Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1 and Avaya Session Border Controller for Enterprise 8.0 with AAPT SIP Voice SIP Trunking Service - Issue 1.0

Abstract

These Application Notes illustrate a sample configuration of Avaya Aura® Communication Manager Release 8.1 and Avaya Aura® Session Manager 8.1 with SIP Trunks to Avaya Session Border Controller for Enterprise (Avaya SBCE) 8.0 when used to connect the AAPT SIP Voice SIP Trunking Service available from AAPT (Australia).

Avaya Aura® Session Manager 8.1 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 8.1 is a telephony application server. Avaya Session Border Controller for Enterprise 8.0 is the point of connection between the Enterprise and the AAPT SIP Voice SIP Trunking service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

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

AAPT is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

Table of Contents

1. Introduction

These Application Notes illustrate a sample configuration Avaya Aura® Communication Manager Release 8.1 and Avaya Aura® Session Manager 8.1 with SIP Trunks to Avaya Session Border Controller for Enterprise (Avaya SBCE) when used to connect the AAPT SIP Voice SIP Trunking Service available from AAPT (Australia).

Avaya Aura® Session Manager 8.1 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 8.1 is a telephony application. Avaya SBCE is the point of connection between the Enterprise and the AAPT SIP Voice SIP Trunking Service and is used to not only secure the SIP trunk, but also to make adjustments to VoIP traffic for interoperability.

The SIP Voice SIP Trunking Service available from AAPT is one of the SIP-based Voice over IP (VoIP) services offered to enterprises in Australia for a variety of voice communications needs. The AAPT SIP Voice SIP Trunking Service allows enterprises in Australia to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

Purely as an example, the lab setup is configured in a non-redundant configuration (single Avaya Aura® Communication Manager, single Avaya Aura® Session Manager and a single Avaya SBCE). Additional resiliency could be built in as per the standard supported configurations documented in other Avaya publications.

On the private (enterprise) side, the Avaya Aura® Communication Manager "Processor Ethernet" or "procr" interface of Avaya Aura® Communication Manager is configured for SIP Trunking and is a SIP entity with associated SIP entity links in Avaya Aura® Session Manager. Additionally, Avaya SBCE is also configured as a SIP entity and has associated SIP entity links assigned within Avaya Aura® Session Manager.

In the documented example, the "Processor Ethernet" of the Avaya server running Avaya Aura® Communication Manager 8.1 is configured for SIP Trunking to Avaya Aura® Session Manager and Avaya SBCE is utilizing TCP transport. Avaya SBCE is connected to the AAPT SIP Voice SIP Trunking Service, and the SIP signaling connectivity from Avaya SBCE toward AAPT uses UDP.

Avaya SBCE performs security and topology-hiding at the enterprise edge. In the sample configuration, all SIP signaling and RTP media between the enterprise and the AAPT SIP Voice SIP Trunking Service solution flow through the Avaya SBCE.

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 AAPT SIP Voice and Customer Premises Equipment (CPE) containing Communication Manager, Session Manager, and Avaya SBCE (see **Section 3** for lab diagram).

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.

2.1 Interoperability Compliance Testing

The compliance testing was based on a standard Avaya GSSCP test plan. The testing covered functionality required for compliance as a solution supported on the AAPT SIP Voice network. Calls were made to and from the PSTN across the AAPT SIP Voice network. The following standard features were tested as part of this effort:

- PSTN incoming and outgoing calls to/from various phone types supported by Avaya IP Office including H.323, SIP, analog and digital stations; and Avaya Equinox Softphone
- Passing of DTMF events and their recognition by navigating automated menus (interacting with Avaya Aura® Messaging 7.1)
- PBX features such as hold, resume, conference and transfer
- G.711A audio
- Network Call Redirection
- Dialing plan including local 8-digit number and 10-digit Full Nation Number (FNN), international number
- Caller ID presentation and restriction
- Basic Call Center scenarios
- Faxing (G.711 pass-through)
- EC500 – call extending to mobile
- Remote Worker scenarios

2.2 Test Results

Interoperability testing of AAPT SIP Voice Service was completed with successful results for all test cases with the exception of the observations/limitations described below.

Please refer to the test case document for a complete list of solution issues found when tested.

- **Calling / Called Party Identity:** In untrusted / authentication required configuration, AAPT may send hexadecimal string in **Contact** header, and may not send **P-Asserted-Identity**. This results in improper calling / called party identity displayed on Avaya clients. This also causes ec500 mapping failure, e.g., Communication Manager could not associate the mobile number with the host extension. An Avaya SBCE Signaling Manipulation (SigMa) script was used to overcome the issues. The purpose of the script is to add the **P-Asserted-Identity** header (if not existed) to the SIP requests/responses from/to AAPT SIP Voice Service. The content of the **P-Asserted-Identity header** is copied from **From/To** headers. See **Section 7.3** for how to create the script.
- **Call Forward** – No ring back tone on PSTN phone when Aura SIP extension set forward all call to another extension. The SIP trace is showing that Communication Manager firstly sends **181 Call is being forwarded** with **100rel** in **Require** header. Therefore, AAPT SIP Voice sends **PRACK**, and is expecting **200OK** for that **PRACK**. However, before sending **200OK** for the **PRACK**, Communication Manager sends **180 Ringing** with **100rel** in the **Require** header. AAPT SIP Voice could not handle this out-of-order **180 Ringing** at this point to generate the ringback tone to the PSTN caller. The issue is being investigated by both AAPT and Avaya.
- **EC500 service with Confirmed Answer enabled -** With Initial IP-IP Direct Media enabled on the SIP signaling group toward to AAPT SIP Voice SIP Trunking Service, the EC500 call leg is established with no voice as soon as EC500 user answers the call on mobile. This results in a call drop after the confirmation timeout (default to 10 seconds). If EC500 service with Confirmed Answer setting is required, the **Initial IP-IP Direct Media** must be disabled on the signaling group which is used for (or shared with) EC500 service.
- **Avaya Network Call Redirection (NCR) is recommended to be disabled** (default) on the Communication Manager SIP trunk group to the AAPT SIP Voice SIP Trunking Service. With NCR is enabled, in call transfer / conference scenarios, AAPT stops media by sending a re-INVITE, followed immediately by a BYE, and does not wait for a complete dialog. No obvious end-user impact was observed during compliance test. However, race condition issues are expected (e.g., Avaya endpoint does not get notified of transfer status properly).

2.3 Support

- **Avaya:** Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com
- **AAPT:** Customers should contact their AAPT Business representative or follow the support links available on http://www.aapt.com.au

3. Reference Configuration

The reference configuration used in these Application Notes is shown in the diagram below and consists of several components.

- Avaya Aura® Communication Manager running on Virtualized Environment.
- Avaya Aura® Session Manager running on Virtualized Environment.
- Avaya Aura® System Manager running on Virtualized Environment.
- Avaya Aura® Messaging running on Virtualized Environment.
- Avaya G450 Media Gateway.
- Avaya Aura® Media Server running on Virtualized Environment. The Media Server can act as a media gateway Gxxx series.
- Avaya IP phones are represented with Avaya 9600/1600 Series IP Telephones running H.323/SIP software.
- Avaya one-X® Communicator 6.2
- Avaya Equinox for Windows 3.5
- Avaya SBCE provided Session Border Controller functionality, including, Network Address Translation, SIP header manipulation, and Topology Hiding between the AAPT SIP Voice SIP Trunking Service and the enterprise internal network.

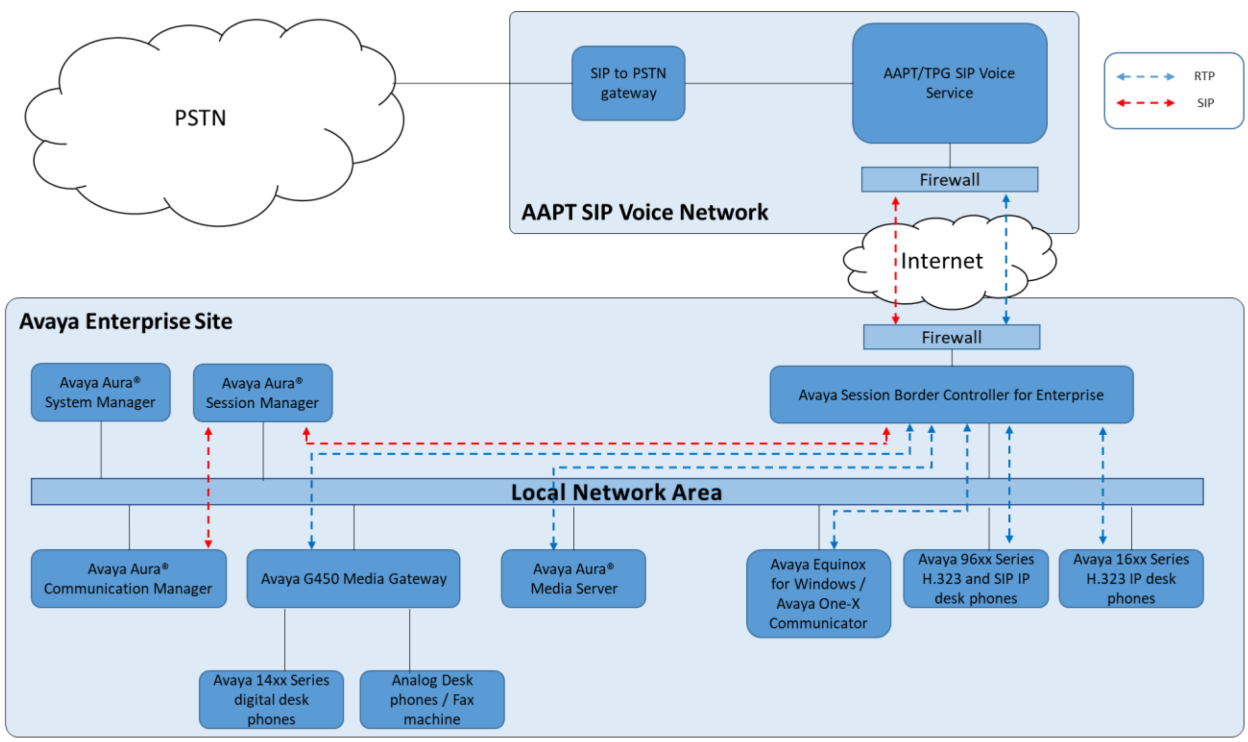


Figure 1: Network Components as Tested

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Communication Manager	8.1.0.0.890-25393
Avaya Aura® Session Manager	8.1.0.0.810007
Avaya Aura® System Manager	Build No. - 8.1.0.0.733078 Software Update Revision No: 8.1.0.0.9814
Avaya Aura® Messaging	7.1.0.0.532
Avaya Session Border Controller for Enterprise	8.0.0.0-19-16991
Avaya G450 Media Gateway	g450_sw_41_9_0
Avaya Aura Media Server	8.0.0.205
Avaya one-X® Communicator	6.2.13.2
Avaya Equinox for Windows	3.5.7.30.1
Avaya one-X® Agent H323	2.5.60313.0
Avaya 96x1 series – SIP Deskphones	7.1.5
Avaya 96xx series – H.323 Deskphones	3.2.8
Avaya 16xx series – H.323 Deskphones	1.3.12
Service Provider – AAPT SIP Voice	
Metaswith cCFS (Softswitch)	V9.4.10_SU5_P90.00
Metaswitch Perimera ISC (SBC)	V4.3.20_SU4_P1203

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.

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. The Communication Manager SAT console, the System Manager Web UI and the Avaya SBCE Web UI captured in this sections are displaying the configuration those have been configured earlier. The actual Communication Manager SAT commands, the System Manager Web UI and the Avaya SBCE Web UI to create/add the configurations may vary.

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.

Follow the steps shown below:

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      1
         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     1
                   Maximum Administered SIP Trunks:  4000     10
  Max Administered Ad-hoc Video Conferencing Ports:  4000      0
   Max Number of DS1 Boards with Echo Cancellation: 80     0
```

2. On **Page 6** of the form, verify that the **Private Networking** and **Processor Ethernet** fields are 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? y      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? n             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

Follow the steps shown below:

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**.

```
display system-parameters features                              Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    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? no
              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
```

2. On **Page 9** verify that a text string has been defined to replace the **Calling Party Number (CPN)** for restricted or unavailable calls. The compliance test used the value of **Restricted** for restricted calls and **Unavailable** for unavailable calls.

```
display system-parameters features                              Page   9 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
   CPN/ANI/ICLID Replacement for Restricted Calls: Restricted
  CPN/ANI/ICLID Replacement for Unavailable Calls: Unavailable

DISPLAY TEXT
                                       Identity When Bridging: principal
                                        User Guidance Display? n
 Extension only label for Team button on 96xx H.323 terminals? n

INTERNATIONAL CALL ROUTING PARAMETERS
                 Local Country Code:
          International Access Code:

SCCAN PARAMETERS
   Enable Enbloc Dialing without ARS FAC? n

CALLER ID ON CALL WAITING PARAMETERS
     Caller ID on Call Waiting Delay Timer (msec): 200
```

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.

Follow the steps shown below:

- Enter the **change dialplan analysis** command to provision the following dial plan.
  - 4-digit extensions with a **Call Type** of **ext** beginning with:
    - The digits **68** for Communication Manager extensions (which is assigned by AAPT as DID numbers).
  - 3-digit dial access code (indicated with a **Call Type** of **dac**), e.g., access code ***** for SIP Trunk Access Codes (TAC).

```
display dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 2

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
 000           3  udp
 1300          10 udp
 18            10 udp
 68            4  ext
 9             1  fac
 *             3  dac
 #             4  fac
```

5.4 IP 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 Entity in **Section 6.3.2**.

Follow the steps shown below:

- 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-ve** and **10.1.20.7**).
  - Avaya Media Server interface (e.g., **ams-ve** and **10.1.20.12**).

```
display node-names ip                                           Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
ams-ve              10.1.20.12
default             0.0.0.0
procr               10.1.20.10
procr6              ::
sm-ve               10.1.20.7
```

5.5 IP Interface for Procr

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 fields.

```
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: 10.1.20.10
```

5.6 IP Network Regions

For the compliance testing, ip-network-region 1 was created by the **change ip-network-region 1** command with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In the compliance testing, the domain name is **sipinterop.net**. This domain name appears in the "From" header of SIP message originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway / Avaya Media Server. By default, both **Intra-region** and **Inter-region IP-IP Direct Audio** are set to **yes**. Shuffling can be further restricted at the trunk level under the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.7**.
- Default values can be used for all other fields.

```
display ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: sipinterop.net
    Name: AAPT                Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 53999
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
```

On **Page 4**, define the IP codec set to be used for traffic in region 1. In the compliance testing, Communication Manager, the G450 Media Gateway, Media Server, IP/SIP extensions, Session Manager and Avaya SBCE were assigned to the same region 1. To configure the IP codec set between regions, enter the desired IP codec set in the **codec set** column of the table with appropriate destination region (**dst rgn**). Default values may be used for all other fields.

```
display 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
 3
 4
 5
 6
 7
 8
 9
 10
 11
 12
 13
```

Non-IP telephones (e.g., analog, digital) derive their network region from the IP interface of the G450 Media Gateway to which the device is connected. IP telephones can be assigned a network region based on an IP address mapping.

To define network region 1 for IP interface **procr**, use **change ip-interface procr** command as shown in the following screen.

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


                 Type: PROCR
                                                      Target socket load: 19660

     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: 10.1.20.10


          Subnet Mask: /24
```

To define network region 1 for the G450 Media Gateway, use **change media-gateway** command as shown in the following screen.

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

                    Type: g450
                    Name: g450
               Serial No: 10IS11367055
    Link Encryption Type: any-ptls/tls       Enable CF? n
          Network Region: 1                   Location: 1
                                             Site Data:
           Recovery Rule: none

              Registered?  y
   FW Version/HW Vintage: 41 .9  .0  /2
       MGP IPV4 Address: 10.1.20.20
       MGP IPV6 Address:
   Controller IP Address: 10.1.20.10
             MAC Address: 00:1b:4f:3e:a5:e0

   Mutual Authentication? optional
```

5.7 IP Codec Parameters

Follow the steps shown below:

1. Enter the **change ip-codec-set x** command, where **x** is the number of the IP codec set specified in **Section 5.6**. On **Page 1** of the **ip-codec-set** form, ensure that **G.711A** and **G.711MU** are included in the codec list. Note that the packet interval size will default to 20ms.

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

                          IP MEDIA PARAMETERS
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711A           n         2        20
 2: G.711MU          n         2        20
 3:
 4:
 5:
 6:
 7:
```

2. On **Page 2** of the ip-codec-set form, set **pass-through** for G.711 pass-through mode**.**

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

                          IP MEDIA PARAMETERS

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

                                            Redun-                      Packet
                          Mode              dancy                       Size(ms)
    FAX                   pass-through      0
    Modem                 off               0
    TDD/TTY               US                3
    H.323 Clear-channel   n                 0
    SIP 64K Data          n                 0                           20
```

5.8 SIP Trunks

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

- SIP Voice SIP Trunking service access – SIP Trunk group 1
- Internal CPE access (ie: Avaya SIP extension) – SIP Trunk group 3

5.8.1 SIP Trunk for SIP Voice SIP Trunking service access

This section describes the steps for administering the SIP trunk to Session Manager. This trunk corresponds to the **cm-ve** SIP Entity defined in **Section 6.3.2**.

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

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tcp**.
- 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-ve**).
- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5060**
- **Far-end Network Region** – Set the IP network region to **1**, as set in **Section 5.6**.
- **Far-end Domain** – Enter **sipinterop.net**. This is the domain provisioned for Session Manager in **Section 6.1**.
- **DTMF over IP** – Set to **rtp-payload** to enable Communication Manager to use DTMF according to RFC 2833.

- **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized directly to the associated stations, to reduce the use of media resources on the Avaya Media Gateway / Avaya Media Server when possible (known as shuffling).
- **Enable Layer 3 Test** – Set to **y**. This directs Communication Manager to send SIP OPTIONS messages to Session Manager to check link status.
- **Initial IP-IP Direct Media – Set** to **y**, indicating that the RTP paths should be initially direct between Avaya SIP stations and the internal interface of ASBCE, to reduce the use of media resources on the Avaya Media Gateway / Avaya Media Server.
- **H.323 Station Outgoing Direct Media** – Set to **y**, indicating that the RTP paths should be also initially direct for the H.323 stations, to avoid the use of media resources on the Avaya Media Gateway / Avaya Media Server.
- Default values may be used for all other fields.

```
display signaling-group 1                                       Page   1 of   3
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? y          Priority 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-ve
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1

Far-end Domain: sipinterop.net
                                             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? y
H.323 Station Outgoing Direct Media? y            Alternate Route Timer(sec): 6
```

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

- **Group Type** – Set to **sip**.
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., ***01**).
- **Direction** – Set to **two-way**.
- **Service Type** – Set to **public-ntwrk**.
- **Signaling Group** – Set to the signaling group administered in **Section 5.8.1** (e.g., **1**).
- **Number of Members** – Enter the maximum number of simultaneous calls desired on this trunk group (based on licensing) (e.g., **10**).

```
display trunk-group 1                                           Page   1 of   4
                                TRUNK GROUP

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: AAPT-Trunk              COR: 1        TN: 1        TAC: *01
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 1
                                                    Number of Members: 10
```

On **Page 3**, set the **Numbering Format** field to **private**. This field specifies the format of the calling party number (CPN) sent to the far-end.

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to **y**. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.2**, if the inbound call enabled CPN block. For outbound calls, these same settings request that CPN block be activated on the far-end destination if a local user requests CPN block on a particular call routed out this trunk. Default values were used for all other fields.

```
display trunk-group 1                                           Page   3 of   4
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: public
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? y
                                               Replace Unavailable Numbers? y

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

On **Page 4**, set the **Network Call Redirection** field should be set to **n**. Setting the **Network Call Redirection** flag to **y** enables use of the SIP REFER message for call transfer; otherwise the SIP INVITE message will be used for call transfer. Refer **Section 2.2** for observations with **Network Call Redirection** / SIP REFER enabled.

Set the **Send Diversion Header** field to **y** and the **Support Request History** field to **n**. The **Send Diversion Header** field provides additional information to the network if the call has been redirected. These header modifications are needed to support the call display for call forwarding of inbound calls back to the PSTN and some Extension to Cellular (EC500) call scenarios.

```
display trunk-group 1                                            Page   4 of   4
                              PROTOCOL VARIATIONS

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

                                                  Send Diversion Header? y
                                                Support Request History? n
                                          Telephone Event Payload Type: 101


                                     Convert 180 to 183 for Early Media? n
                               Always Use re-INVITE for Display Updates? y
                                    Identity for Calling Party Display: From
                         Block Sending Calling Party Location in INVITE? n
                              Accept Redirect to Blank User Destination? y
                                                         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 SIP Trunk for Internal CPE access (Avaya SIP extensions)

This section describes the steps for administering the SIP trunk to Session Manager. This trunk corresponds to the **cm-ve-optim** SIP Entity defined in **Section 6.3.3**.

Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **1**), 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-ve**).
- **Near-end Listen Port** and **Far-end Listen Port** – Set to **5061**
- **Far-end Network Region** – Set the IP network region to **1**, as set in **Section 5.6**.

- **Far-end Domain** – Enter **sipinterop.net**. This is the domain provisioned for Session Manager in **Section 6.1**.
- **DTMF over IP** – Set to **rtp-payload** to enable Communication Manager to use DTMF according to RFC 2833.
- **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized directly to the associated stations, to reduce the use of media resources on the Media Gateway / Media Server when possible (known as shuffling).
- **Enable Layer 3 Test** – Set to **y**. This directs Communication Manager to send SIP OPTIONS messages to Session Manager to check link status.
- **Initial IP-IP Direct Media – Set to y**, indicating that the RTP paths should be initially direct between Avaya SIP stations and the internal interface of ASBCE, to the use of media resources on the Media Gateway / Media Server.
- **H.323 Station Outgoing Direct Media** – Set to **y**, indicating that the RTP paths should be also initially direct for the H.323 stations, to avoid the use of media resources on the Avaya Media Gateway / Avaya Media Server.
- Default values may be used for all other fields.

