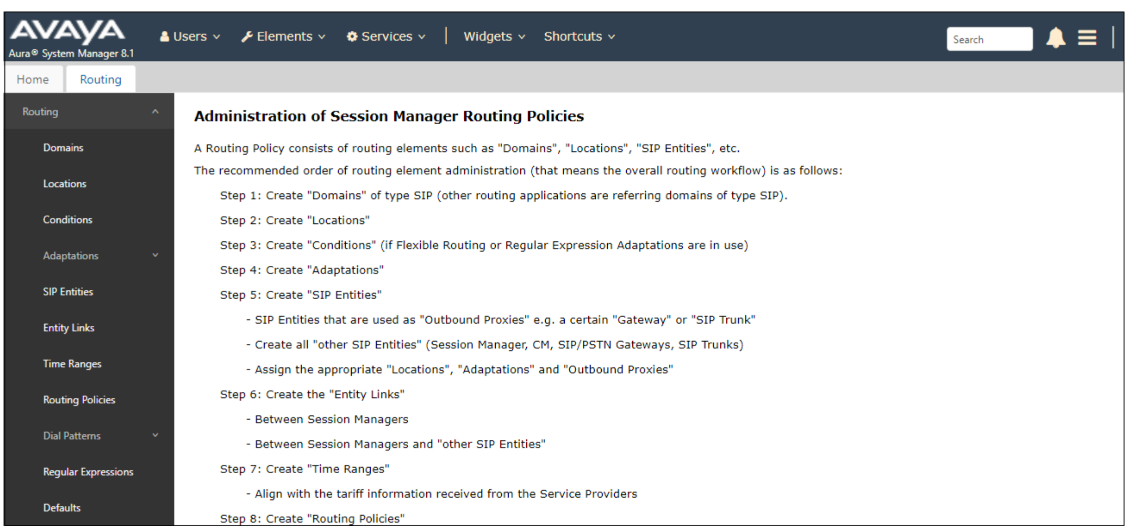
- Define a SIP Domain.
- Define a Location for Customer Premises Equipment (CPE).
- Configure the Adaptation Modules that will be associated with the SIP Entities for Communication Manager and the Avaya SBCE.
- Define SIP Entities corresponding to Session Manager, Communication Manager, the Avaya SBCE,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

6.1. System Manager Login and Navigation

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



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



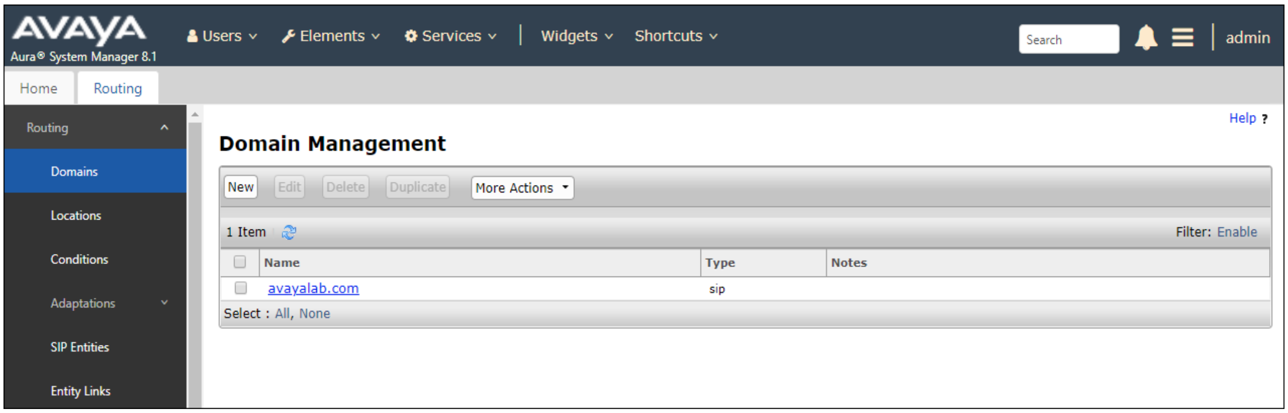
6.2. SIP Domain

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

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

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

Step 3 - Click **Commit** to save (not shown).



6.3. Locations

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

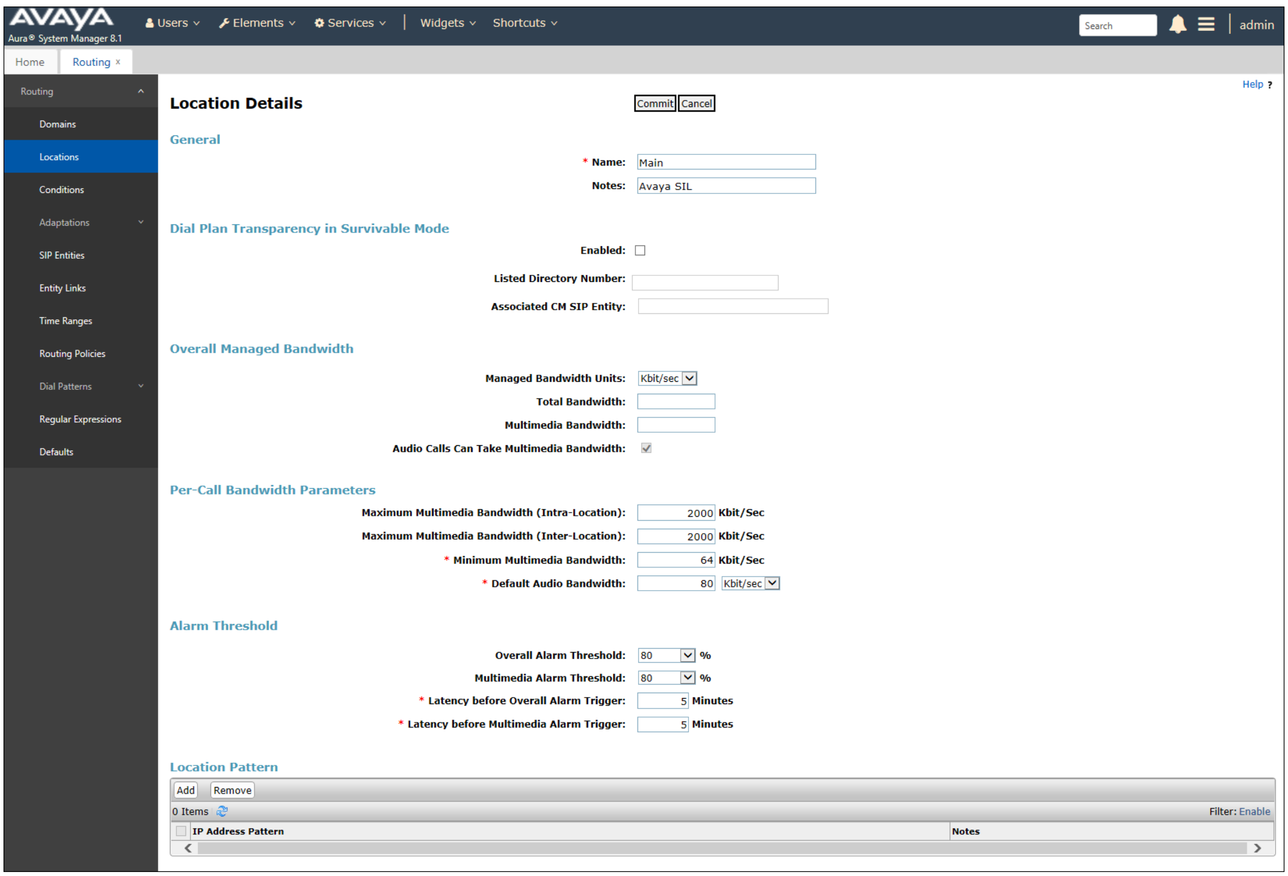
- **Main** – The customer site containing System Manager, Session Manager, Communication Manager, SIP endpoints, etc.
- **Common SBCs**– This site contains the Avaya SBCE.

6.3.1. Main Location

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

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

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



6.3.2. Common-SBCs Location

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

- **Name**: Enter a descriptive name for the Location (e.g., **Common-SBCs**).

6.4. Configure Adaptations

Session Manager can be configured to use Adaptation Modules to convert SIP headers sent to/from AT&T to Communication Manager.

- Inbound messages - Modification of SIP messages sent to Communication Manager extensions. (**Section 6.4.1**)
    - The AT&T called number digit string in the Request URI is replaced with the associated Communication Manager extensions defined for Agent skill queue VDNs/telephones.
- Outbound messages - Modification of SIP messages sent by Communication Manager extensions. (**Section 6.4.2**)
    - The History-Info header is removed automatically by the **AttAdapter**.
    - Avaya SIP headers not required by AT&T are removed.

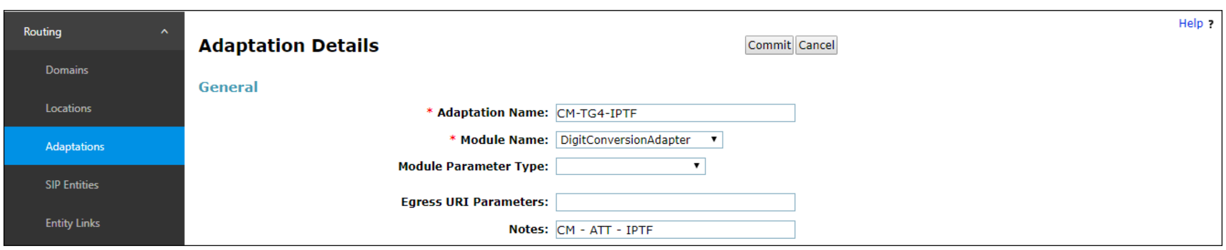
6.4.1. Adaptation for Avaya Aura® Communication Manager Extensions

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

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

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

1. A descriptive **Name**, (e.g., **CM-TG4-IPTF**).
2. Select **DigitConversionAdapter** from the **Module Name** drop down menu (if no module name is present, select **<click to add module>** and enter **DigitConversionAdapter**).



Step 3 - Scroll down to the **Digit Conversion for Outgoing Calls from SM** section (the inbound digits from AT&T that need to be replaced with their associated Communication Manager extensions before being sent to Communication Manager). 0000011041 is a DNIS string sent in the Request URI by the IPTF service that is associated with Communication Manager Agent/VDN skill queue 71041.

- Enter **0000011041** in the **Matching Pattern** column.
- Enter **10** in the **Min/Max** columns.
- Enter **6** in the **Delete Digits** column.
- Enter **7** in the **Insert Digits** column to convert the number to 71041, a Vector Directory Number (VDN) in Communication Manager.
- Specify that this should be applied to the SIP **destination** headers in the **Address to modify** column.
- Enter any desired notes.

Step 4 - Repeat **Step 3** for all additional IPTF DNIS numbers//Communication Manager extensions.

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

Digit Conversion for Outgoing Calls from SM

Add Remove

5 Items Filter: Enable

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
* 0000011041	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion
* 0000021042	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion
* 0000031043	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion
* 0000041044	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion
* 0000051045	* 10	* 10		* 6	7	destination		10 digit DNIS to VDN Conversion

Select : All, None

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

6.4.2. Adaptation for the AT&T IP Toll Free Service

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

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

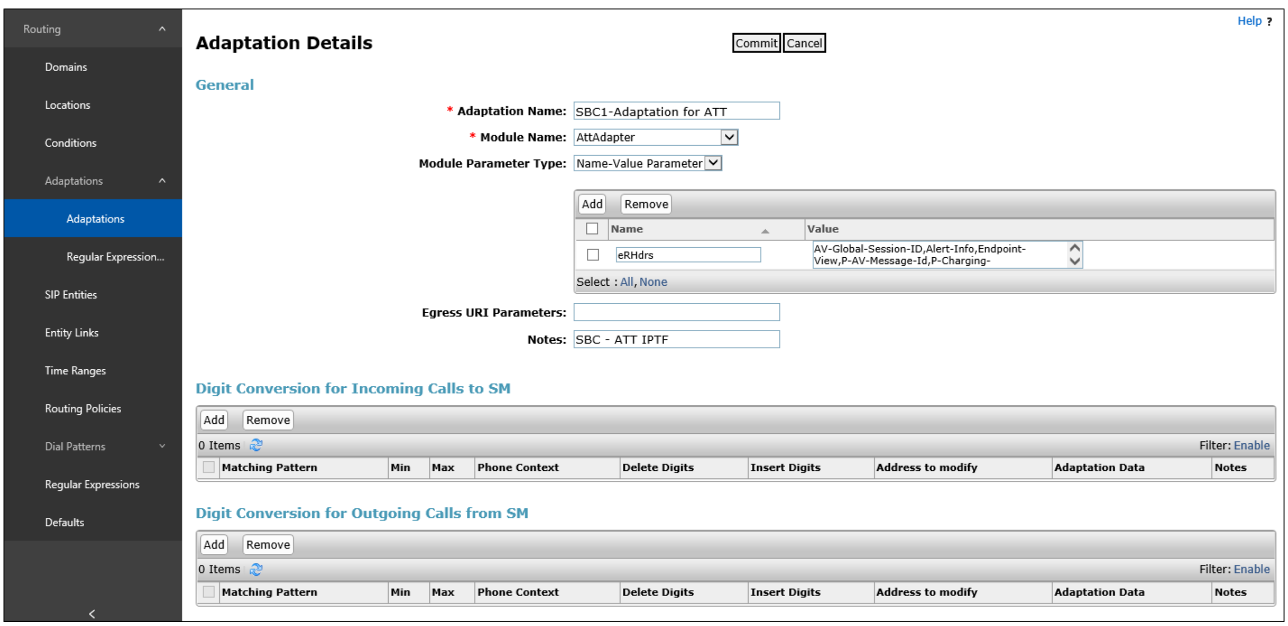
- A descriptive **Name**, (e.g., **SBC1-Adaptation for ATT**).
- Select **AttAdapter** from the **Module Name** drop down menu (if no module name is present, select **<click to add module>** and enter **AttAdapter**). The AttAdapter will automatically remove History-Info headers, (which the IPTF service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

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

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

- **Name** – Enter **eRHdrs**
- **Value** – Enter the following Avaya headers to be removed by Session Manager. Note that each header name is separated by a comma with no spaces in between. If spaces are used after the comma, the string needs to be enclosed in quotes:
  AV-Global-Session-ID,Alert-Info, Endpoint-View,P-AV-Message-Id,P-Charging-Vector,P-Location, AV-Correlation-ID,Av-Secure-Indication

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



6.5. SIP Entities

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

- Session Manager (**Section 6.5.1**). Note that this Entity is normally created during Session Manager installation but is shown here for completeness.
- Communication Manager for AT&T access (**Section 6.5.2**) – This entity, and its associated Entity Link (using TLS with port 5064, is for calls from the IPTF service to Communication Manager via the Avaya SBCE.
- Communication Manager for local access (**Section 6.5.3**) – This entity, and its associated Entity Link (using TLS with port 5061), is primarily used for traffic between Avaya SIP telephones and Communication Manager.
- Avaya SBCE (**Section 6.5.4**) – This entity, and its associated Entity Link (using TLS and port 5061), is for calls from the IPTF service via the Avaya SBCE.
- Experience Portal (**Section 6.5.5**) – 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 5064), and to the Avaya SBCE (port 5061). The connection between the Avaya SBCE and the AT&T IPTF service uses UDP/5060 per AT&T requirements.

6.5.1. Avaya Aura® Session Manager SIP Entity

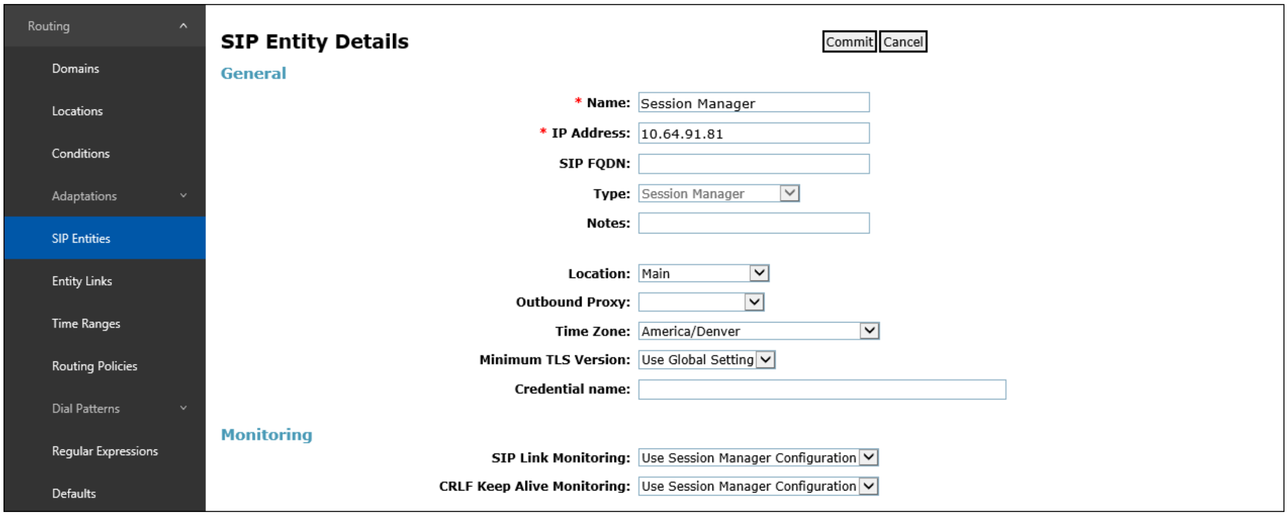
Step 1 - In the left pane under **Routing**, click on **SIP Entities**. In the **SIP Entities** page click on **New** (not shown).

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

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

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

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



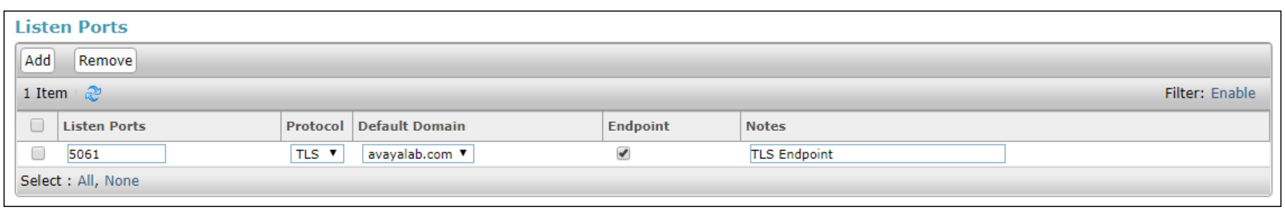
Step 4 - Scrolling down to the **Port** section of the **SIP Entity Details** page, click on **Add** and provision entries as follow:

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

Step 5 - Repeat **Step 4** to provision entries for any other listening ports used by Session Manager for SIP telephones. These are separate from the ports defined for the Entity Links in **Section 6.6**.

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

Step 7 - Click on **Commit**.



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

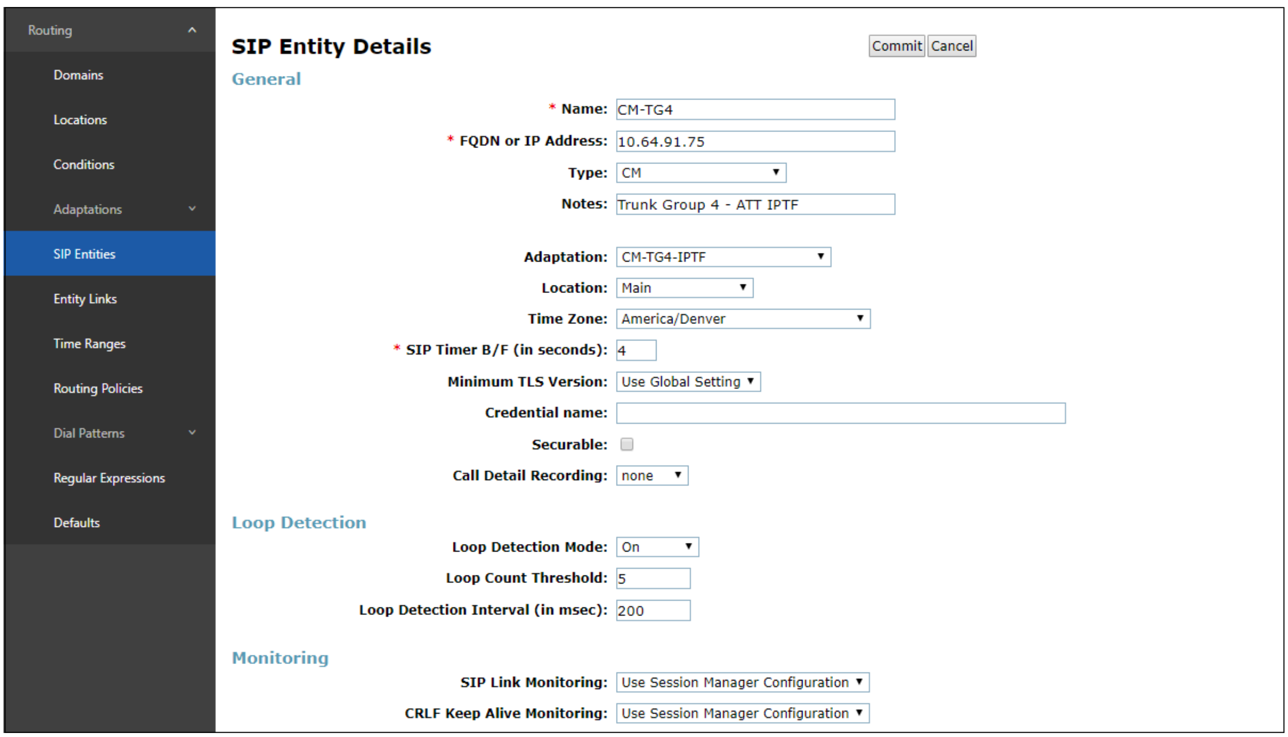
6.5.2. Avaya Aura® Communication Manager SIP Entity – Public Trunk

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

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

- **Name –** Enter a descriptive name (e.g., **CM-TG4**).
- **FQDN or IP Address –** Enter the IP address of Communication Manager Processor Ethernet (procr) described in **Sections 5.4** and **5.5** (e.g., **10.64.91.75**).
- **Type** – Select **CM**.
- **Adaptation –** Select the Adaptation **CM-TG4-IPTF** administered in **Section 6.4.1**.
- **Location** – Select a Location **Main** administered in **Section 6.3.1**.
- **Time Zone** – Select the time zone in which Communication Manager resides.
- In the **SIP Link Monitoring** section of the **SIP Entity Details** page select:
  - Select **Use Session Manager Configuration** for **SIP Link Monitoring** field and use the default values for the remaining parameters.

Step 3 - Click on **Commit**.



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

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

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

6.5.4. Avaya Session Border Controller for Enterprise SIP Entity

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

- **Name –** Enter a descriptive name (e.g., **SBCE-Toll Free**).
- **FQDN or IP Address –** Enter the IP address of the A1 (private) interface of the Avaya SBCE (e.g., **10.64.91.41**), see **Section 8.3**.
- **Type –** Select **SIP Trunk**.
- **Adaptations –** Select Adaptation **SBC1-Adaptation for ATT** (**Section 6.4.2**).
- **Location –** Select Location **Common-SBCs** administered in **Section 6.3.2**.

6.5.5. Avaya Aura® Experience Portal SIP Entity

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

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

6.6. Entity Links

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

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

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

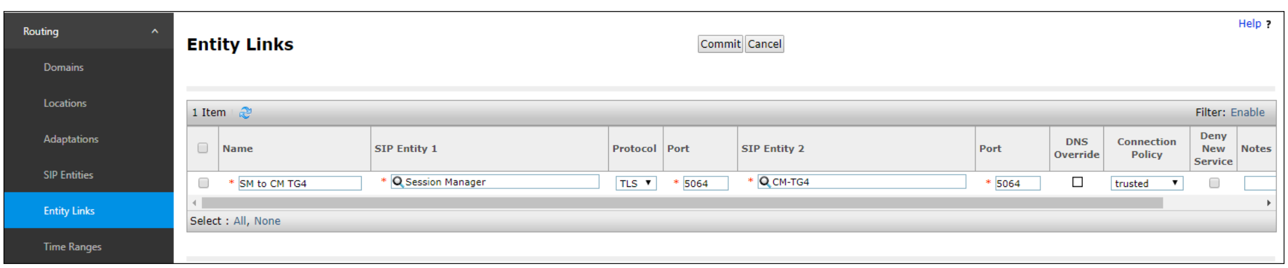
Note – See the information in **Section 6.5** regarding the transport protocols and ports used in the reference configuration.

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

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

- **Name** – Enter a descriptive name for this link to Communication Manager (e.g., **SM to CM TG4**).
- **SIP Entity 1** – Select the SIP Entity administered in **Section 6.5.1** for Session Manager (e.g., **Session Manager**).
- **SIP Entity 1 Port** – Enter **5064**.
- **Protocol –** Select **TLS** (see **Section 5.8.1**).
- **SIP Entity 2** –Select the SIP Entity administered in **Section 6.5.2** for the Communication Manager public entity (e.g., **CM-TG4**).
- **SIP Entity 2 Port –** Enter **5064** (see **Section 5.8.1**).
- **Connection Policy –** Select **trusted**.

Step 3 - Click on **Commit**.



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

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

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

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

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

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

6.6.4. Entity Link to Avaya Aura® Experience Portal

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

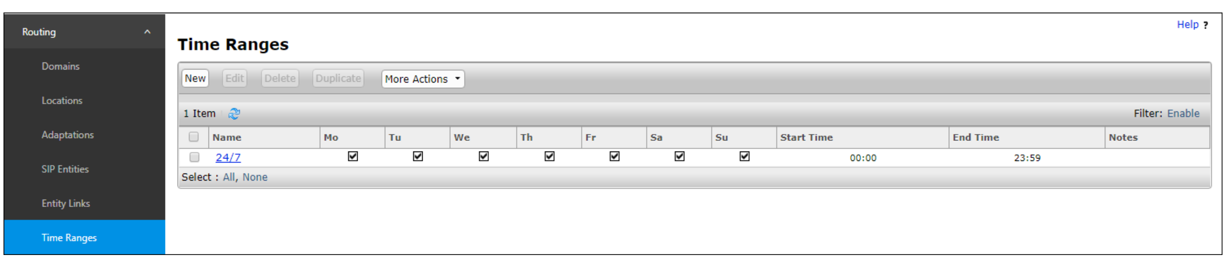
- **Name** – Enter a descriptive name for this link to Messaging (e.g., **SM to ExperiencePortal**).
- **SIP Entity 1 Port** – Enter **5061**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.5.5** for the Experience Portal entity (e.g., **ExperiencePortal**).
- **SIP Entity 2 Port**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

6.7. Time Ranges – (Optional)

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

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

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



6.8. Routing Policies

In this section, the following Routing Policies are administered:

- Inbound calls to Communication Manager extensions (**Section 6.8.1**).
- Inbound calls to Experience Portal (**Section 6.8.2**).

