Avaya Solution & Interoperability Test Lab

Application Notes for Configuring TELUS SIP Trunking with Avaya Aura® Communication Manager Access Element 5.2.1, Avaya Aura® Session Manager 6.1 and Acme Packet 3800 Net-Net Session Border Controller – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between TELUS SIP Trunking and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager 6.1, Avaya Aura® Communication Manager Access Element 5.2.1, Acme Packet 3800 Net-Net Session Border Controller and various Avaya endpoints.

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

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between TELUS SIP Trunking and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager 6.1, Avaya Aura® Communication Manager Access Element 5.2.1, Acme Packet 3800 Net-Net Session Border Controller and various Avaya endpoints.

Customers using this Avaya SIP-enabled enterprise solution with TELUS SIP Trunking are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. 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 TELUS SIP Trunking via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of Communication Manager, Session Manager and the Acme Packet 3800 Net-Net SBC.

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

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test. Please note that enterprise SIP endpoints were not tested since SIP endpoints are not supported on a Communication Manager Access Element.

- Response to SIP OPTIONS queries
- Incoming PSTN calls to various enterprise phone types
  Phone types included H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various enterprise phone types
  Phone types included H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator (soft client)
  Avaya one-X® Communicator can place calls from the local computer or control a remote phone. Both of these modes were tested. Avaya one-X® Communicator also supports two Voice Over IP (VoIP) protocols: H.323 and SIP. Only the H.323 version of Avaya one-X® Communicator was tested.
- Inbound and outbound calls to/from TELUS Derived Voice endpoints

- Inbound and outbound calls to/from TELUS Mobility endpoints
- Various call types including: local, long distance, international, outbound toll-free, operator assisted calls, and local directory assistance (411).
- Codec 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 enterprise mobility (extension to cellular)
- T.38 Fax (established from an initial G.711MU/A call)
- Network Call Redirection using the SIP REFER method

Items not supported or not tested included the following:

- Inbound toll-free and emergency calls are supported but were not tested.
- Call redirection requested by a 302 response is not supported by TELUS.
- Establishment of a T.38 fax from a G.729 call could not be tested due to a limitation of the lab environment.

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/limitations described below.

- **OPTIONS 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. Thus, the SBC was used to modify this value when the SBC sent the OPTIONS message to the network. (See **Section 7.10.2.15**) The SBC can also be configured to originate its own OPTIONS message to the network with Max-Forward set to zero. See the **ping-method** setting in **Section 7.8.1**. This was done as part of the compliance test but it is not necessary since Session Manager was sending OPTIONS message to determine link status.
- **Use of SA8965**: TELUS requires re-INVITEs to contain Session Description Protocol (SDP) information. Thus, the Communication Manager special application SA8965 must be enabled. (See **Section 5.2**)
- **No Error Indication if No Matching Codec Offered**: If the Communication Manager SIP trunk is improperly configured to have no matching codec with the service provider and an inbound call is placed, the Communication Manager returns a "500 Service Unavailable" response instead of a "488 Not Acceptable Here" response. The user hears fast busy.
- **Call Fowarding and EC500**: Inbound PSTN calls that are call forwarded back to the PSTN or ring to an EC500 (enterprise mobility) endpoint, will display the forwarding

party/EC500 host at the destination instead of the original PSTN caller. This is the result of differences in the interpretation/implementation of the SIP Diversion header between TELUS and Communication Manager. A SIP header manipulation was created on the SBC to modify the P-Asserted-ID header with information contained in the Diversion header. (See **Sections 7.10.2.12** and **7.10.2.13**) This allows the call to complete but results in the incorrect calling party displayed at the destination as described above.

- **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 displays the number of the transferring party and not the actual connected party. Communication Manager provides the new connected party information by updating the Contact header in an UPDATE message. TELUS does not use the UPDATE message for this purpose.
- **Coverage to Voicemail for TELUS Mobility Users**: Calls from the enterprise to TELUS mobility users that cover to voicemail could result in one-way audio. If this occurs, the caller will not be able to hear the voicemail announcements and menus. A software change was made on Communication Manager to address this issue and was built on top of Release 5.2.1 Service Pack 11. The change was tested and passed compliance testing using an early development release. The change will be available in a future service pack release. Customers who encounter this problem should use the standard escalation process to request a patch from Avaya Global Services.
- **Use of REFER**: Enabling of the Network Call Redirection feature on the Communication Manager SIP trunk activates the use of the SIP REFER method for various inbound PSTN calls redirected calls back to the PSTN. The use of the REFER method resulted in dropped calls for blind transfer and vector redirection scenarios. Enabling of Network Call Redirection is not recommended.
- **T.38 Fax – Network Coverage**: Not all media gateways in the TELUS network support T.38 fax. Communication Manager does not support fallback to G.711 pass-through fax from T.38 fax. Thus, if a T.38 fax call encounters a media gateway in the TELUS network that does not support T.38 then the call will terminate.
- **Transitioning to T.38 for Outbound Calls**: In general, the answering side of a fax call will 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 requires Communication Manager to transition to T.38 for both inbound and outbound fax calls. Relying on Communication Manager to transition to T.38 on an outbound call may have the following impact:
  - On an outbound call, sending of the T.38 INVITE happens on detection of the V.21 preamble of the originating fax machine's Digital Command Signal (DCS) message. This is part of the T.30 exchange. This request to transition to T.38 may happen too late for some terminating gateways to accommodate the switch to T.38.
  - If the initial call is using the G.729 codec, the compression of the V.21 preamble may cause its detection to be less reliable then if the call was initially using G.711.
  - The ability to transition to T.38 in the middle of the T.30 exchange is supported on the following Avaya media platforms (G430/G450/TN2602). Older platforms (G350/G700/TN2302) may have different behavior.

Compliance testing was conducted with the TN2602 media platform (part of the G650 media gateway) using codec G.711MU to initially establish the call. Outbound T.38 fax calls in this environment were successful.

- **G.711 Pass-through Fax**: Communication Manager does not support G.711 pass-through fax over SIP trunks. These calls are treated like any other voice call by Communication Manager. If a customer chooses to use G.711 pass-through fax, success is not guaranteed.

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. Selecting the **Support Contact Options** link followed by **Maintenance Support** provides the worldwide support directory for Avaya Global Services. Specific numbers are provided for both customers and partners based on the specific type of support or consultation services needed. Some services may require specific Avaya service support agreements. 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 TELUS SIP Trunking. This is the configuration used for compliance testing.

The Avaya components used to create the simulated customer site included:

- Avaya S8300D Server running Communication Manager
- Avaya G450 Media Gateway
- Avaya S8800 Server running Session Manager
- Avaya S8800 Server running System Manager
- Avaya 9600-Series IP telephones (H.323)
- Avaya 4600-Series IP telephones (H.323)
- Avaya 1600-Series IP telephones (H.323)
- Avaya one-X® Communicator (H.323)
- Avaya digital and analog telephones

Located at the edge of the enterprise is the 3800 Net-Net SBC. 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 3800 Net-Net SBC. In this way, the 3800 Net-Net SBC can protect the enterprise against any SIP-based attacks. The 3800 Net-Net SBC 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. Similarly, any references to real routable PSTN numbers have also been changed to numbers that can not be routed by the PSTN.

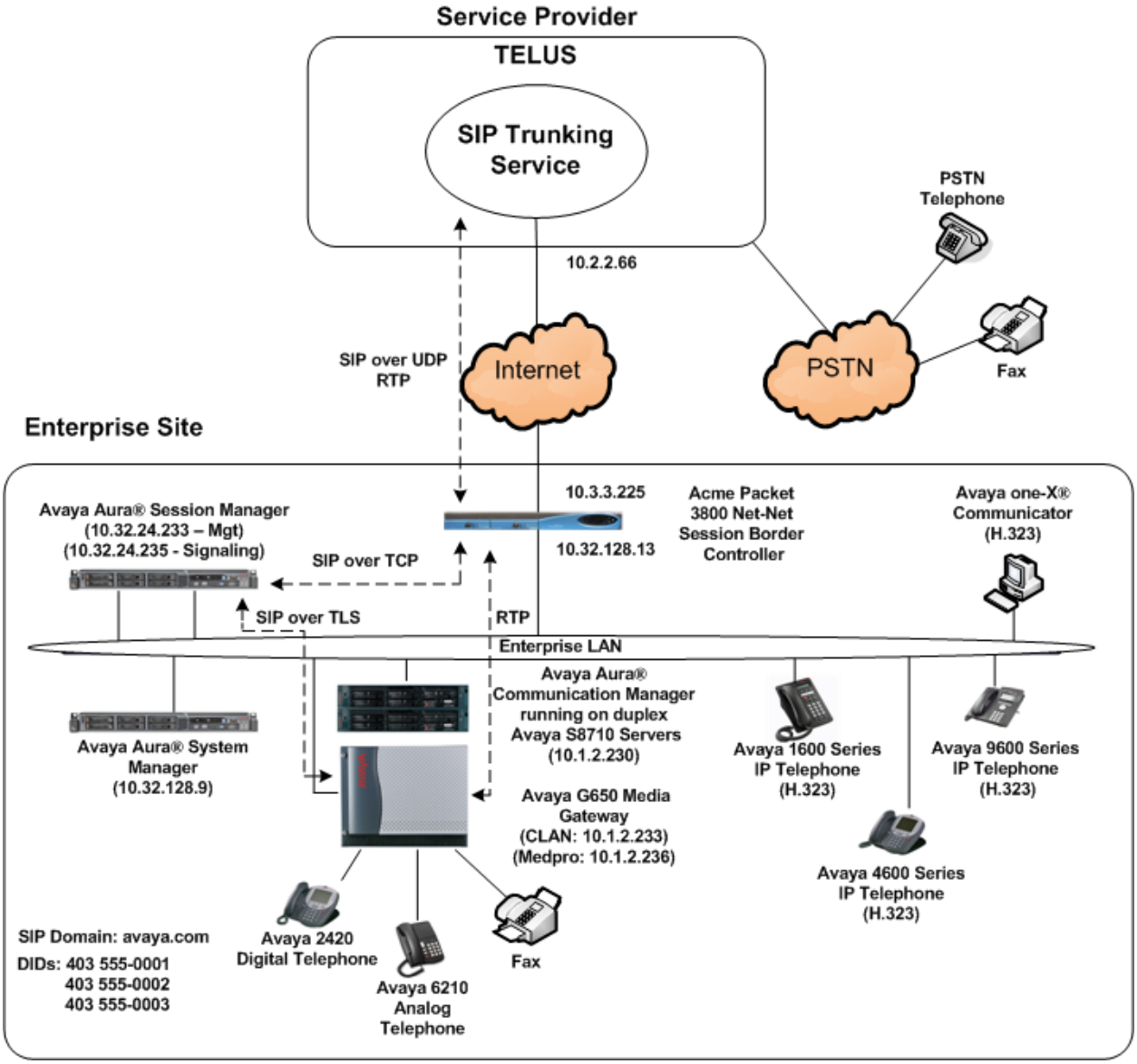


Figure 1: Avaya IP Telephony Network using TELUS SIP Trunking

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 settings 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 3800 Net-Net SBC then to Session Manager. Session Manager uses the configured dial patterns (or regular expressions) and routing policies to determine the recipient (in this case the 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. The Session Manager once again uses the configured dial patterns (or regular expressions) to determine the route to the 3800 Net-Net SBC. From the 3800 Net-Net SBC, the call is sent to TELUS SIP Trunking.