```
display signaling-group 3                                       Page   1 of   3
                                SIGNALING GROUP

 Group Number: 1             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority 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-ve
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain: sipinterop.net
                                              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? y
H.323 Station Outgoing Direct Media? y             Alternate Route Timer(sec): 6
```

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

- **Group Type** – Set to **sip**.
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., ***03**).
- **Direction** – Set to **two-way**.
- **Service Type** – Set to **tie**.
- **Signaling Group** – Set to the signaling group administered in **5.8.1** (e.g., **3**).
- **Number of Members** – Enter the maximum number of simultaneous calls desired on this trunk group (based on licensing) (e.g., **10**).

```
display trunk-group 3                                           Page   1 of   4
                                TRUNK GROUP

Group Number: 3                   Group Type: sip          CDR Reports: y
  Group Name: SIP-OPTIM                  COR: 1        TN: 1        TAC: *03
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 3
                                                     Number of Members: 10
```

On **Page 3**, set the **Numbering Format** field to **private**. This field specifies the format of the calling party number (CPN) sent to the far-end. Default values may be used for all other fields.

```
display trunk-group 1                                           Page   3 of   4
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                     Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

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

On **Page 4**, Set **Telephone Event Payload Type** to **101** to be consistent with the **Telephone Event Payload Type** of the SIP Trunk (Trunk 1) toward AAPT SIP Voice. Default values may be used for all other fields

```
display trunk-group 3                                               Page   4 of   4
                              PROTOCOL VARIATIONS

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

                                Overwrite Calling Identity? 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
```

5.9 Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers.

Use the **change private-unknown-numbering** command to create an entry for each extension which has a DID assigned. The DID number will be assigned by the service provider. It is used to authenticate the caller.

In the sample configuration, the DID numbers provided for testing were assigned to the extensions 68xx. Thus, these same DID numbers were used in the outbound calling party information on the service provider trunk (trunk 1) when calls were originated from these extensions.

Use the **change public-unknown-numbering** command to create an entry for each extension which has a DID assigned. The DID number will be assigned by the SIP service provider. It is used to authenticate the caller. AAPT will accept the full E.164 string including + prefix. The following example is to construct an E.164 11-digit calling number from 4-digit extension, '+' will be automatically inserted if the SIP Signaling group is connected to Session Manager. The actual number is masked with 'x' for security reason.

```
display public-unknown-numbering 0                            Page   1 of   2
                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                                 Total
Ext Ext              Trk       CPN                CPN
Len Code             Grp(s)    Prefix             Len
                                                     Total Administered: 3
 4  68               1         612xxxx            11    Maximum Entries: 240

                                                    Note: If an entry applies to
                                                     a SIP connection to Avaya
                                                     Aura(R) Session Manager,
                                                     the resulting number must
                                                     be a complete E.164 number.

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

In order to presenting calling party number to Avaya SIP extension in 4-digit internal format, an additional private-numbering entry is administered for the SIP extension access trunk (trunk group 3).

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

Ext Ext               Trk        Private          Total
Len Code              Grp(s)     Prefix           Len
 4  68                1                           4  Total Administered: 2
                                                        Maximum Entries: 540
```

5.10 Incoming Call Handling Treatment

In general, the incoming call handling treatment for a trunk group can be used to manipulate the digits received for an incoming call if necessary. The DID numbers sent by AAPT can be mapped to Communication Manager extensions using the incoming call handling treatment of the receiving trunk-group. Use the **change inc-call-handling-trmt trunk-group** command to create an entry for each DID.

```
display inc-call-handling-trmt trunk-group 1                  Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10  02xxxx          6
```

Note: the actual number is masked with "xxxx" for security reason.

5.11 Outbound Routing

In these Application Notes, the **Automatic Route Selection** (ARS) feature is used to route an outbound call via the SIP trunk to the service provider. In the compliance testing, a single digit 9 was used as the ARS access code. An enterprise caller will dial 9 to reach an outside line. To define feature access code (**fac**) **9**, use the **change dialplan analysis** command as shown below.

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all            Percent Full: 2

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
 000           3  udp
 1300          10 udp
 18            10 udp
 68            4  ext
 *             3  dac
 #             4  fac
 9             1  fac
```

Use the **change feature-access-codes** command to define **9** as the **Auto Route Selection (ARS) – Access Code 1**.

```
display feature-access-codes                                    Page   1 of  12
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2: 6
                 Automatic Callback Activation: #002   Deactivation: #003
Call Forwarding Activation Busy/DA: #004   All: #005   Deactivation: #006
   Call Forwarding Enhanced Status: #007   Act: #008   Deactivation: #009
                         Call Park Access Code: #010
                       Call Pickup Access Code: #011
CAS Remote Hold/Answer Hold-Unhold Access Code: #012
                  CDR Account Code Access Code: #013
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit **9**. The example below shows a subset of the dialed strings tested as part of the compliance testing. All dialed strings are mapped to route pattern **1** for an outbound call which contains the SIP trunk to the service provider (as defined next).

```
display ars analysis 0                                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 0

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    000                   3    3     1        emer         n
    0011                  12   20    1        pubu         n
    02                    10   10    1        pubu         n
    03                    10   10    1        pubu         n
    04                    10   10    1        pubu         n
    06                    10   10    1        pubu         n
    07                    10   10    1        pubu         n
    08                    10   10    1        pubu         n
    1300                  10   10    1        pubu         n
    18                    10   10    1        pubu         n
    xxxxxxxx              8    8     1        pubu         n
```

As mentioned above, the route pattern defines which trunk group will be used for the outbound calls and performs necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for **route pattern 1** in the following manner.

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the service provider. For the compliance testing, trunk group **1** was used.
- **FRL**: Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **Numbering Format**: **pub-unk**. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.9**.

```
display route-pattern 1                                         Page   1 of   4
                    Pattern Number: 1      Pattern Name: AAPT-EV-Route
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 1    0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    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                               pub-unk   none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
```

5.12 Avaya SIP Extension Routing

Route Patterns are used to direct calls to the local SIP trunk for access to SIP extensions or other destinations in the CPE. Use the **change route-pattern** command to configure the parameters for **route pattern 3** in the following manner.

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the trunk group for the Avaya SIP Extension Routing. For the compliance testing, trunk group **3** was used.
- **FRL**: Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **Numbering Format**: **lev0-pvt**. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.9**.

```
display route-pattern 3                                         Page   1 of   4
                    Pattern Number: 3      Pattern Name: SIP-OPTIM
    SCCAN? n    Secure SIP? n     Used for SIP stations? y
    Primary SM: sm-ve          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
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    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
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
```

5.13 Automatic Alternate Routing (AAR) Dialing

Use the **change aar analysis** command to configure the routing for Avaya SIP Extensions. The example below shows a subset of the SIP extensions used as part of the compliance testing.

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

          Dialed           Total     Route     Call    Node   ANI
          String          Min  Max  Pattern    Type    Num    Reqd
    68xx                  4    4     3         lev0           n
                                                              n
```

5.14 Avaya G450 Media Gateway Provisioning

In the reference configuration, a G450 Media Gateways is provisioned. The G450 Media Gateway is used for local DSP resources, announcements, Music On Hold, etc.

Note – Only the Media Gateway provisioning associated with the G450 registration to Communication Manager is shown below.

1. SSH to the G450 (not shown). Note that the Media Gateway prompt will contain *???* if the Media Gateway is not registered to Communication Manager (e.g., **g450-*???(super)#***).
2. Enter the **show system** command and note the G450 serial number (e.g., **10IS11367055**).
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., **10.1.20.10**).
4. Enter the **copy run copy start command** to save the G450 configuration.
5. On Communication Manager, enter the **add media-gateway x** command where x is an available Media Gateway identifier (e.g., **1**). The Media Gateway form will open (not shown).
   Enter the following parameters:
   - Set **Type** = **G450**.
   - Set **Name** = Enter a descriptive name (e.g., **g450**).
   - Set **Serial Number** = Enter the serial number copied from **Step 2**.
   - Set the **Encrypt Link** parameter as desired (**any-ptls/tls** was used in the reference configuration).
   - Set **Network Region** = **1**.

When the Media Gateway registers, the SSH connection prompt will change to reflect the Media Gateway Identifier assigned in **Step 5** (e.g., **g450-*001(super)#***).

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

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

                    Type: g450
                    Name: g450
               Serial No: 10IS11367055
    Link Encryption Type: any-ptls/tls       Enable CF? n
          Network Region: 1                   Location: 1
                                             Site Data:
           Recovery Rule: none

              Registered?  y
   FW Version/HW Vintage: 41 .9  .0  /2
        MGP IPV4 Address: 10.1.20.20
        MGP IPV6 Address:
   Controller IP Address: 10.1.20.10
             MAC Address: 00:1b:4f:3e:a5:e0

   Mutual Authentication? optional
```

5.15 Avaya Aura® Media Server Provisioning

In the reference configuration, a Media Server is provisioned. The Media Server is located in the enterprise and is used, along with the G450 Media Gateway, for local DSP resources, announcements, and Music On Hold.

Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **2**), and provision the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tls**.
- Verify that **Peer Detection Enabled?** is 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-ve**).
- **Near-end Listen Port –** Set to **9061**.
- **Far-end Listen Port** – Set to **5061**.
- **Far-end Network Region** – Set the IP network region to **1**, as set in **Section 5.6**.
- **Far-end Domain** – Automatically populated with the IP address of the Media Server.

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

 Group Number: 2              Group Type: sip
                        Transport Method: tls


  Peer Detection Enabled? n  Peer Server: AMS



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

Far-end Domain: 10.1.20.12
```

Enter the **add media-server x** command where **x** is an available Media Server identifier (e.g., **1**), and provision the followings:

- **Signaling Group** – Enter the signaling group previously configured for Media Server (e.g., **2**).
- **Voip Channel License Limit** – Enter the number of VoIP channels for this Media Server (based on licensing) (e.g., **10**).
- **Dedicated Voip Channel Licenses** – Enter the number of VoIP channels licensed to this Media Server (e.g., **10**).
- Remaining fields are automatically populated based on the signaling group provisioning for the Media Server.

```
display media-server 1
                                  MEDIA SERVER

                    Media Server ID: 1

                    Signaling Group: 2
         Voip Channel License Limit: 10
    Dedicated Voip Channel Licenses: 10