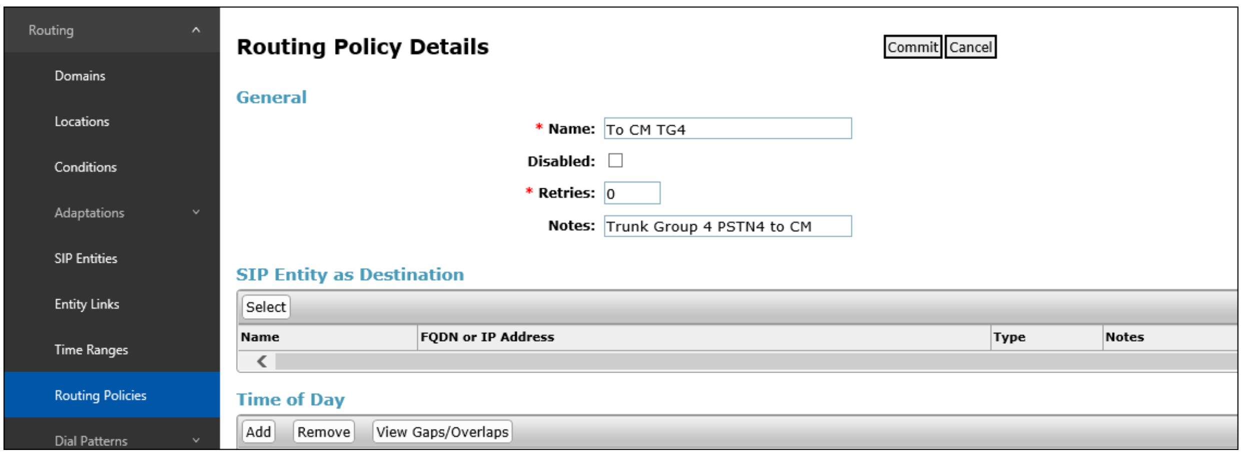
6.8.1. Routing Policy for AT&T Routing to Avaya Aura® Communication Manager

This Routing Policy is used for inbound calls from IPTF.

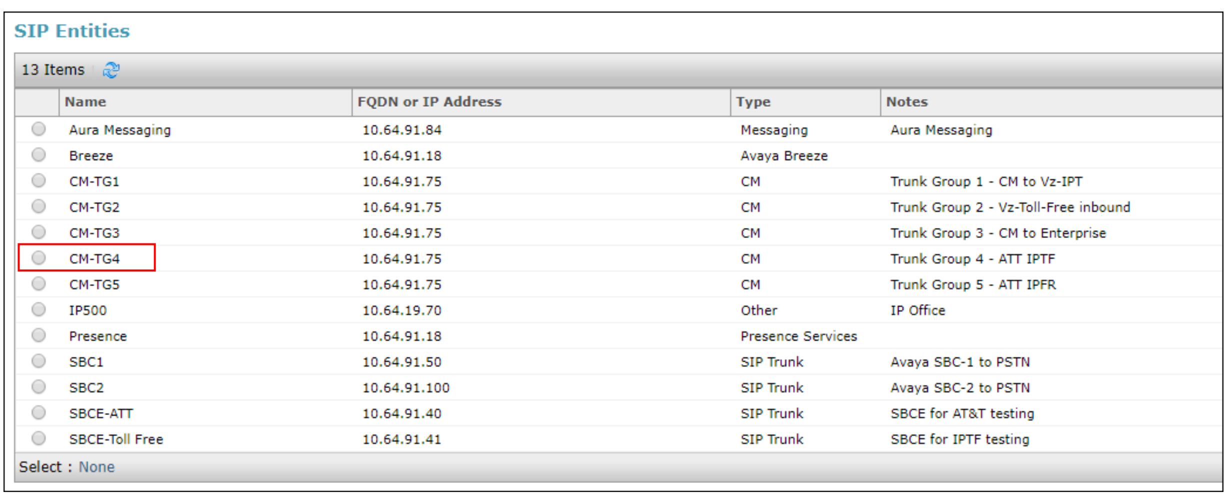
Step 1 - In the left pane under **Routing**, click on **Routing Policies**. In the **Routing Policies** page click on **New** (not shown).

Step 2 - In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing AT&T calls to Communication Manager (e.g., **To CM TG4**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.

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



Step 4 - In the **SIP Entities List** page, select the SIP Entity administered in **Section 6.5.2** for the Communication Manager public SIP Entity (**CM-TG4**), and click on **Select**.



SIP Entities

13 Items

Name	FQDN or IP Address	Type	Notes
Aura Messaging	10.64.91.84	Messaging	Aura Messaging
Breeze	10.64.91.18	Avaya Breeze	
CM-TG1	10.64.91.75	CM	Trunk Group 1 - CM to Vz-IPT
CM-TG2	10.64.91.75	CM	Trunk Group 2 - Vz-Toll-Free inbound
CM-TG3	10.64.91.75	CM	Trunk Group 3 - CM to Enterprise
CM-TG4	10.64.91.75	CM	Trunk Group 4 - ATT IPTF
CM-TG5	10.64.91.75	CM	Trunk Group 5 - ATT IPFR
IP500	10.64.19.70	Other	IP Office
Presence	10.64.91.18	Presence Services	
SBC1	10.64.91.50	SIP Trunk	Avaya SBC-1 to PSTN
SBC2	10.64.91.100	SIP Trunk	Avaya SBC-2 to PSTN
SBCE-ATT	10.64.91.40	SIP Trunk	SBCE for AT&T testing
SBCE-Toll Free	10.64.91.41	SIP Trunk	SBCE for IPTF testing

Select : None

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

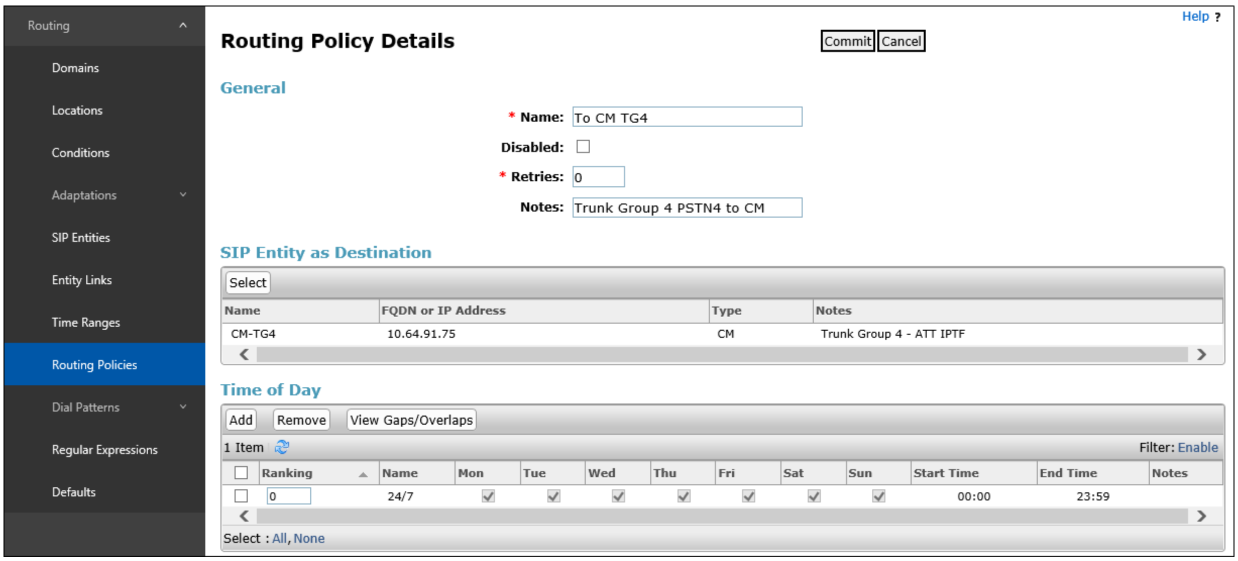
Step 6 - In the **Time Range List** page (not shown), check the checkbox(s) corresponding to one or more Time Ranges administered in **Section 6.7**, and click on **Select**.

Step 7 - Returning to the **Routing Policy Details** page in the **Time of Day** section, enter a **Ranking** of **0**.

Step 8 - No **Regular Expressions** were used in the reference configuration.

Step 9 - Click on **Commit**.

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



6.8.2. Routing Policy for Inbound Calls to Experience Portal

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

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

6.9. Dial Patterns

In this section, the following task are administered:

- Origination Dial Pattern for inbound calls arriving from the local area code.
- Dial Pattern for inbound PSTN calls via the IPTF service to Communication Manager.
- Dial Pattern for inbound PSTN calls via the IPTF service to Experience Portal.

6.9.1. Origination Dial Patterns – (Optional)

One of the routing enhancements in Session Manager release 8.1 is the addition of Origination Dial Patterns functionality. This configuration is optional. Origination Dial Pattern sets can be created to include digits patterns, which are matched by Session Manager to make more granular routing decisions, allowing the use of different routes for calls arriving to Session Manager from the same Originating Location. This is done by matching the number present in the From header of the incoming INVITE. More information can be found on [**2**] on the References section if necessary.

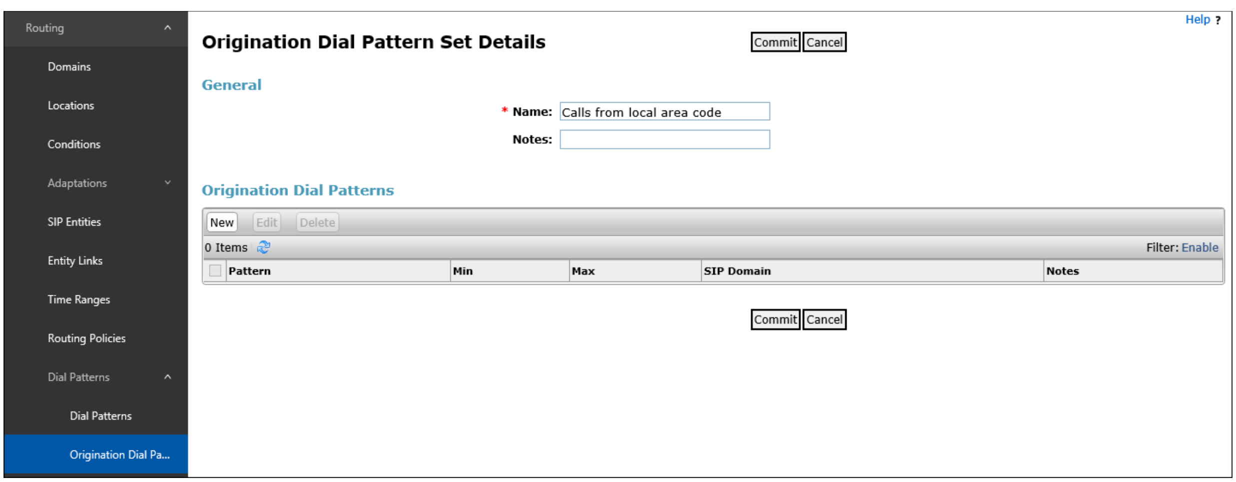
In the reference configuration, an Origination Dial Pattern set was created to route inbound calls originating from the local area code to Experience Portal, while calls from other area codes are routed to Communication Manager.

Note: To enable the use of Origination Dial Patterns, **Enable Flexible Routing** needs to be checked, under **Elements → Session Manager → Global Settings**.

Step 1 - In the left pane under **Routing**, expand the **Dial Patterns** tab. Select **Origination Dial Patterns Sets** and click on **New** (not shown).

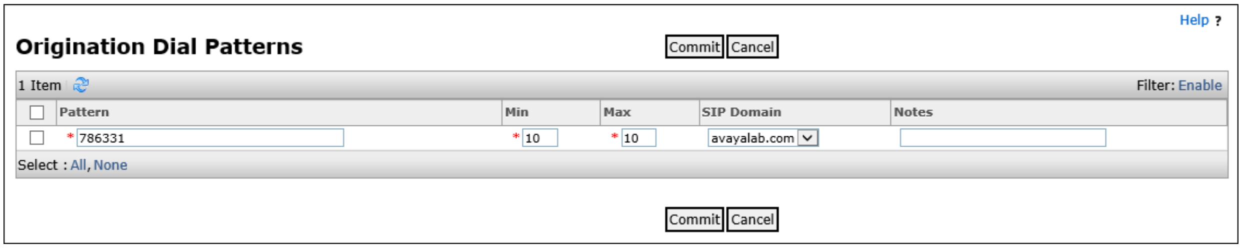
Step 2 - In the **General** section of the **Origination Dial Pattern Set Details** page, enter a descriptive name (e.g., **Calls from local area code**).

Step 3 - In the **Origination Dial Patterns** section, click on **New**.

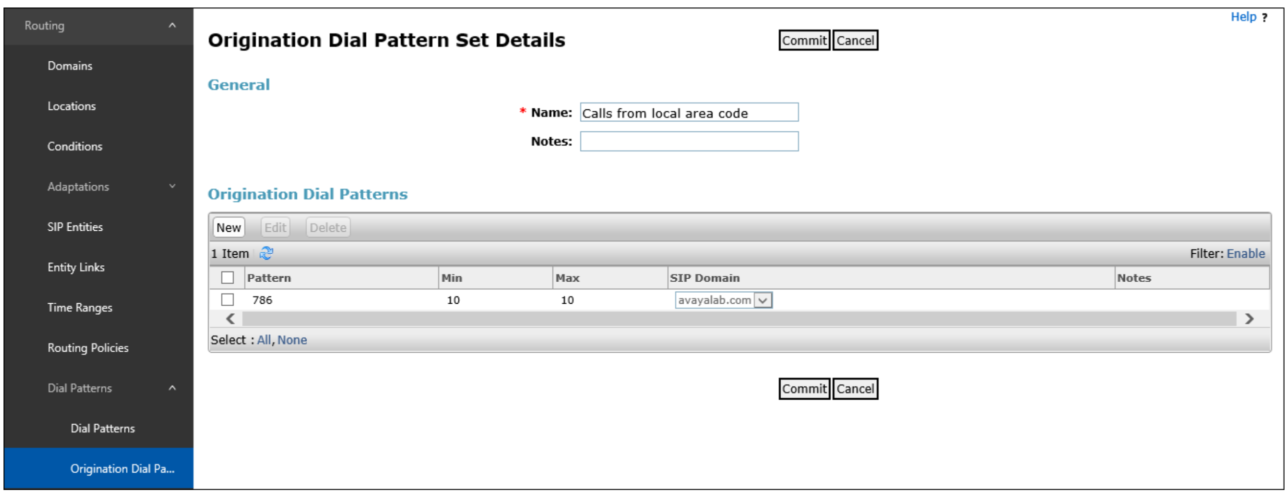


Step 3 - In the **Origination Dial Patterns** page, provision the following:

- **Pattern** – Enter **786**, the starting digits corresponding to the local area code.
- **Min** and **Max** – Enter **10**.
- **SIP Domain** – Select the enterprise SIP domain, e.g., **avayalab.com**.
- Click on **Commit**.



Step 4 – Back at the **Origination Dial Pattern Set Details** page, click on **Commit**.



6.9.2. Dial Pattern for Inbound Calls to Communication Manager

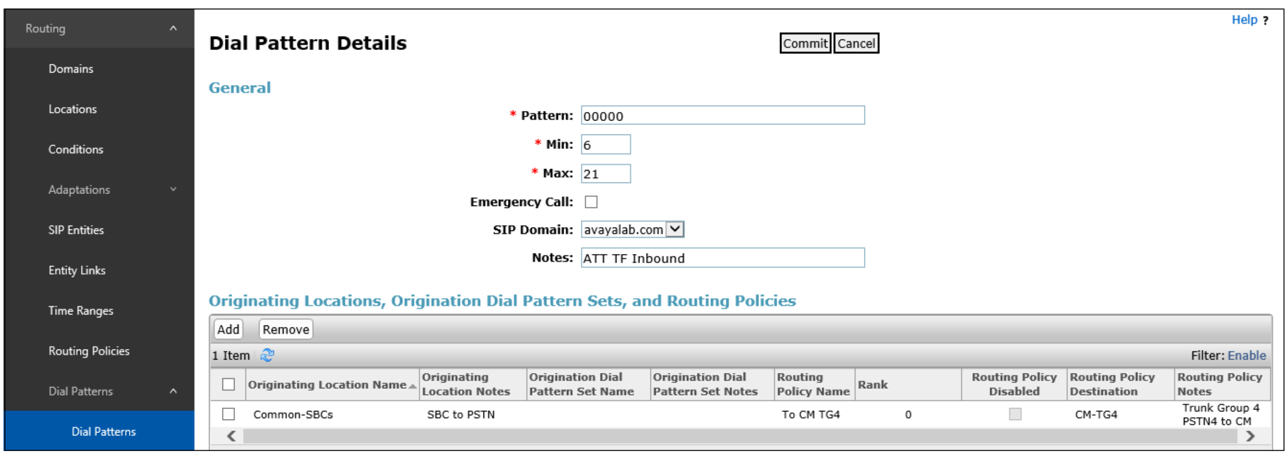
Note – In the reference configuration inbound calls from the IPTF service sent 10 DNIS digits in the SIP Request URI. Be sure to match on the digit string specified in the AT&T Request URI, not the digit string of the number dialed. They may be different.

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

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

- **Pattern** – In the reference configuration, AT&T sends a 10-digit number in the Request URI with the format 00000xxxxx. Enter **00000**.
- **Min** – Enter **6**.
- **Max** – Enter **21**.
- **SIP Domain** – Select the enterprise SIP domain, e.g., **avayalab.com**.

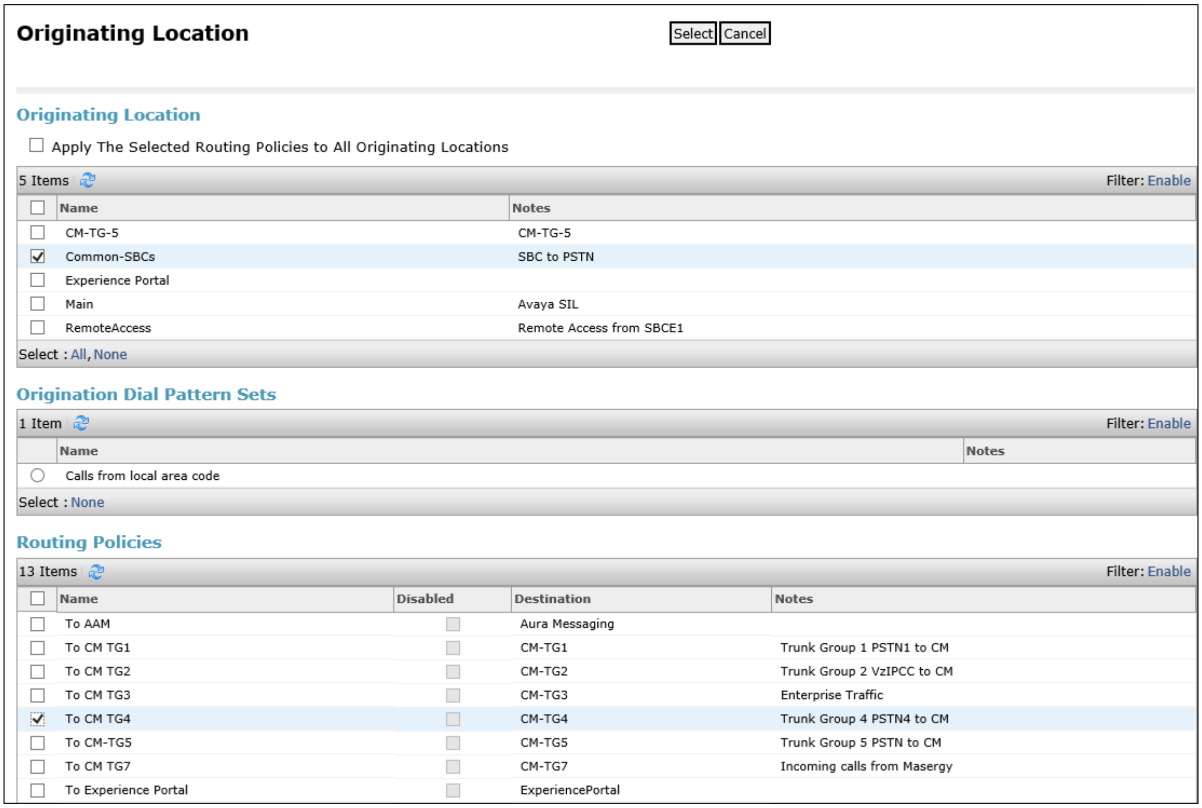
Note – The Adaptation defined for Communication Manager in **Section 6.4.1** will convert the various 00000xxxxx numbers into their corresponding Communication Manager extensions.



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

Step 4 - In the **Originating Location** section of the **Originating Locations, Origination Dial Pattern Sets and Routing Policies** page, check the checkbox corresponding to the location assigned to the Avaya SBCE in **Section 6.3.2**, e.g., **Common-SBCs**.

Step 5 - In the **Routing Policies** section, check the checkbox corresponding to the Routing Policy administered for routing calls to the Communication Manager public trunk in **Section 6.8.1** (e.g., **To CM TG4**). Click on **Select** (not shown).



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

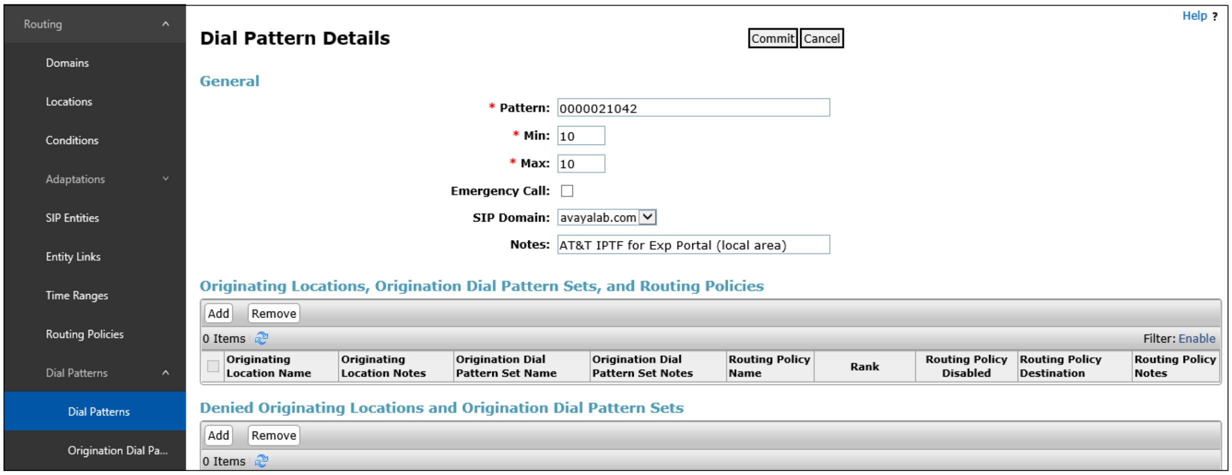
Step 7 - Repeat **Steps 1-6** for any additional inbound dial patterns from AT&T to Communication Manager.

6.9.3. Dial Pattern for Inbound Calls to Experience Portal

In the reference configuration, one the AT&T IPTF numbers, corresponding to DNIS 0000021042, was assigned for inbound calls to Experience Portal.

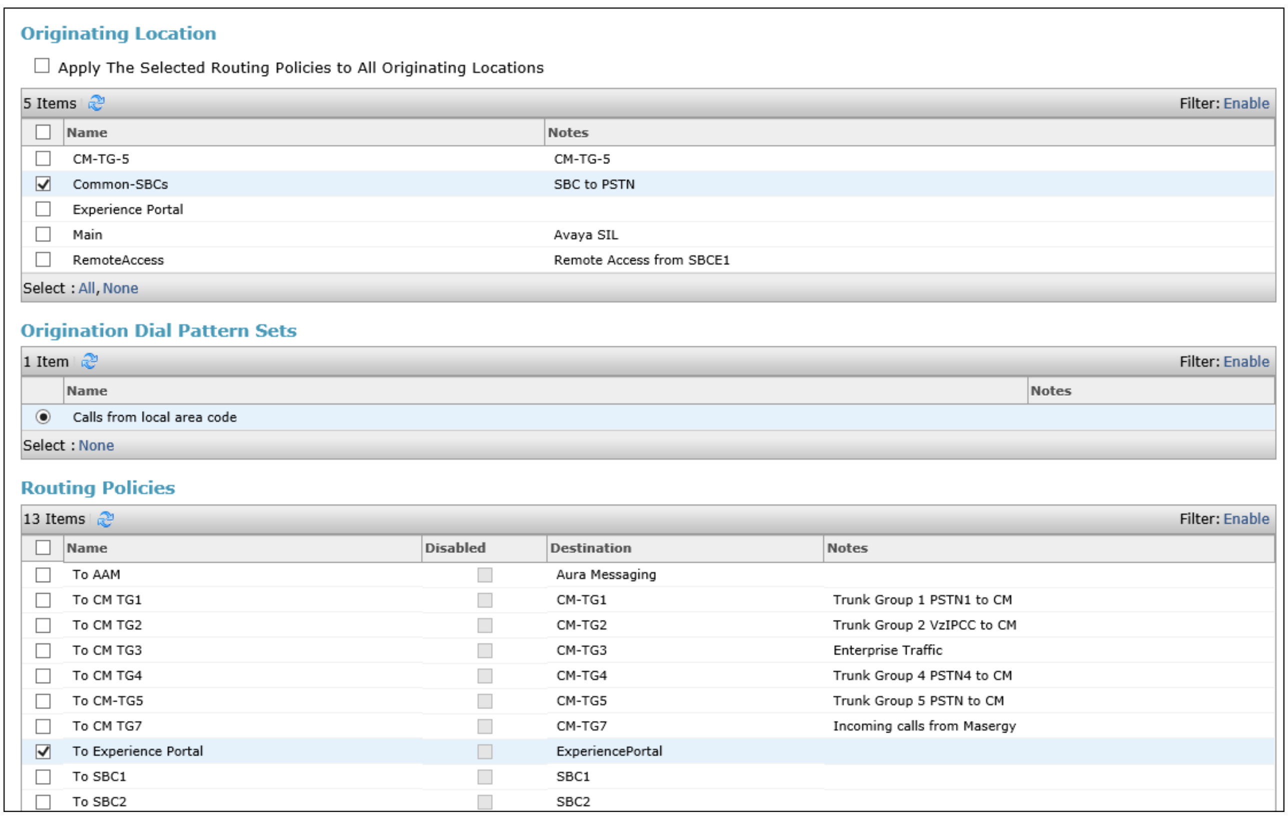
Step 1 - In the **General** section of the **Dial Pattern Details** page, repeat the steps shown in **Section 6.9.2**, with the following changes:

- **Pattern** – Enter the DNIS digits corresponding to the AT&T IPTF number assigned for calls to Experience Portal (e.g., **0000021042**).
- **Min** – Enter **10**.
- **Max** – Enter **10**



Step 2 – On the **Originating Locations, Origination Dial Patterns Sets and Routing Policies** page, repeat the steps shown in **Section 6.9.2** with the following addition:

- Check the checkbox for the Origination Dial pattern Set corresponding to calls from the local area code, defined in **Section 6.9.1** (e.g., **Calls from local area code**).