For outbound calls, the enterprise was configured to send 11 digits in the SIP destination headers (Request URI and To) and 10 digits in the SIP source headers (i.e., From, Contact, and P-Asserted-Identity). For inbound calls, TELUS sent 10 digits in both the source headers and destination headers.

4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Component	Release
Avaya Aura® Communication Manager running on duplex Avaya S8710 Servers	5.2.1 SP11 + dev patch (R015x.02.1.016.4-19308) + dev patch
Avaya G650 Media Gateway • IP Server Interface (IPSI) TN2312BP • Control LAN (CLAN) TN799DP • IP Media Processor (MEDPRO) TN2602AP	HW15 FW054 HW01 FW040 HW02 FW061
Avaya Aura® System Manager running on an Avaya S8800 Server	6.1 SP5 (Build 6.1.0.0.7345-6.1.5.502) (System Platform 6.0.3.3.3)
Avaya Aura® Session Manager running on an Avaya S8800 Server	6.1 SP5 (Build asm-6.1.5.0.615006)
Avaya 1608 IP Telephone (H.323)	Avaya one-X® Deskphone Value Edition 1.3.00B
Avaya 4621SW IP Telephone (H.323)	2.9.2
Avaya 9640 IP Telephone (H.323)	Avaya one-X® Deskphone Edition 3.1 SP2 (3.1.02S)
Avaya one-X® Communicator (H.323)	6.1 (Build 6.1.0.12-GA-30334)
Avaya 2420 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Acme Packet 3800 Net-Net Session Border Controller	SCX6.2.0 MR-3 GA (Build 619)
TELUS SIP Trunking Solution Components	
Component	Release
Acme Packet 4520 Net-Net Session Border Controller	6.1m7p5
Nokia Siemens Networks HiQ 4200	Version 14.0

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 TELUS SIP Trunking. 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, 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 800 SIP trunks are available and 208 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  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 800   200
          Maximum Concurrently Registered IP Stations: 18000 5
            Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
             Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 800   208
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                            Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   2
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

5.2. Special Application SA8965

TELUS requires that all INVITE messages contain SDP information, including re-INVITEs. In general, when Communication Manager sends a re-INVITE to perform a media shuffling operation (redirect media directly between two endpoints) the re-INVITE will not include SDP information. In order to change this behavior, special application SA8965 must be enabled. This is done via the **change system-parameters special-applications** command. Navigate to **Page 7** and enter a **y** next to the special application titled **SA8965 - SIP Shuffling with SDP** in the list below. By enabling this feature, a new protocol variation parameter will appear on **Page 3** of the trunk form (See **Section 5.8**).

```
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? y
  (SA8967) - Mask CLI and Station Name for QSIG/ISDN Calls? y
```

5.3. System Features

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

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 **unknown** for both.

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

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

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:

ENBLOC DIALING 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 CLAN circuit pack (**clan1**) and for Session Manager (**bvSM**). 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
bvSM                   10.32.24.235
clan1                  10.1.2.233
default                0.0.0.0
medpro2                10.1.2.236
procr                   . . .
procr1                 10.1.2.11
procr2                 10.1.2.21
```

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. For the compliance test, G.729A, G.711A and G.711MU were defined in IP codec set 4. To use these codecs, enter **G.729A**, **G.711A** and **G.711MU** in the **Audio Codec** column of the table in the order of preference. Default values can be used for all other fields.

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

                           IP Codec Set

    Codec Set: 4

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

On **Page 2**, to enable T.38 fax, set the **Fax Mode** to **t.38-standard**. Otherwise, set the Fax Mode to **off**.

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

                           IP Codec Set

                               Allow Direct-IP Multimedia? n

                    Mode                Redundancy
    FAX             t.38-standard        0
    Modem           off                  0
    TDD/TTY         US                   3
    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 4 was chosen for the service provider trunk. Use the **change ip-network-region 4** command to configure region 4 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 4                                    Page   1 of  19
                              IP NETWORK REGION
  Region: 4
Location:         Authoritative Domain: avaya.com
    Name: SP Region
MEDIA PARAMETERS                 Intra-region IP-IP Direct Audio: yes
      Codec Set: 4               Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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 3**, define the IP codec set to be used for traffic between region 4 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 4 will be used for calls between region 4 (the service provider region) and region 1 (the rest of the enterprise). Creating this table entry for IP network region 4 will automatically create a complementary table entry on the IP network region 1 form for destination region 4. This complementary table entry can be viewed using the **display ip-network-region 1** command and navigating to **Page 4**.

```
change ip-network-region 4                                    Page   3 of  19

 Source Region: 4     Inter Network Region Connection Management     I       M
                                                                    G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    4     y    NoLimit                                          n       t
 2
 3
 4    4                                                            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 34 was used 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). The transport method specified here is used between the Communication Manager and Session Manager.
- Set the **IMS Enabled** field to **n**. This specifies the Communication Manager will serve as an Access Element for Session Manager.
- Set the **Near-end Node Name** to **clan1**. This node name maps to the IP address of the CLAN circuit pack as defined in **Section 5.4**.
- Set the **Far-end Node Name** to **bvSM**. 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 the chosen transport protocol. (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. 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 **5066**.

- 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 **Direct IP-IP Audio Connections** to **y**. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the SIP trunk and the enterprise endpoint. If this value is set to **n**, then the Avaya Media Gateway will remain in the media path of all calls between the SIP trunk and the endpoint. Depending on the number of media resources available in the Avaya Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- Set the **DTMF over IP** field to **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- 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 34                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 34              Group Type: sip
                         Transport Method: tls
  IMS Enabled? n




   Near-end Node Name: clan1                 Far-end Node Name: bvSM
 Near-end Listen Port: 5066                Far-end Listen Port: 5066
                                         Far-end Network Region: 4
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                  Direct IP-IP Early 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 34 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 the **Signaling Group** to the signaling group shown in **Section 5.7**.
- 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 34                                          Page   1 of  21
                              TRUNK GROUP

Group Number: 34                    Group Type: sip           CDR Reports: y
  Group Name: SP Trunk                      COR: 1        TN: 1        TAC: 134
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                      Signaling Group: 34
                                                    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 **600** seconds was used.

```
add trunk-group 34                                              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): 600
```

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

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to ***y***. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.3**, 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 34                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                 UUI Treatment: service-provider

                                                  Replace Restricted Numbers? y
                                                 Replace Unavailable Numbers? y


 Show ANSWERED BY on Display? y
```

On **Page 4**, set the **Network Call Redirection** field to **n**. Set the **Send Diversion Header** field to **y** and the **Support Request History** field to **n**. The **Send Diversion Header** and **Support Request History** fields provide additional information to the network if the call has been re-directed. 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 preferred by TELUS.

Set the **Shuffling with SDP** field to **y**. This will instruct Communication Manager to send SDP information in shuffling re-INVITEs on calls that use this trunk. This parameter only appears if special application SA8965 is enabled. See **Section 5.2** for full details.

```
add trunk-group 34                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                    Send Diversion Header? y
                  Support Request History? n
             Telephone Event Payload Type: 101
                       Shuffling with SDP? y
```

5.9. Calling Party Information

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

In the sample configuration, multiple DID numbers were assigned for testing. These numbers were assigned to the extensions 30023, 30024 and 30025. 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 public-unknown-numbering 0                              Page   1 of   1
                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext  Ext             Trk        CPN       CPN
Len  Code            Grp(s)     Prefix    Len
                                                    Total Administered: 4
 5   3                                     5          Maximum Entries: 9999
 5   30023           34         4035550001 10
 5   30024           34         4035550002 10
 5   30025           34         4035550003 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 public-unknown-numbering entry can be applied for all extensions. In the example below, all stations with a 5-digit extension beginning with 3 will send the calling party number as the **CPN Prefix** plus the extension number.

```
change public-unknown-numbering 0                              Page   1 of   1
                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext  Ext             Trk        CPN       CPN
Len  Code            Grp(s)     Prefix    Len
                                                    Total Administered: 1
 5   3               34         40355     10         Maximum Entries: 9999
```

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:    2

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     1            3    dac
     2            5    ext
     222          5    aar
     3            5    ext
     3234         7    ext
     4            5    ext
     5            5    ext
     6            5    ext
     7            7    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   8
                              FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *01
         Abbreviated Dialing List2 Access Code: *02
         Abbreviated Dialing List3 Access Code: *03
Abbreviated Dial - Prgm Group List Access Code: *04
                      Announcement Access Code: *05
                       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: *13    All: *11    Deactivation: *12
```

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 34 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: 2

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern  Type   Num   Reqd
    0                     1    1    34       op           n
    0                     11   11   34       op           n
    011                   10   18   34       intl         n
    1403                  11   11   34       fpna         n
    1647                  11   11   34       fpna         n
    1780                  11   11   34       fpna         n
    1800                  11   11   34       fpna         n
    1877                  11   11   34       fpna         n
```

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the service provider trunk route pattern in the following manner. The example below shows the values used for route pattern 34 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 34 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.
- **Pfx Mrk**: **1** The prefix mark (**Pfx Mrk**) of one will prefix any FNPA 10-digit number with a 1 and leave numbers of any other length unchanged. This will ensure 1 + 10 digits are sent to the service provider for long distance North American Numbering Plan (NANP) numbers.
- **LAR**: **next**