                          Node Name: ams-ve
                     Network Region: 1
                           Location: 1
          Announcement Storage Area: ANNC-b2bf4c0a-205a-41e8-84c1-000c2963b6c0
```

5.16 Save Communication Manager Translations

After the Communication Manager provisioning is completed, enter the command **save translation** (not shown).

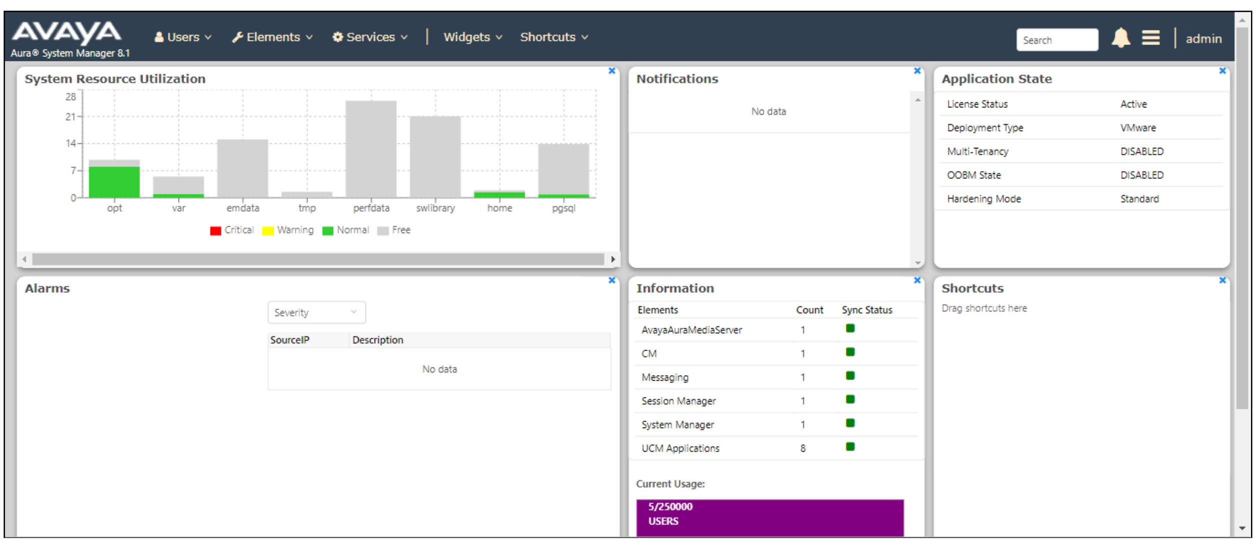
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain.
- Logical/physical Location that can be used by SIP Entities.
- SIP Entities corresponding to Communication Manager, Session Manager and Avaya SBCE.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Routing Policies, which control call routing between the SIP Entities.
- Dial Patterns, which govern to which SIP Entity a call is routed.
- Session Manager, corresponding to the Session Manager server to be managed by System Manager.

It may not be necessary to configure all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

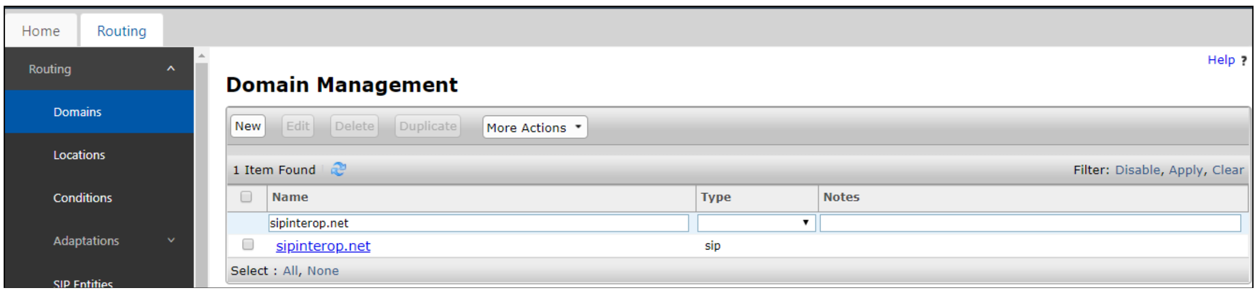
Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR,** where **<ip-address>** is the IP address of System Manager. In the **Log On** screen (not shown), enter appropriate **User ID** and **Password** and press the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



6.1 Configure SIP Domain

Follow the steps shown below:

1. Select **Domains** from the left navigation menu. In the reference configuration, domain **sipinterop.net** was defined.
2. Click **New** (not shown). Enter the following values and use default values for remaining fields.
   - **Name**: Enter the enterprise SIP Domain Name. In the sample screen below, **sipinterop.net** is shown.
   - **Type**: Verify **sip** is selected.
   - **Notes**: Add a brief description.
3. Click **Commit** to save (not shown).



6.2 Configure Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside. In the reference configuration, location **aapt** is configured.

Follow the steps shown below:

1. Select **Locations** from the left navigational menu. Click **New**. In the **General** section (not shown), enter the following values and use default values for remaining fields.
   - **Name**: Enter a descriptive name for the Location (e.g., **aapt**).
   - **Notes**: Add a brief description.
2. Click **Commit** to save.

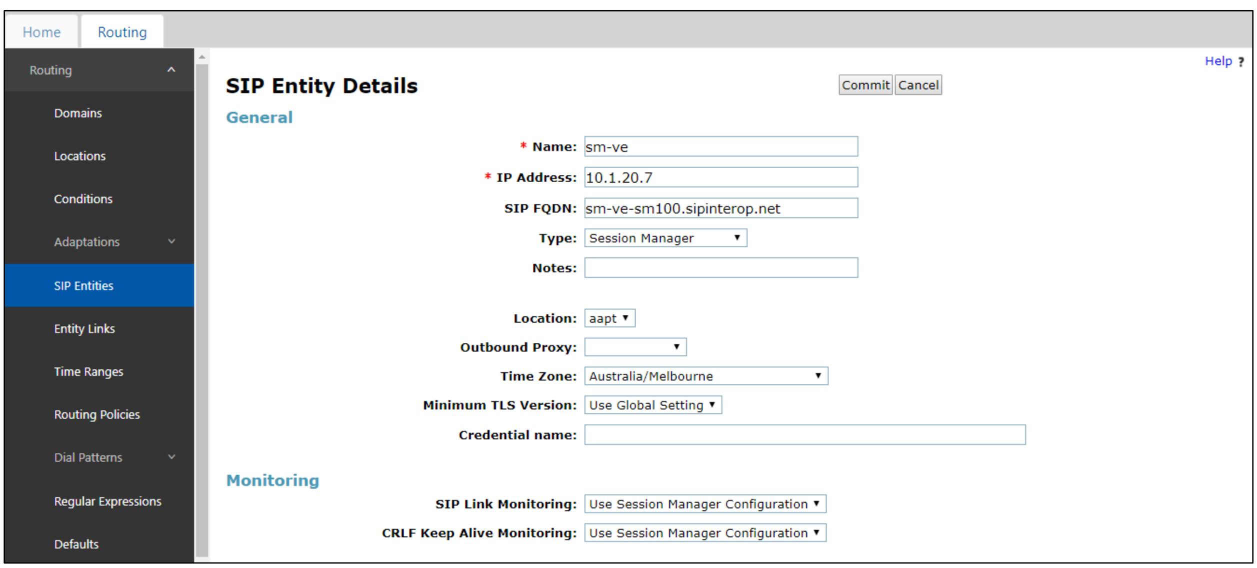
6.3 Configure SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it which includes Communication Manager and Avaya SBCE.

6.3.1 Configure Session Manager SIP Entity

Follow the steps shown below

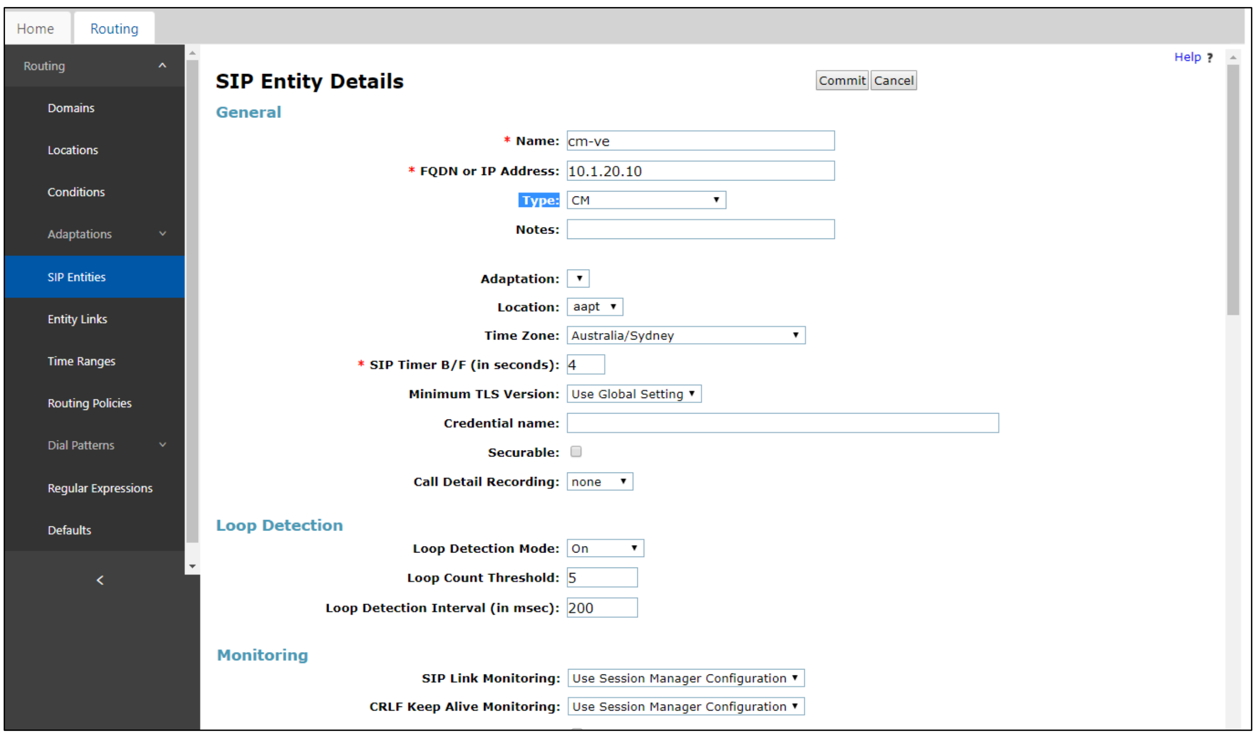
1. In the left pane under **Routing**, click on **SIP Entities**. In the **SIP Entities** page, click on **New** (not shown).
2. In the **General** section of the **SIP Entity Details** page, provision the following:
   - **Name –** Enter a descriptive name (e.g., **sm-ve**).
   - **IP Address –** Enter the IP address of Session Manager signaling interface, (*not* the management interface), provisioned during installation (e.g., **10.1.20.7**).
   - **SIP FQDN –** (Optional) Leave blank or enter the SIP FQDN of Session Manager signaling interface (e.g., **sm-ve-sm100.sipinterop.net**)
   - **Type –** Verify **Session Manager** is selected.
   - **Location** – Select location **aapt.**
   - **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.
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.



6.3.2 Configure Communication Manager SIP Entity – Outbound SIP Trunk

Follow the steps shown below:

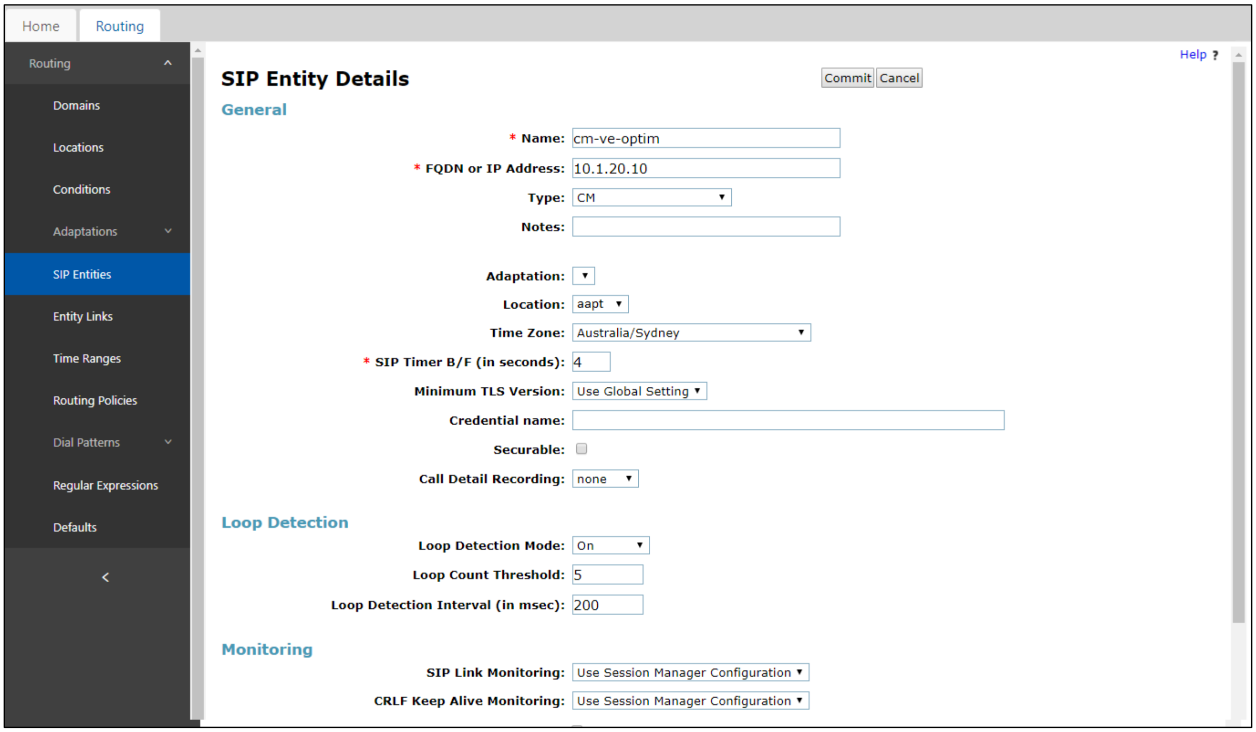
1. In the **SIP Entities** page, click on **New** (not shown).
2. In the **General** section of the **SIP Entity Details** page, provision the following:
   - **Name –** Enter a descriptive name (e.g. **cm-ve**).
   - **FQDN or IP Address –** Enter the IP address of Communication Manager Processor Ethernet (procr) (e.g. **10.1.20.10**).
   - **Type –** Select **CM**.
   - **Location** – Select a Location **aapt** administered in **Section 6.2**.
   - **Time Zone** – Select the time zone in which Communication Manager resides.
3. In the **Monitoring** section of the **SIP Entity Details** page select:
   - **a.** Select **Use Session Manager Configuration** for **SIP Link Monitoring** field
   - **b.** Use the default values for the remaining parameters.
4. Click on **Commit**.



6.3.3 Configure Communication Manager SIP Entity – CPE Access

Repeat the steps in **Section 6.3.2** with the following changes:

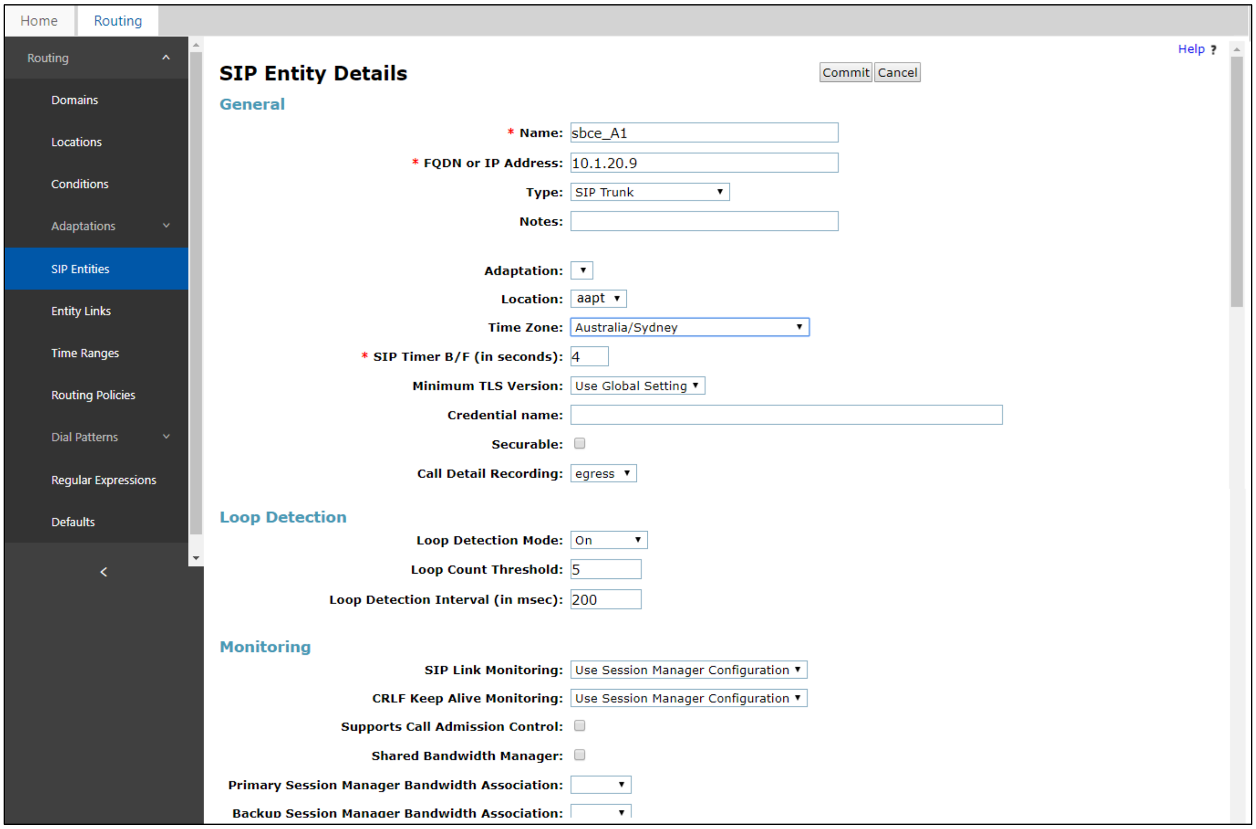
- **Name –** Enter a different CM descriptive name (e.g., **cm-ve-optim**).
- **FQDN or IP Address –** Enter the same IP address of Communication Manager Processor Ethernet (procr) (e.g. **10.1.20.10**).
- Other fields as same as in **Section 6.3.2**.



6.3.4 Configure Avaya SBCE SIP Entity

Repeat the steps in **Section 6.3.2** with the following changes:

- **Name –** Enter a descriptive name (e.g., **sbce_A1**).
- **FQDN or IP Address –** Enter the IP address of the A1 (private) interface of the Avaya SBCE (e.g., **10.1.20.9**).
- **Type –** Verify **SIP Trunk** is selected.
- **Location** – Select location **aapt** (**Section 6.2**).



6.4 Configure Entity Links

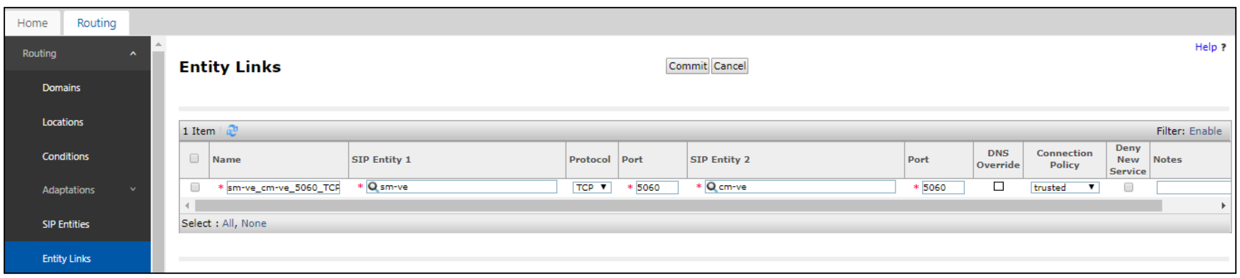
A SIP trunk between Session Manager and a telephony system is described by an Entity Link. During compliance testing, two Entity Links were created, one for Communication Manager and another one for Avaya SBCE. To add an Entity Link, navigate to **Routing → Entity Links** in the left navigation pane and click the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager defined in **Section 6.3.1**.
