Avaya Solution & Interoperability Test Lab

Application Notes for Optus Evolve Voice SIP Trunking Service with Avaya Aura® Communication Manager 7.0, Avaya Aura® Session Manager 7.0 and Avaya Session Border Controller for Enterprise 7.0 - Issue 1.0

Abstract

These Application Notes illustrate a sample configuration of Avaya Aura® Communication Manager Release 7.0 and Avaya Aura® Session Manager 7.0 with SIP Trunks to the Avaya Session Border Controller for Enterprise (Avaya SBCE) when used to connect the Optus Evolve SIP Trunking Service available from Optus (Australia).

Purely as an example, the lab setup is configured in a non-redundant configuration. Additional resiliency could be built in as per the standard supported configurations documented in other Avaya publications.

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

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes illustrate a sample configuration Avaya Aura® Communication Manager Release 7.0 and Avaya Aura® Session Manager 7.0 with SIP Trunks to the Avaya Session Border Controller for Enterprise (Avaya SBCE) when used to connect the Optus Evolve Voice SIP Trunking Service available from Optus (Australia).

Avaya Aura® Session Manager 7.0 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 7.0 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. The Avaya SBCE is the point of connection between Avaya Aura® Session Manager and the Optus Evolve 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 Enterprise SIP Trunking Service available from Optus (Australia) is one of many SIP-based Voice over IP (VoIP) services offered to Enterprises in Australia for a variety of voice communications needs. The Optus Evolve 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 the 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, the Avaya SBCE is also configured as a SIP entity and has associated SIP entity links assigned within the Avaya Aura® Session Manager.

In the documented example, the "Processor Ethernet" of the Avaya server running Avaya Aura® Communication Manager 7.0 is configured for SIP Trunking to Avaya Aura® Session Manager and the Avaya SBCE is utilizing TCP transport. The Avaya SBCE is connected to the Optus Evolve Voice SIP Trunk Service, and as it is an industry default amongst SIP Service Providers to use UDP for SIP signaling, the SIP signaling connectivity from the Avaya SBCE toward Optus Evolve Voice uses UDP.

The Avaya SBCE performs conversion between TCP transport for SIP signaling used by Avaya Aura® Session Manager to UDP transport commonly used by SIP Service Providers. The Avaya SBCE also performs security and topology-hiding at the enterprise edge. In the sample configuration, all SIP signaling and RTP media between the enterprise and Optus Evolve SIP Trunking Service solution flows through the Avaya SBCE.

A customer interested in SIP Trunk survivability may want a redundant pair of Avaya SBCEs at each site. Although the sample configuration verified in these Application Notes used only a single Avaya SBCE configuration, actual verification testing of the Avaya SBCE in a High Availability configuration with Avaya Aura® Communication Manager has been performed as part of Avaya DevConnect compliance testing.

2. General Test Approach and Test Results

The general test approach was to make calls through the Avaya SBCE while DoS policies are in place using various codec settings and exercising common and advanced PBX features.

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.

2.1 Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound and outbound call flows between Avaya Aura® Session Manager, Avaya Aura® Communication Manager, the Avaya SBCE, and the Optus Evolve Voice SIP Trunking Service.

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 Optus Evolve SIP Trunk network. Additional test cases supplied by Optus were also executed. Calls were made to and from the PSTN across the Optus Evolve Voice network. The following standard features were tested as part of this effort:

- SIP trunking (incoming and outgoing calls)
- Passing of DTMF events and their recognition by navigating automated menus (interacting with Avaya Aura® Messaging 6.3.3)
- PBX features such as hold, resume, conference and transfer
- EC500 – call extending to mobile
- G.711A and G.729A audio
- Network Call Redirection
- Basic Call Center scenarios
- Faxing (using T.38)

2.2 Test Results

Interoperability testing of Optus Evolve Voice SIP Trunking 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.

- **Faxing** using T.38 is supported, and was tested successfully.
- **Legacy unresolved issues** such as Blank Invites (Invites without SDP) were resolved with a change to the Server Interworking Profile for Optus (Delayed SDP handling was enabled).
- **Emergency '000' Services Limitations and Restrictions** - Although Optus provides Emergency Services dialing on '000", Optus does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with Optus Evolve SIP Trunk Service to complete '000' calls; therefore, it is the customer's responsibility to ensure proper operation with its equipment/software vendor.

  While the Optus Evolve SIP Trunking Service does support '000' calling capabilities under certain Calling Plans, there are circumstances when that '000' service may not be available. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the customer's location in the automatic location information database.
- **Avaya Network Call Redirection (NCR) must be disabled** (default) on the Avaya Aura® Communication Manager SIP trunk to the Optus Evolve Voice SIP Trunking Service as Optus Evolve Voice does not support REFER.

2.3 Support

- **Avaya:** Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com
- **Optus:** Customers should contact their Optus Business representative or follow the support links available on http://optus.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 VMware ESXi 5.5.
- Avaya Aura® Session Manager running on VMware ESXi 5.5.
- Avaya Aura® System Manager running on VMware ESXi 5.5.
- Avaya Aura® Messaging running on VMware ESXi 5.5.
- Avaya G450 Media Gateway.

- Avaya Aura® Media Server running on VMware ESXi 5.5. The Media Server can act as a media gateway Gxxx series.
- Avaya IP phones are represented with Avaya 9600 Series IP Telephones running H.323/SIP software.
- Avaya one-X® Communicator 6.2
- Avaya Communicator for Windows 2.1
- The Avaya SBCE provided Session Border Controller functionality, including, Network Address Translation, SIP header manipulation, and Topology Hiding between the Optus SIP Trunking Service and the enterprise internal network.
- Outbound calls were originated from a phone provisioned on Avaya Aura® Communication Manager. Signaling passed from Avaya Aura® Communication Manager and Avaya Aura® Session Manager to the Avaya SBCE, before being sent to the Telecom network for termination.
- Inbound calls were sent from Optus, through the Avaya SBCE to the Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Communication Manager terminated the call to the appropriate phone extension. The analog and H.323/SIP phones on the enterprise side registered to the Communication Manager or Session Manager.

All IP addresses shown in the diagram are private IP addresses.

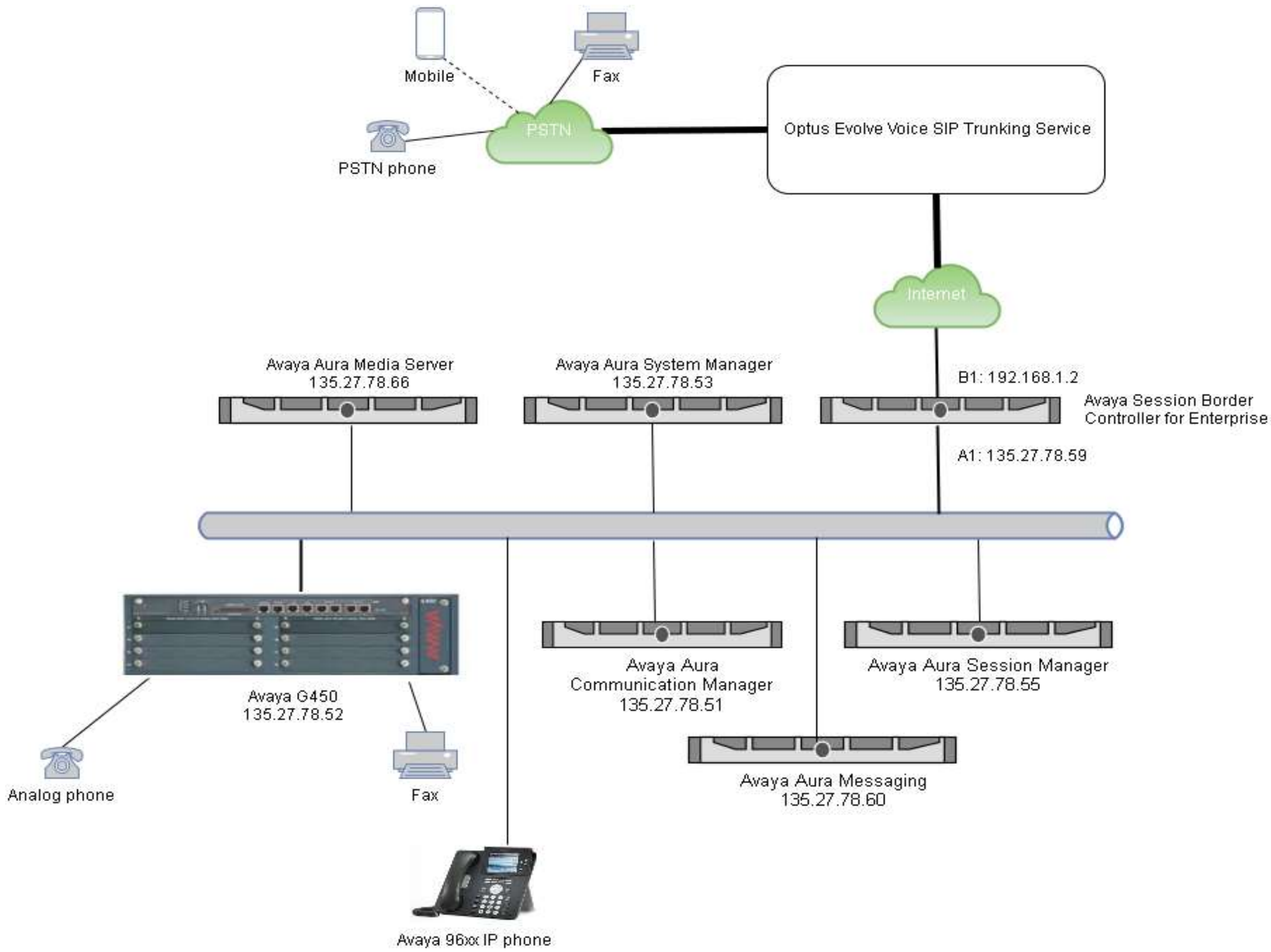


Figure 1: Network Components as Tested

4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya Aura® Communication Manager 7.0 SP2	7.0.0.2.0.441.22684
Avaya Aura® Session Manager 7.0	7.0.0.0.700007
Avaya Aura® System Manager 7.0	Build No. - 7.0.0.0.16266-7.0.9.912 Software Update Revision No: 7.0.0.0.4016
Avaya Aura® Messaging 6.3.3	6.3.3.0.11348
Avaya Session Border Controller for Enterprise 7.0	7.0.0-21-6602

Component	Version
Avaya Media Gateway G450	G450_sw_37_20_0
Avaya Aura® Media Server 7.7	7.7.0.281
Avaya one-X® Communicator 6.2	6.2.10.03
Avaya Communicator for Windows 2.1	2.1.1.74
Avaya one-X® Agent H323 2.5.8	2.5.58020.0
Avaya 96xx Series Deskphone – SIP phone	S96x1_SALBR7_0_0r40_V4r83
Avaya 96xx Series Deskphone – H.323 phone	S9608_11HALBR6_6_1_15_V474
Service Provider	
Optus Evolve Voice	Genband CS2100 CVN16 (14.1.0.12), Acme Packet SBC edge (6.2)

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.

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: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                        Maximum Video Capable Stations: 41000 0
                   Maximum Video Capable IP Softphones: 1000  2
                       Maximum Administered SIP Trunks: 24000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0







        (NOTE: You must logoff & login to effect the permission changes.)
```

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
                No-License Mode Disabled? y   System Management Data Transfer? n
            Personal Station Access (PSA)? y              Tenant Partitioning? y
                       PNC Duplication? n        Terminal Trans. Init. (TTI)? y
                   Port Network Support? y                Time of Day Routing? y
                       Posted Messages? y        TN2501 VAL Maximum Capacity? y
                                                           Uniform Dialing Plan? y
                     Private Networking? y      Usage Allocation Enhancements? y
                Processor and System MSP? y
                     Processor Ethernet? y                 Wideband Switching? y
                                                                     Wireless? n
                          Remote Office? y
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y


        (NOTE: You must logoff & login to effect the permission changes.)
```

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

F1=Cancel  F2=Refresh  F3=Submit  F4=Clr Fld  F5=Help  F6=Update  F7=Nxt Pg  F8=Prv Pg
```

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.

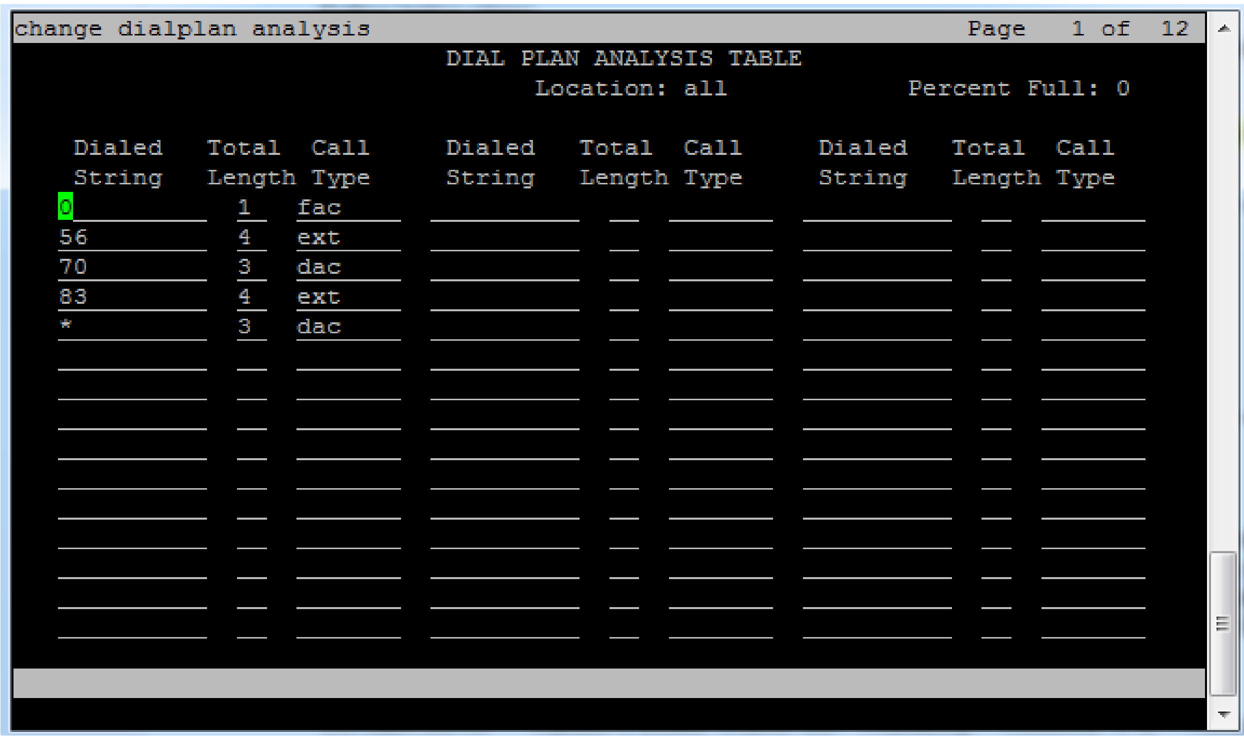
```
display system-parameters features                              Page   9 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

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: 61
          International Access Code: 0011

SCCAN PARAMETERS
   Enable Enbloc Dialing without ARS FAC? n

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


F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

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 83 and 56 for Communication Manager extensions.
  - 3-digit dial access code (indicated with a **Call Type** of **dac**), e.g., access code 70x for SIP Trunk Access Codes (TAC).



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 Entities in **Section 6.3.2**.

Follow the steps shown below:

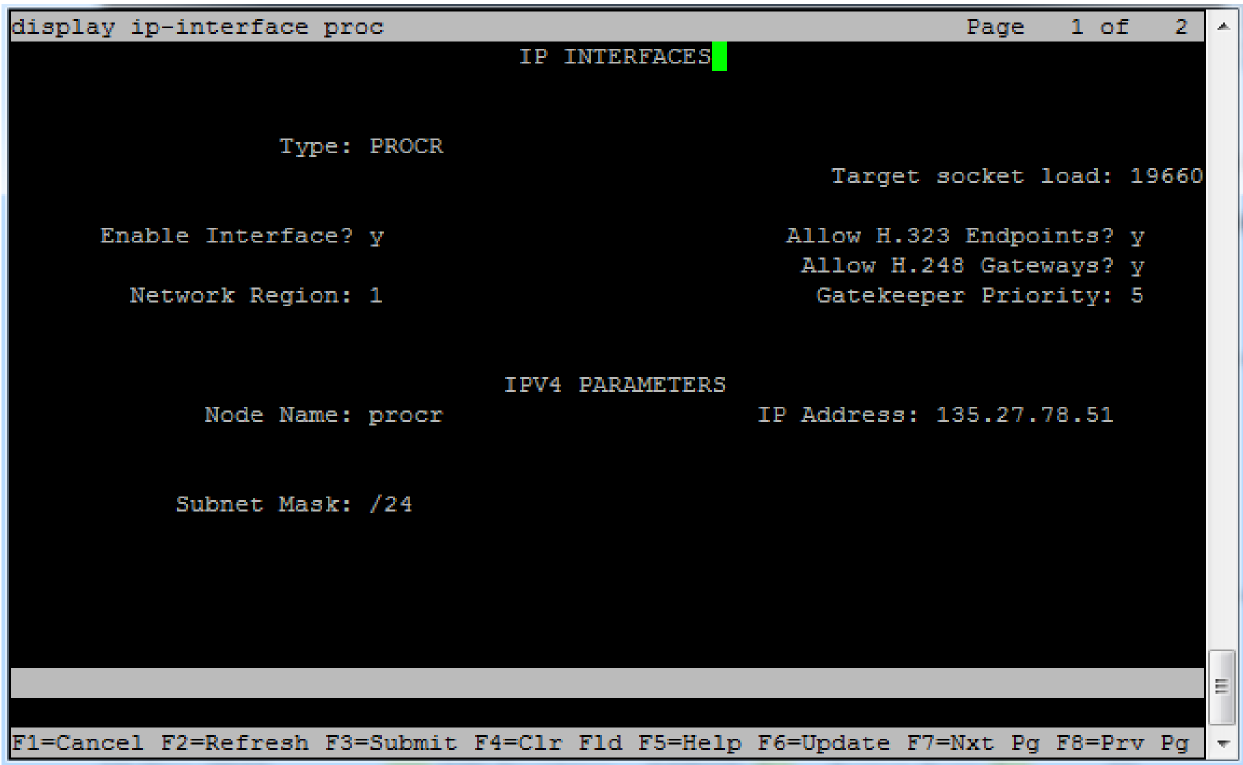
- Enter the **change node-names ip** command, and add a node name and IP address for the following:
  - Avaya SBCE private network interface (e.g., **A1-SBCE** and **135.27.78.59**).
  - Session Manager SIP signaling interface (e.g., **ve3-sm** and **135.27.78.55**).

```
display node-names ip                                           Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
A1-SBCE             135.27.78.59
default             0.0.0.0
procr               135.27.78.51
procr6              ::
ve3-sm              135.27.78.55

( 5  of 5    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

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 parameters.