```
change route-pattern 34                                          Page   1 of   3
                     Pattern Number: 34  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: 34   0       1                                                   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                                      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
```

6. Configure Avaya Aura® Session Manager

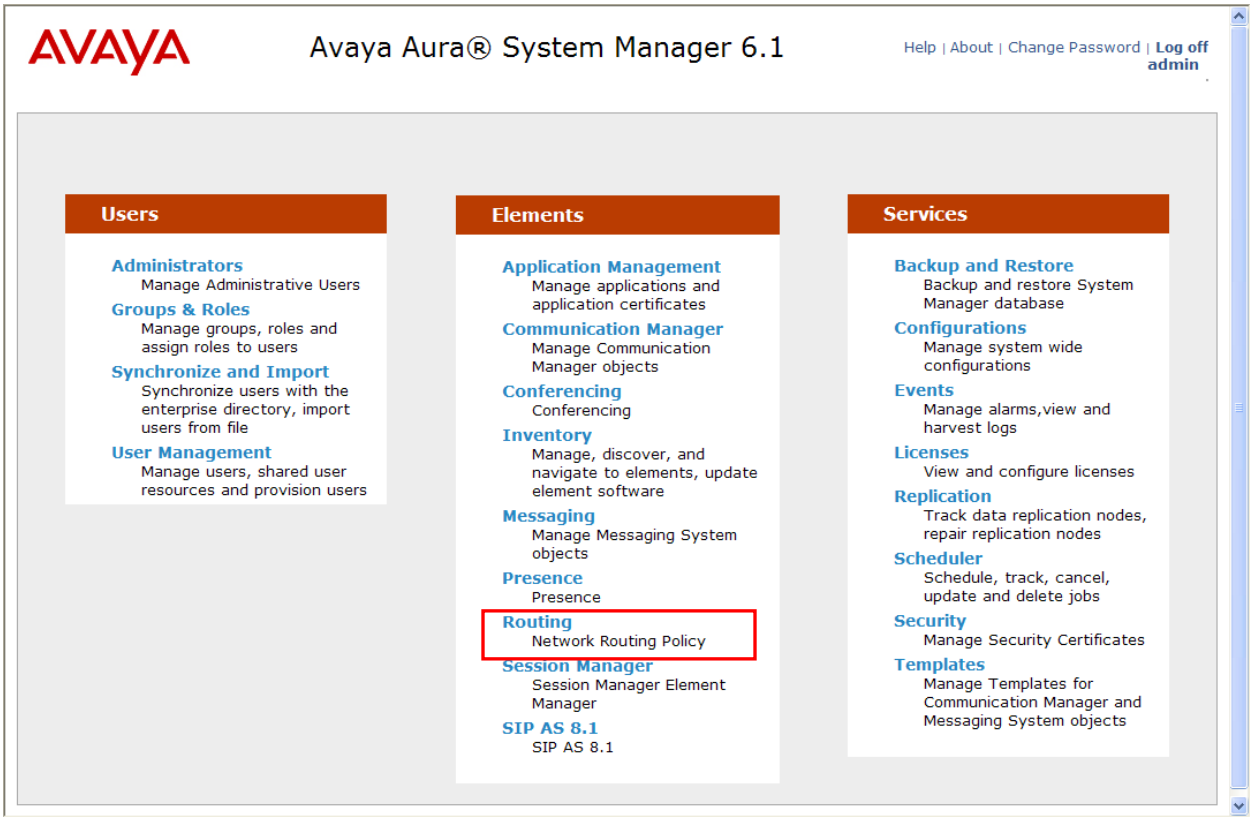
This section provides the procedures for configuring Session Manager. The procedures include configuring the following items:

- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- Adaptation module to perform dial plan manipulation
- SIP Entities corresponding to Communication Manager, the SBC and Session Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which governs which Routing Policy is used to service a call.
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager.

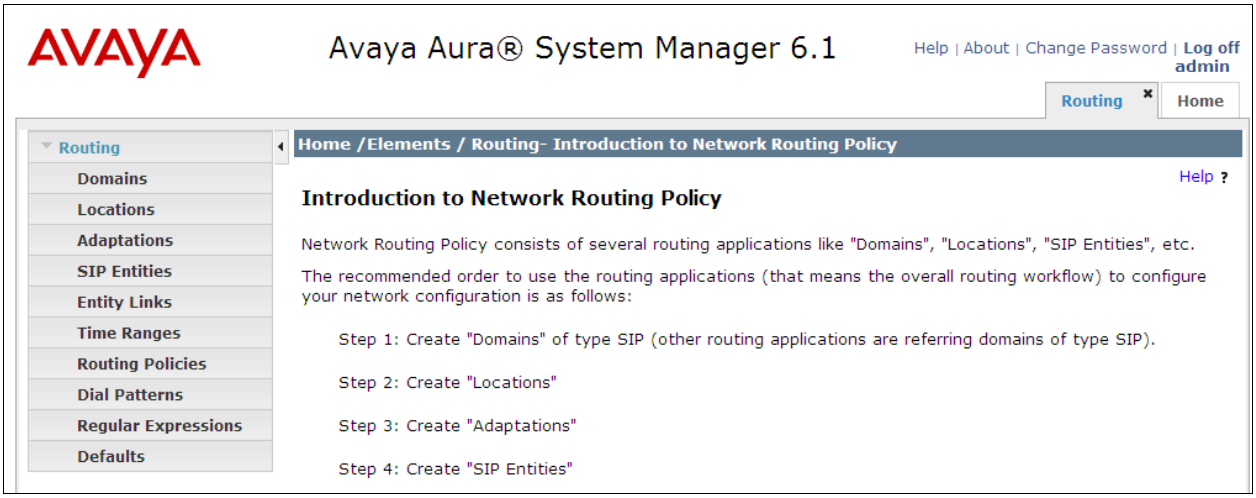
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session 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 **Home** 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.

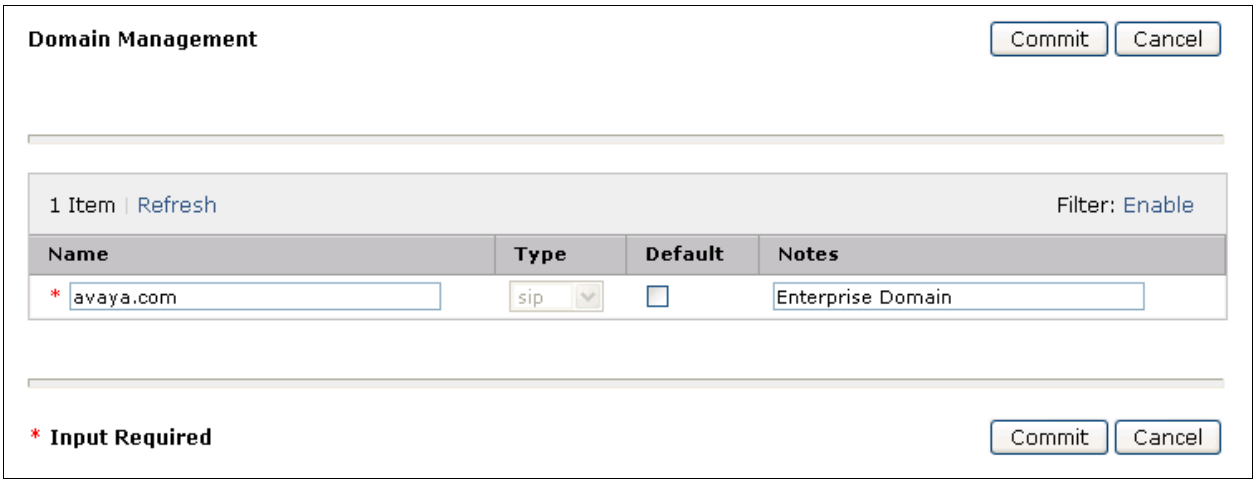


6.2. Specify SIP Domain

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**). 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.



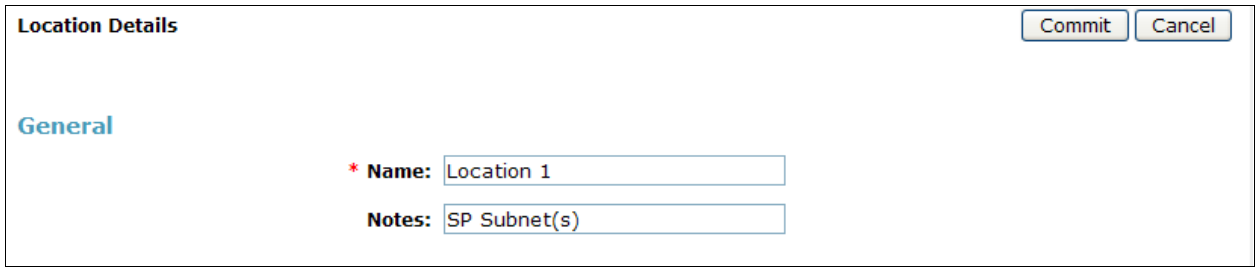
6.3. Add Location

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 **Location 1**, which includes all equipment on the enterprise including Communication Manager, Session Manager and the 3800 Net-Net SBC.

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.

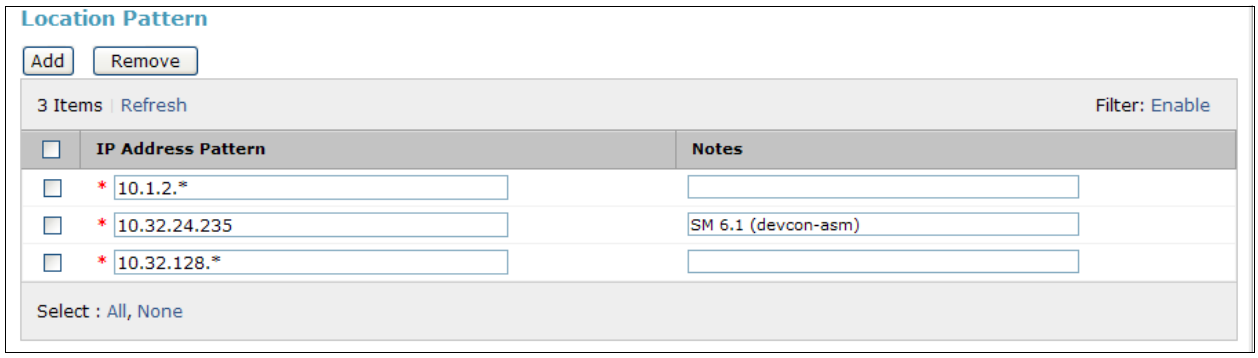
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).



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

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

Click **Commit** to save.



6.4. Add Adaptation Module

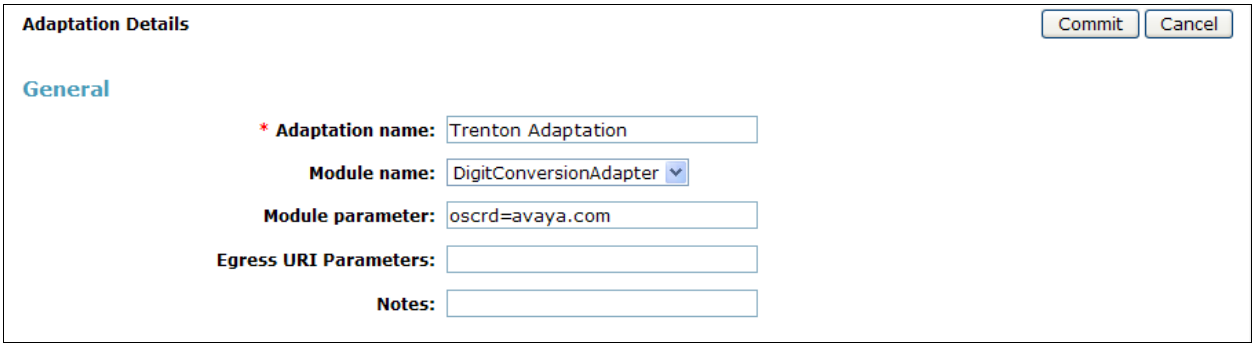
Session Manager can be configured with adaptation modules that can modify SIP messages before or after routing decisions have been made. A generic adaptation module **DigitConversionAdapter** supports digit conversion of telephone numbers in specific headers of SIP messages. Other adaptation modules are built on this generic, and can modify other headers to permit interoperability with third party SIP products.

For the compliance test, two adaptations were needed. The first adaptation was applied to the Communication Manager SIP entity and converts the domain part of the inbound PAI header to the enterprise domain (**avaya.com**). In addition, this adaptation maps inbound DID numbers from TELUS to local Communication Manager extensions. The second adaptation is applied to the 3800 Net-Net SBC SIP entity and converts the domain part of the outbound Request URI header from Session Manager containing the enterprise domain to the TELUS SIP proxy IP address.

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.

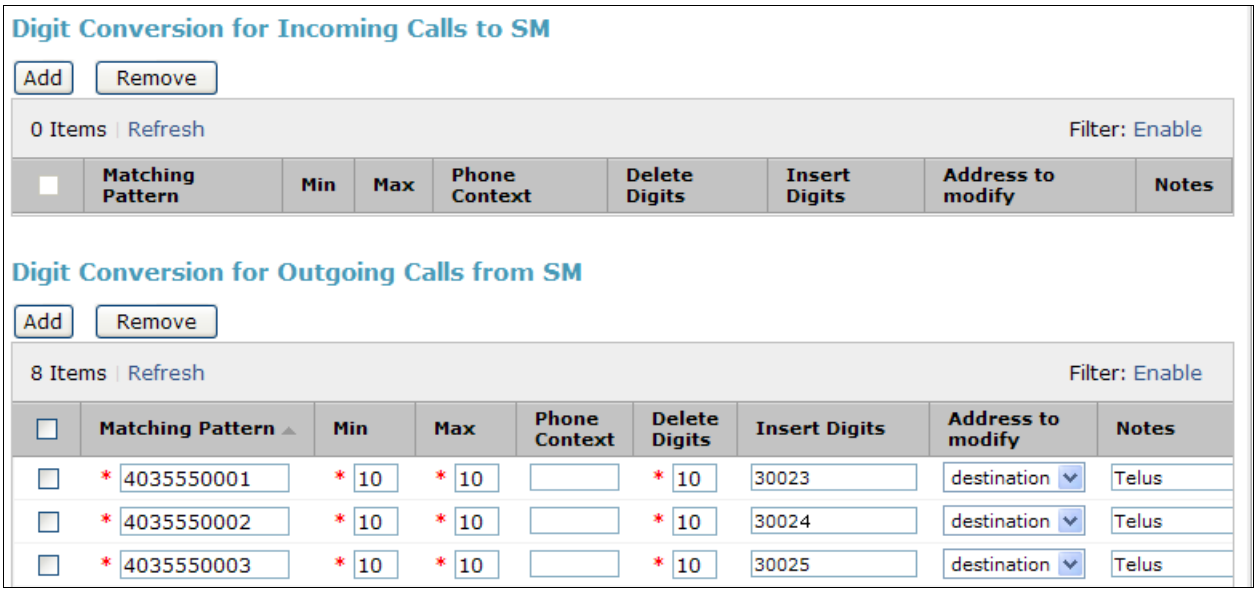
- **Adaptation name:** Enter a descriptive name for the adaptation.
- **Module name:** Enter **DigitConversionAdapter**.
- **Module parameter:** Enter **osrcd=avaya.com**. This is the OverrideSourceDomain parameter. This parameter replaces the domain in the inbound PAI header with the given value. This parameter must match the value used for the **Far-end Domain** setting on the Communication Manager signaling group form in **Section 5.7**.



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 number of 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.

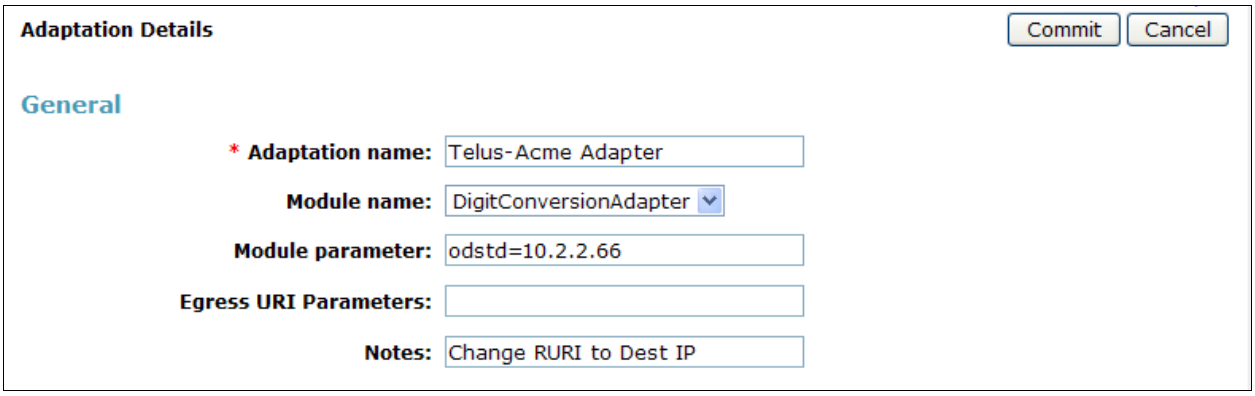


To create the adaptation that will be applied to the 3800 Net-Net SBC SIP entity, navigate to **Routing → Adaptations** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

- **Adaptation name:** Enter a descriptive name for the adaptation.
- **Module name:** Enter **DigitConversionAdapter**.
- **Module parameter:** Enter **odstd=10.2.2.66.** This is the OverrideDestinationDomain parameter. This parameter replaces the domain in the Request URI header with the given value for outbound only.
- **Notes:** Add a brief description (optional).

Click **Commit** to save.



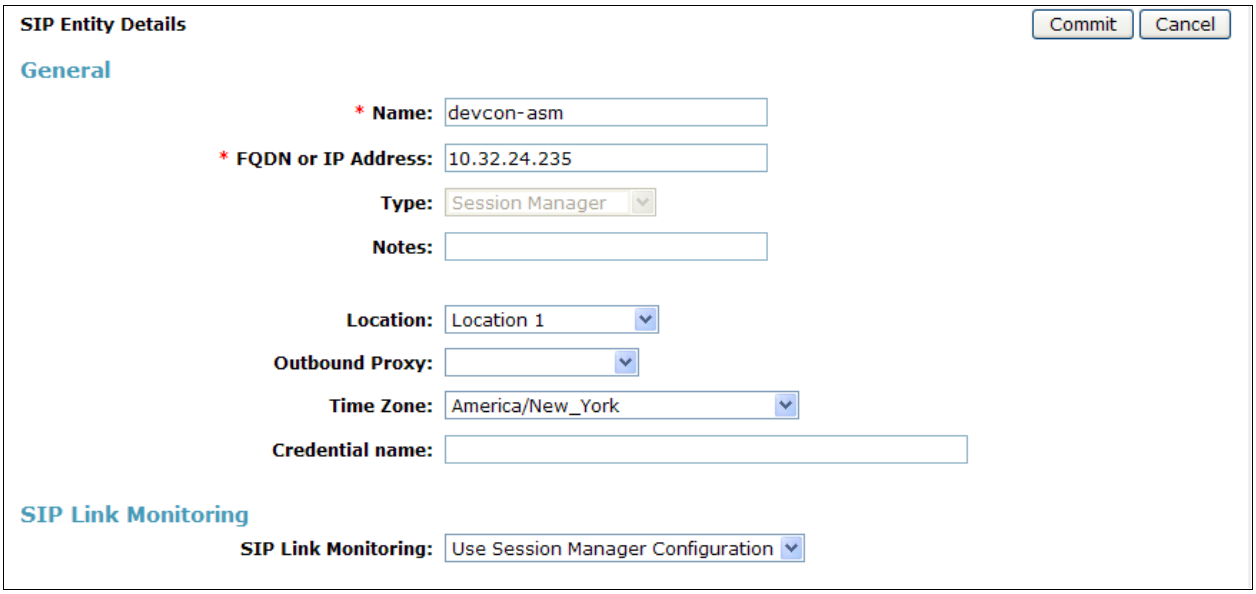
6.5. Add SIP Entities

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 3800 Net-Net SBC. 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 3800 Net-Net SBC.
- **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 **Location 1**.
- **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**.



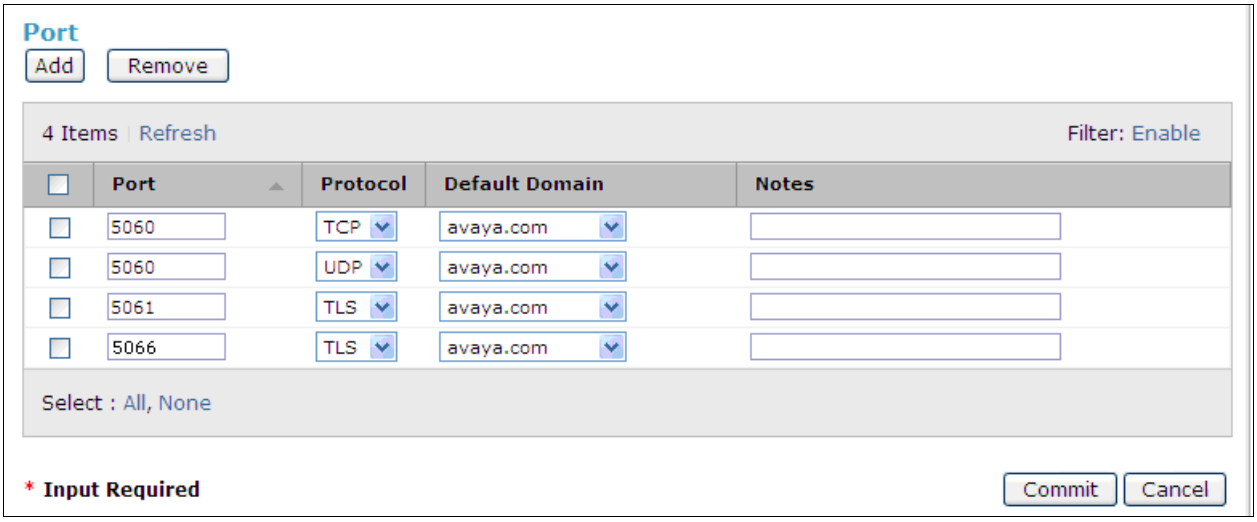
To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP entities.

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

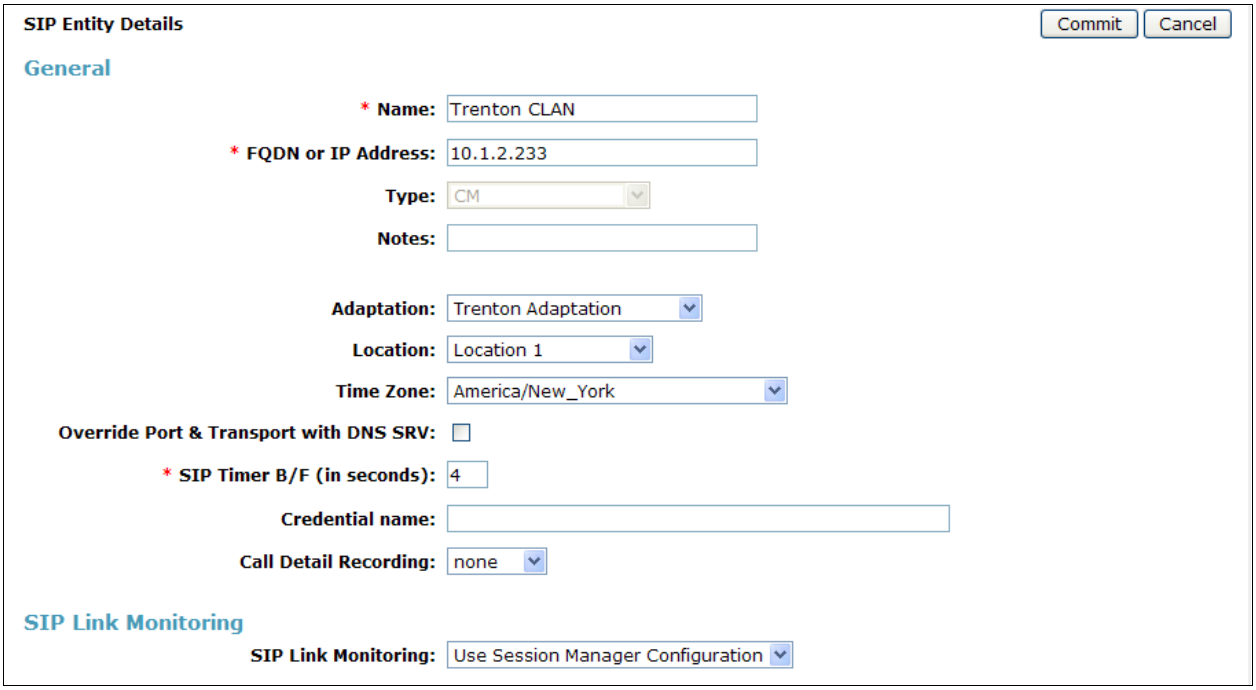
- **Port:** Port number on which the Session Manager can listen for SIP requests.
- **Protocol:** Transport protocol to be used with this port.
- **Default Domain:** The default domain associated with this port. For the compliance test, this was the enterprise SIP domain.

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

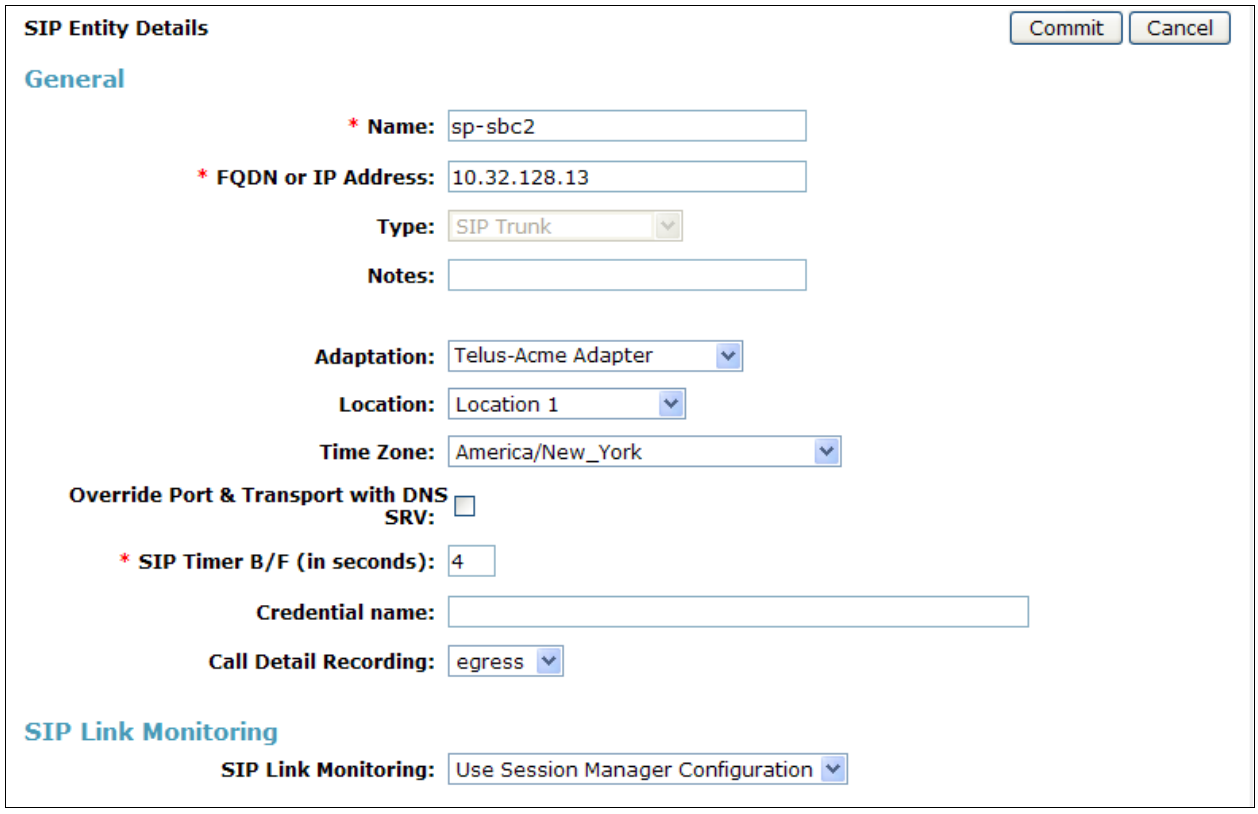