- **Protocol:** Select the transport protocol used for this link, **TCP** for the Entity Link to Communication Manager and **TCP** for the Entity Link to the Avaya SBCE.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For Communication Manager, this must match the **Far-end Listen Port** defined on the Communication Manager.
- **SIP Entity 2:** Select the name of the other systems. For Communication Manager, select the Communication Manager SIP Entity defined in **Section 6.3.2**. For Avaya SBCE, select Avaya SBCE SIP Entity defined in **Section 6.3.4**.
- **Port:** Port number on which the other system receives SIP requests from Session Manager. For Communication Manager, this must match the **Near-end Listen Port** defined on the Communication Manager.
- **Connection Policy:** Select **Trusted**.
- Click **Commit** to save.

6.4.1 Configure Entity Link to Communication Manager – Outbound SIP Trunk

Follow the steps shown below:

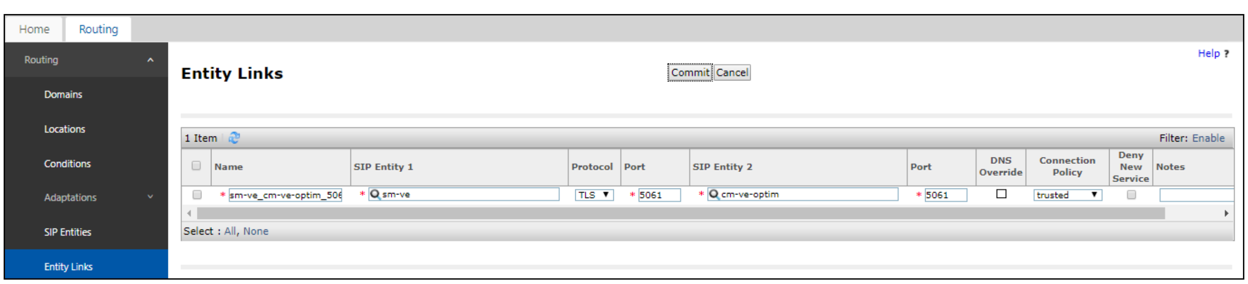
1. In the left pane under **Routing**, click on **Entity Links**, then click on **New** (not shown).
2. Continuing in the **Entity Links** page, provision the following:
   - **Name** – Enter a descriptive name (or have it created automatically) for this link to Communication Manager (e.g., **sm-ve_cm-ve_5060_TCP**).
   - **SIP Entity 1** – Select the SIP Entity administered in **Section 6.3.1** for Session Manager (e.g., **sm-ve**).
   - **SIP Entity 1 Port** – Enter **5060**.
   - **Protocol** – Select **TCP**.
   - **SIP Entity 2** –Select the SIP Entity administered in **Section 6.3.2** for the Communication Manager entity (e.g., **cm-ve**).
   - **SIP Entity 2 Port** - Enter **5060**.
   - **Connection Policy** – Select **Trusted**.
3. Click on **Commit**.



6.4.2 Configure Entity Link to Communication Manager – CPE Access

To configure this Entity Link, repeat the steps in **Section 6.4.1**, with the following changes:

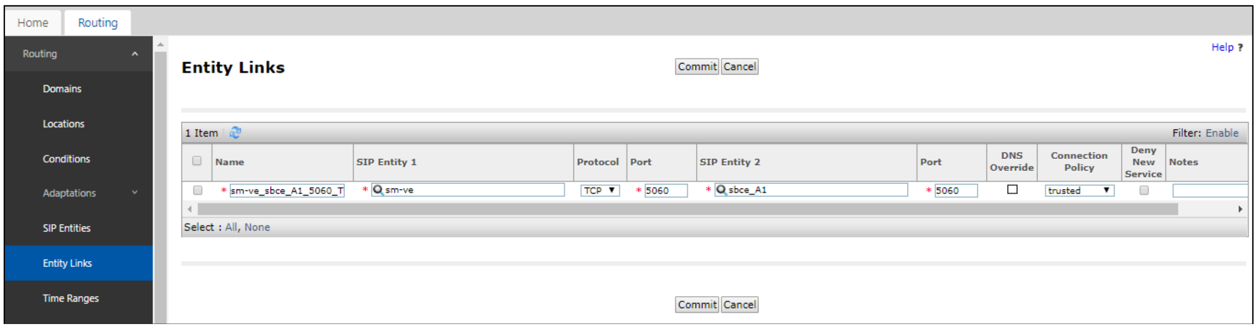
- **Name** – Enter a descriptive name (or have it created automatically) for this link to the Avaya SBCE (e.g., **sm-ve_cm-ve-optim_5061_TLS**).
- **SIP Entity 1 Port** – Enter **5061**.
- **Protocol** – Select **TLS**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.3.3** for the Communication Manager entity (e.g., **cm-ve-optim**).
- **SIP Entity 2 Port** - Enter **5061**.
- Click on **Commit**.



6.4.3 Configure Entity Link for Avaya SBCE

To configure this Entity Link, repeat the steps in **Section 6.4.1**, with the following changes:

- **Name** – Enter a descriptive name (or have it created automatically) for this link to the Avaya SBCE (e.g., **sm-ve_sbce_A1_5060_TCP**).
- **SIP Entity 1 Port** – Enter **5060**.
- **Protocol –** Select **TCP**.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.3.4** for the Avaya SBCE entity (e.g., **sbce_A1**).
- **SIP Entity 2 Port** - Enter **5060**.



6.5 Configure Routing Policies

Routing Policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.3**. Two routing policies were added, one for Communication Manager and another for Avaya SBCE. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click the **New** button in the right pane (not shown). The following screen is displayed.

In the **General** section, enter the following values:

- **Name:** Enter a descriptive name.
- **Notes:** Add a brief description (optional).

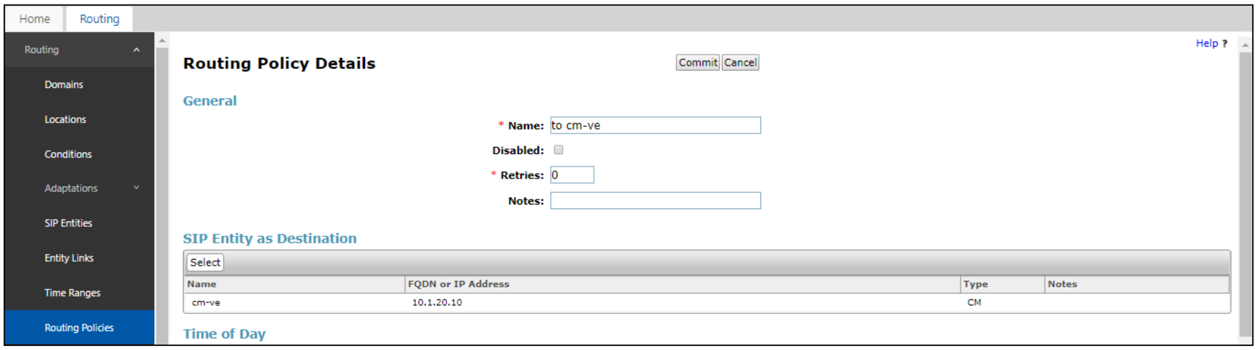
In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Select appropriate SIP entity to which this routing policy applies and click **Select**. The selected SIP Entity is displayed in the **Routing Policy Details** page as shown below. Use default values for remaining fields. Click **Commit** to save.

6.5.1 Configure Routing Policy for Communication Manager

This Routing Policy is used for inbound calls from AAPT.

1. In the left pane under **Routing**, click on **Routing Policies**. In the **Routing Policies** page click on **New** (not shown).
2. In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing AAPT calls to Communication Manager (e.g., **to cm-ve**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.

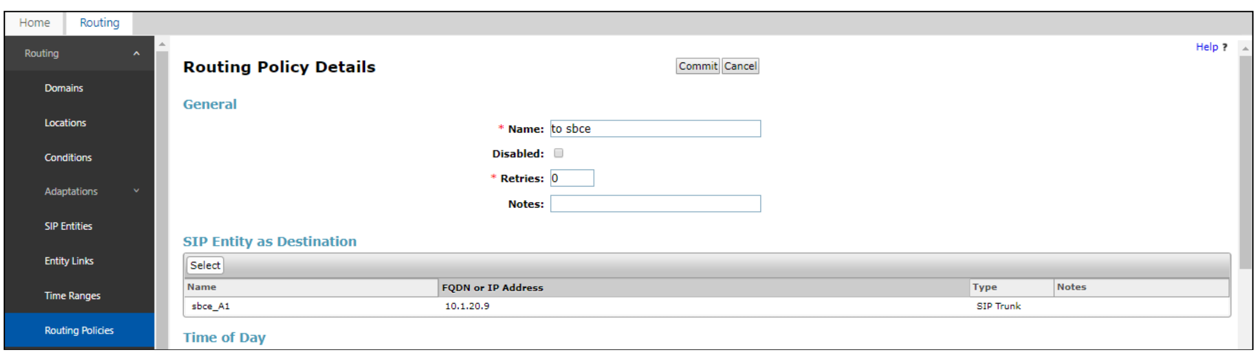
3. In the **SIP Entity as Destination** section of the **Routing Policy Details** page, click on **Select** and the SIP Entity list page will open.
4. In the **SIP Entity List** page, select the SIP Entity administered in **Section 6.3.2** for the Communication Manager SIP Entity (**cm-ve**), and click on **Select**.
5. Note that once the **Dial Patterns** are defined they will appear in the **Dial Pattern** section of this form.
6. No **Regular Expressions** were used in the reference configuration.
7. Click on **Commit**.



6.5.2 Configure Routing Policy for Avaya SBCE

This Routing Policy is used for outbound calls to the service provider. Repeat the steps in **Section 6.5.1**, with the following changes:

- **Name** – Enter a descriptive name for this link to the Avaya SBCE (e.g., **to sbce**).
- **SIP Entity List** –Select the SIP Entity administered in **Section 6.3.4** for the Avaya SBCE entity (e.g., **sbce_A1**).