```
display ip-interface proc                                       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: 135.27.78.51


           Subnet Mask: /24





F1=Cancel  F2=Refresh  F3=Submit  F4=Clr Fld  F5=Help  F6=Update  F7=Nxt Pg  F8=Prv Pg
```

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 Media Gateway. 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
Location: 1        Authoritative Domain: sipinterop.net
    Name: Default                 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: 35000                       IP Audio Hairpinning? y
   UDP Port Max: 39999
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

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

On **Page 4**, define the IP codec set to be used for traffic between region 1 and other regions. In the compliance testing, Communication Manager, the Avaya G450 Media Gateway, IP/SIP phones, Session Manager and the 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
 14
 15


F1=Cancel  F2=Refresh  F3=Submit  F4=Clr Fld  F5=Help  F6=Update  F7=Nxt Pg  F8=Prv Pg
```

Non-IP telephones (e.g., analog, digital) derive their network region from the IP interface of the Avaya 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 proc                                        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: 135.27.78.51


           Subnet Mask: /24






F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

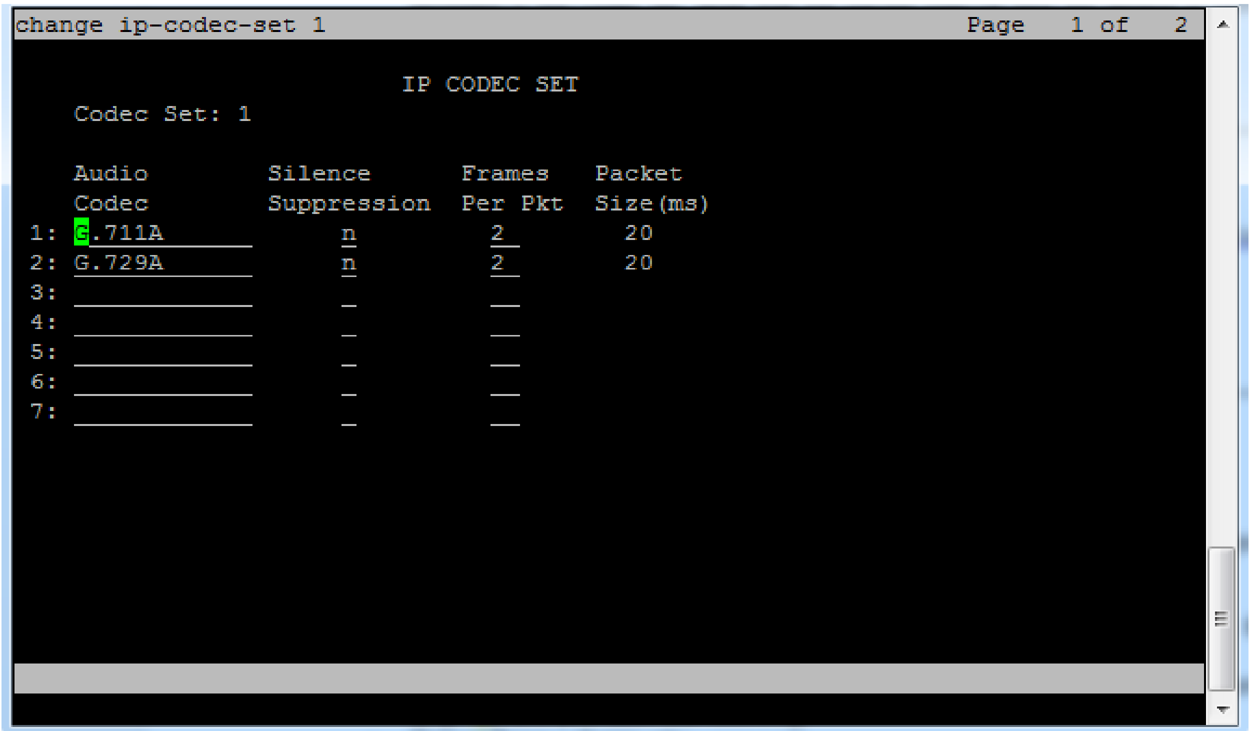
To define network region 1 for the Avaya G450 Media Gateway, use **change media-gateway** command as shown in the following screen.

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

                      Type: g430
                      Name: g430
                 Serial No: 10IS07356033
      Link Encryption Type: any-ptls/tls
            Network Region: 1                   Location: 1
                                               Site Data:
             Recovery Rule: none

               Registered?  y
     FW Version/HW Vintage: 37 .20 .0  /1
         MGP IPV4 Address: 135.27.78.52
         MGP IPV6 Address:
     Controller IP Address: 135.27.78.51
               MAC Address: 00:1b:4f:3c:d0:c9

    Mutual Authentication? n
```

5.7 IP Codec Parameters

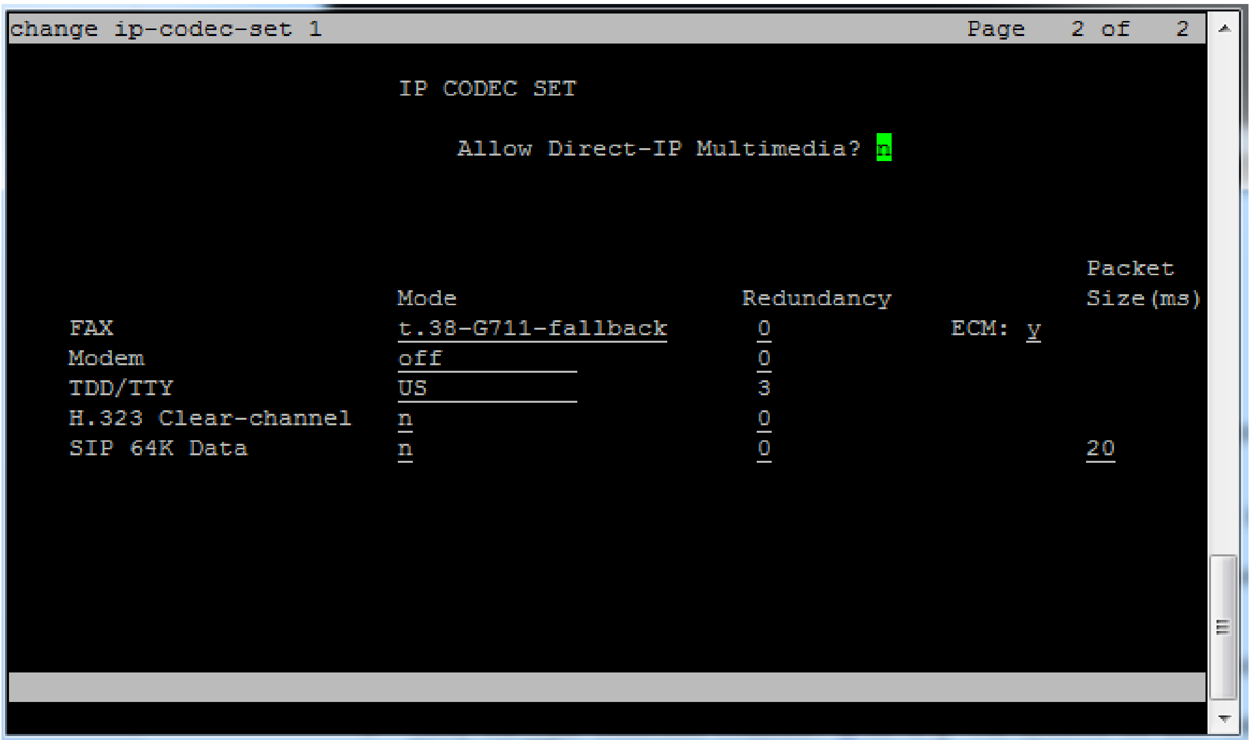
Follow the steps shown below:

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.729A** are included in the codec list. Note that the packet interval size will default to 20ms.



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

                          IP CODEC SET

    Codec Set: 1

    Audio          Silence      Frames   Packet
    Codec          Suppression  Per Pkt  Size(ms)
 1: G.711A              n          2        20
 2: G.729A              n          2        20
 3:
 4:
 5:
 6:
 7:
```

2. On **Page 2** of the ip-codec-set form, set **FAX Mode** to **t.38-G711-fallback.**



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

                          IP CODEC SET

                              Allow Direct-IP Multimedia? n


                                                                     Packet
                          Mode                 Redundancy            Size(ms)
    FAX                   t.38-G711-fallback     0           ECM: y
    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.

5.8.1 SIP trunk for public calls

This section describes the steps for administering the SIP trunk to Session Manager. This trunk corresponds to the **ve3-cm-external** SIP Entity defined in **Section 6.3.2**.

1. Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **3**), 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., **ve3-sm**).
   - **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 Avaya Media Gateway when possible (known as shuffling).
- **Enable Layer 3 Test** – Set to **y**. This directs Communication Manager to send SIP OPTIONS messages to Session Manager to check link status.
- **OPTIONAL**: If desired, set **Initial IP-IP Direct Media** is set to **Y**. Otherwise leave it disable (default).
- Use the default parameters on **Page 2** of the form (not shown).

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

 Group Number: 3              Group Type: sip
  IMS Enabled? y        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? y        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                   Far-end Node Name: ve3-sm
 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? n          Alternate Route Timer(sec): 6
```

2. Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g., **3**). On **Page 1** of the **trunk-group** form, provision the following:
   - **Group Type** – Set to **sip**.
   - **Group Name** – Enter a descriptive name (e.g., **SIP to SM**).
   - **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., **703**).
   - **Direction** – Set to **two-way**.
   - **Service Type** – Set to **public-ntwrk**.
   - **Signaling Group** – Set to the signaling group administered in **Step 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**).

```
change trunk-group 3                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip          CDR Reports: y
  Group Name: SIP to SM                   COR: 1        TN: 1        TAC: 703
   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: 3
                                                     Number of Members: 10
```

3. On **Page 2** of the **Trunk Group** form:
   - Set the **Preferred Minimum Session Refresh Interval(sec):** to **600**.

```
change trunk-group 3                                              Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                                Redirect On OPTIM Failure: 5000

            SCCAN? n                                   Digital Loss Group: 18
                       Preferred Minimum Session Refresh Interval(sec): 600

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n




 Caller ID for Service Link Call to H.323 1xC: station-extension
```

4. On **Page 3** of the **Trunk Group** form:
   - Set **Numbering Format:** to **private**.

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



                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

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




 Show ANSWERED BY on Display? y

 DSN Term? n                      SIP ANAT Supported? n
```

5. On **Page 4** of the **Trunk Group** form:
   - Set **Telephone Event Payload Type** to the RTP payload type recommended by the Optus service (e.g., **101**).

```
change trunk-group 3                                            Page   4 of  21
                              PROTOCOL VARIATIONS

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

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


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

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

5.8.2 SIP trunks for local calls

Repeat the same steps as in **5.8.1** with the following changes:

1. Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g., **4**), and provision the following:
   - **Near-end Listen Port** – Set to **5061** and **Far-end Listen Port** – Set to **5062**

```
change signaling-group 4                                        Page   1 of   3
                                SIGNALING GROUP

 Group Number: 4              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 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: ve3-sm
 Near-end Listen Port: 5061                   Far-end Listen Port: 5062
                                           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? n              Alternate Route Timer(sec): 6
```

2. Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g., **4**). On **Page 1** of the **trunk-group** form, provision the following:
   - **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g., **704**).
   - **Direction** – Set to **two-way**.
   - **Service Type** – Set to **tie**.
   - **Signaling Group** – Set to the signaling group administered in **Step 1** (e.g., **4**).
   - **Number of Members** – Enter the maximum number of simultaneous calls desired on this trunk group (based on licensing) (e.g., **10**).

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

Group Number: 4                  Group Type: sip            CDR Reports: y
  Group Name: internal                              COR: 1        TN: 1        TAC: 704
   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: 4
                                                     Number of Members: 10
```

5.9 Calling Party Information

The calling party number is sent in the SIP "From", "Contact" and "PAI" headers. Since private numbering is selected to define the format of this number, both the private numbering table and the public-unknown-numbering tables will need to be configured. Use the **change public-unknown-numbering** command to create an entry for each extension which has a DID assigned. DID numbers are provided by the service provider. They are used to authenticate the caller. The screen below shows a subset of the 10-digit DID numbers assigned for testing. These numbers were mapped to the enterprise extensions 83xx. These same 10-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these extensions.

```
change public-unknown-numbering 0                                 Page   1 of   2
                              NUMBERING - PUBLIC/UNKNOWN FORMAT
                                                Total
Ext Ext               Trk         CPN            CPN
Len Code              Grp(s)      Prefix         Len
                                                      Total Administered: 2
 4  83                3           029020         10      Maximum Entries: 9999
 4  83                4                          4
                                                      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.
```

Repeat the same steps for the private numbering table using the **change private-numbering** command:

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

Ext Ext                Trk          Private           Total
Len Code               Grp(s)       Prefix            Len
 5  2                  3                              5     Total Administered: 3
 4  83                 3            029020            10       Maximum Entries: 540
 4  83                 4                              4
```

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. DID number sent by Optus can be mapped to an extension 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.

```
change inc-call-handling-trmt trunk-group 3                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number       Del Insert
 Feature         Len      Digits
 public-ntwrk     10 029020           6
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
```

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 0 was used as the ARS access code. An enterprise caller will dial 0 to reach an outside line. To define feature access code (**fac**) **0**, use the **change dialplan analysis** command as shown below.

```
change dialplan analysis                                       Page   1 of  12
                            DIAL PLAN ANALYSIS TABLE
                                 Location: all           Percent Full: 0

    Dialed    Total  Call      Dialed    Total  Call      Dialed    Total  Call
    String    Length Type      String    Length Type      String    Length Type
   0            1   fac
   56           4   ext
   70           3   dac
   83           4   ext
   *            3   fac
```

Use the **change feature-access-codes** command to define **0** as the **Auto Route Selection (ARS) – Access Code 1**.

```
change feature-access-codes                                    Page   1 of  11
                               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: 0      Access Code 2:
                 Automatic Callback Activation:          Deactivation:
Call Forwarding Activation Busy/DA:          All:        Deactivation:
   Call Forwarding Enhanced Status:          Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                   CDR Account Code Access Code:
                        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 **0**. 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 **3** for an outbound call which contains the SIP trunk to the service provider (as defined next).

```
change 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     3          emer           n
    0011                    12   20    3          pubu           n
    02                      10   10    3          pubu           n
    03                      10   10    3          pubu           n
    04                      10   10    3          pubu           n
    07                      10   10    3          pubu           n
    08                      10   10    3          pubu           n
    13                      6    6     3          pubu           n
    1300                    10   10    3          pubu           n
    1800                    10   10    3          pubu           n
                                                                 n
                                                                 n
                                                                 n
                                                                 n
                                                                 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 **3** in the following manner.

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the SIP service provider. 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**: **unk-unk** All calls using this route pattern will use the private numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.9**.

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3       Pattern Name: SIP to SM
    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: 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                               unk-unk   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
 6: y y y y y n  n            rest                                         none
```

5.12 Avaya G450 Media Gateway Provisioning

In the reference configuration, a G450 Media Gateways is provisioned. The G450 is located in the 123ER site and is used for local DSP resources, announcements, Music On Hold, etc.

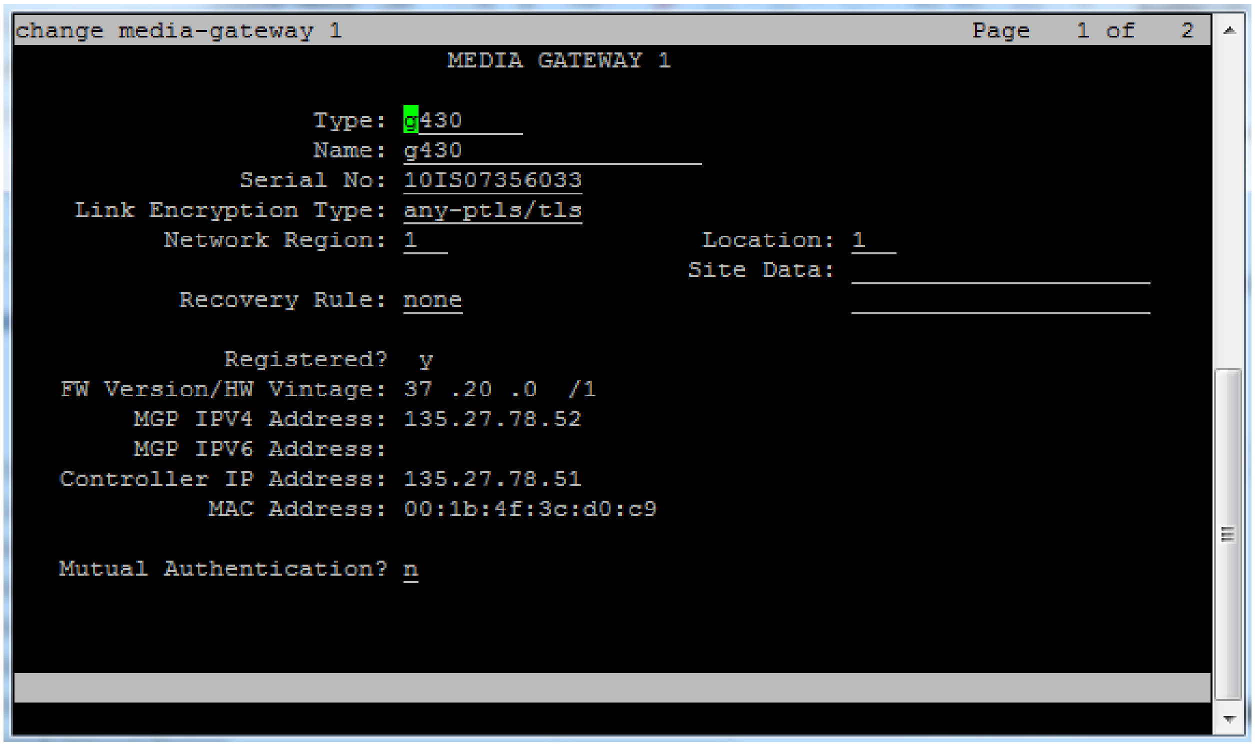
Note – Only the Media Gateway provisioning associated with the G450 registration to Communication Manager is shown below.

