Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the TELUS SIP Trunking Service (Release 2 Platform – No Registration) with Avaya Aura® Communication Manager Evolution Server 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between the TELUS SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager 6.3, Avaya Aura® Communication Manager Evolution Server 6.3, Avaya Session Border Controller for Enterprise and various Avaya endpoints. TELUS is a member of the Avaya DevConnect Service Provider program.

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

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.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between the TELUS SIP Trunking Service (R2 Platform – No Registration) and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager 6.3, Avaya Aura® Communication Manager Evolution Server 6.3, Avaya Session Border Controller for Enterprise and various Avaya endpoints. In addition, Avaya Aura® System Manager 6.3 is used to configure Avaya Aura® Session Manager.

The TELUS SIP Trunking Service can be deployed using private MPLS connections from the TELUS network to the enterprise or can be deployed across the Internet. Deployment across the Internet requires registration by the enterprise while the MPLS connections do not. These Application Notes cover the MPLS deployment configuration.

Customers using this Avaya SIP-enabled enterprise solution with the TELUS SIP Trunking Service are able to place and receive PSTN calls via a broadband WAN connection with SIP. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the TELUS SIP Trunking Service via a broadband connection and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of Communication Manager, Session Manager and Avaya Session Border Controller for Enterprise (Avaya SBCE).

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 for 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

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test.

- Sending and receiving SIP OPTIONS queries to the service provider
- Inbound and outbound PSTN calls (via the SIP trunk) to/from SIP and H.323 telephones at the enterprise
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator (soft client) using multiple protocols (H.323 and SIP) and multiple modes (Local Computer and Other Phone mode)
- Inbound and outbound PSTN calls to/from Avaya Flare® Experience for Windows
- Inbound and outbound PSTN calls to/from TELUS Business VoIP endpoints (SIP)
- Inbound and outbound PSTN calls to/from TELUS Mobility endpoints

- Various call types including: local (10 digit), long distance (1 + 10 digits), international, outbound toll-free, operator, operator-assisted calls (0 + 10 digits) and local directory assistance (411)
- Codecs G.711MU and G.729A
- DTMF transmission using RFC 2833
- Caller ID presentation and Caller ID restriction
- Response to incomplete call attempts and trunk errors
- Voicemail navigation for inbound and outbound calls
- Voicemail Message Waiting Indicator (MWI)
- User features such as hold and resume, internal call forwarding, transfer, and conference
- Off-net call forwarding and mobility (Extension to cellular – EC500)
- T.38 Fax
- Network Call Redirection using REFER and a 302 response
- Initial IP-IP Direct Media

Emergency 911 calls, and inbound toll-free calls are supported but were not tested as part of the compliance test. Also, Remote Worker functionality could not be tested due to the test configuration used. A Remote Worker located on the Internet could not register to the enterprise Avaya SBCE since the Avaya SBCE did not have a public IP address but was connected via a VPN to the TELUS SIP Trunking Service.

The following item was not supported:

- SIP User to User Information (UUI)

2.2. Test Results

Interoperability testing of TELUS SIP Trunking was completed with successful results for all test cases with the exception of the observations and/or limitations described below.

- **OPTIONS to TELUS (Max-Forwards Value)**: TELUS requires that SIP OPTIONS messages sent from the enterprise contain a Max-Forwards value of zero. These messages originate from Session Manager with a non-zero Max-Forwards value when link monitoring is enabled on Session Manager. Thus, the Avaya SBCE was used to modify this value when the Avaya SBCE sent the OPTIONS message to the network (**Section 7.6.1**).
- **OPTIONS from TELUS (Request-URI)**: TELUS sends OPTIONS messages whose user part of the Request URI is not routable by the Session Manager which results in a 404 User Not Found response to TELUS. For interoperability, the Avaya SBCE was configured to return a 200 OK response to all OPTIONS messages instead of sending the messages to the Session Manager (**Section 7.10.2**).
- **Session Refresh Time**: The Communication Manager SIP trunk parameter **Preferred Minimum Session Refresh Interval (sec)** on both the internal SIP trunk and service provider SIP trunk should be increased from the default of 300 seconds. Otherwise, outbound INVITEs would receive the response "422 Session Interval Too Small" from TELUS. The compliance test used a value of 900 seconds on **Page 2** of the Communication Manager trunk group form (**Section 5.8**).

- **Use of SA8965 is no longer a requirement**: In previous compliance tests, TELUS required re-INVITEs to contain Session Description Protocol (SDP) information. Thus, the Communication Manager special application SA8965 was enabled (**Section 5.2**). The sending of re-INVITEs with SDP is no longer a requirement. Thus, SA8965 is no longer needed and trunk group parameter **Shuffling with SDP** must be set to **no** (**Section 5.8**).
- **Remove P-Asserted-Identity (PAI) header in 200 OK from TELUS**: In response to an outbound INVITE from the enterprise, TELUS always sends a 200 OK response with "anonymous" as the user part of the PAI header. This lead to "anonymous" being displayed as the connected party on the enterprise phone. As a workaround, the Avaya SBCE was configured to remove the PAI header in the 200 OK (**Section 7.10.2**). In this case, the Session Manager will generate a new PAI header to send to Communication Manager using the contents of the From header. The contents of the new PAI header are used to provide the connected party to the phone.
- **Call Forwarding and EC500**: For inbound PSTN calls that are forwarded back to the PSTN or ring to an EC500 (enterprise mobility) PSTN endpoint, TELUS requires the originating calling number be present in the P-Asserted-Identity (PAI) header. Normally, Communication Manager puts this information in the Diversion header. A SIP header manipulation was created on the Avaya SBCE to modify the P-Asserted-Identity (PAI) header with information contained in the Diversion header received from Session Manager (**Section 7.6.1**). This allowed the call to complete successfully.
- **Calling Party Number (PSTN transfers)**: The calling party number displayed on the PSTN phone is not updated to reflect the true connected party on calls that are transferred to the PSTN. After the call transfer is complete, the calling party number displayed the number of the transferring party and not the actual connected party. The PSTN phone display is ultimately controlled by the terminating PSTN provider, thus this behavior is not necessarily indicative of a limitation of the combined Avaya/TELUS solution. It is listed here simply as an observation.
- **T.38 Fax**
    - **Network Coverage**: Not all media gateways in the TELUS network support T.38 fax. Communication Manager supports fallback to G.711 pass-through fax from T.38 fax if configured on the ip-codec-set form (**See Section 5.5**). This is the recommended setting if all gateways in the service provider network do not support T.38 fax.
    - **Transitioning to T.38 for Outbound Calls**: In general, the answering side of a fax call should send a re-INVITE to transition to T.38. For outbound fax calls to the PSTN, this means the network would typically send the re-INVITE to transition to T.38. However, TELUS never sends a T.38 re-INVITE for outbound calls even if the TELUS gateway supports T.38. The impact is that all outbound fax calls will fallback to G.711 pass-through fax regardless of the TELUS gateway support for T.38. All inbound fax calls will use T.38 if supported on the specific TELUS gateway.
- **Operator-assisted calls routed as direct dialed calls**: Operated-assisted calls (0 + 10 digits) were routed the same as direct dialed long distance calls (1 + 11 digits). This was believed to be a routing problem in the TELUS test lab and would not occur in the production environment.

- **Four-party conference with PSTN users**: A call is established to the enterprise (either inbound or outbound), then the enterprise party conferences in two other PSTN parties for a total of four parties. In the resulting conference, all parties have two-way audio except for the 3rd party which can hear but cannot be heard. This only happens if the call is set-up using the G.729 codec. This is under investigation by TELUS. The current workaround is to set G.711MU as the preferred codec in the Communication Manager ip-codec-set form (**Section 5.5**).
- **Avaya one-X® Communicator SIP in "Other Phone" Mode**: In this mode, an outbound call is issued to the associated "Other Phone" when Avaya one-X® Communicator initiates/receives a call so that Avaya one-X® Communicator controls the call but voice media is to/from the physical "Other Phone". The following items are Avaya issues and not specific to TELUS interoperability. As a workaround to the following issue(s), Avaya one-X® Communicator H.323 may be used for "Other Phone" mode.
    - **No Audio after Hold/Release**: Inbound and outbound calls to/from Avaya one-X® Communicator SIP that were placed on hold and then released from hold resulted in no audio. Attempting to hold/release the call a second time did not clear the problem. The call had to be disconnected. Similar behavior has been observed and documented in Communication Manager Modification Request (MR) defsw130619 and in Avaya one-X® Communicator change request ONEXC-6770.
    - **Enterprise extension in PAI header**: In previous testing, when Communication Manager placed the call to the "Other Phone", the call was rejected with a 500 Server Error from the TELUS network. It was rejected because the initial INVITE from Communication Manager included a PAI header containing the enterprise extension instead of the DID number for that station. This no longer occurs; however, a subsequent re-INVITE still contained the extension in the PAI header which had no impact on the success of the call. As a precaution, a Session Manager Adaptation for the Avaya SBCE SIP Entity was configured to convert the Communication Manager extension number to the associated DID number for populating the PAI header (**Section 6.4**). This Session Manager configuration is only needed for Avaya one-X® Communicator SIP in "Other Phone" mode.
- **Initial IP-IP Direct Media**: Compliance testing was performed both with the Communication Manager feature "Initial IP-IP Direct Media" enabled and disabled (**Section 5.7**). "Initial IP-IP Direct Media" is an optimization of the SIP signaling which attempts to connect the final two ends of the call together initially without first having to connect media to the media gateway and later move the media to the final destination. At least one large TELUS customer has reported that enabling "Initial IP-IP Direct Media" improved voice clipping issues they experienced when calling from the enterprise to TELUS Mobility endpoints. However, this feature had interactions with other Communication Manager functionality outlined below. The following items are Avaya issues and not specific to TELUS interoperability.
    - **Initial IP-IP Direct Media and Avaya one-X® Communicator SIP in "Other Phone" Mode**: No ringback is heard on the second call leg using attended transfer or conference. Specifically, if the Avaya one-X® Communicator SIP endpoint has an inbound/outbound PSTN call up, then performs an attended

transfer or conference to another party, no ringback is heard by the Avaya one-X® Communicator SIP endpoint while ringing the new party.

  - **Initial IP-IP Direct Media and EC500 with confirmed answer**: The Communication Manager EC500 feature allows an inbound call to ring the enterprise phone and simultaneously ring a remote phone (usually a cell phone). Optionally, this feature can be configured for "confirmed answer" which the user must enter a digit to answer the call on the remote phone. This prevents the call from being mistakenly answered by the cell phone voicemail. If confirmed answer is enabled on EC500 and "Initial IP-IP Direct Media" is being used on the SIP Trunk, then when the call is answered on the remote phone the call has one way audio and the enterprise host phone does not stop ringing. This will only be an issue for the customer if they are using EC500 with confirmed answer. A fix for this problem has been tested in a development release and will be available in a future Communication Manager generally available (GA) release. Related MRs include defsw140186 and defsw130477.
  - **Initial IP-IP Direct Media and the response to the "no matching codec" condition using codec G.726**: While testing the error condition of "no matching codec", it was observed that if the Communication Manager SIP trunk to TELUS is configured only for codec G.726, Communication Manager will not send the 488 Not Acceptable Here" error response immediately on receipt of an inbound call but will only send it after the called party has answered. A customer will not be impacted by this behavior if the codecs are properly configured. A fix for this problem has been tested in a development release and will be available in a future Communication Manager generally available (GA) release. Related MRs include defsw140201, defsw140034, and defsw130882.
- **Interactions with TELUS Mobility Endpoints**: These items are TELUS issues and are expected to be addressed by a TELUS equipment vendor.
  - **Attended transfer and REFER:** If the Communication Manager is using the SIP REFER method for transfers and has "Initial IP-IP Direct Media" disabled, then an outbound call from an enterprise SIP endpoint to a TELUS mobility endpoint that is then attended transferred to another TELUS mobility endpoint will result in one-way audio. Inbound calls are unaffected as are outbound calls from H.323 endpoints. This issue may be worked around by either using blind transfer in this scenario or by disabling the use of REFER or by enabling "Initial IP-IP Direct Media" on the Communication Manager. REFER can be disabled by setting the **Network Call Redirection** parameter to "**n**" on **Page 4** of the SIP trunk group form (see **Section 5.8**). "Initial IP-IP Direct Media" can be enabled on **Page 1** of the signaling group form (see **Section 5.7**).
  - **Mobility endpoint covers to voicemail results in no audio**: Calls from enterprise endpoints to TELUS mobility endpoints that covered to voicemail resulted in no audio if the call was established with the G.729 codec and if "Initial IP-IP Direct Media" was enabled on Communication Manager. This can be worked around by setting G.711MU as the preferred codec in the Communication Manager ip-codec-set form (**Section 5.5**) or disabling "Initial IP-IP Direct Media" in the Communication Manager signaling form (**Section 5.7**).
- **Interactions with TELUS BVoIP Endpoints:** This item is a TELUS issue.

- **Outbound calls from the enterprise to a TELUS BVoIP endpoint results in one-way audio or no audio.** During initial testing, outbound calls from the enterprise to TELUS BVoIP endpoints were successful with two-way audio. During subsequent regression testing, it was observed that these calls resulted in one-way audio or no audio. This change in behavior is being investigated by TELUS.

2.3. Support

For technical support on the TELUS system, please contact your TELUS Account Executive or visit http://telus.com.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

Figure 1 illustrates a sample Avaya SIP-enabled enterprise solution connected to the TELUS SIP Trunking Service. In a true customer MPLS deployment, TELUS would provide a MPLS connection from their network directly to the customer site. To simulate this type of deployment in the test environment, an IPSec VPN tunnel was established across the public Internet between the TELUS and Avaya labs. This is the configuration used for compliance testing.

The components used to create the simulated customer site included:

- System Manager
- Session Manager
- Communication Manager
- Avaya G450 Media Gateway
- Avaya Session Border Controller for Enterprise
- Avaya 1600 Series IP Deskphones (H.323)
- Avaya 9600 Series IP Deskphones (H.323 and SIP)
- Avaya A175 Desktop Video Device
- Avaya one-X® Communicator (H.323 and SIP)
- Avaya Flare® Experience for Windows

Located at the edge of the enterprise is the Avaya SBCE. It has a public side that connects to the external network and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise flows through the Avaya SBCE. In this way, the Avaya SBCE can protect the enterprise against any SIP-based attacks. The Avaya SBCE provides network address translation at both the IP and SIP layers. For security reasons, any actual public IP addresses used in the configuration have been replaced with private IP addresses in this document. Similarly, any references to real routable PSTN numbers have also been changed to numbers that cannot be routed by the PSTN.

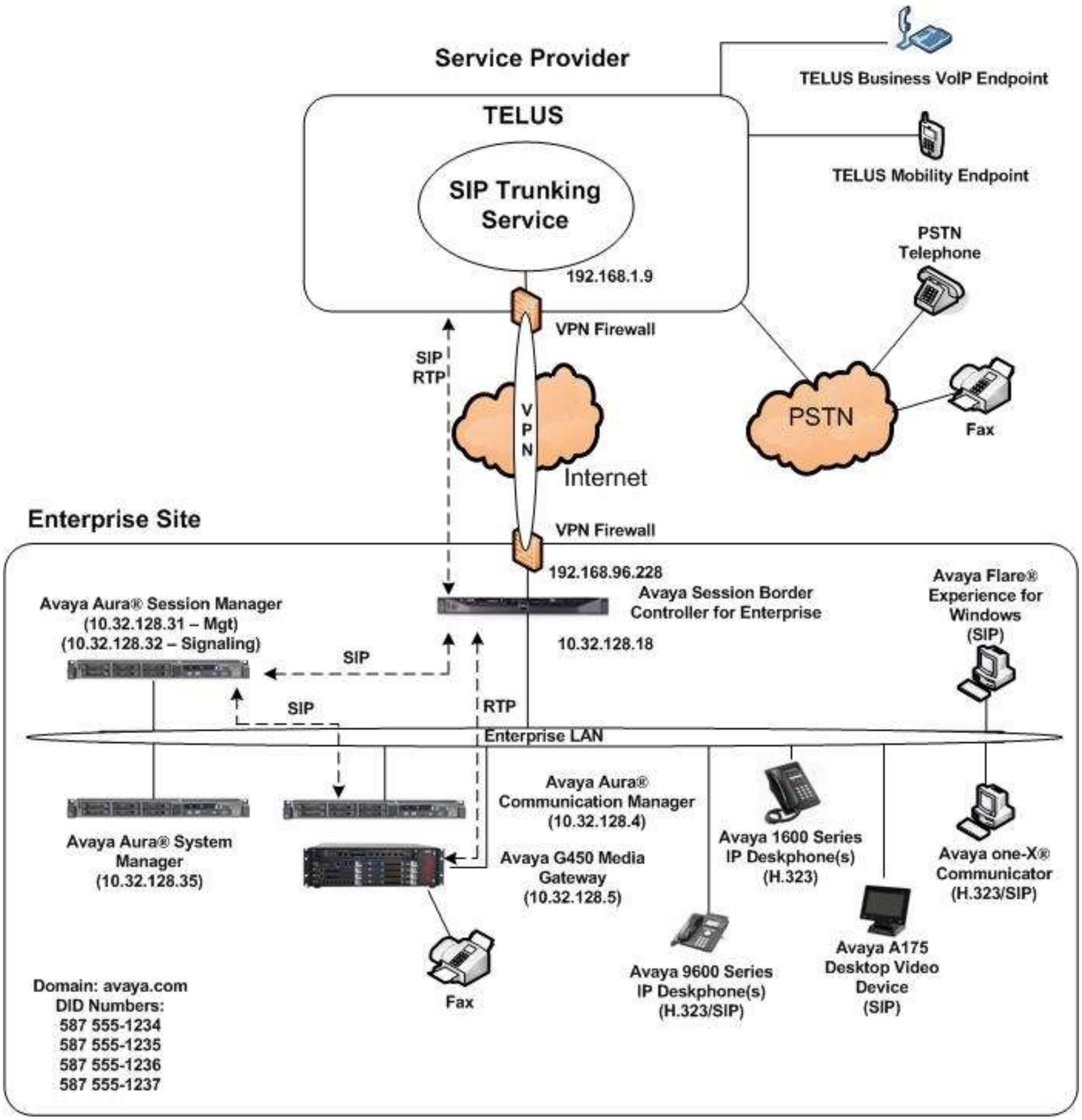


Figure 1: Avaya IP Telephony Network using the TELUS SIP Trunking Service

A separate trunk was created between Communication Manager and Session Manager to carry the service provider traffic. This was done so that any trunk or codec setting required by the service provider could be applied only to this trunk and not affect other enterprise SIP traffic. In addition, this trunk carried both inbound and outbound traffic.

For inbound calls, the calls flow from the service provider to the Avaya SBCE then to Session Manager. Session Manager uses the configured dial patterns (or regular expressions) and routing policies to determine the recipient (in this case Communication Manager) and on which link to send the call. Once the call arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed.

Outbound calls to the PSTN are first processed by Communication Manager and may be subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects the proper SIP trunk, the call is routed to Session Manager. Session Manager once again uses the configured dial patterns (or regular expressions) to determine the route to the Avaya SBCE. From the Avaya SBCE, the call is sent to the TELUS SIP Trunking Service.

TELUS requires 11 digits (1+10 digits) be sent in the Request URI header for long distance calls and 10 digits for local calls.

4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Equipment/Software	Release/Version
Avaya Aura® System Manager running on a HP ProLiant DL360 G7 Server	6.3 SP11 (Build 6.3.0.8.5682-6.3.8.4751) (Software Update Revision 6.3.11.8.2933) System Platform 6.3.5.01003.0
Avaya Aura® Session Manager running on a HP ProLiant DL360 G7 Server	6.3 SP11 (Build 6.3.11.0.631103)
Avaya Aura® Communication Manager running on an Avaya S8300 Server	6.3 SP9.1 + patch (R016x.03.0.124.0-22230) System Platform 6.3.5.01003.0 Patch includes fixes for defsw141241, defsw141360 and CM-3712
Avaya G450 Media Gateway	34.5.1
Avaya Session Border Controller for Enterprise	6.3.1-22-4653
Avaya 1616 IP Deskphone (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP5 (1.3.50B)
Avaya 9641G IP Deskphone (H.323) running Avaya one-X® Deskphone Edition	6.4.0 (6.4014)
Avaya 9611 IP Deskphone (SIP) running Avaya one-X® Deskphone SIP Edition	6.5.0 (6.5.0.17)
Avaya A175 Desktop Video Device with Avaya Flare® Experience	1.1.3
Avaya one-X® Communicator (H.323 or SIP)	6.2 SP5 (Build 6.2.5.03-SP5)
Avaya Flare® Experience for Windows	1.1.4 (1.1.4.23)
TELUS SIP Trunking Service Release 2 Platform Components	
Equipment/Software	Release/Version
Oracle AP6300 Session Border Controller	7.1.2 MR 4
Genband EXPERiUS Application Server	MCP-17.0.18.4
Genband C20 Call Session Controller	CVM17

Table 1: Equipment and Software Tested

The specific configuration above was used for the compliance testing. Note that this solution will be compatible with other Avaya Server and Media Gateway platforms running similar versions of Communication Manager and Session Manager.

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager for the TELUS SIP Trunking Service. A SIP trunk is established between Communication Manager and Session Manager for use by traffic to and from TELUS. It is assumed the general installation of Communication Manager, the Avaya Media Gateway and Session Manager has been previously completed and is not discussed here.

The Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. Note that the IP addresses and phone numbers shown throughout these Application Notes have been edited so that the actual public IP addresses of the network elements and public PSTN numbers are not revealed.

5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to the service provider. The example shows that **4000** SIP trunks are available and **70** are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 4000  36
           Maximum Concurrently Registered IP Stations: 2400  2
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  1
                   Maximum Video Capable IP Softphones: 2400  4
                       Maximum Administered SIP Trunks: 4000  70
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
```

5.2. Special Application SA8965

Special application SA8965 controls whether a new protocol variation parameter, **Shuffling with SDP**, appears on **Page 4** of the trunk form (see **Section 5.8**). In previous compliance tests, TELUS required that all INVITE messages contain SDP information, including re-INVITEs. Thus, the trunk parameter made available by SA8965 needed to be enabled.

In the current compliance test, re-INVITEs are no longer required to contain SDP information. Thus, SA8965 should be disabled but only after the **Shuffling with SDP** parameter on the trunk form has been disabled (**Section 5.8**).

SA8965 is controlled via the **change system-parameters special-applications** command. To disable this special application, navigate to **Page 7** and enter **n** next to the special application titled **SA8965 - SIP Shuffling with SDP** in the list below.

```
change system-parameters special-applications                   Page   7 of   9
                             SPECIAL APPLICATIONS

                       (SA8888) - Per Station Music On Hold? n
     (SA8889) - Verizon VoiceGenie SIP MIME Message Bodies? n
                  (SA8891) - Verizon VoiceGenie SIP Headers? n
                                (SA8893) - Blast Conference? n
                       (SA8896) - IP Softphone Lamp Control? n
                  (SA8900) - Support for NTT Call Screening? n
               (SA8904) - Location Based Call Type Analysis? n
                   (SA8911) - Expanded Public Unknown Table? n
      (SA8917) - LSP Redirect using special coverage point? n
                          (SA8927) - Increase Paging Groups? n
     (SA8928) - Display Names on Bridged Appearance Labels? n
            (SA8931) - Send IE with EC500 Extension Number? n
          (SA8942) - Multiple Unicode Message File Support? n
          (SA8944) - Multiple Logins for Single IP Address? n
                             (SA8946) - Site Data Expansion? n
  (SA8958) - Increase BSR Polling/Interflow Pairs to 40000? n
                          (SA8965) - SIP Shuffling with SDP? n
  (SA8967) - Mask CLI and Station Name for QSIG/ISDN Calls? n
```

5.3. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to **all** to allow incoming calls from the PSTN to be transferred to another PSTN endpoint. If for security reasons incoming calls should not be allowed to transfer back to the PSTN then leave the field set to **none**.

```
change system-parameters features                               Page   1 of  20
                            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
```

On **Page 9**, verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. This text string is entered in the two fields highlighted below. The compliance test used the value of **anonymous** for both.

```
change system-parameters features                               Page   9 of  20
                        FEATURE-RELATED SYSTEM PARAMETERS

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

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

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

SCCAN PARAMETERS
   Enable Enbloc Dialing without ARS FAC? n

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

5.4. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of the server running Communication Manager **(procr**) and for Session Manager (**sessionMgr**). These node names will be needed for defining the service provider signaling group in **Section 5.7**.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
cmm                 10.32.128.4
default             0.0.0.0
procr               10.32.128.4
procr6              ::
sessionMgr          10.32.128.32
```

5.5. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. To configure the codecs, enter the codecs in the **Audio Codec** column of the table in the order of preference defined by the service provider. For the compliance test, ip-codec-set 3 was used for this purpose. To mitigate the impact of some issues documented in **Section 2.2**, it is recommended to set G.711MU as the preferred codec. Default values can be used for all other fields.

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

                          IP Codec Set

    Codec Set: 3

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

On **Page 2**, set the **FAX Mode** to **t.38-G711-fallback**. In general, TELUS supports T.38 fax but not on all media gateways in the network. Using the **t.38-G711-fallback** setting will allow all fax calls to succeed, though some may use G.711 fax instead of T.38. See **Section 2.2** for details.

```
change ip-codec-set 3                                           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
```

5.6. IP Network Region

Create a separate IP network region for the service provider trunk. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the service provider versus calls within the enterprise or elsewhere. For the compliance test, IP network region 3 was chosen for the service provider trunk. Use the **change ip-network-region 3** command to configure region 3 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is **avaya.com**. This name appears in the "From" header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Set both **Intra-region** and **Inter-region IP-IP Direct Audio** to **yes.** This is the default setting. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.5**.
- Default values can be used for all other fields.