6.6 Configure Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance testing, dial patterns were needed to route calls from Communication Manager to AAPT and vice versa. Dial Patterns define which routing policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left navigation pane and click **New** button in the right pane (not shown).

In the **General** section, enter the following values:

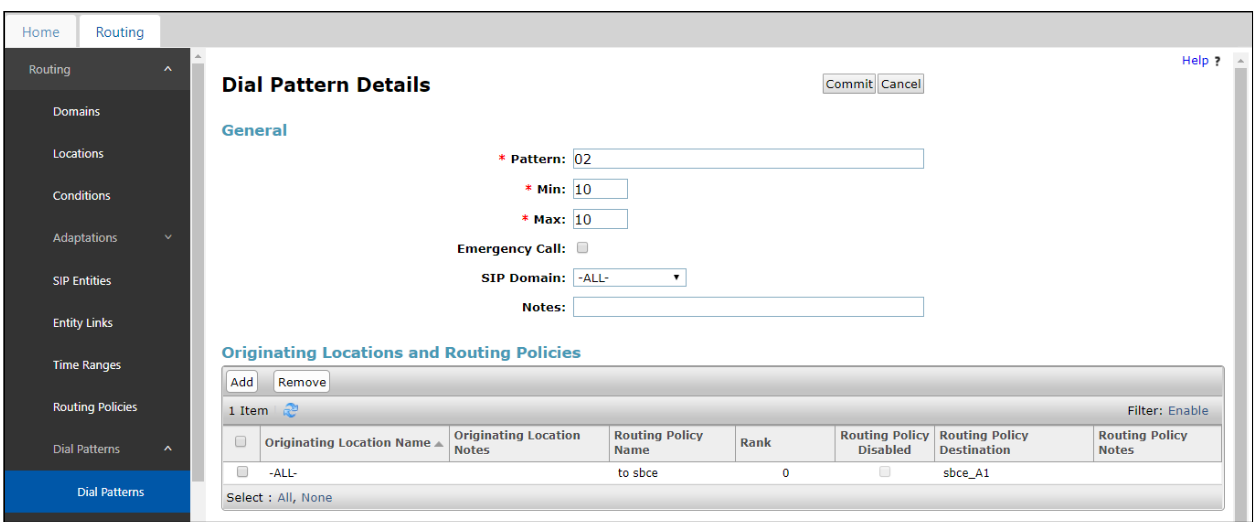
- **Pattern:** Enter a dial string that will be matched against the "Request-URI" of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria.
- **Notes:** Add a brief description (optional).

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria. Click **Select**.

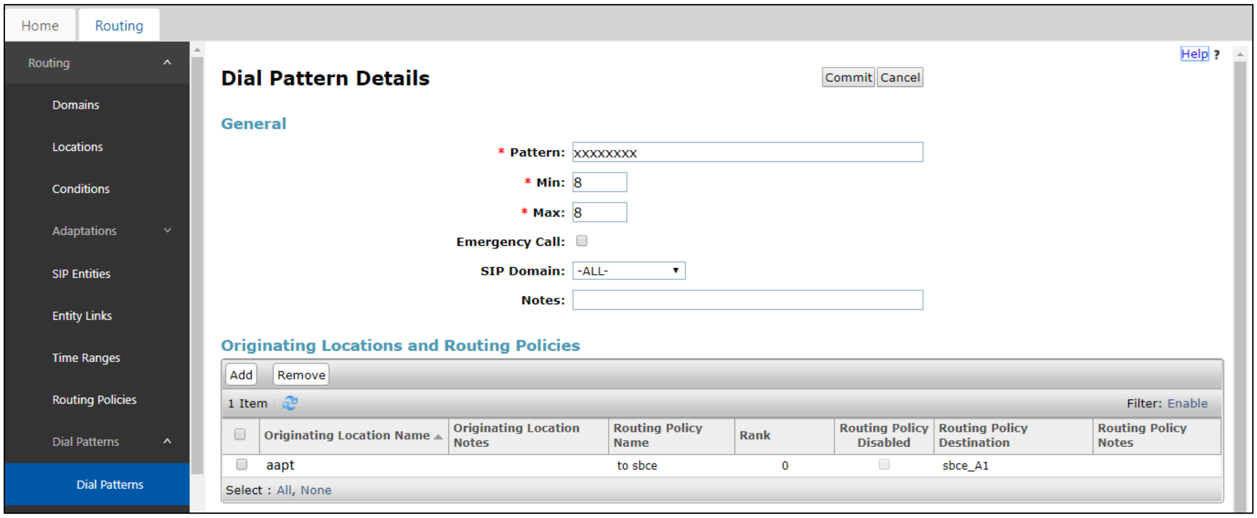
Default values can be used for the remaining fields. Click **Commit** to save.

Four examples of the dial patterns used for the compliance testing were shown below, two for outbound calls from the enterprise to the PSTN, one for inbound calls from the PSTN to the enterprise and another one for outbound calls to emergency number.

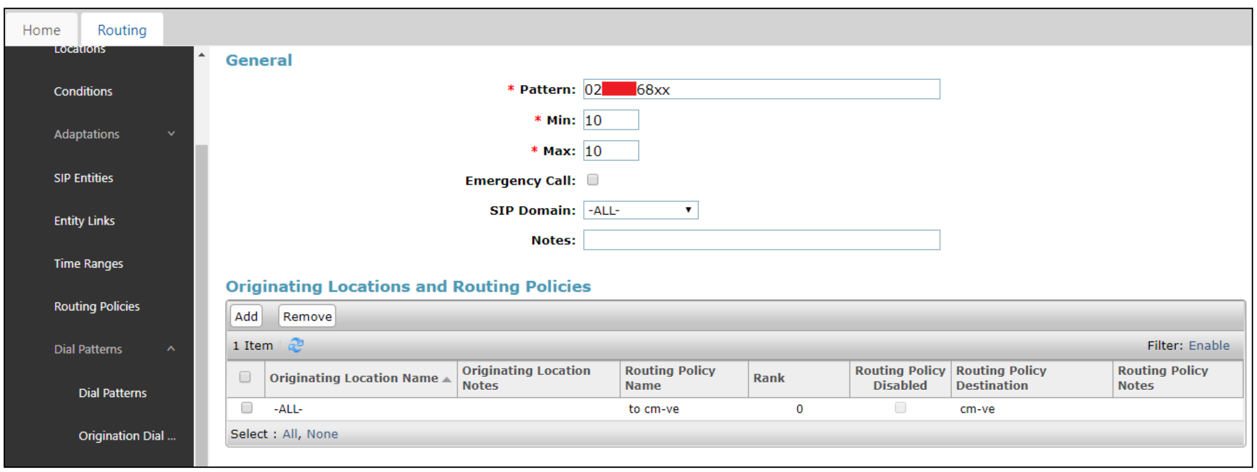
The first example shows that 10-digit dialed numbers starting with 02 that has a destination domain of "All" uses route policy **to sbce** as defined in **Section 6.5.2**



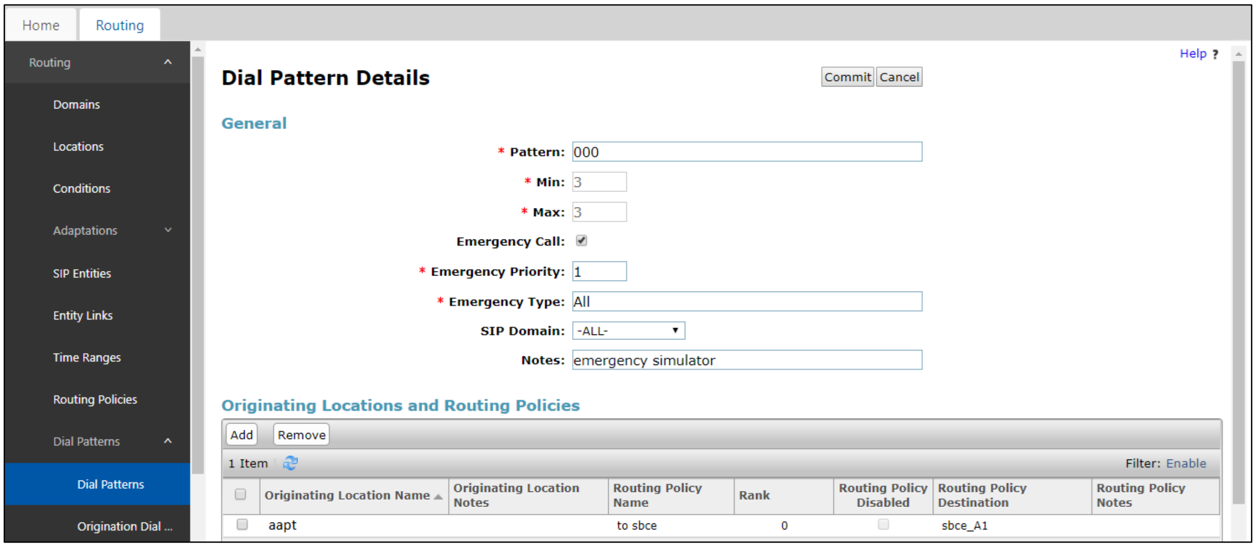
The second example shows that outbound any 8-digit numbers uses route policy **to sbce** as defined in **Section 6.5.2** for PSTN calls**.**



The third example shows that 10-digit pattern that starts with 02xxxx68 is used for inbound calls from AAPT to DID numbers on Avaya Aura® Communication Manager.



The fourth example shows that 000 dialed number is used for emergency service in Australia.

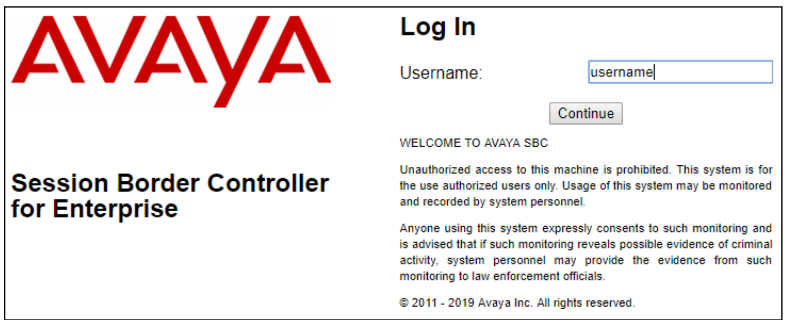


7. Configure Avaya Session Border Controller for Enterprise

Note: The installation and initial provisioning of the Avaya SBCE is beyond the scope of this document.

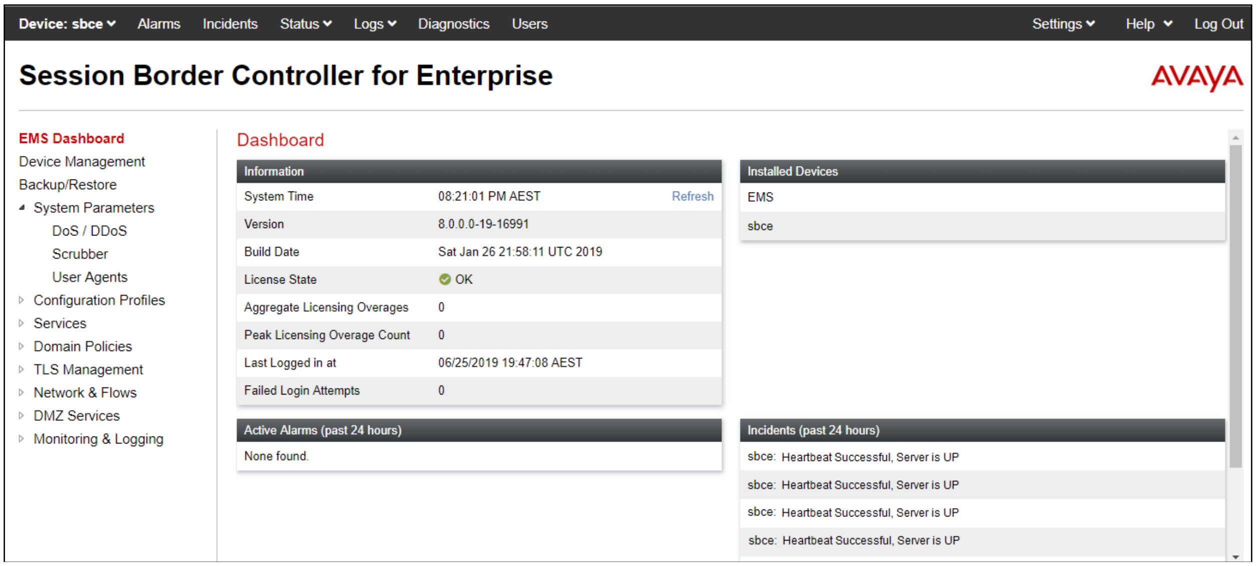
As described in **Section 3**, the reference configuration places the private interface (A1) of Avaya SBCE in the Common site, (10.1.20.9), with access to the **AAPT** site. The connection to AAPT uses the Avaya SBCE public interface B1 (IP address 10.239.192.234). The following provisioning is performed via the Avaya SBCE GUI interface, using the "M1" management LAN connection on the chassis.

1. Access the web interface by typing "**https://x.x.x.x**" (where x.x.x.x is the management IP address of the Avaya SBCE).
2. Enter the **Username** and click on **Continue**.



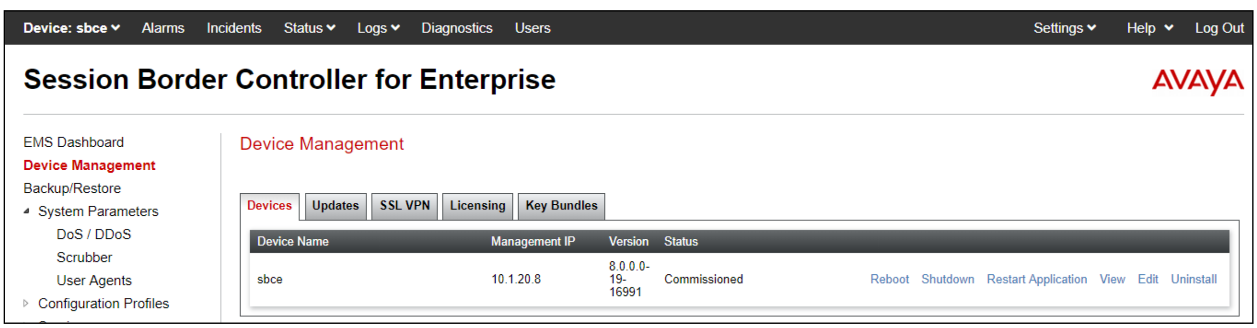
3. Enter the password and click on **Log In**.

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

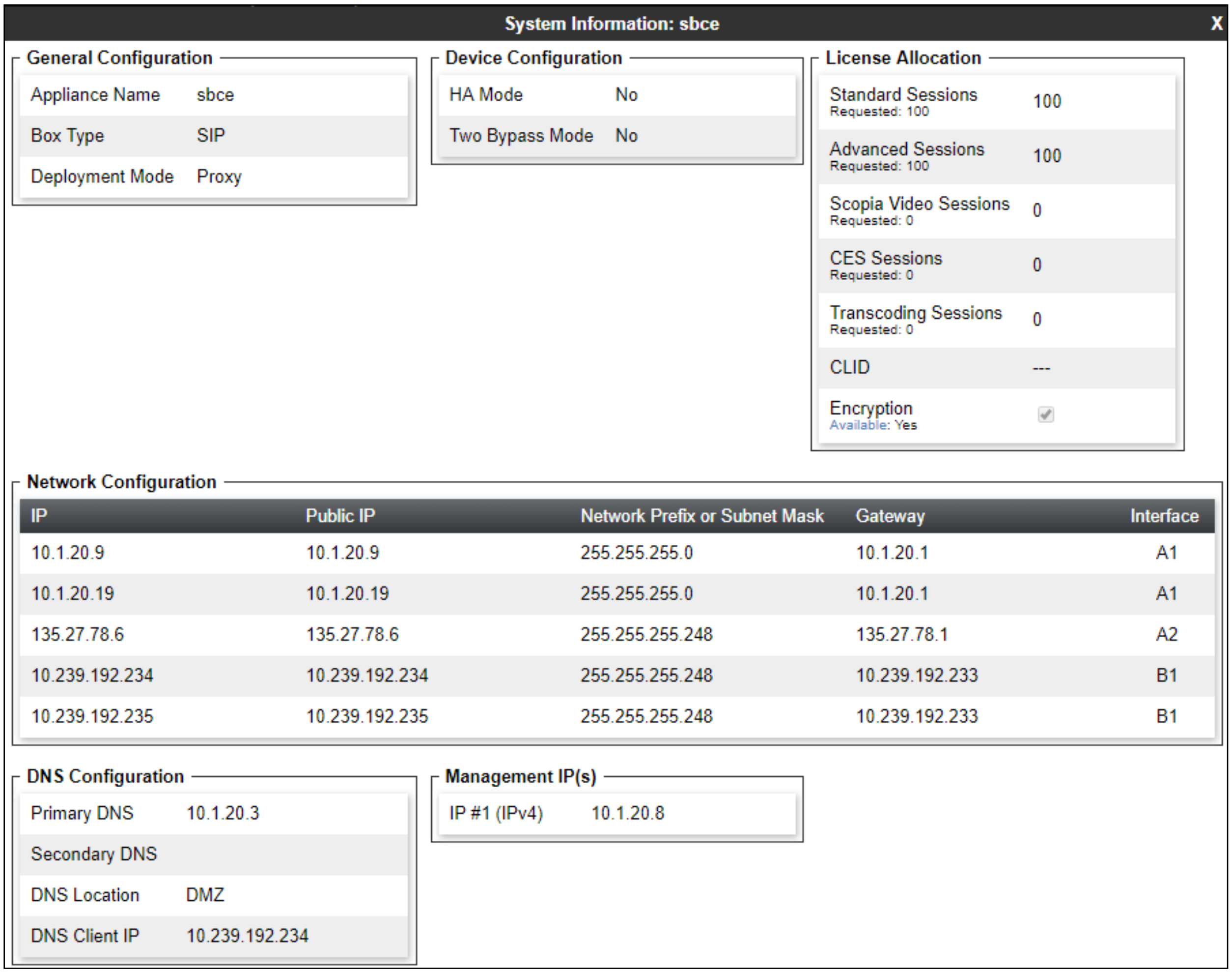


7.1 Device Management – Status

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



2. Click on **View** (shown above) to display the **System Information** screen. Note that DNS servers are AAPT DNS servers and DNS client must be B1 IP address that is used for SIP trunk with AAPT.

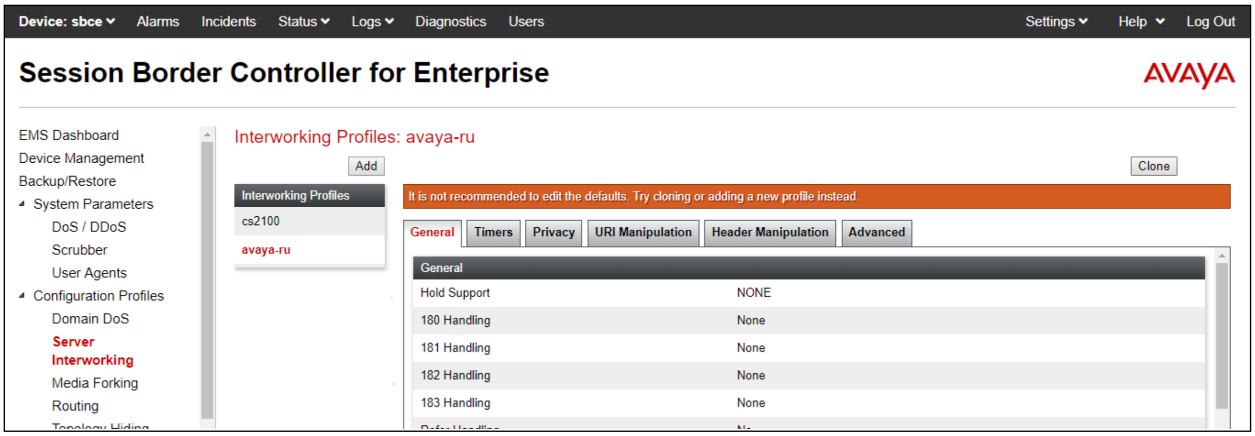


7.2 Server Interworking Profiles

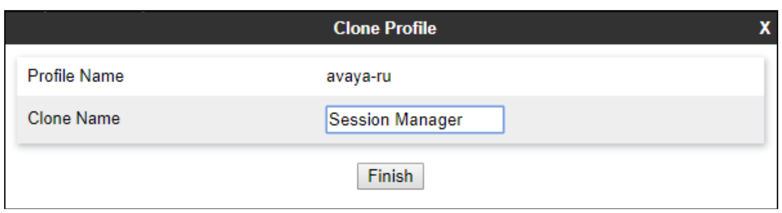
7.2.1 Server Interworking – Session Manager

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

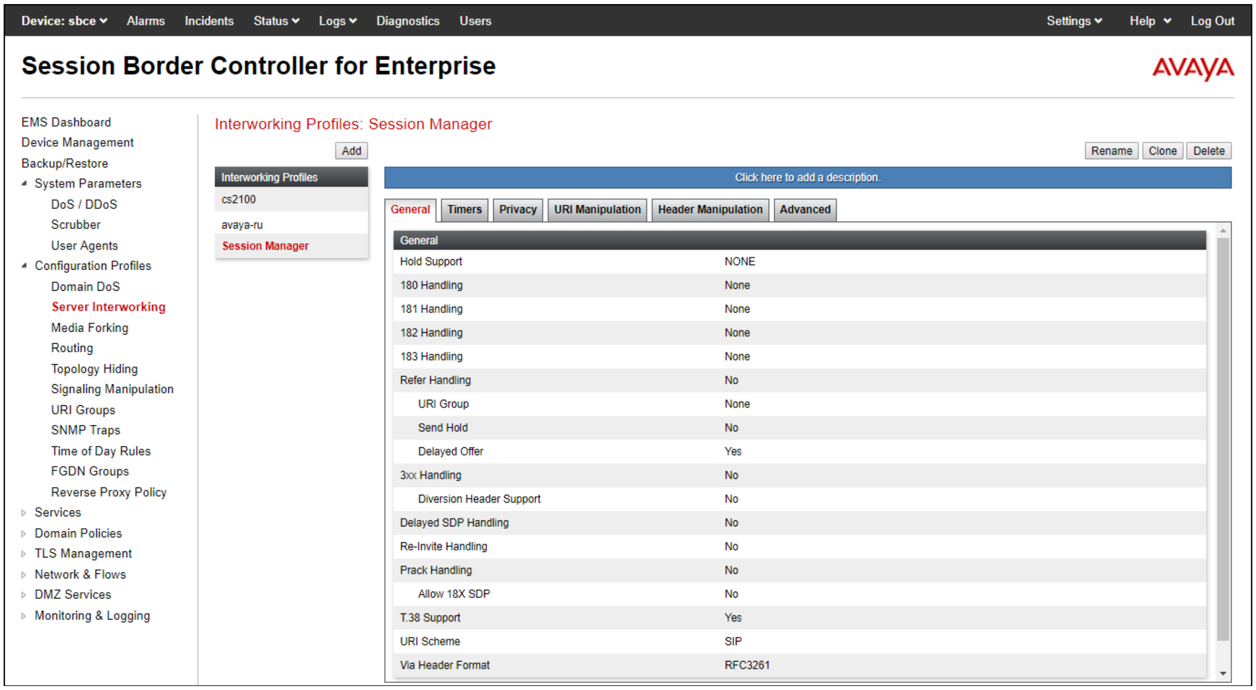
1. Select **Configuration Profiles → Server Interworking** from the left-hand menu.
2. Select the pre-defined **avaya-ru** profile and click the **Clone** button.



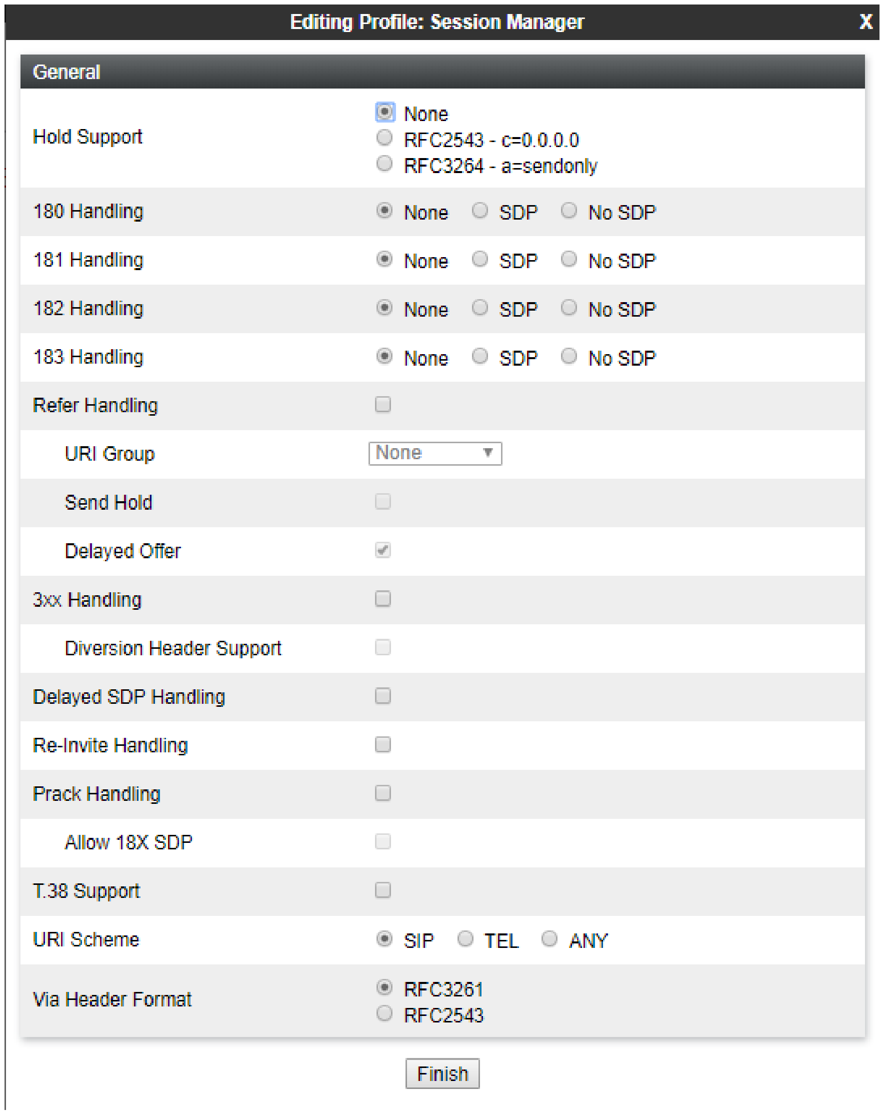
3. Enter profile name: (e.g., **Session Manager**), and click **Finish**.



4. The new Session Manager profile will be listed. Select it, scroll to the bottom of the Profile screen, and click on **Edit**.

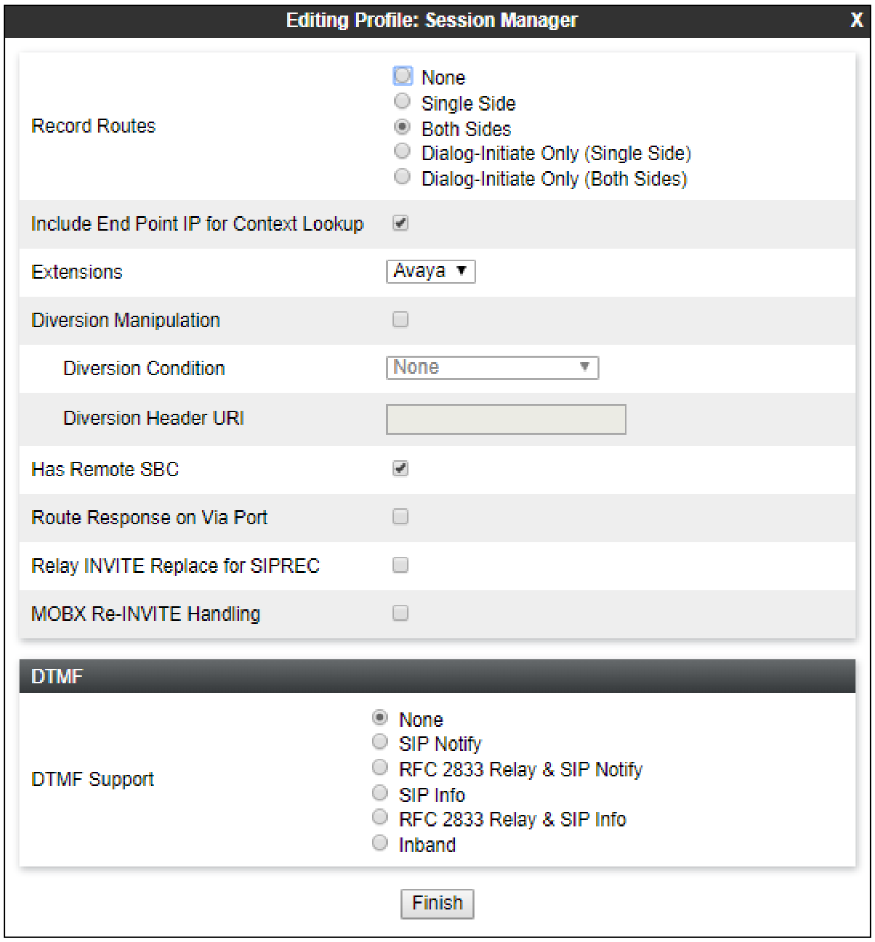


5. The **General** screen will open.
   - Uncheck **T.38 Support**.
   - All other options can be left with default values, and click **Finish**.



6. On the **Privacy, URI Manipulation, Header Manipulation** windows as, select **Finish** to accept default values.

7. On the **Advanced** window, configure;
   - **Record Routes**: choose **Both Sides**.
   - **Include End Point IP for Context Lookup**: choose **Yes**.
   - **Has Remote SBC**: choose **Yes**.

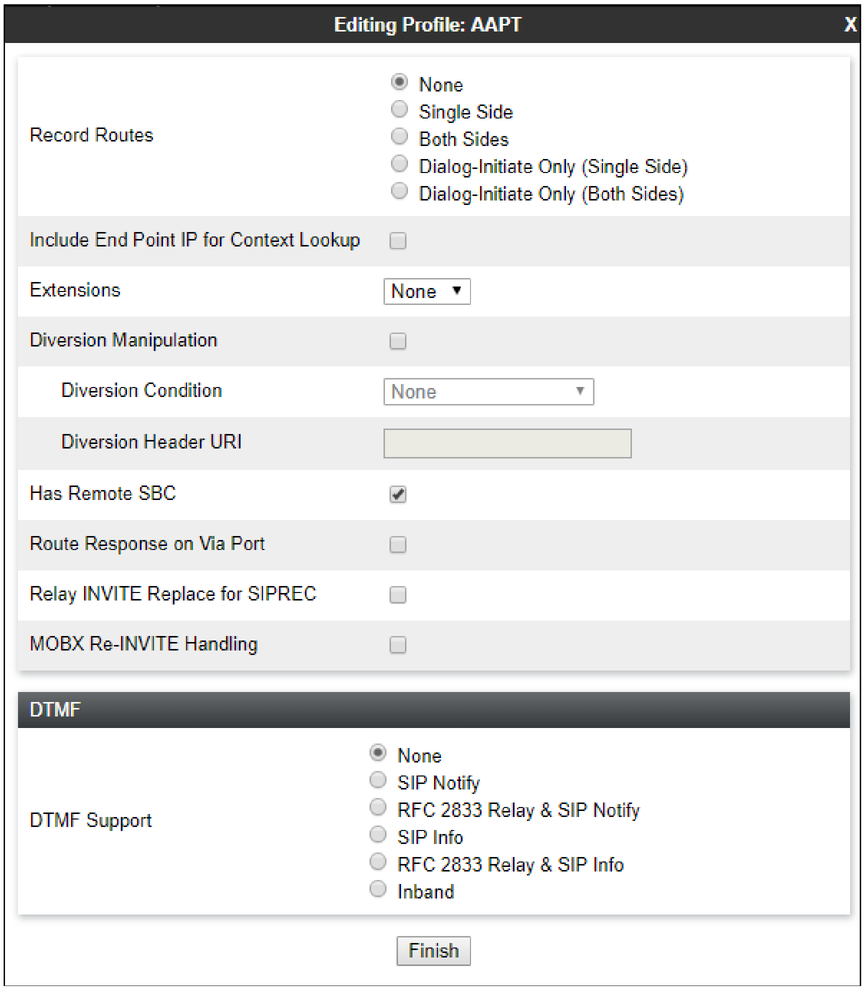


7.2.2 Server Interworking – AAPT

Repeat the steps shown in **Section 7.2.1** to add an Interworking Profile for the connection to AAPT via the public network, with the following changes:

1. Click **Add** to add a new profile, enter **AAPT** then click **Next** (not shown)
2. The **General** screen will open:
   - Uncheck **T.38 Support**.
   - All other options can be left as default.
   - Click **Next**.
   - The **Privacy/DTMF, SIP Timers/Transport Timers** screens will open (not shown), accept default values for all the screens by clicking **Next**.

The **Advanced** window is configured as below, click **Finish** to save the profile:



7.3 Signaling Manipulation Script

Signaling Manipulation Script **Add-PAI-header** is required to:

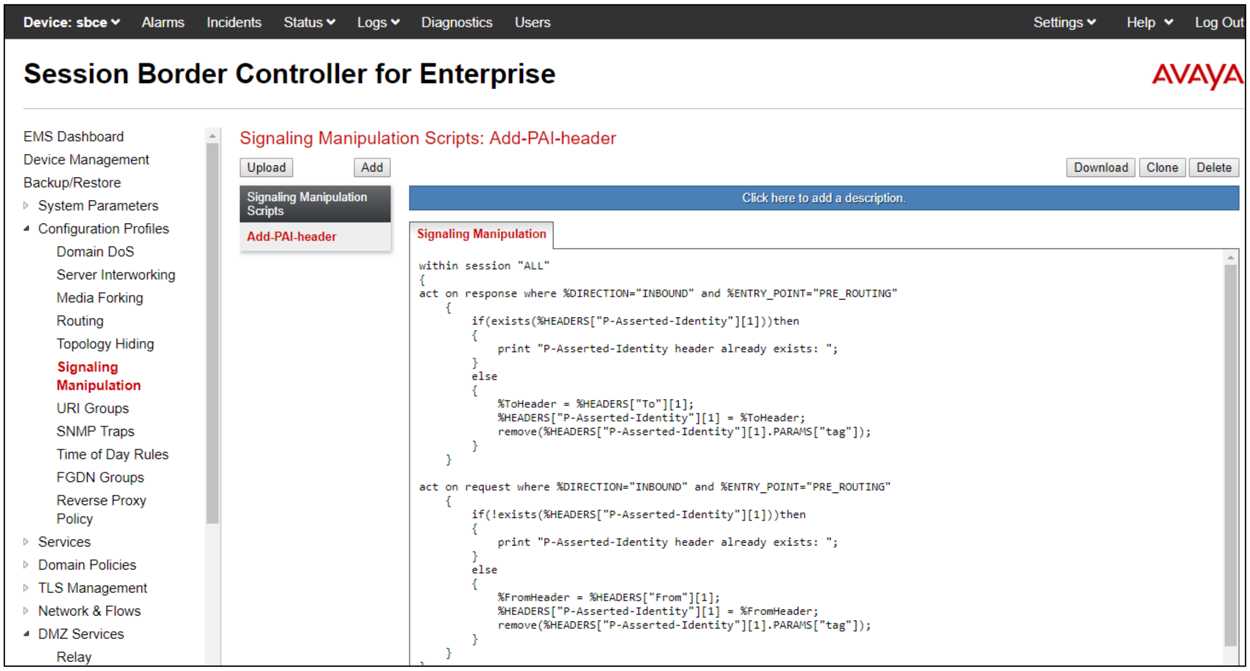
- If not existed, add PAI header to SIP requests and responses from/to AAPT SIP Voice
- The PAI header content is copied from From/To headers

Follow below steps to create Signaling Manipulation Script **Add-PAI-header**:

1. Select **Configuration Profiles → Signaling Manipulation** from the left-hand menu
2. Select **Add** and the **Signaling Manipulation Editor** window will open
3. Enter the script name into **Title** (e.g., **Add-PAI-header**)
4. Copy and paste the content in the below text box into the editor, and click **Save**