For the compliance test, four port entries were used. The first three are the standard ports used for SIP traffic: port 5060 for UDP/TCP and port 5061 for TLS. In addition, port 5066 defined in **Section 5.7** for use with service provider SIP traffic between Communication Manager and Session Manager 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 the Avaya Server running Communication Manager. For the **Adaptation** field, select the adaptation module previously defined for use with Communication Manager in **Section 6.4**. The **Location** field is set to **Location 1** which is the location defined for the subnet where Communication Manager resides.



The following screen shows the addition of the 3800 Net-Net SBC. 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 module previously defined for the SBC in **Section 6.4**. The **Location** field is set to **Location 1** which is the location defined for the subnet where the 3800 Net-Net SBC resides.

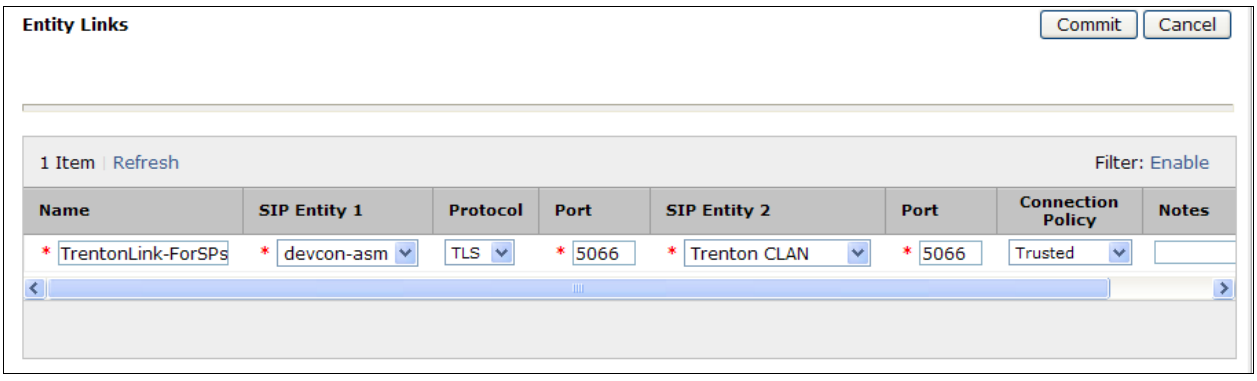


6.6. Add Entity Links

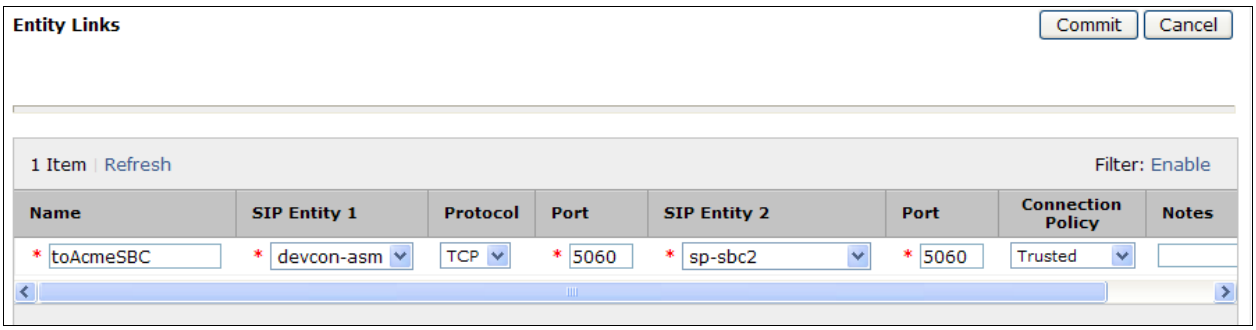
A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created: one to Communication Manager for use only by service provider traffic and one to the 3800 Net-Net SBC. To add an Entity Link, navigate to **Routing → Entity Links** in the left-hand 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.
- **Protocol:** Select the transport protocol used for this link.
- **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 the 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**.
- **Trusted:** Check this box. Note: If this box is not checked, calls from the associated SIP Entity specified in **Section 6.5** will be denied.

Click **Commit** to save. The following screen illustrates the Entity Link to Communication Manager. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.7**.



The following screen illustrates the Entity Link to the 3800 Net-Net SBC.



6.7. Add Routing Policies

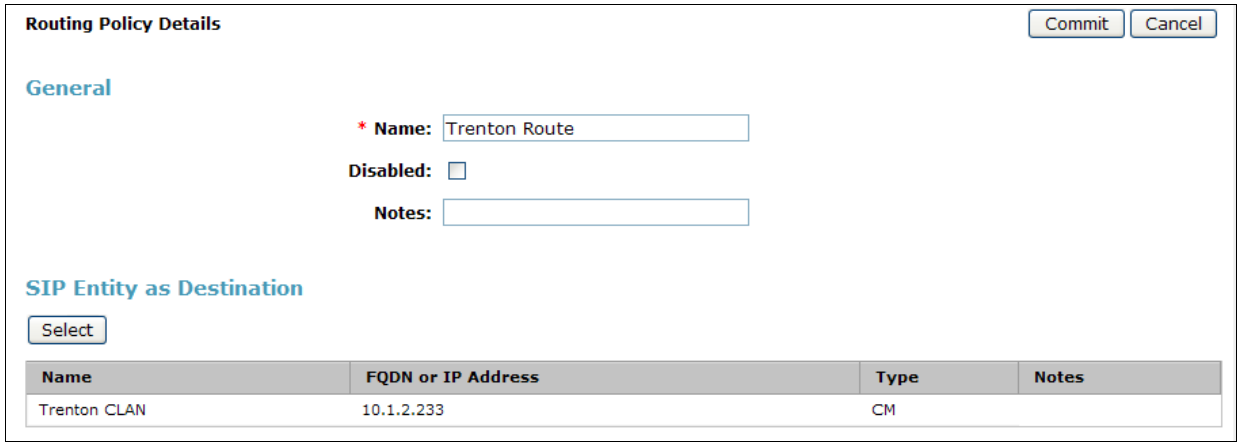
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Two routing policies must be added: one for Communication Manager and one for the 3800 Net-Net SBC. To add a routing policy, navigate to **Routing → Routing Policies** in the left-hand 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:

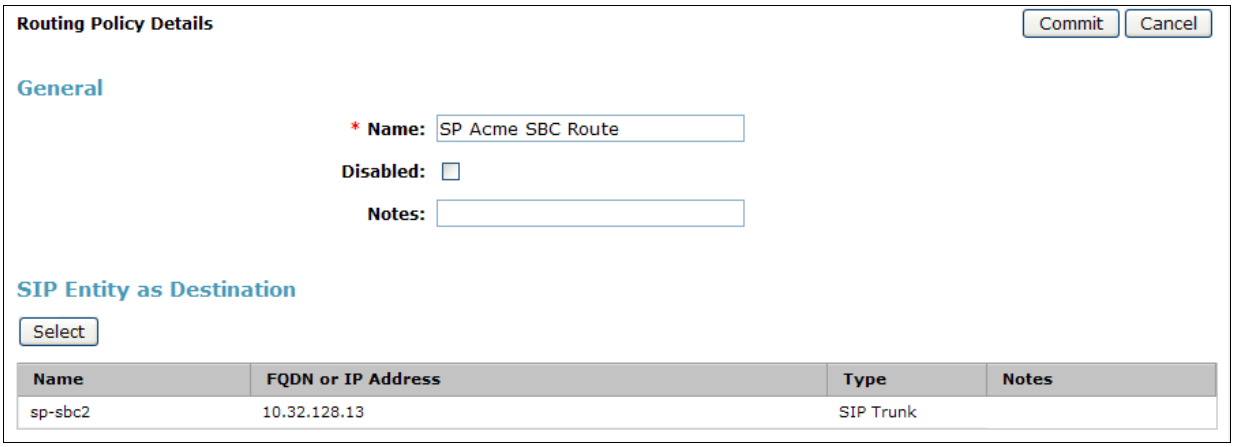
In the **General** section, enter the following values. Use default values for all remaining fields.

- **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 screens show the Routing Policies for Communication Manager and the 3800 Net-Net SBC.





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.

Two examples of the dial patterns used for the compliance test are shown below. The first example shows that 11 digit numbers that begin with a 1 and have a destination domain of **avaya.com** from **ALL** locations uses route policy **SP Acme SBC route**.

Dial Pattern Details Commit Cancel

General

* **Pattern:** 1
* **Min:** 11
* **Max:** 11
Emergency Call: ☐
SIP Domain: avaya.com
Notes:

Originating Locations and Routing Policies

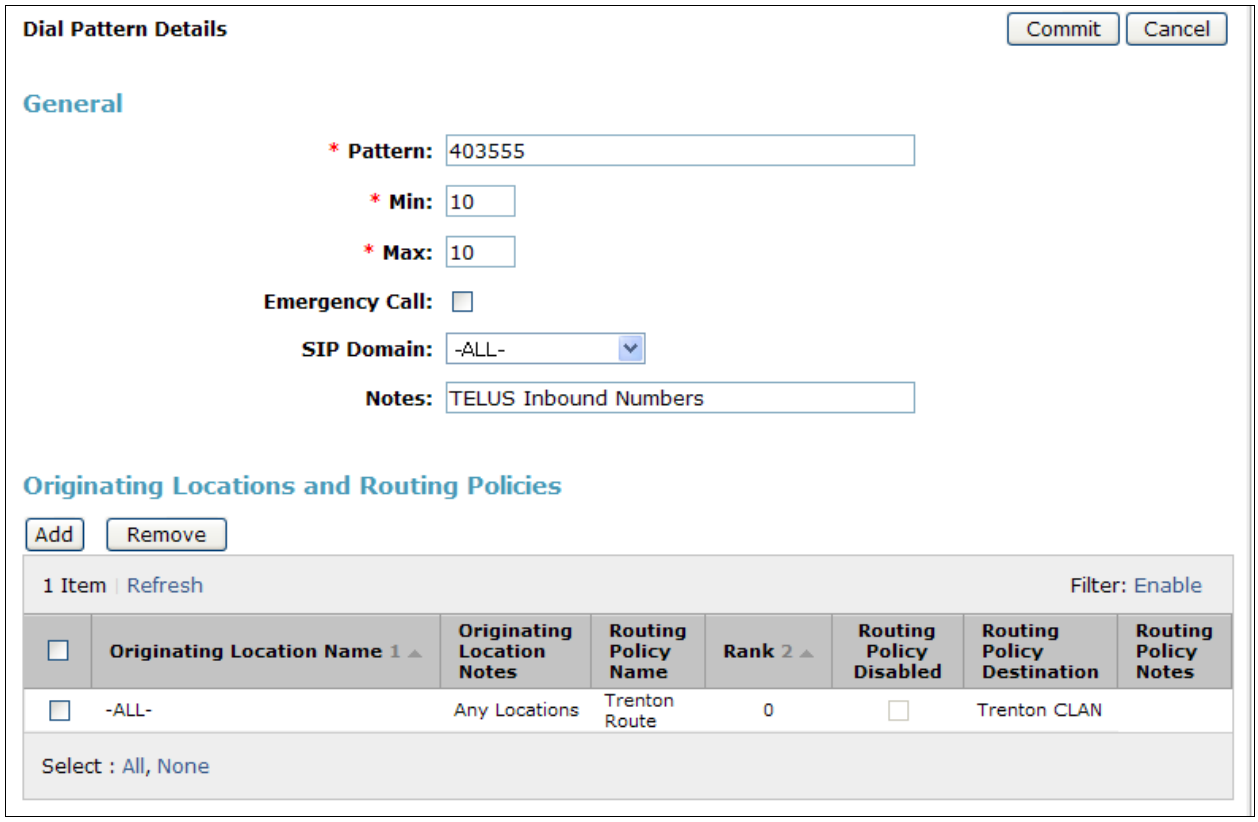
Add Remove

1 Item | Refresh Filter: Enable

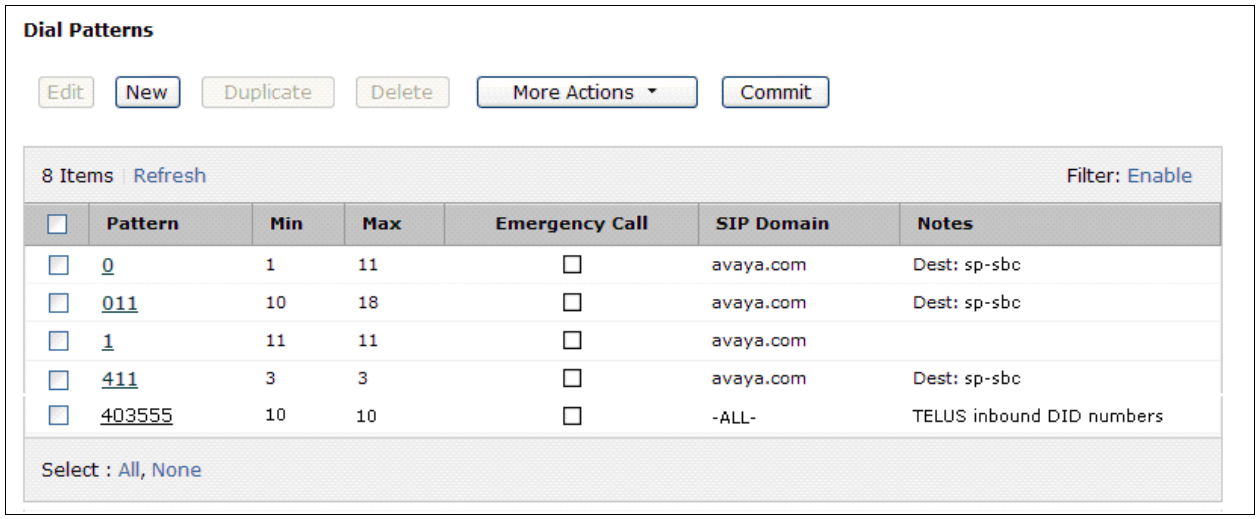
☐	Originating Location Name 1	Originating Location Notes	Routing Policy Name	Rank 2	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
☐	-ALL-	Any Locations	SP Acme SBC Route	0	☐	sp-sbc2	

Select : All, None

The second example shows that 10 digit numbers that start with **403555** to any domain and originating from any location uses route policy **Trenton Route**. These are the DID numbers assigned to the enterprise from TELUS.



The complete list of dial patterns defined for the compliance test is shown below.



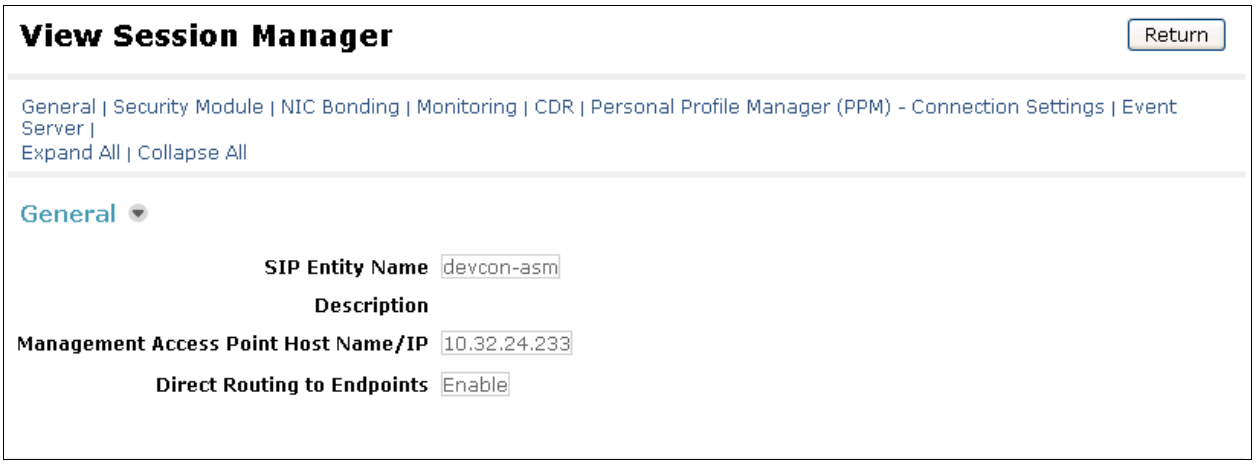
6.9. Add/View Avaya Aura® Session Manager

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 IP address of the Session Manager management interface.

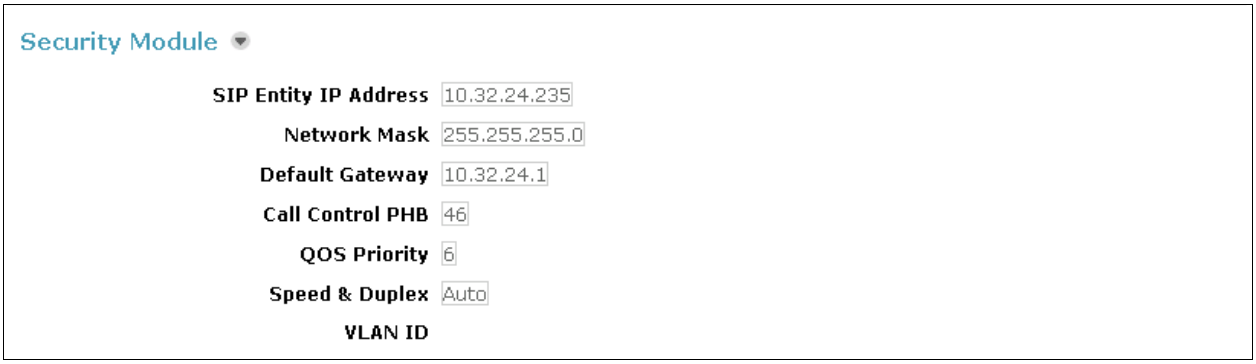
The screen below shows the Session Manager values used for the compliance test.



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 IP address of 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 Acme Packet 3800 Net-Net Session Border Controller

The following sections describe the provisioning of the Acme Packet 3800 Net-Net SBC. Only the Acme Packet provisioning required for the reference configuration is described in these Application Notes. The resulting SBC configuration file is shown in **Appendix A**.

The Acme Packet SBC was configured using the Acme Packet CLI via a serial console port connection. An IP remote connection to a management port is also supported. The following are the generic steps for configuring various elements.

1. Log in with the appropriate credentials.
2. Enable the Superuser mode by entering **enable** and the appropriate password (prompt will end with #).
3. In Superuser mode, type **configure terminal** and press <ENTER>. The prompt will change to (configure)#.
4. Type the name of the element that will be configured (e.g., **session-router**).
5. Type the name of the sub-element, if any (e.g., **session-agent**).
6. Type the name of the parameter followed by its value (e.g., **ip-address**).