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., **ve3-gw-*???(super)#***).
2. Enter the **show system** command and note the G450 serial number (e.g., **10IS07356033**).
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., **135.27.78.51**).
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., **ve3-gw**).
   - Set **Serial Number** = Enter the serial number copied from **Step 2** (e.g., **10IS07356033**).

- Set the **Encrypt Link** parameter as desired (**n** 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., **ve3-gw-*001(super)#***).

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



5.13 Save Communication Manager Translations

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

6. Configure Avaya Aura® Session Manager

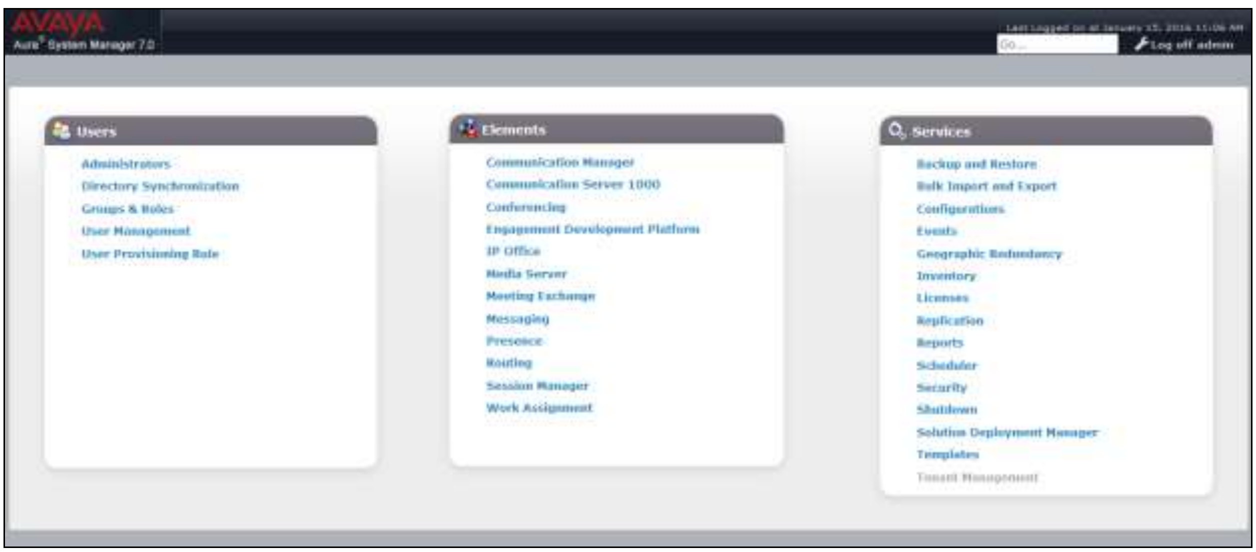
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Location that can be used by SIP Entities
- SIP Entities corresponding to Communication Manager, Session Manager and the 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.

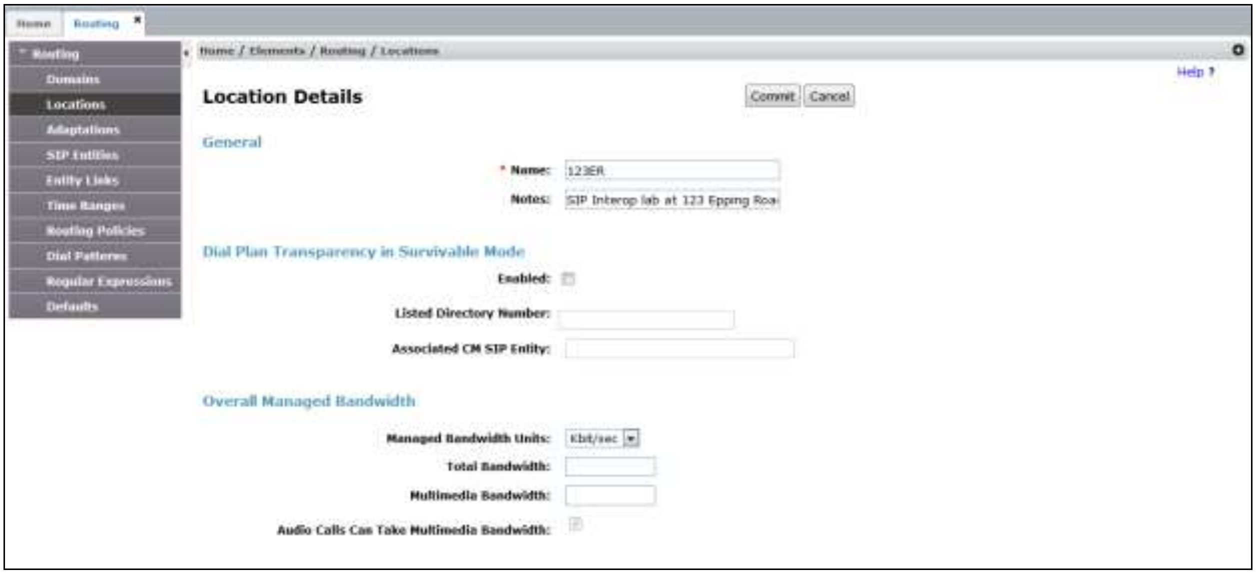
3. Click **Commit** to save (not shown).

6.2 Configure Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside. In the reference configuration, location **123ER** is configured.

Follow the steps shown below:

1. Select **Locations** from the left navigational menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.
    - **Name**: Enter a descriptive name for the Location (e.g., **123ER**).
    - **Notes**: Add a brief description.
2. In the **Location Pattern** section, click **Add** and enter the following value
    - **IP Address Pattern**: Leave blank.
3. Click **Commit** to save.



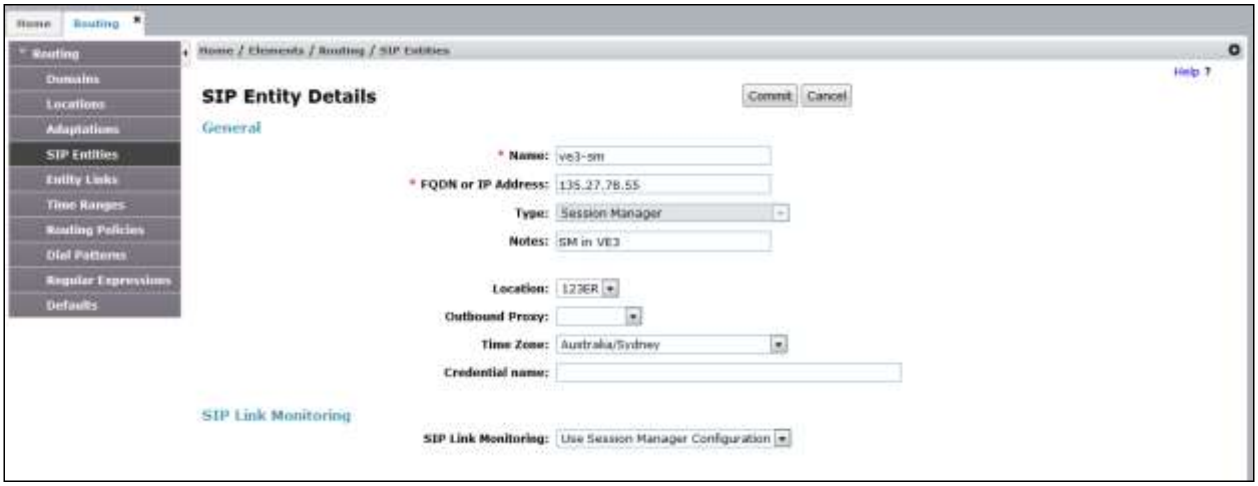
6.3 Configure SIP Entities

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., **ve3-sm**).
    - **FQDN or IP Address –** Enter the IP address of Session Manager signaling interface, (*not* the management interface), provisioned during installation (e.g., **135.27.78.54**).
    - **Type –** Verify **Session Manager** is selected.
    - **Location** – Select location **123ER.**
    - **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.



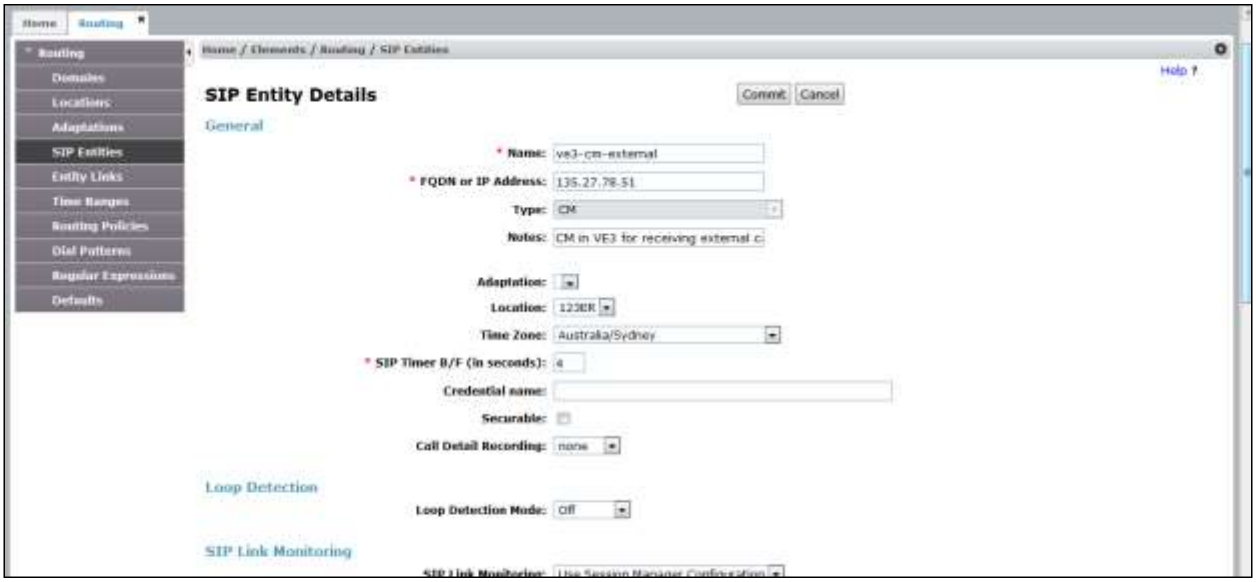
6.3.2 Configure Communication Manager SIP Entity

As there are two SIP trunks configured on Avaya Aura® Communication Manager in this compliance test, one trunk is used for incoming/outgoing from/to PSTN and another trunk is used for Avaya SIP extension and calls to Avaya Aura® Messaging, it is necessary to create two SIP Entities for Avaya Aura® Communication Manager: **ve3-cm-external** and **ve3-cm-internal**.

6.3.2.1 Communication Manager – Public

Follow the steps shown below:

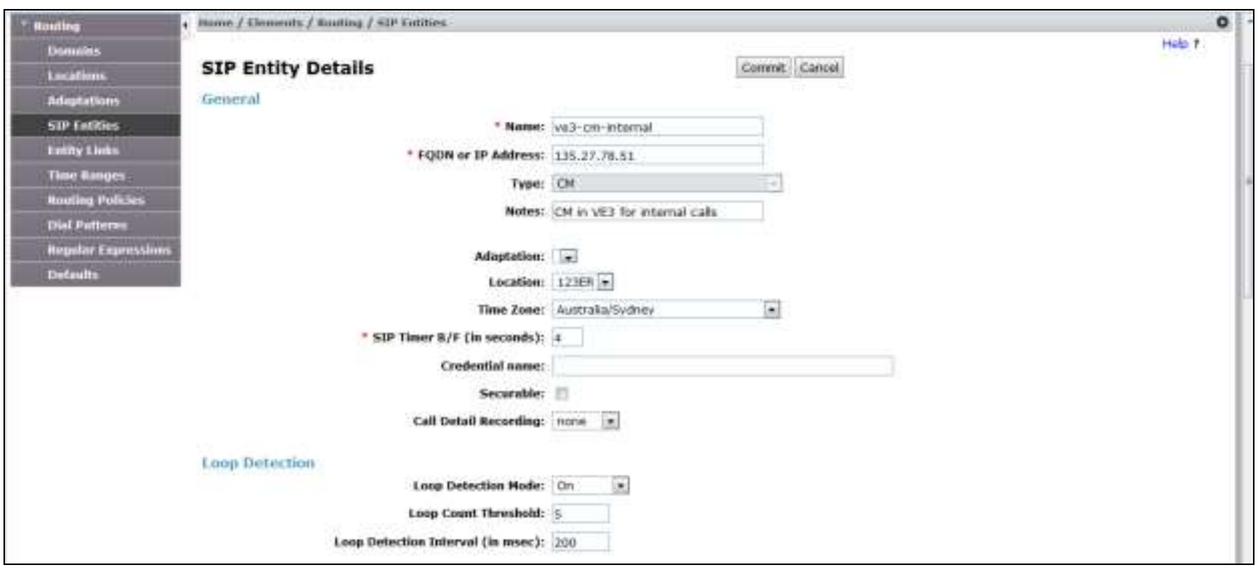
1. In the **SIP Entities** page, click on **New** (not shown).
2. In the **General** section of the **SIP Entity Details** page, provision the following:
   - **Name –** Enter a descriptive name (e.g. **ve3-cm-external**).
   - **FQDN or IP Address –** Enter the IP address of Communication Manager Processor Ethernet (procr) (e.g. **135.27.78.51**).
   - **Type –** Select **CM**.
   - **Location** – Select a Location **123ER** administered in **Section 6.2**.
   - **Time Zone** – Select the time zone in which Communication Manager resides.
   - In the **SIP Link Monitoring** section of the **SIP Entity Details** page select:
     - Select **Use Session Manager Configuration** for **SIP Link Monitoring** field, and use the default values for the remaining parameters.
3. Click on **Commit**.



6.3.2.2 Communication Manager – Local

Repeat the steps in **Section 6.3.2.1** with the following changes:

- **Name –** Enter a descriptive name (e.g., **ve3-cm-internal**).

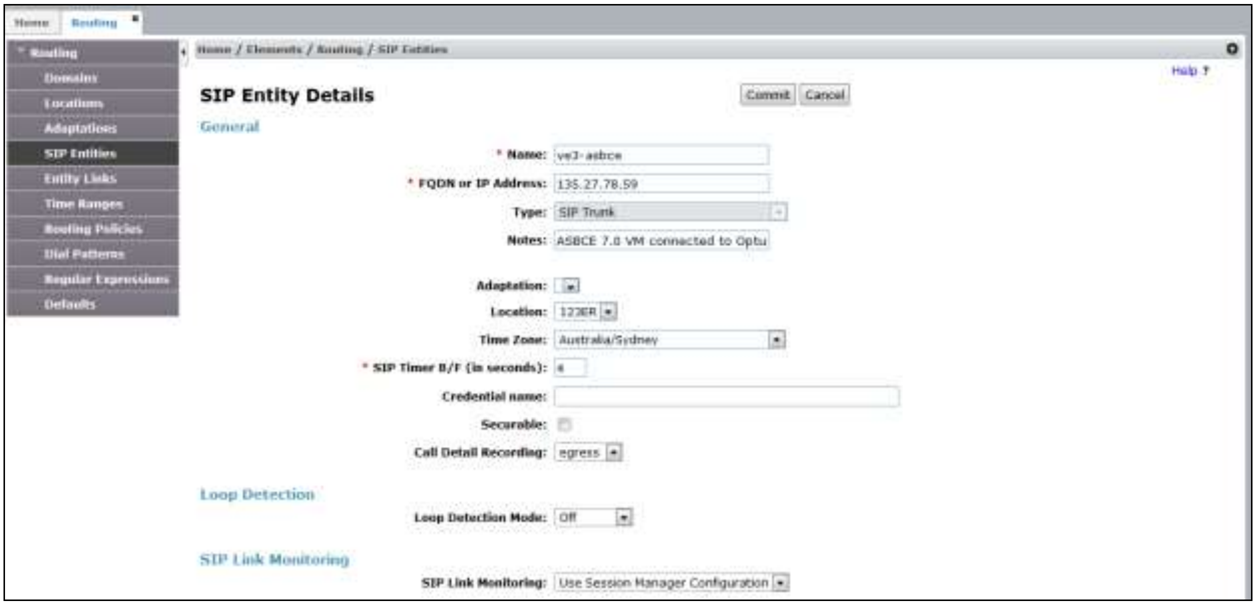


6.3.3 Configure Avaya SBCE SIP Entity

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

- **Name –** Enter a descriptive name (e.g., **ve3-asbce**).