```
within session "ALL"
{
act on response where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
    {
        if(exists(%HEADERS["P-Asserted-Identity"][1]))then
        {
            print "P-Asserted-Identity header already exists: ";
        }
        else
        {
            %ToHeader = %HEADERS["To"][1];
            %HEADERS["P-Asserted-Identity"][1] = %ToHeader;
            remove(%HEADERS["P-Asserted-Identity"][1].PARAMS["tag"]);
        }
    }

act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
    {
        if(!exists(%HEADERS["P-Asserted-Identity"][1]))then
        {
            print "P-Asserted-Identity header already exists: ";
        }
        else
        {
            %FromHeader = %HEADERS["From"][1];
            %HEADERS["P-Asserted-Identity"][1] = %FromHeader;
            remove(%HEADERS["P-Asserted-Identity"][1].PARAMS["tag"]);
        }
    }
}
```

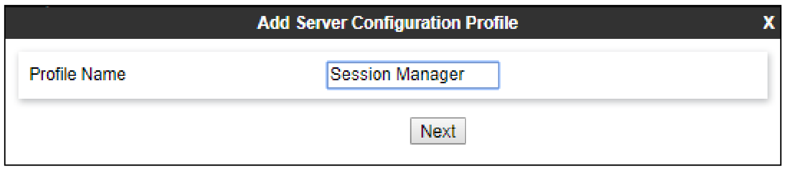

Device: sbce
Alarms
Incidents
Status
Logs
Diagnostics
Users
Settings
Help
Log Out
Session Border Controller for Enterprise
AVAYA
EMS Dashboard
Device Management
Backup/Restore
System Parameters
Configuration Profiles
Domain DoS
Server Interworking
Media Forking
Routing
Topology Hiding
Signaling Manipulation
URI Groups
SNMP Traps
Time of Day Rules
FGDN Groups
Reverse Proxy Policy
Services
Domain Policies
TLS Management
Network & Flows
DMZ Services
Relay
Signaling Manipulation Scripts: Add-PAI-header
Upload
Add
Download
Clone
Delete
Signaling Manipulation Scripts
Add-PAI-header
Click here to add a description.
Signaling Manipulation
within session "ALL"
{
act on response where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
{
if(exists(%HEADERS["P-Asserted-Identity"][1]))then
{
print "P-Asserted-Identity header already exists: ";
}
else
{
%ToHeader = %HEADERS["To"][1];
%HEADERS["P-Asserted-Identity"][1] = %ToHeader;
remove(%HEADERS["P-Asserted-Identity"][1].PARAMS["tag"]);
}
}
act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
{
if(!exists(%HEADERS["P-Asserted-Identity"][1]))then
{
print "P-Asserted-Identity header already exists: ";
}
else
{
%FromHeader = %HEADERS["From"][1];
%HEADERS["P-Asserted-Identity"][1] = %FromHeader;
remove(%HEADERS["P-Asserted-Identity"][1].PARAMS["tag"]);
}
}

7.4 SIP Server Profiles

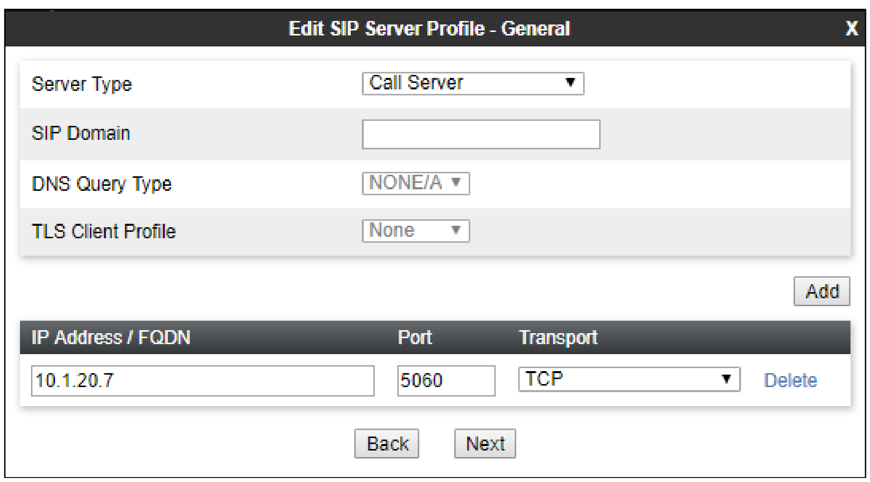
7.4.1 SIP Server – Session Manager

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

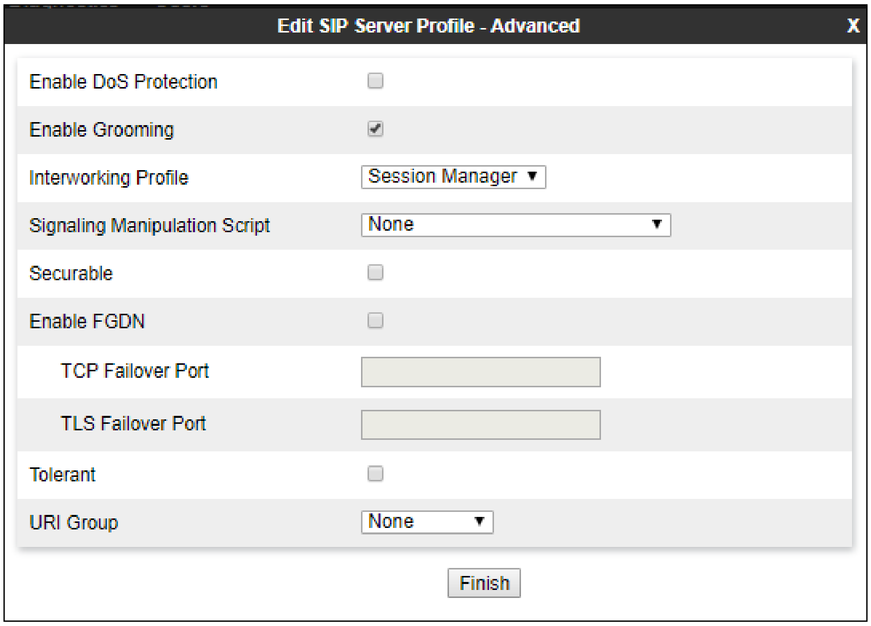
1. Select **Services → SIP Server** from the left-hand menu.
2. Select **Add** and the **Profile Name** window will open. Enter a Profile Name (e.g., **Session Manager**) and click **Next**.



3. The **Add SIP Server Profile** window will open.
   - Select **Server Type**: **Call Server**.
   - **IP Address / FQDN**: **10.1.20.7** (Session Manager signaling IP Address)
   - **Transport**: Select **TCP**.
   - **Port**: **5060**.
   - Select **Next**.



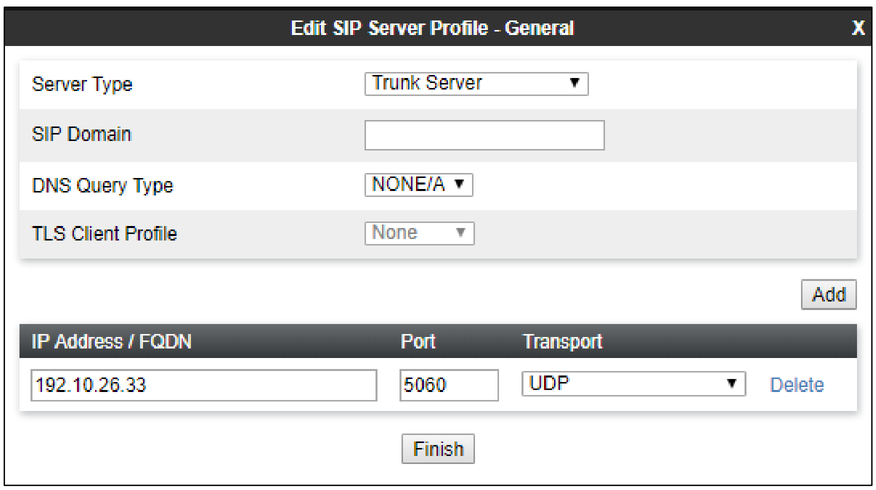
4. The **Authentication** and **Heartbeat** windows will open (not shown).
   - Select **Next** to accept default values.
5. The **Advanced** window will open.
   - For **Interworking Profile**, select the profile created for Session Manager in **Section 7.2.1**.
   - Check **Enable Grooming**.
   - Select **Finish**.



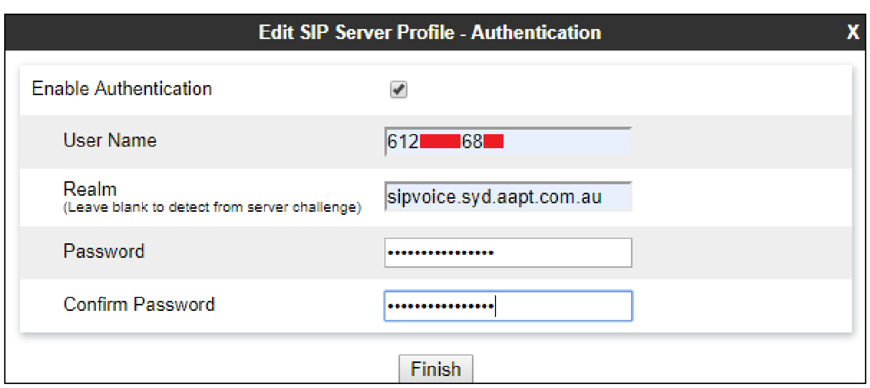
7.4.2 SIP Server – AAPT

Repeat the steps in **Section 7.4.1**, with the following changes, to create a SIP Server Profile for the Avaya SBCE connection to AAPT Trunk Group. AAPT supports both trusted network / static IP address configuration and untrusted network / authentication required configuration. Step 3 and 5 in below example are for untrusted network configuration only.

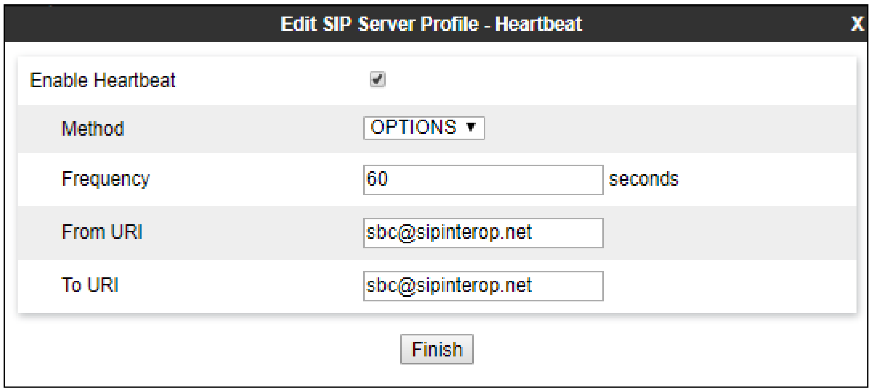
1. Select **Add Profile** and enter a Profile Name (e.g., **AAPT**) and select **Next**.
2. On the **General** window (not shown), enter the following.
    - Select **Server Type: Trunk Server**.
    - **IP Address / FQDN: 192.10.26.33** (outbound proxy of AAPT).
    - **Transport**: Select **UDP**.
    - **Port: 5060**.
    - Select **Next**.



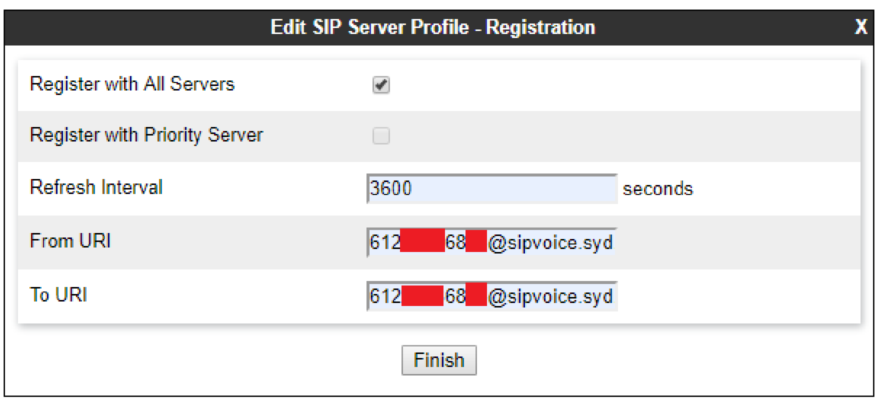
3. Under Authentication window:
    - Select **Enable Authentication**, if requested by AAPT. Otherwise, skip all the settings in this window.
    - **User Name**: enter Authentication name for outbound proxy.
    - **Realm**: enter SIP realm provided by AAPT.
    - **Password** and **Confirm Password**: enter Password provided by AAPT.



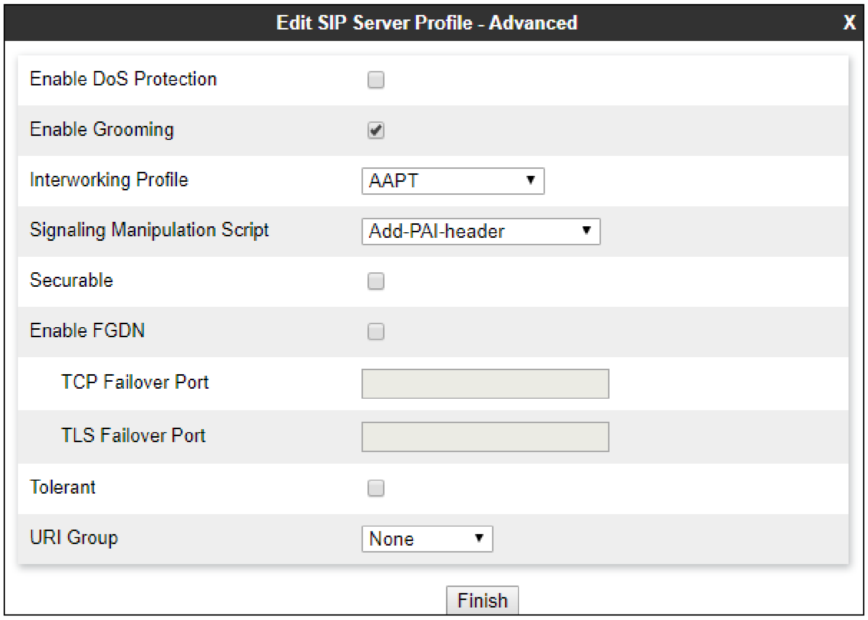
4. Under Heartbeat window:
   - Select **Enable Heartbeat**.
   - **Method**: choose **OPTIONS**.
   - **Frequency**: enter **60**.
   - **From URI** and **To URI**: enter **sbc@sipinterop.net**.



5. Under Registration window:
   - Select **Register with All Servers**, if requested by AAPT. Otherwise, skip all the settings in this window.
   - **Refresh Interval**: enter **3600**.
   - **From URI** and **To URI**: enter SIP URI provided by AAPT.



6. Under **Advanced** window:
   - Check **Enable Grooming**.
   - Select **AAPT** for Interworking Profile
   - Select **Add-PAI-header** for the Signaling Manipulation Script (see **Section 7.3**)

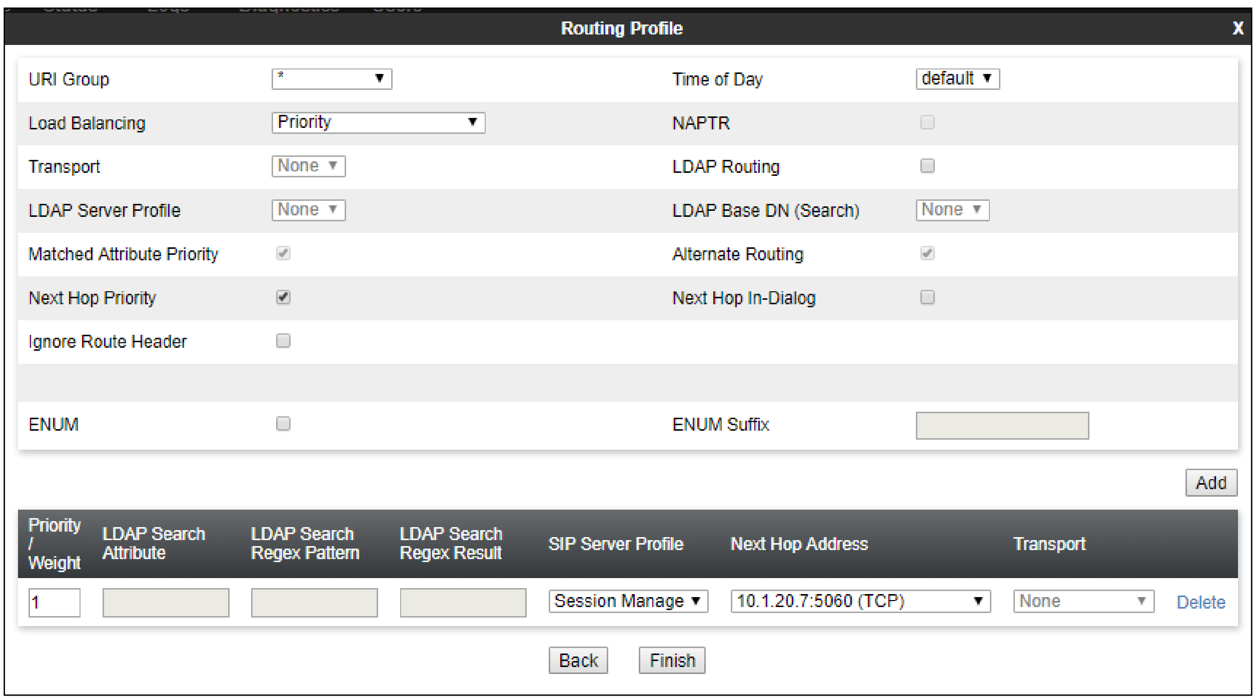


7.5 Routing Profiles

7.5.1 Routing – To Session Manager

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

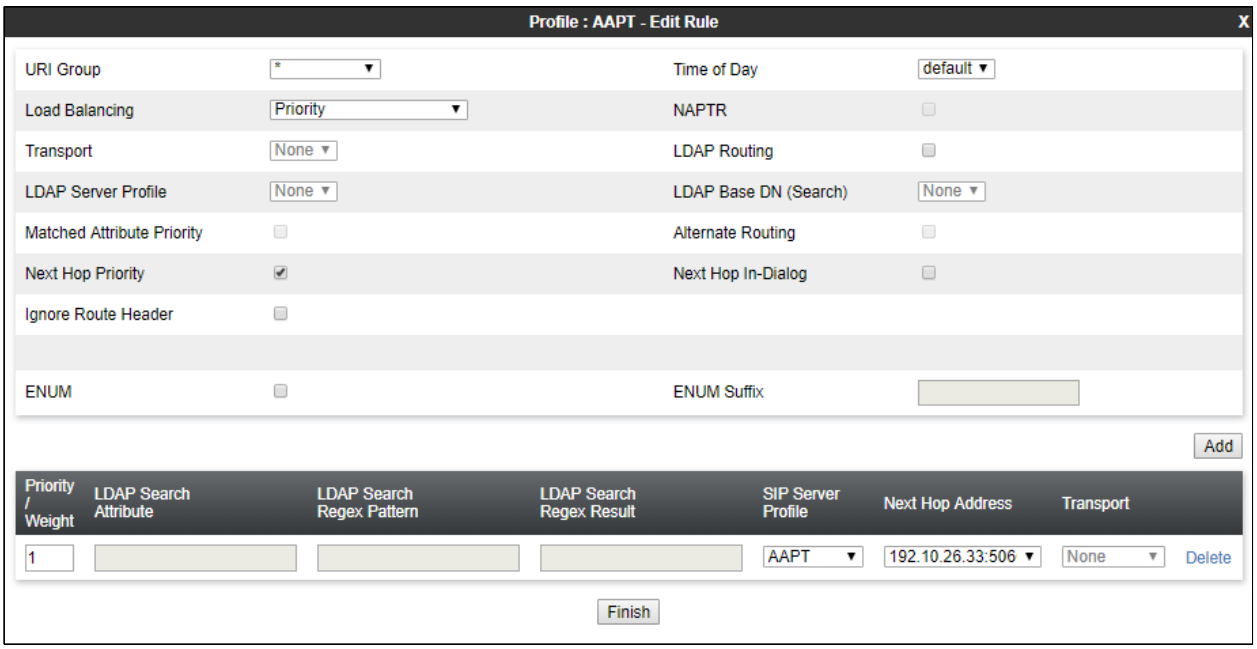
1. Select **Configuration Profiles → Routing** from the left-hand menu, and select **Add** (not shown).
2. Enter a **Profile Name**: (e.g., **Session Manager)** and click **Next**.
3. The **Routing Profile** window will open. Using the default values shown, click on **Add**.
4. The **Next-Hop Address** window will open. Populate the following fields:
   - **Priority/Weight** = **1**.
   - **SIP Server Profile** = **Session Manager**.
   - **Next Hop Address:** Verify that the **10.1.20.7:5060 (TCP)** entry from the drop down menu is selected (Session Manager IP address). Also note that the **Transport** field is grayed out.
   - Click on **Finish**.



7.5.2 Routing – To AAPT

Repeat the steps in **Section 7.5.1**, with the following changes, to add a Routing Profile for the Avaya SBCE connection to AAPT.