7. Type **done**.
8. Type **exit** to return to the previous menu.
9. Repeat steps 4-8 to configure all the elements. When finished, exit from the configuration mode by typing **exit** until returned to the Superuser prompt.
10. Type **save-configuration** to save the configuration.
11. Type **activate-configuration** to activate the configuration.

Once the provisioning is complete, the configuration may be verified by entering the **show running-config** command.

7.1. Physical Interfaces

This section defines the physical interfaces to the private enterprise and public networks.

7.1.1. Public Interface

Create a phy-interface to the public side of the Acme.

1. Enter **system → phy-interface**
2. Enter **name → s0p0**
3. Enter **operation-type → Media**
4. Enter **port → 0**
5. Enter **slot → 0**
6. Enter **duplex-mode → FULL**
7. Enter **speed → 100**
8. Enter **done**
9. Enter **exit**

7.1.2. Private Interface

Create a phy-interface to the private enterprise side of the Acme.

1. Enter **system → phy-interface**
2. Enter **name → s1p0**
3. Enter **operation-type → Media**
4. Enter **port → 0**
5. Enter **slot → 1**
6. **virtual-mac → 00:08:25:a0:f4:8a**
   Virtual MAC addresses are assigned based on the MAC address assigned to the Acme. This MAC address is found by entering the command → **show prom-info mainboard** (e.g. **00 08 25 a0 fa 80**). To define a virtual MAC address, replace the last digit with **8** thru **f**.
7. Enter **duplex-mode → FULL**
8. Enter **speed → 100**
9. Enter **done**
10. Enter **exit**

7.2. Network Interfaces

This section defines the network interfaces to the private enterprise and public IP networks.

7.2.1. Public Interface

Create a network-interface to the public side of the Acme. The compliance test was performed with a direct Internet connection to the service using the settings below.

1. Enter **system → network-interface**
2. Enter **name → s0p0**
3. Enter **ip-address → 10.3.3.225**
4. Enter **netmask → 255.255.255.224**
5. Enter **gateway → 10.3.3.254**
6. Enter **dns-ip-primary → 10.3.184.199**
7. Enter **hip-ip-list → 10.3.3.225**
8. Enter **icmp-ip-list → 10.3.3.225**
9. Enter **done**
10. Enter **exit**

7.2.2. Private Interface

Create a network-interface to the private enterprise side of the Acme.

1. Enter **system → network-interface**
2. Enter **name → s1p0**
3. Enter **ip-address → 10.32.128.13**
4. Enter **netmask → 255.255.255.0**
5. Enter **gateway → 10.32.128.254**
6. Enter **hip-ip-list → 10.32.128.13**

7. Enter **icmp-ip-list → 10.32.128.13**
8. Enter **done**
9. Enter **exit**

7.3. Realms

Realms are used as a basis for determining egress and ingress associations between physical and network interfaces as well as applying header manipulation such as NAT.

7.3.1. Outside Realm

Create a realm for the external network.

1. Enter **media-manager → realm-config**
2. Enter **identifier → EXTERNAL**
3. Enter **network-interfaces → s0p0:0**
4. Enter **done**
5. Enter **exit**

7.3.2. Inside Realm

Create a realm for the internal network.

1. Enter **media-manager → realm-config**
2. Enter **identifier → INTERNAL2**
3. Enter **network-interfaces → s1p0:0**
4. Enter **done**
5. Enter **exit**

7.4. Steering-Pools

Steering pools define sets of ports that are used for steering media flows thru the 3800 Net-Net SBC.

7.4.1. Outside Steering-Pool

Create a steering-pool for the outside network. The start-port and end-port values should specify a range acceptable to the service provider. For the compliance test, no specific range was specified by the service provider, so the start and end ports shown below were chosen arbitrarily.

1. Enter **media-manager → steering-pool**
2. Enter **ip-address → 10.3.3.225**
3. Enter **start-port → 49152**
4. Enter **end-port → 65535**
5. Enter **realm-id → EXTERNAL**
6. Enter **done**
7. Enter **exit**

7.4.2. Inside Steering-Pool

Create a steering-pool for the inside network. The start-port and end-port values should specify a range acceptable to the internal enterprise network and include the port range used by Communication Manager. For the compliance test, a wide range was selected that included the default port range that Communication Manager uses and shown on the ip-network-region form in **Section 5.6**.

1. Enter **media-manager → steering-pool**
2. Enter **ip-address → 10.32.128.13**
3. Enter **start-port → 2048**
4. Enter **end-port → 65535**
5. Enter **realm-id → INTERNAL2**
6. Enter **done**
7. Enter **exit**

7.5. Media-Manager

Verify that the media-manager process is enabled.

1. Enter **media-manager → media-manager**
2. Enter **select → show** Verify that the media-manager state is enabled. If not, perform steps 3 -5.
3. Enter **state → enabled**
4. Enter **done**
5. Enter **exit**

7.6. SIP Configuration

This command sets the values for the 3800 Net-Net SBC SIP operating parameters. The home-realm is the internal default realm for the 3800 Net-Net SBC and the egress-realm is the realm that will be used to send a request if a realm is not specified elsewhere. If the egress-realm is blank, the home-realm is used instead.

1. Enter **session-router → sip-config**