```
change ip-network-region 3                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 3
Location:         Authoritative Domain: avaya.com
    Name: SP Region             Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

On **Page 4**, define the IP codec set to be used for traffic between region 3 and region 1. Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) **1**. Default values may be used for all other fields. The example below shows the settings used for the compliance test. It indicates that codec set 3 will be used for calls between region 3 (the service provider region) and region 1 (the rest of the enterprise). Creating this table entry for IP network region 3 will automatically create a complementary table entry on the IP network region 1 form for destination region 3. This complementary table entry can be viewed using the **display ip-network-region 1** command and navigating to **Page 4** (not shown).

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

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

5.7. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and Session Manager for use by the service provider trunk. This signaling group is used for inbound and outbound calls between the service provider and the enterprise. For the compliance test, signaling group 3 was used for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** to the recommended default value of **tls** (Transport Layer Security). For ease of troubleshooting during testing, some of the compliance test was conducted with the **Transport Method** set to **tcp**. The transport method specified here is used between Communication Manager and Session Manager. If TLS is used here, it must also be used on the Session Manager entity link defined in **Section 6.6**.
- Set the **IMS Enabled** field to **n**. This specifies Communication Manager will serve as an Evolution Server for Session Manager.
- Set the **Peer Detection Enabled** field to **y**. The **Peer-Server** field will initially be set to **Others** and cannot be changed via administration. Later, the **Peer-Server** field will automatically change to **SM** once Communication Manager detects its peer as a Session Manager.
- Set the **Near-end Node Name** to **procr**. This node name maps to the IP address of Communication Manager as defined in **Section 5.4**.
- Set the **Far-end Node Name** to **sessionMgr**. This node name maps to the IP address of Session Manager as defined in **Section 5.4**.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061 and for TCP the well-known port value is 5060). At the time of Session Manager installation, a SIP connection between Communication Manager and Session Manager would have been established for use by all Communication Manager SIP traffic using the well-known port

value for TLS or TCP. By creating a new signaling group with a separate port value, a separate SIP connection is created between Communication Manager and Session Manager for SIP traffic to the service provider. As a result, any signaling group or trunk group settings (**Section 5.8**) will only affect the service provider traffic and not other SIP traffic at the enterprise. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to **5063**.

- Set the **Far-end Network Region** to the IP network region defined for the service provider in **Section 5.6**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set the **DTMF over IP** field to **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Set **Direct IP-IP Audio Connections** to **y**. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic from the Avaya Media Gateway and allow it to flow directly between the SIP trunk and the enterprise endpoint.
- Set **Initial IP-IP Direct Media** to **n** or **y** depending on the customer requirements. This option attempts to directly connect the media traffic between the SIP trunk and the enterprise endpoint at initial call-setup instead of establishing a media connection to the Avaya Media Gateway which is later redirected to the endpoints. However, if this option is set on the service provider signaling group, it must be set the same on the signaling group associated with the SIP trunk used by the enterprise SIP endpoints. In the test configuration, this was signaling group 1 (not shown). If the customer has no requirement for **Initial IP-IP Direct Media**, then the recommendation is to set the parameter to **n**.
- Set the **Alternate Route Timer** to **15**. This defines the number of seconds that Communication Manager will wait for a response (other than 100 Trying) to an outbound INVITE before selecting another route. If an alternate route is not defined, then the call is cancelled after this interval.
- Default values may be used for all other fields.

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

 Group Number: 3             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: sessionMgr
 Near-end Listen Port: 5063               Far-end Listen Port: 5063
                                       Far-end Network Region: 3

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 15
```

5.8. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.7**. For the compliance test, trunk group 3 was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Service Type** field to **public-ntwrk**.
- Set **Member Assignment Method** to **auto**.
- Set the **Signaling Group** to the signaling group shown in the previous section.
- Set the **Number of Members** field to the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk.
- Default values were used for all other fields.

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

Group Number: 3                  Group Type: sip          CDR Reports: y
  Group Name: SP Trunk                  COR: 1        TN: 1        TAC: 1003
   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
```

On **Page 2**, the **Redirect On OPTIM Failure** value is the amount of time (in milliseconds) that Communication Manager will wait for a response (other than 100 Trying) to a pending INVITE sent to an EC500 remote endpoint before selecting another route. If another route is not defined, then the call is cancelled after this interval. This time interval should be set to a value equal to the **Alternate Route Timer** on the signaling group form described in **Section 5.7**.

Verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. For the compliance test, the value of **900** seconds was used.

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

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 15000

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

 Disconnect Supervision - In? y  Out? y


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

On **Page 3**, set the **Numbering Format** field to **private**. This field specifies the format of the calling party number (CPN) sent to the far-end. Beginning with Communication Manager 6.0, public numbers are automatically preceded with a + sign (E.164 numbering format) when passed in the SIP From, Contact and P-Asserted Identity headers. To remove the + sign, the **Numbering Format** was set to **private** and the **Numbering Format** in the route pattern was set to **unk-unk** (see **Section 5.10**).

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to **y**. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.3**, if the inbound call enabled CPN block. For outbound calls, these same settings request that CPN block be activated on the far-end destination if a local user requests CPN block on a particular call routed out this trunk. Default values were used for all other fields.

```
add 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? y
                                                 Replace Unavailable Numbers? y


                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y

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

On **Page 4**, set **Mark Users as Phone** as **y**. This is recommended by TELUS. The **Network Call Redirection** field may be set to **y** or **n**. Setting the **Network Call Redirection** flag to **y** enables use of the SIP REFER message for call transfer; otherwise the SIP INVITE message will be used for call transfer. Both approaches are supported with this solution.

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

Set the **Telephone Event Payload Type** to **101**, the value used by TELUS.

Set the **Shuffling with SDP** field to **n** if it appears on the form. This parameter only appears if special application SA8965 is enabled. In addition, this setting must be repeated for the internal SIP trunk used by the enterprise SIP endpoints. Since calls between the enterprise SIP endpoints and TELUS traverse two SIP trunks: the internal SIP trunk for intra-enterprise traffic (trunk 1 in the test configuration) and the service provider SIP trunk to TELUS (trunk 3), the **Shuffling with SDP** parameter must be set the same on both. Once **Shuffling with SDP** has been set to **n** on **both** trunks (trunks 3 and 1), disable special application SA8965 to remove this parameter from the form. See **Section 5.2** for full details.

Set **Always Use re-INVITE for Display Updates** to **y**. TELUS returned a 488 Not Acceptable Here response to some of the Communication Manager display update messages. To avoid these errors, the Communication Manager was configured to use re-INVITEs for display updates instead of UPDATE messages.

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

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

                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? y
                        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
```

5.9. Calling Party Information

The calling party number is sent in the SIP "From", "Contact" and "PAI" headers. Since private numbering was selected to define the format of this number (**Section 5.8**), use the **change private-numbering** command to create an entry for each extension which has a DID assigned. The DID number will be assigned by the SIP service provider. It is used to authenticate the caller.

In the sample configuration, four DID numbers were assigned for testing. These four numbers were assigned to the four extensions 40006, 40008, 40022, and 40024. Thus, 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 private-numbering 5                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 5  4                                          5     Total Administered: 5
 5  40006          3          5875551234       10       Maximum Entries: 540
 5  40008          3          5875551235       10
 5  40022          3          5875551236       10
 5  40024          3          5875551237       10
```

In a real customer environment, normally the DID number is comprised of the local extension plus a prefix. If this is true, then a single private numbering entry can be applied for all extensions. In the example below, all stations with a 5-digit extension beginning with 4 will send the calling party number as the **Private Prefix** plus the extension number.

```
change private-numbering 5                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext            Trk        Private      Total
Len Code           Grp(s)     Prefix        Len
 5  4                                       5     Total Administered: 2
 5  4              3          58755         10       Maximum Entries: 540
```

5.10. Outbound Routing

In these Application Notes, the Automatic Route Selection (ARS) feature is used to route outbound calls via the SIP trunk to the service provider. In the sample configuration, the single digit 9 is used as the ARS access code. Enterprise callers will dial 9 to reach an "outside line". This common configuration is illustrated below with little elaboration. Use the **change dialplan analysis** command to define a dialed string beginning with 9 of length 1 as a feature access code (**fac**).

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all            Percent Full: 3

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   1           4   dac
   3           5   ext
   4           5   ext
   8           1   fac
   9           1   fac
   *           3   fac
   #           3   fac
```

Use the **change feature-access-codes** command to configure **9** 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: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *01    All: *02    Deactivation: *03
   Call Forwarding Enhanced Status:        Act:        Deactivation:
```

TELUS requires the sending of 11 (1 + 10) digits for long distance calls and sending of 10 digits for local calls. Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example below shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to route pattern **2** 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: 1

          Dialed           Total      Route     Call    Node   ANI
          String          Min  Max   Pattern    Type    Num    Reqd
    0                     1    1      2         op             n
    0                     11   11     2         op             n
    011                   10   18     2         intl           n
    1647                  11   11     2         natl           n
    1732                  11   11     2         natl           n
    1800                  11   11     2         natl           n
    1877                  11   11     2         natl           n
    1908                  11   11     2         natl           n
    411                   3    3      2         svcl           n
    587                   10   10     2         natl           n
```

The route pattern defines which trunk group will be used for an outgoing call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the service provider route pattern in the following manner. The example below shows the values used for route pattern 2 during the compliance test.

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the SIP service provider. For the compliance test, 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 5.8**.
- **LAR**: **next**

```
change route-pattern 2                                          Page   1 of   3
                    Pattern Number: 4   Pattern Name: SP Route
                             SCCAN? n     Secure SIP? 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  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                              unk-unk   next
 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
```

Use the **save translation** command to save all Communication Manager configuration described in **Section 5**.

6. Configure Avaya Aura® Session Manager

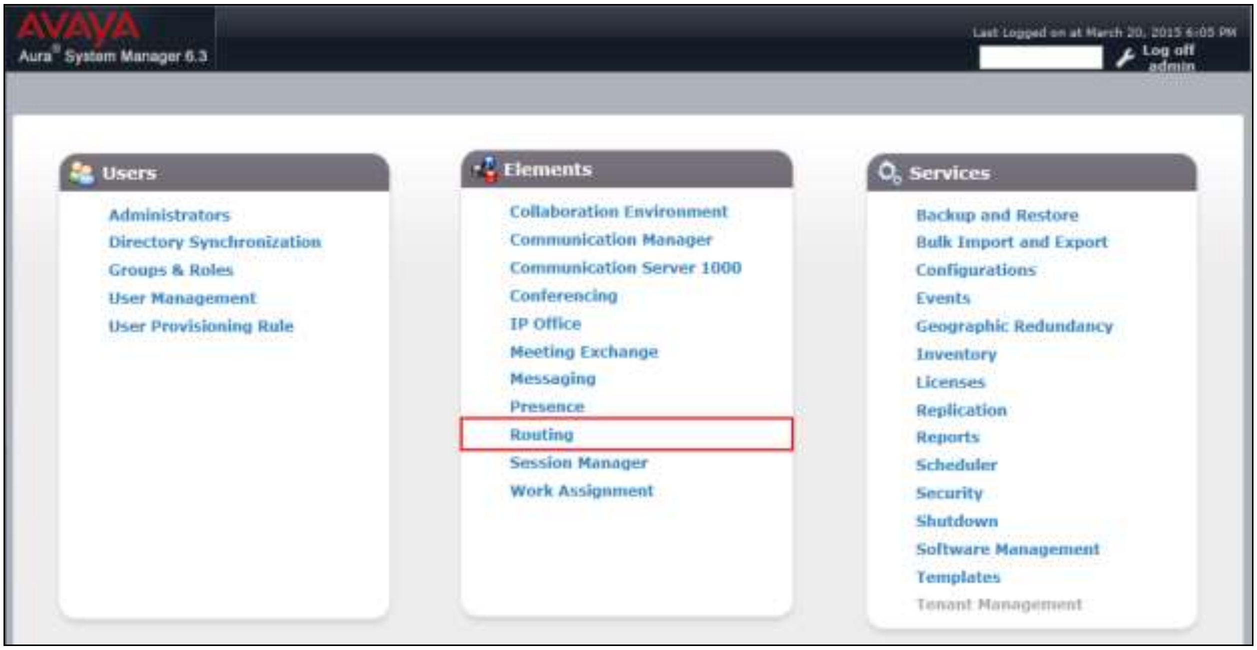
This section provides the procedures for configuring Session Manager. The procedures include configuring the following items:

- SIP Domain
- Location
- Adaptation Modules
- SIP Entities
- Entity Links
- Routing Policies
- Dial Patterns
- Session Manager

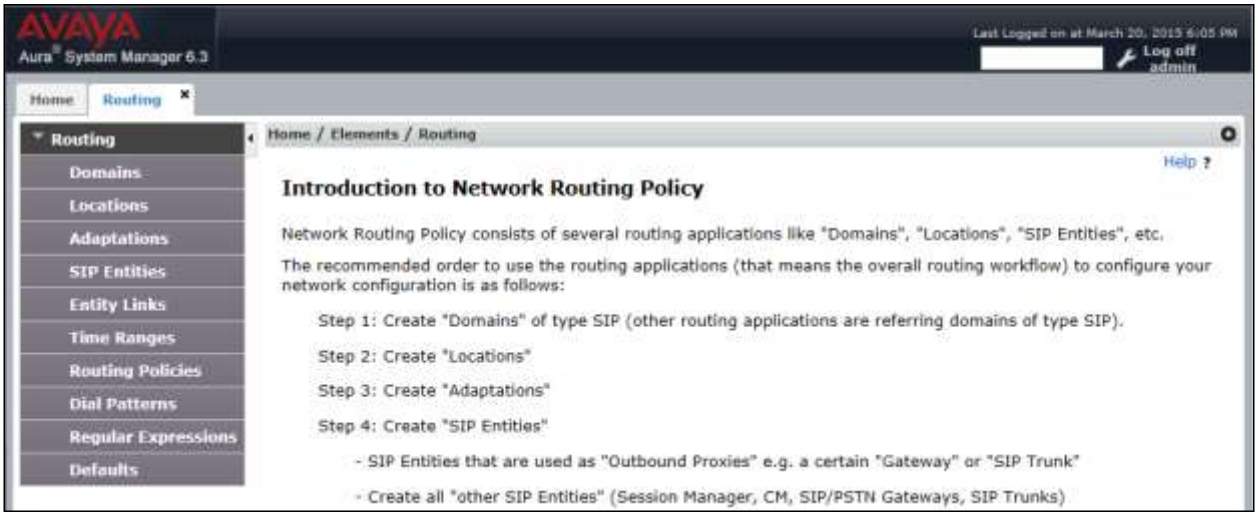
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP Domains, Locations, SIP Entities, and Session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

6.1. Avaya Aura® System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "https://<ip-address>/SMGR", where "<ip-address>" is the IP address of System Manager. Log in with the appropriate credentials and click on **Login** (not shown). The following page is displayed. The links displayed below will be referenced in subsequent sections to navigate to items requiring configuration. Most items will be located under the **Elements → Routing** link highlighted below.



Clicking the **Elements → Routing** link, displays the **Introduction to Network Routing Policy** page. In the left-hand pane is a navigation tree containing many of the items to be configured in the following sections.

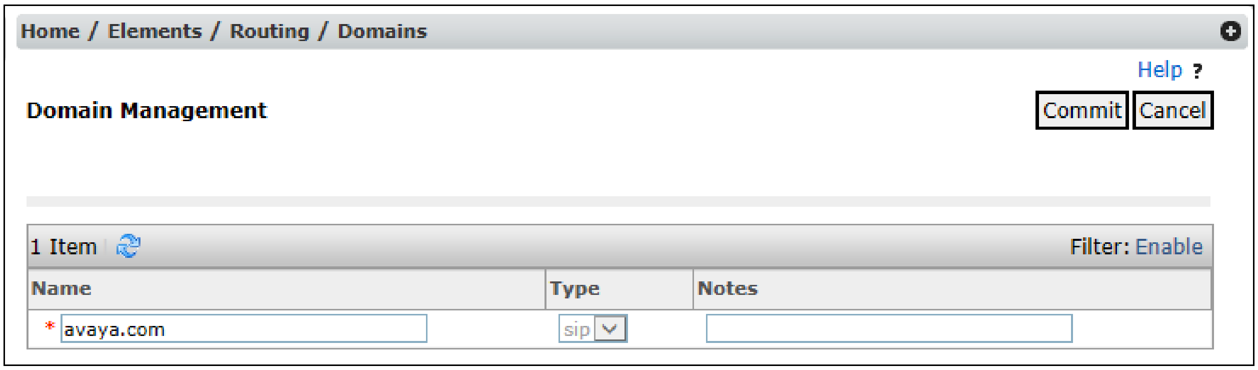


6.2. Specify SIP Domain

Create a SIP Domain for each domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this includes the enterprise domain (**avaya.com**) as defined in **Section 5.6**. Navigate to **Routing → Domains** in the left-hand navigation pane (**Section 6.1**) and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).

Click **Commit**. The screen below shows the entry for the enterprise domain.



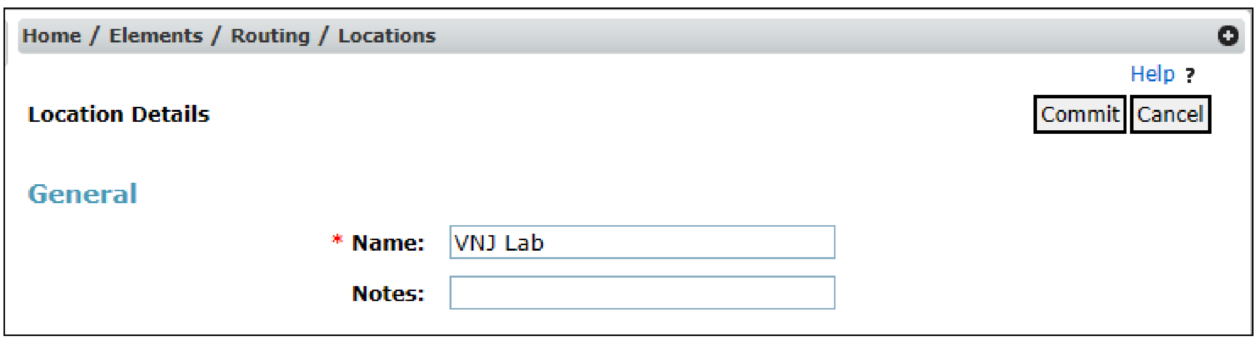
6.3. Add Location

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. A single Location was defined for the enterprise even though multiple subnets were used. The screens below show the addition of the Location named **VNJ Lab**, which includes all equipment at the enterprise including Communication Manager, Session Manager and the Avaya SBCE.

To add a Location, navigate to **Routing → Locations** in the left-hand navigation pane (**Section 6.1**) and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

In the **General** section, enter the following values. Use default values for all remaining fields.

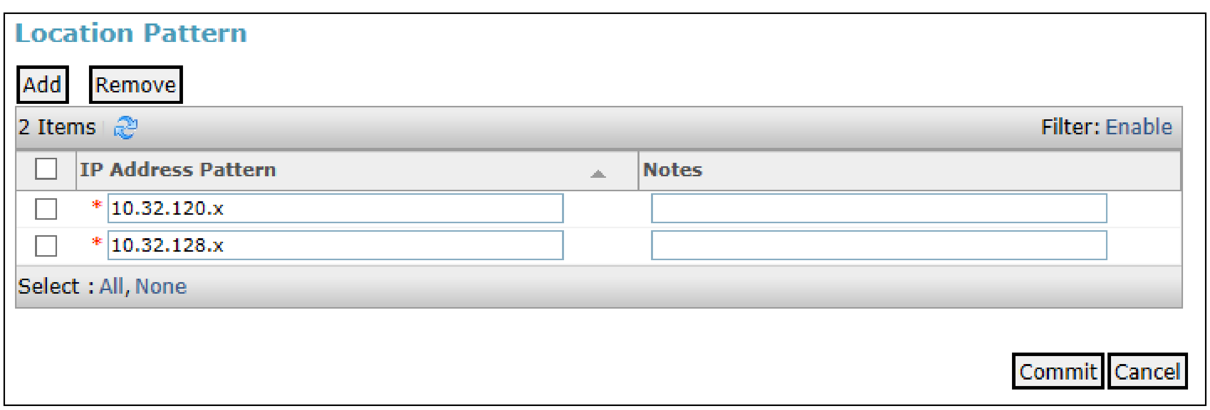
- **Name:** Enter a descriptive name for the Location.
- **Notes:** Add a brief description (optional).



Scroll down to the **Location Pattern** section. Click **Add** and enter the following values. Use default values for all remaining fields.

- **IP Address Pattern:** Add all IP address patterns used to identify the location.
- **Notes:** Add a brief description (optional).

Click **Commit** to save.



6.4. Add Adaptation

Session Manager can be configured with Adaptations that can modify SIP messages before or after routing decisions have been made or perform digit manipulation. The Adaptation **DigitConversionAdapter** supports digit conversion of telephone numbers in specific headers of SIP messages.

For the compliance test, two Adaptations were used. The first Adaptation is applied to the Communication Manager SIP Entity and performs the following:

- Mapping inbound DID numbers from TELUS to local Communication Manager extensions.

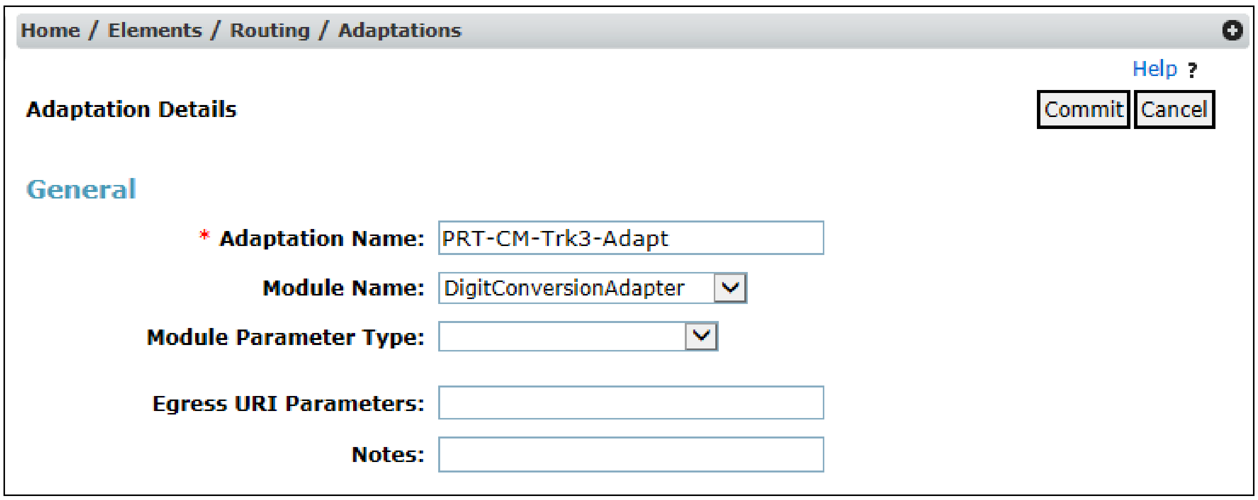
The second Adaptation is applied to the Avaya SBCE SIP Entity and performs the following:

- Mapping the internal extension number of the Avaya one-X® Communicator SIP softphone to the assigned DID number. This Adaptation is not strictly required as described in **Section 2.2** but was used in the compliance test as a precaution. For more details, see the **Avaya one-X® Communicator SIP in "Other Phone" Mode** item in the observation/limitation list in **Section 2.2**.

To create the Adaptation that will be applied to the Communication Manager SIP Entity, navigate to **Routing → Adaptations** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

In the **General** section, enter the following values. Use default values for all remaining fields.

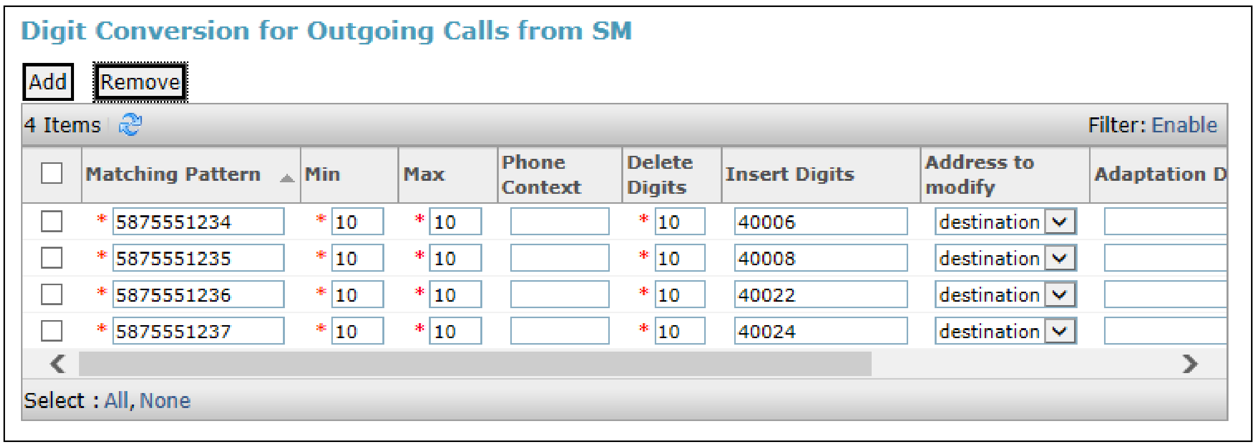
- **Adaptation name:** Enter a descriptive name for the Adaptation.
- **Module name:** Select **DigitConversionAdapter** from the drop-down menu.
- **Module Parameter Type:** Leave blank.
- **Notes:** Enter a description (optional).



To map inbound DID numbers from TELUS to Communication Manager extensions, scroll down to the **Digit Conversion for Outgoing Calls from SM** section. Create an entry for each DID to be mapped. Click **Add** and enter the following values for each mapping. Use default values for all remaining fields.