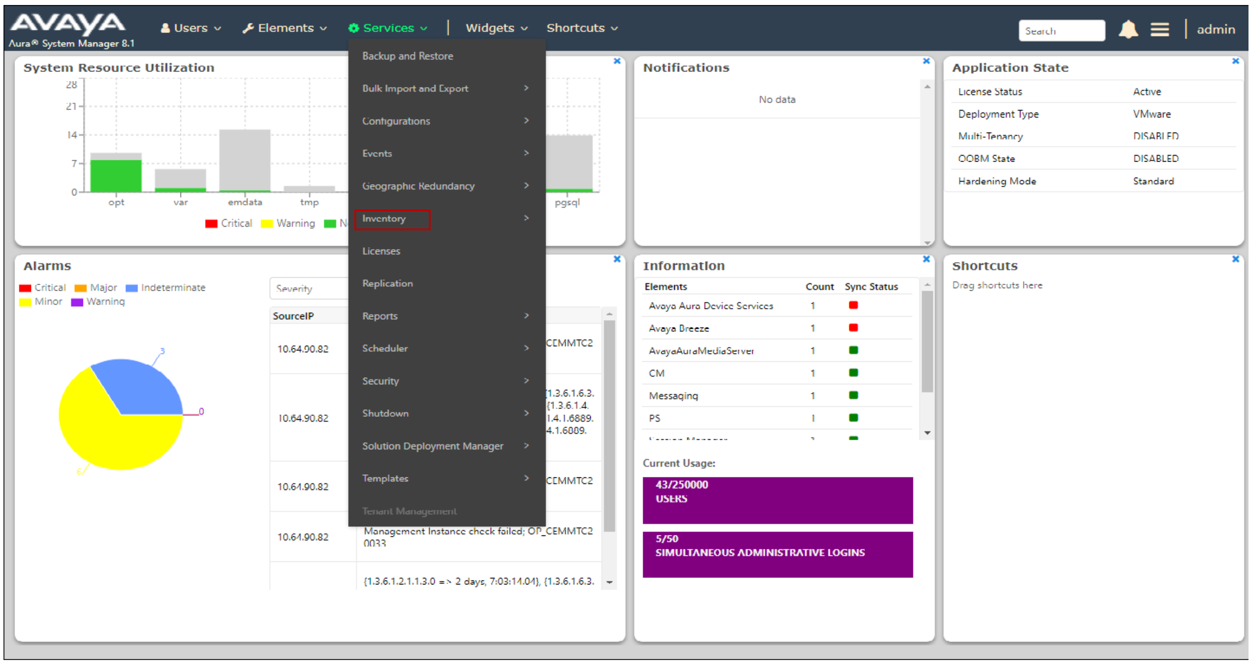
With this configuration, calls to this IPTF number originating from the local area code will be routed to Experience Portal, while calls to this same number originating from area codes other than the local area will still be routed to Communication Manager, following the dial pattern shown previously in **Section 6.9.2**.

6.10. Verify TLS Certificates – Session Manager

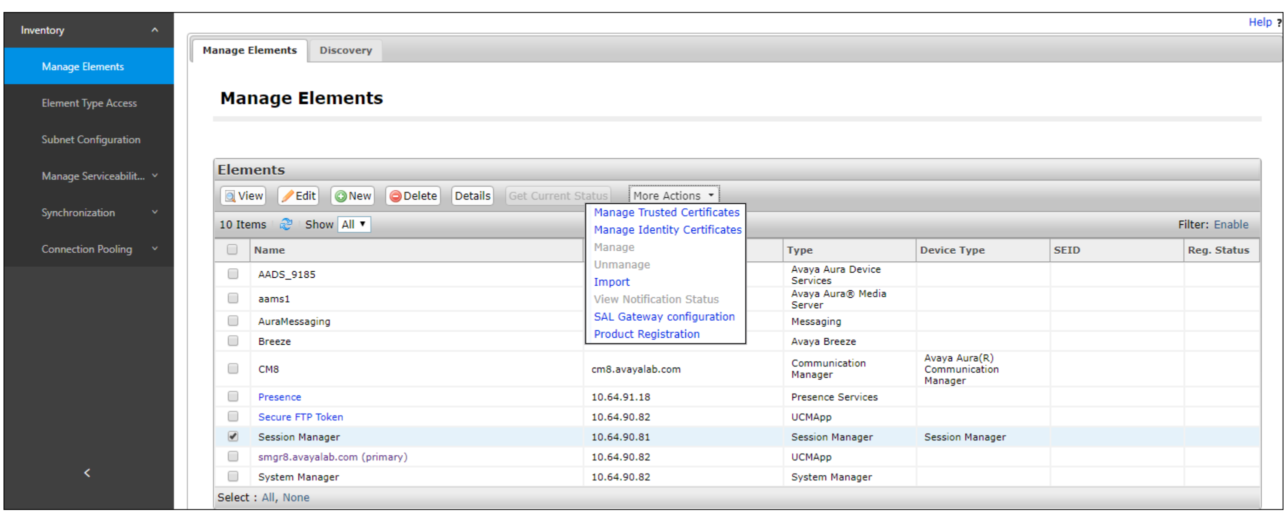
Note – Testing was done with System Manager signed identity certificates. The procedure to obtain and install certificates is outside the scope of these Application Notes.

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

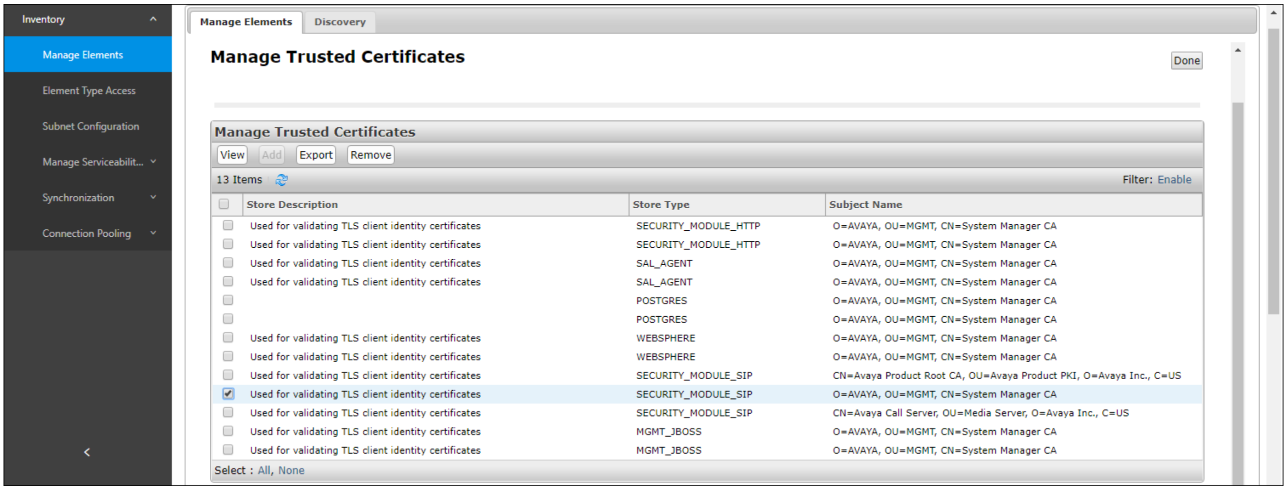
Step 1 - From the **Home** screen, under the **Services** heading, select **Inventory**.



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

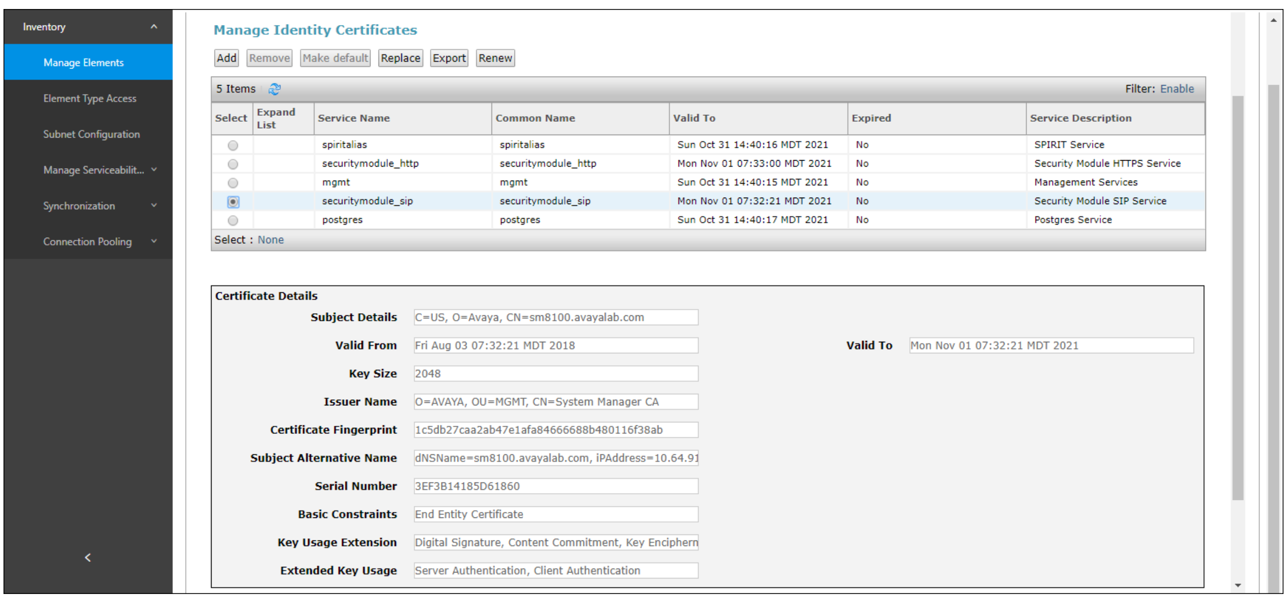


Step 3 - Verify the System Manager Certificate Authority certificate is listed in the trusted store, **SECURITY_MODULE_SIP**. Click **Done** to return to the previous screen.



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

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



7. Configure Avaya Aura® Experience Portal

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

7.1. Background

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

References to the Voice XML and/or Call Control XML applications are administered on Experience Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Experience Portal, the called party DNIS number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match is found, Experience Portal informs the caller that the call cannot be handled, and disconnects the call[2].

For the sample configuration described in these Application Notes, a simple VXML test application was used to exercise various SIP call flow scenarios with the AT&T IPTF 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.

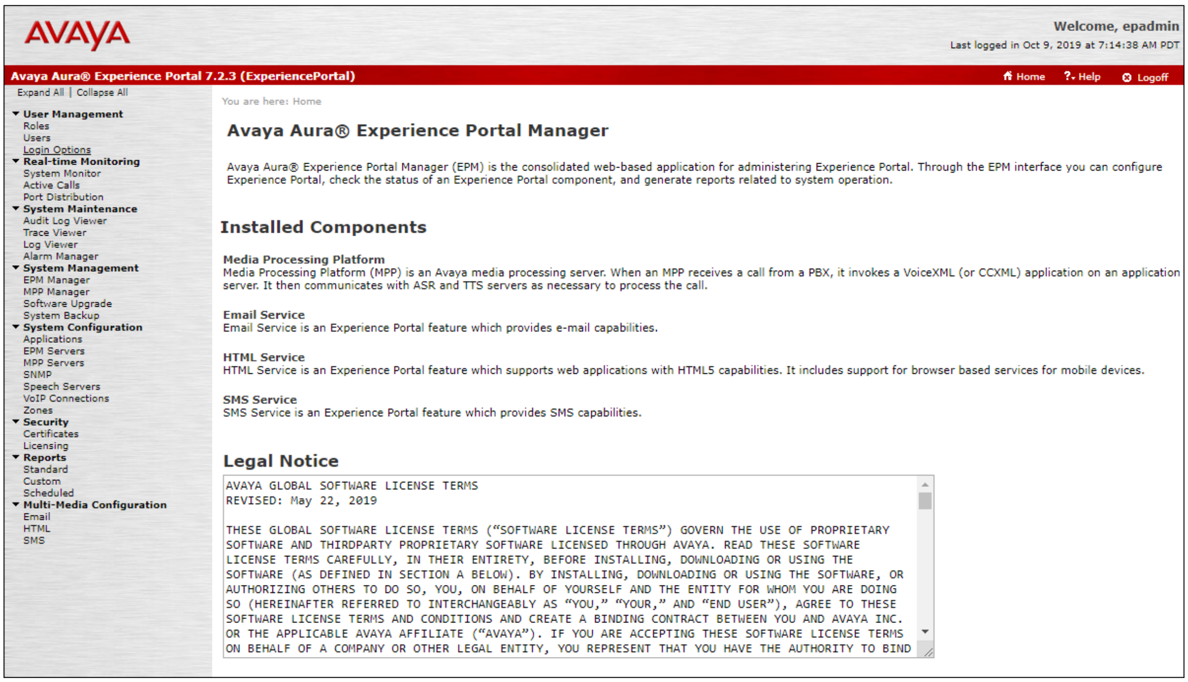
[2] An application may be configured with "inbound default" as the called number, to process all inbound calls that do not match any other application references.

7.2. Logging In and Licensing

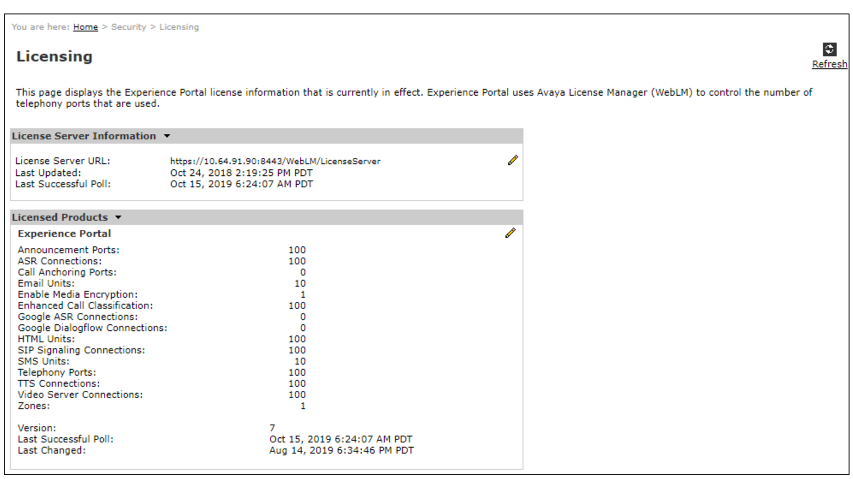
This section describes the steps on Experience Portal for administering a SIP connection to the Session Manager.

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

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



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

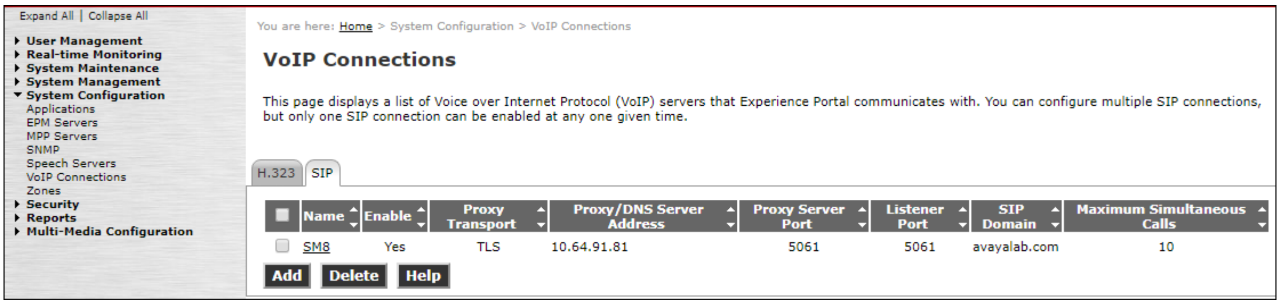


7.3. VoIP Connection

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

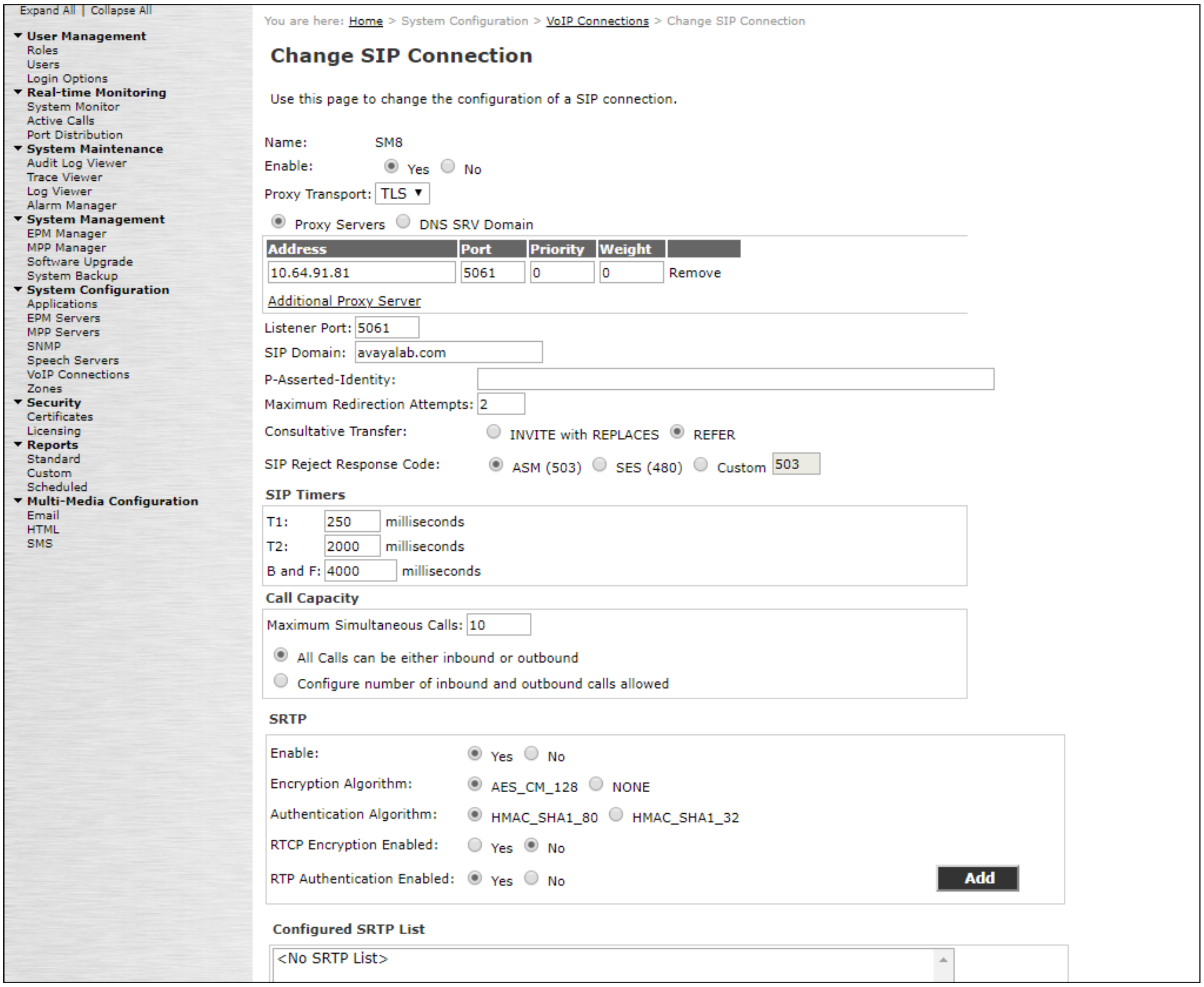
Step 1 - In the left pane, navigate to **System Configuration→VoIP Connections**. On the **VoIP Connections** page, select the **SIP** tab and click **Add** to add a SIP trunk.

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



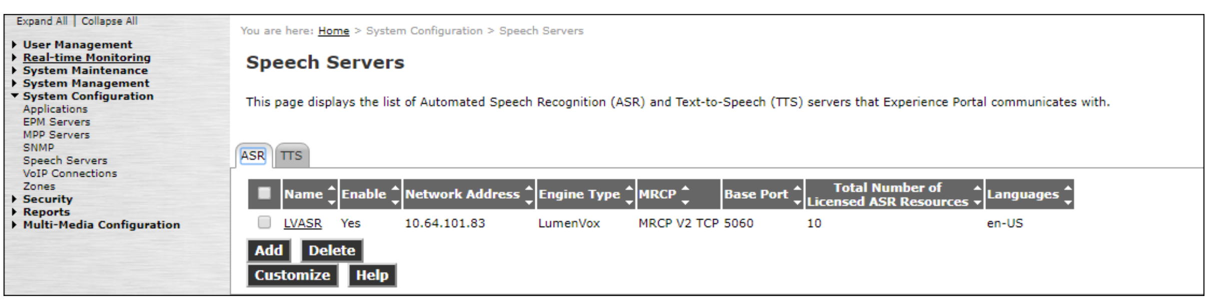
Step 2 - Configure a SIP connection as follows:

- **Name** – Set to a descriptive name (e.g., **SM8**).
- **Enable** – Set to **Yes**.
- **Proxy Server Transport** – Set to **TLS**.
- Select **Proxy Servers**, and enter:
  - **Proxy Server Address** = **10.64.91.81** (the IP address of the Session Manager signaling interface defined in **Section 6.5.1**).
  - **Port** = **5061**
  - **Priority** = **0** (default)
  - **Weight** = **0** (default)
- **Listener Port** – Set to **5061**.
- **SIP Domain** – Set to **avayalab.com** (**Section 6.2**).
- **Consultative Transfer** – Select **REFER**.
- **SIP Reject Response Code** – Select **ASM (503)**.
- **Maximum Simultaneous Calls** – Set to a number in accordance with licensed capacity. In the reference configuration a value of **10** was used.
- Select **All Calls can be either inbound or outbound**.
- **SRTP Enable** = **Yes**
- **Encryption Algorithm** = **AES_CM_128**
- **Authentication Algorithm** = **HMAC_SHA1_80**
- **RTCP Encryption Enabled** = **No**
- **RTP Authentication Enabled** = **Yes**
- Use default values for all other fields.
- Click **Save**.



7.4. Speech Servers

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

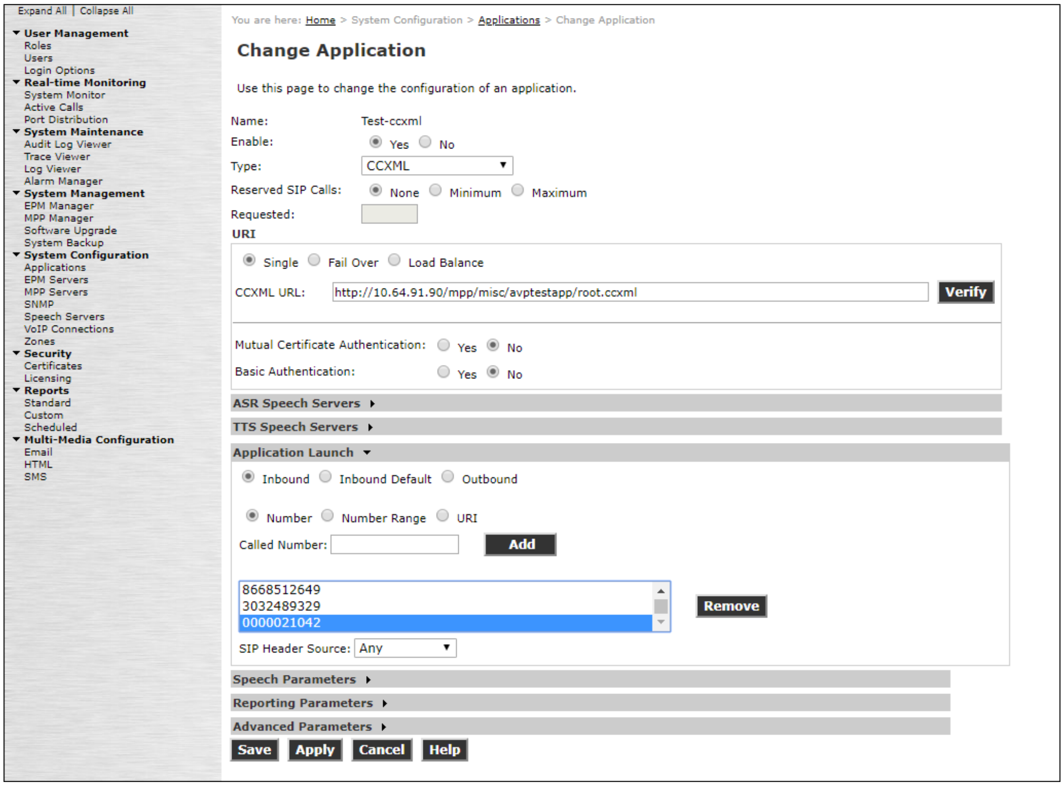


7.5. Application References

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

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

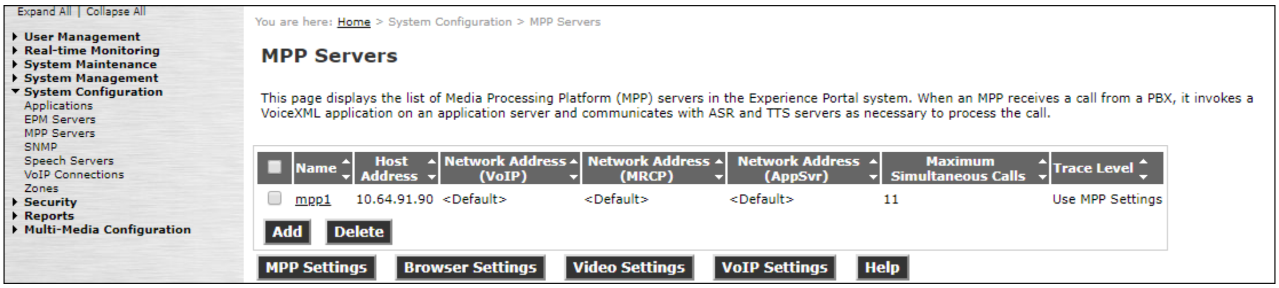
- **Name** – Set to a descriptive name (e.g., **Test-ccxml**).
- **Enable** – Set to **Yes**. This field determines which application(s) will be executed based on their defined criteria.
- **Type** – Select **VoiceXML**, **CCXML**, or **CCXML/VoiceXML** according to the application type.
- **VoiceXML** and/or **CCXML URL** – Enter the necessary URL(s) to access the VXML and/or CCXML application(s) on the application server. In the sample screen below, the Experience Portal test application on a single server is referenced.
- **ASR and TTS Speech Servers** – Select the appropriate ASR and/or TTS servers as necessary.
- **Application Launch** – Set to **Inbound**.
- **Called Number** – Enter the number to match against an inbound SIP INVITE message, and click **Add**. In the sample configuration illustrated in these Application Notes, the IPTF DNIS number 0000021042 was used (**Section 6.9.3**). Repeat to define additional called party numbers as needed. Inbound AT&T IPTF calls with these called party numbers will be handled by the application defined in this section.