2. Enter **state → enabled**
3. Enter **operation-mode → dialog**
4. Enter **home-realm-id → INTERNAL2**
5. Enter **egress-realm-id →**
6. Enter **nat-mode → Public**
7. Enter **done**
8. Enter **exit**

7.7. SIP Interfaces

The SIP interface defines the SIP signaling interface (IP address and port) on the 3800 Net-Net SBC.

7.7.1. Outside SIP Interface

Create a sip-interface for the outside network.

1. Enter **session-router → sip-interface**
2. Enter **state → enabled**
3. Enter **realm-id → EXTERNAL**
4. Enter **sip-port**
    a. Enter **address → 10.3.3.225**
    b. Enter **port → 5060**
    c. Enter **transport-protocol → UDP**
    d. Enter **allow-anonymous → agents-only**
    e. Enter **done**
    f. Enter **exit**
5. Enter **stop-recurse → 401,407**
6. Enter **done**
7. Enter **exit**

7.7.2. Inside SIP Interface

Create a sip-interface for the inside network.

1. Enter **session-router → sip-interface**
2. Enter **state → enabled**
3. Enter **realm-id → INTERNAL2**
4. Enter **sip-port**
    a. Enter **address → 10.32.128.13**
    b. Enter **port → 5060**
    c. Enter **transport-protocol → TCP**
    d. Enter **allow-anonymous → all**
    e. Enter **done**
    f. Enter **exit**
5. Enter **stop-recurse → 401,407**
6. Enter **done**
7. Enter **exit**

7.8. Session-Agents

A session-agent defines an internal "next hop" signaling entity for the SIP traffic. A realm is associated with a session-agent to identify sessions coming from or going to the session-agent. A session-agent is defined for the service provider (outside) and Session Manager (inside). SIP header manipulations can be applied to the session-agent level.

7.8.1. Outside Session-Agent

Create a session-agent for the outside network.

1. Enter **session-router → session-agent**
2. Enter **hostname → 10.2.2.66**

3. Enter **ip-address → 10.2.2.66**
4. Enter **port → 5060**
5. Enter **state → enabled**
6. Enter **app-protocol → SIP**
7. Enter **transport-method → UDP**
8. Enter **realm-id → EXTERNAL**
9. Enter **description → TELUS**
10. Enter **ping-method → OPTIONS;hops=0**
11. Enter **ping-interval → 60**
12. Enter **ping-send-mode → keep-alive**
13. Enter **in-manipulationid →**
14. Enter **out-manipulationid → outManToSP**
15. Enter **done**
16. Enter **exit**

7.8.2. Inside Session-Agent

Create a session-agent for the inside network.

1. Enter **session-router → session-agent**
2. Enter **hostname → 10.32.24.235**
3. Enter **ip-address → 10.32.24.235**
4. Enter **port → 5060**
5. Enter **state → enabled**
6. Enter **app-protocol → SIP**
7. Enter **transport-method → StaticTCP**
8. Enter **realm-id → INTERNAL2**
9. Enter **description → SM_SPenv**
10. Enter **ping-method →**
11. Enter **ping-interval → 0**
12. Enter **ping-send-mode → keep-alive**
13. Enter **in-manipulationid → inManFromSM**
14. Enter **done**
15. Enter **exit**

7.9. Local Policies

Local policies allow SIP requests from the **INTERNAL2** realm to be routed to the service provider session agent in the **EXTERNAL** realm (and vice-versa).

7.9.1. INTERNAL2 to EXTERNAL

Create a local-policy for the **INSIDE** realm.

1. Enter **session-router → local-policy**
2. Enter **from-address → ***
3. Enter **to-address → ***
4. Enter **source-realm → INTERNAL2**

5. Enter **state → enabled**
6. Enter **policy-attributes**
    a. Enter **next-hop → 10.2.2.66**
    b. Enter **realm → EXTERNAL**
    c. Enter **terminate-recursion → enabled**
    d. Enter **app-protocol → SIP**
    e. Enter **state → enabled**
    f. Enter **done**
    g. Enter **exit**
7. Enter **done**
8. Enter **exit**

7.9.2. EXTERNAL to INTERNAL2

Create a local-policy for the **EXTERNAL** realm.

1. Enter **session-router → local-policy**
2. Enter **from-address → ***
3. Enter **to-address → ***
4. Enter **source-realm → EXTERNAL**
5. Enter **state → enabled**
6. Enter **policy-attributes**
    a. Enter **next-hop → 10.32.24.235**
    b. Enter **realm → INTERNAL2**
    c. Enter **terminate-recursion → enabled**
    d. Enter **app-protocol → SIP**
    e. Enter **state → enabled**
    f. Enter **done**
    g. Enter **exit**
7. Enter **done**
8. Enter **exit**

7.10. SIP Manipulations

SIP manipulation specifies rules for manipulating the contents of specified SIP headers. Two separate sets of SIP manipulations were required for the compliance test listed below.

- inManFromSM – A set of SIP header manipulation rules (HMRs) on traffic from Session Manager to the SBC.
- outManToSP - A set of SIP header manipulation rules on traffic from the SBC to service provider (TELUS).

7.10.1. Session Manager to SBC

The following set of SIP HMRs is applied to traffic from the Session Manager to the SBC. In some call flows the user part of the SIP Contact header received from the Session Manager was not passed unaltered to the public side of the SBC. To correct this, the user part of the Contact header is stored when received from the Session Manager and used to create a temporary header

called X-Contact that will be deleted on the outbound (public) side of the SBC. The information contained in the X-Contact header will be used to recreate the proper Contact header on the public side of the SBC as shown in **Sections 7.10.3.8** and **7.10.3.9**.