- **Matching Pattern:** Enter a digit string used to match the inbound DID number.
- **Min:** Enter a minimum dialed number length used in the match criteria.
- **Max:** Enter a maximum dialed number length used in the match criteria.
- **Delete Digits** Enter the number of digits to delete from the beginning of the received number.
- **Insert Digits:** Enter the digits to insert at the beginning of the received number.
- **Address to modify:** Select **destination** since this digit conversion only applies to the destination number.

Click **Commit** to save.



Digit Conversion for Outgoing Calls from SM

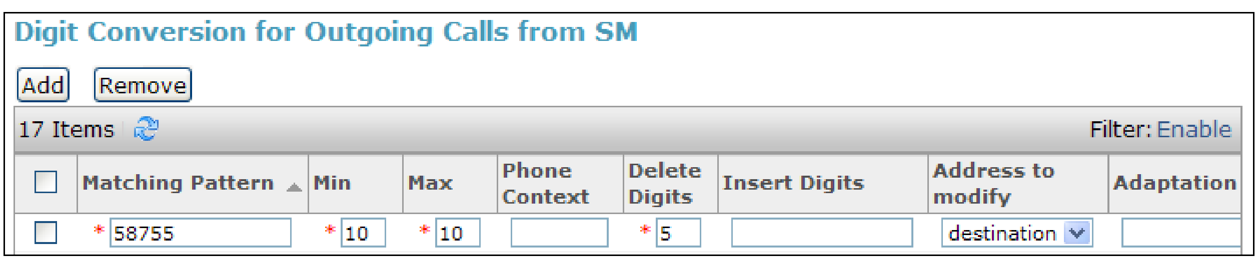
Add Remove

4 Items Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation D
☐	* 5875551234	* 10	* 10		* 10	40006	destination	
☐	* 5875551235	* 10	* 10		* 10	40008	destination	
☐	* 5875551236	* 10	* 10		* 10	40022	destination	
☐	* 5875551237	* 10	* 10		* 10	40024	destination	

Select : All, None

In a real customer environment, often the DID number is comprised of the local extension plus a prefix. If this is true, then a single digit conversion entry can be created for all extensions. In the example below, a 5 digit prefix is deleted from each incoming DID number leaving a 5 digit extension to be routed by Session Manager.



Digit Conversion for Outgoing Calls from SM

Add Remove

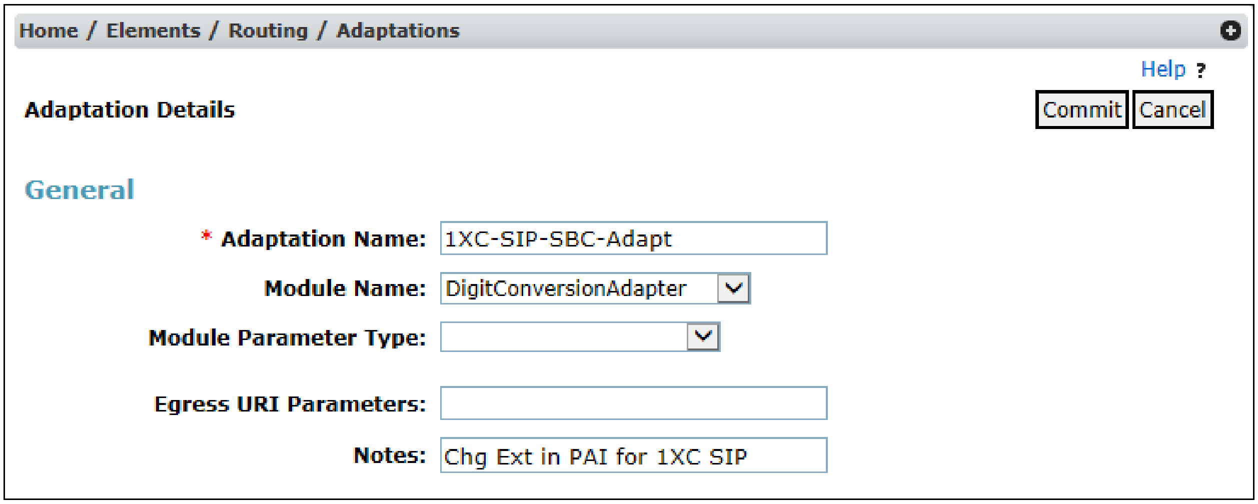
17 Items Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation
☐	* 58755	* 10	* 10		* 5		destination	

To create the Adaptation that will be applied to the Avaya SBCE SIP Entity, navigate to **Routing → Adaptations** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

In the **General** section, enter the following values. Use default values for all remaining fields.

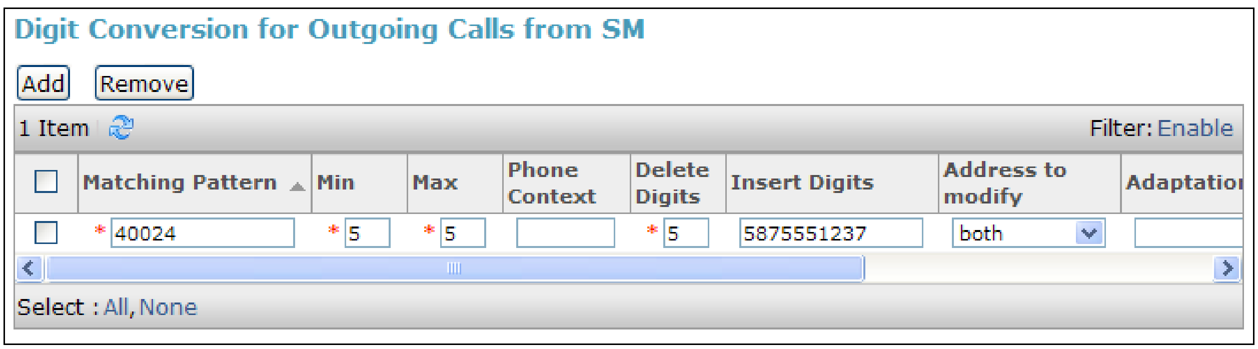
- **Adaptation name:** Enter a descriptive name for the Adaptation.
- **Module name:** Enter **DigitConversionAdapter**.
- **Notes:** Add a brief description (optional).



To map the Communication Manager extension number for the Avaya one-X® Communicator SIP softphone to the TELUS DID number assigned to the extension, scroll down to the **Digit Conversion for Outgoing Calls from SM** section. Create an entry for each Avaya one-X® Communicator SIP softphone extension to be mapped. Click **Add** and enter the following values for each mapping. Use default values for all remaining fields.

- **Matching Pattern:** Enter the Avaya one-X® Communicator SIP softphone extension.
- **Min:** Enter a minimum dialed number length used in the match criteria.
- **Max:** Enter a maximum dialed number length used in the match criteria.
- **Delete Digits** Enter the number of digits to delete from the beginning of the received number.
- **Insert Digits:** Enter the DID number assigned to the Avaya one-X® Communicator SIP softphone extension.
- **Address to modify:** Select **both**.

Click **Commit** to save.



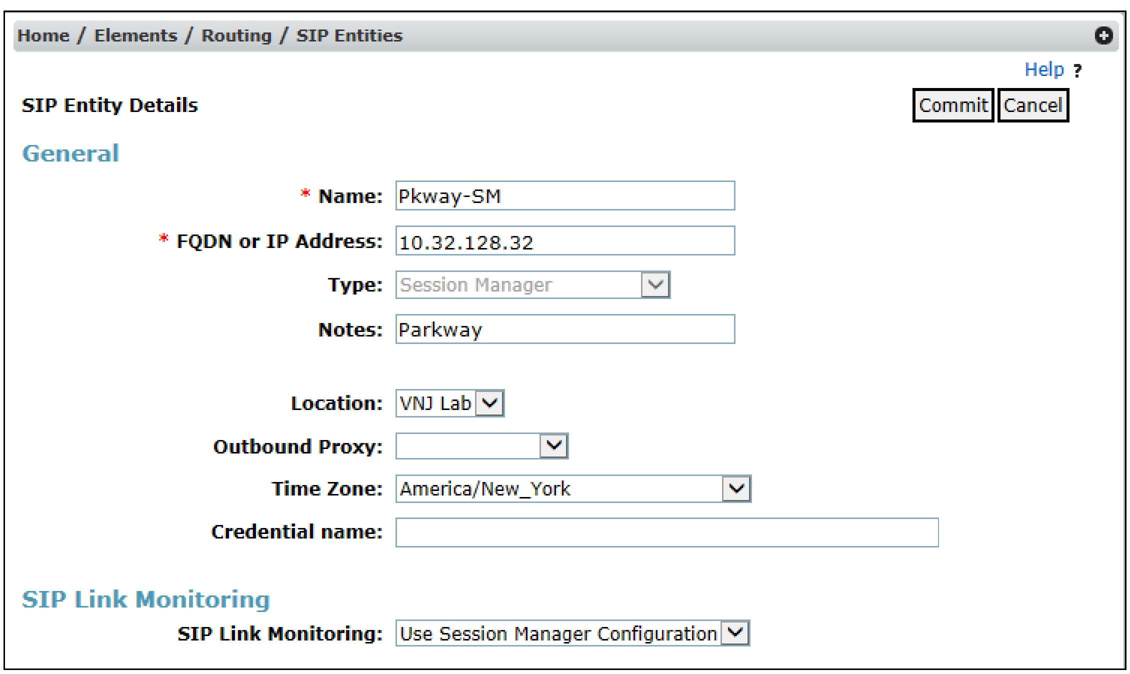
6.5. Add SIP Entity

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to Session Manager which includes Communication Manager and the Avaya SBCE. Navigate to **Routing → SIP Entities** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

In the **General** section, enter the following values. Use default values for all remaining fields.

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Enter **Session Manager** for Session Manager, **CM** for Communication Manager and **SIP Trunk** for the Avaya SBCE.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If applicable, select the appropriate **Adaptation name** created in **Section 6.4** that will be applied to this entity.
- **Location:** Select the Location that applies to the SIP Entity being created. For the compliance test, all components were located in Location **VNJ Lab** created in **Section 6.3**.
- **Time Zone:** Select the time zone for the Location above.

The following screen shows the addition of Session Manager. The IP address of the virtual SM-100 Security Module is entered for **FQDN or IP Address**.



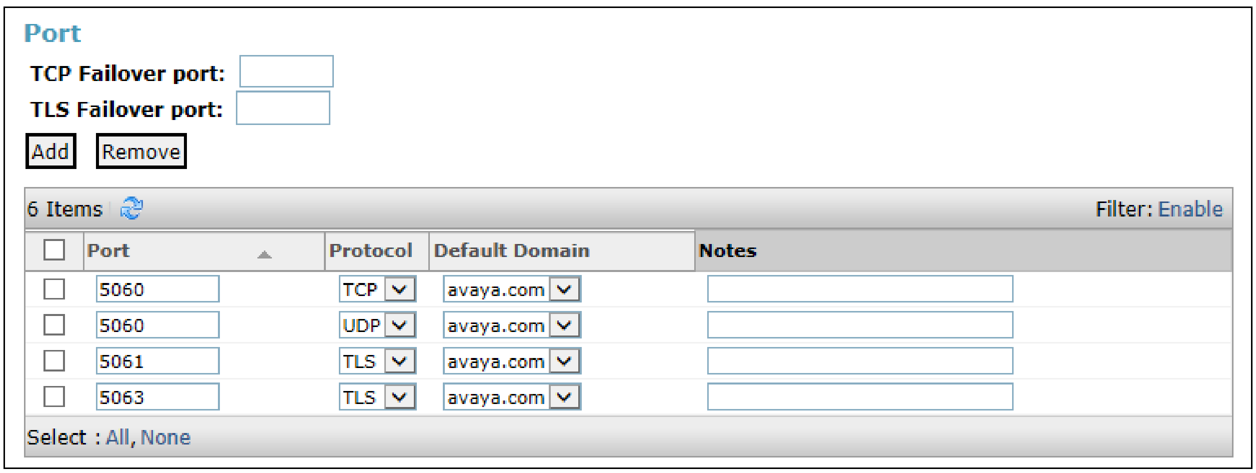
To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP Entities.

In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:

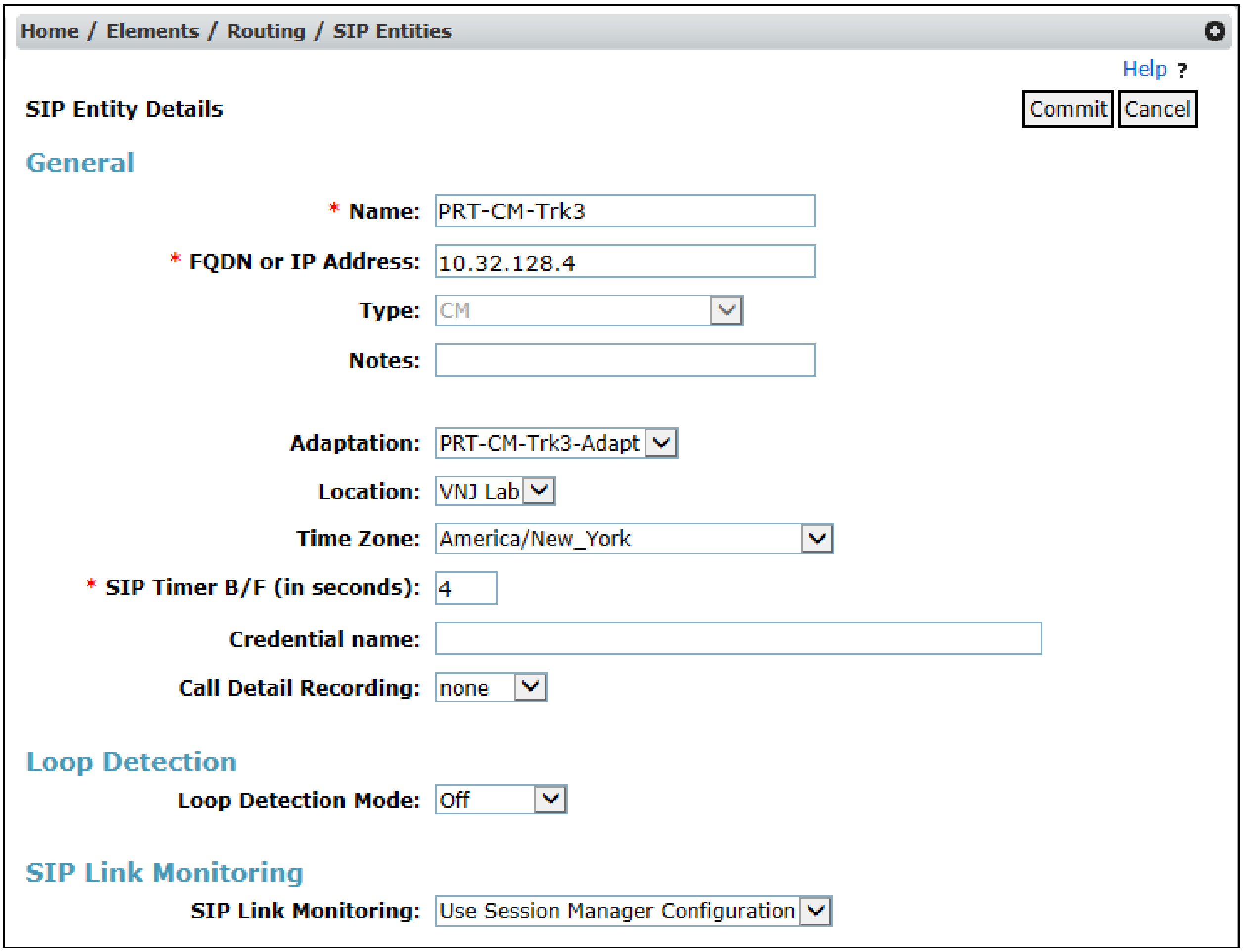
- **Port:** Port number on which Session Manager can listen for SIP requests.
- **Protocol:** Transport protocol to be used with this port.
- **Default Domain:** The default domain associated with this port. For the compliance test, this was the enterprise SIP domain.

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

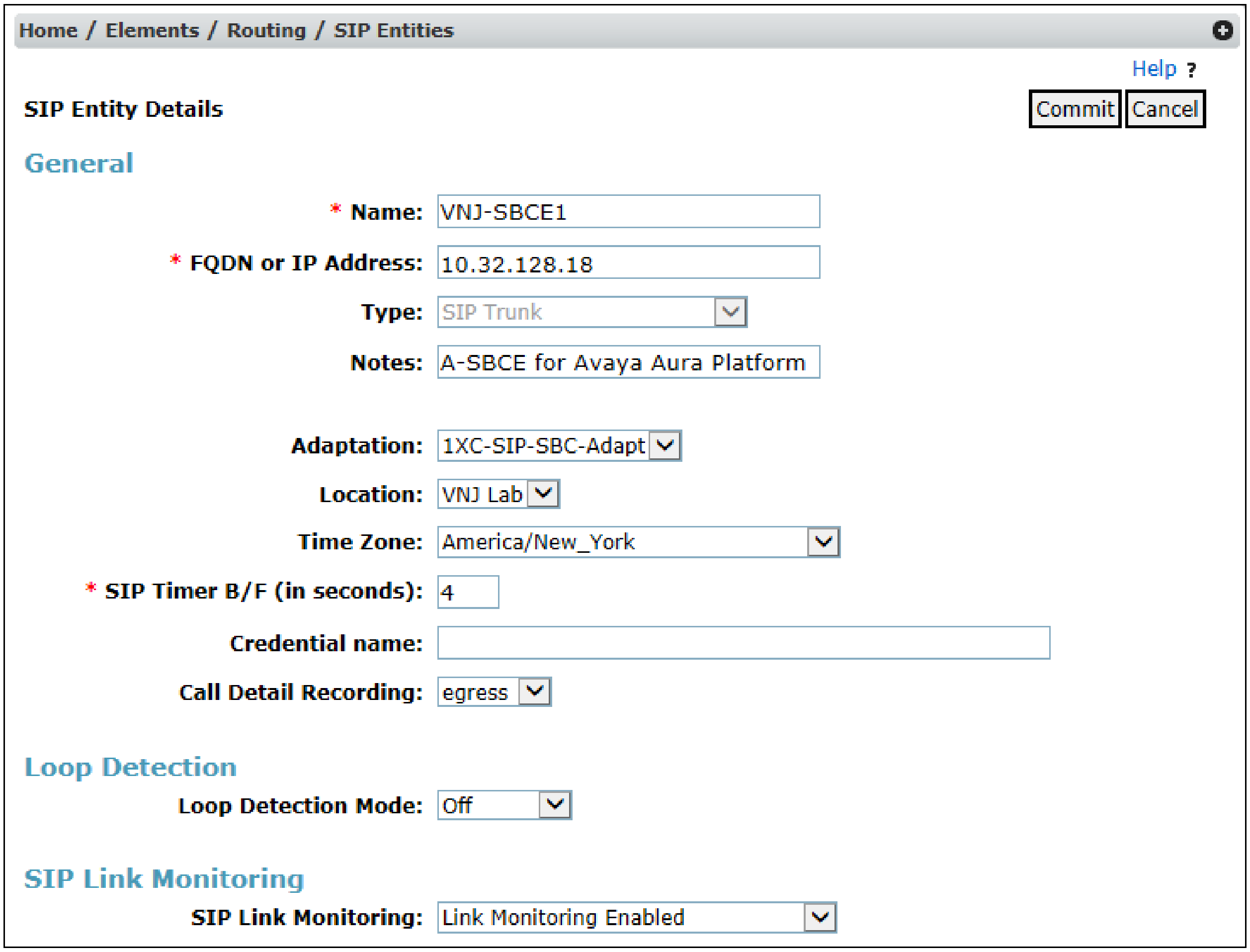
For the compliance test, four port entries were used. The first three are the standard ports used for SIP traffic: port 5060 for UDP/TCP and port 5061 for TLS. These ports were provisioned as part of the Session Manager installation not covered by this document. In addition, port 5063 defined in **Section 5.6** for use with service provider SIP traffic between Communication Manager and Session Manager was added to the list.



The following screen shows the addition of Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, this requires the creation of a separate SIP Entity for Communication Manager other than the one created at Session Manager installation for use with all other SIP traffic. The **FQDN or IP Address** field is set to the IP address of Communication Manager. For the **Adaptation** field, select the Adaptation previously defined for dial plan digit manipulation in **Section 6.4**. The **Location** field is set to **VNJ Lab** which is the Location defined for the subnet where Communication Manager resides. See **Section 6.3**.



The following screen shows the addition of the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of its private network interface (see **Figure 1**). For the **Adaptation** field, select the Adaptation previously defined for the Avaya SBCE in **Section 6.4**. The **Location** field is set to **VNJ Lab** which is the Location defined for the subnet where the Avaya SBCE resides.

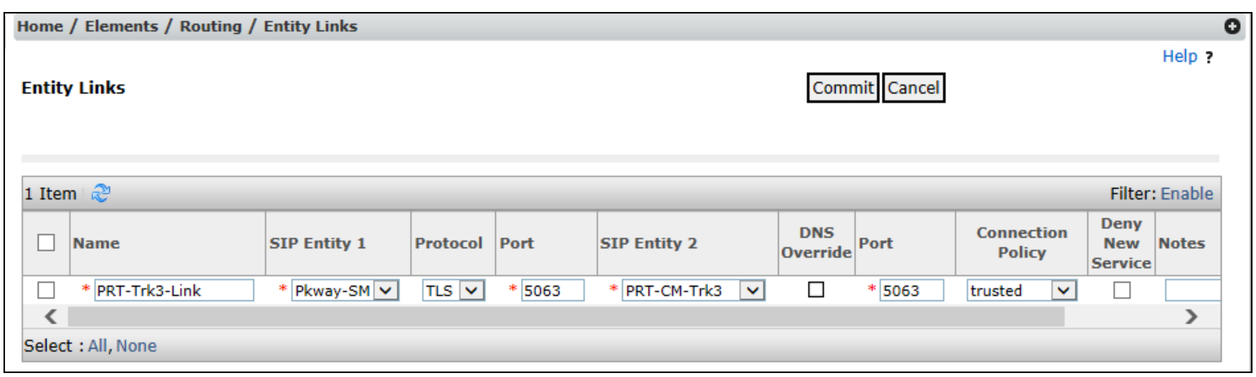


6.6. Add Entity Links

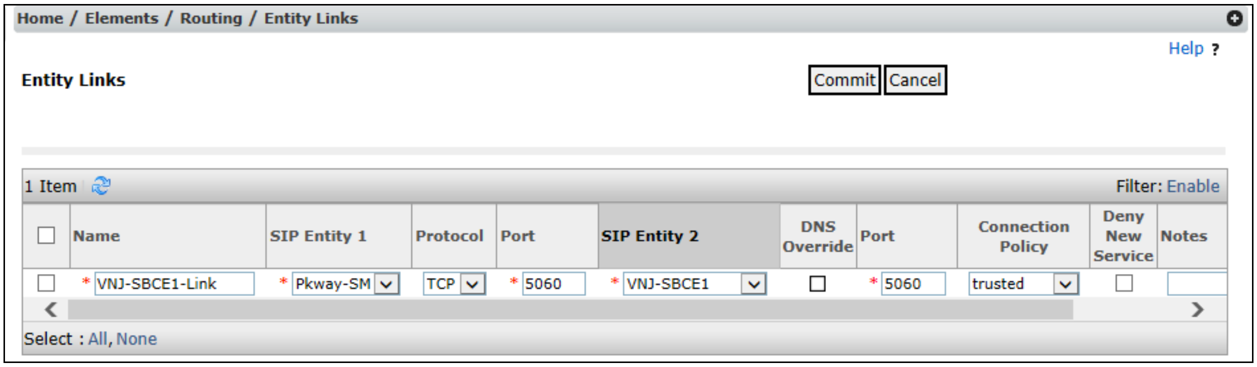
A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created: one to Communication Manager for use only by service provider traffic and one to the Avaya SBCE. To add an Entity Link, navigate to **Routing → Entity Links** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager SIP Entity.
- **Protocol:** Select the transport protocol used for this link. For the Communication Manager Entity Link, this must match the protocol used in the Communication Manager signaling group in **Section 5.7**.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For the Communication Manager Entity Link, this must match the **Far-end Listen Port** defined on the Communication Manager signaling group in **Section 5.7**.
- **SIP Entity 2:** Select the name of the other system. For the Communication Manager Entity Link, select the Communication Manager SIP Entity defined in **Section 6.5**.
- **Port:** Port number on which the other system receives SIP requests from Session Manager. For the Communication Manager Entity Link, this must match the **Near-end Listen Port** defined on the Communication Manager signaling group in **Section 5.7**.
- **Connection Policy:** Select **trusted** from pull-down menu.

Click **Commit** to save. The following screen illustrates the Entity Link to Communication Manager (**PRT-Trk3-Link**). The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.7**. For part of the compliance test, the TCP protocol was used but the recommended configuration is to use TLS.



The following screen illustrates the Entity Link to the Avaya SBCE (**VNJ-SBCE1-Link**).