- **FQDN or IP Address –** Enter the IP address of the A1 (private) interface of the Avaya SBCE (e.g., **135.27.78.59**).
- **Type –** Verify **Other** is selected.
- **Location** – Select location **123ER** (**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, three Entity Links were created, two for Communication Manager and another 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, ***TLS*** 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 6.3.3**
- **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

6.4.1.1 Entity Link for local (internal) calls

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., **ve3-sm_ve3-cm-internal_5061_TLS**).
   - **SIP Entity 1** – Select the SIP Entity administered in **Section 6.3.1** for Session Manager (e.g., **ve3-sm**).
   - **SIP Entity 1 Port** – Enter **5062**.
   - **Protocol –** Select **TLS**
   - **SIP Entity 2** –Select the SIP Entity administered in **Section 6.3.2** for the Communication Manager internal entity (e.g., **ve3-cm-internal**).
   - **SIP Entity 2 Port** - Enter **5061.**
   - **Connection Policy –** Select **Trusted**.
3. Click on **Commit**.

6.4.1.2 Entity Link for public calls

Repeat the steps in **6.4.1.1** with the following changes:

- **Name** – Enter a descriptive name (or have it created automatically) for this link to Communication Manager (e.g., **ve3-sm_ve3-cm-external_5061_TLS**).
- **SIP Entity 1 Port** – Enter **5061. Note that this port is different from the port in 6.4.1.1**
- **SIP Entity 2** –Select the SIP Entity administered in **Section 6.3.2** for the Communication Manager external entity (e.g., **ve3-cm-external**).

6.4.2 Configure Entity Link for Avaya SBCE

To configure this Entity Link, repeat the steps in **Section 6.4.1.1**, with the following changes:

- **Name** – Enter a descriptive name (or have it created automatically) for this link to the Avaya SBCE (e.g., **ve3-sm_ve3-asbce_5060_TCP**).
- **SIP Entity 1 Port** – Enter **5060**
- **Protocol –** Select **TCP**
- **SIP Entity 2** – Select the SIP Entity administered in **Section 6.3.3** for the Avaya SBCE entity (e.g., **ve3-asbce**).
- **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 6.4**. 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).

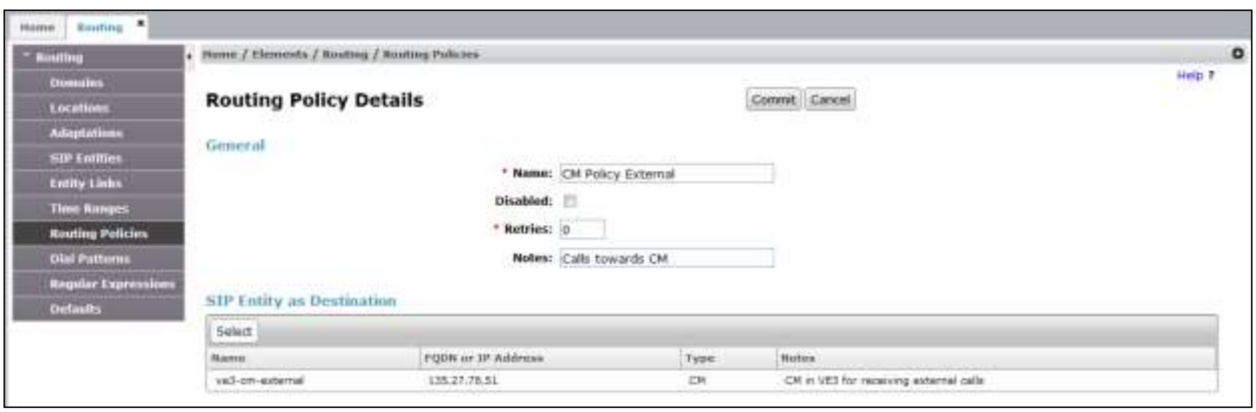
In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown). Select appropriate SIP entity to which this routing policy applies and click **Select**. The selected SIP Entity is displayed in the **Routing Policy Details** page as shown below. Use default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

6.5.1 Configure Routing Policy for Communication Manager

6.5.1.1 Routing Policy - Public

This Routing Policy is used for inbound calls from Optus.

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 Optus calls to Communication Manager (e.g., **CM Policy External**), 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 External Entity (**ve3-cm-external**), 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**.



6.5.1.2 Routing Policy - Local

This Routing Policy is used for local (internal) calls to Avaya SIP extensions or calls to Avaya Aura® Messaging. Repeat the steps in **6.5.1.1** with the following changes:

1. In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing internal enterprise calls to Communication Manager (e.g., **CM Policy Internal**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.
2. In the **SIP Entity List** page, select the SIP Entity administered in **Section 6.3.2** for the Communication Manager SIP External Entity (**ve3-cm-internal**), and click on **Select**.

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., **SBC Outgoing Policy**).
- **SIP Entity List** –Select the SIP Entity administered in **Section 6.3.3** for the Avaya SBCE entity (e.g., **ve3-asbce**).

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 Optus 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:

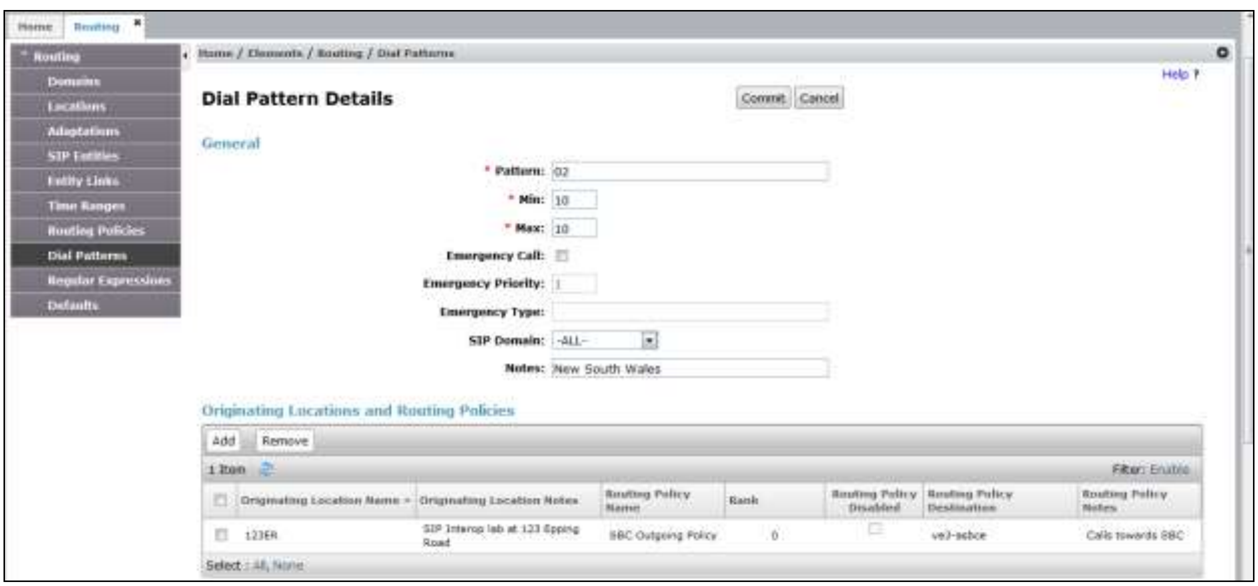
- **Pattern:** Enter a dial string that will be matched against the "Request-URI" of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria.
- **Notes:**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

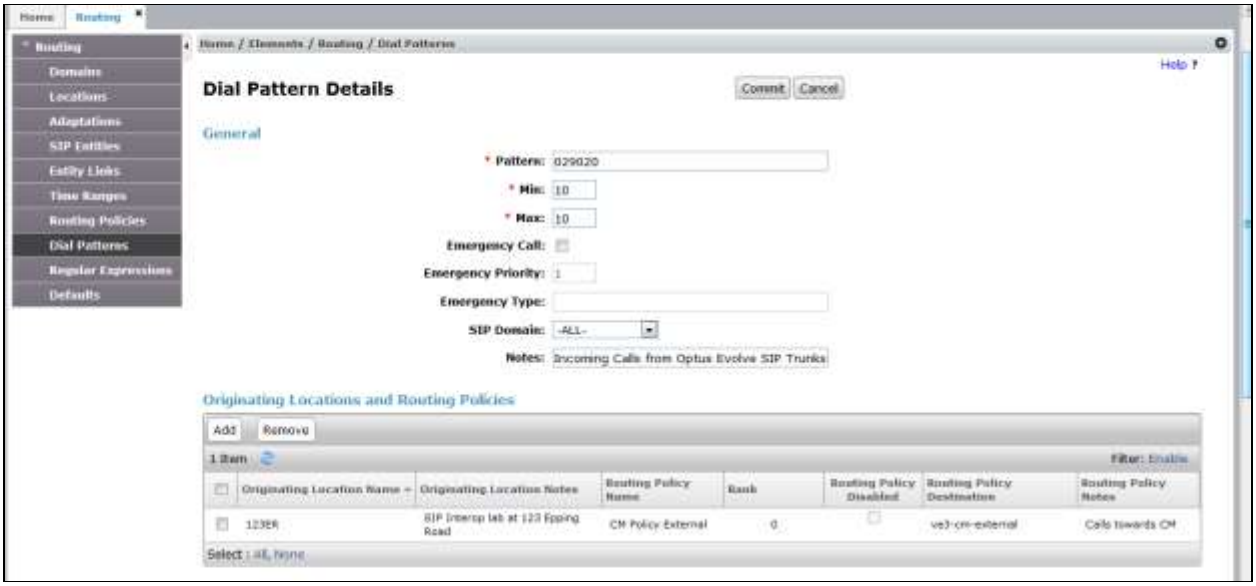
Default values can be used for the remaining fields. Click **Commit** to save.

Three examples of the dial patterns used for the compliance testing were shown below, one for outbound calls from the enterprise to the PSTN, one for inbound calls from the PSTN to the enterprise and another one for Avaya SIP extension.

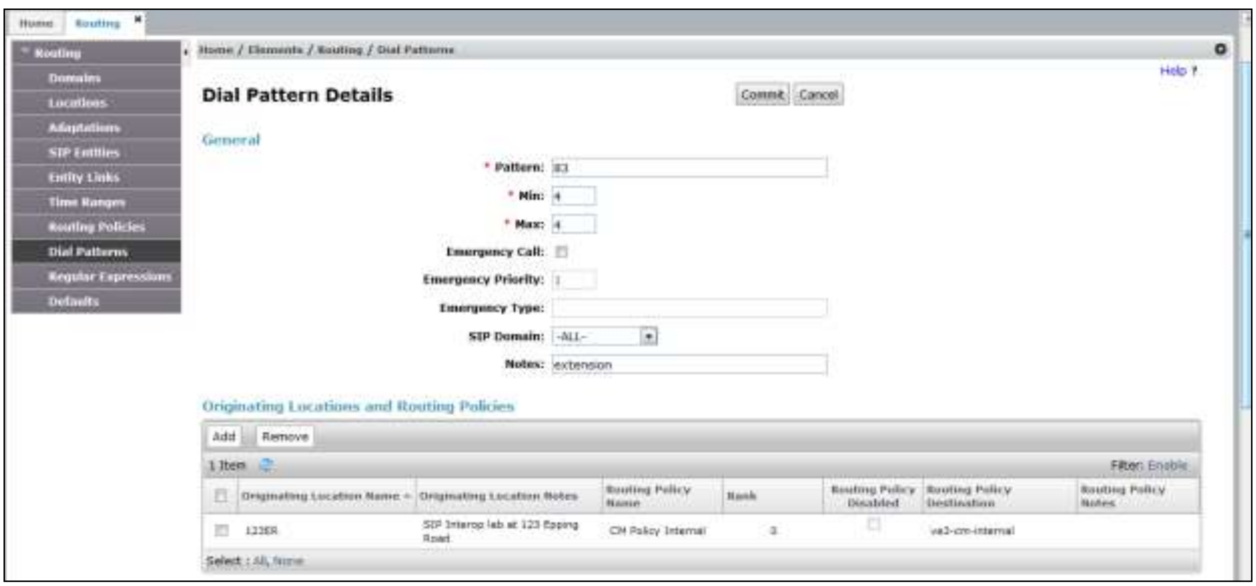
The first example shows that 10-digit dialed numbers that has a destination domain of "sipinterop.net" uses route policy to Avaya SBCE as defined in **Section 6.5.2**



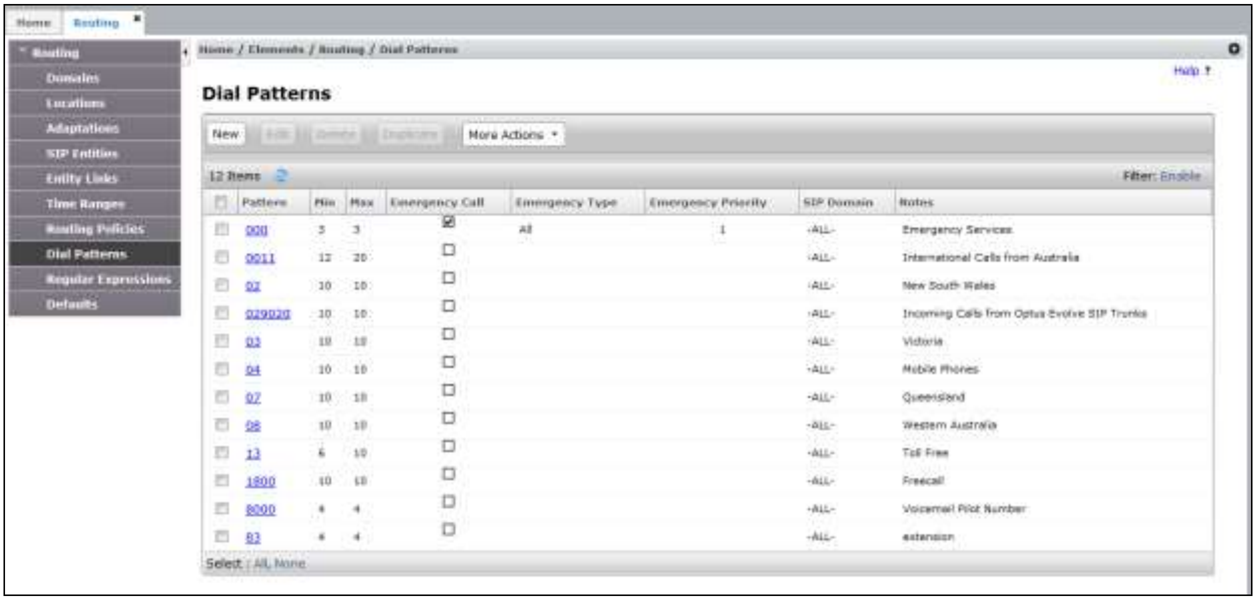
The second example shows that inbound 10-digit numbers that start with 020920 to domain "sipinterop.net" uses route policy to Communication Manager as defined in **Section 6.5.1.** These are the DID numbers assigned to the enterprise by Optus.



The third example shows that 4-digit pattern that start with 83 is used for Avaya SIP extension local calls.



All Dial Patterns used in the test:



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.

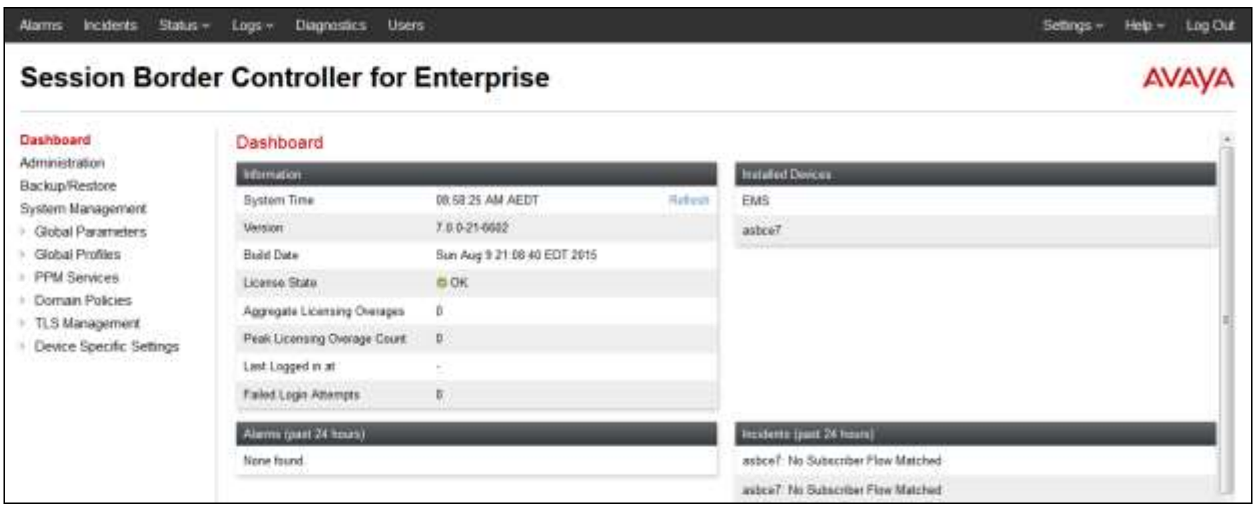
IMPORTANT! – During the Avaya SBCE installation, the Management interface of the Avaya SBCE must be provisioned on a different subnet than either of the Avaya SBCE private and public network interfaces (e.g., A1 and B1). If this is not the case, contact your Avaya representative to get this condition resolved.

As described in Section 3, the reference configuration places the private interface (A1) of the Avaya SBCE in the Common site, (135.27.78.59), with access to the **123ER** site. The connection to Optus uses the Avaya SBCE public interface B1 (IP address 192.168.1.2). The follow 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**.