1. On the **Configuration Profiles → Routing** window (not shown), enter a **Profile Name**: (e.g., **AAPT)**.
2. **Load Balancing**: select **Priority**.
3. On the **Next-Hop Address** window (not shown), populate the following fields:
   - **SIP Server Profile = AAPT**.
   - **Next Hop Address:** Verify that the **192.10.26.33:5060** entry from the drop down menu is selected. Also note that the **Transport** field is grayed out.
   - Use default values for the rest of the parameters.
4. Click **Finish**.

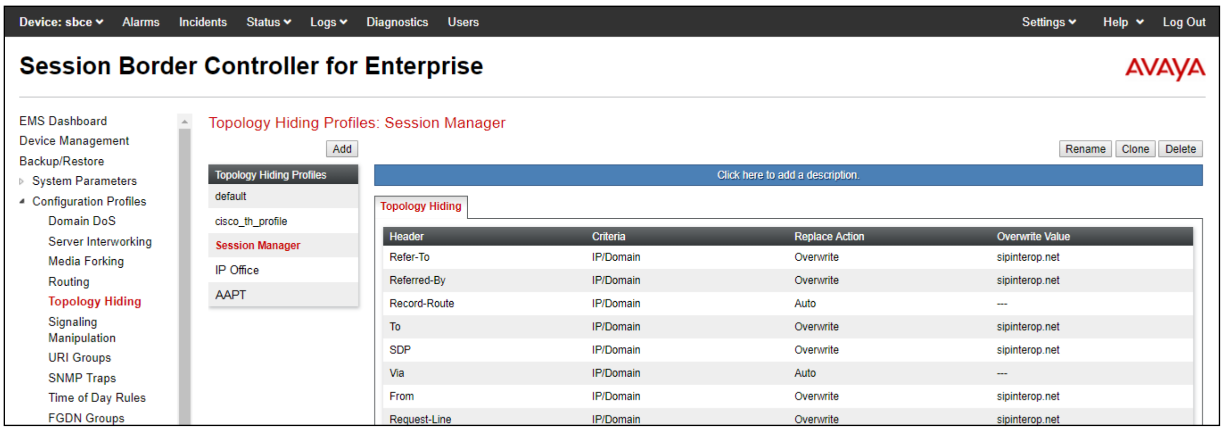


7.6 Topology Hiding

7.6.1 Topology Hiding – Session Manager

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

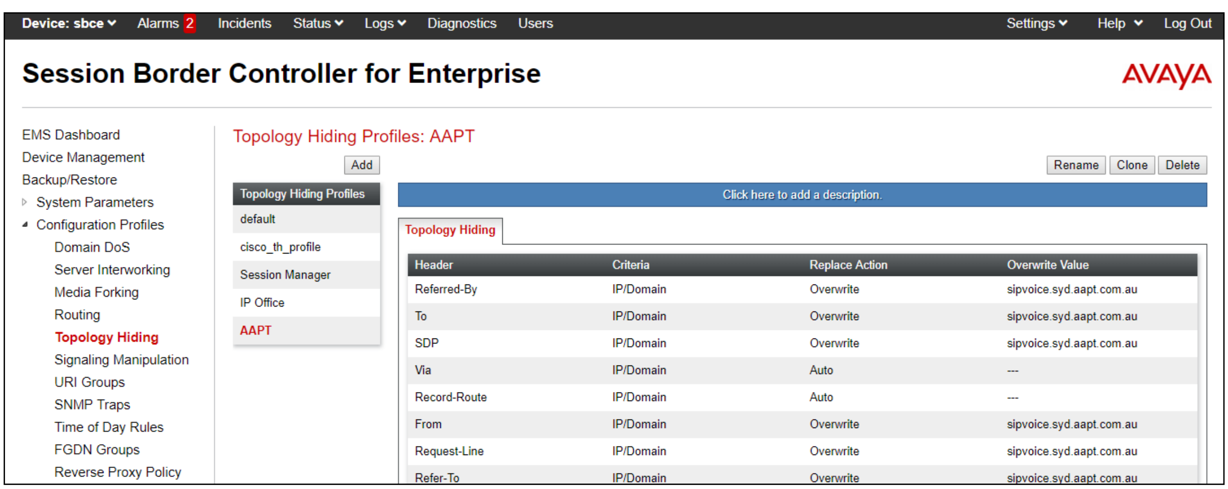
1. Select **Configuration Profiles → Topology Hiding** from the left-hand side menu.
2. Select the **Add** button, enter **Profile Name**: (e.g., **Session Manager**), and click **Next**.
3. The **Topology Hiding Profile** window will open. Click on the **Add Header** button repeatedly until all headers are added.
4. Populate the fields as shown below, and click **Finish**.



7.6.2 Topology Hiding – AAPT

Repeat the steps in **Section 7.6.1,** with the following changes, to create a Topology Hiding Profile for the Avaya SBCE connection to AAPT.

1. Enter a **Profile Name**: (e.g., **AAPT**).
2. Click on the **Add Header** button repeatedly until all headers are added.
3. Populate the fields as shown below, and click **Finish**.

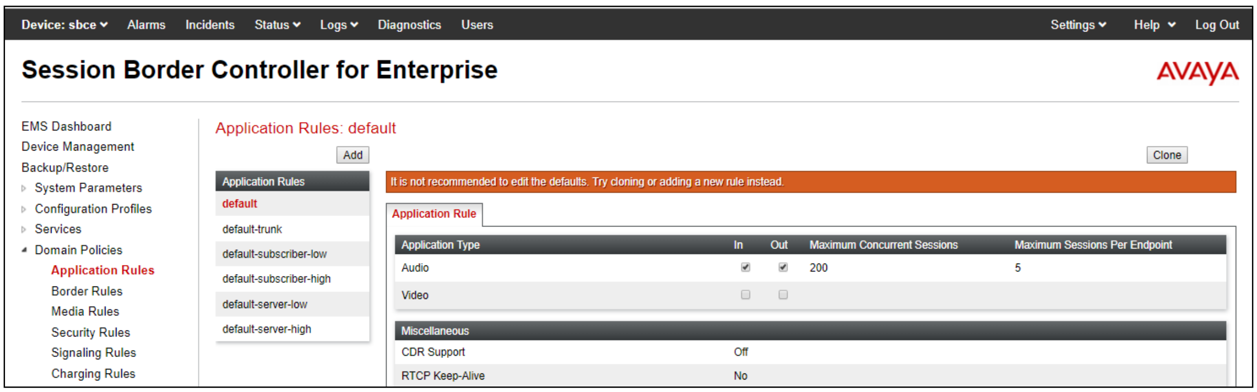


7.7 Domain Policies

The Domain Policies feature allows users to configure, apply and manage various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating in the enterprise. Avaya SBCE has pre-defined / default Rules and Policies under Domain Policies. Although the default Rules and Policies are editable, it is highly recommended to clone the Rules and/or Policies before modification as needed. The compliance test was commenced using the default rules and policies without any modification.

7.7.1 Application Rules

Ensure that the Application Rule used in the End Point Policy Group reflects the licensed sessions that the customer has purchased. In the lab setup, Avaya SBCE was licensed for 200 Voice sessions, and the default rule was amended accordingly. Other Application Rules could be utilized on an as needed basis.

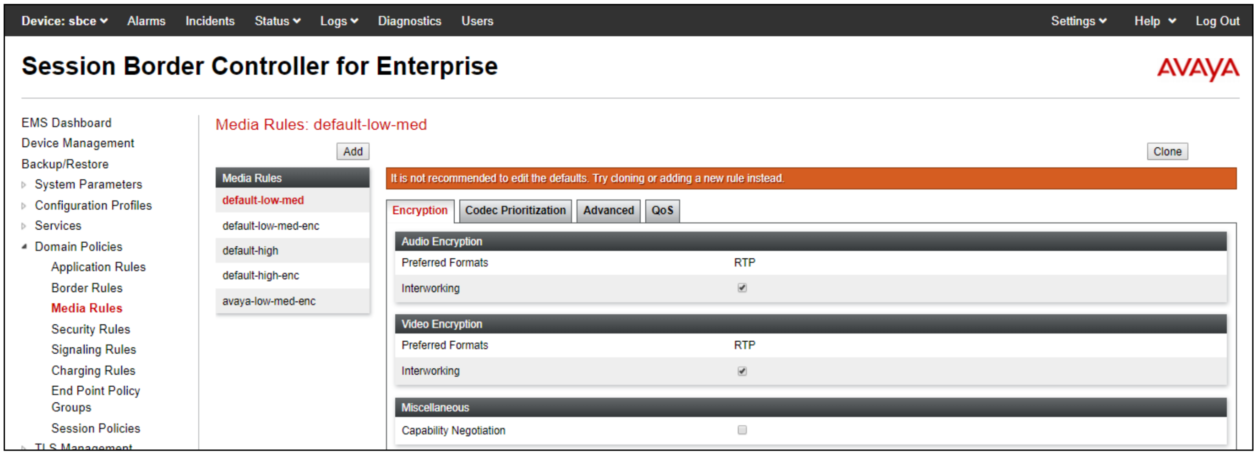


7.7.2 Border Rules

The Border Rule specifies if NAT is utilized (on by default), as well as detecting SIP and SDP Published IP addresses.

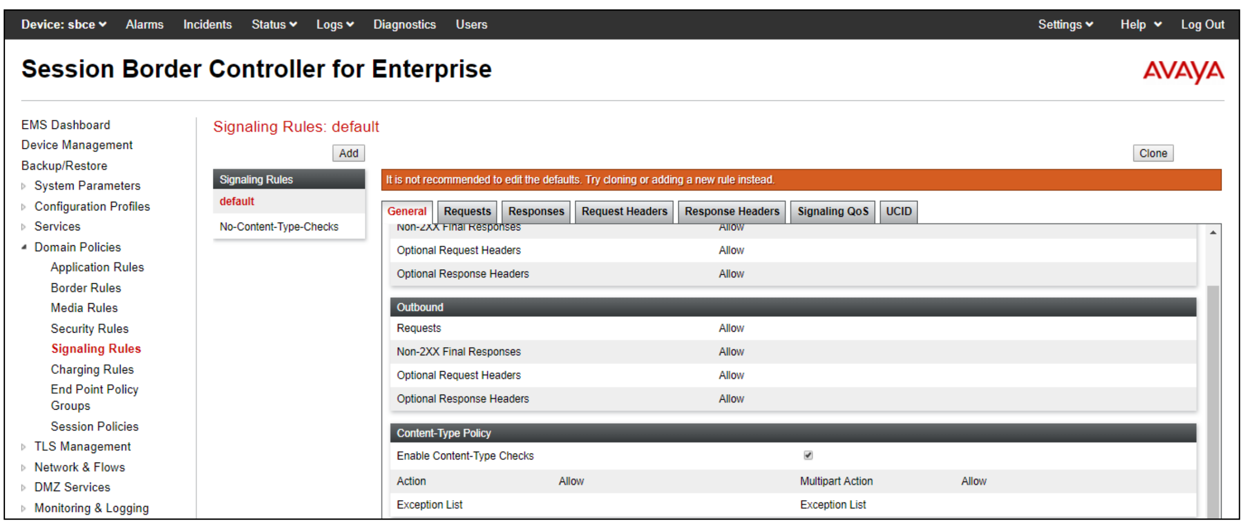
7.7.3 Media Rules

This Media Rule will be applied to both directions and therefore, only one rule is needed. In the solution as tested, the **default-low-med** rule was utilized. No customization was required.



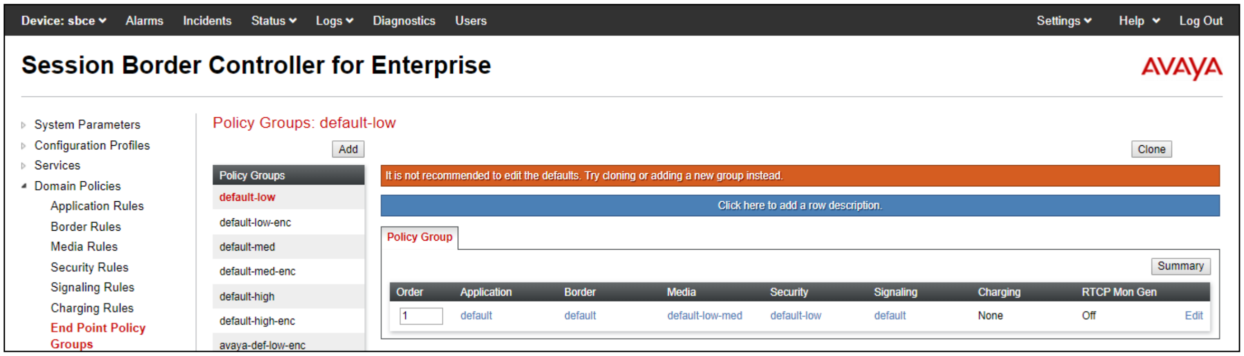
7.7.4 Signaling Rules

The **default** Signaling Rule was utilized. No customization was required.



7.7.5 Endpoint Policy Groups

In the solution as tested, the **default-low** rule was utilized. This rule incorporated the media and Signaling Rules specified above, as well as other policies.



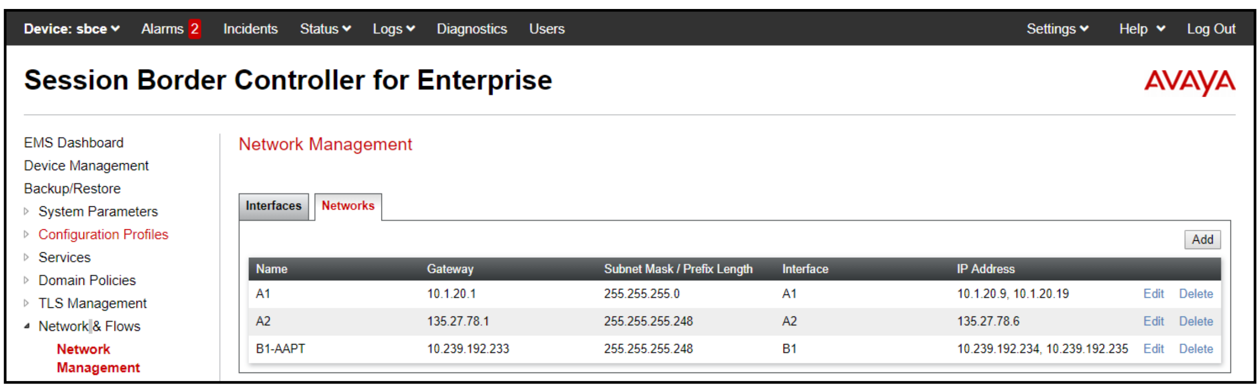
7.8 Network & Flows

The **Network & Flows** feature for SIP allows you to view aggregate system information, and manage various device-specific parameters which determine how a particular device will function when deployed in the network. Specifically, you have the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality, end-point and session call flows.

7.8.1 Network Management

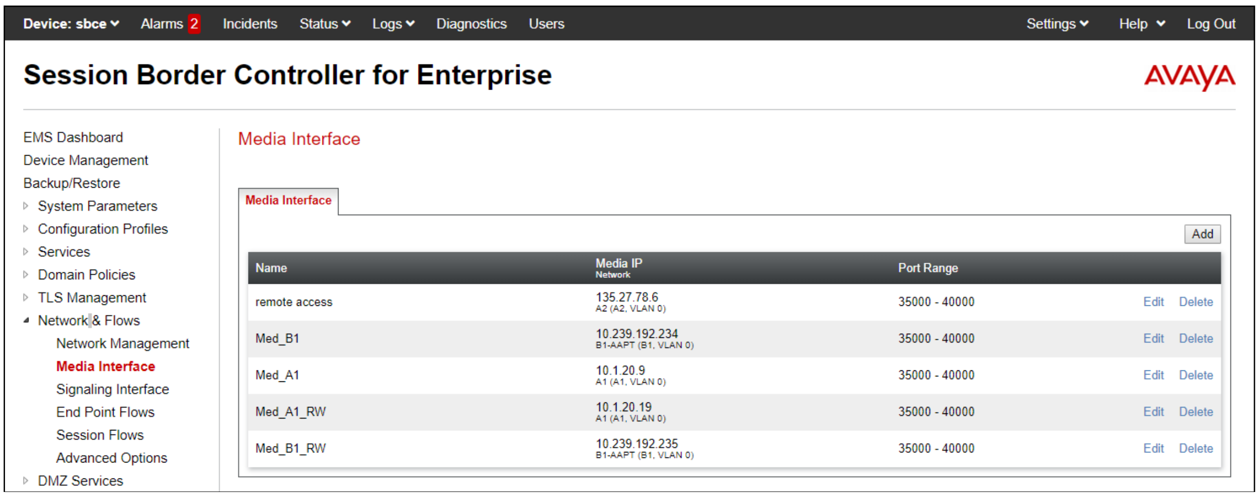
1. Select **Networks & Flows → Network Management** from the menu on the left-hand side.
2. The **Interfaces** tab displays the enabled/disabled interfaces. In the reference configuration, interfaces A1 (private) and B1 (public) interfaces are used.
3. Select the **Networks** tab to display the IP provisioning for the A1 and B1 interfaces. These values are normally specified during installation. These can be modified by selecting **Edit**; however some of these values may not be changed if associated provisioning is in use.

Note: B1 has two IP Addresses configured for each interface. One is used for SIP trunking, another one is used for Remote Worker. Configuration for Remote Worker is out of scope of this document.



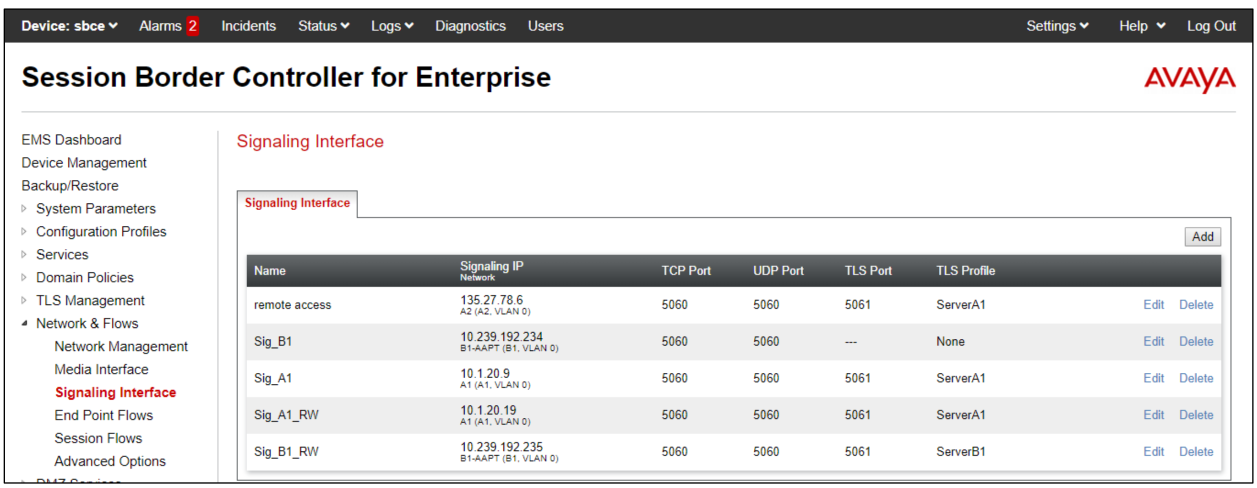
7.8.2 Media Interfaces

1. Select **Networks & Flows** from the menu on the left-hand side (not shown).
2. Select **Media Interface.**
3. Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:
    - **Name**: **Med_A1**.
    - **IP Address**: **10.1.20.9** (Avaya SBCE A1 address).
    - **Port Range**: **35000-40000**.
4. Click **Finish** (not shown).
5. Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:
    - **Name**: **Med_B1**.
    - **IP Address**: **10.239.192.234** (Avaya SBCE B1 address).
    - **Port Range**: **35000-40000**.
6. Click **Finish** (not shown). Note that changes to these values require an application restart. The completed **Media Interface** screen is shown below.



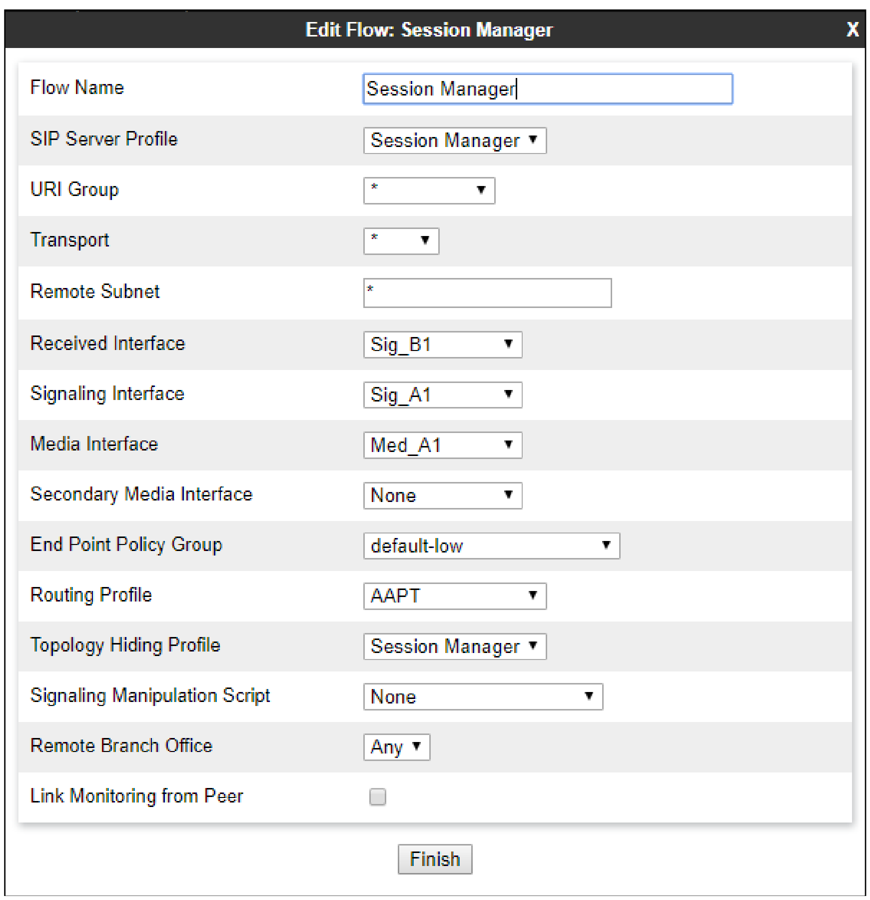
7.8.3 Signaling Interface

1. Select **Networks & Flows** from the menu on the left-hand side (not shown).
2. Select **Signaling Interface**.
3. Select **Add** (not shown) and enter the following:
   - **Name**: **Sig_A1**.
   - **IP Address**: **10.1.20.9** (Avaya SBCE A1 address).
   - **TCP Port**: **5060**.
   - **UDP Port**: **5060**.
   - **TLS Port**: **5060**.
4. Click **Finish** (not shown).
5. Select **Add** again, and enter the following:
   - **Name**: **Sig_B1**.
   - **IP Address**: **10.239.192.234** (Avaya SBCE B1 address).
   - **TCP Port**: **5060**.
   - **UDP Port**: **5060**.
6. Click **Finish** (not shown). Note that changes to these values require an application restart.



7.8.4 Endpoint Flows – For Session Manager

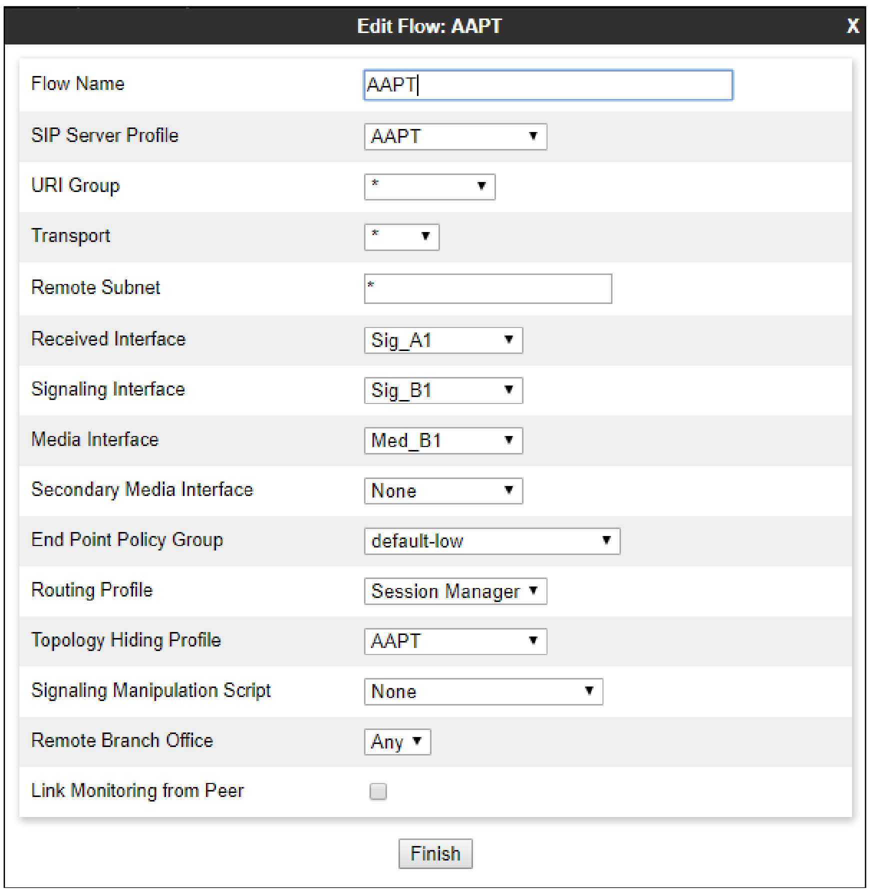
1. Select **Networks & Flows → Endpoint Flows** from the menu on the left-hand side (not shown).
2. Select the **Server Flows** tab (not shown).
3. Select **Add**, (not shown) and enter the following:
   - **Name**: **Session Manager**.
   - **SIP Server Profile**: **Session Manager**.
   - **URI Group**: *****.
   - **Transport**: *****.