6.7. Add Routing Policies

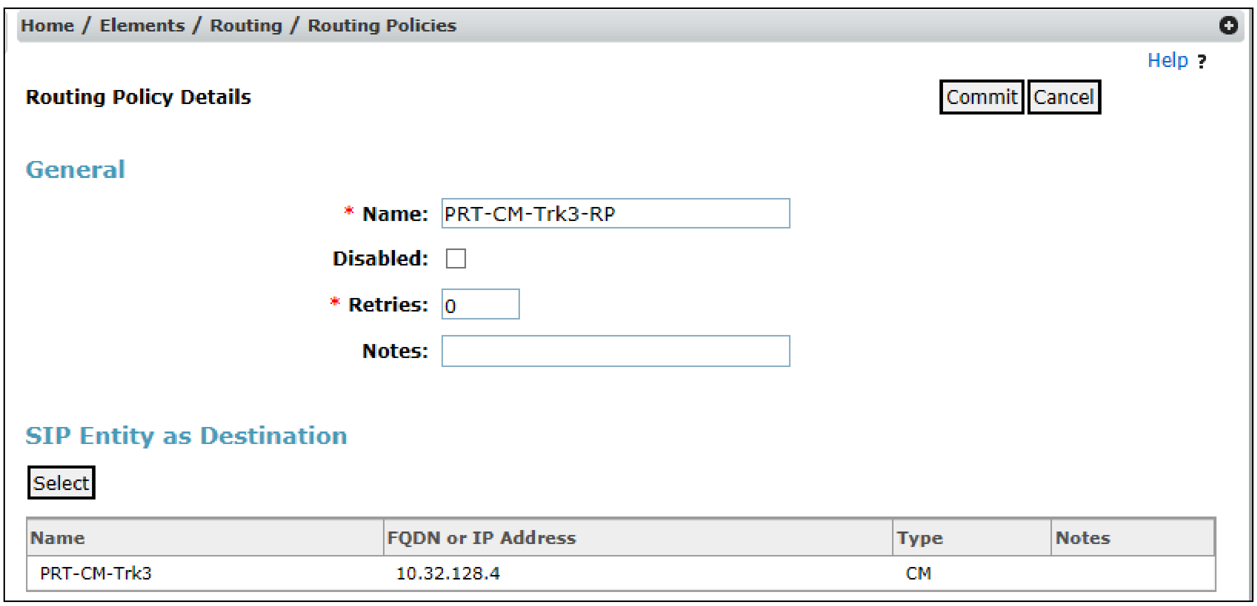
Routing Policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Two Routing Policies must be added: one for Communication Manager and one for the Avaya SBCE. To add a Routing Policy, navigate to **Routing → Routing Policies** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

In the **General** section, enter the following values. Use default values for all remaining fields.

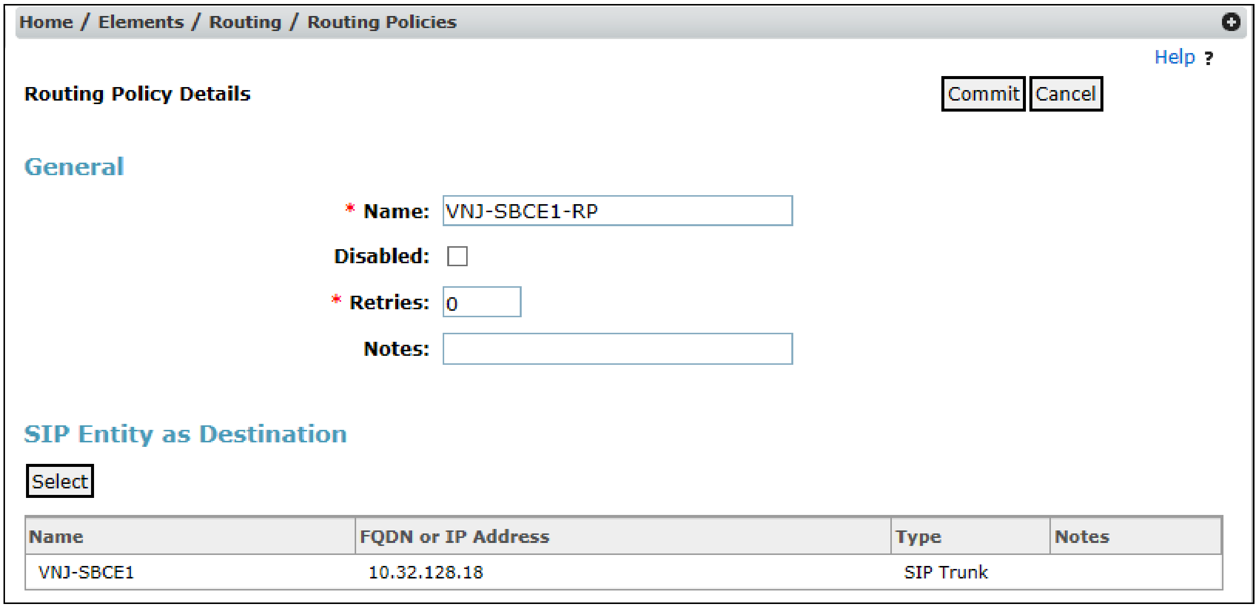
- **Name:** Enter a descriptive name.
- **Notes:** Add a brief description (optional).

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

The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy for the Avaya SBCE.



6.8. Add Dial Patterns

Dial Patterns are needed to route calls through Session Manager. For the compliance test, Dial Patterns were needed to route calls from Communication Manager to TELUS and vice versa. Dial Patterns define which Route 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-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

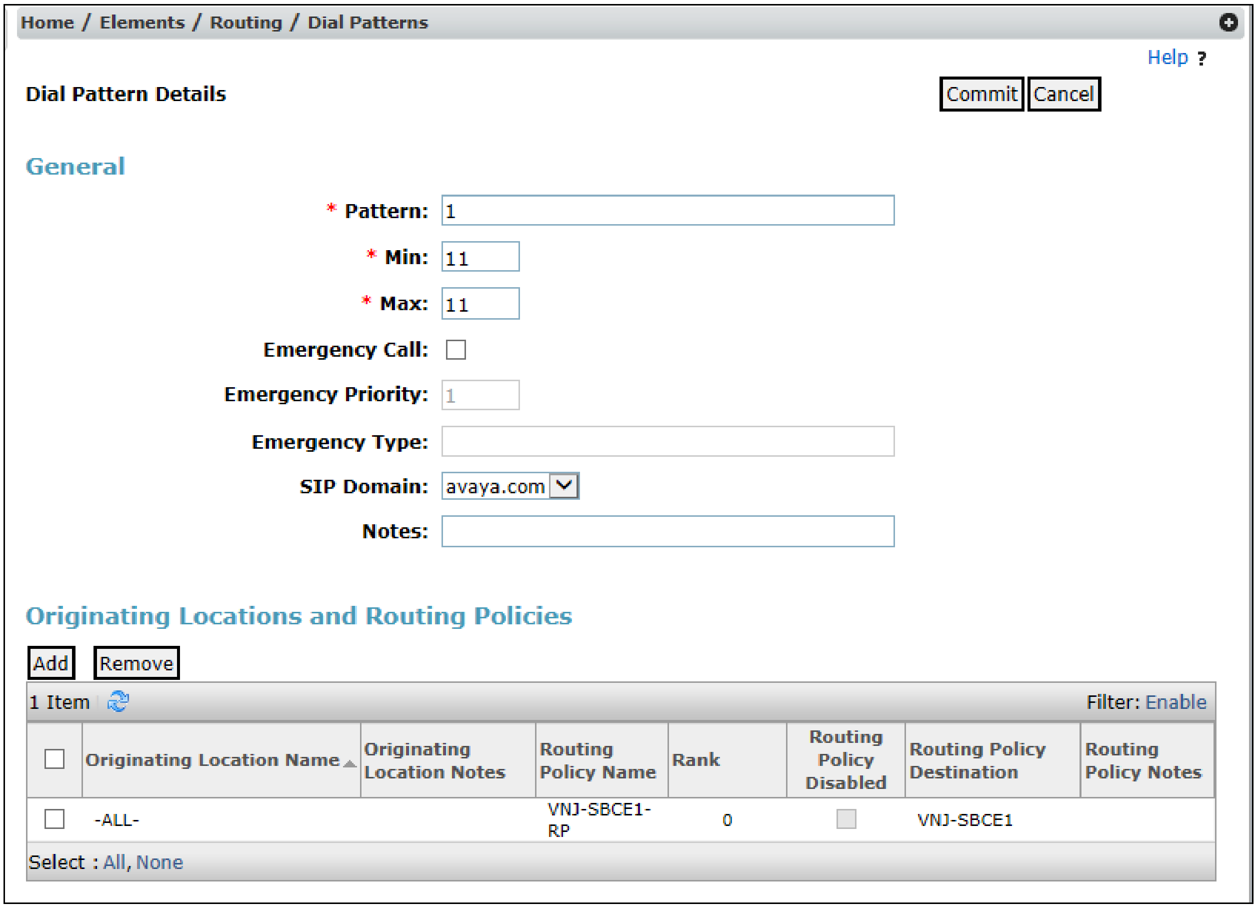
In the **General** section, enter the following values. Use default values for all remaining fields.

- **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).

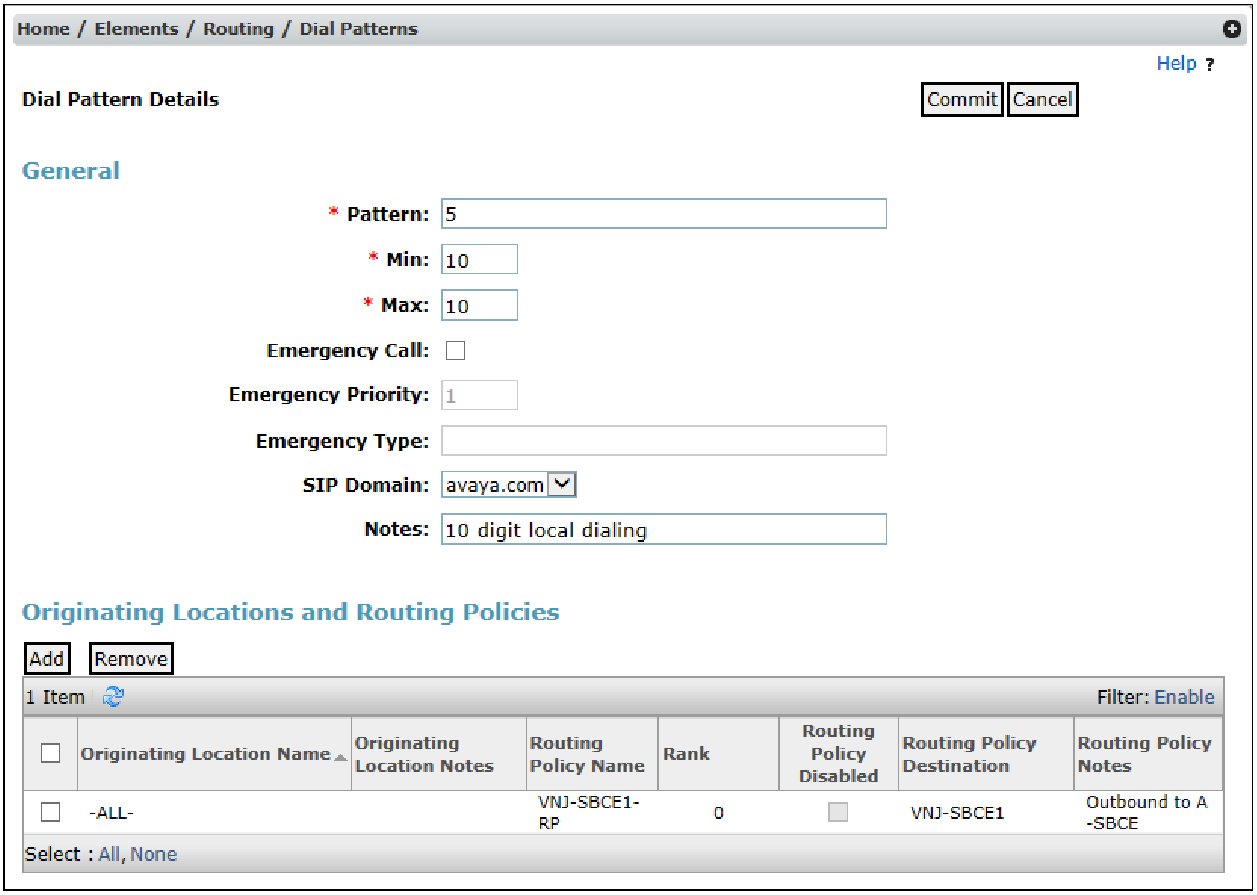
In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the Routing Policy from the list that will be used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

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

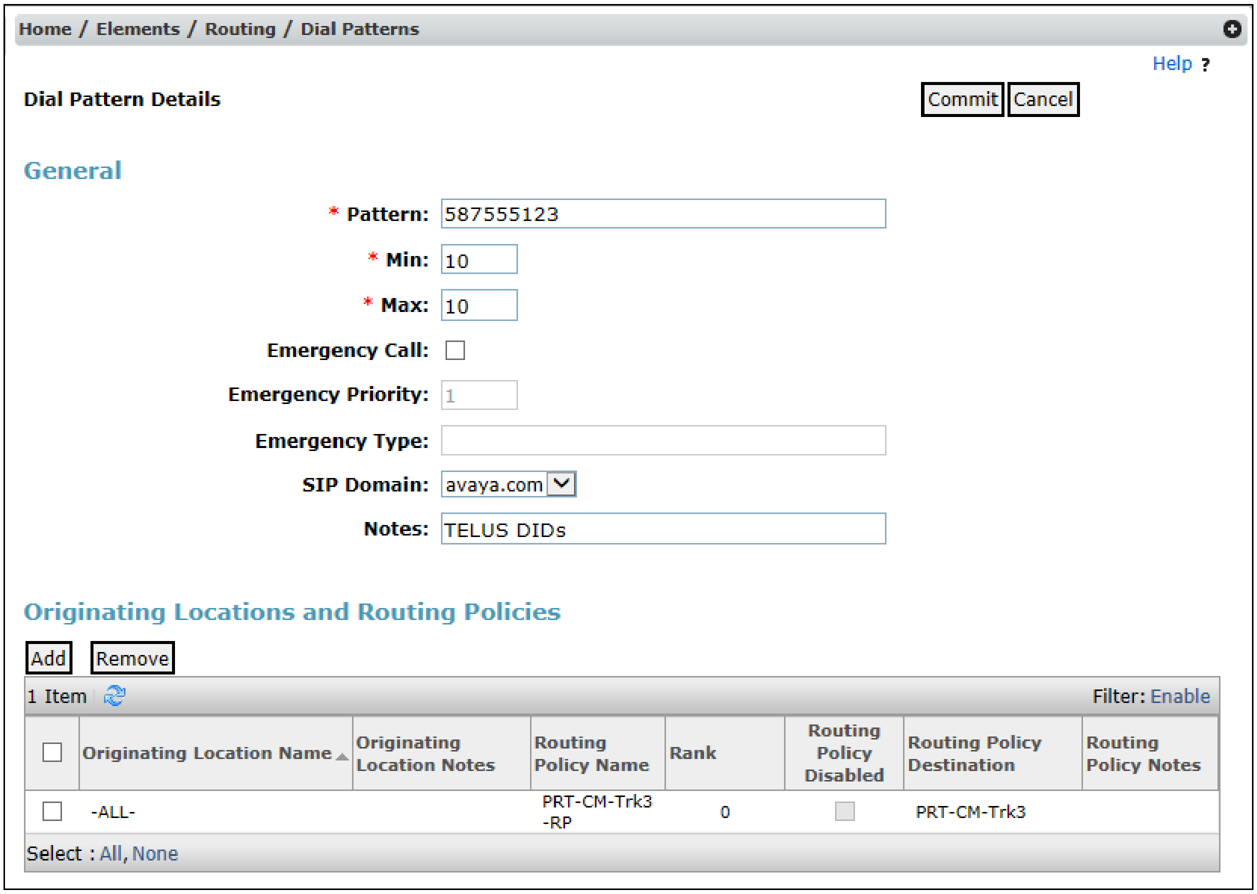
Three examples of the Dial Patterns used for the compliance test are shown below. The first example shows that outbound long distance numbers (11 digits) that begin with **1** and have a destination domain of **avaya.com** from **ALL** locations use route policy **VNJ-SBCE1-RP**.



The second example shows that outbound local 10 digit numbers that start with **5** to domain **avaya.com** and originating from **ALL** locations use route policy **VNJ-SBCE1-RP**.



The third example shows that incoming 10 digit numbers that start with **587555123** to domain **avaya.com** and originating from **ALL** locations use route policy **PRT-CM-Trk3-RP**. These are the DID numbers assigned to the enterprise from TELUS. All other Dial Patterns used as part of the compliance test were configured in a similar manner.



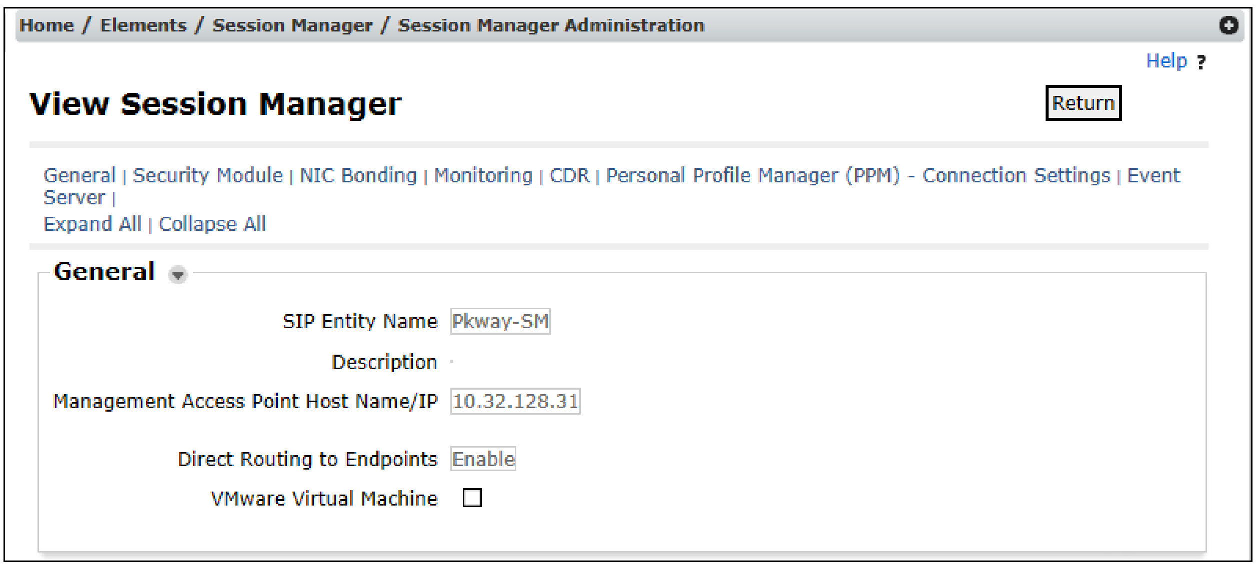
6.9. Add/View Session Manager

The creation of a Session Manager element provides the linkage between System Manager and Session Manager. This was most likely done as part of the initial Session Manager installation. To add a Session Manager, from the **Home** page, navigate to **Elements → Session Manager → Session Manager Administration** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). If the Session Manager already exists, select the appropriate Session Manager and click **View** (not shown) to view the configuration. Enter/verify the data as described below and shown in the following screen:

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

- **SIP Entity Name:** Select the SIP Entity created for Session Manager.
- **Description**: Add a brief description (optional).
- **Management Access Point Host Name/IP:** Enter the host name or IP address of the Session Manager management interface.

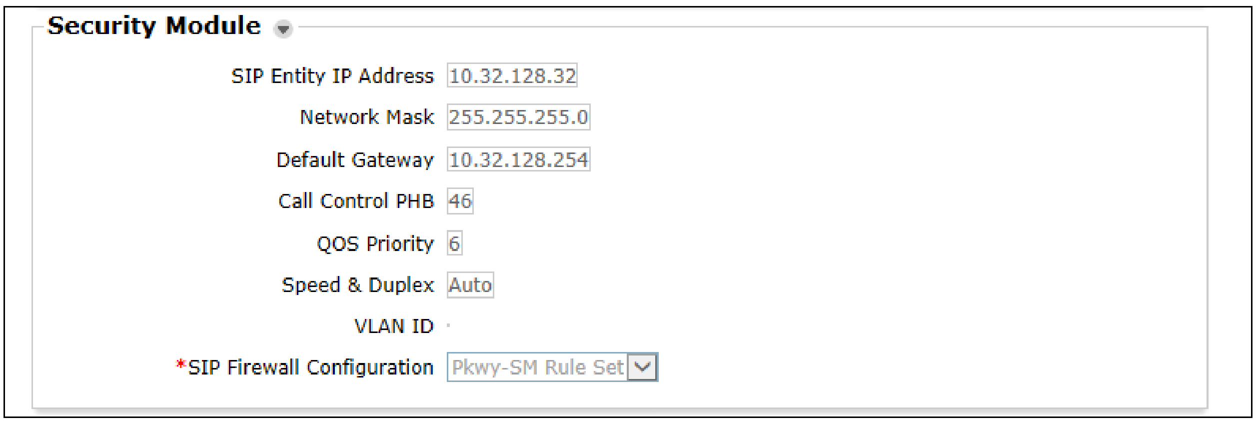
The screen below shows the Session Manager values used for the compliance test.



In the **Security Module** section, enter the following values:

- **SIP Entity IP Address:** Should be filled in automatically based on the SIP Entity Name. Otherwise, enter the IP address of the Session Manager signaling interface.
- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager.
- **Default Gateway**: Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields. Click **Save** (not shown) to add this Session Manager. The screen below shows the remaining Session Manager values used for the compliance test.



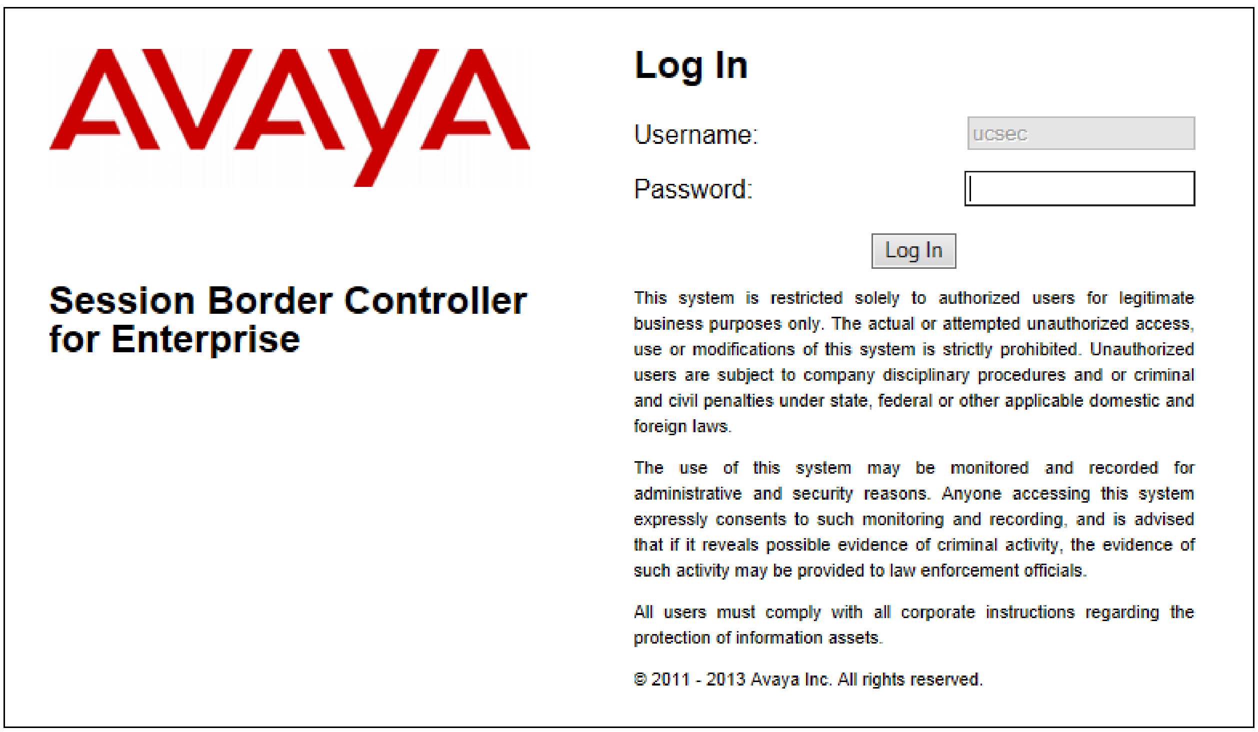
7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Avaya SBCE. It is assumed that the initial installation of the Avaya SBCE has been completed including the assignment of a management IP address. The management interface **must** be provisioned on a different subnet than either the Avaya SBCE private or public network interfaces (e.g., A1 and B1). If the management interface has not been configured on a separate subnet, then contact your Avaya representative for guidance in correcting the configuration.

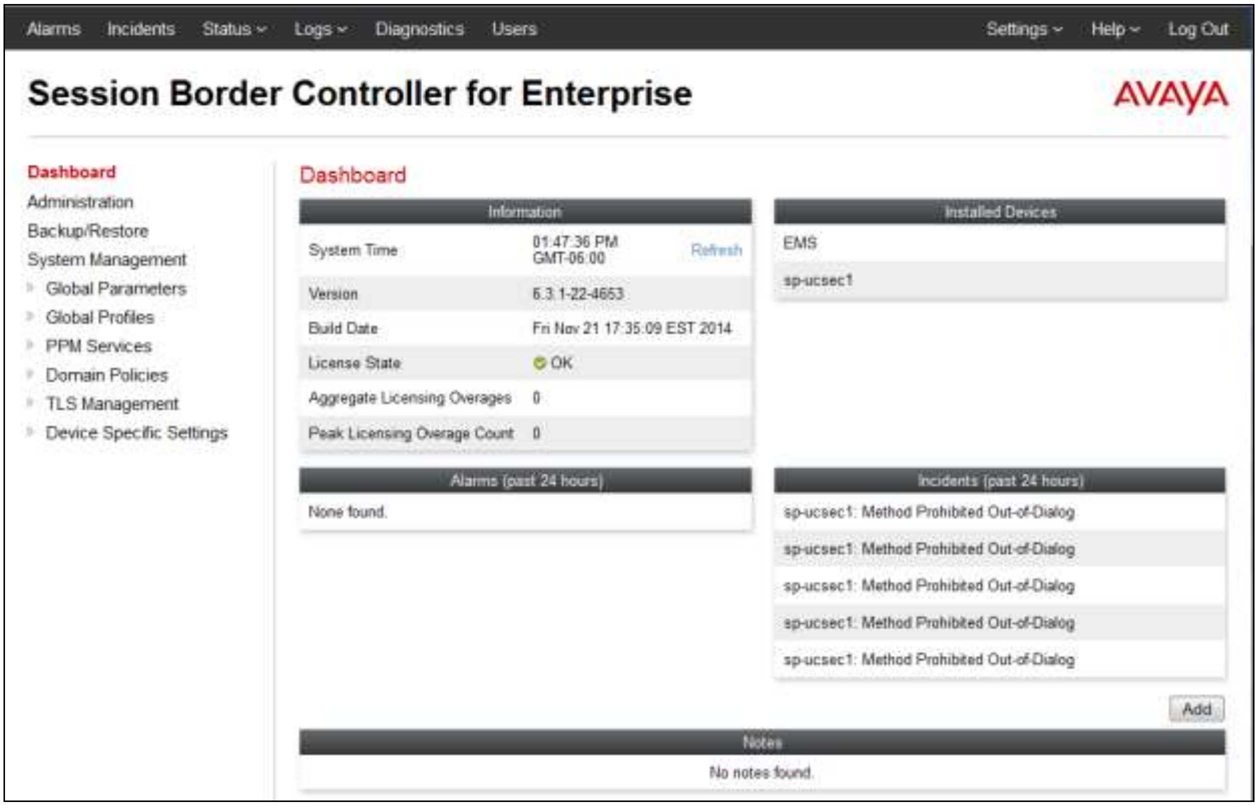
On all screens described in this section, it is to be assumed that parameters are left at their default values unless specified otherwise.