7.6. MPP Servers and VoIP Settings

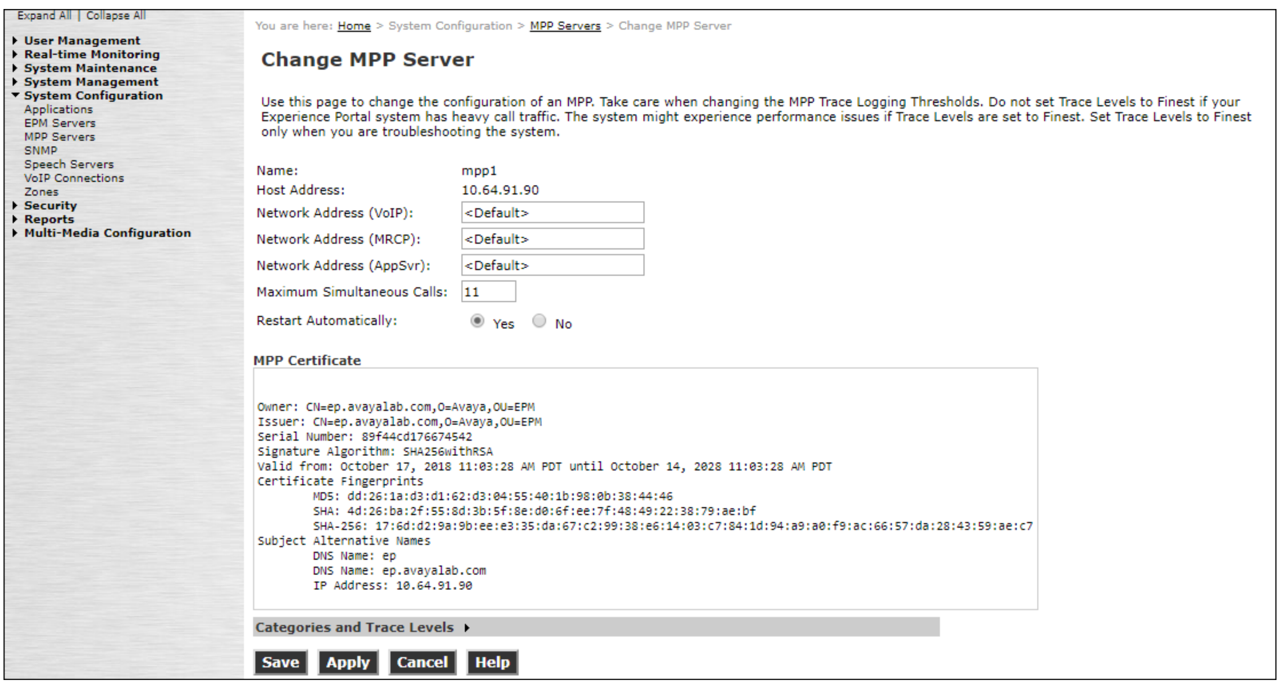
This section illustrates the procedure for viewing or changing the MPP Settings. In the sample configuration, the MPP Server is co-resident on a single server with the Experience Portal Management server (EPM).

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



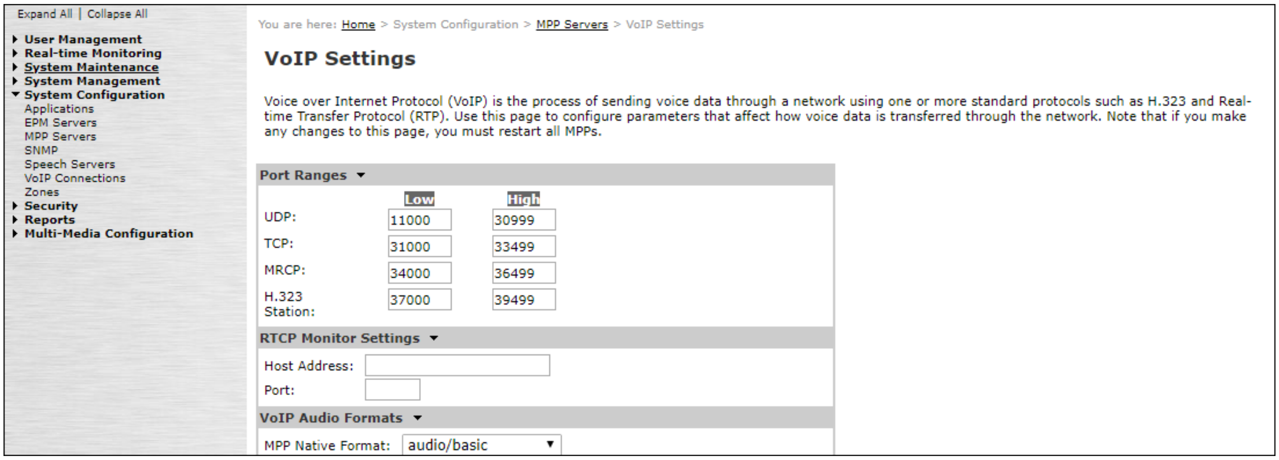
Step 2 - Enter any descriptive name in the **Name** field (e.g., **mpp1**) and the IP address of the MPP server in the **Host Address** field and click **Continue** (not shown).

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



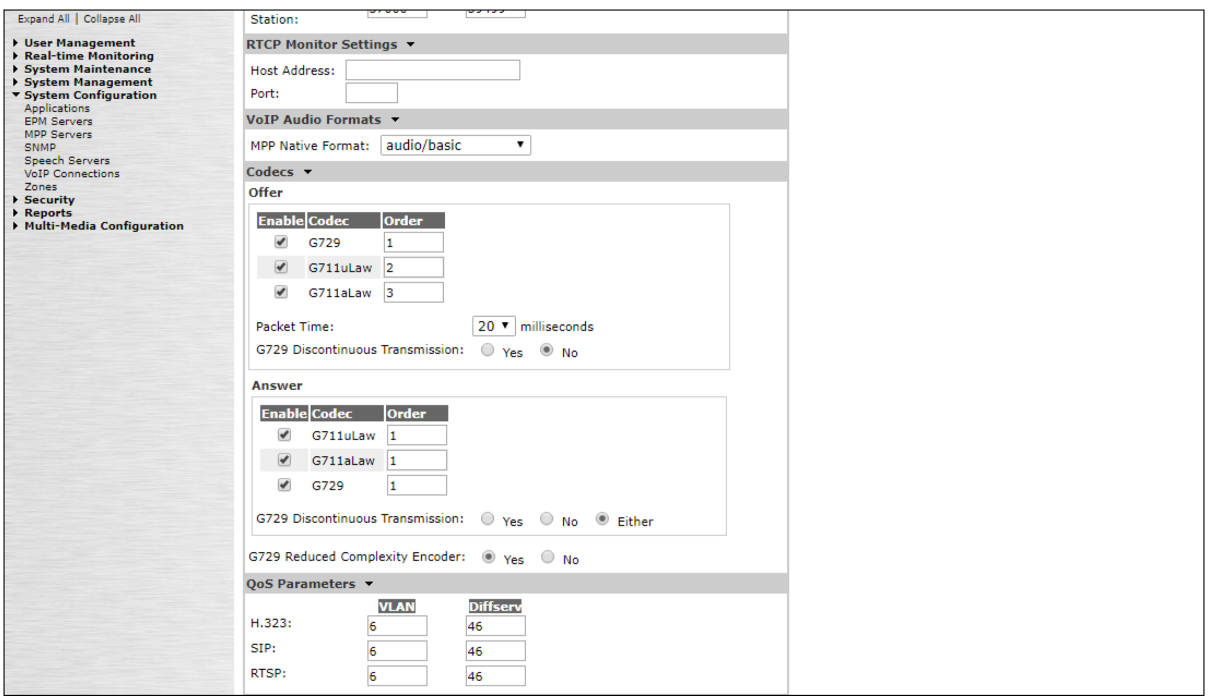
Step 4 - Click **VoIP Settings** tab on the screen displayed in **Step 1**, and the following screen is displayed.

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



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

Step 5 - Click on **Save**.



7.7. Configuring RFC2833 Event Value Offered by Experience Portal

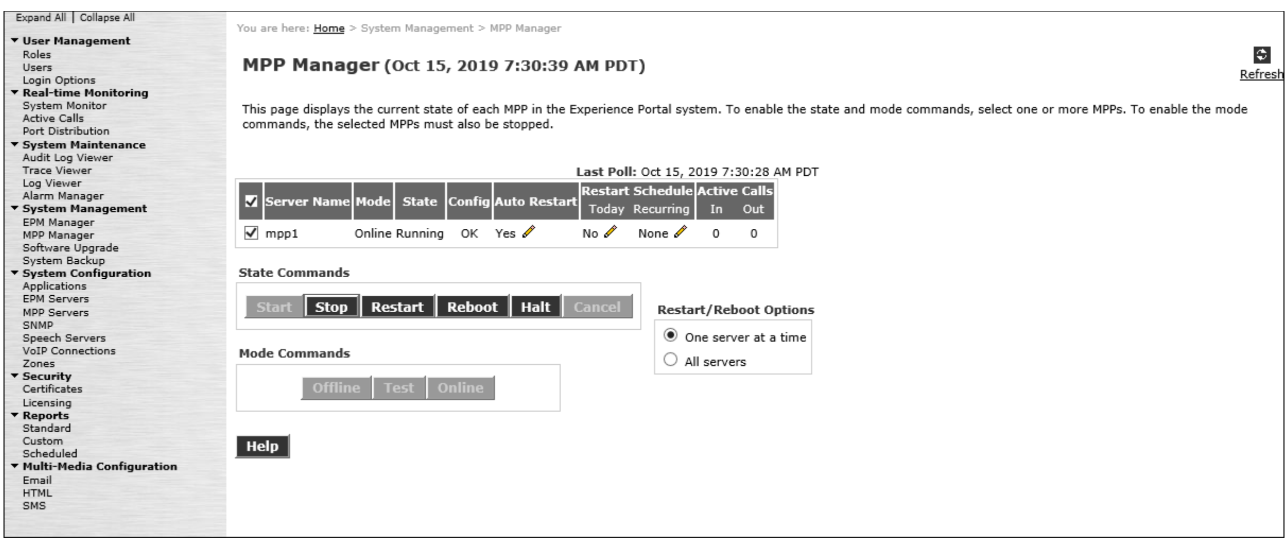
For incoming calls from AT&T IPTF services to Experience Portal, AT&T specifies the value 100 for the RFC2833 telephone-events that signal DTMF digits entered by the user. When Experience Portal answers, the SDP from Experience Portal matches this offered value.

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

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

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

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



8. Configure Avaya Session Border Controller for Enterprise

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

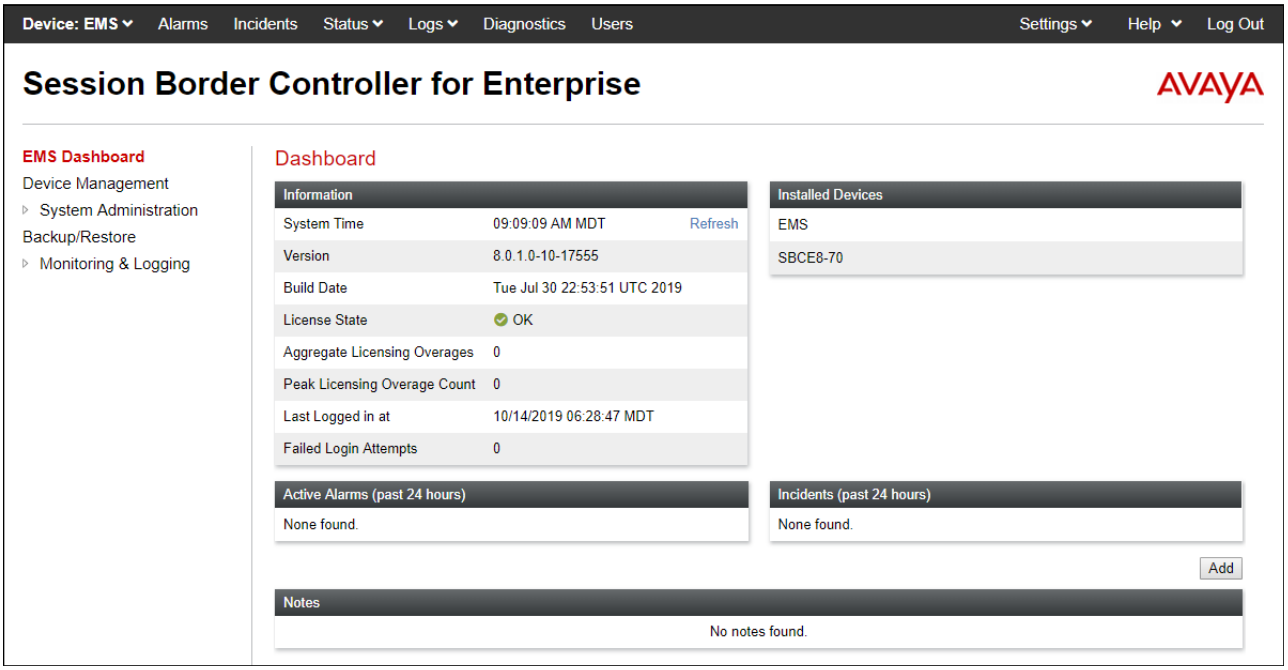
Note: The installation and initial provisioning of the Avaya SBCE is beyond the scope of this document. Refer to **[11]** and **[12]** in the References section for additional information.

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

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

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

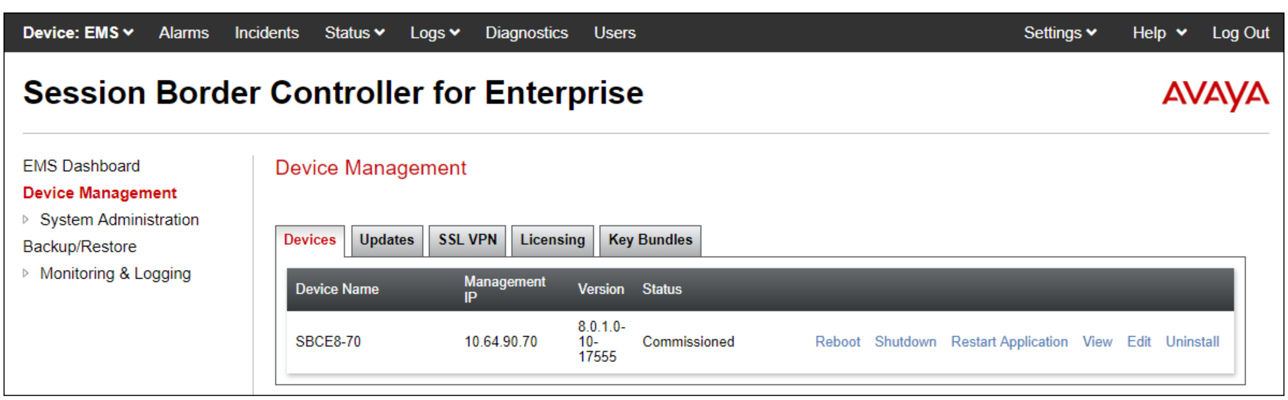
Note – The provisioning described in the following sections use the menu options listed in the left-hand column shown below.



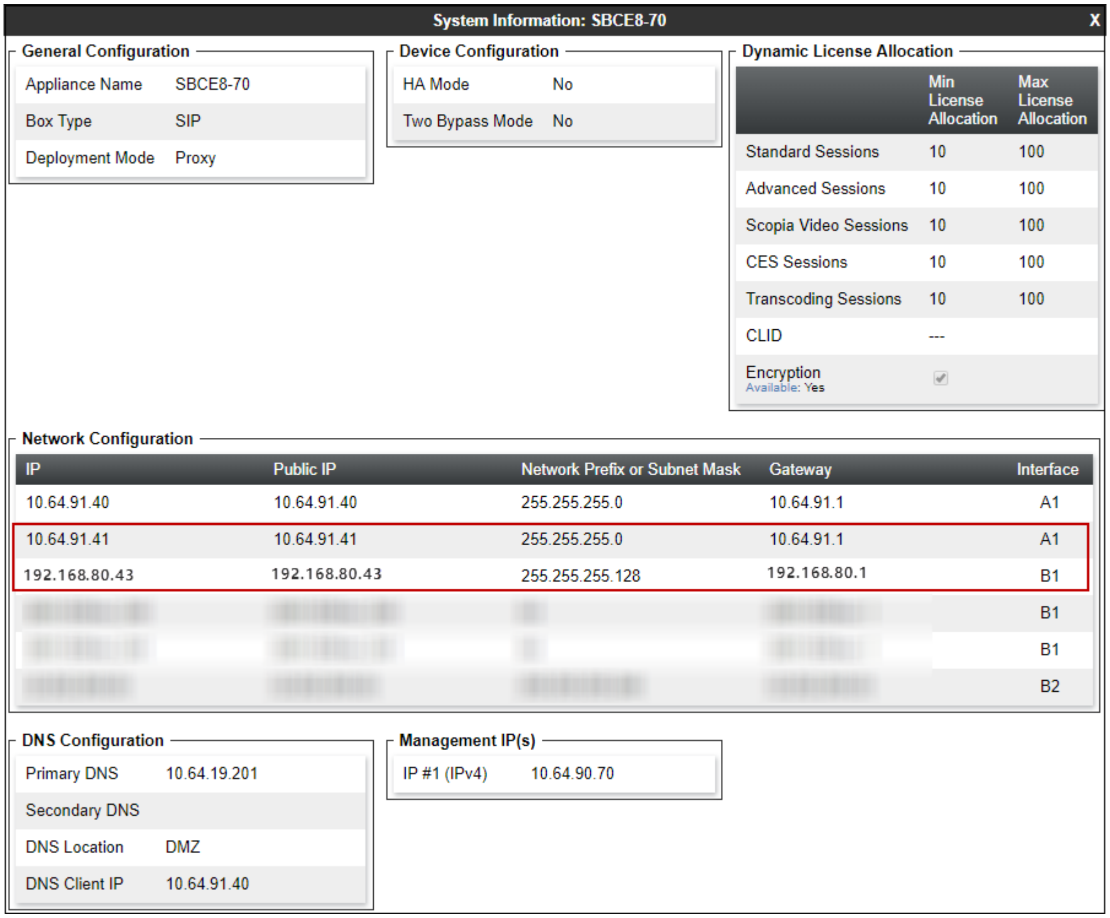
8.1. Device Management – Status

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

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



Step 2 - Click on **View** to display the **System Information** screen. The screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**. In the shared test environment, the highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to AT&T.



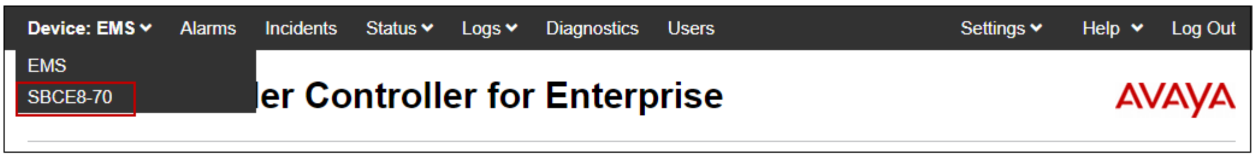
8.2. TLS Management

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

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

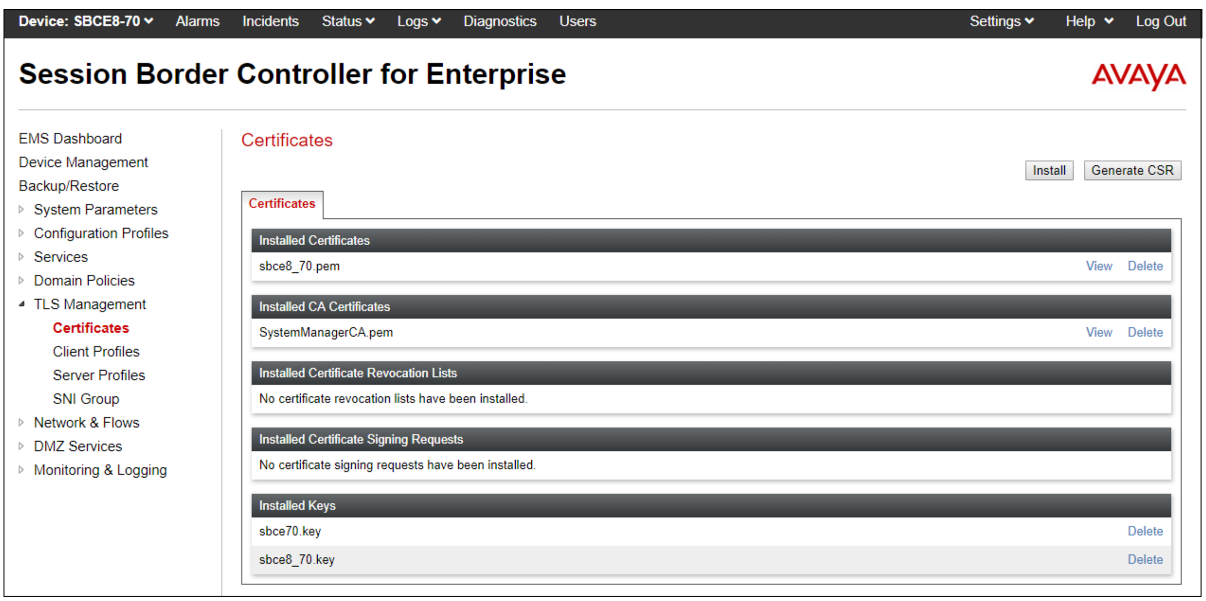
8.2.1. Verify TLS Certificates – Avaya Session Border Controller for Enterprise

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



Step 1 - Select **TLS Management → Certificates** from the left-hand menu. Verify the following:

- System Manager CA certificate is present in the **Installed CA Certificates** area.
- System Manager CA signed identity certificate is present in the **Installed Certificates** area.
- Private key associated with the identity certificate is present in the **Installed Keys** area.

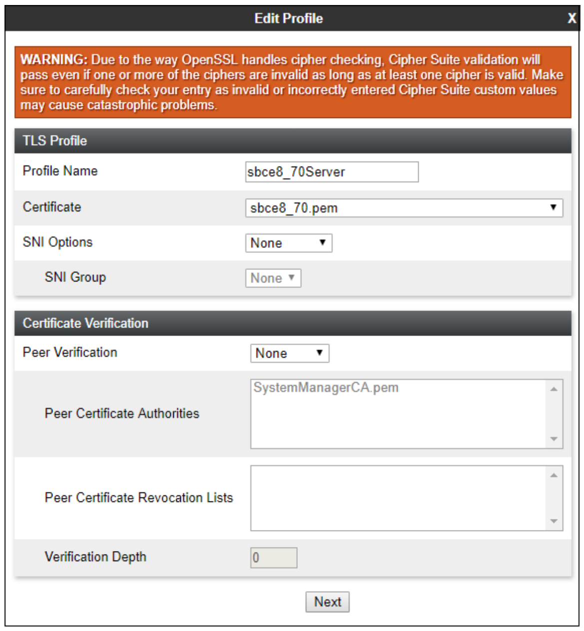


8.2.2. Server Profiles

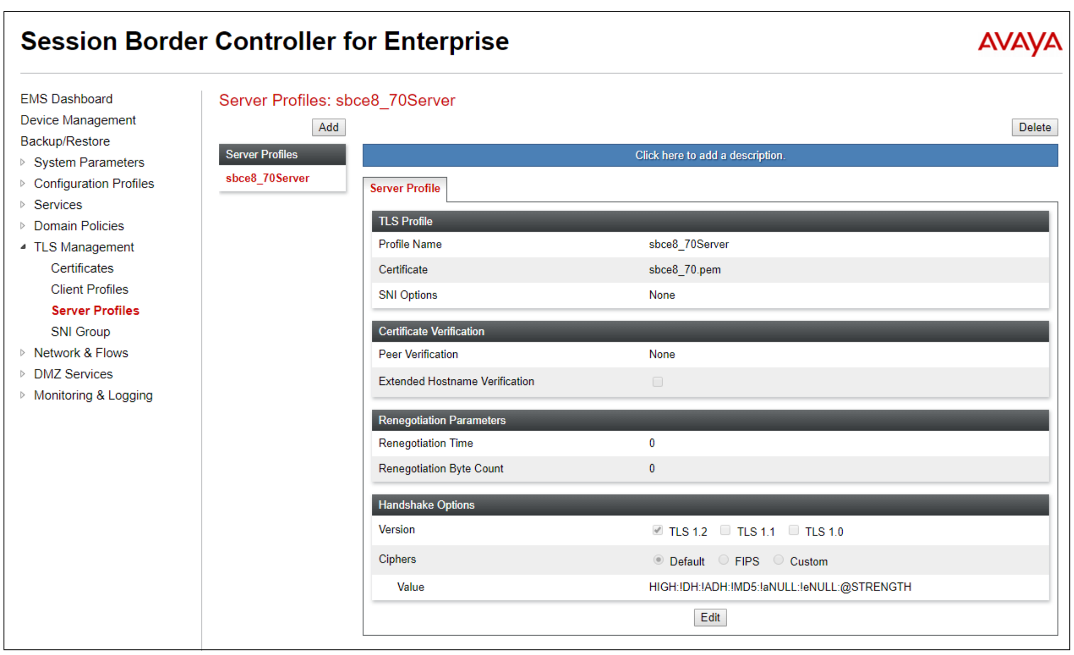
Step 1 - Select **TLS Management → Server Profiles** and click on **Add**. Enter the following:

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

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



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

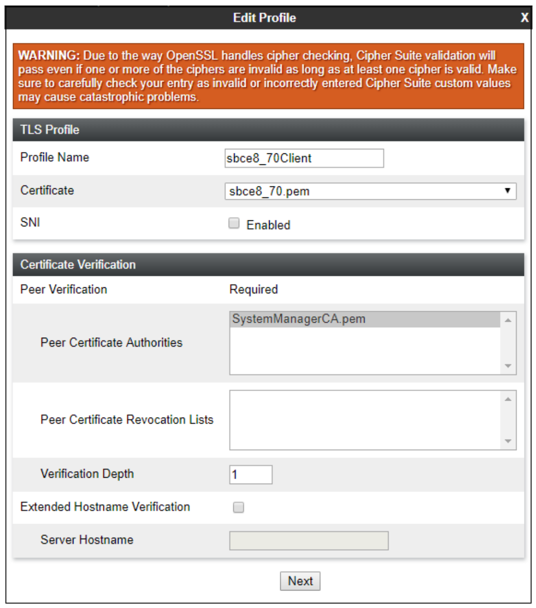


8.2.3. Client Profiles

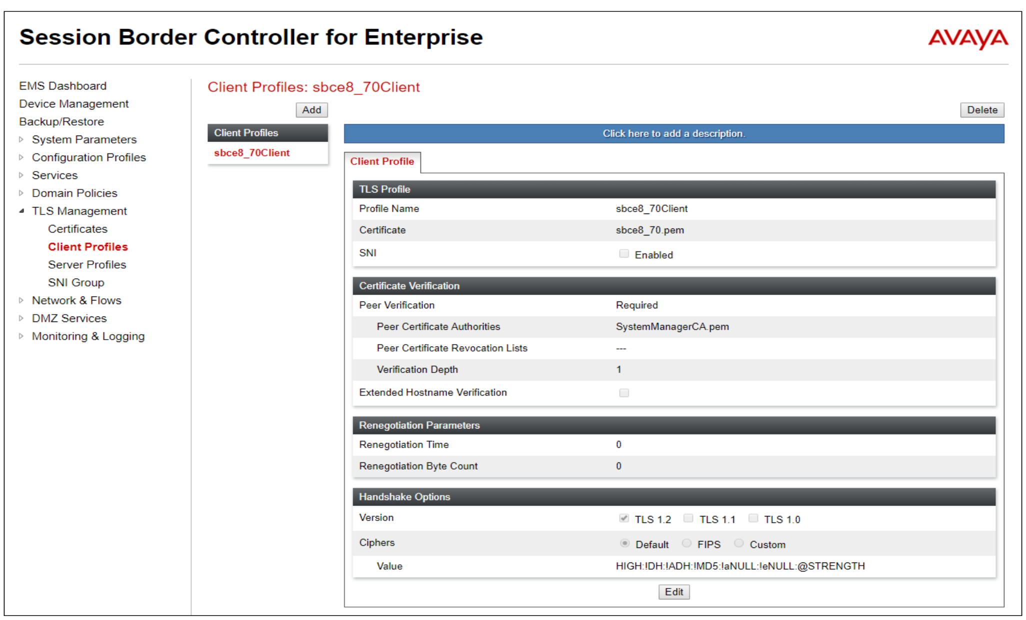
Step 1 - Select **TLS Management → Server Profiles**, and click on **Add**. Enter the following:

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

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



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



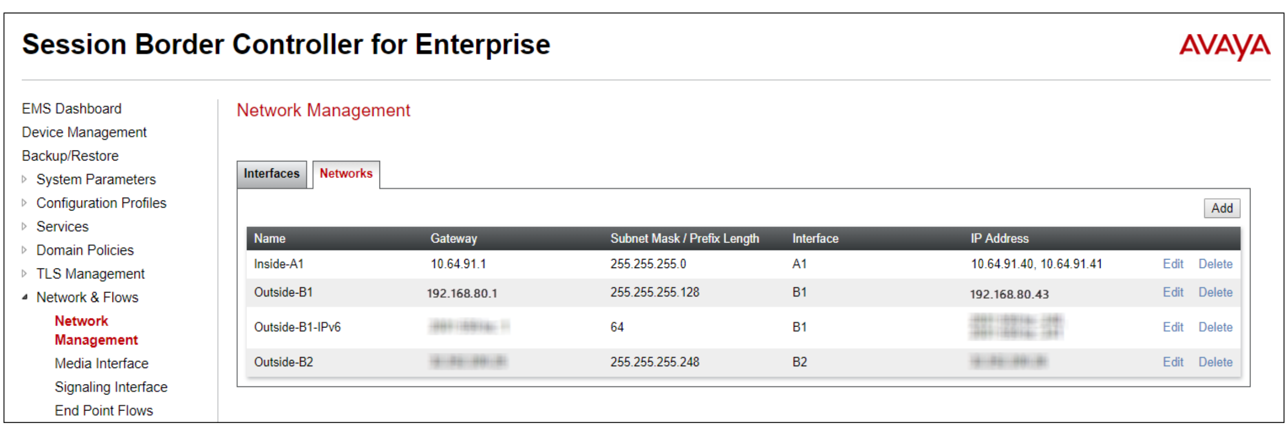
8.3. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of Avaya SBCE, certain network-specific information is defined such as device IP address(es), public IP address(es), netmask, gateway, etc., to interface the device to the network. It is this information that populates the various Network Management tab displays, which can be edited as needed to optimize device performance and network efficiency. Navigate to **Networks & Flows → Network Management**. On the **Networks** tab, verify the IP addresses assigned to the interfaces. The following screen shows the enterprise interface is assigned to **A1** and the interface towards AT&T is assigned to **B1**.