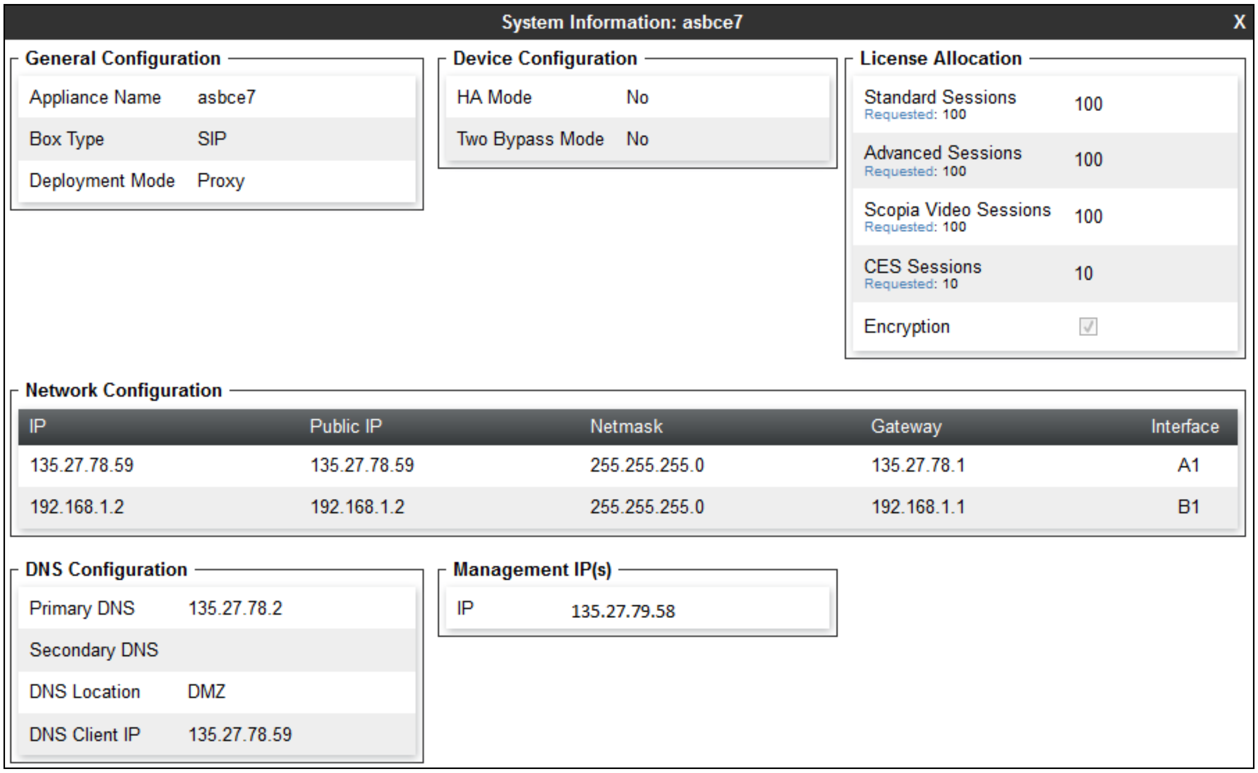
The main menu window will open. Note that the installed software version is displayed. Verify that the **License State** is **OK**. The SBCE will only operate for a short time without a valid license. Contact your Avaya representative to obtain a license.



7.1 System Management – Status

1. Select **System 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.



7.2 Global Profiles

7.2.1 Uniform Resource Identifier (URI) Groups

URI Group feature allows a user to create any number of logical URI Groups that are comprised of individual SIP subscribers located in that particular domain or group. These groups are used by the various domain policies to determine which actions (Allow, Block, or Apply Policy) should be used for a given call flow.

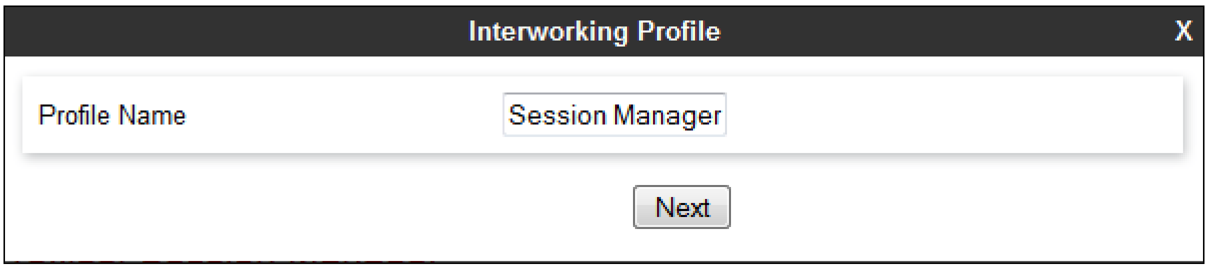
For this configuration testing, “*” is used for all incoming and outgoing traffic.

7.2.2 Server Interworking – Avaya

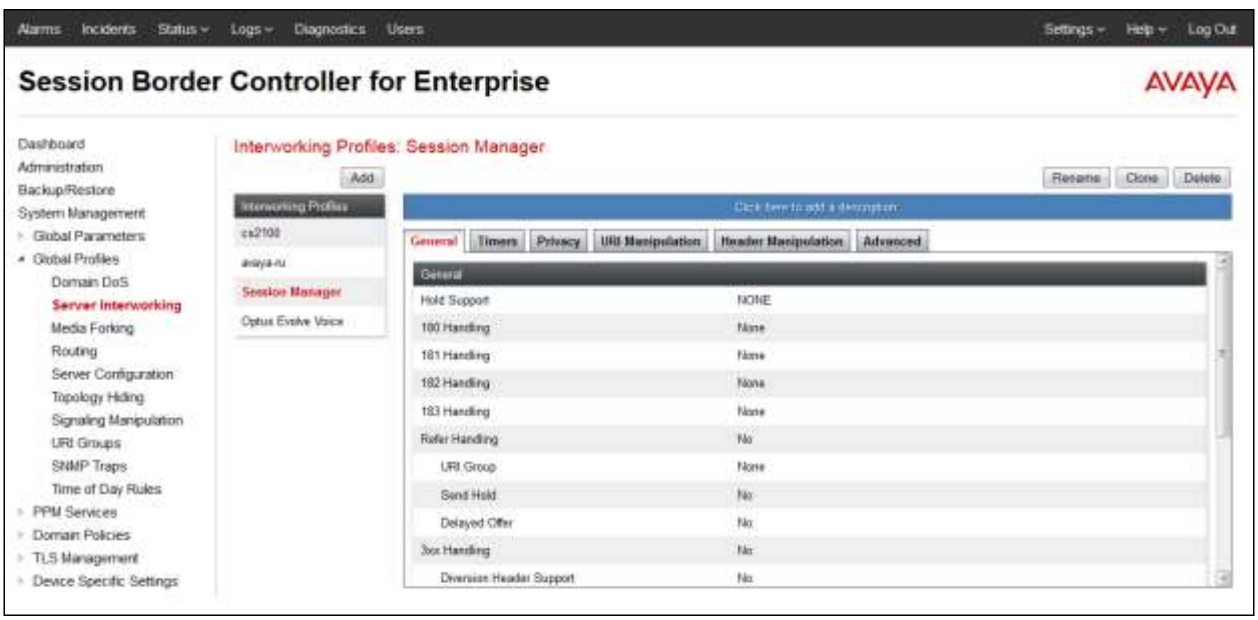
Server Interworking allows users to configure and manage various SIP call server-specific capabilities such as call hold and T.38 faxing. This section defines the profile for the connection to Session Manager.

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

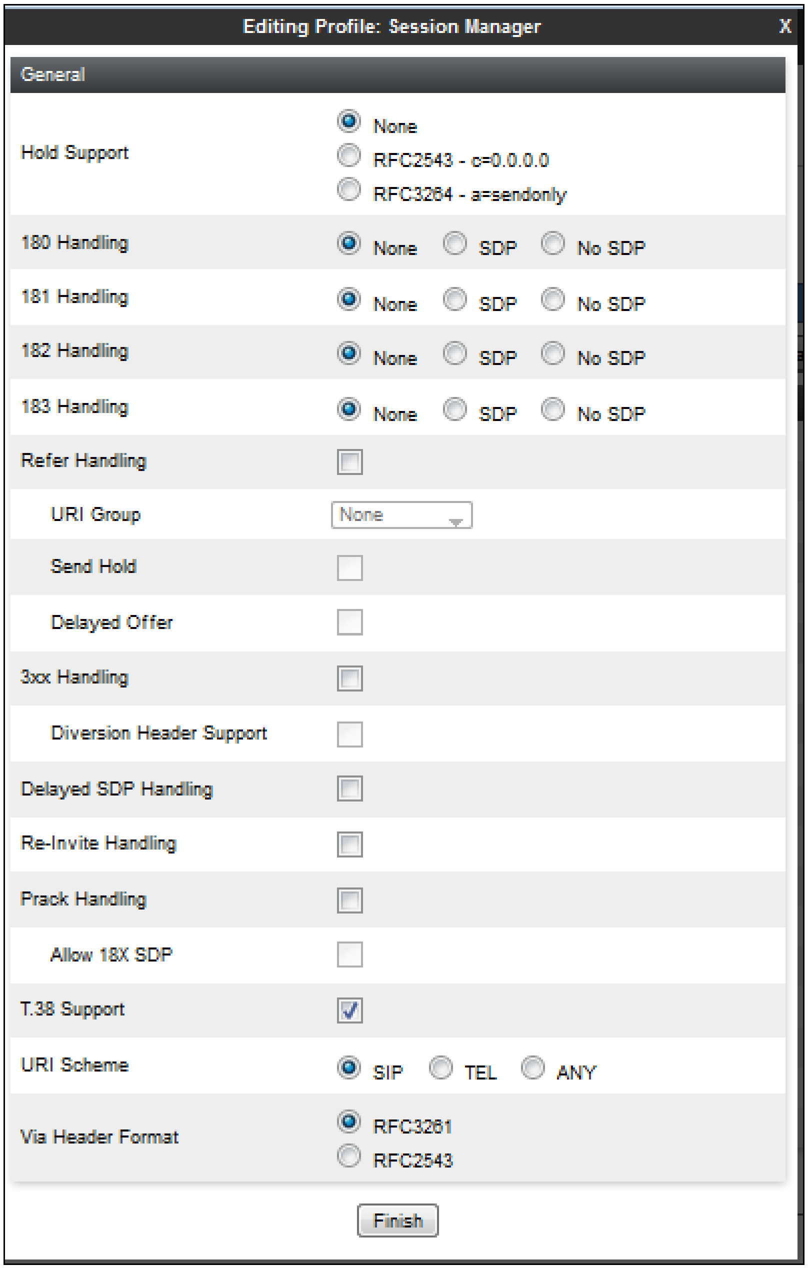
3. Enter profile name: (e.g., **Session Manager**), and click **Finish**.



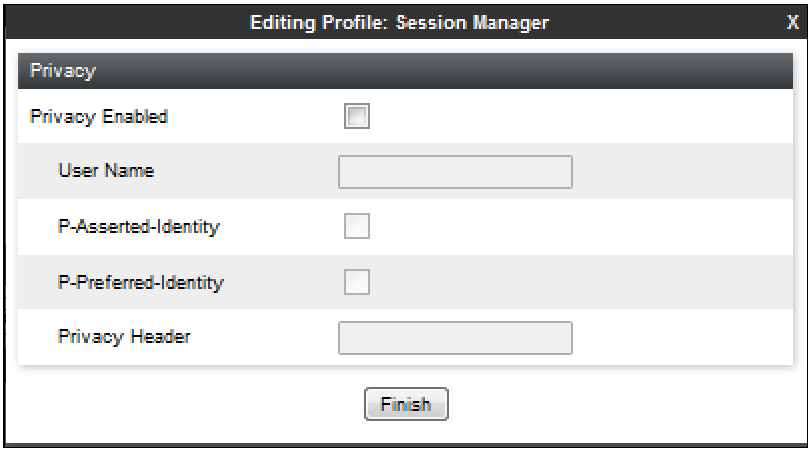
4. The new Session Manager profile will be listed. Select it, scroll to the bottom of the Profile screen, and click on **Edit**.



5. The General screen will open.
   - Check **T38 Support**.
   - All other options can be left with default values, and click **Finish**.



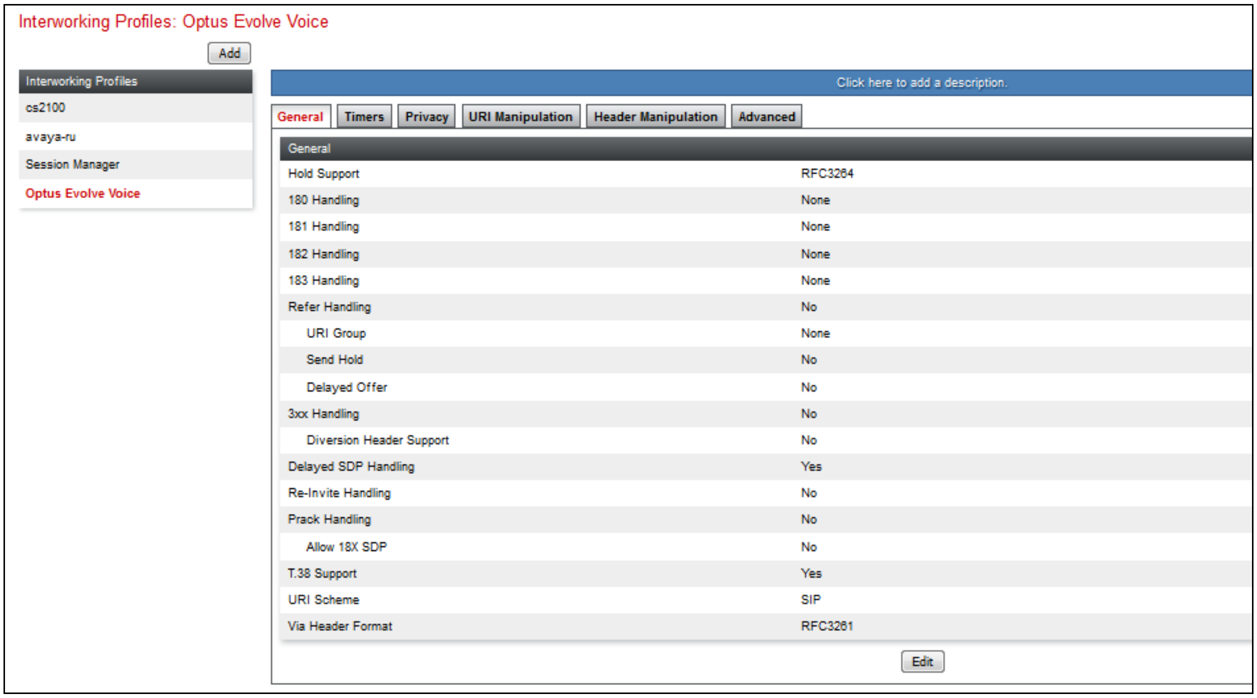
6. On the Privacy window, select **Finish** to accept default values.



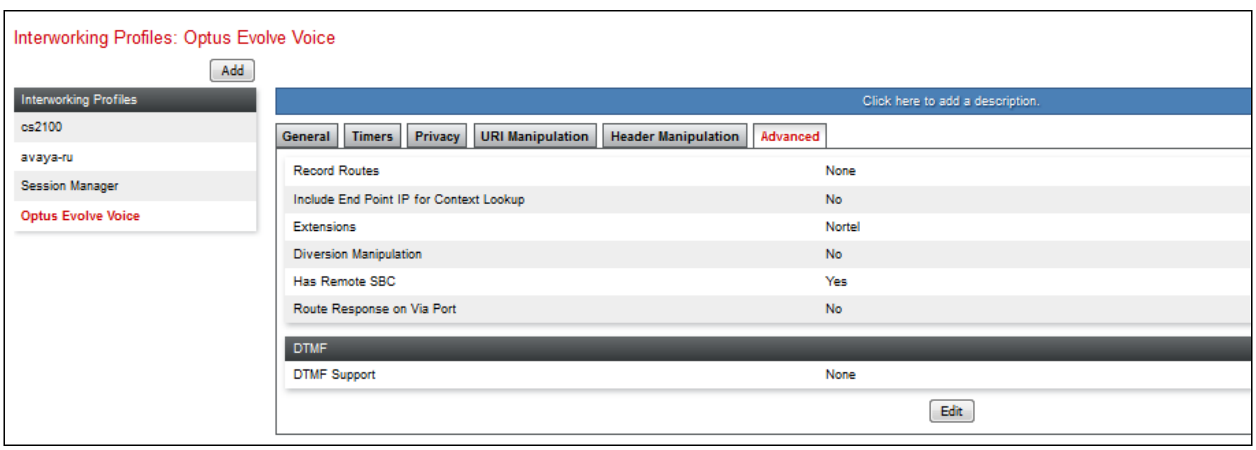
7.2.3 Server Interworking – Optus

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

1. Select cs2100 **Profile** (not shown) and **Clone** with a new name (e.g., **Optus Evolve Voice**) and click **Next** (not shown).
2. The **General** screen will open (not shown):
   - Check **T38 Support**.
   - All other options can be left as default.
   - Click **Next**.
   - The **Privacy/DTMF, SIP Timers/Transport Timers**, and **Advanced** screens will open (not shown), accept default values for all the screens by clicking **Next**, then clicking on **Finish** when completed.