7.1. Access the Management Interface

Use a web browser to access the web interface by entering the URL **https://<ip-addr>**, where **<ip-addr>** is the management IP address assigned during installation. The Avaya SBCE login page will appear as shown below. Log in with appropriate credentials.



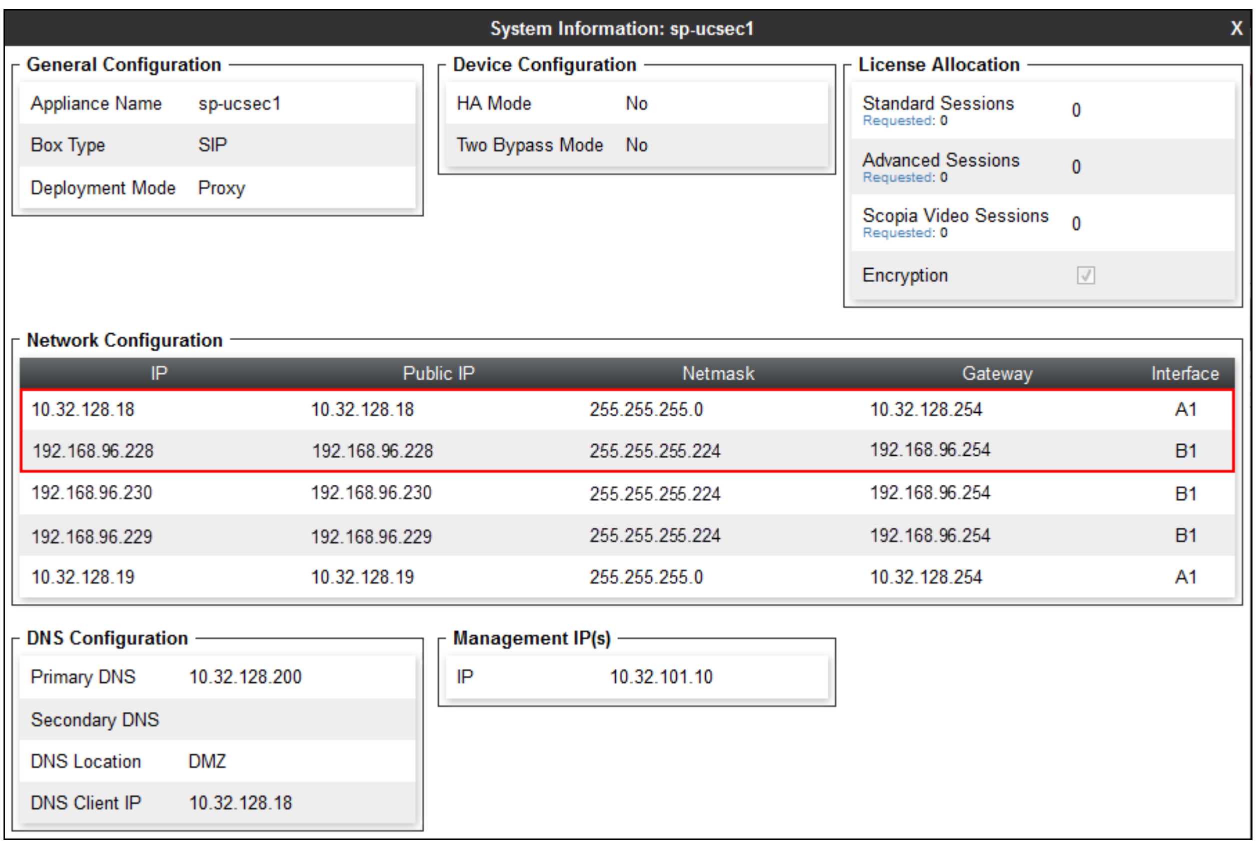
After logging in, the Dashboard screen will appear as shown below. All configuration screens of the Avaya SBCE are accessed by navigating the menu tree in the left pane.



7.2. Verify Network Configuration and Enable Interfaces

To view the network information provided during installation, navigate to **System Management**. In the right pane, click **View** highlighted below.

A System Information page will appear showing the information provided during installation. In the **Appliance Name** field is the name of the device (**sp-ucsec1**). This name will be referenced in other configuration screens. The two **Network Configuration** entries highlighted below are the only two IP addresses that are directly related to the SIP trunking solution described in these Application Notes. Interfaces **A1** and **B1** represent the private and public interfaces of the Avaya SBCE respectively. Each of these interfaces must be enabled after installation.



To enable the interfaces, first navigate to **Device Specific Settings → Network Management** in the left pane and select the device being managed in the center pane. In the right pane, click on the **Interfaces** tab. Verify the **Status** is **Enabled** for both the **A1** and **B1** interfaces. If not, click the status **Enabled/Disabled** to toggle the state of the interface.

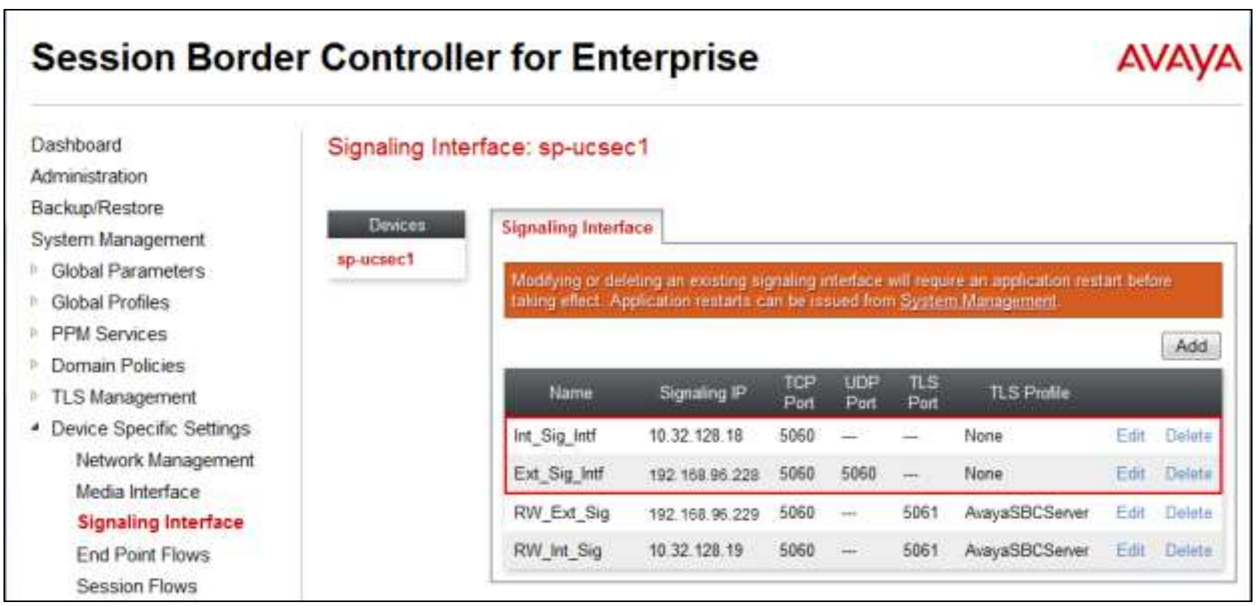
7.3. Signaling Interface

A signaling interface defines an IP address, protocols and listen ports that the Avaya SBCE can use for signaling. Create a signaling interface for both the internal and external sides of the Avaya SBCE.

To create a new interface, navigate to **Device Specific Settings → Signaling Interface** in the left pane. In the center pane, select the Avaya SBCE device (**sp-ucsec1**) to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by series of pop-up windows in which the interface parameters can be configured. Once complete, the settings are shown in the far right pane.

For the compliance test, signaling interface **Int_Sig_Intf** was created for the Avaya SBCE internal interface and signaling interface **Ext_Sig_Intf** was created for the Avaya SBCE external interface. Each is highlighted below. When configuring the interfaces, configure the parameters as follows:

- Set **Name** to a descriptive name.
- For the internal interface, set the **Signaling IP** to the IP address associated with the private interface (A1) defined in **Section 7.2**. For the external interface, set the **Signaling IP** to the IP address associated with the public interface (B1) defined in **Section 7.2**.
- In the **UDP Port**, **TCP Port** and **TLS Port** fields, enter the port the Avaya SBCE will listen on for each transport protocol. For the internal interface, the Avaya SBCE was configured to listen for TCP on port 5060. For the external interface, the Avaya SBCE was configured to listen for UDP or TCP on port 5060. Since TELUS uses UDP on port 5060, it would have been sufficient to simply configure the Avaya SBCE for UDP.



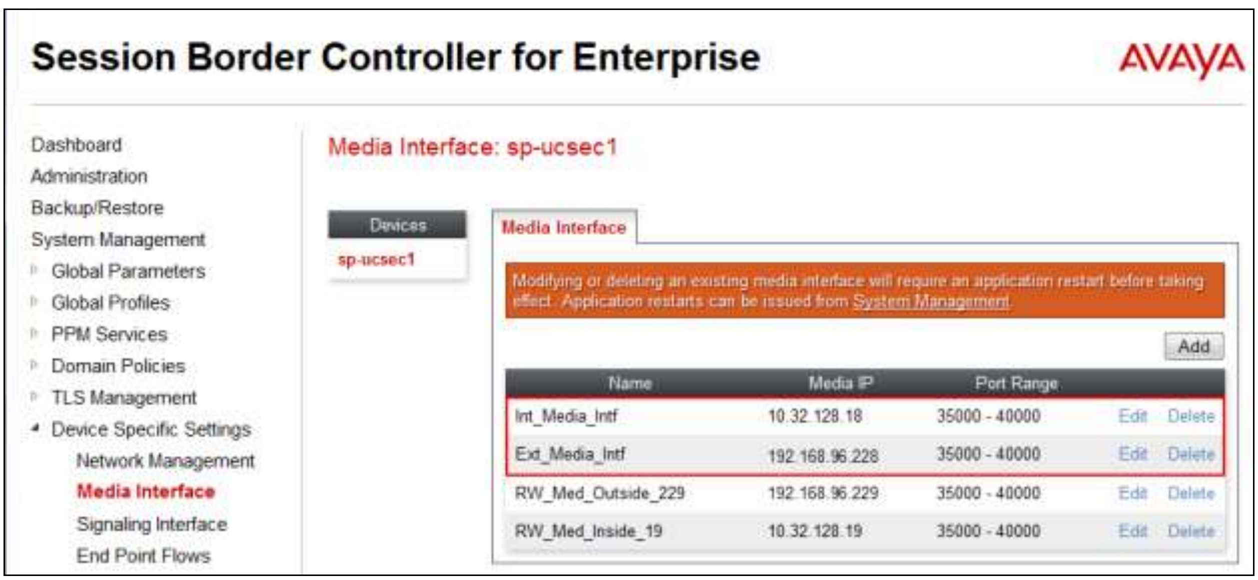
7.4. Media Interface

A media interface defines an IP address and port range for transmitting media. Create a media interface for both the internal and external sides of the Avaya SBCE.

To create a new interface, navigate to **Device Specific Settings → Media Interface** in the left pane. In the center pane, select the Avaya SBCE device (**sp-ucsec1**) to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by series of pop-up windows in which the interface parameters can be configured. Once complete, the settings are shown in the far right pane.

For the compliance test, media interface **Int_Media_Intf** was created for the Avaya SBCE internal interface and media interface **Ext_Media_Intf** was created for the Avaya SBCE external interface. Each is highlighted below. When configuring the interfaces, configure the parameters as follows:

- Set **Name** to a descriptive name.
- For the internal interface, set the **Media IP** to the IP address associated with the private interface (A1) defined in **Section 7.2**. For the external interface, set the **Media IP** to the IP address associated with the public interface (B1) defined in **Section 7.2**.
- Set **Port Range** to a range of ports acceptable to both the Avaya SBCE and the far-end. For the compliance test, the default port range was used for both interfaces.



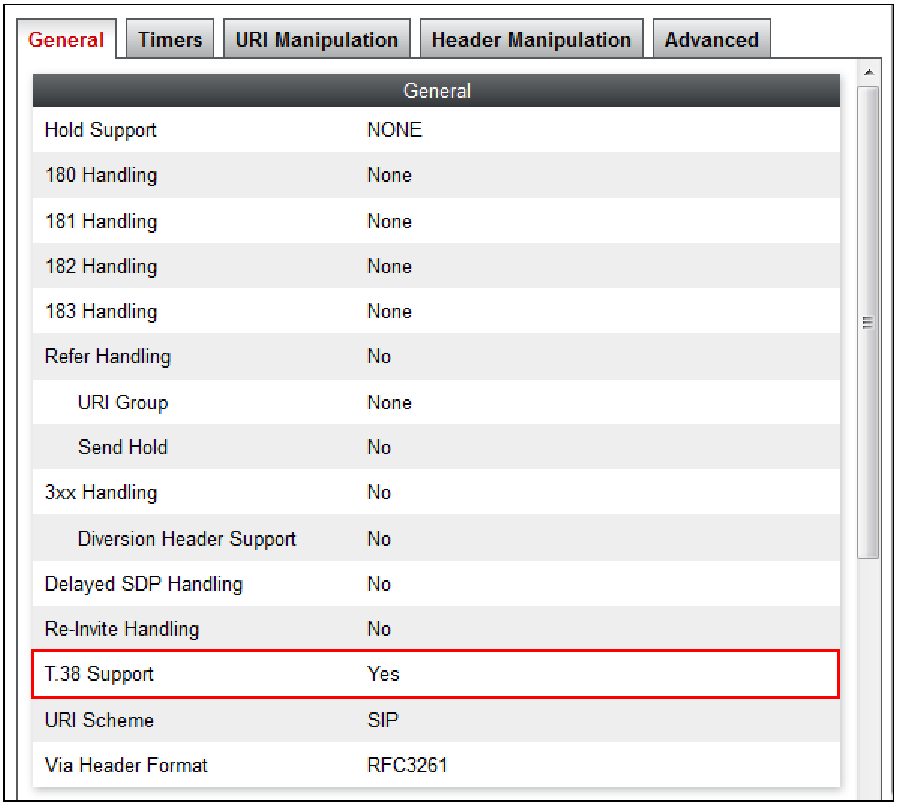
7.5. Server Interworking

A server interworking profile defines a set of parameters that aid in interworking between the Avaya SBCE and a connected server. Create a server interworking profile for Session Manager and the service provider SIP server. These profiles will be applied to the appropriate server in **Sections 7.7.1** and **7.7.2**.

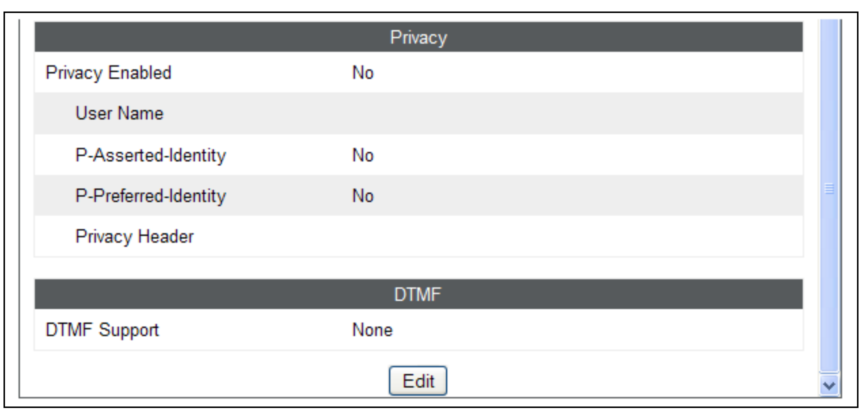
To create a new profile, navigate to **Global Profiles → Server Interworking** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by series of pop-up windows in which the profile parameters can be configured. Once complete, the settings are shown in the far right pane. Alternatively, a new profile may be created by selecting an existing profile in the center pane and clicking the **Clone** button in the right pane. This will create a copy of the selected profile which can then be edited as needed. To view the settings of an existing profile, select the profile from the center pane. The settings will appear in the right pane.

7.5.1. Server Interworking – Session Manager

For the compliance test, server interworking profile **PkwySM** was created for Session Manager by cloning the existing profile **avaya-ru**. The highlighted values are values that differ from the default. The **General** tab parameters are shown below. **T.38 Support** is set to **Yes** since TELUS supports T.38 fax.

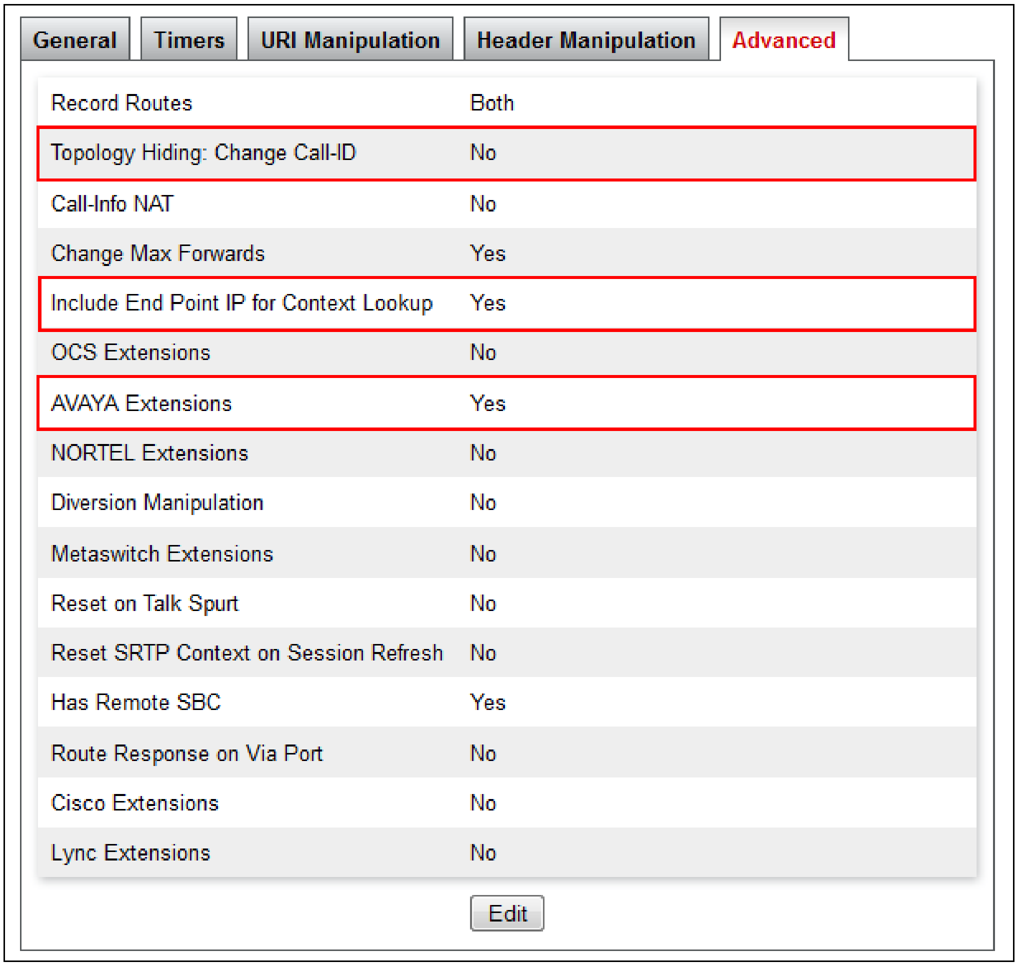


Scroll down to see the rest of the **General** tab.



The **Timers**, **URI Manipulation**, **Header Manipulation** tabs have no entries.

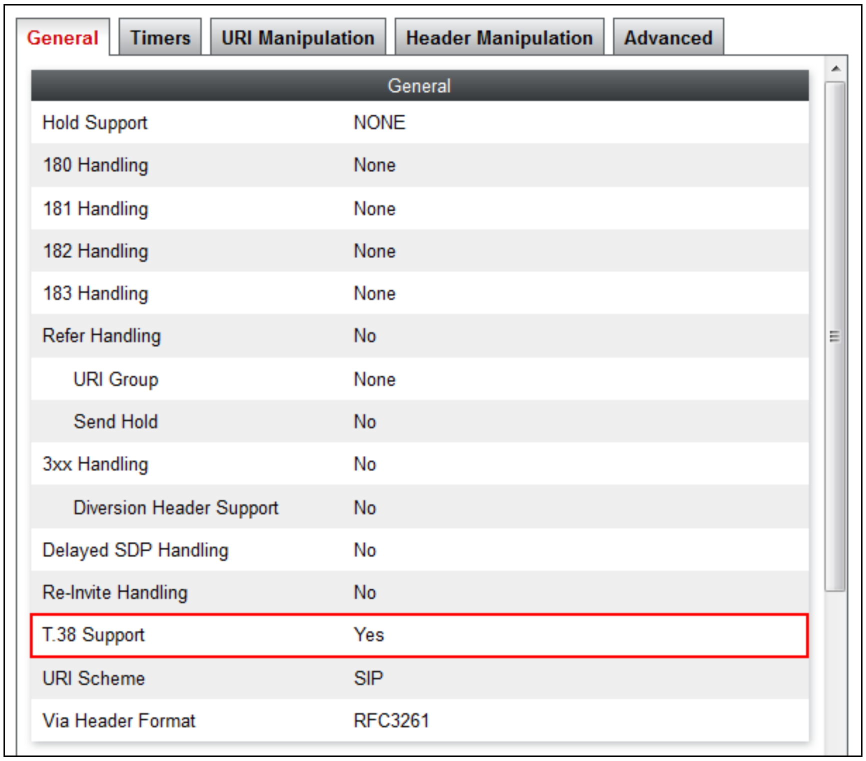
The **Advanced** tab parameters are shown below.



Record Routes	Both
Topology Hiding: Change Call-ID	No
Call-Info NAT	No
Change Max Forwards	Yes
Include End Point IP for Context Lookup	Yes
OCS Extensions	No
AVAYA Extensions	Yes
NORTEL Extensions	No
Diversion Manipulation	No
Metaswitch Extensions	No
Reset on Talk Spurt	No
Reset SRTP Context on Session Refresh	No
Has Remote SBC	Yes
Route Response on Via Port	No
Cisco Extensions	No
Lync Extensions	No

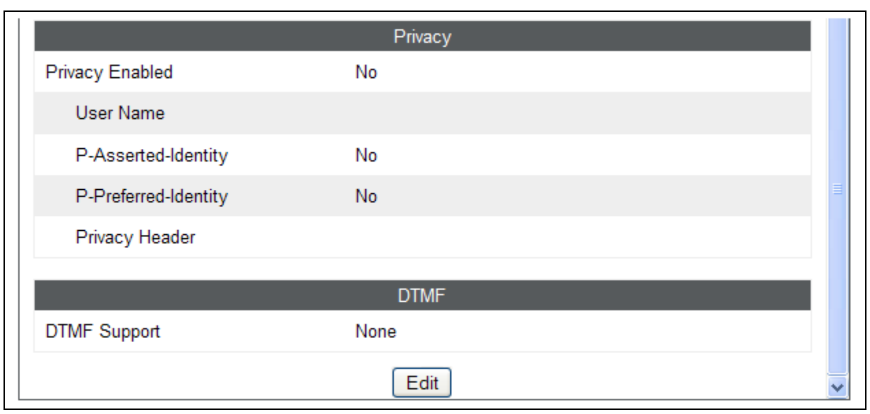
7.5.2. Server Interworking – TELUS

For the compliance test, server interworking profile **SP-General-T38** was created for the TELUS SIP server. When creating the profile, the default values were used for all parameters with the exception of **T.38 Support** which was set to **Yes**. The **General** tab parameters are shown below.



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

Scroll down to see the rest of the **General** tab.