Step 1 - Select **Networks & Flows → Network Management** from the menu on the left-hand side.
Step 2 - The **Interfaces** tab displays the enabled/disabled interfaces. In the reference configuration, interfaces A1 (private) and B1 (public) interfaces are used. To enable an interface, click the corresponding **Disabled** link under the Status column to change it to **Enabled**.

Step 3 - Select the **Networks** tab to display the IP provisioning for the A1 and B1 interfaces. The following Avaya SBCE IP addresses and associated interfaces were used in the sample configuration:

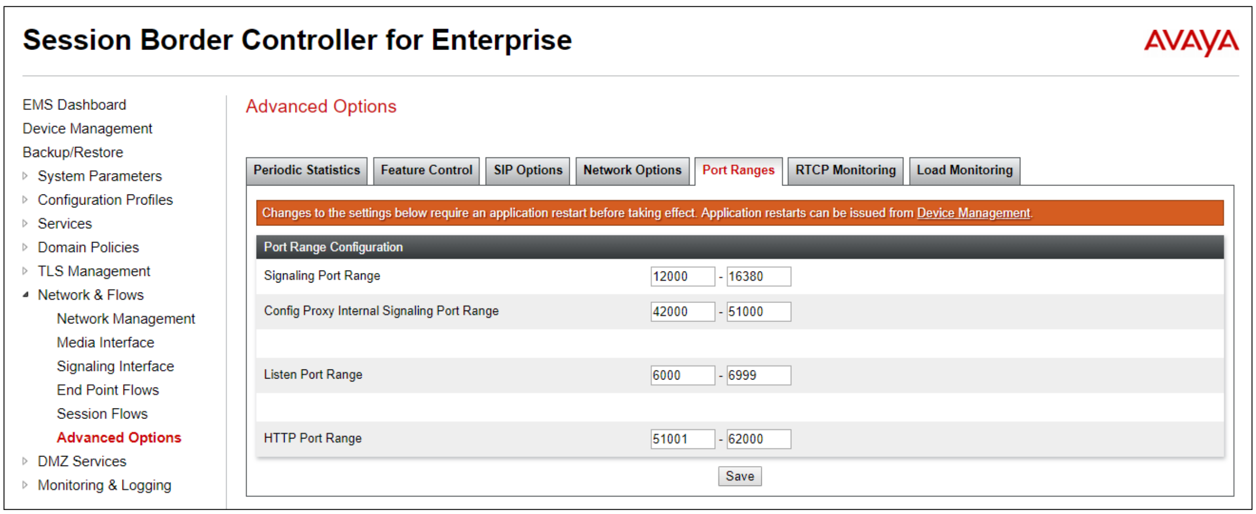
- **B1: 192.168.80.43** – IP address configured for the AT&T IPTF service. This address is known to AT&T. See **Section 3**.
- **A1**: **10.64.91.41** – IP address configured for AT&T IPTF service to Session Manager.



8.4. Advanced Options

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

Step 1 - Select **Network & Flows → Advanced Options** from the menu on the left-hand side.
Step 2 - Select the **Port Ranges** tab.
Step 3 - In the **Signaling Port Range** row, change the range to **12000** – **16380**
Step 4 - In the **Config Proxy Internal Signaling Port Range** row, change the range to **42000** – **51000**.
Step 5 – In the **Listen Port Range** row, change the range to **6000** – **6999**.
Step 6 – In the **HTTP Port Range** row, change the range to **51001** – **62000**.
Step 7 - Select **Save**. Note that changes to these values require an application restart (see **Section 8.1**).



8.5. Media Interfaces

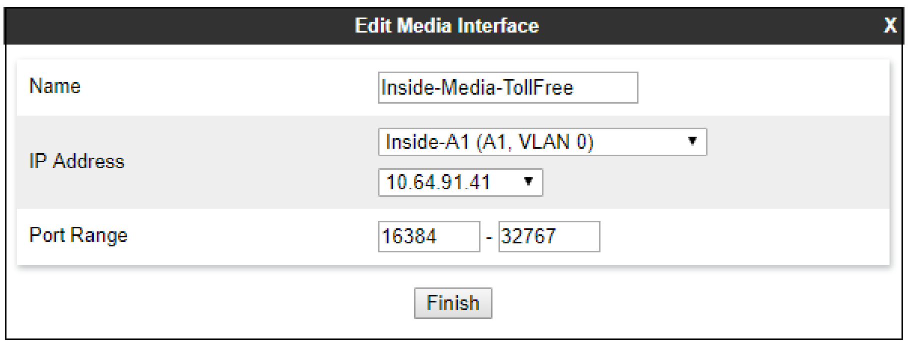
Media Interfaces are created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address, and one of the ports in this range as the listening IP address and port in which the SBCE will accept media from the connected server. Note that some ports in the range required by AT&T were already allocated by the Avaya SBCE for internal use, by default. **Section 8.4** shows the steps required to reallocate the port ranges used by the Avaya SBCE, so the range required by AT&T could be accommodated.

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

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

- **Name**: **Inside-Media-TollFree**
- **IP Address**: Select **Inside-A1 (A1, VLAN0)** and **10.64.91.41**
- **Port Range**: **16384 – 32767**

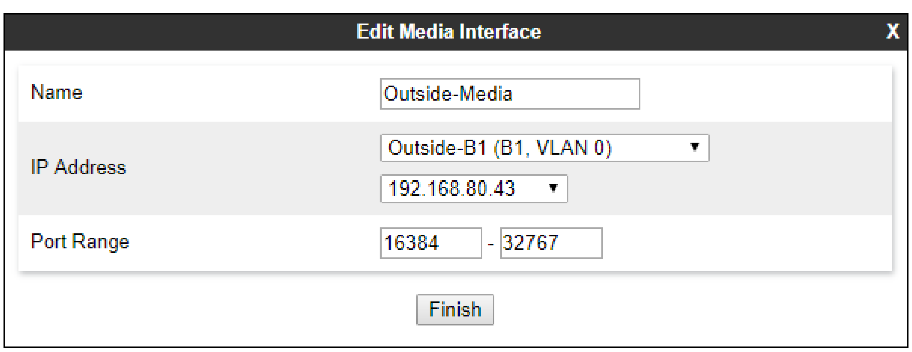
Step 3 - Click **Finish**.



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

- **Name**: **Outside-Media**
- **IP Address**: Select **Outside-B1 (B1, VLAN0)** and **192.168.80.43**
- **Port Range**: **16384 – 32767**

Step 5 - Click **Finish**



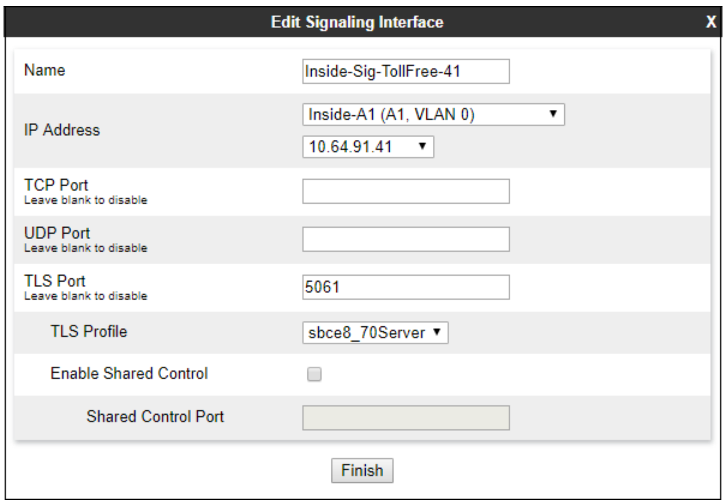
8.6. Signaling Interfaces

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

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

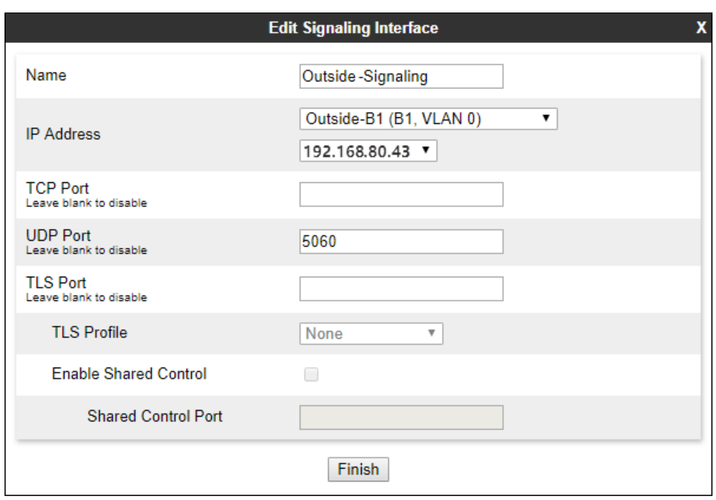
- **Name**: **Inside-Sig-TollFree-41**
- **IP Address**: Select **Inside-A1 (A1, VLAN0)** and **10.64.91.41**
- **TLS Port**: **5061**
- **TLS Profile**: Select the TLS server profile created in **Section 8.2.2**

Step 3 - Click **Finish**



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

- **Name**: **Outside-Signaling**
- **IP Address**: Select **Outside-B1 (B1, VLAN0)** and **192.168.80.43**
- **UDP Port**: **5060**. Click **Finish.**



8.7. Server Interworking Profiles

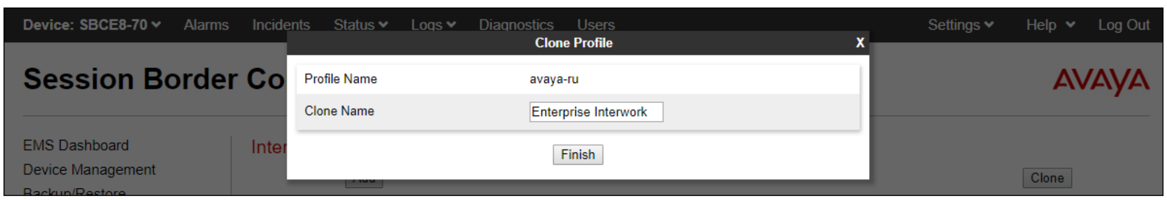
The Server Internetworking profiles include parameters to make the Avaya SBCE function in an enterprise VoIP network using different implementations of the SIP protocol. There are default profiles available that may be used as is, or modified, or new profiles can be configured as described below.

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

8.7.1. Server Interworking Profile – Enterprise

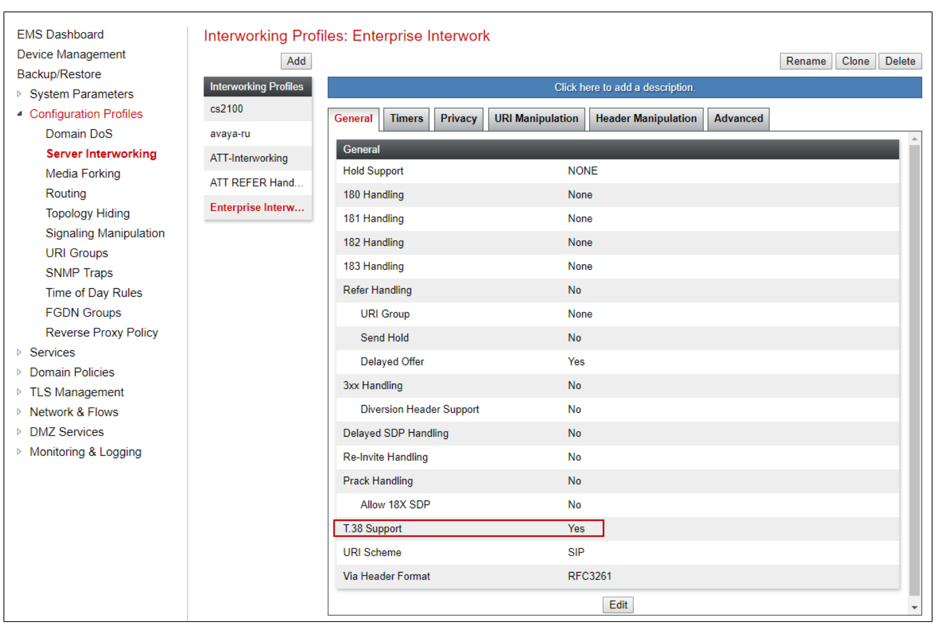
In the sample configuration, the enterprise Server Interworking profile was cloned from the default **avaya-ru** profile and then modified.

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



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

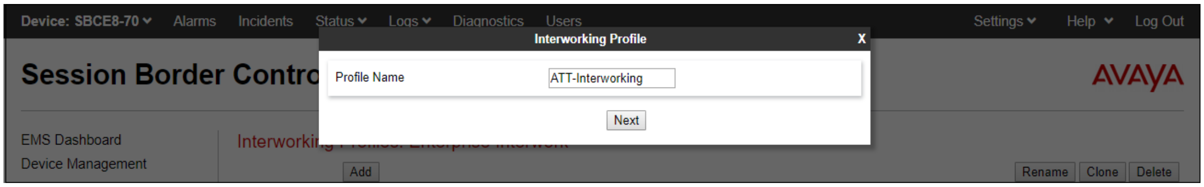
- Check **T38 Support**.
- All other options can be left with default values. Click **Finish** (not shown).



8.7.2. Server Interworking – AT&T

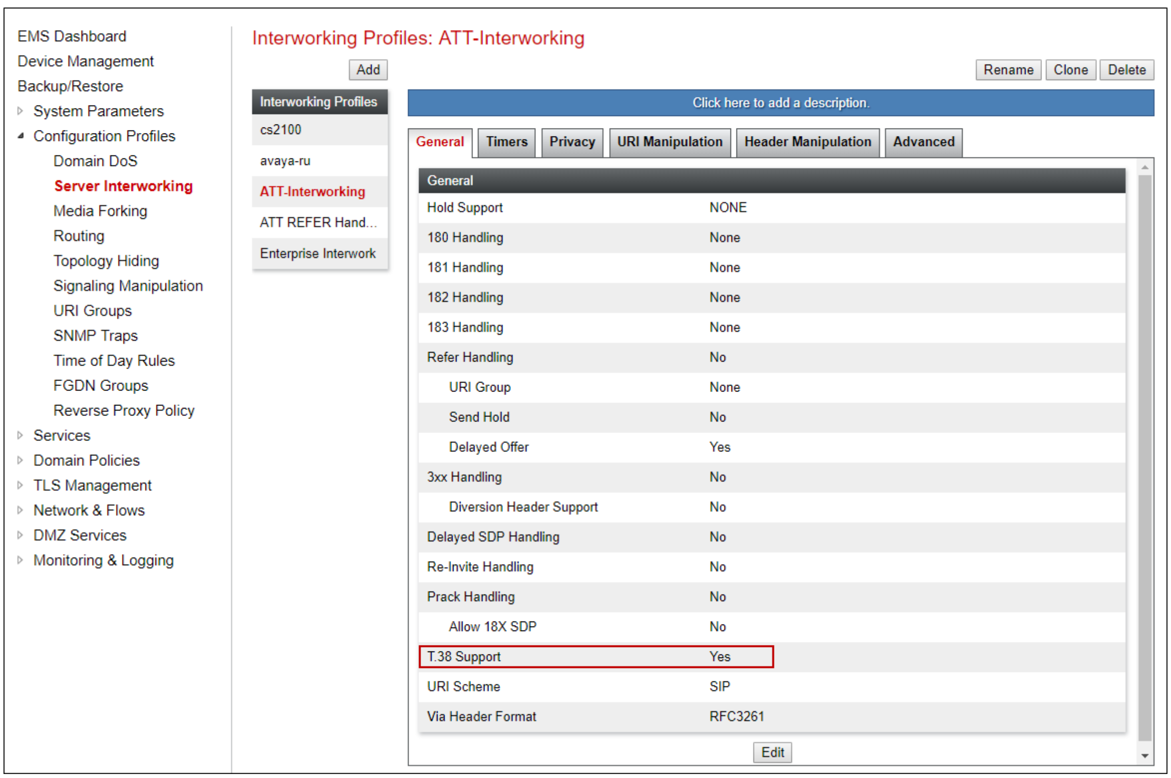
Repeat the steps shown in **Section 8.7.1** to add an Interworking Profile for the connection to AT&T via the public network, with the following changes:

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

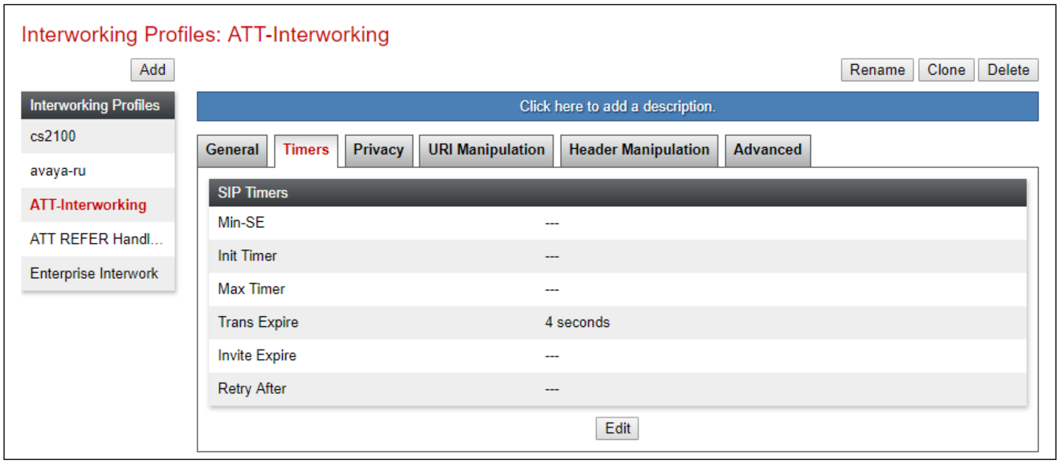


Step 2 - The **General** screen will open:

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



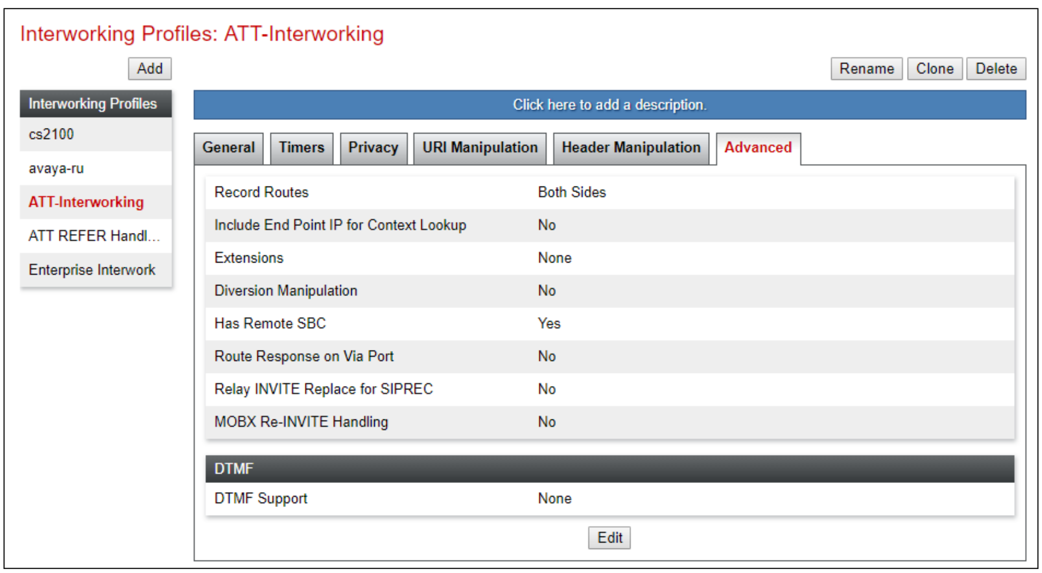
Step 3 – On the **Timers** tab, the **Trans Expire** timer is set to the allotted time the Avaya SBCE will try the first primary server before trying the secondary server, if one exists.



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

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

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



8.8. Signaling Manipulation

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

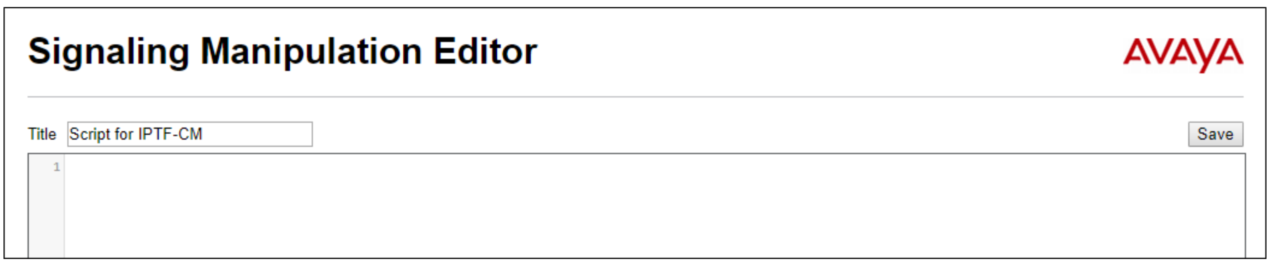
Note – Use of the Signaling Manipulation scripts require higher processing requirements on the Avaya SBCE. Therefore, this method of header manipulation should only be used in cases where the use of Server Interworking Profiles (**Section 8.7**) or Signaling Rules (**Section 8.14**) do not meet the desired result. Refer to References **[11]** for information on the Avaya SBCE scripting language.

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

- Remove the gsid and epv parameters from outbound Contact headers. (**Section 2.2**, **Item 7**).
- Remove the Bandwidth headers sent by some Avaya SIP endpoints. (**Section 2.2**, **Item 8**).

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

- Enter a name for the script in the **Title** box (e.g., **Script for IPTF-CM**).



Step 3 - Copy and paste the script below in the editor window.