Interworking Profiles: Optus Evolve Voice
Add
Interworking Profiles
cs2100
avaya-ru
Session Manager
Optus Evolve Voice
Click here to add a description.
General
Timers
Privacy
URI Manipulation
Header Manipulation
Advanced
General
Hold Support RFC3264
180 Handling None
181 Handling None
182 Handling None
183 Handling None
Refer Handling No
URI Group None
Send Hold No
Delayed Offer No
3xx Handling No
Diversion Header Support No
Delayed SDP Handling Yes
Re-Invite Handling No
Prack Handling No
Allow 18X SDP No
T.38 Support Yes
URI Scheme SIP
Via Header Format RFC3261
Edit

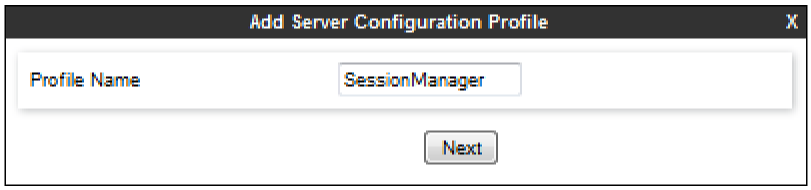

Interworking Profiles: Optus Evolve Voice
Add
Interworking Profiles
cs2100
avaya-ru
Session Manager
Optus Evolve Voice
General
Timers
Privacy
URI Manipulation
Header Manipulation
Advanced
Privacy
Privacy Enabled No
User Name
P-Asserted-Identity No
P-Preferred-Identity No
Privacy Header
Edit


Interworking Profiles: Optus Evolve Voice
Add
Interworking Profiles
cs2100
avaya-ru
Session Manager
Optus Evolve Voice
Click here to add a description.
General
Timers
Privacy
URI Manipulation
Header Manipulation
Advanced
Record Routes None
Include End Point IP for Context Lookup No
Extensions Nortel
Diversion Manipulation No
Has Remote SBC Yes
Route Response on Via Port No
DTMF
DTMF Support None
Edit

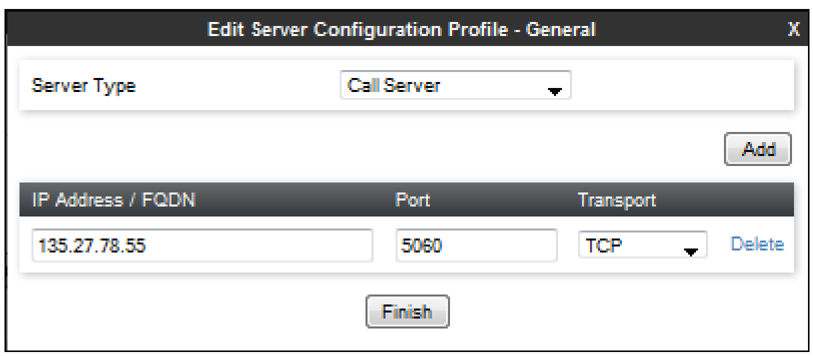
7.2.4 Server Configuration – Session Manager

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

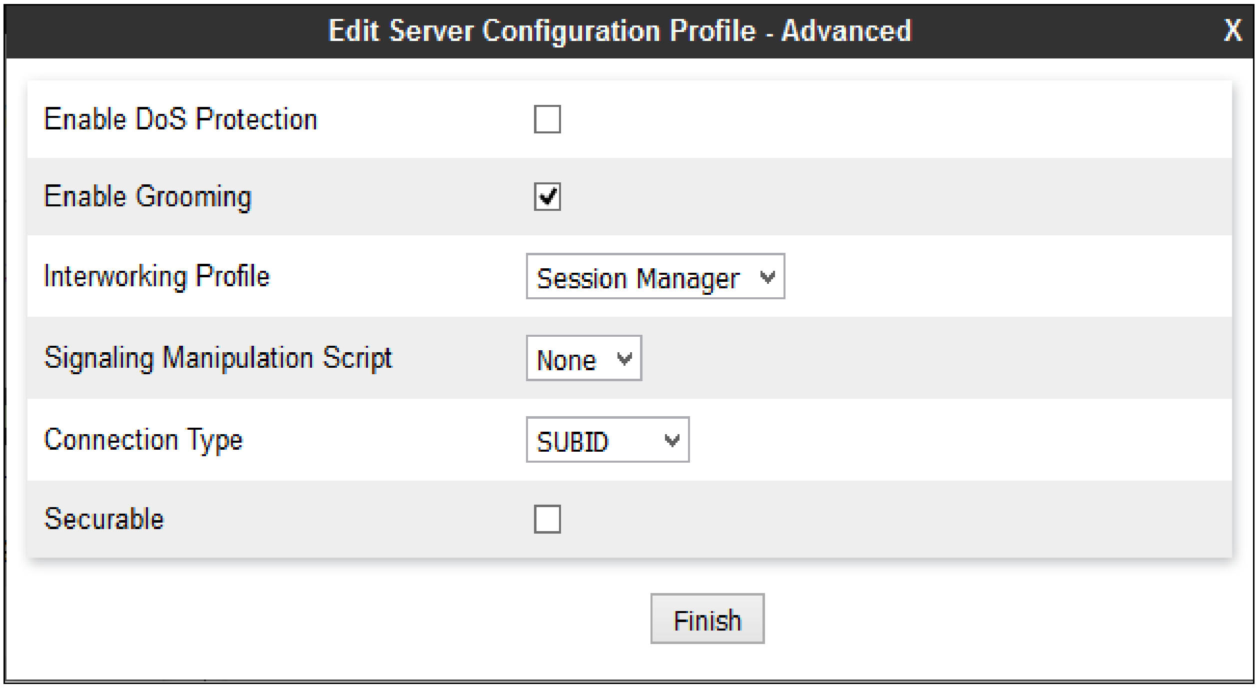
1. Select **Global Profiles → Server Configuration** from the left-hand menu.
2. Select **Add Profile** and the **Profile Name** window will open. Enter a Profile Name (e.g., **Session Manager**) and click **Next**.



3. The **Add Server Configuration Profile** window will open.
   - Select **Server Type**: **Call Server**.
   - **IP Address / FQDN**: **135.27.78.55** (Session Manager signaling IP Address)
   - **Transport**: Select **TCP**.
   - **Port**: **5060**.
   - Select **Next**.



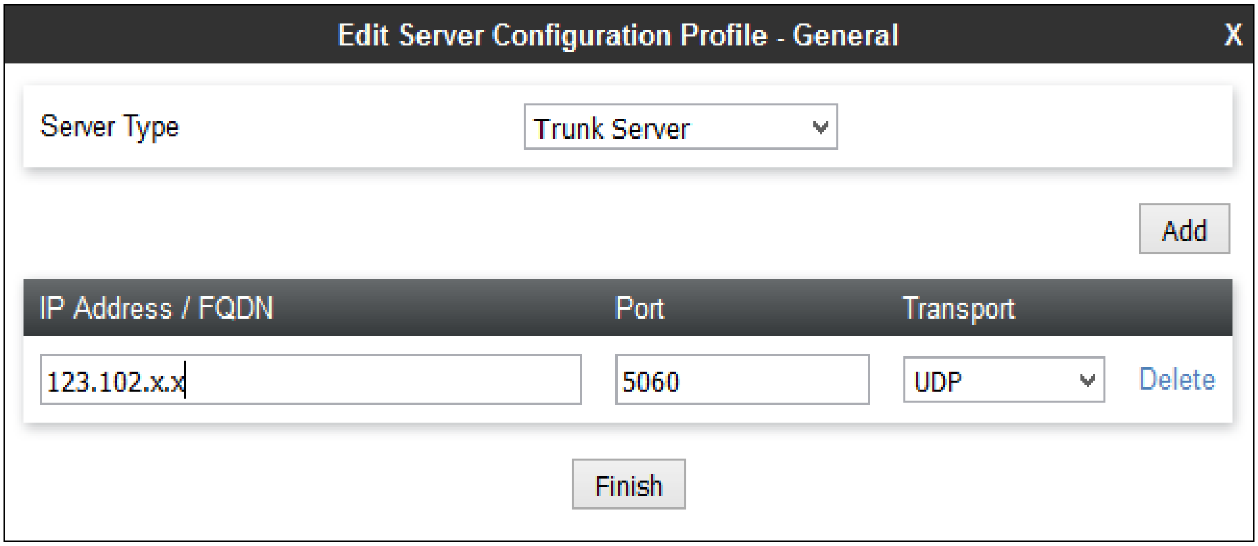
4. The **Authentication** and **Heartbeat** windows will open (not shown).
   - Select **Next** to accept default values.
5. The **Advanced** window will open.
   - For **Interworking Profile**, select the profile created for Session Manager in **Section 7.2.2**.
   - In the **Signaling Manipulation Script** field select **none.**
   - Select **Finish**.

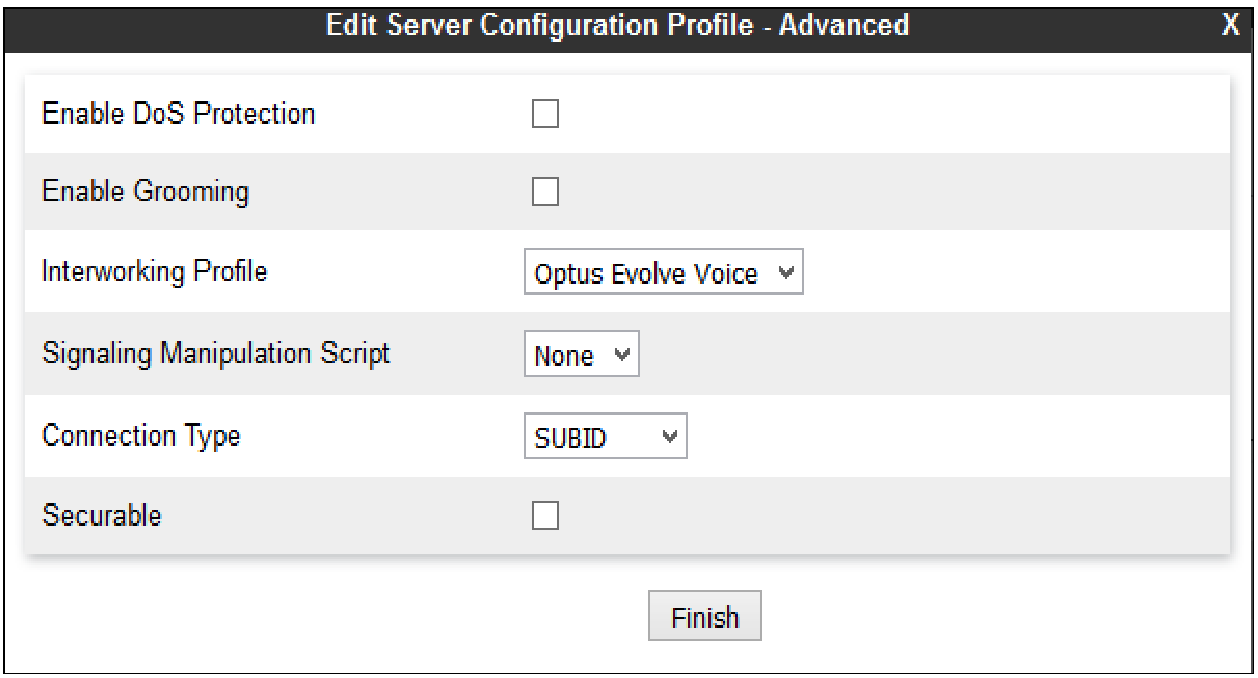


7.2.5 Server Configuration – Optus

Repeat the steps in **Section 7.2.4**, with the following changes, to create a Server Configuration for the Avaya SBCE connection to Optus.

1. Select **Add Profile** and enter a Profile Name (e.g., **Optus Evolve Voice**) and select **Next**.
2. On the **General** window (not shown), enter the following.
   - Select Server Type**: Trunk Server**.
   - **IP Address / FQDN: 123.102.x.x (**because of security reason, the real IP address is not shown here)
   - **Transport**: Select **UDP**.
   - **Port: 5060**
   - Select **Next**.
3. On the **Advanced** window, enter the following.
   - For **Interworking Profile**, select the profile created for Optus in **Section 7.2.3**.
   - Select **Finish**.



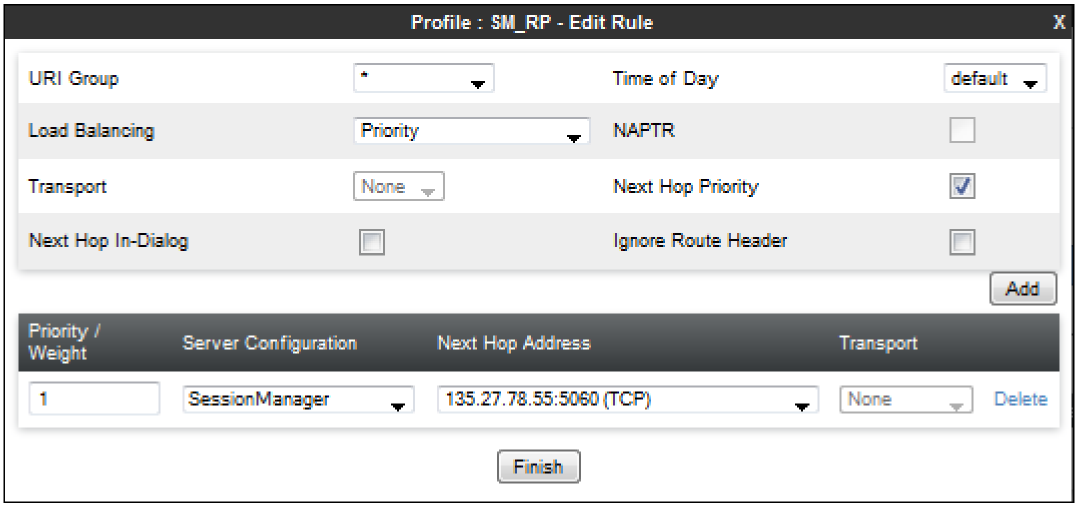


7.2.6 Routing – To Session Manager

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

1. Select **Global 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**

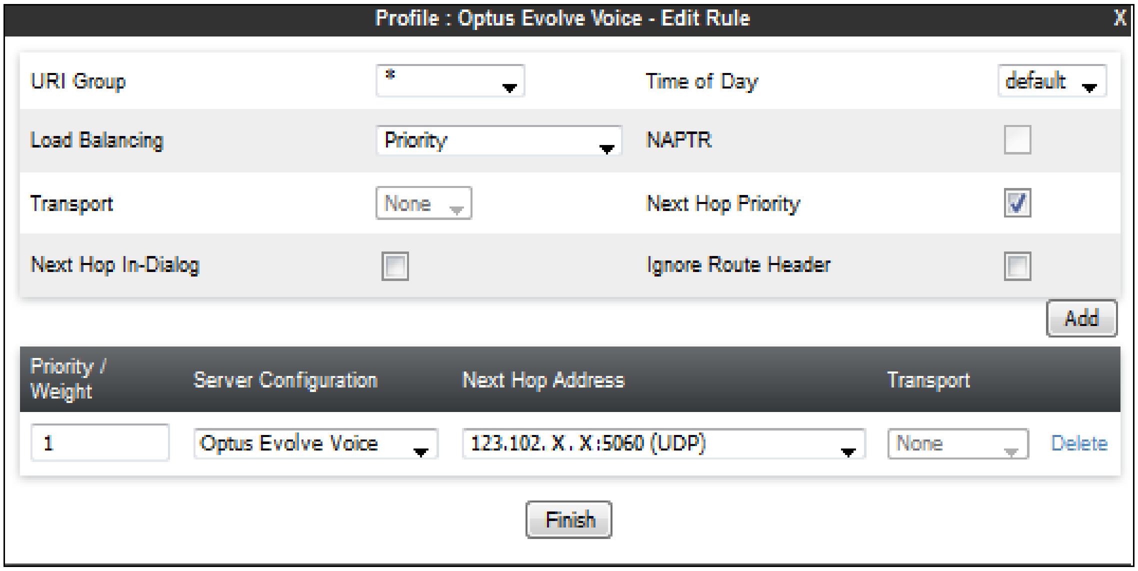
- **Server Configuration** = **SessionManager**.
- **Next Hop Address** = Verify that the **135.27.78.55:5060 (TCP)** entry from the drop down menu is selected (Session Manager IP address). Also note that the **Transport** field is grayed out.
- Click on **Finish.**



7.2.7 Routing – To Optus

Repeat the steps in **Section 7.2.6**, with the following changes, to add a Routing Profile for the Avaya SBCE connection to Optus.

1. On the **Global Profiles → Routing** window (not shown), enter a **Profile Name**: (e.g., **Optus Evolve Voice)**.
2. On the **Next-Hop Address** window (not shown), populate the following fields:
   - **Priority/Weight** = **1**
   - **Server Configuration** = **Optus Evolve Voice**.
   - **Next Hop Address:** Verify that the **123.102.x.x:5060** entry from the drop down menu is selected (Optus Border Element IP address).
   - Use default values for the rest of the parameters.
3. Click **Finish**.



7.2.8 Topology Hiding – Avaya

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

1. Select **Global 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 **To** header is added.
4. Populate the fields as shown below, and click **Finish**. Note that **sipinterop.net** is the domain used.

7.2.9 Topology Hiding – Optus

Repeat the steps in **Section 7.2.8,** with the following changes, to create a Topology Hiding Profile for the Avaya SBCE connection to Optus.

1. Enter a **Profile Name**: (e.g., **Optus Evolve Voice**).
2. Use the default values for all fields and click **Finish**.

7.2.10 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.

7.2.11 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 Optus lab setup, the Avaya SBCE was licensed for 25 Voice sessions, and the default rule was amended accordingly. Other Application Rules could be utilized on an as needed basis.

7.2.12 Border Rules

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

7.2.13 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.

7.2.14 Signaling Rules

The default Signaling Rule was utilized and customized accordingly.

Next two images are detailed Request and Response Headers that are proprietary to Avaya, and originate from Communication Manager or Session Manager. The inclusion of these headers increase the Packet size sent to the Optus Network. In an effort to reduce the packet size, these proprietary headers are removed before being sent to the Optus network.

7.2.15 Endpoint Policy Groups

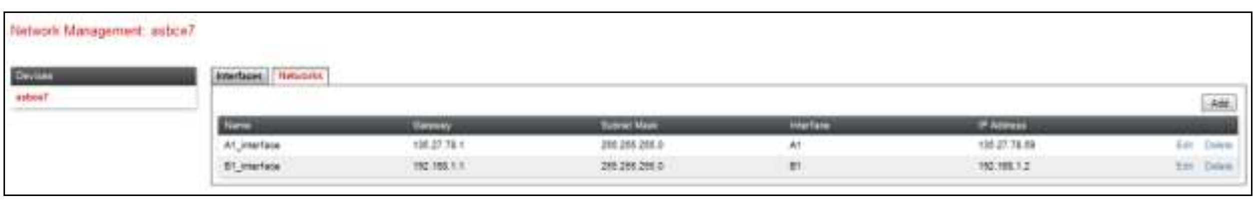
In the solution as tested, the **default-low** rule was utilized. This rule incorporated the media and Signaling Rules specified above, as well as other policies.

7.3 Device Specific Settings

The **Device Specific Settings** 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.