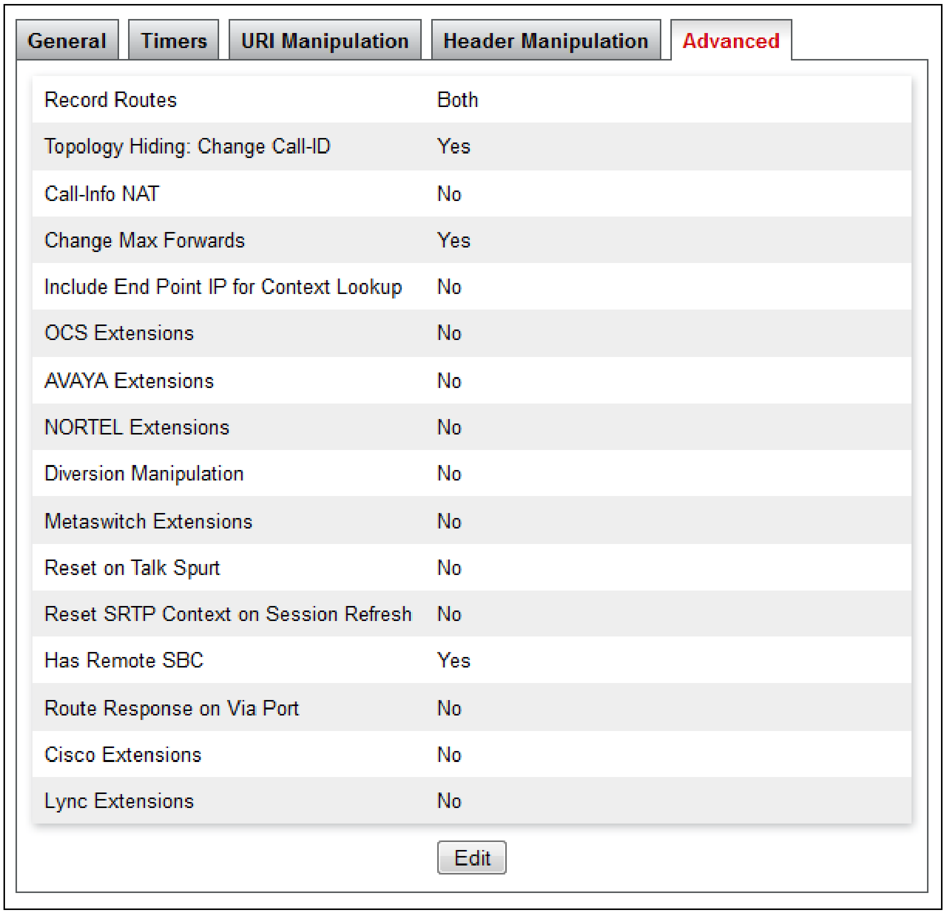


Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

DTMF	
DTMF Support	None

The **Timers**, **URI Manipulation**, **Header Manipulation** tabs have no entries.

The **Advanced** tab parameters are shown below.

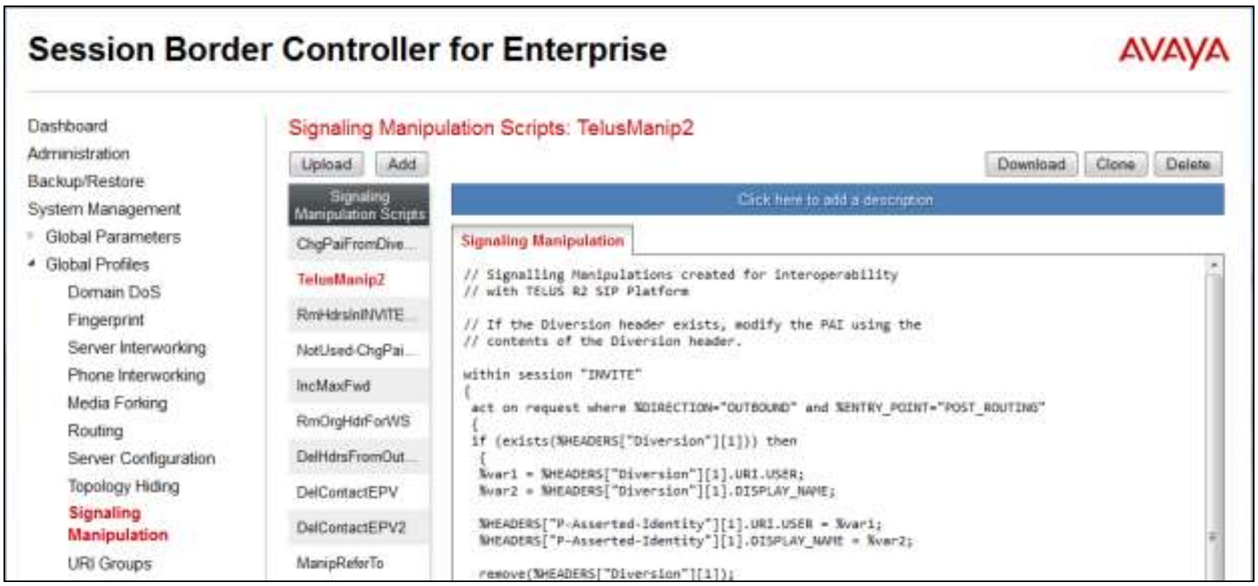


Record Routes	Both
Topology Hiding: Change Call-ID	Yes
Call-Info NAT	No
Change Max Forwards	Yes
Include End Point IP for Context Lookup	No
OCS Extensions	No
AVAYA Extensions	No
NORTEL Extensions	No
Diversion Manipulation	No
Metaswitch Extensions	No
Reset on Talk Spurt	No
Reset SRTP Context on Session Refresh	No
Has Remote SBC	Yes
Route Response on Via Port	No
Cisco Extensions	No
Lync Extensions	No

7.6. Signaling Manipulation

Signaling manipulation scripts provides for the manipulation of SIP messages which cannot be done by other configuration within the Avaya SBCE. TELUS required the signaling manipulation script defined in **Section 7.6.1**. It is applied to the TELUS SIP server in **Section 7.7.2**.

To create a script, navigate to **Global Profiles → Signaling Manipulation** in the left pane. In the center pane, select **Add**. A script editor window (not shown) will appear in which the script can be entered line by line. The **Title** box at the top of the editor window (not shown) is where the name of the script is entered. Once complete, the script is shown in the far right pane. To view an existing script, select the script from the center pane. The settings will appear in the right pane as shown in the example below.



7.6.1. Signaling Manipulation Script – TELUS

For the compliance test, signaling manipulation script **TelusManip2** was created for the TELUS SIP server. The script contains two manipulations. The first checks to see if a Diversion header is present in the outbound INVITE, and if so it will overwrite the user and display name in the PAI header with the contents of the Diversion Header. This is necessary for call forwarding and EC500. In these scenarios, TELUS expects the information provided by Communication Manager in the Diversion header to be present in the PAI. The script instructions to perform this manipulation are shown below.

Signaling Manipulation

```
// Signalling Manipulations created for interoperability
// with TELUS R2 SIP Platform

// If the Diversion header exists, modify the PAI using the
// contents of the Diversion header.

within session "INVITE"
{
 act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
 {
 if (exists(%HEADERS["Diversion"][1])) then
  {
  %var1 = %HEADERS["Diversion"][1].URI.USER;
  %var2 = %HEADERS["Diversion"][1].DISPLAY_NAME;

  %HEADERS["P-Asserted-Identity"][1].URI.USER = %var1;
  %HEADERS["P-Asserted-Identity"][1].DISPLAY_NAME = %var2;

  remove(%HEADERS["Diversion"][1]);
  }
 }
}
```

The second manipulation is in the same script file as the first and is shown below. It sets the Max-Forwards value to 0 in the outbound OPTIONS messages. This is a TELUS requirement. The complete file is shown in **Appendix A**.

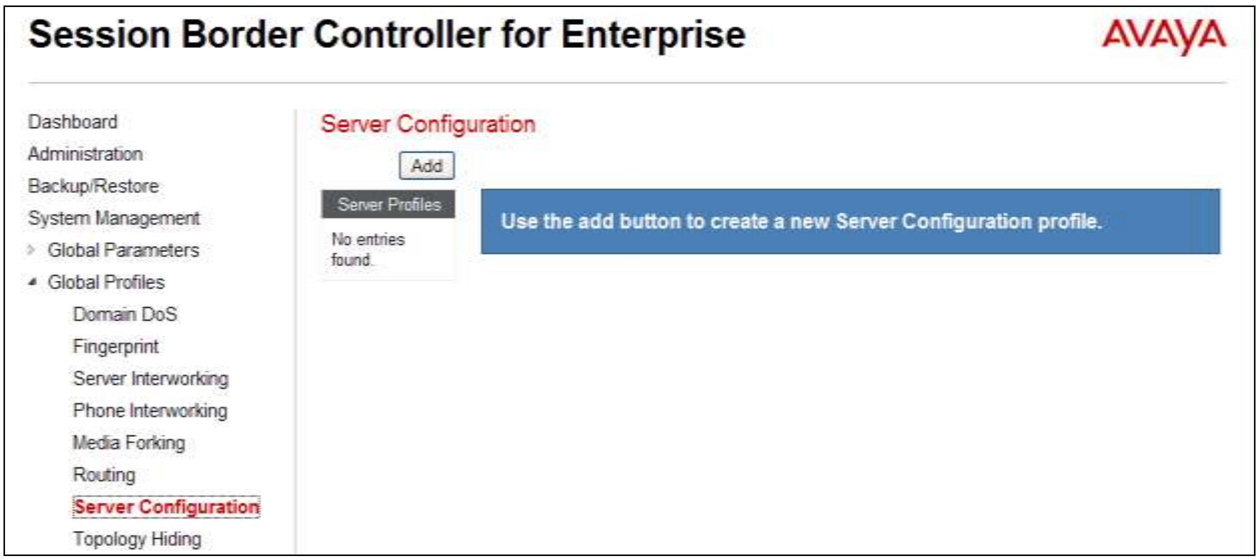
```
// Set Max-Forwards header to 0 in OPTIONS

within session "OPTIONS"
{
 act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
 {
  if (exists(%HEADERS["Max-Forwards"][1])) then
  {
  %HEADERS["Max-Forwards"][1] = "0";
  }
 }
}
```

Edit

7.7. Server Configuration

A server configuration profile defines the attributes of the physical server. Create a server configuration profile for Session Manager and the service provider SIP server.

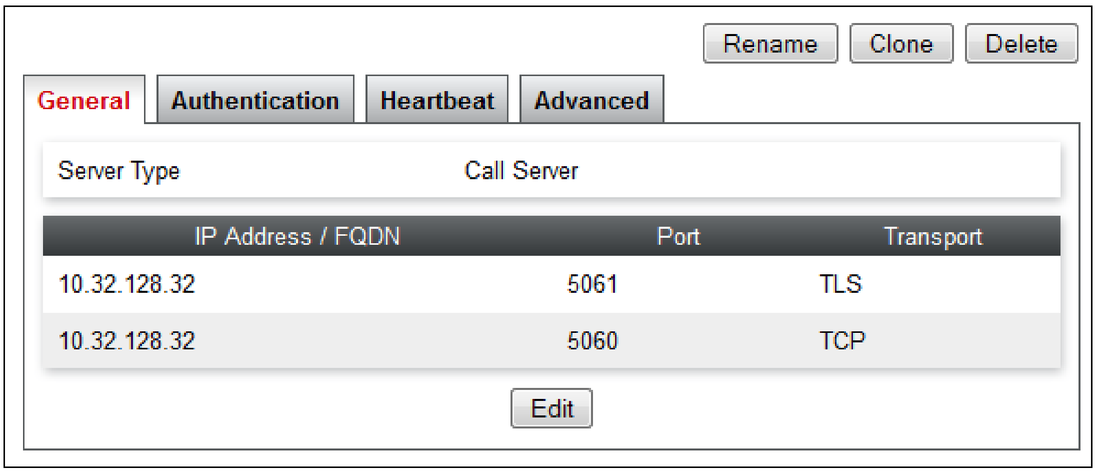
To create a new profile, navigate to **Global Profiles → Server Configuration** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by series of pop-up windows in which the profile parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing profile, select the profile from the center pane. The settings will appear in the right pane.



7.7.1. Server Configuration – Session Manager

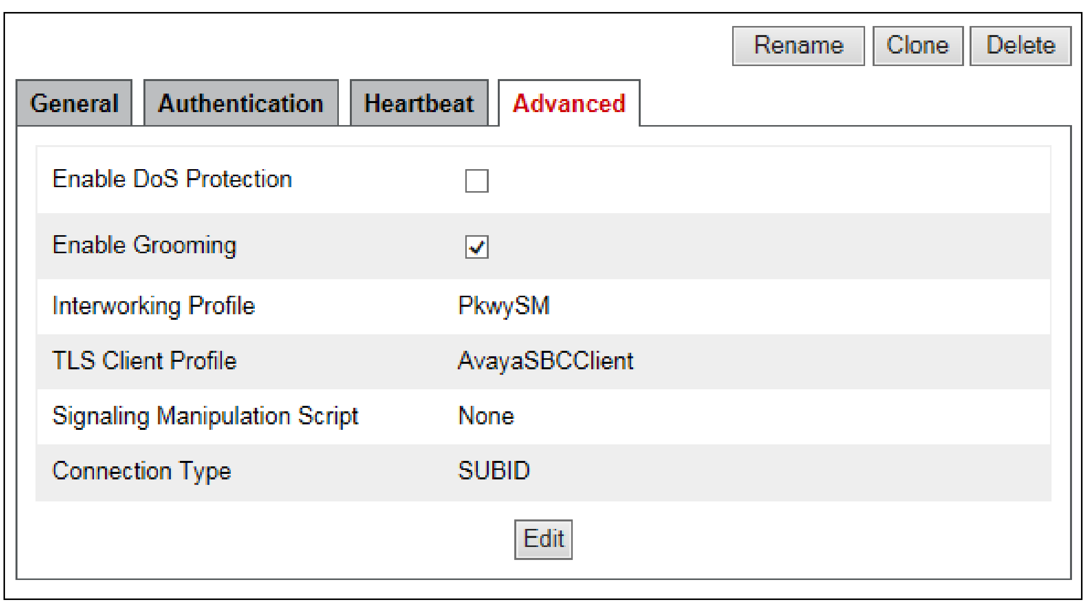
For the compliance test, server configuration profile **Pkwy-SM** was created for Session Manager. When creating the profile, configure the **General** tab parameters as follows:

- Set **Server Type** to **Call Server.**
- Enter a valid combination of **IP Address / FQDN**, **Port** and **Transport** that Session Manager will use to listen for SIP requests. The standard SIP UDP/TCP port is 5060. The standard SIP TLS port is 5061. Additional combinations can be entered by clicking the **Add** button (not shown).



The **Authentication** and **Heartbeat** tabs have no entries.

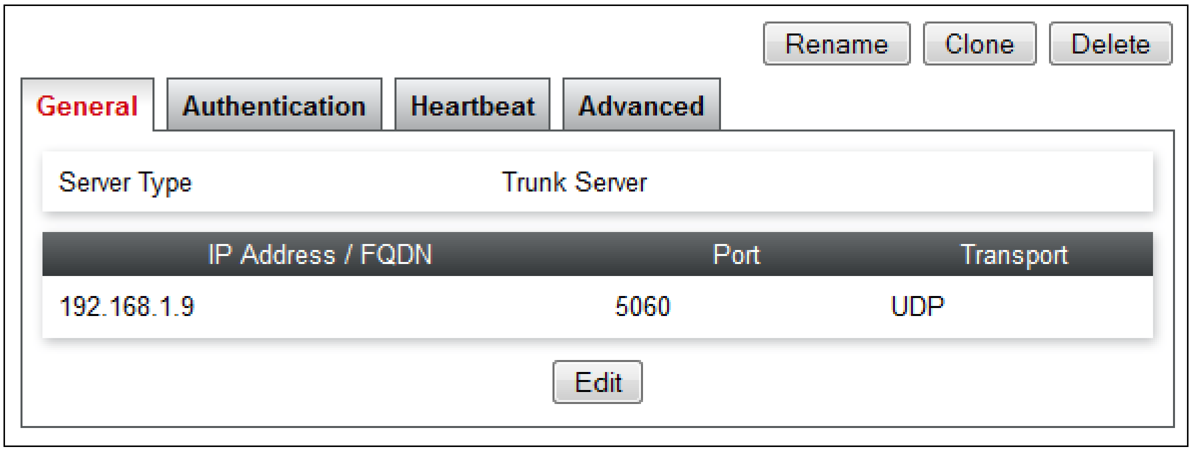
On the **Advanced** tab, check **Enable Grooming** and set the **Interworking Profile** field to the interworking profile for Session Manager defined in **Section 7.5.1**.



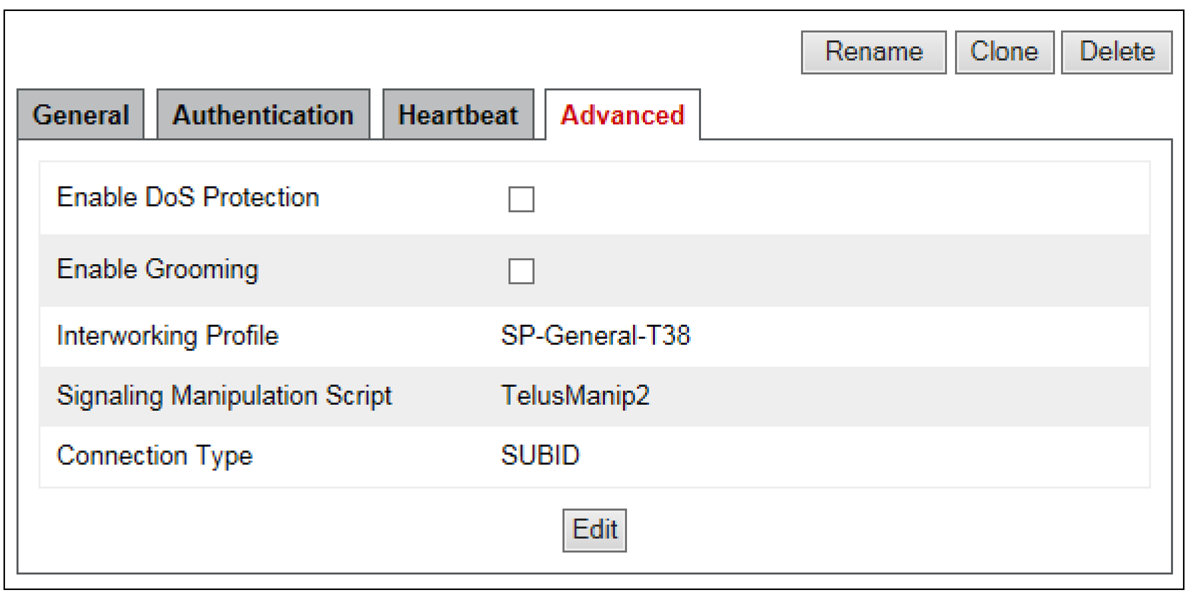
7.7.2. Server Configuration – TELUS

For the compliance test, server configuration profile **SP-TELUS2** was created for TELUS. When creating the profile, configure the **General** tab parameters as follows:

- Set **Server Type** to **Trunk Server**.
- Enter a valid combination of **IP Address / FQDN**, **Port** and **Transport** that the TELUS SIP proxy will use to listen for SIP requests. The standard SIP UDP/TCP port is 5060. Additional combinations can be entered by clicking the **Add** button (not shown).



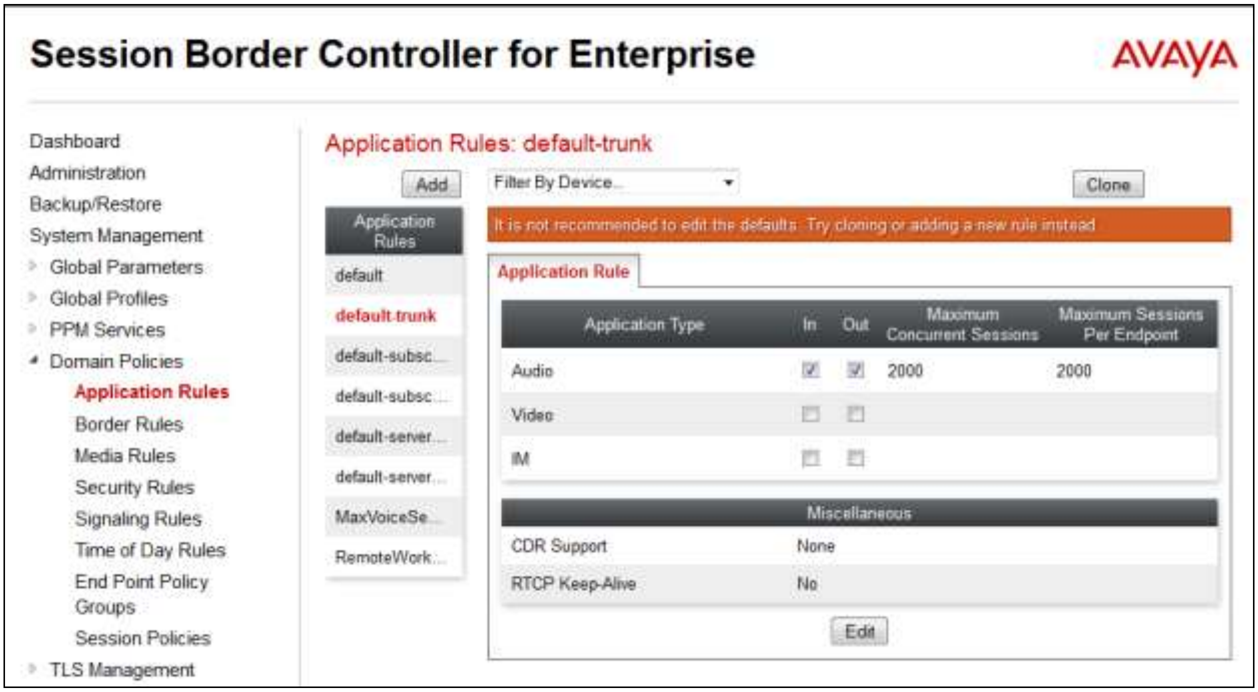
On the **Advanced** tab, set the **Interworking Profile** field to the interworking profile for TELUS defined in **Section 7.5.2**. Set the **Signaling Manipulation Script** field to the script created for TELUS in **Section 7.6.1**.



7.8. Application Rules

An application rule defines the allowable SIP applications and associated parameters. An application rule is one component of the larger endpoint policy group defined in **Section 7.11**. For the compliance test, the predefined **default-trunk** application rule (shown below) was used for both Session Manager and the TELUS SIP server.

To view an existing rule, navigate to **Domain Policies → Application Rules** in the left pane. In the center pane, select the rule (e.g., **default-trunk**) to be viewed.



7.9. Media Rules

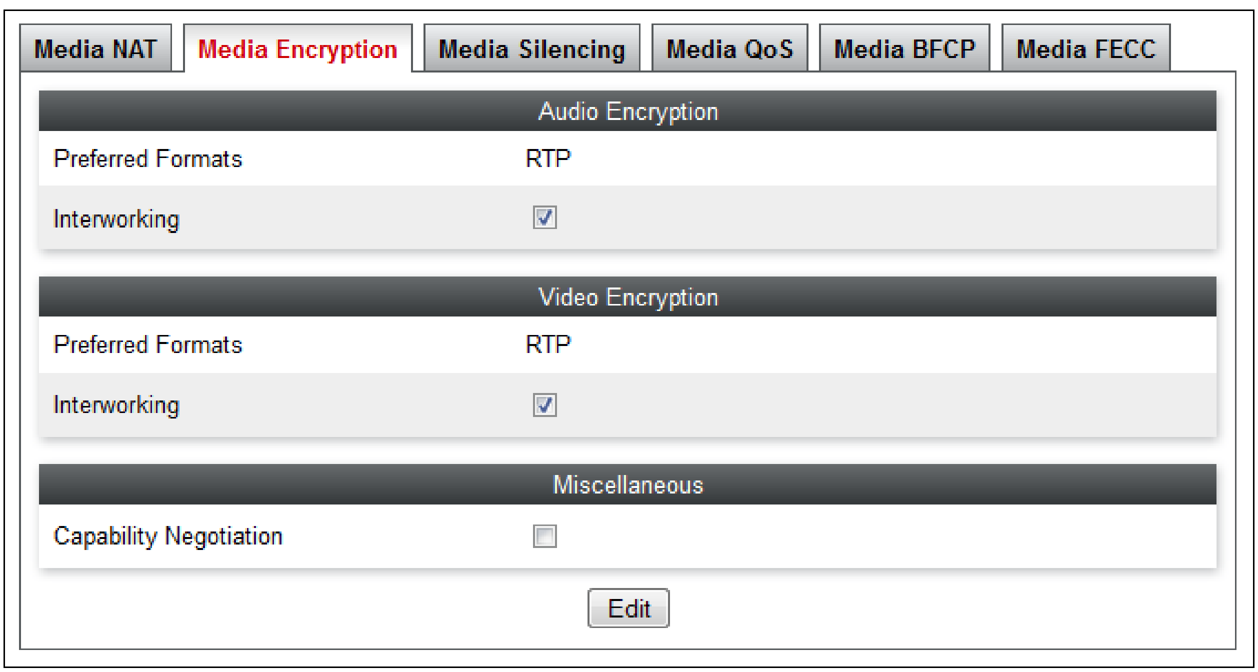
A media rule defines the processing to be applied to the selected media. A media rule is one component of the larger endpoint policy group defined in **Section 7.11**. For the compliance test, the predefined **default-low-med** media rule (shown below) was used for both Session Manager and the TELUS SIP server.

To view an existing rule, navigate to **Domain Policies → Media Rules** in the left pane. In the center pane, select the rule (e.g., **default-low-med**) to be viewed.

The contents of the **default-low-med** media rule are described below.

The **Media NAT** tab has no entries.