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

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

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

    }
 }
-----------------------------------------------------------------------------------------
```

Step 4 - Click on **Save**. The script editor will test for any errors, and the window will close. This script is applied to the AT&T SIP Server profile in **Section 8.9.2**.

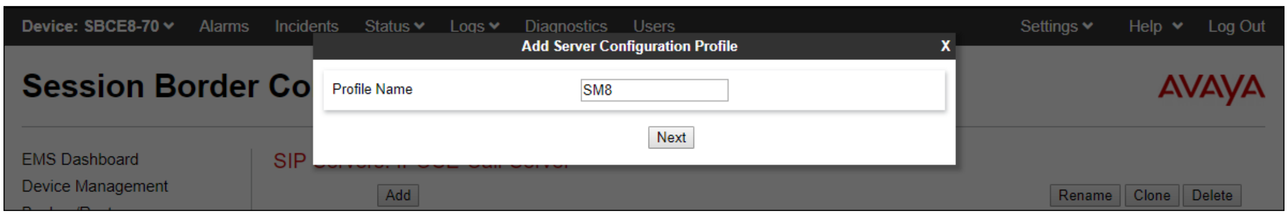
8.9. SIP Server Profiles

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

8.9.1. SIP Server Profile – Session Manager

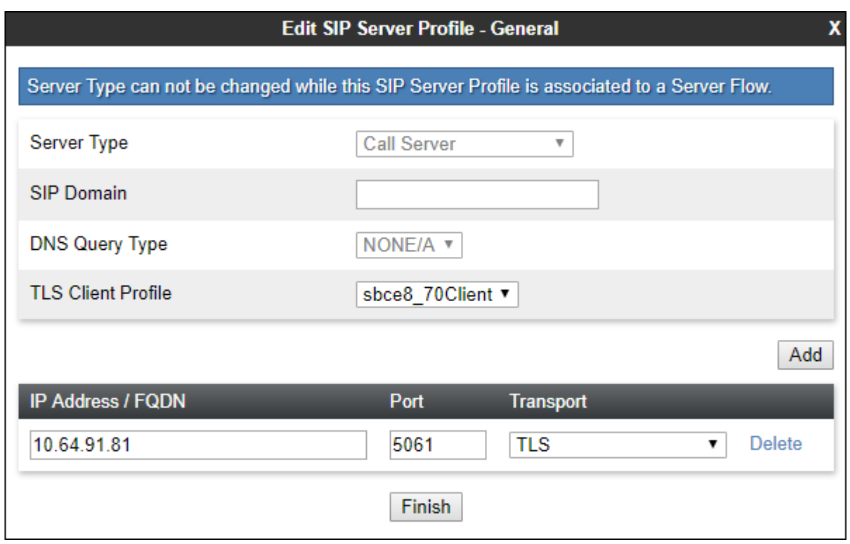
This section defines the SIP Server Profile for the Avaya SBCE connection to Session Manager.

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



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

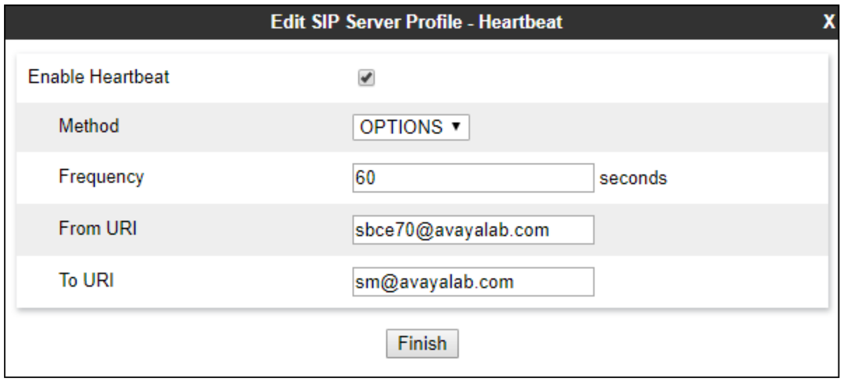
- Select **Server Type**: **Call Server**
- **SIP Domain**: Leave blank (default)
- **DNS Query Type**: Select **NONE/A** (default)
- **TLS Client Profile**: Select the profile create in **Section 8.2.3** (e.g., **sbce8_70Client**)
- **IP Address/FQDN**: **10.64.91.81** (Session Manager Security Module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



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

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

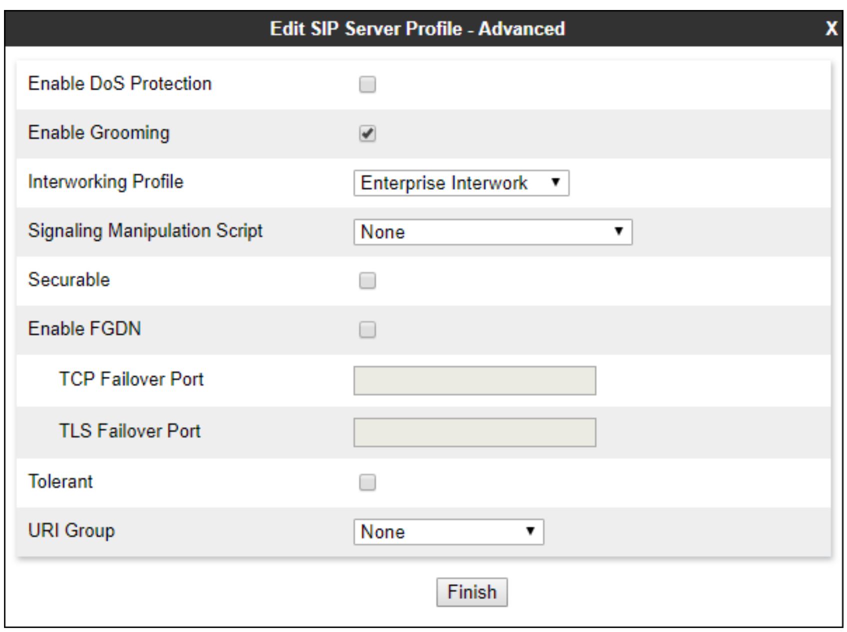
- Select **OPTIONS** from the **Method** drop-down menu.
- Select the desired frequency that the SBCE will source OPTIONS toward Session Manager.
- Make logical entries in the **From URI** and **To URI** fields that will be used in the OPTIONS headers.



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

Step 7 – On the **Advanced** tab:

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



8.9.2. SIP Server Profile – AT&T

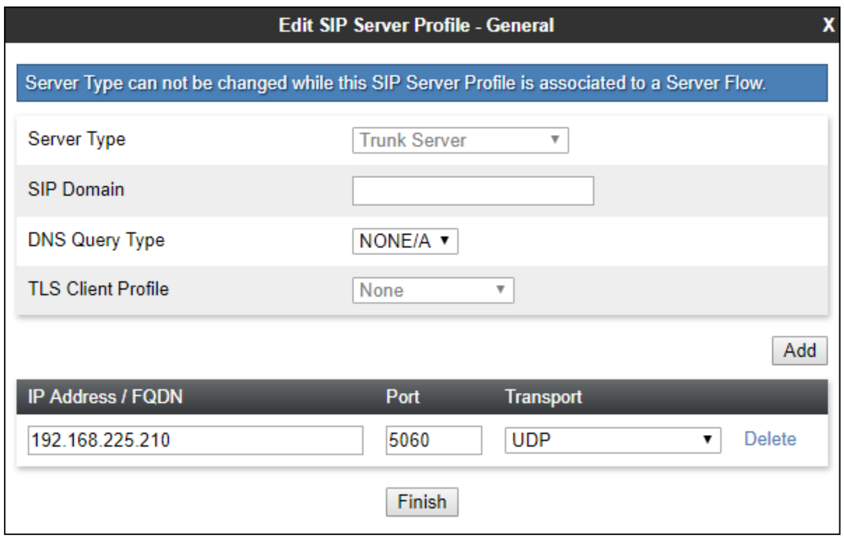
Note – The AT&T IPTF service may provide a Primary and Secondary Border Element. This section describes the connection to a single (Primary) Border Element.

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

Step 1 - Select **Add** and enter a Profile Name (e.g., **ATT-TollFree-trk-svr**) and select **Next** (not shown).

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

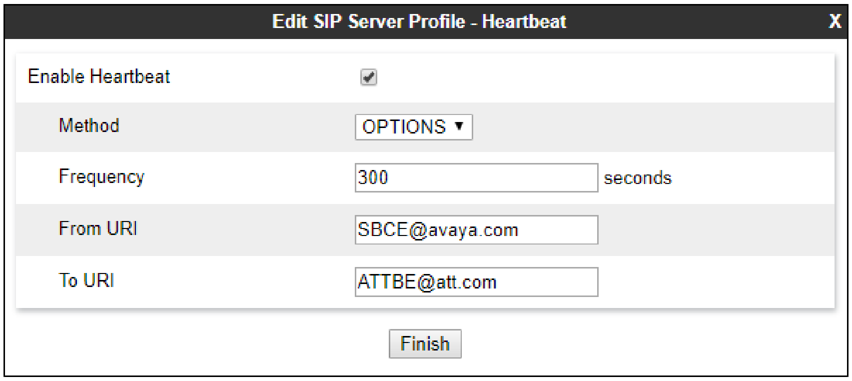
- Select **Server Type**: **Trunk Server**
- **IP Address/FQDN**: **192.168.225.210** (AT&T Border Element IP address)
- **Port**: **5060**
- Select **Transport**: **UDP**
- Click **Next** (not shown) to proceed. If editing an existing profile, click **Finish**.



Step 3 – Default values can be used on the **Authentication** tab.

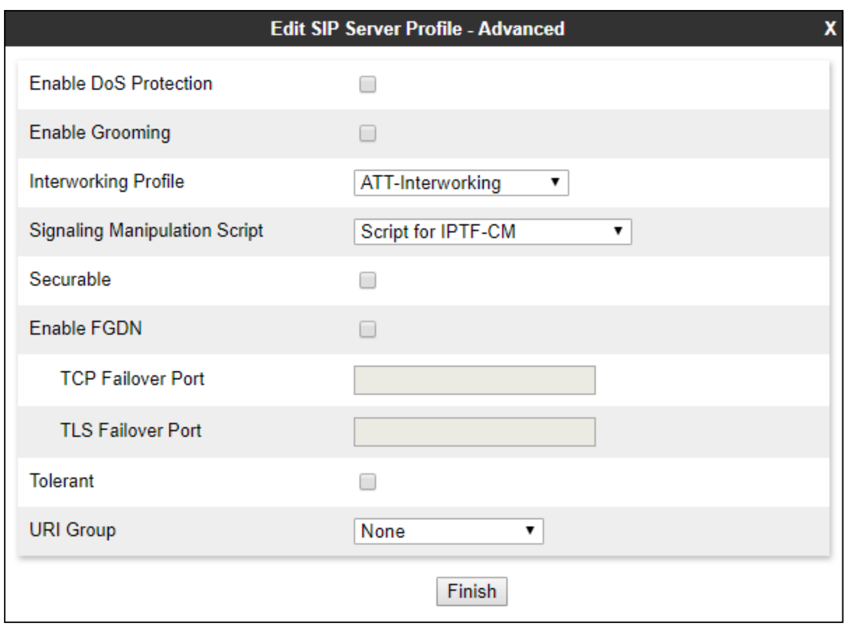
Step 4 – On the **Heartbeat** tab, check the **Enable Heartbeat** box to have the Avaya SBCE source "heartbeats" toward AT&T. This configuration is optional.

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



Step 5 - On the **Advanced** window, enter the following.

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



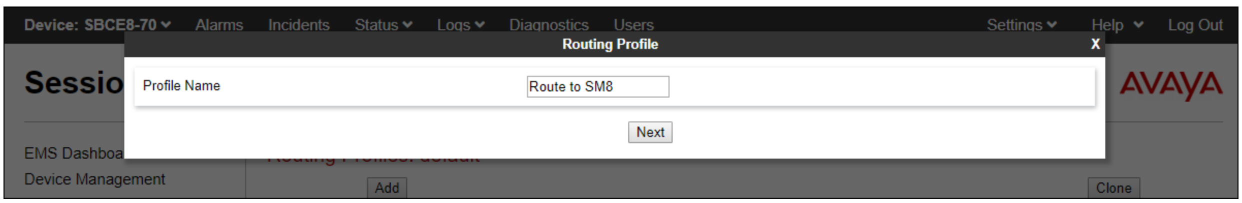
8.10. Routing Profiles

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

8.10.1. Routing Profile – Session Manager

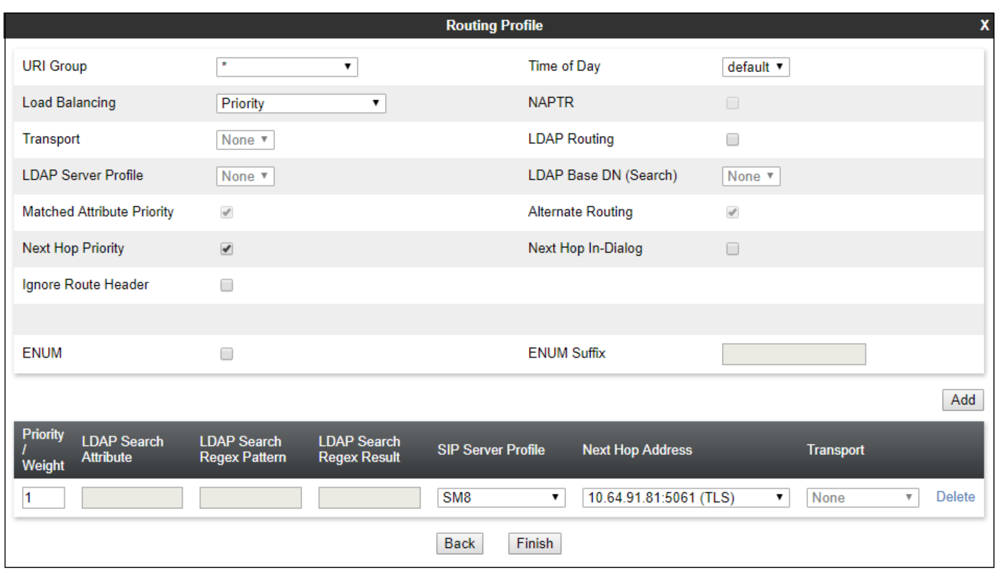
This provisioning defines the Routing Profile for the connection to Session Manager.

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



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

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

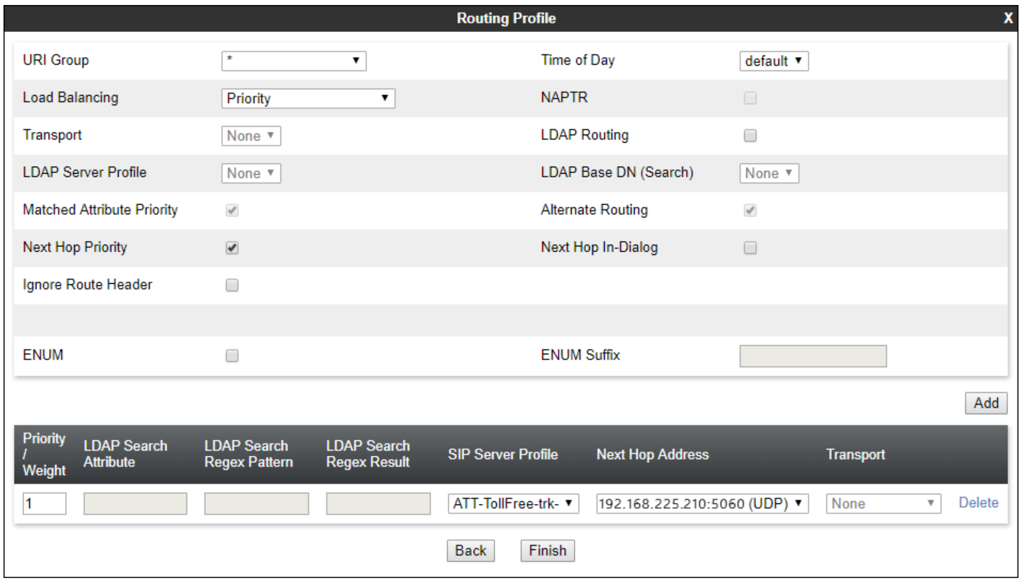


8.10.2. Routing Profile – AT&T

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

Step 1 - Enter a Profile Name: (e.g., **Route to ATT IPTF**).
Step 2 - On the **Next-Hop Address** window, populate the following fields:

- **Priority/Weight**: **1**
- **Server Configuration**: **ATT-TollFree-trk-svr** (**from Section 8.9.2**).
- **Next Hop Address**: Verify that the **192.168.225.210:5060 (UDP)** entry from the drop-down menu is selected (AT&T Border Element IP address).
- Click **Finish**.



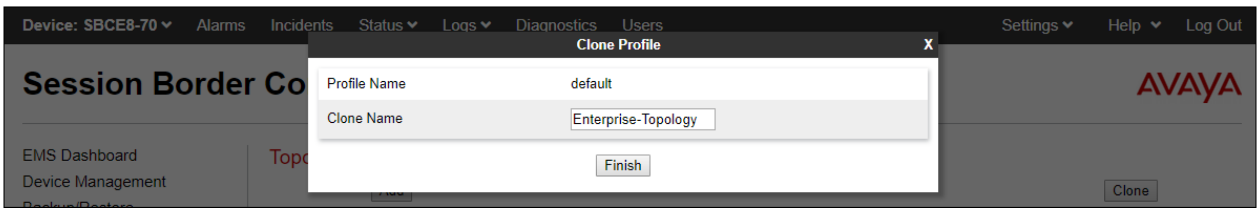
8.11. Topology Hiding Profiles

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

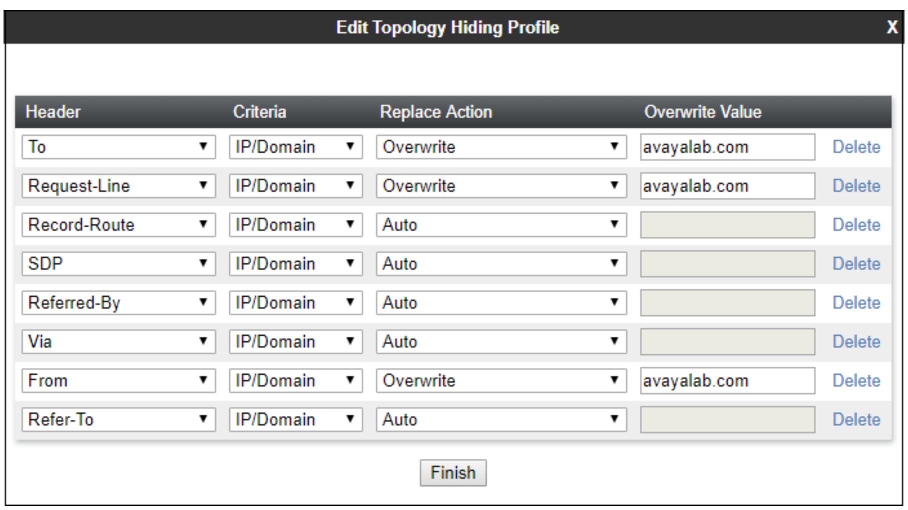
8.11.1. Topology Hiding – Enterprise Side

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

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



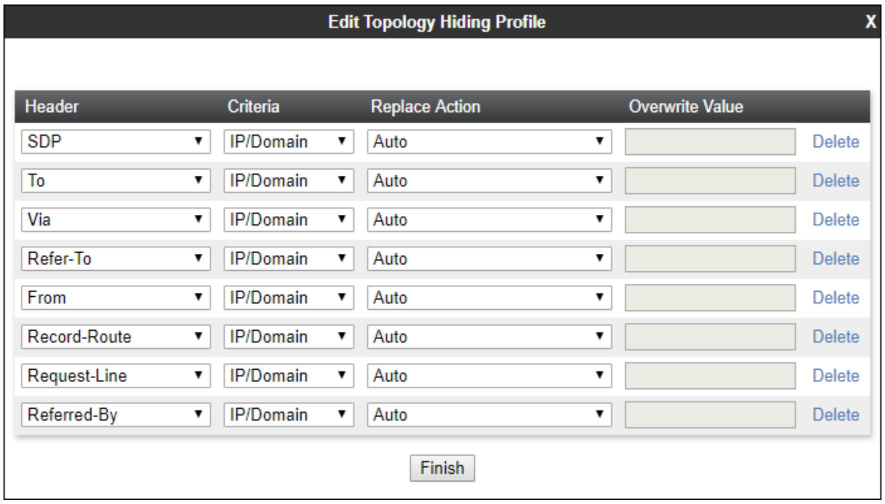
Step 4 - Edit the newly created **Enterprise-Topology** profile.
Step 5 - For the **Request-Line**, **To** and **From** headers select **Overwrite** under the **Replace Action** column. Enter the domain of the enterprise (e.g., **avayalab.com**) on the **Overwrite Value** field.
Step 6 - Click **Finish**.



8.11.2. Topology Hiding – AT&T Side

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

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

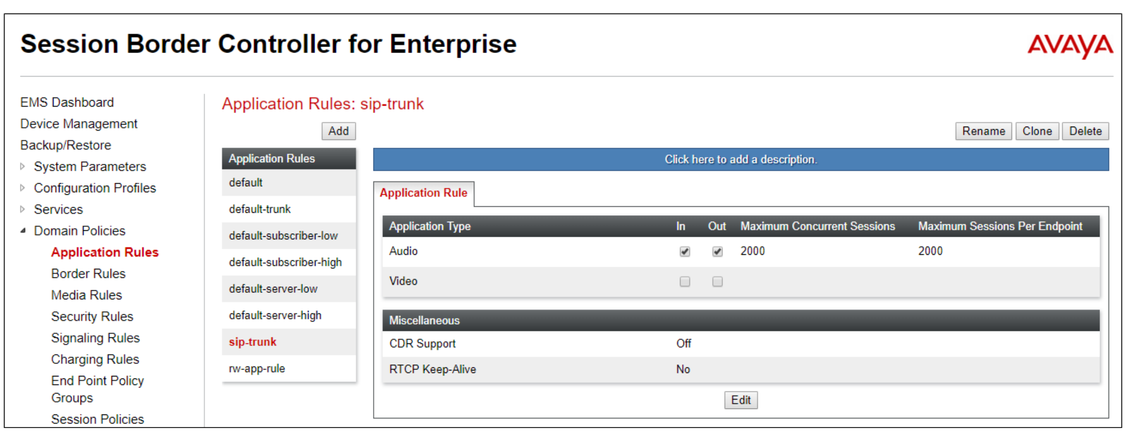


8.12. Application Rules

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

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

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



8.13. Media Rules

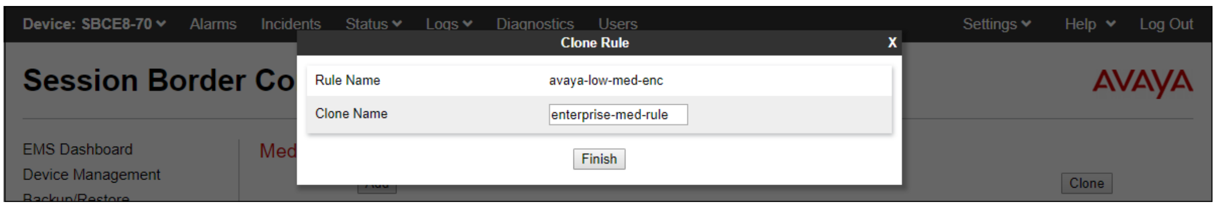
Media Rules are used to define media encryption and QoS parameters. Separate media rules are created for the enterprise and AT&T.

8.13.1. Enterprise – Media Rule

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

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

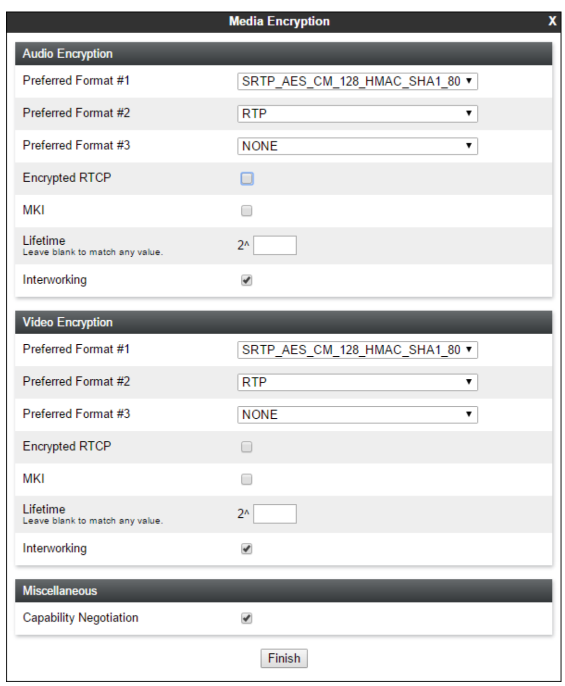
- In the **Clone Name** field enter the new Media Rule name (e.g., **enterprise-med-rule**)
- Click **Finish.** The newly created rule will be displayed.



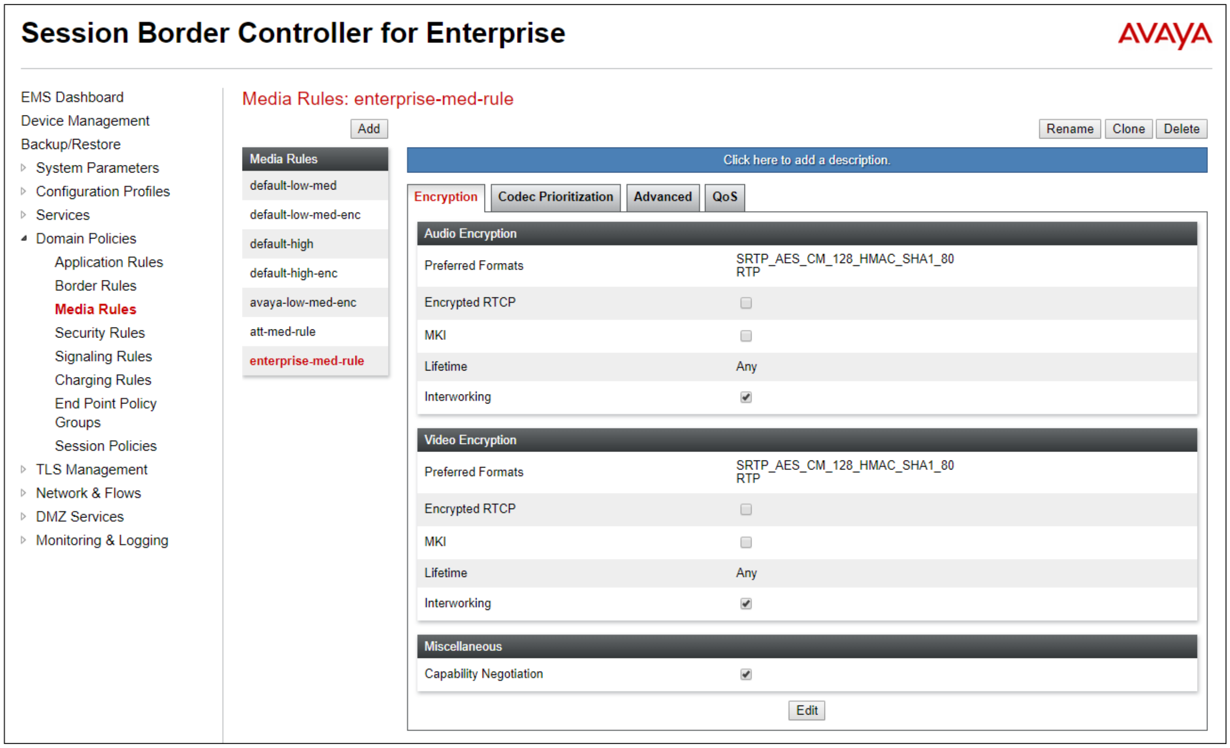
Step 4 - On the **enterprise med rule** just created, select the **Encryption** tab.

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

Step 5 - Click **Finish**.



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

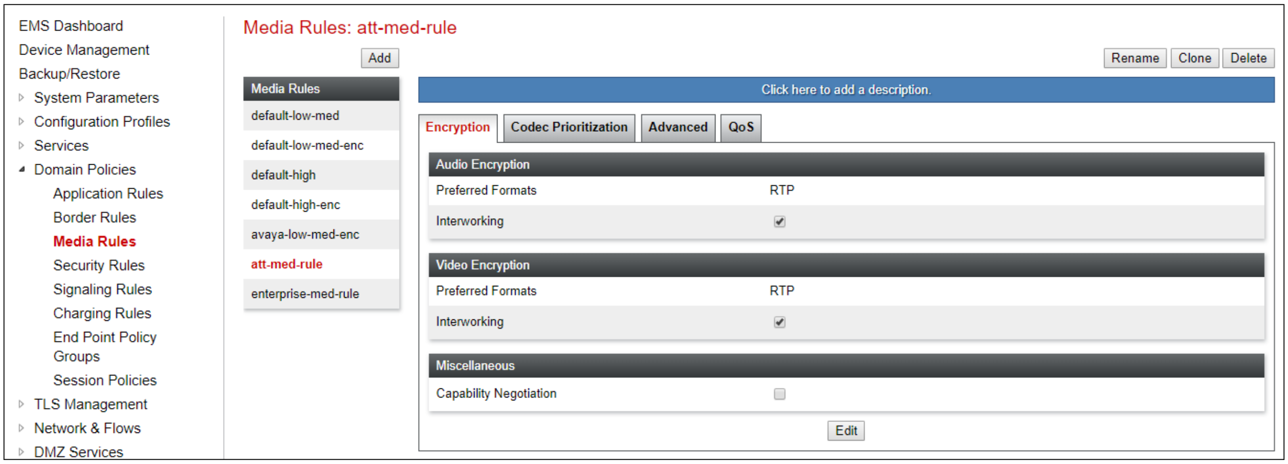


8.13.2. AT&T – Media Rule

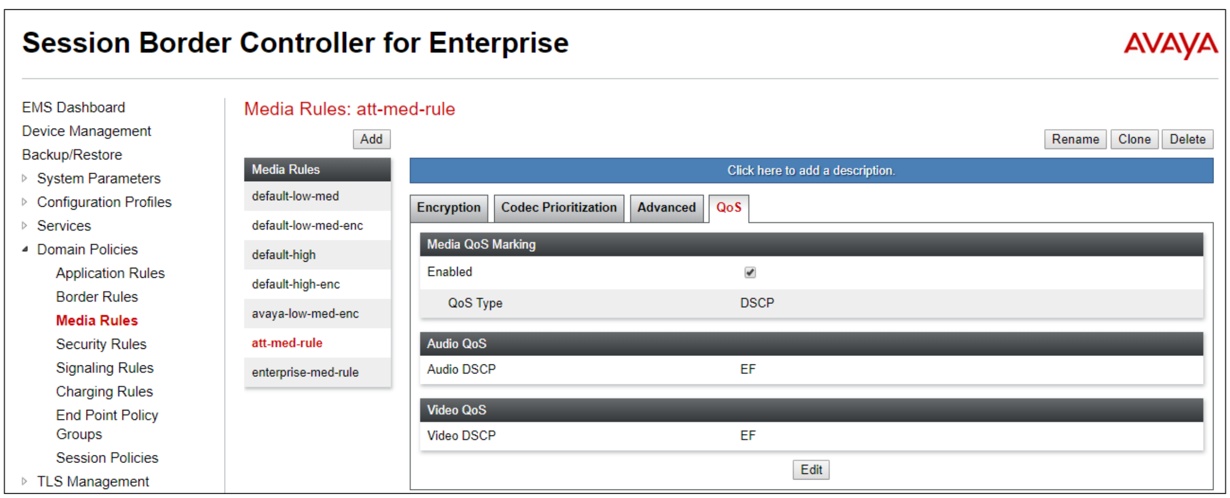
Repeat the steps in **Section 8.13.1**, with the following changes, to create a Media Rule for AT&T.

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

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



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



8.14. Signaling Rules

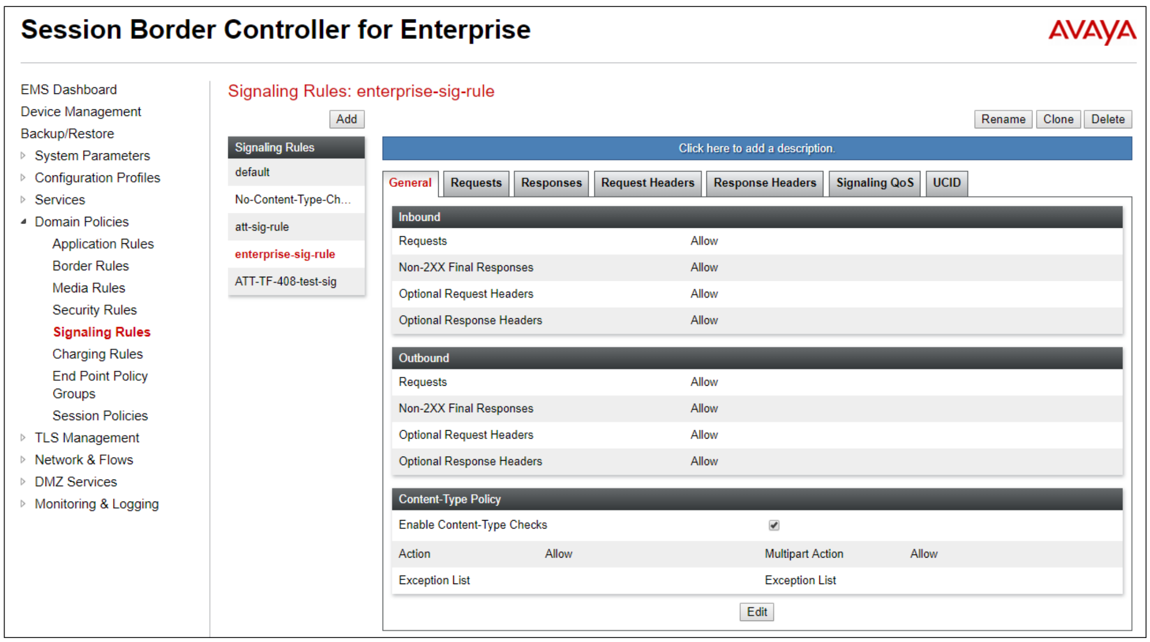
Signaling Rules are used to define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message, and to specify QoS parameters for the SIP signaling packets.

8.14.1. Signaling Rule – Enterprise

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

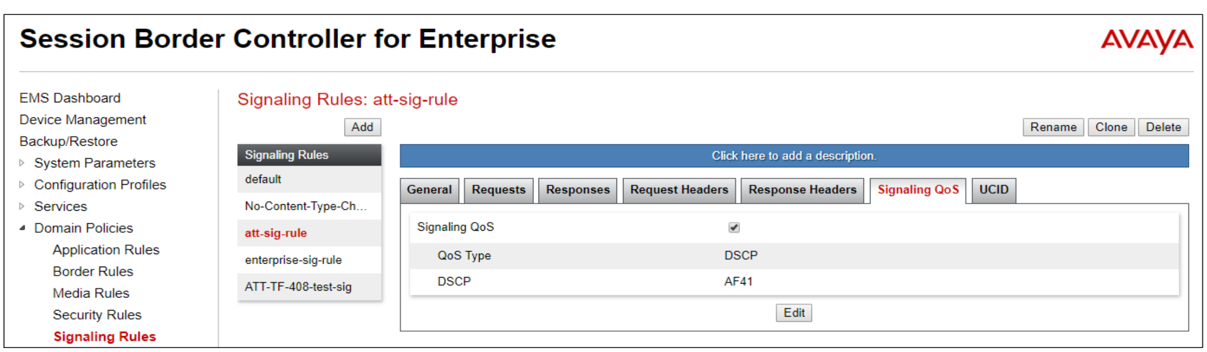
- In the **Rule Name** field enter the new Signaling Rule name (e.g., **enterprise-sig-rule)**
- Click **Finish**.

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



8.14.2. Signaling Rule – AT&T

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



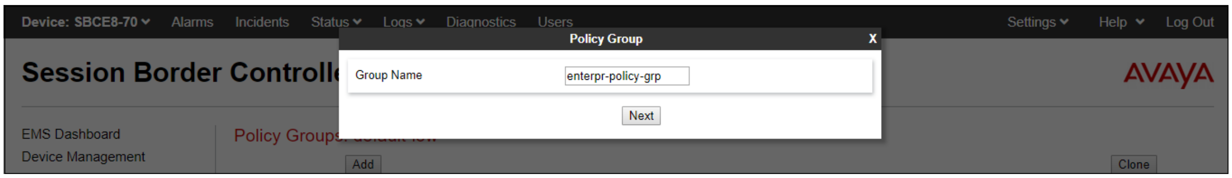
8.15. Endpoint Policy Groups

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

8.15.1. Endpoint Policy Group – Enterprise

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

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

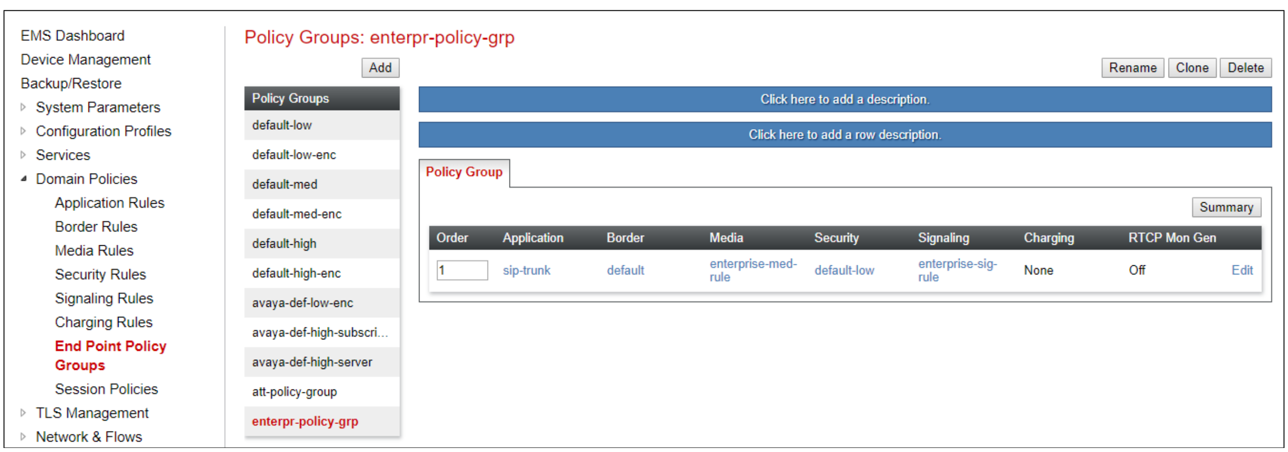


Step 3 – On the **Policy Group** window (not shown), select the following.

- **Application Rule**: **sip-trunk** (created in **Section 8.128.12**).
- **Border Rule**: **default**.
- **Media Rule**: **enterprise-med-rule** (created in **Section 8.13.1**).
- **Security Rule**: **default-low**.
- **Signaling Rule**: **enterprise-sig-rule** (created in **Section 8.14.1**).

Step 4 - Select **Finish**.

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



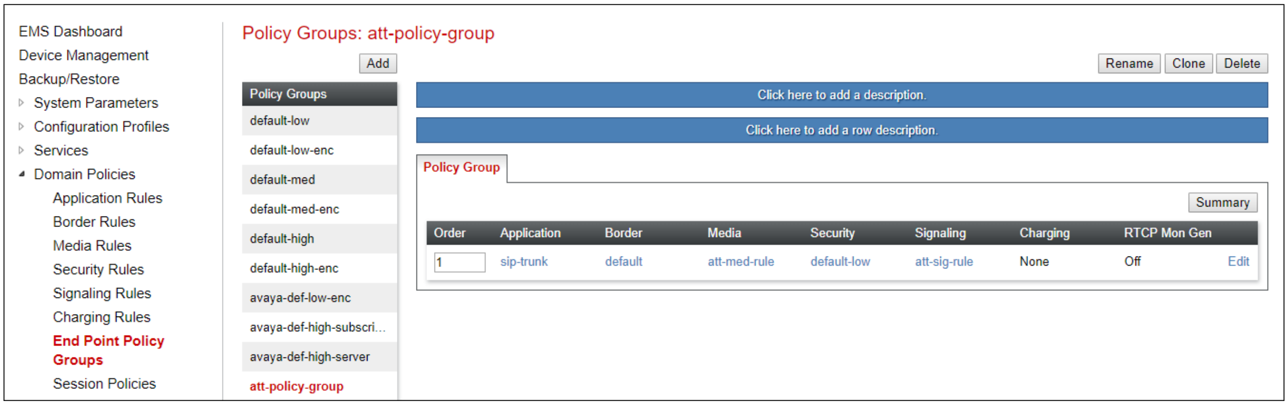
8.15.2. Endpoint Policy Group – AT&T

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

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

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

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



8.16. Endpoint Flows – Server Flows

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

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

8.16.1. Server Flows – Enterprise

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

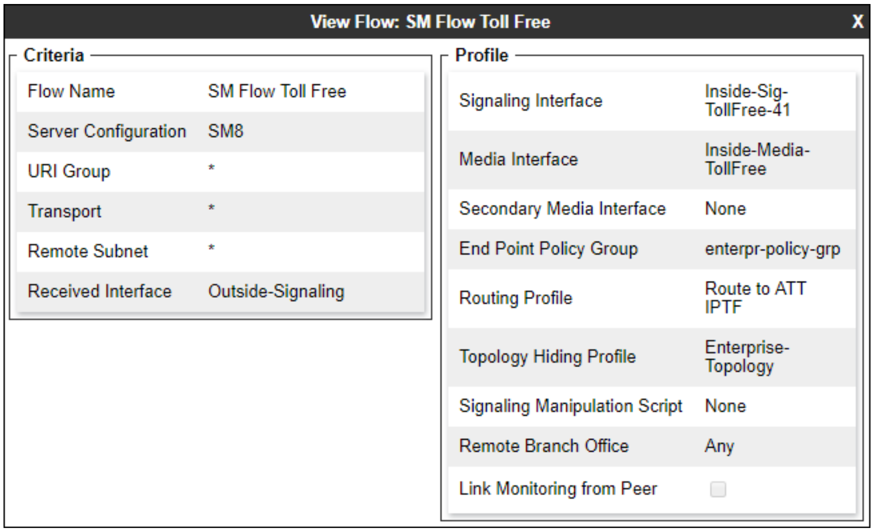
Step 2 - Select the **Server Flows** tab (not shown).

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

- **Flow Name**: Enter a name for the flow, e.g., **SM Flow Toll Free**
- **Server Configuration**: **SM8** (**Section 8.9.1**).
- **URI Group**: *****
- **Transport**: *****
- **Remote Subnet**: *****
- **Received Interface**: **Outside-Signaling** (**Section 8.6**).
- **Signaling Interface**: **Inside-Sig-TollFree-41** (**Section 8.6**).
- **Media Interface**: **Inside-Media-TollFree** (**Section 8.5**).

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

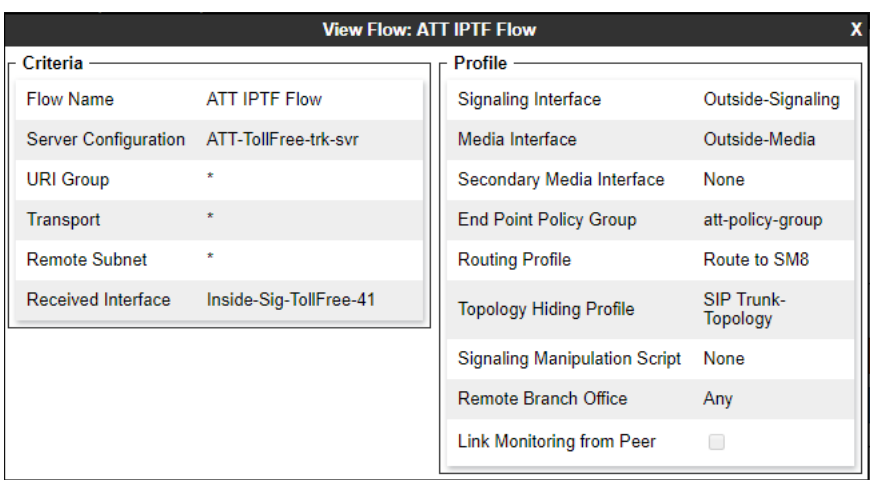
Step 4 - Click **Finish** (not shown).



8.16.2. Server Flow – AT&T

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

- **Flow Name**: Enter a name for the flow, e.g., **ATT IPTF Flow**.
- **Server Configuration**: **ATT-TollFree-trk-svr** (**Section 8.9.2**).
- **Received Interface**: **Inside-Sig-TollFree-41** (**Section 8.6**).
- **Signaling Interface**: **Outside-Signaling (Section 8.6**).
- **Media Interface**: **Outside-Media** (**Section 8.5**).
- **End Point Policy Group**: **att-policy-group** (**Section 8.15.2**).
- **Routing Profile**: **Route to SM8** (**Section 8.10.1**).
- **Topology Hiding Profile**: **SIP-Trunk-Topology** (**Section 8.11.2**).



9. AT&T IP Toll Free Service Configuration

AT&T provides the IPTF service border element IP address, the access DID numbers, and the associated DNIS digits used in the reference configuration. In addition, the AT&T IPTF features, and their associated access numbers, are also assigned by AT&T. AT&T requires that the Avaya SBCE public (B1) IP address be provided to the IPTF service, as part of the provisioning process. For more information, consult reference **[15]**.

10. Verification Steps

The following steps may be used to verify the configuration.

10.1. AT&T IP Toll Free Service

The following scenarios may be executed to verify functionality with the AT&T IPTF service:

1. Place an inbound call, answer the calls, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnects properly.
2. Verify basic call functions such as hold, transfer, and conference.
3. Verify the use of DTMF signaling.
4. Using the appropriate IPTF access numbers and DTMF codes, verify that the following IPTF features are successful:
    a. Legacy Transfer Connect DTMF triggered Agent Hold, Conference and Transfer capabilities
    b. Alternate Destination Routing call redirection capabilities based on Busy, Ring-No-Answer, and other SIP error codes.
5. Inbound fax using T.38 or G.711. See **Section 2.2** for limitations.
6. SIP OPTIONS monitoring of the health of the SIP trunk.

10.2. Avaya Aura® Communication Manager Verification

The following examples are only a few of the monitoring commands available on Communication Manager. See **[6]** for more information.

- Tracing a SIP trunk.
    1. From the Communication Manager console connection enter the command ***list trace tac xxx***, where ***xxx*** is a trunk access code defined for the SIP trunk to AT&T (e.g., *04). Note that in the trace shown below, Session Manager has previously converted the IPTF DNIS number included in the Request URI, to the Communication Manager VDN 71041, before sending the INVITE to Communication Manager.

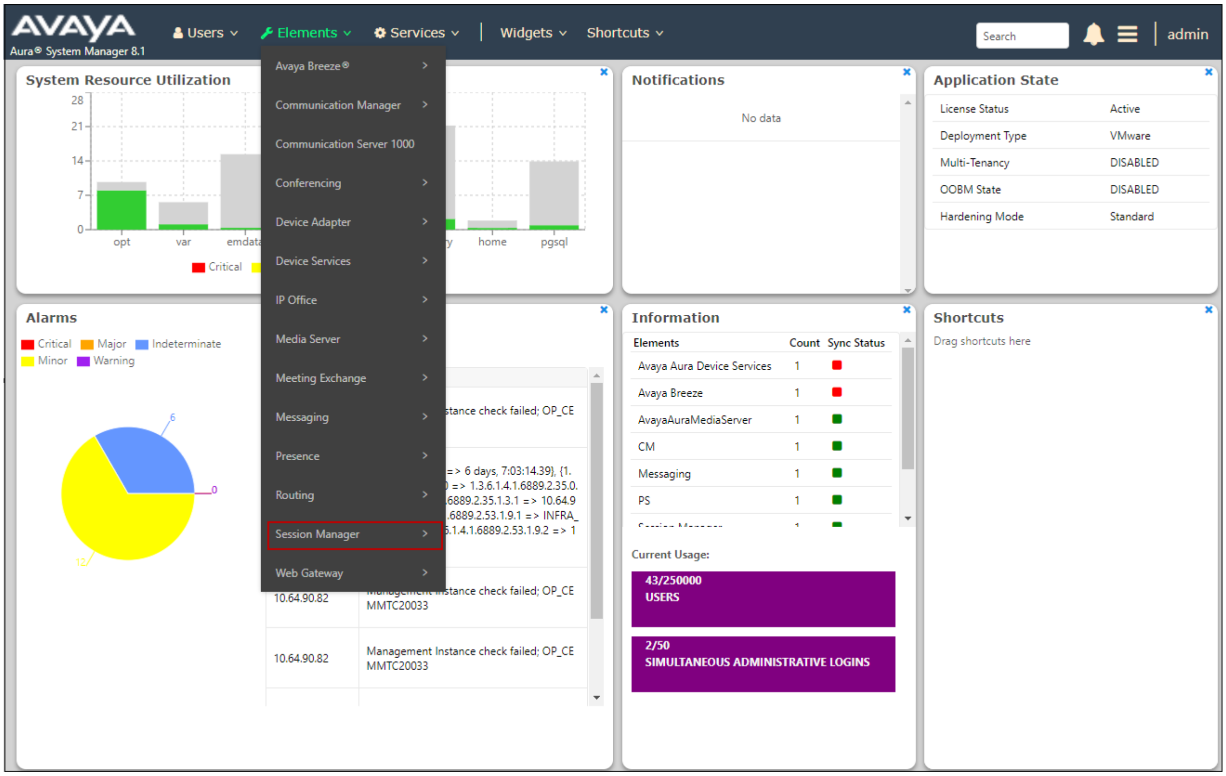
```
list trace tac *04                                                  Page   1
                                LIST TRACE
time            data