7.3.1 Network Management

1. Select **Device Specific Settings → 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.



7.3.2 Media Interfaces

1. Select **Device Specific Settings** from the menu on the left-hand side (not shown).
2. Select **Media Interface.**

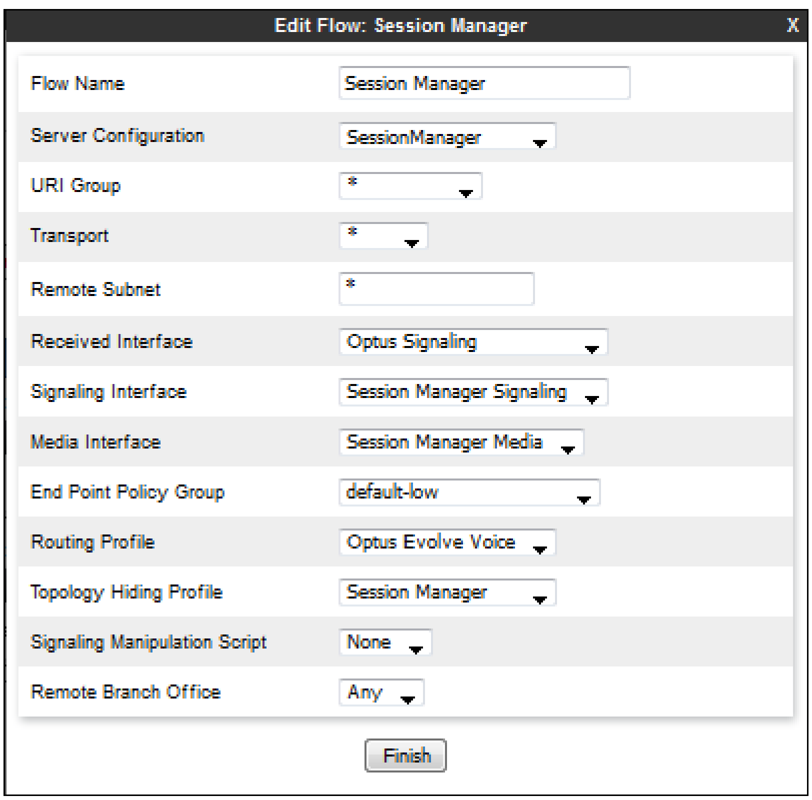
3. Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:
    - **Name**: **Session Manager Media**.
    - **IP Address**: **135.27.78.59** (Avaya SBCE A1 address).
    - **Port Range**: **35000-40000**.
4. Click **Finish** (not shown).
5. Select **Add** (not shown). The **Add Media Interface** window will open. Enter the following:
    - **Name**: **Optus Media**.
    - **IP Address**: **192.168.1.2** (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.

7.3.3 Signaling Interface

1. Select **Device Specific Settings** from the menu on the left-hand side (not shown).
2. Select **Signaling Interface**.
3. Select **Add** (not shown) and enter the following:
    - **Name**: **Session Manager Signaling**.
    - **IP Address**: **135.27.78.59** (Avaya SBCE A1 address).
    - **TCP Port**: **5060**.
4. Click **Finish** (not shown).
5. Select **Add** again, and enter the following:
    - **Name**: **Optus Signaling**.
    - **IP Address**: **192.168.1.2** (Avaya SBCE B1 address).
    - **UDP Port**: **5060**.
6. Click **Finish** (not shown). Note that changes to these values require an application restart.

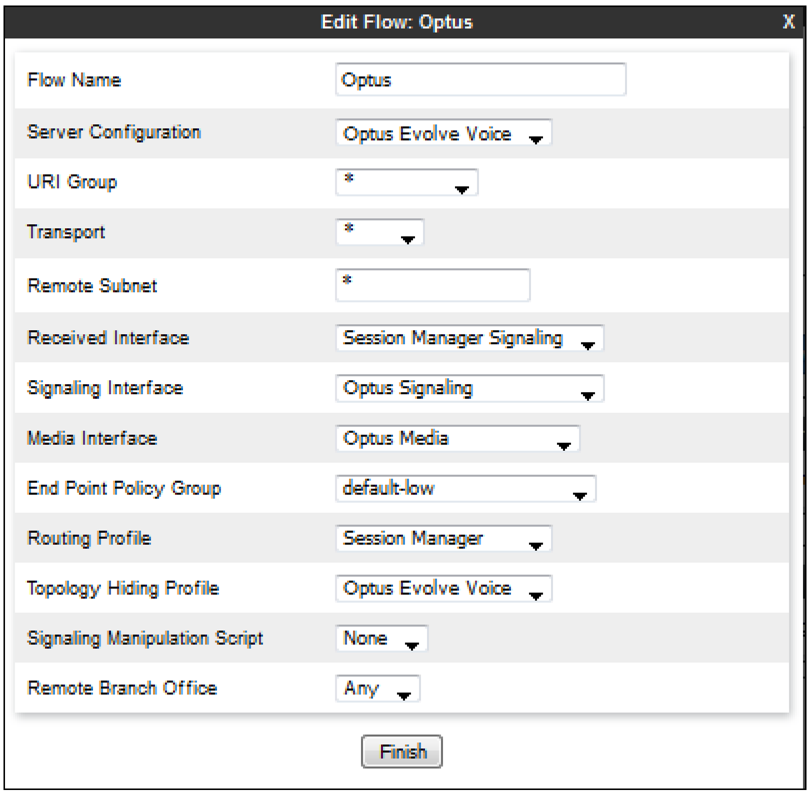
7.3.4 Endpoint Flows – For Session Manager

1. Select **Device Specific Settings → 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**.
   - **Server Configuration**: **Session Manager**.
   - **URI Group**: *****
   - **Transport**: *****
   - **Remote Subnet**: *****
   - **Received Interface**: **Optus Signaling.**
   - **Signaling Interface**: **Session Manager Signaling**.
   - **Media Interface**: **Session Manager Media**.
   - **End Point Policy Group**: **default-low**.
   - **Routing Profile**: **Optus Evolve Voice**.
   - **Topology Hiding Profile**: **Session Manager**.
   - Let other values default.
4. Click **Finish** .



7.3.5 Endpoint Flows – For Optus

1. Repeat step **1** through **4** from **Section 7.3.4**, with the following changes:
   - **Name**: **Optus Evolve Voice**.
   - **Server Configuration**: **Optus Evolve Voice**.
   - **URI Group**: *
   - **Transport**: *
   - **Remote Subnet**: *
   - **Received Interface**: **Session Manager Signaling**.
   - **Signaling Interface**: **Optus Signaling**.
   - **Media Interface**: **Optus Media**
   - **End Point Policy Group**: **default_low**.
   - **Routing Profile**: **Session Manager**.
   - **Topology Hiding Profile**: **Optus Evolve Voice**.



8. Verification Steps

The following steps may be used to verify the configuration.

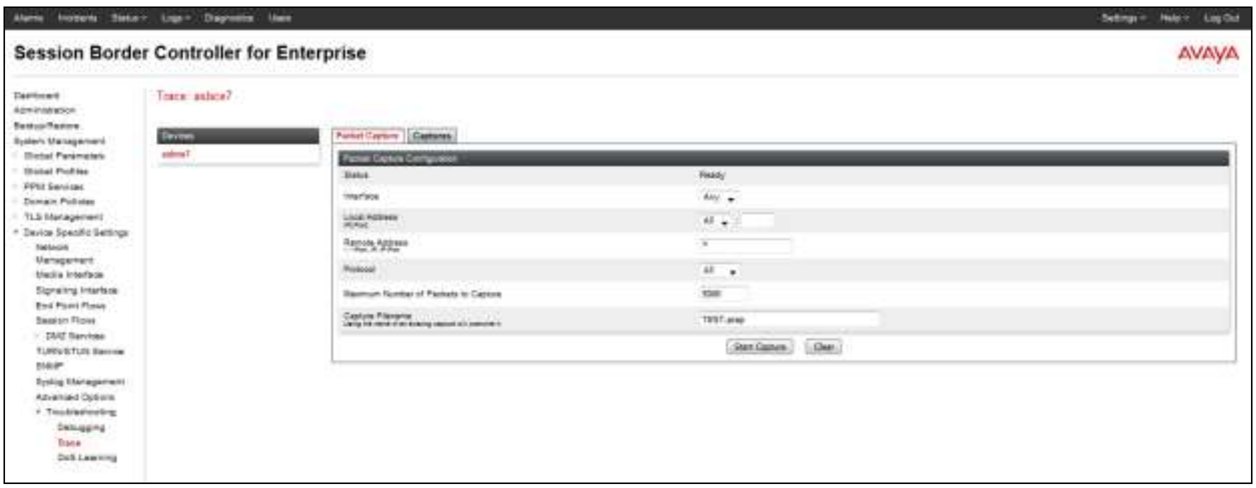
8.1 Avaya Session Border Controller for Enterprise

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.

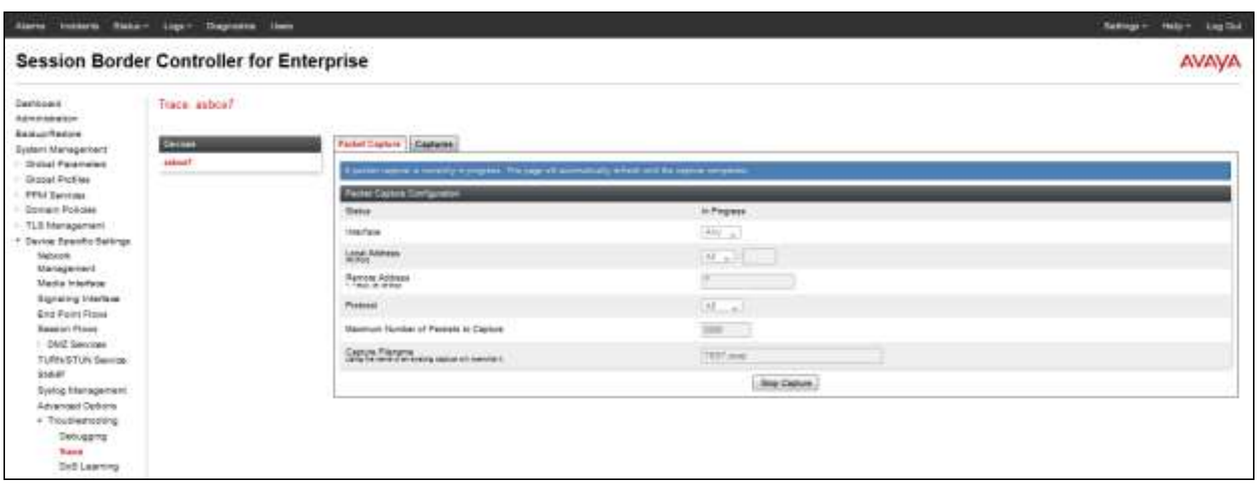
Protocol Traces

The Avaya SBCE can take internal traces of specified interfaces.

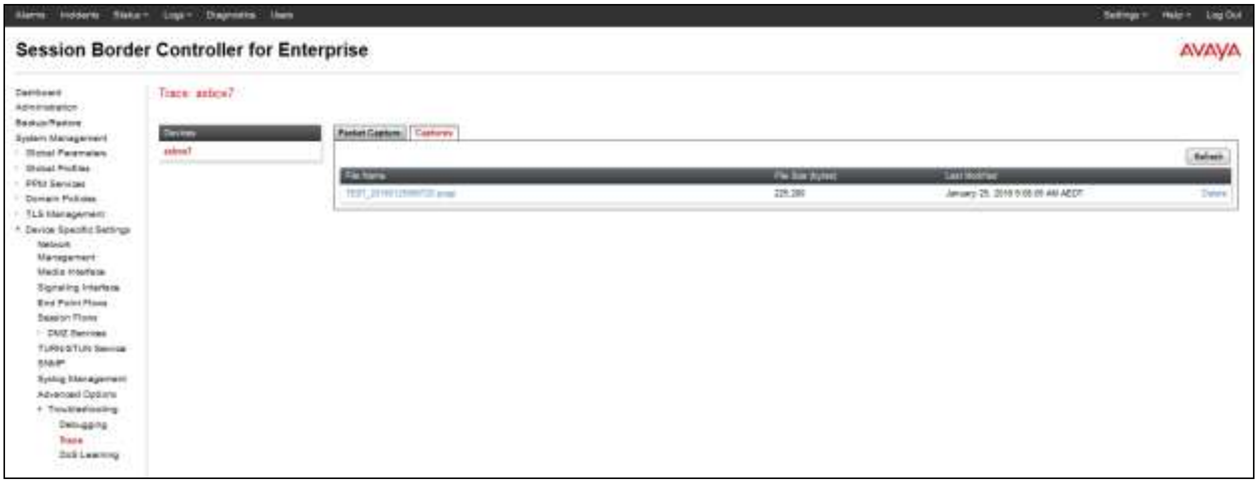
1. Navigate to **Device Specific Settings → Troubleshooting → Trace**.
2. Select the **Packet Capture** tab and select the following:
   - Select the desired **Interface** from the drop down menu (e.g., **All**).
   - Specify the **Maximum Number of Packets to Capture** (e.g., **5000**).
   - Specify 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:

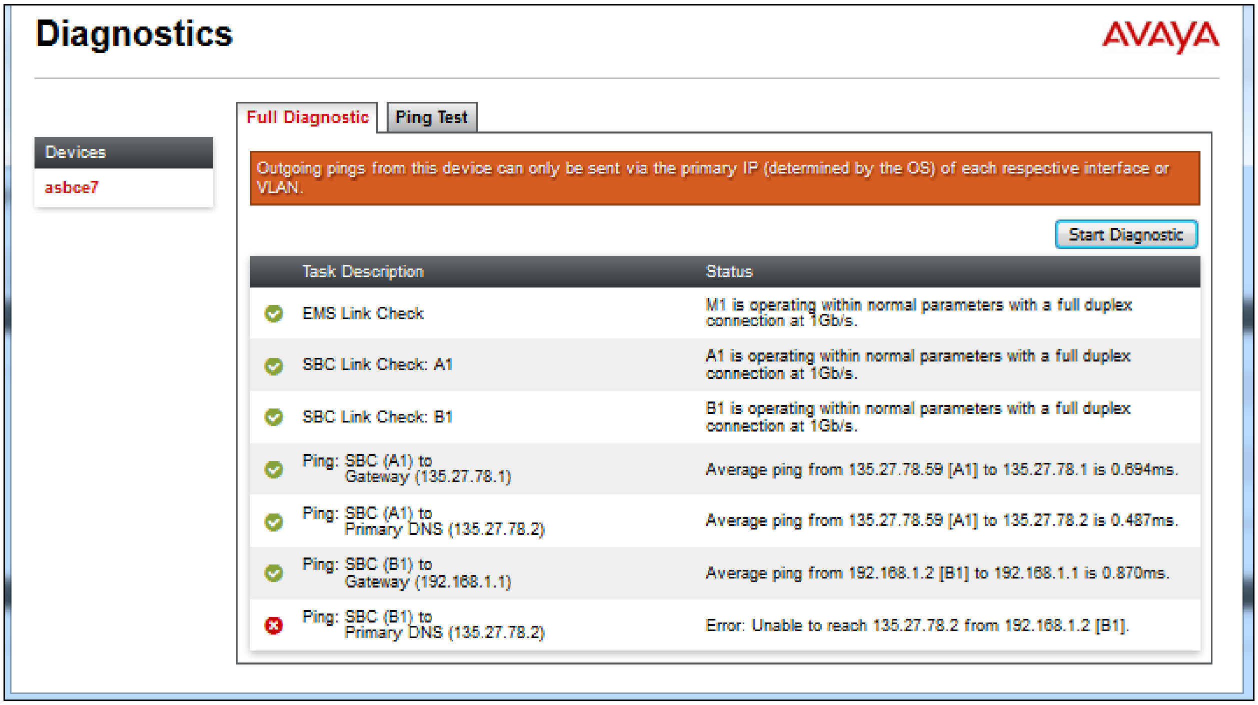


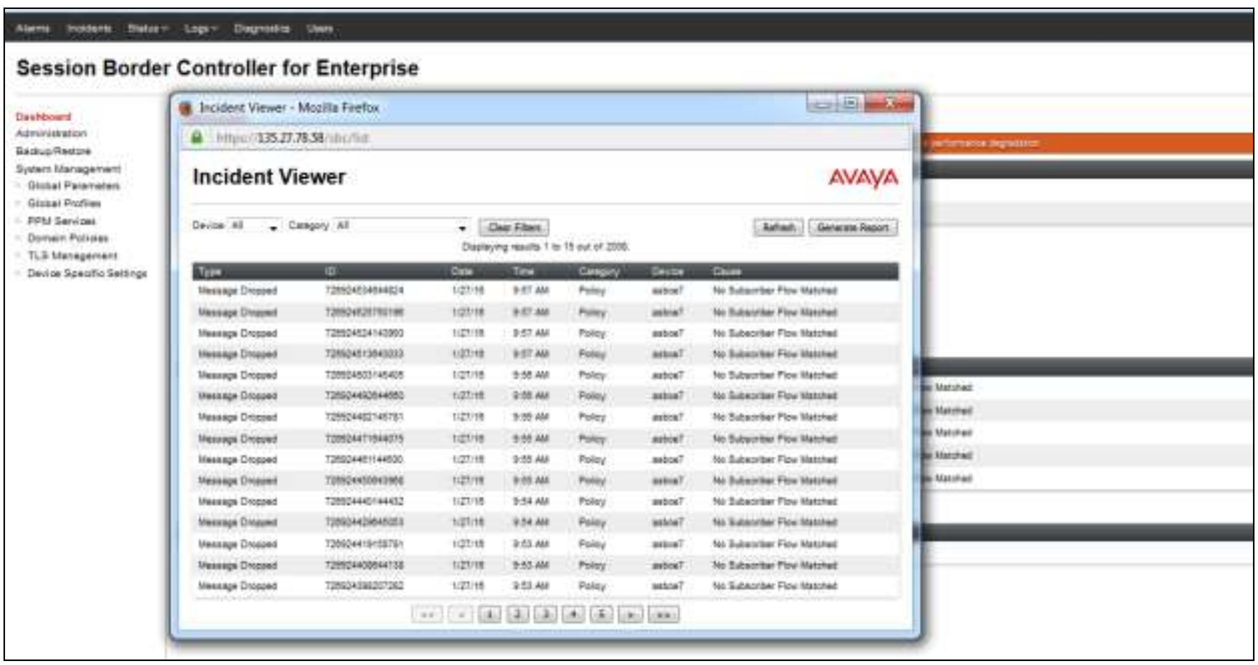
3. Run the test.
4. When the test is completed, select the **Stop Capture** button shown above.
5. Click on the **Captures** tab and the packet capture is listed as a *.pcap* file with the date and time added to filename specified in **Step 2**.
6. Click on the **File Name** link to download the file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