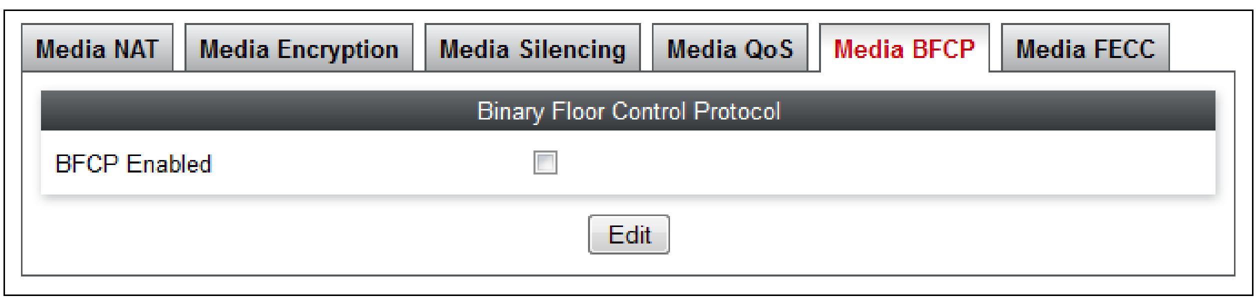
The **Media Encryption** tab indicates that no encryption was used.



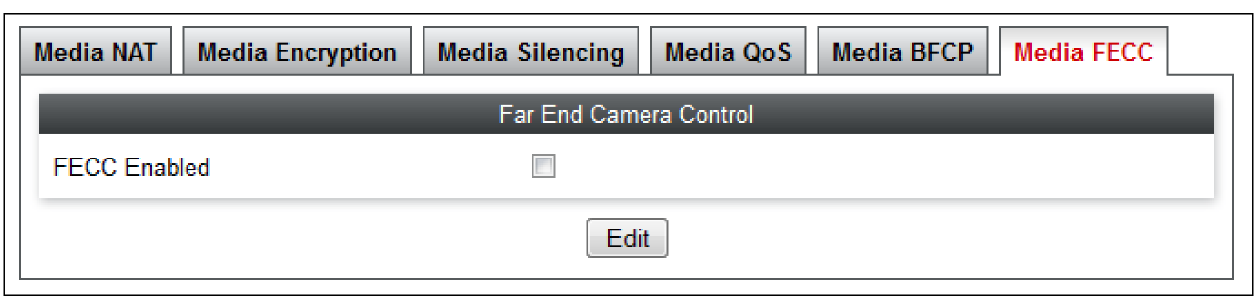
On the **Media Silencing** tab, **Media Silencing** is disabled.

The **Media QoS** settings are shown below.

On the **Media BFCP** tab, BFCP is disabled.



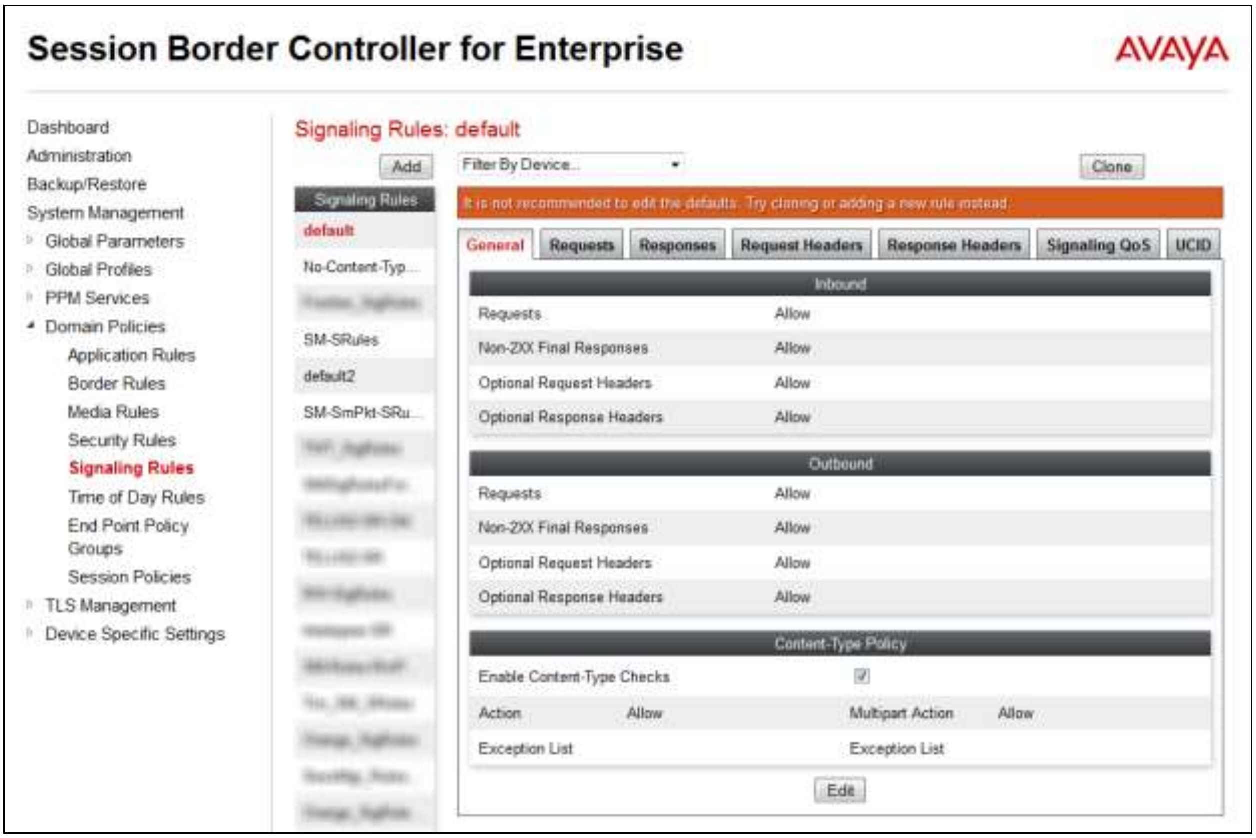
On the **Media FECC** tab, FECC is disabled.



7.10. Signaling Rules

A signaling rule defines the processing to be applied to the selected signaling traffic. A signaling rule is one component of the larger endpoint policy group defined in **Section 7.11**. A specific signaling rule was created for Session Manager and the TELUS SIP server.

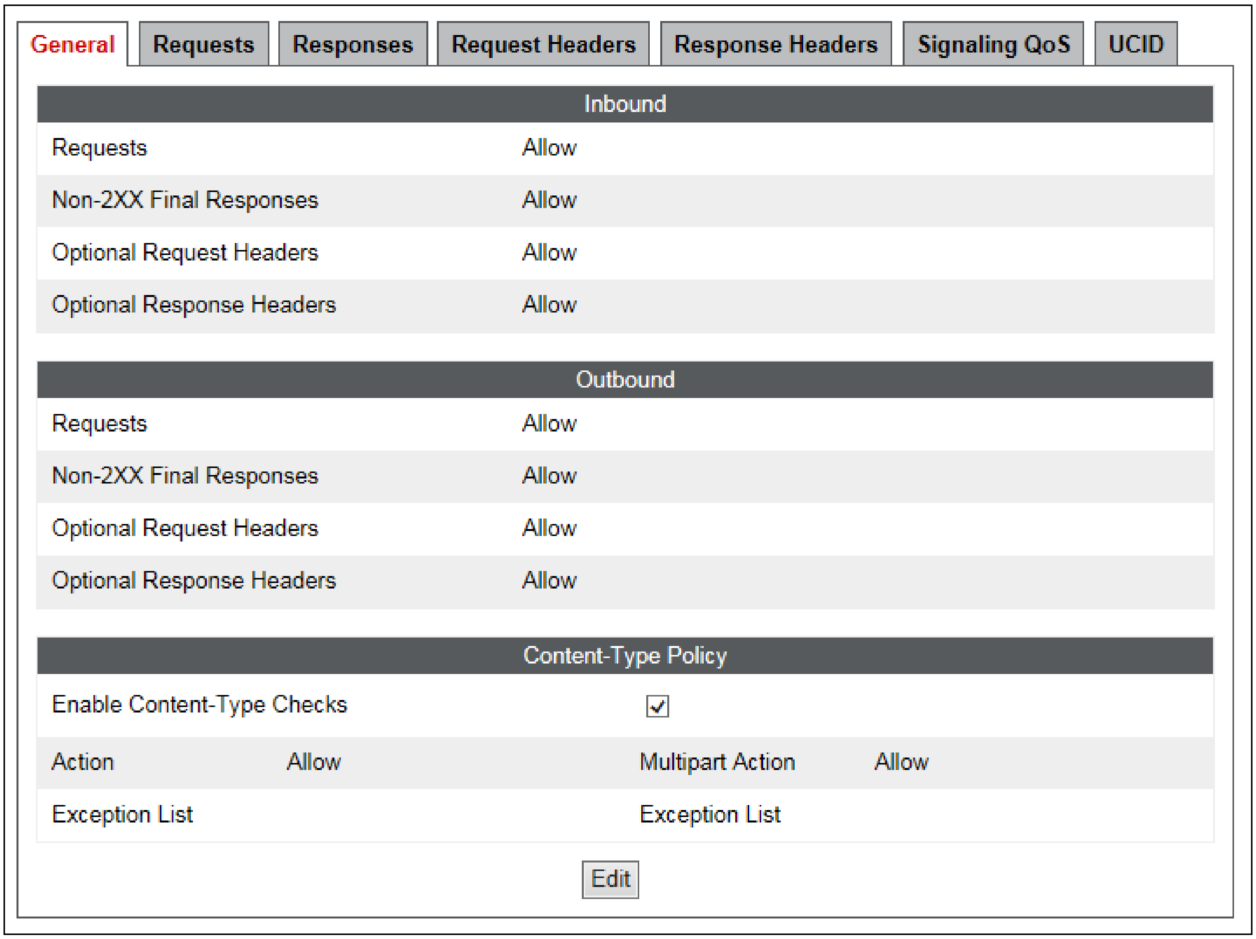
To create a new rule, navigate to **Domain Policies → Signaling Rules** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new rule, followed by series of pop-up windows in which the rule parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing rule, select the rule from the center pane. The settings will appear in the right pane.



7.10.1. Signaling Rules – Session Manager

For the compliance test, signaling rule **SM-SRules** was created for Session Manager to prevent some proprietary headers in the SIP messages, sent from the Session Manager, from being propagated to TELUS. A header was blocked if it contained internal addresses or other information about the internal network.

SM-SRules was created using the default values on all tabs except the **Request Headers**, **Response Headers**, and **Signaling QoS** tabs. The **General** tab settings are shown below.



The **Requests** and **Responses** tabs have no entries.

The **Request Headers** tab shows the manipulations performed on the headers of request messages such as the initial INVITE or UPDATE message. An entry is created by clicking the **Add In Header Control** or **Add Out Header Control** button depending on the direction (relative to the Avaya SBCE) of the message to be modified. Entries were created to perform the following actions:

1. Removes the **AV-Correlation-ID** header from **INVITE** messages in the **IN** direction (Session Manager to Avaya SBCE).
2. Removes the **Endpoint-View** header from **ALL** messages in the **IN** direction.

General | Requests | Responses | Request Headers | Response Headers | Signaling QoS | UCID

Add In Header Control | Add Out Header Control

Row	Header Name	Method Name	Header Criteria	Action	Proprietary	Direction		
1	AV-Correlation-ID	INVITE	Forbidden	Remove Header	Yes	IN	Edit	Delete
2	Endpoint-View	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete

Similarly, manipulations can be performed on the headers of SIP response messages. These can be viewed by selecting the **Response Header** tab as shown below. Entries were created in the same manner as was done on the **Request Headers** tab. The entries shown perform the following actions:

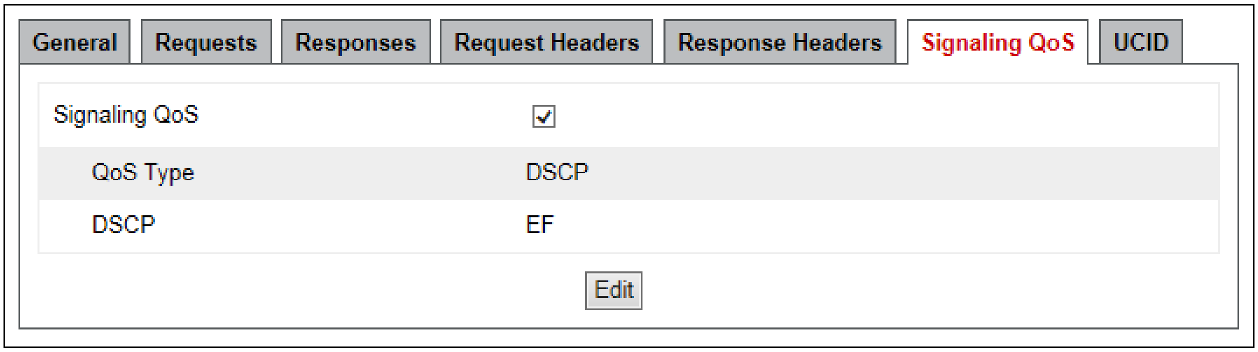
1. Removes the **Endpoint-View** header from any **2XX** response to **ALL** messages in the **IN** direction (Session Manager to Avaya SBCE).
2. Removes the **Endpoint-View** header from any **1XX** response to an **INVITE** message in the **IN** direction.

General | Requests | Responses | Request Headers | Response Headers | Signaling QoS | UCID

Add In Header Control | Add Out Header Control

Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction		
1	Endpoint-View	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
2	Endpoint-View	1XX	INVITE	Forbidden	Remove Header	Yes	IN	Edit	Delete

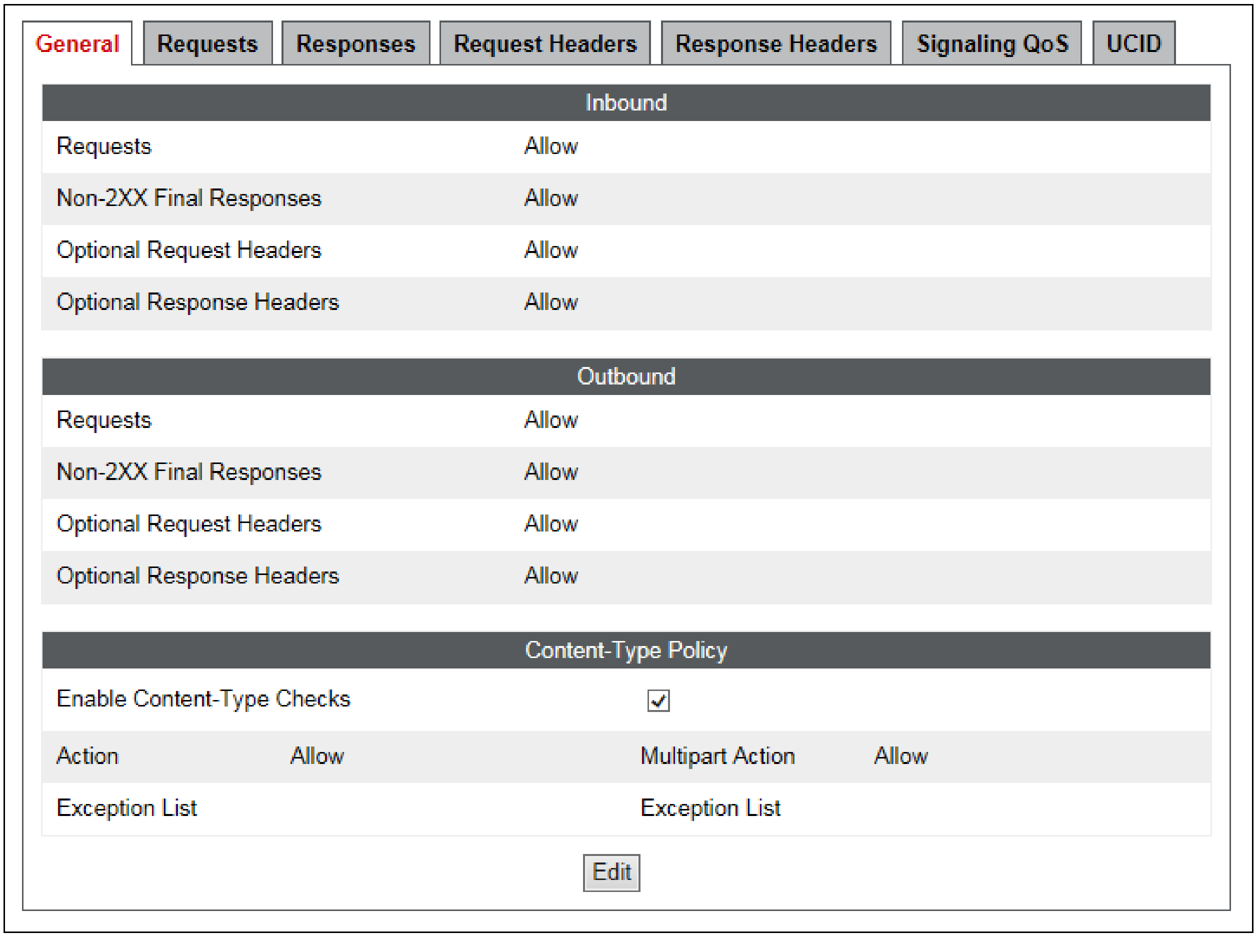
The **Signaling QoS** settings are shown below.



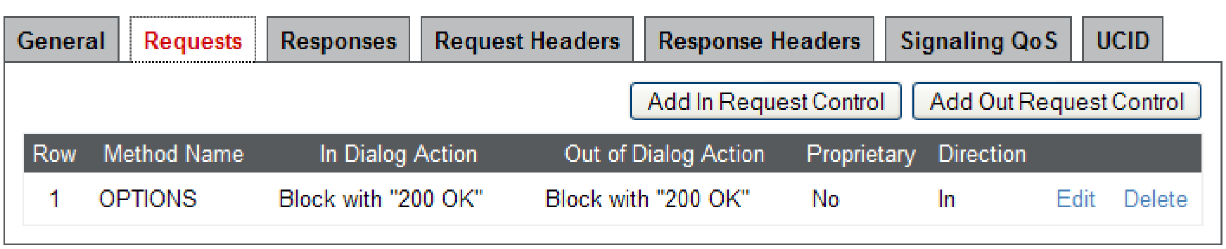
The **UCID** settings are shown below.

7.10.2. Signaling Rules – TELUS

The **TELUS2-SR** signaling rule (shown below) was used for the TELUS SIP server. The **General** tab settings use the default values and are shown below.

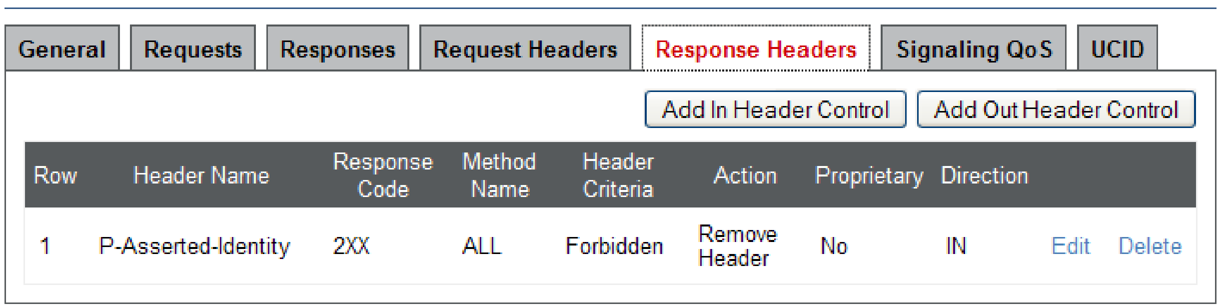


The **Requests** tab shows the actions performed on request messages. An entry is created by clicking the **Add In Header Control** or **Add Out Header Control** button depending on the direction (relative to the Avaya SBCE) of the message to be modified. The entry shown below blocks incoming OPTIONS messages and returns a 200 OK response. See **Section 2.2** for full details.

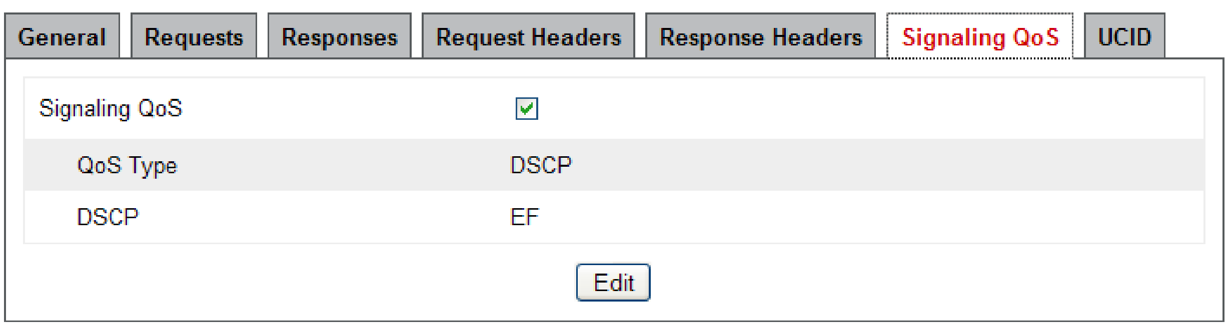


The **Responses** and **Requests Headers** tabs have no entries.

The **Response Header** tab shows the manipulations performed on headers of response messages. The entry was created in the same manner as was done on the **Request Headers** and **Response Headers** tabs in **Section 7.10.1**. The entry shown removes the P-Asserted-Identity header from any 2XX response to ALL messages in the IN direction (TELUS to Avaya SBCE). See **Section 2.2** for full details.



The **Signaling QoS** settings are shown below.



The **UCID** settings are shown below.

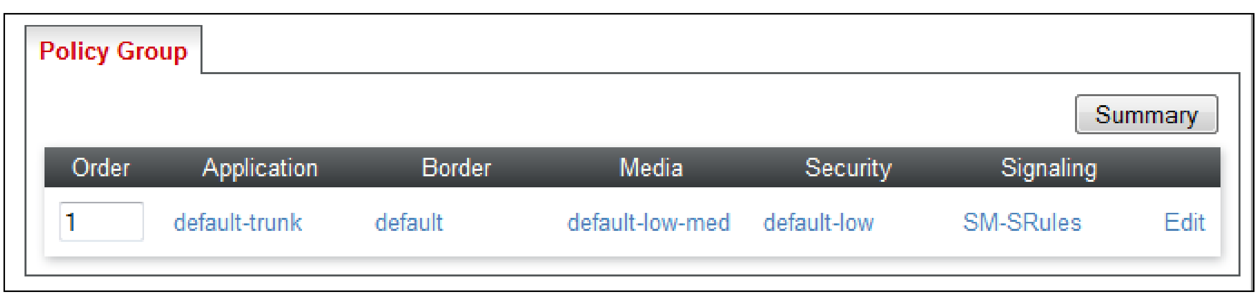
7.11. Endpoint Policy Groups

An endpoint policy group is a set of policies that will be applied to traffic between the Avaya SBCE and an endpoint (connected server). Thus, an endpoint policy group must be created for Session Manager and the service provider SIP server. The endpoint policy group is applied to the traffic as part of the endpoint flow defined in **Section 7.14**.

To create a new group, navigate to **Domain Policies → End Point Policy Groups** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new group, followed by series of pop-up windows in which the group parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing group, select the group from the center pane. The settings will appear in the right pane.

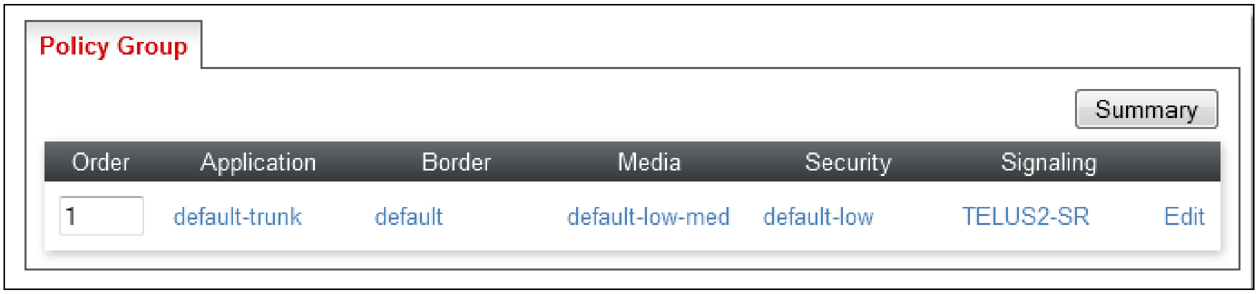
7.11.1. Endpoint Policy Group – Session Manager

For the compliance test, endpoint policy group **SM** was created for Session Manager. Default values were used for each of the rules which comprise the group with the exception of **Application** and **Signaling**. For **Application**, enter the application rule created in **Section 7.8**. For **Signaling**, enter the signaling rule created in **Section 7.10.1**. The details of the default settings for **Media** are showed in **Section 7.9**.



7.11.2. Endpoint Policy Group – TELUS

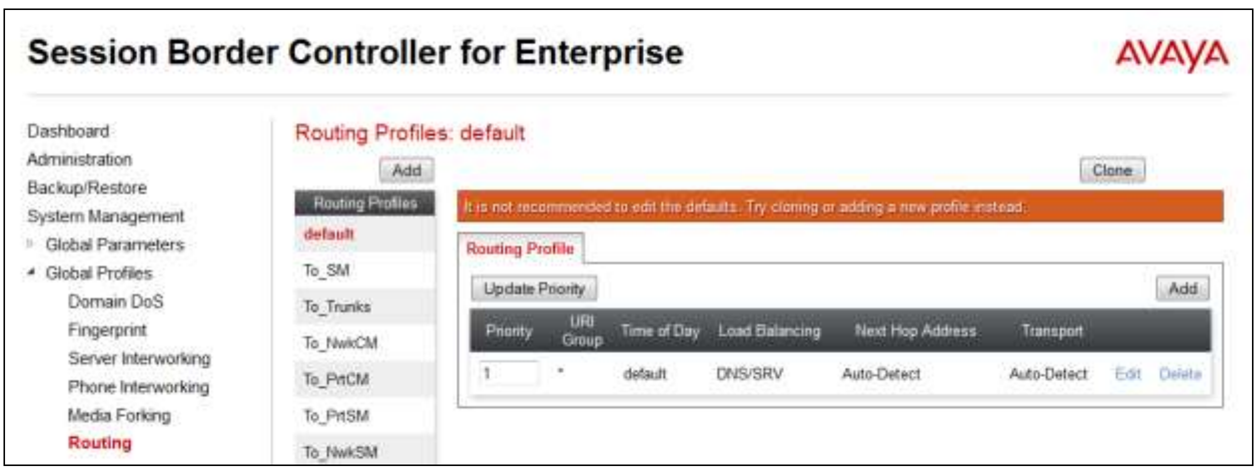
For the compliance test, endpoint policy group **TELUS2-PolicyGrp** was created for the TELUS SIP server. Default values were used for each of the rules which comprise the group with the exception of **Application** and **Signaling**. For **Application**, enter the application rule created in **Section 7.8**. For **Signaling**, enter the signaling rule created in **Section 7.10.2**. The details of the default settings for **Media** are showed in **Section 7.9**.