13:35:53 TRACE STARTED 11/06/2019 CM Release String cold-01.0.890.0-25578
13:36:04 SIP<INVITE sips:71041@avayalab.com SIP/2.0
13:36:04     Call-ID: 31ebc87eee7ec97b24e184164efeae18
13:36:04     active trunk-group 4 member 1    cid 0xf6b
13:36:04    0  0 ENTERING TRACE cid 3947
13:36:04    4  1 vdn e71041 bsr appl   0 strategy 1st-found override n
13:36:04    4  1 AVDN: 71041 AVRD:
13:36:04    4  1 # Wait hearing ringback...
13:36:04    4  2 wait 2 secs hearing ringback
13:36:04 SIP>SIP/2.0 180 Ringing
13:36:04     Call-ID: 31ebc87eee7ec97b24e184164efeae18
13:36:04     dial 71041
13:36:04     ring vector 4      cid 0xf6b
13:36:04     G729 ss:off ps:20
             rgn:4 [10.64.91.41]:16924
             rgn:1 [10.64.91.91]:16394
13:36:04     xoip options: fax:T38 modem:off tty:US  uid:0x50001f
             xoip ip: [10.64.91.91]:16394
13:36:06    4  3 # Play greeting and collect 1 d...
13:36:06    4  4 collect 1 digits after annc 11001 for none
13:36:06 SIP>SIP/2.0 200 OK
```

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

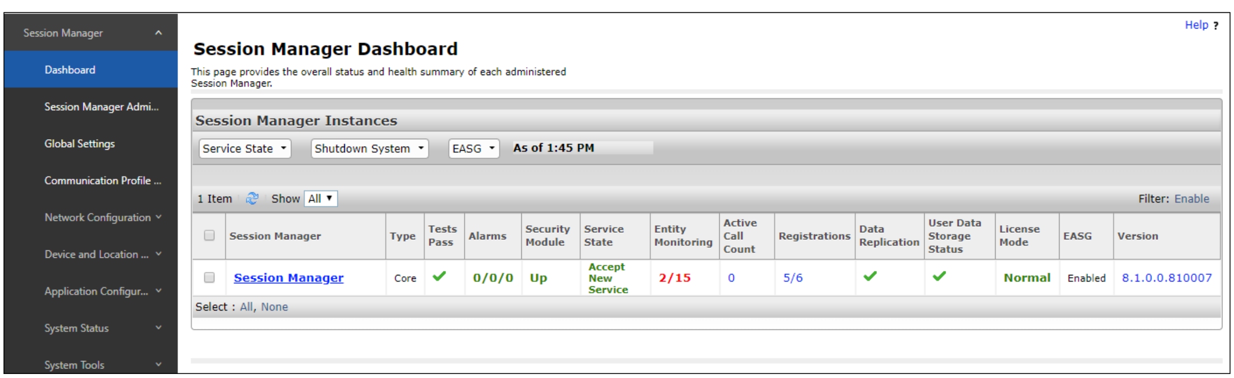
10.3. Avaya Aura® Session Manager Verification

The Session Manager configuration may be verified via System Manager.

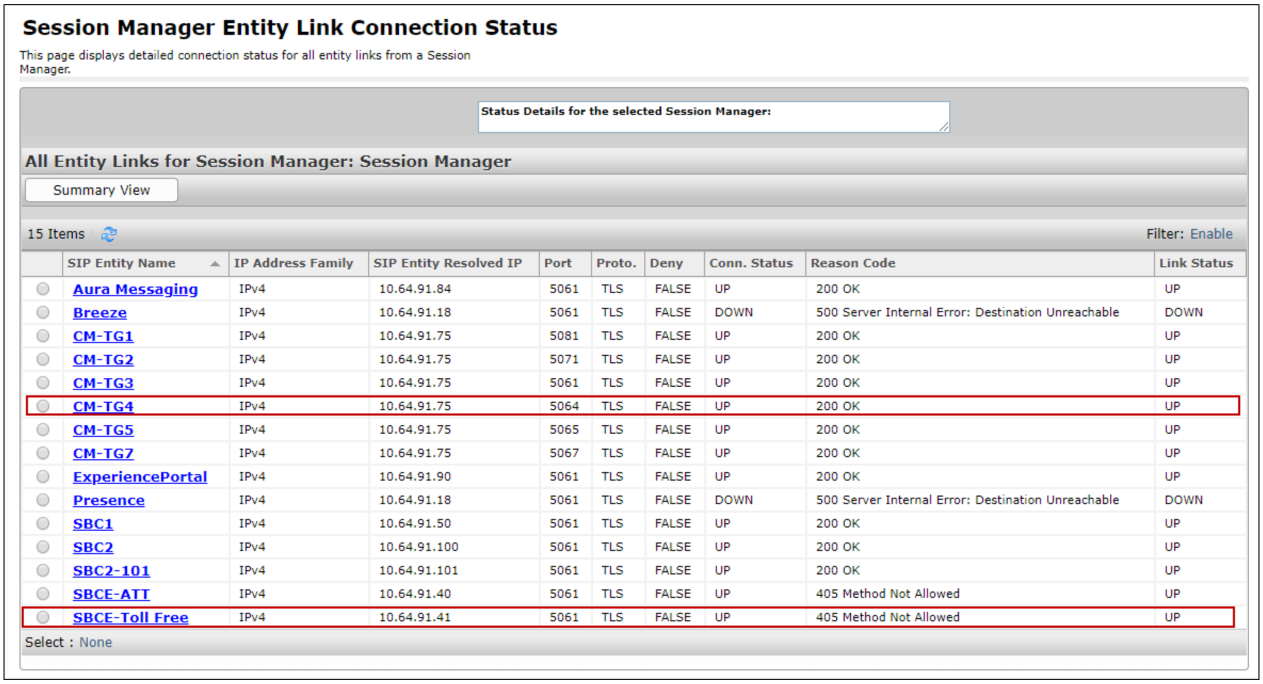
Using the procedures described in **Section 6**, access the System Manager GUI. From the **Home** screen, under the **Elements** heading, select **Session Manager**.



The Session Manager Dashboard is displayed. Verify that the **Tests Pass**, **Alarms**, **Service State**, and **Data Replication** columns, all show good status.



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



Session Manager Entity Link Connection Status

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

Status Details for the selected Session Manager:

All Entity Links for Session Manager: Session Manager

Summary View

15 Items | Filter: Enable

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

Select : None

Note – On the **SBCE-Toll Free** Entity from the list of monitored entities above, the **Reason Code** column indicates that Session Manager has received a SIP **405 Method Not Allowed** response to the SIP OPTIONS it generated. This response is sufficient for SIP Link Monitoring to consider the link up. Also note that the Avaya SBCE sends the Session Manager generated OPTIONS on to the AT&T IPTF Border Element, and it is the AT&T Border Element that is generating the 405 response, and the Avaya SBCE sends it back to Session Manager.

Other Session Manager useful verification and troubleshooting tools include:

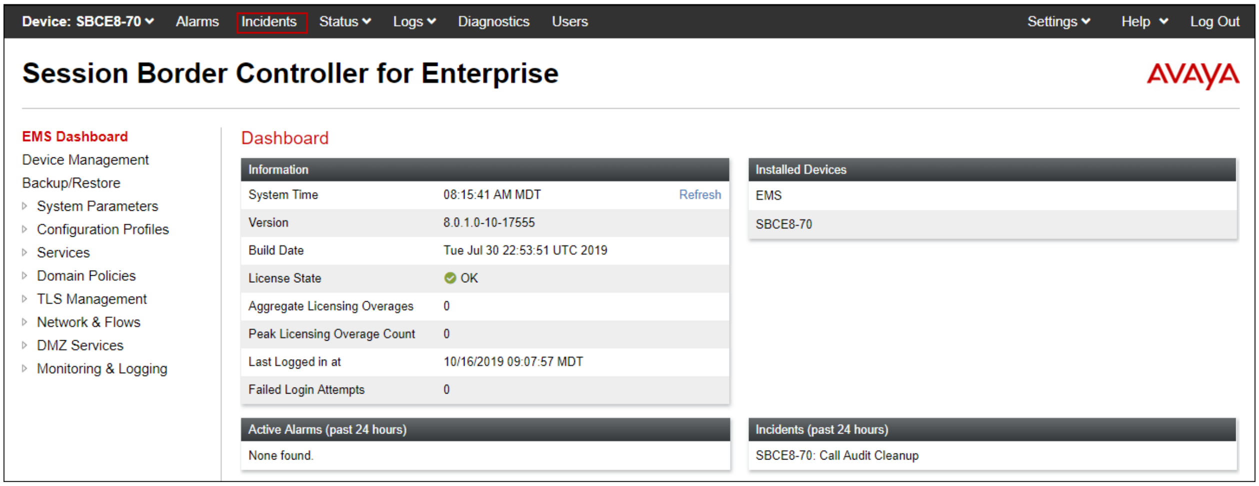
- **traceSM** – Session Manager command line tool for traffic analysis. Login to the Session Manager command line management interface to run this command.
- **Call Routing Test** - The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, from the System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

10.4. Avaya Session Border Controller for Enterprise Verification

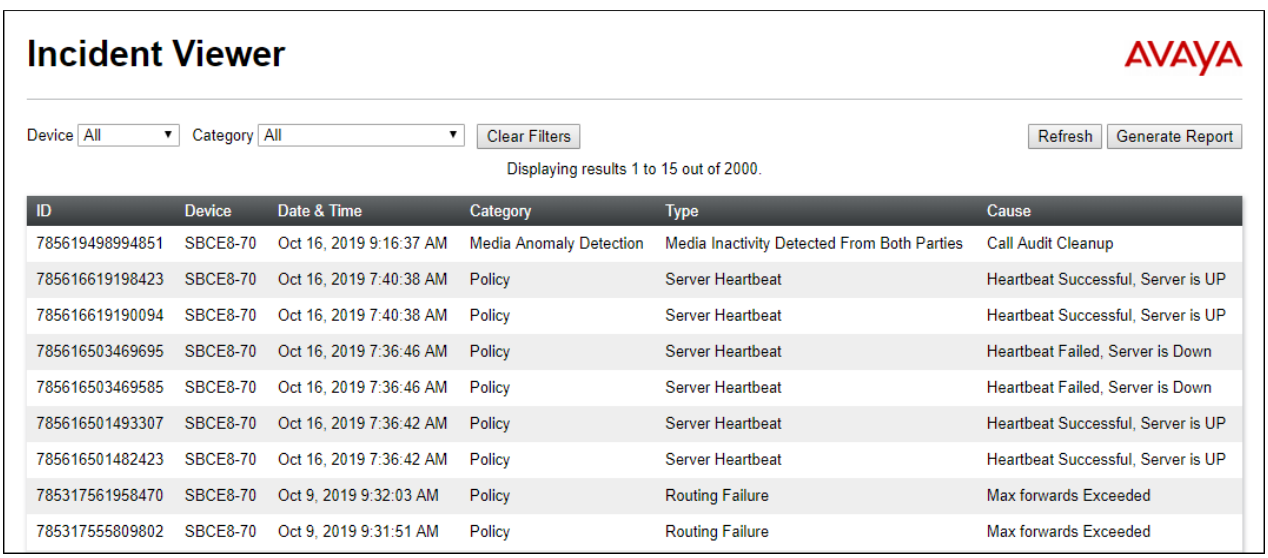
This section provides verification steps that may be performed with the Avaya SBCE.

10.4.1. Incidents

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



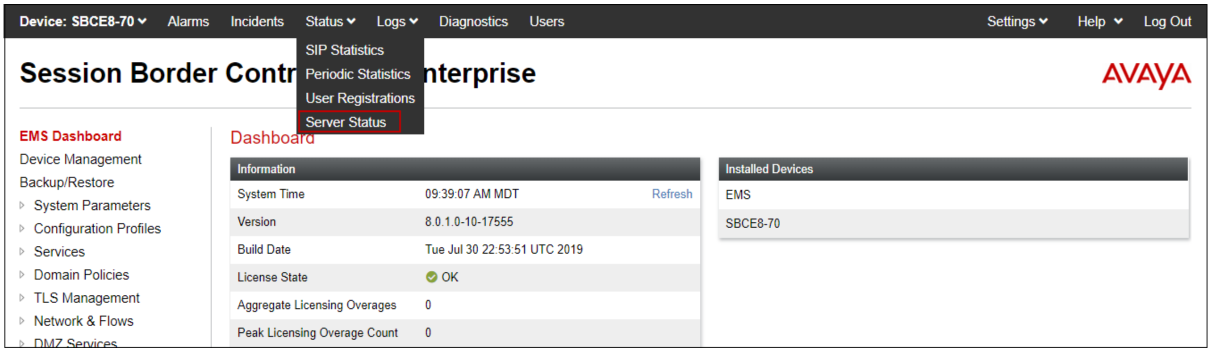
Use the Incident Viewer to verify Server Heartbeat and to troubleshoot routing failures. Further Information can be obtained by clicking on an incident in the incident viewer.



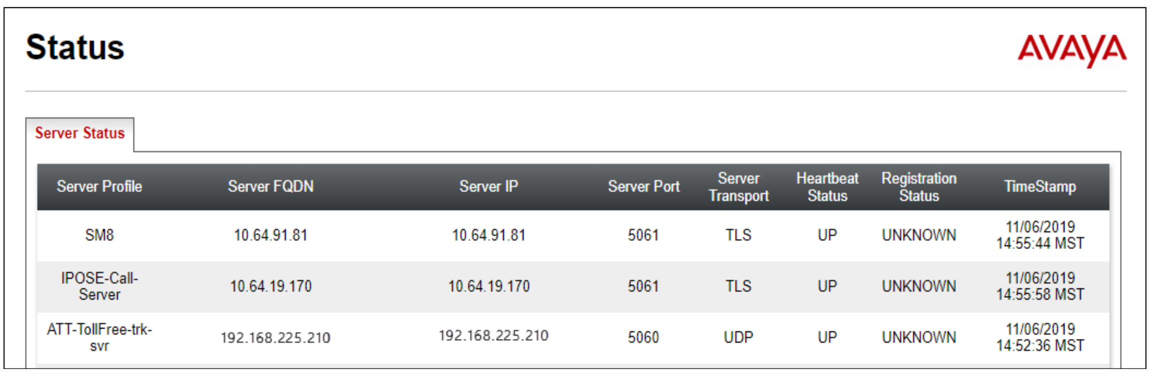
ID	Device	Date & Time	Category	Type	Cause
785619498994851	SBCE8-70	Oct 16, 2019 9:16:37 AM	Media Anomaly Detection	Media Inactivity Detected From Both Parties	Call Audit Cleanup
785616619198423	SBCE8-70	Oct 16, 2019 7:40:38 AM	Policy	Server Heartbeat	Heartbeat Successful, Server is UP
785616619190094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

10.4.2. Server Status

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



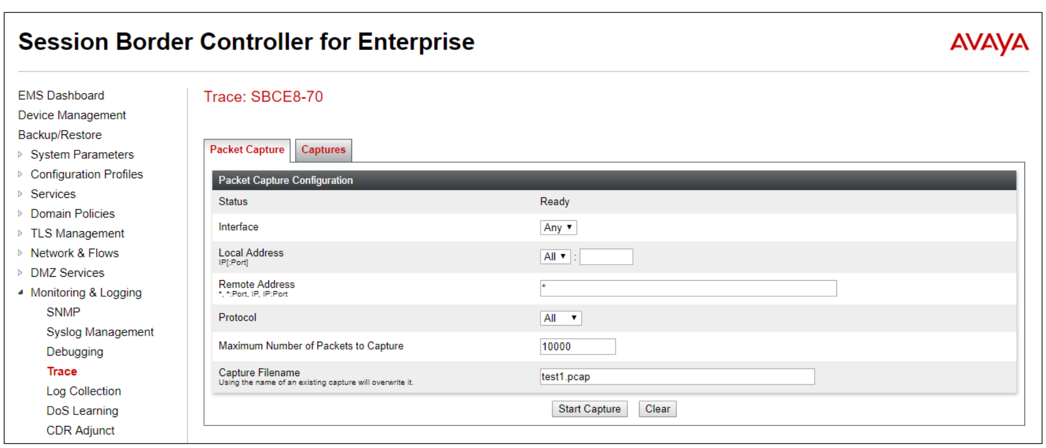
The **Server Status** screen provides information about the condition of the connection to the connected SIP Servers. This functionality requires Heartbeat to be enabled on the SIP Server Configuration profiles, as configured in **Section 8.9**.



Server Profile	Server FQDN	Server IP	Server Port	Server Transport	Heartbeat Status	Registration Status	TimeStamp
SM8	10.64.91.81	10.64.91.81	5061	TLS	UP	UNKNOWN	11/06/2019 14:55:44 MST
IPOSE-Call-Server	10.64.19.170	10.64.19.170	5061	TLS	UP	UNKNOWN	11/06/2019 14:55:58 MST
ATT-TollFree-trk-svr	192.168.225.210	192.168.225.210	5060	UDP	UP	UNKNOWN	11/06/2019 14:52:36 MST

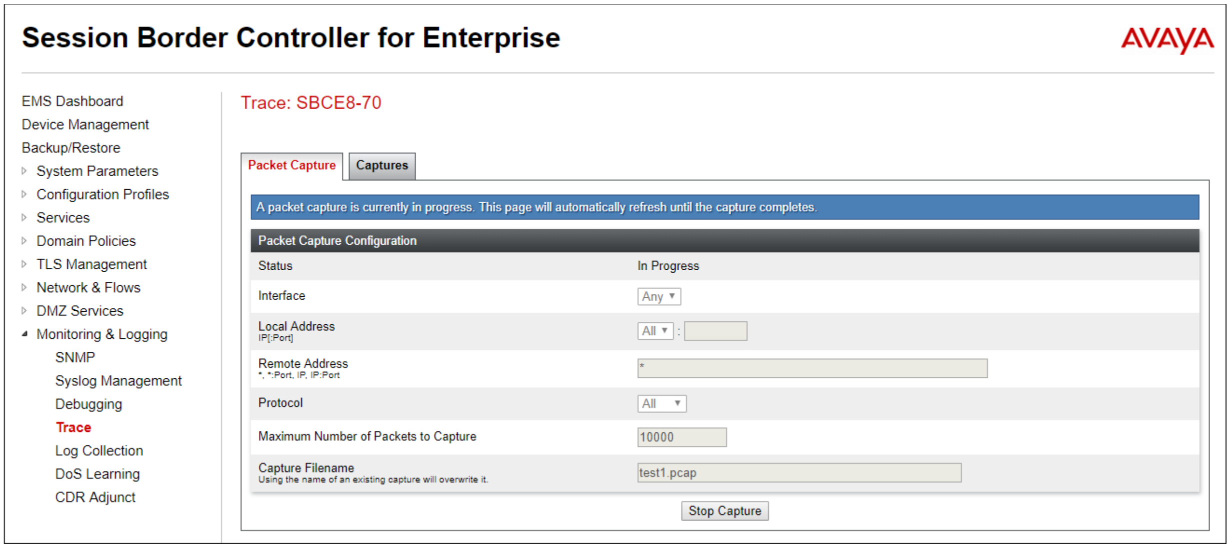
10.4.3. Protocol Traces

The Avaya SBCE can take internal traces of specified interfaces. To take a call trace, navigate to **Monitoring & Logging → Trace** and select the **Packet Capture** tab. Populate the fields for the capture parameters and click **Start Capture** as shown below.

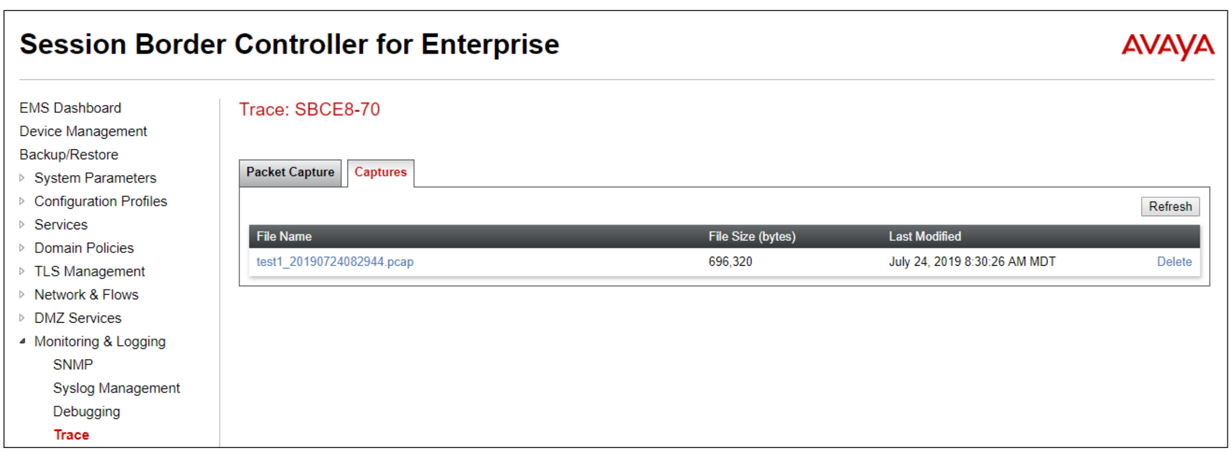


Note – Specifying **All** in the **Interface** field will result in the Avaya SBCE capturing traffic from both the A1 and B1 interfaces defined in the reference configuration. Also, when specifying the **Maximum Number of Packets to Capture**, estimate a number large enough to include all packets for the duration of the test.

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



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



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

11. Conclusion

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

Testing was performed on a simulated AT&T IP Toll Free service circuit. The reference configuration shown in these Application Notes is intended to provide configuration guidance to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program.

12. References

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

Avaya Aura® Session Manager/System Manager

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

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

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

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

Avaya Aura® Communication Manager

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

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

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

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

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

[10] *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010

Avaya Session Border Controller for Enterprise

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

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

Avaya Aura® Experience Portal

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

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

AT&T IP Toll Free Service

[15] *AT&T IP Toll Free Service – Product Description* https://www.business.att.com/products/ip-toll-free.html

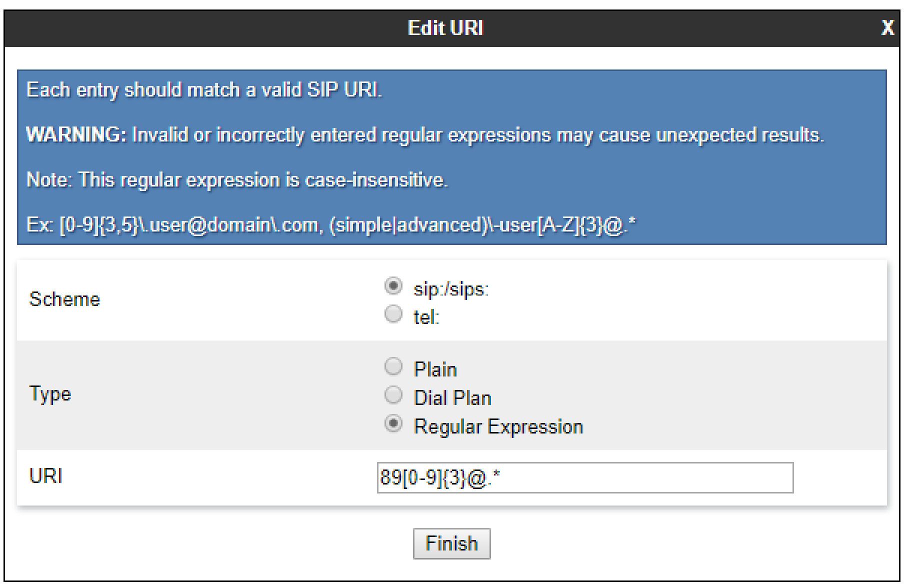
13. Appendix A – Refer Handling by Avaya SBCE

One of the important capabilities to the Experience Portal environment is the Avaya SBCE Refer Handling option. As described in **Section 3.2.2**, Experience Portal inbound call processing may include call redirection to Communication Manager agents, or other CPE destinations. This redirection is accomplished by having Experience Portal send SIP REFER messaging to the Avaya SBCE. Enabling the Refer Handling option causes the Avaya SBCE to intercept and process the REFER and generate a new SIP INVITE messages back to the CPE (e.g., Communication Manager).

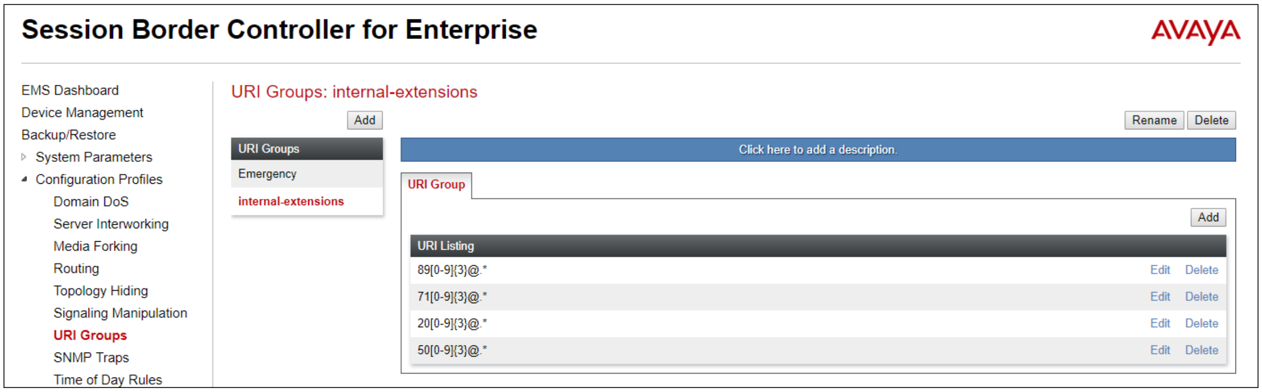
As an additional option, the Refer Handling feature can also specify *URI Group* criteria as a discriminator, whereby SIP REFER messages matching the URI Group criteria are processed by the Avaya SBCE, while SIP REFER messages that do not match the URI Group criteria, are passed through to AT&T.

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

- **Scheme**: **sip:/sips:**
- **Type: Regular Expression**
- **URI: 89[0-9]{3}@.*** This will match 5-digit local extensions starting with 89, e.g., 89001.
- Select **Finish**.



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

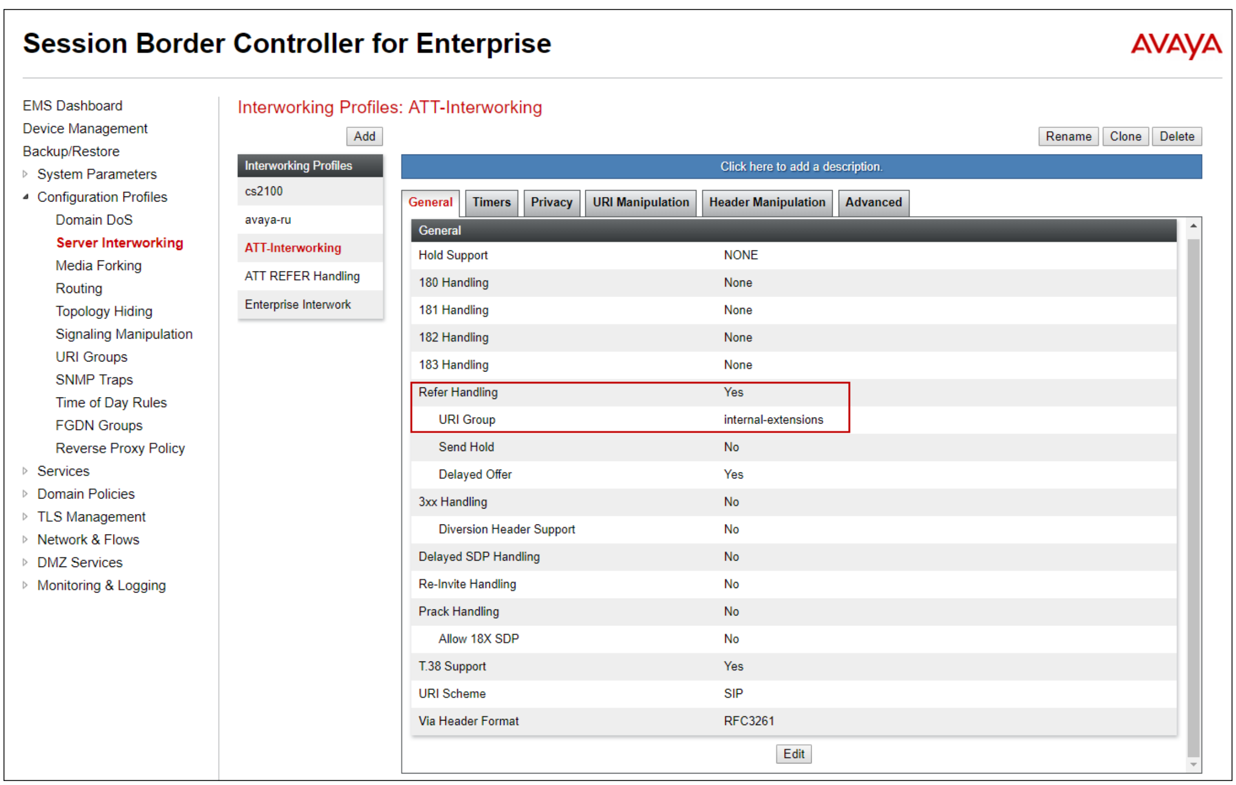


Edit the existing AT&T Server Interworking Profile to enable Refer Handling and assign the newly created URI Group.

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

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

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



Note that with the Refer Handling option enabled on the Avaya SBCE, the INVITE generated by the SBCE towards the 5 digit internal extensions is routed by Session Manager to Communication Manager via the local trunk (Trunk Group 3 in the reference configuration), not on the trunk group assigned for inbound calls from AT&T (Trunk Group 4). See **Section 5.8**. Depending on the codec priorities listed on the IP Codec Sets on the network regions associated to each trunk, this may cause the codec originally negotiated on the inbound call from AT&T to Experience Portal (i.e., G729A) to be re-negotiated to a different codec (i.e., G711U), once the call is transferred by Experience Portal to an internal extension in Communication Manager.

To avoid the issue described above, use the **change-ip-network-map** in Communication Manager to assign the IP address of the internal interface of the Avaya SBCE (10.61.91.41) to the network region associated to inbound calls from AT&T (e.g., network region 4, **Section 5.6.2**). With this setting, all calls arriving from the inside interface of the Avaya SBCE to a Communication Manager destination will be associated to network region 4, which uses IP Codec Set 4 for calls between region 4 (IPTF calls) and region 1 (the rest of the CPE).

Step 1 - Enter the **change ip-network-map** command in Communication Manager and enter the following:

- **FROM / TO**: **10.64.91.41**
- **Subnet Bits**: **32**
- **Network Region**: **4**