To create this set of SIP HMRs:

1. Enter **session-router → sip-manipulation**
2. Enter **name → inManFromSM**
3. Enter **description → "Inbound SIP HMRs From SM"**
4. Proceed to the following sections. Once all sections are completed then proceed with **Steps 5** and **6** below.
5. Enter **done**
6. Enter **exit**

7.10.1.1 Store Contact

This rule stores the user part of the incoming Contact header.

1. Enter **header-rule**
2. Enter **name → strcon**
3. Enter **header-name → Contact**
4. Enter **action → manipulate**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → request**
7. Enter **methods → INVITE,UPDATE**
8. Enter **element-rule**
    a. Enter **name → strval**
    b. Enter **type → uri-user**
    c. Enter **action → store**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → case-sensitive**
    f. Enter **match-value → (.*)**
    g. Enter **done**
    h. Enter **exit**
9. Enter **done**
10. Enter **exit**

7.10.1.2 Create X-Contact

This rule creates a temporary header called X-Contact containing only the user part of the incoming Contact header as stored by the rule defined in the previous section.

1. Enter **header-rule**
2. Enter **name → addXcontact**
3. Enter **header-name → X-Contact**
4. Enter **action → add**
5. Enter **comparison-type → pattern-rule**

6. Enter **msg-type → request**
7. Enter **methods → INVITE,UPDATE**
8. Enter **element-rule**
    a. Enter **name → addX**
    b. Enter **type → header-value**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → pattern-rule**
    f. Enter **new-value → $strcon.$strval.$0**
    g. Enter **done**
    h. Enter **exit**
9. Enter **done**
10. Enter **exit**

7.10.2. SBC to TELUS

The following set of SIP HMRs is applied to traffic from the SBC to TELUS.

To create this set of SIP HMRs:

1. Enter **session-router → sip-manipulation**
2. Enter **name → outManFromSP**
3. Enter **description → "outbound SIP HMRs From SP"**
4. Proceed to the following sections. Once all sections are completed then proceed with **Steps 5** and **6** below.
5. Enter **done**
6. Enter **exit**

7.10.2.1 Change Host of the To Header

This rule replaces the host part of the To header with the service provider's IP address. A similar manipulation is performed on the Request-URI by the Session Manager. The Request-URI could have also been manipulated by the SBC.

1. Enter **header-rule**
2. Enter **name → manipTo**
3. Enter **header-name → To**
4. Enter **action → manipulate**
5. Enter **comparison-type → pattern-rule**
6. Enter **msg-type → request**
7. Enter **element-rule**
    a. Enter **name → chgToHost**
    b. Enter **type → uri-host**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → case-sensitive**
    f. Enter **new-value → $REMOTE_IP**

g. Enter **done**
h. Enter **exit**

8. Enter **done**
9. Enter **exit**

7.10.2.2 Change Host of the From Header

This rule replaces the host part of the From header with the public IP address of the SBC.

1. Enter **header-rule**
2. Enter **name → manipFrom**
3. Enter **header-name → From**
4. Enter **action → manipulate**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → request**
7. Enter **element-rule**
    a. Enter **name → From**
    b. Enter **type → uri-host**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → case-sensitive**
    f. Enter **new-value → $LOCAL_IP**
    g. Enter **done**
    h. Enter **exit**
8. Enter **done**
9. Enter **exit**

7.10.2.3 Change Host of the History Info Header

This rule replaces the host part of the History-Info header with the public IP address of the SBC.

1. Enter **header-rule**
2. Enter **name → manipHistInfo**
3. Enter **header-name → History-Info**
4. Enter **action → manipulate**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → request**
7. Enter **element-rule**
    a. Enter **name → HistoryInfo**
    b. Enter **type → uri-host**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → case-sensitive**
    f. Enter **new-value → $LOCAL_IP**
    g. Enter **done**
    h. Enter **exit**
8. Enter **done**

9. Enter **exit**

7.10.2.4 Change Host of the PAI Header

This rule replaces the host part of the P-Asserted-Identity header with the public IP address of the SBC.

1. Enter **header-rule**
2. Enter **name → manipPAI**
3. Enter **header-name → P-Asserted-Identity**
4. Enter **action → manipulate**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → request**
7. Enter **element-rule**
    a. Enter **name → Pai**
    b. Enter **type → uri-host**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → case-sensitive**
    f. Enter **new-value → $LOCAL_IP**
    g. Enter **done**
    h. Enter **exit**
8. Enter **done**
9. Enter **exit**

7.10.2.5 Change Host of the Diversion Header

This rule replaces the host part of the Diversion header with the public IP address of the SBC.

1. Enter **header-rule**
2. Enter **name → manipDiversion**
3. Enter **header-name → Diversion**
4. Enter **action → manipulate**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → request**
7. Enter **element-rule**
    a. Enter **name → Diversion**
    b. Enter **type → uri-host**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → case-sensitive**