7.12. Routing

A routing profile defines where traffic will be directed based on the contents of the Request-URI. A routing profile is applied only after the traffic has matched an endpoint server flow defined in **Section 7.14**. Create a routing profile for Session Manager and the service provider SIP server.

To create a new profile, navigate to **Global Profiles → Routing** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by series of pop-up windows in which the profile parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing profile, select the profile from the center pane. The settings will appear in the right pane.

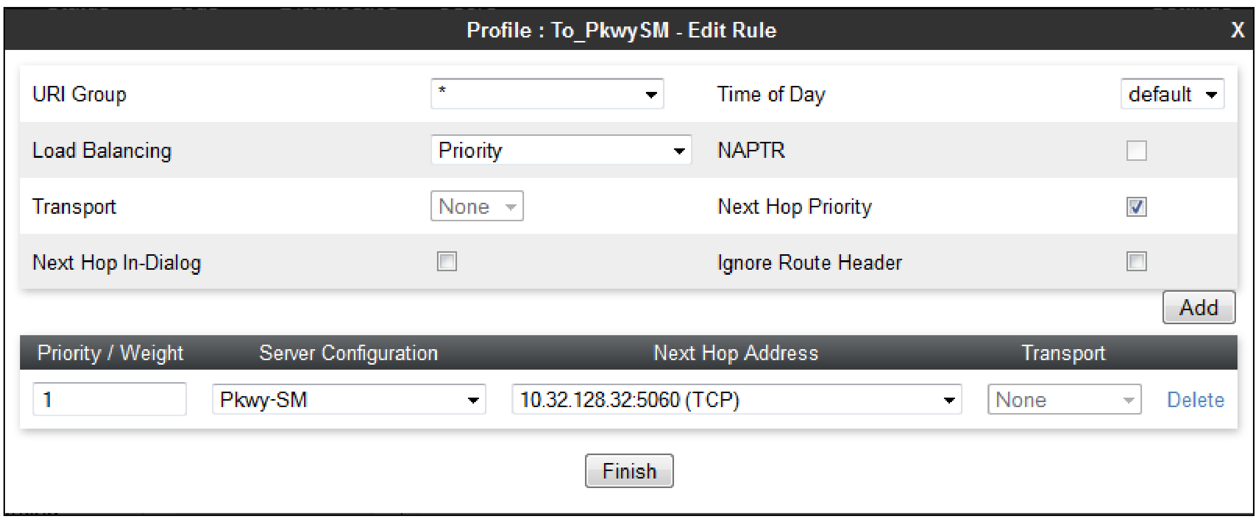


7.12.1. Routing – Session Manager

For the compliance test, routing profile **To_PkwySM** was created for Session Manager. When creating the profile, configure the parameters as follows:

- Set the **URI Group** to the wild card ***** to match on any URI.
- Set **Load Balancing** to **Priority** from the pull-down menu.
- Enable **Next Hop Priority**.
- Click **Add** to enter the following for the Next Hop Address:
  - Set **Priority/Weight** to **1**.
  - For **Server Configuration**, select **Pkwy-SM** from the pull-down menu. The **Next Hop Address** will be filled-in automatically.

Click **Finish**.

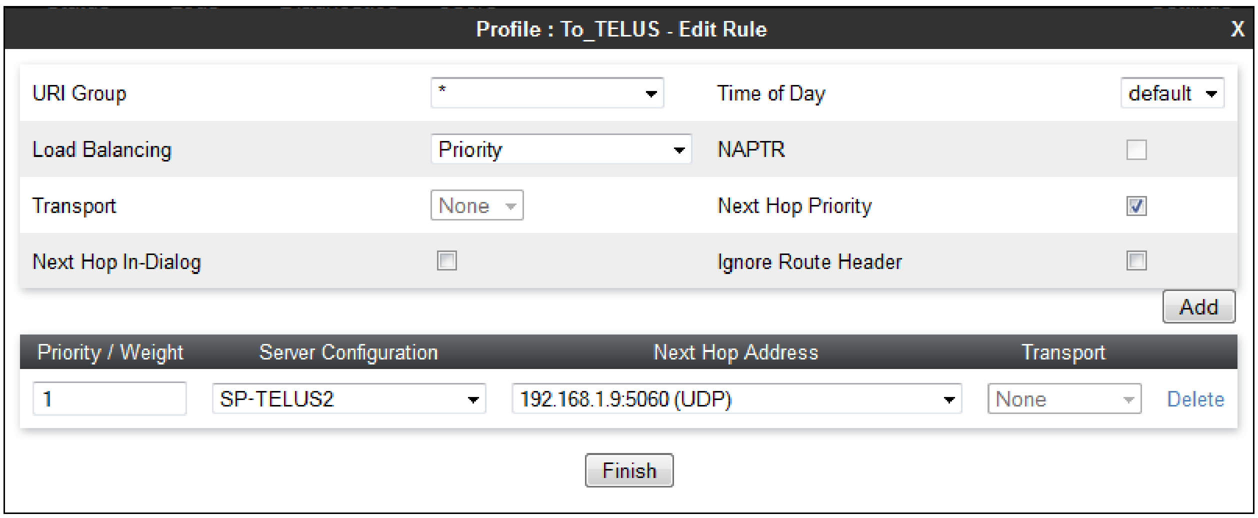


7.12.2. Routing – TELUS

For the compliance test, routing profile **To_TELUS** was created for TELUS. When creating the profile, configure the parameters as follows:

- Set the **URI Group** to the wild card ***** to match on any URI.
- Set **Load Balancing** to **Priority** from the pull-down menu.
- Enable **Next Hop Priority**.
- Click **Add** to enter the following for the Next Hop Address:
    - Set **Priority/Weight** to **1**.
    - For **Server Configuration**, select **SP-TELUS2** from the pull-down menu. The **Next Hop Address** will be filled-in automatically.

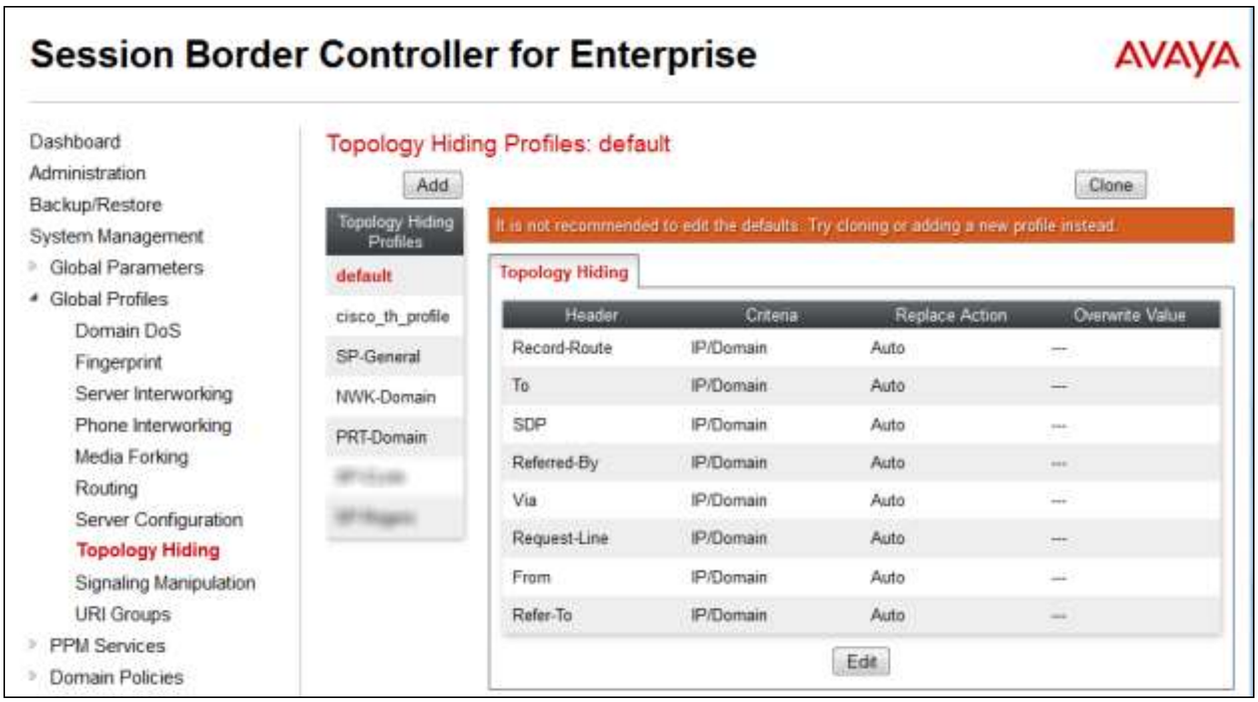
Click **Finish**.



7.13. Topology Hiding

Topology hiding allows the host part of some SIP message headers to be modified in order to prevent private network information from being propagated to the untrusted public network. It can also be used as an interoperability tool to adapt the host portion of these same headers to meet the requirements of the connected servers. The topology hiding profile is applied as part of the endpoint flow in **Section 7.14**.

To create a new profile, navigate to **Global Profiles → Topology Hiding** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by a pop-up window in which a header can be selected and configured. Additional headers can be added in this window. Once complete, the settings are shown in the far right pane. To view the settings of an existing profile (e.g., **default**), select the profile from the center pane. The settings will appear in the right pane.



7.13.1. Topology Hiding – Session Manager

For the compliance test, topology hiding profile **PRT-Domain** was created for Session Manager. This profile will be applied to traffic from the Avaya SBCE to Session Manager. When creating the profile, configure the parameters as follows:

- Set **Header** to the header whose host part of the URI is to be modified.
- Set **Criteria** to **IP/Domain** to indicate that the host part should be modified if it is an IP address or a domain.
- Set **Replace Action** to **Auto** for all headers except **Request-Line**, **Referred-By**, **Refer-To**, **From** and **To** which should be set to **Overwrite**.
- For those headers to be overwritten, the **Overwrite Value** is set to the enterprise domain (**avaya.com**).

Topology Hiding

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

Edit

7.13.2. Topology Hiding – TELUS

For the compliance test, topology hiding profile **SP-General** was created for TELUS. This profile will be applied to traffic from the Avaya SBCE to TELUS. When creating the profile, configure the parameters as follows:

- Set **Header** to the header whose host part of the URI is to be modified.
- Set **Criteria** to **IP/Domain** to indicate that the host part should be modified if it is an IP address or a domain.
- Set **Replace Action** to **Auto** for all headers.

Topology Hiding

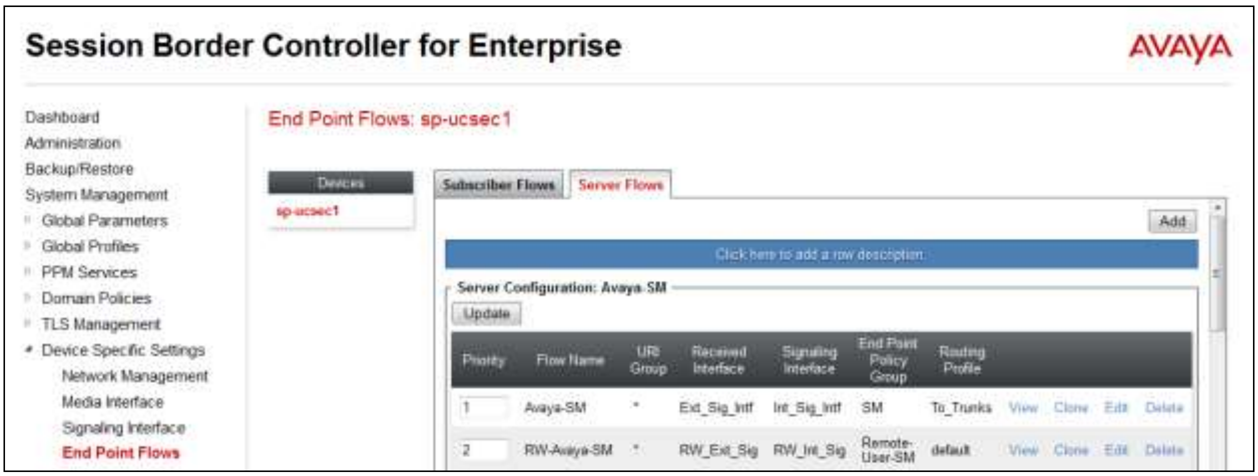
Header	Criteria	Replace Action	Overwrite Value
Record-Route	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Request-Line	IP/Domain	Auto	---
Referred-By	IP/Domain	Auto	---
To	IP/Domain	Auto	---
From	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
Via	IP/Domain	Auto	---

Edit

7.14. End Point Flows

Endpoint flows are used to determine the endpoints (connected servers) involved in a call in order to apply the appropriate policies. When a packet arrives at the Avaya SBCE, the content of the packet (IP addresses, URIs, etc) is used to determine which flow it matches. Once the flow is determined, the flow points to policies and profiles which control processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for the destination endpoint are applied. Thus, two flows are involved in every call: the source endpoint flow and the destination endpoint flow. In the case of the compliance test, the endpoints are Session Manager and the service provider SIP server.

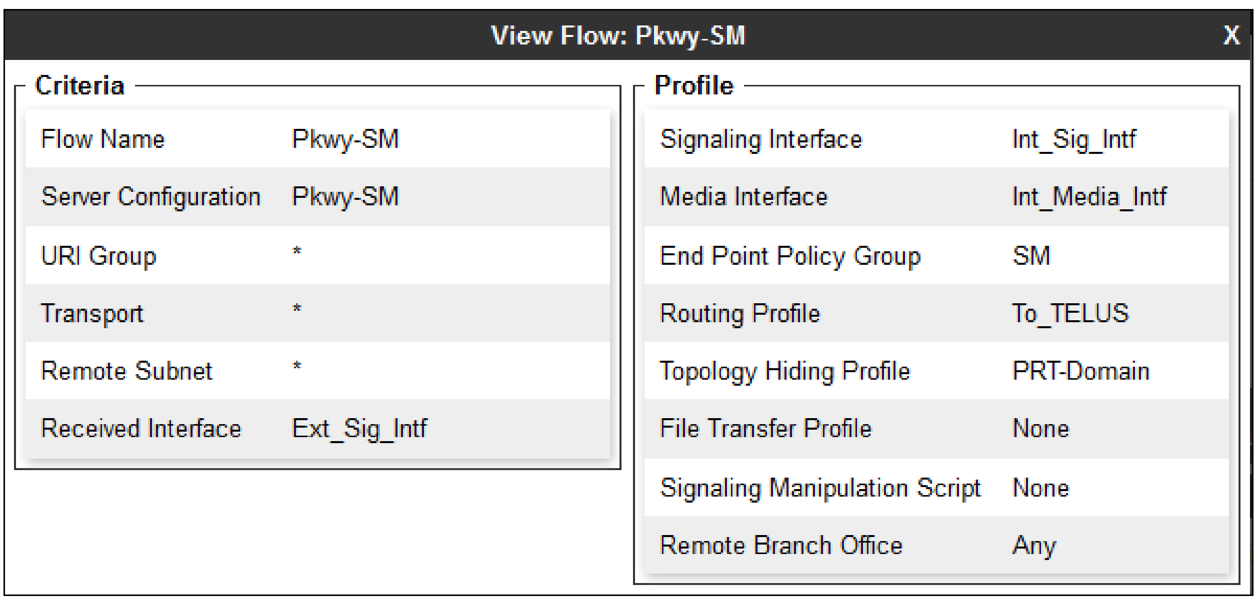
To create a new flow for a server endpoint, navigate to **Device Specific Settings → End Point Flows** in the left pane. In the center pane, select the Avaya SBCE device (**sp-ucsec1**) to be managed. In the right pane, select the **Server Flows** tab and click the **Add** button. A pop-up window (not shown) will appear requesting the name of the new flow and the flow parameters. Once complete, the settings are shown in the far right pane.



7.14.1. End Point Flow – Session Manager

For the compliance test, endpoint flow **Pkwy-SM** was created for Session Manager. All traffic from Session Manager will match this flow as the source flow and use the specified **Routing Profile To_TELUS** to determine the destination server and corresponding destination flow. The **End Point Policy** and **Topology Hiding Profile** will be applied as appropriate. When creating the flow, configure the parameters as follows:

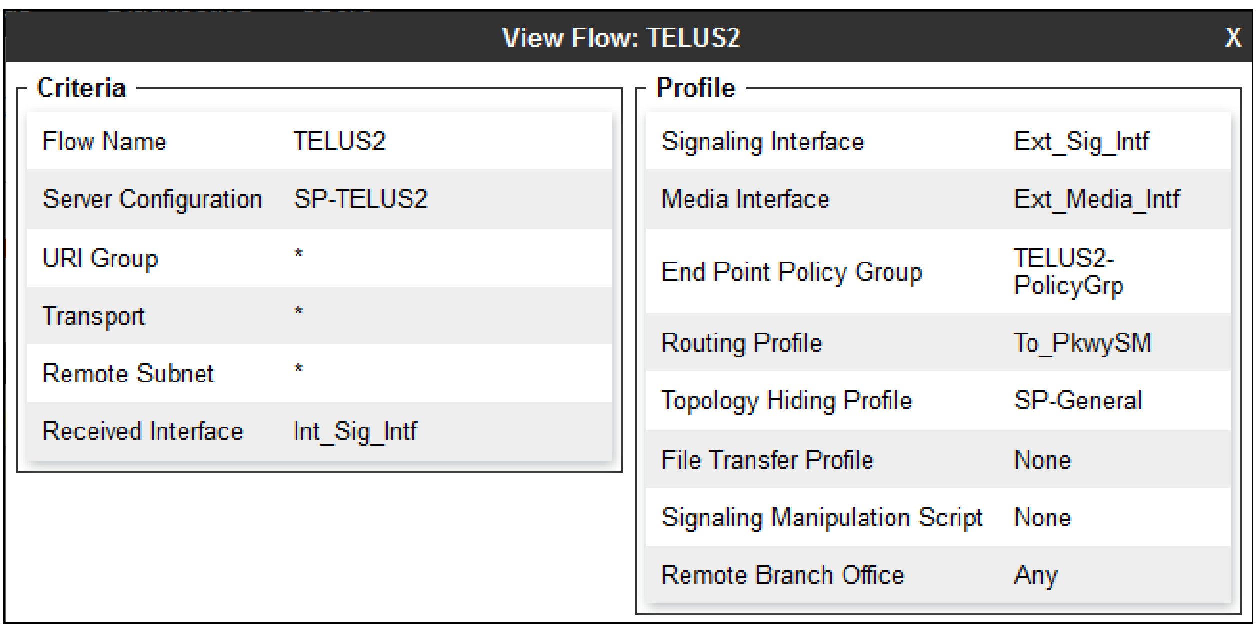
- For the **Flow Name**, enter a descriptive name.
- For **Server Configuration**, select the Session Manager server created in **Section 7.7.1**.
- To match all traffic, set the **URI Group**, **Transport**, and **Remote Subnet** to *.
- Set the **Received Interface** to the external signaling interface.
- Set the **Signaling Interface** to the internal signaling interface.
- Set the **Media Interface** to the internal media interface.
- Set the **End Point Policy Group** to the endpoint policy group defined for Session Manager in **Section 7.11.1**.
- Set the **Routing Profile** to the routing profile defined in **Section 7.12.2** used to direct traffic to the TELUS SIP server.
- Set the **Topology Hiding Profile** to the topology hiding profile defined for Session Manager in **Section 7.13.1**.



7.14.2. End Point Flow – TELUS

For the compliance test, endpoint flow **TELUS2** was created for the TELUS SIP server. All traffic from TELUS will match this flow as the source flow and use the specified **Routing Profile To_PkwySM** to determine the destination server and corresponding destination flow. The **End Point Policy** and **Topology Hiding Profile** will be applied as appropriate. When creating the flow, configure the parameters as follows:

- For the **Flow Name**, enter a descriptive name.
- For **Server Configuration**, select the TELUS SIP server created in **Section 7.7.2**.
- To match all traffic, set the **URI Group**, **Transport**, and **Remote Subnet** to *****.
- Set the **Received Interface** to the internal signaling interface.
- Set the **Signaling Interface** to the external signaling interface.
- Set the **Media Interface** to the external media interface.
- Set the **End Point Policy Group** to the endpoint policy group defined for TELUS in **Section 7.11.2**.
- Set the **Routing Profile** to the routing profile defined in **Section 7.12.1** used to direct traffic to Session Manager.
- Set the **Topology Hiding Profile** to the topology hiding profile defined for TELUS in **Section 7.13.2**.



8. TELUS SIP Trunking Service Configuration

TELUS is responsible for the network configuration and deployment of the TELUS SIP Trunking Service.

TELUS will require that the customer provide the IP address and port number used to reach the Avaya SBCE at the edge of the enterprise. TELUS will provide the IP address and port number of the TELUS SIP proxy/SBC, IP addresses/ports of media sources and Direct Inward Dialed (DID) numbers assigned to the enterprise. This information is used to complete the Communication Manager, Session Manager and Avaya SBCE configuration discussed in the previous sections.

9. Verification Steps

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting commands that can be used to troubleshoot the solution.

Verification Steps:

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
3. Verify that a user on the PSTN can end an active call by hanging up.
4. Verify that an endpoint at the enterprise site can end an active call by hanging up.

Troubleshooting:

1. Communication Manager:
    - **list trace station** <extension number> - Traces calls to and from a specific station.
    - **list trace tac** <trunk access code number> - Trace calls over a specific trunk group.
    - **status station** <extension number> - Displays signaling and media information for an active call on a specific station.
    - **status trunk** <trunk access code number> - Displays real-time trunk group information.
    - **status trunk** <trunk access code number/channel number> - Displays real-time signaling and media information for an active trunk channel.

2. Session Manager:
    - **Call Routing Test** - The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, navigate to **Elements → Session**

Manager → System Tools → Call Routing Test. Enter the requested data to run the test.

3. Avaya Session Border Controller for Enterprise:
   There are several links and menus located on the taskbar at the top of the screen of the web interface that can provide useful diagnostic or troubleshooting information.

   - **Alarms**: This option provides information about active alarms.
   - **Incidents**: This option provides detailed reports of anomalies, errors, policies violations, etc.
   - **Status**: This option provides statistical and current status information.
   - **Diagnostics**: This option provides a variety of tools to test and troubleshoot the Avaya SBCE network connectivity.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and the Avaya Session Border Controller for Enterprise to the TELUS SIP Trunking Service. The TELUS SIP Trunking Service provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. Please refer to **Section 2.2** for any exceptions or workarounds.

11. References

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

[1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.3.4, July 2014.
[2] *Administering Avaya Aura® System Platform*, Release 6.3.4, July 2014.
[3] *Administering Avaya Aura® Communication Manager*, Release 6.3, May 2014, Document Number 03-300509.
[4] *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 6.3, October 2014, Document Number 555-245-205.
[5] *Upgrading Avaya Aura® System Manager*, Release 6.3, October 2014.
[6] *Administering Avaya Aura® System Manager*, Release 6.3, February 2015.
[7] *Installing and Configuring Avaya Aura® Session Manager*, Release 6.1, April 2011, Document Number 03-603473.
[8] *Administering Avaya Aura® Session Manager*, Release 6.3, September 2013.
[9] *Deploying Avaya Session Border Controller for Enterprise*, Release 6.3, October 2014.
[10] *Administering Avaya Session Border Controller for Enterprise*, Release 6.3, October 2014.
[11] *Avaya 1600 Series IP Deskphones Administrator Guide Release 1.3.6*, August 2014, Document Number 16-601443.

[12] *Avaya one-X® Deskphone Edition H.323 for 9600 Series IP Deskphones Administrator Guide*, Release 3.2, January 2013, Document Number 16-300698.
[13] *Administering 9608,9808G,9611G,9621G IP Deskphones Edition H.323*, June 2014, Document Number 16-300698.
[14] *Administering 9608,9808G,9611G,9621G IP Deskphones Edition SIP*, January 2015, Document Number 16-601944.
[15] *Administering Avaya one-X® Communicator*, October 2014.
[16] RFC 3261 *SIP: Session Initiation Protocol*, http://www.ietf.org/
[17] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, http://www.ietf.org/

12. Appendix A: TELUS SIP Manipulation Script

```
// Signalling Manipulations created for interoperability
// with TELUS R2 SIP Platform

// If the Diversion header exists, modify the PAI using the
// contents of the Diversion header.

within session "INVITE"
{
 act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
 {
 if (exists(%HEADERS["Diversion"][1])) then
  {
  %var1 = %HEADERS["Diversion"][1].URI.USER;
  %var2 = %HEADERS["Diversion"][1].DISPLAY_NAME;

  %HEADERS["P-Asserted-Identity"][1].URI.USER = %var1;
  %HEADERS["P-Asserted-Identity"][1].DISPLAY_NAME = %var2;

  remove(%HEADERS["Diversion"][1]);
  }
 }
}

// Set Max-Forwards header to 0 in OPTIONS

within session "OPTIONS"
{
 act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
 {
  if (exists(%HEADERS["Max-Forwards"][1])) then
  {
  %HEADERS["Max-Forwards"][1] = "0";
  }
 }
}
```

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