```
change ip-network-map                                          Page   1 of  63
                              IP ADDRESS MAPPING

                                                 Subnet Network   Emergency
 IP Address                                      Bits   Reg  VLAN Location Ext
 ----------------------------------------------- ------ ---- ---- ----------------
  FROM: 10.64.91.41                               /32    4    n
    TO: 10.64.91.41
  FROM:                                           /           n
    TO:
```

14. Appendix B – Configuration for G.711 Fax Testing

During the compliance test, in order to perform G.711 pass-through fax testing, the network region assigned to the G430 Media Gateway where the fax machine was connected was changed from region 1 (**Section 5.14**) to region 3. This network region utilized IP Codec Set 3 for calls between region 3 and region 4 (IPTF calls). Creating a dedicated network region and ip-codec-set for G.711 pass-through fax allowed for fax calls from this G430 Media Gateway to begin with codec G.711MU, while voice calls to other Media Gateways, Media Servers, and IP endpoints belonging to region 1, will continue to request G.729A as the first codec choice. (**Section 5.7.2**).

This configuration is shown here for completeness and is only needed if G.711 pass-through is preferred to T.38 fax. See **Section 2.2** for limitations.

To create the IP Network Region 3 used for G.711 fax testing, repeat the steps in **Section 5.6.1** with the following changes:

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

- Enter a descriptive name (e.g., **G711 Fax**).
- Enter **3** for the **Codec Set** parameter.

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

- Set codec set **3** for **dst rgn 4**.
- Note that **dst rgn 3** is pre-populated with codec set **3** (from page 1 provisioning).

```
change ip-network-region 3                                  Page   4 of  20

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

Repeat the steps in **Section 5.7.1** to create IP Codec Set 3 with the following changes:

Step 1 - On **Page 1** of the form

- Provision the codecs in the order shown below. Note that **G.711MU** is listed as the preferred codec.
- Set **Frames Per Pkt** to **3**. This will auto-populate **30** for the **Packet Size (ms)** field, and specify a PTIME value of 30 in the SDP.

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

- Set the **Fax Mode** to **off**.

```
change ip-codec-set 3                                           Page   1 of   2

                          IP CODEC SET
    Codec Set: 3
    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          3        30
 2: G.729A           n          3        30
 3: G.729B           n          3        30

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

change ip-codec-set 3                                           Page   2 of   2
                          IP CODEC SET
                              Allow Direct-IP Multimedia? n
                                                                       Packet
                          Mode                   Redundancy            Size(ms)
    FAX                   off                     0
    Modem                 off                     0
    TDD/TTY               US                      3
    H.323 Clear-channel   n                       0
```

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