   - **Remote Subnet**: *****.
   - **Received Interface**: **Sig_B1**.
   - **Signaling Interface**: **Sig_A1**.
   - **Media Interface**: **Med_A1**.
   - **End Point Policy Group**: **default-low**.
   - **Routing Profile**: **AAPT**.
   - **Topology Hiding Profile**: **Session Manager**.
   - Let other values default.
4. Click **Finish** .



7.8.5 Endpoint Flows – For AAPT

Repeat step **1** through **4** from **Section 7.8.4**, with the following changes:

- **Name**: **AAPT**.
- **SIP Server Profile**: **AAPT**.
- **URI Group**: *****.
- **Transport**: *****.
- **Remote Subnet**: *****.
- **Received Interface**: **Sig_A1**.
- **Signaling Interface**: **Sig_B1**.
- **Media Interface**: **Med_B1**.
- **End Point Policy Group**: **AAPT**.
- **Routing Profile**: **Session Manager**.
- **Topology Hiding Profile**: **AAPT**.



8. Verification Steps

The following steps may be used to verify the configuration.

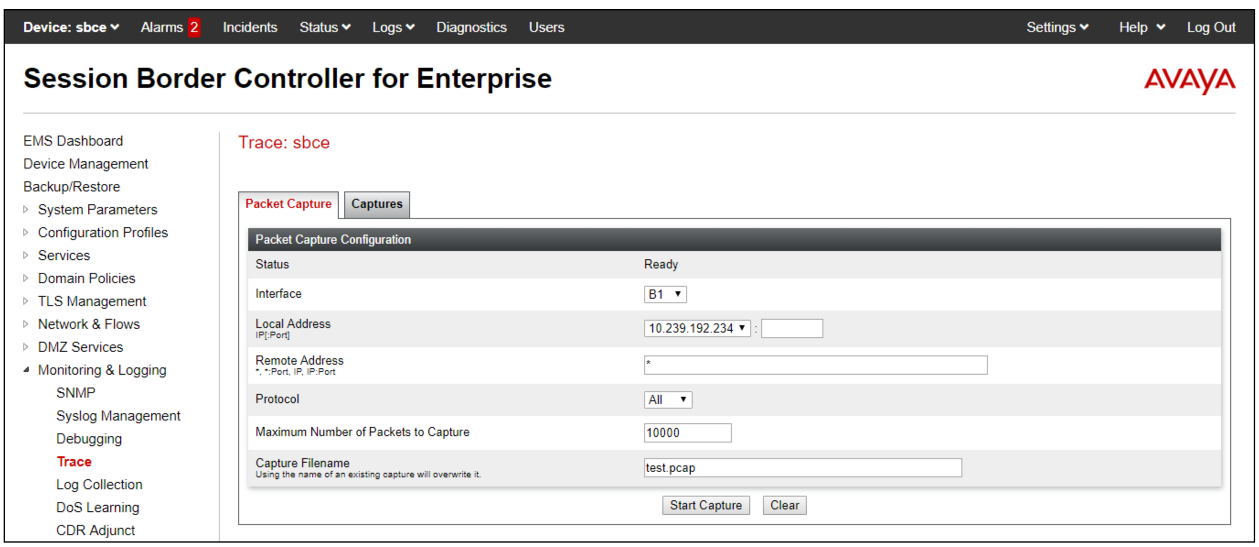
8.1 Avaya Session Border Controller for Enterprise

Log into the Avaya SBCE as shown in **Section 7**. Across the top of the display are options to display **Alarms**, **Incidents**, **Logs**, and **Diagnostics**. In addition, the most recent Incidents are listed in the lower right of the screen.

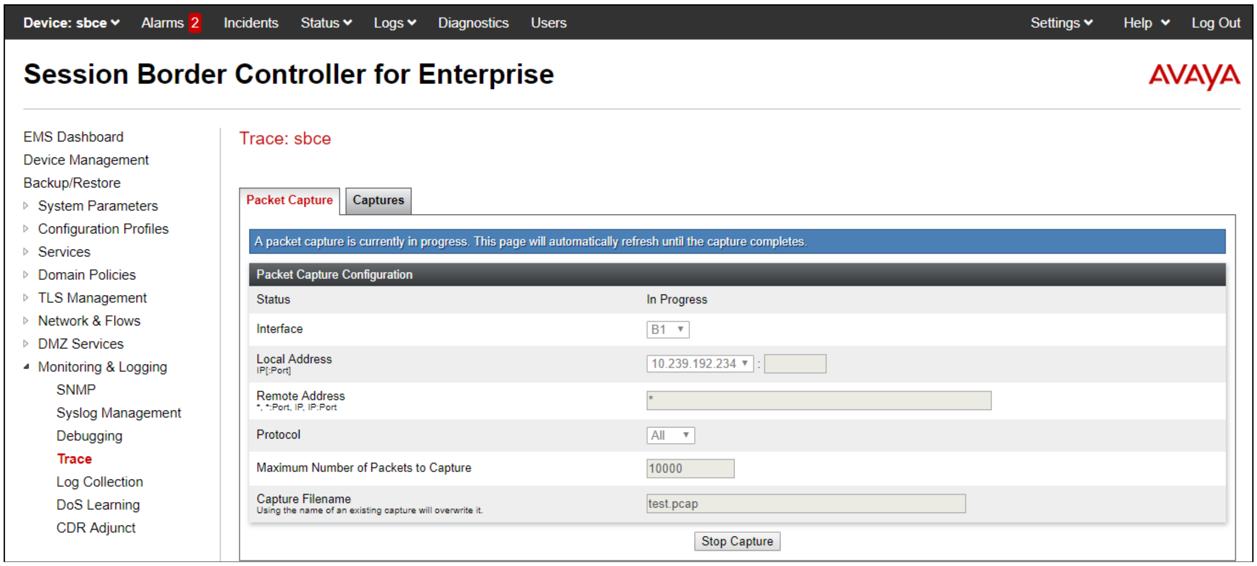
Protocol Traces

Avaya SBCE can take internal traces of specified interfaces.

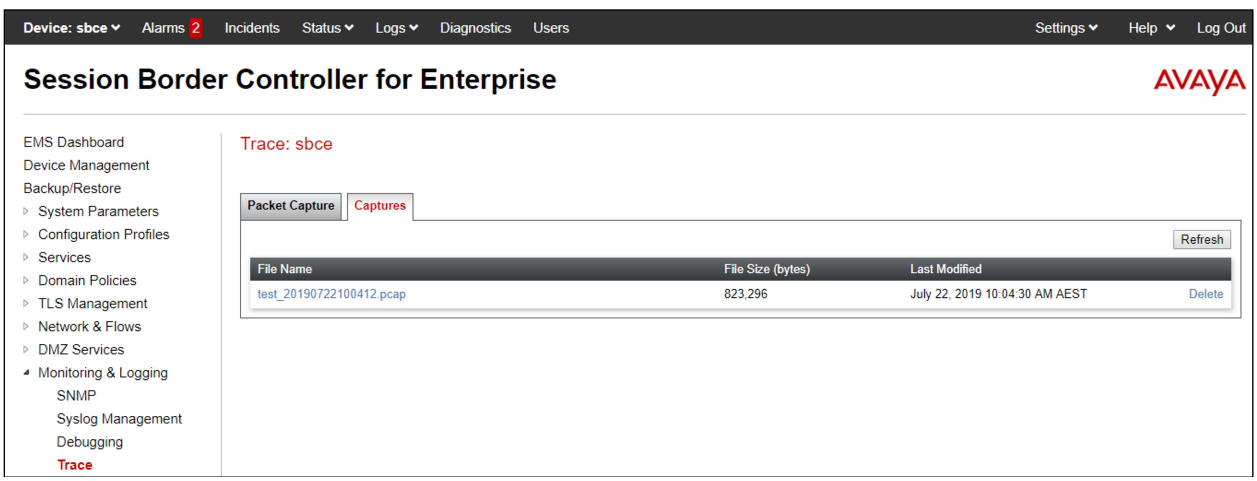
1. Navigate to **Monitoring & Logging → Trace**.
2. Select the **Packet Capture** tab and select the following:
   - Select the desired **Interface** from the drop down menu (e.g., **B1**).
   - Specify the **Maximum Number of Packets to Capture** (e.g., **10000**).
   - Specify a **Capture Filename** (e.g., **test.pcap**).
   - Unless specific values are required, the default values may be used for the **Local Address**, **Remote Address**, and **Protocol** fields.
   - Click **Start Capture** to begin the trace.



The capture process will initialize and then display the following **In Progress** status window:



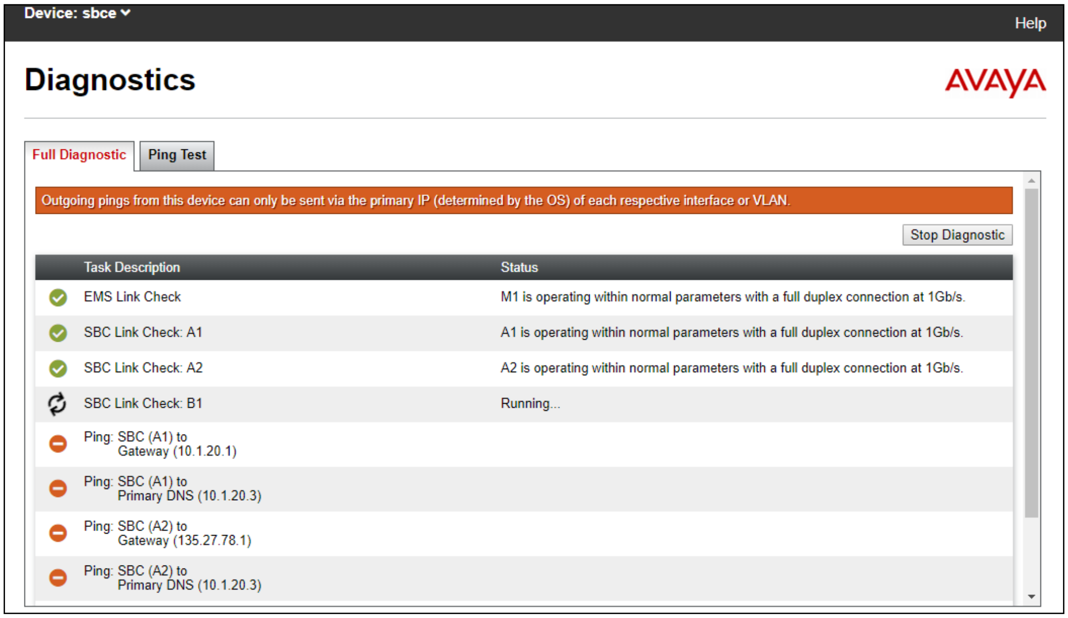
3. Run the test.
4. When the test is completed, select the **Stop Capture** button shown above.
5. Click on the **Captures** tab and the packet capture is listed as a *.pcap* file with the date and time added to filename specified in **Step 2**.
6. Click on the **File Name** link to download the file and use Wireshark to open the trace.

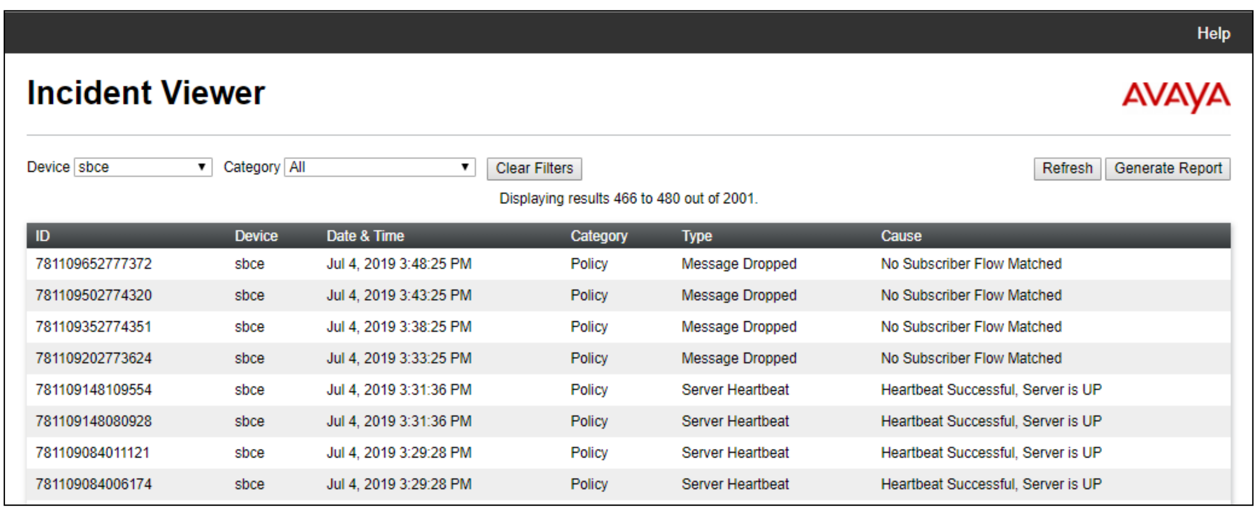


The following section details various methods and procedures to help diagnose call failure or service interruptions. As detailed in previous sections, the demarcation point between the AAPT SIP Trunk Service and the customer SIP PABX is the customer SBC.
On either side of the SBC, various diagnostic commands and tools may be used to determine the cause of the service interruption. These diagnostics can include:

- Ping from the SBC to the AAPT network gateway.

- Ping from the SBC to the Session Manager.
- Ping from the AAPT network towards the customer SBC.
- Note any Incidents or Alarms on the Dashboard screen of the SBC.

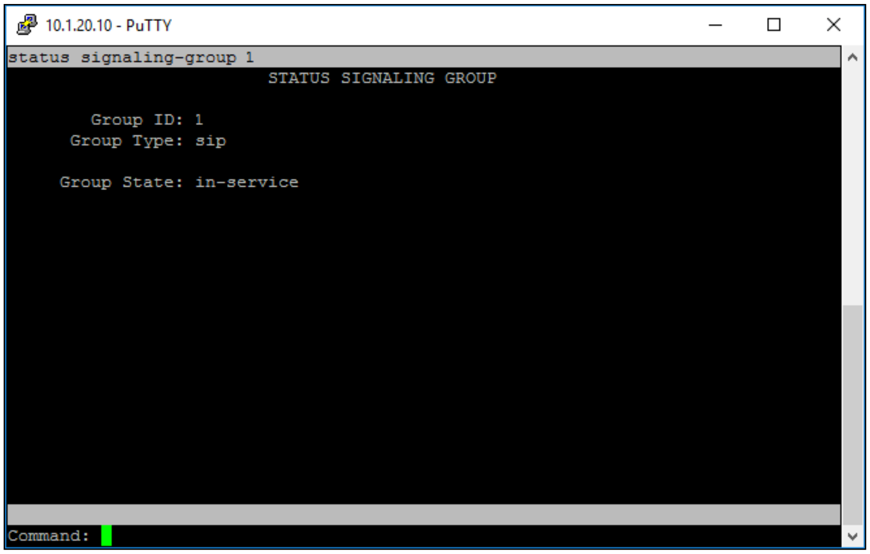




8.2 Avaya Aura® Communication Manager

The following examples are only a few of the monitoring commands available on Communication Manager.

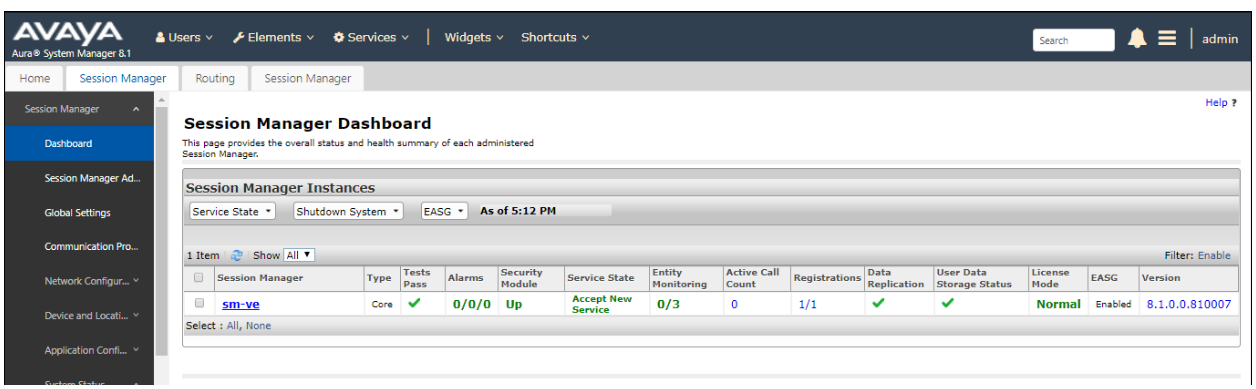
- Verify signaling status, trunk status



8.3 Avaya Aura® Session Manager Status

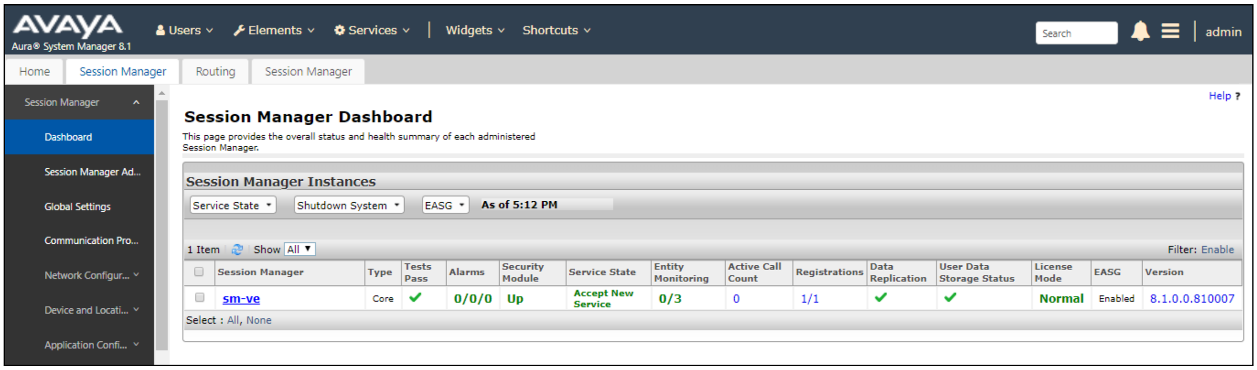
The Session Manager configuration may be verified via System Manager.

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

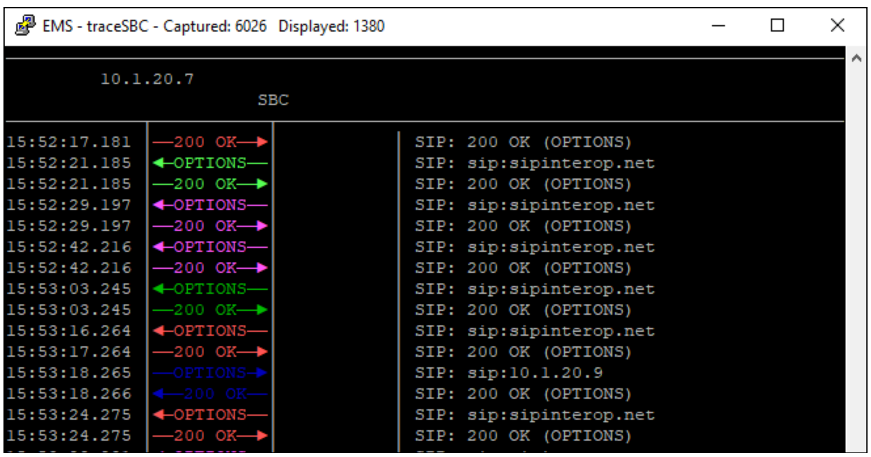


2. The Session Manager Dashboard is displayed. Note that the **Test Passed**, **Alarms**, **Service State**, and **Data Replication** columns, all show good status. In the **Entity Monitoring Column**, Session Manager shows that there are **0** (zero) alarms out of the **3** Entities defined.

3. Clicking on the **0/3** entry in the **Entity Monitoring** column, results in the following display:



Options messages between Avaya SBCE and Session Manager:



8.4 Telephony Services

1. Place inbound/outbound calls, 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.

9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, and Avaya Session Border Control for Enterprise 8.0 can be configured to interoperate successfully with AAPT SIP Voice SIP Trunking service. This solution allows enterprise users access to the PSTN using the AAPT SIP Voice SIP Trunking service connection. Please refer to **Section 2** for exceptions.

10. Additional References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1] *Deploying Avaya Aura® Communication Manager in Virtualized Environment R8.1*, Jun 2019

[2] *Administering Avaya Aura® Communication Manager R8.1*, Jun 2019

[3] *Upgrading Avaya Aura® Communication Manager R8.1*, Jun 2019

[4] *Deploying Avaya Aura® System Manager in Virtualized Environment Release 8.1*, Jun 2019

[5] *Upgrading Avaya Aura® System Manager to Release 8.1*, Jun 2019

[6] *Administering Avaya Aura® System Manager Release 8.1*, Jun 2019

[7] *Deploying Avaya Aura® Session Manager in Virtualized Environment Release 8.1*, Jun 2019

[8] *Upgrading Avaya Aura® Session Manager Release 8.1*, Jun 2019

[9] *Administering Avaya Aura® Session Manager Release 8.1*, Jun 2019

[10] *Deploying Avaya Session Border Controller for Enterprise Release 8.0*, Mar 2019

[11] *Upgrading Avaya Session Border Controller for Enterprise Release 8.0*, Feb 2019

[12] *Administering Avaya Session Border Controller for Enterprise Release 8.0*, Feb 2018

[13] *Deploying and Updating Avaya Aura Media Server Appliance Release 8.0*, Mar 2019

[14] *Implementing and Administering Avaya Aura Media Server Release 8.0*, Apr 2019

[15] *Deploying and Upgrading Avaya G450 Branch Gateway Release 8.1*, Jun 2019

[16] *Administering Avaya G450 Branch Gateway Release 8.1*, Jun 2019

[17] *Deploying Avaya Aura® Messaging using VMware® in the Virtualized Environment*, Mar 2019

[18] *Administering Avaya Aura® Messaging*, Mar 2019

[19] *RFC 3261 SIP: Session Initiation Protocol*, http://www.ietf.org/

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.