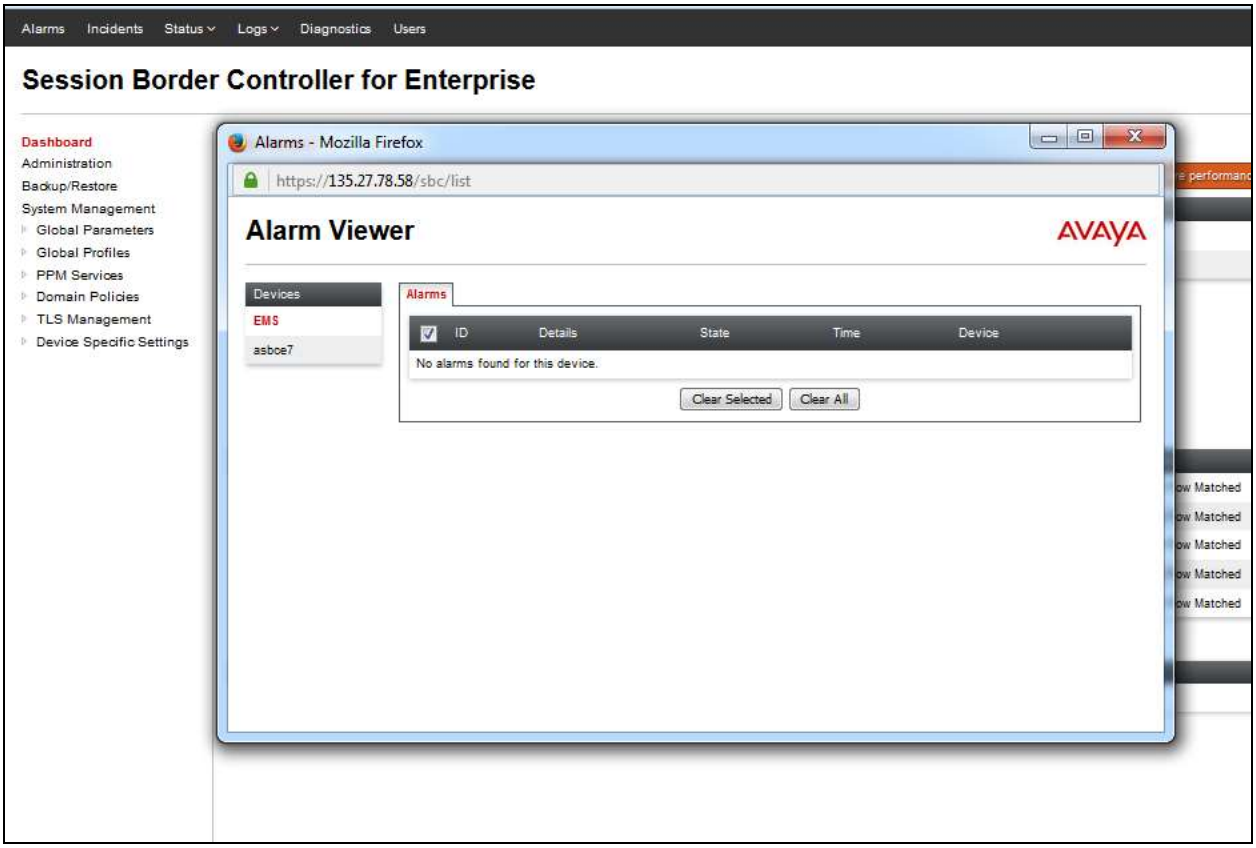


The following section details various methods and procedures to help diagnose call failure or service interruptions. As detailed in previous sections, the demarcation point between the Optus Evolve SIP Trunk Service and the customer SIP PABX is the customer SBC.
On either side of the SBC, various diagnostic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

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


Diagnostics
Full Diagnostic
Ping Test
Devices
asbce7
Outgoing pings from this device can only be sent via the primary IP (determined by the OS) of each respective interface or VLAN.
Start Diagnostic
Task Description
Status
EMS Link Check
M1 is operating within normal parameters with a full duplex connection at 1Gb/s.
SBC Link Check: A1
A1 is operating within normal parameters with a full duplex connection at 1Gb/s.
SBC Link Check: B1
B1 is operating within normal parameters with a full duplex connection at 1Gb/s.
Ping: SBC (A1) to Gateway (135.27.78.1)
Average ping from 135.27.78.59 [A1] to 135.27.78.1 is 0.694ms.
Ping: SBC (A1) to Primary DNS (135.27.78.2)
Average ping from 135.27.78.59 [A1] to 135.27.78.2 is 0.487ms.
Ping: SBC (B1) to Gateway (192.168.1.1)
Average ping from 192.168.1.2 [B1] to 192.168.1.1 is 0.870ms.
Ping: SBC (B1) to Primary DNS (135.27.78.2)
Error: Unable to reach 135.27.78.2 from 192.168.1.2 [B1].


Session Border Controller for Enterprise
Incident Viewer - Mozilla Firefox
Incident Viewer
Dashboard
Administration
Backup/Restore
System Management
Global Parameters
Global Profiles
PPM Services
Domain Policies
TLS Management
Device Specific Settings
Type
ID
Date
Time
Category
Device
Cause



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

```
status signaling-group 3
                        STATUS SIGNALING GROUP

        Group ID: 3
      Group Type: sip

     Group State: in-service




Command:
```

```
status trunk 3                                                  Page   1

                           TRUNK GROUP STATUS

Member    Port      Service State       Mtce Connected Ports
                                        Busy

0003/001 T00006   in-service/idle     no
0003/002 T00007   in-service/idle     no
0003/003 T00008   in-service/idle     no
0003/004 T00009   in-service/idle     no
0003/005 T00010   in-service/idle     no
0003/006 T00001   in-service/idle     no
0003/007 T00002   in-service/idle     no
0003/008 T00003   in-service/idle     no
0003/009 T00004   in-service/idle     no
0003/010 T00005   in-service/idle     no
0003/011 T00016   in-service/idle     no
0003/012 T00017   in-service/idle     no
0003/013 T00018   in-service/idle     no
0003/014 T00019   in-service/idle     no

                press CANCEL to quit --  press NEXT PAGE to continue
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

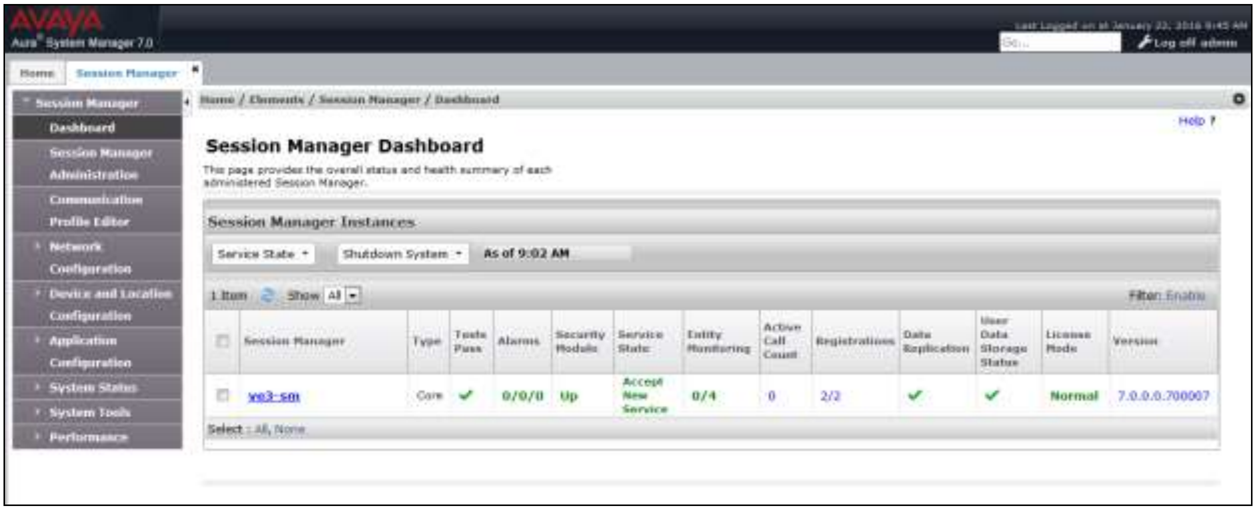
```
status signaling-group 4
                             STATUS SIGNALING GROUP

        Group ID: 4
      Group Type: sip

     Group State: in-service





Command:
```

```
status trunk 4

                             TRUNK GROUP STATUS

Member    Port     Service State      Mtce Connected Ports
                                      Busy

0004/001 T00011   in-service/idle    no
0004/002 T00012   in-service/idle    no
0004/003 T00013   in-service/idle    no
0004/004 T00014   in-service/idle    no
0004/005 T00015   in-service/idle    no



Command successfully completed
Command:
```

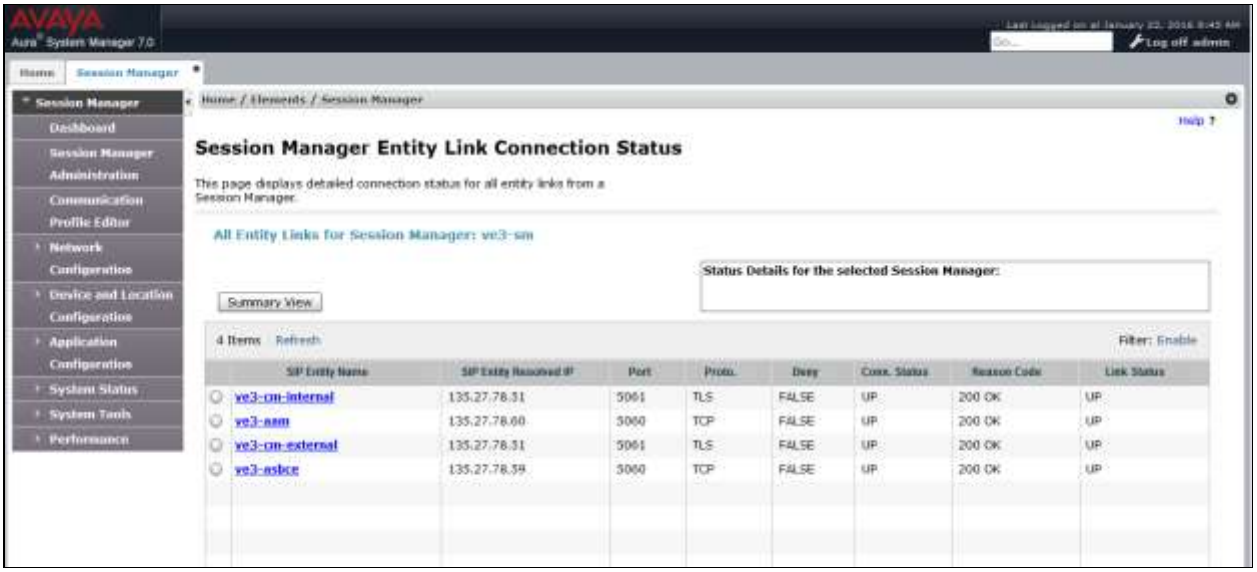
8.3 Avaya Aura® Session Manager Status

The Session Manager configuration may be verified via System Manager.

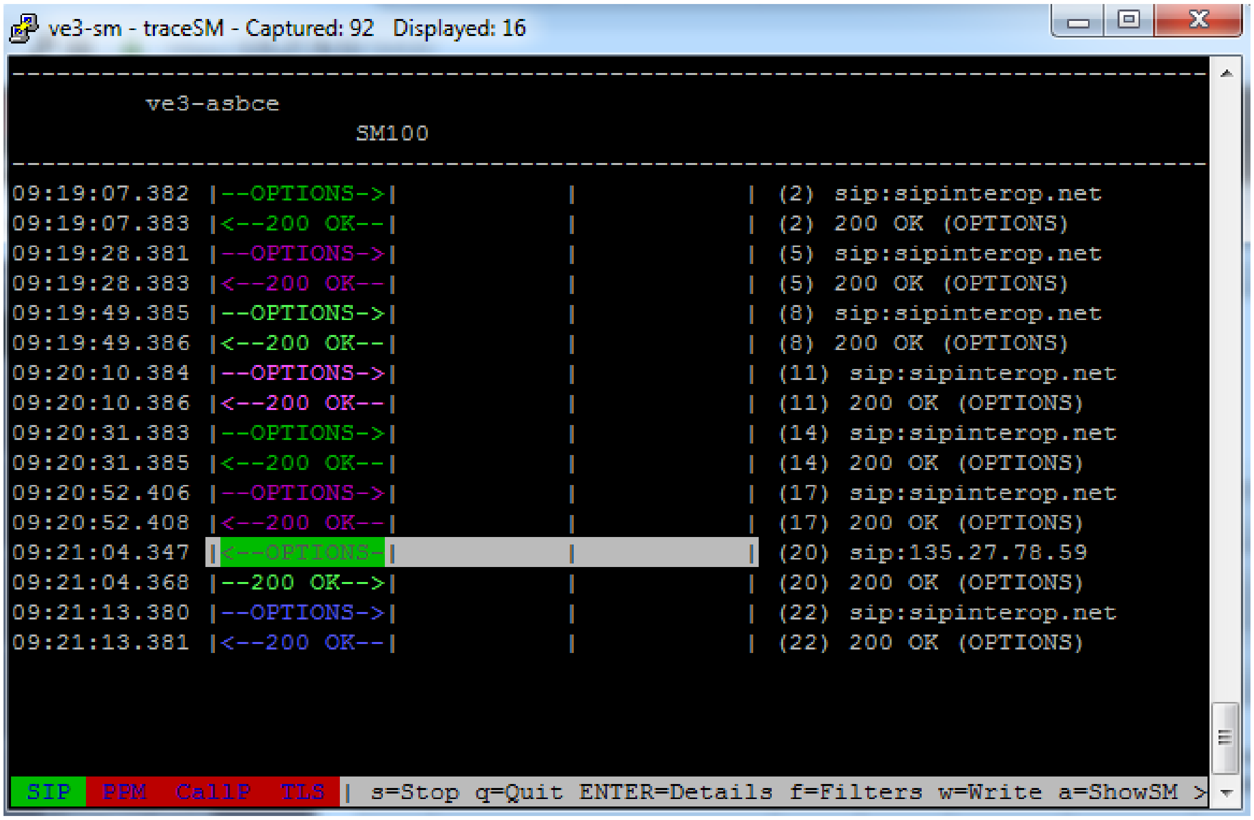
1. Using the procedures described in **Section 6**, access the System Manager GUI. From the **Home** screen, under the **Elements** heading, select **Session Manager**.



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 **4** Entities defined.
3. Clicking on the **0/4** 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 7.0, Avaya Aura® Session Manager 7.0, and Avaya Session Border Control for Enterprise 7.0 can be configured to interoperate successfully with Optus Evolve Voice SIP Trunking service. This solution allows enterprise users access to the PSTN using the Optus Evolve Voice SIP Trunking service connection.

10. Additional References

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

[1] *What's New in Avaya Aura Release 7.0,* Release 7.0, 03-601818, Issue 1, August 2015.
[2] *Deploying Avaya Aura® System Manager*, Release 7.0, Issue 1, October 2015.
[3] *Administering Avaya Aura® System Manager for Release 7.0*, Issue 1, August 2015.
[4] *Administering Avaya Aura® Session Manager*, Release 7.0, Issue 1, August 2015.
[5] *Deploying Avaya Aura Communication Manager in Virtualized Environment*, Release7, Issue 1, August 2015.
[6] *Avaya Session Border Controller for Enterprise Overview and Specification*, Release 7.0, Issue
1, August 2015.
[7] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, Issue 1, August 2015.
[8] *Deploying Avaya Session Border Controller in Virtualized Environment*, Release 7.0, Issue 1, August 2015.
[9] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0, Issue 1, August 2015.
[10] *Deploying and Updating Avaya Aura Media Server Appliance, Release 7.7*, Issue 1, August 2015.
[11] *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager, Release 7.7,* August 2015.
[12] *Deploying Avaya Aura® Messaging for Single Server Systems 6.3.3, Release 6.3.3*, August 2015.
[13] *Administering Avaya Aura® Messaging 6.3.3, Release 6.3.3*, August 2015.
[14] *9600 Series IP Deskphones Overview and Specification, Release 7.0*, Issue 1, August 2015.
[15] *Installing and Maintaining Avaya 9601/9608/9611G/9621G/9641G/9641GS IP Deskphones SIP*, Release 7.0, Issue 1, August 2015.
[16] *Administering Avaya 9601/9608/9611G/9621G/9641G/9641GS IP Deskphones SIP*, *Release 7.0*, Issue 2, August 2015.
[17] *Administering Avaya one-X® Communicator*, Release 6.2, April 2015.
[18] *Configuring Remote Workers with Avaya Session Border Controller for Enterprise Rel. 6.2, Avaya Aura® Communication Manager Rel. 6.3 and Avaya Aura® Session Managers Rel. 6.3.* Issue 1.
[19] *RFC 3261 SIP: Session Initiation Protocol*, http://www.ietf.org/
[20] *RFC 3515, The Session Initiation Protocol (SIP) Refer Method*, http://www.ietf.org/
[21] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, http://www.ietf.org/

Product documentation for Optus Evolve Voice SIP Trunking Solution is available from Optus.

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.