    f. Enter **new-value → $LOCAL_IP**
    g. Enter **done**
    h. Enter **exit**
8. Enter **done**
9. Enter **exit**

7.10.2.6 Change Host of the Refer-To Header

This rule replaces the host part of the Refer-To header with the service provider's IP address.

1. Enter **header-rule**
2. Enter **name → manipRefer**
3. Enter **header-name → Refer-To**
4. Enter **action → manipulate**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → request**
7. Enter **element-rule**
    a. Enter **name → chgHostRefer**
    b. Enter **type → uri-host**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → case-sensitive**
    f. Enter **new-value → $REMOTE_IP**
    g. Enter **done**
    h. Enter **exit**
8. Enter **done**
9. Enter **exit**

7.10.2.7 Store X-Contact Header

This rule stores the contents of the X-Contact header so it can be used later. The X-Contact header contains only the user part of the Contact header as it was originally received from the Session Manager as described in **Section 7.10.1**.

1. Enter **header-rule**
2. Enter **name → storexcontact**
3. Enter **header-name → X-Contact**
4. Enter **action → manipulate**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → request**
7. Enter **methods → INVITE,UPDATE**
8. Enter **element-rule**
    a. Enter **name → storexcontact**
    b. Enter **type → header-value**
    c. Enter **action → store**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → case-sensitive**
    f. Enter **match-value → (.*)**
    g. Enter **done**
    h. Enter **exit**
9. Enter **done**
10. Enter **exit**

7.10.2.8 Replace Contact Header

This rule uses the data stored from the X-Contact header to overwrite the user part of the outbound Contact header.

1. Enter **header-rule**
2. Enter **name → replacecontact**
3. Enter **header-name → Contact**
4. Enter **action → manipulate**
5. Enter **comparison-type → pattern-rule**
6. Enter **msg-type → request**
7. Enter **methods → INVITE,UPDATE**
8. Enter **element-rule**
    a. Enter **name → replacecontact**
    b. Enter **type → uri-user**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → pattern-rule**
    f. Enter **match-value → (.*)**
    g. Enter **new-value $storexcontact.$storexcontact.$0**
    h. Enter **done**
    i. Enter **exit**
9. Enter **done**
10. Enter **exit**

7.10.2.9 Delete X-Contact Header

This rule deletes the temporary X-Contact header before sending the message to the service provider.

1. Enter **header-rule**
2. Enter **name → delxcontact**
3. Enter **header-name → X-Contact**
4. Enter **action → delete**
5. Enter **comparison-type → pattern-rule**
6. Enter **msg-type → request**
7. Enter **methods → INVITE,UPDATE**
8. Enter **done**
9. Enter **exit**

7.10.2.10 Delete P-Location Header

This rule deletes the P-Location header. This header is not used by the service provider and it may contain internal IP addresses which should not be shared outside of the enterprise. Thus, the header was removed.

1. Enter **header-rule**
2. Enter **name → delPloc**

3. Enter **header-name → P-Location**
4. Enter **action → delete**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → any**
7. Enter **methods →**
8. Enter **done**
9. Enter **exit**

7.10.2.11 Delete Alert-Info Header

This rule deletes the Alert-Info header. This header is not used by the service provider and it may contain internal IP addresses which should not be shared outside of the enterprise. Thus, the header was removed.

1. Enter **header-rule**
2. Enter **name → delAlert**
3. Enter **header-name → Alert-Info**
4. Enter **action → delete**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → any**
7. Enter **methods →**
8. Enter **done**
9. Enter **exit**

7.10.2.12 Store Diversion Header User and Display Name

This rule stores the user and the display name parts of the Diversion header to be used in the next section to modify the PAI header. This is necessary to complete calls from the PSTN which are redirected back to the PSTN. See **Section 2.2** for more details.

1. Enter **header-rule**
2. Enter **name → strDivNum**
3. Enter **header-name → Diversion**
4. Enter **action → manipulate**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → request**
7. Enter **methods → INVITE**
8. Enter **element-rule**
    a. Enter **name → strval**
    b. Enter **type → uri-user**
    c. Enter **action → store**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → case-sensitive**
    f. Enter **match-value → (.*)**
    g. Enter **done**
    h. Enter **exit**
9. Enter **element-rule**

a. Enter **name → strDisName**
b. Enter **type → uri-display**
c. Enter **action → store**
d. Enter **match-val-type → any**
e. Enter **comparison-type → case-sensitive**
f. Enter **match-value → (.*)**
g. Enter **done**
h. Enter **exit**

10. Enter **done**
11. Enter **exit**

7.10.2.13 Modify PAI Header with Diversion Header Information

This rule modifies the user and the display name parts of the PAI header using the information stored from the Diversion header in the last section. It only modifies the PAI in the INVITE message if the Diversion header is present. This is determined by checking if any value was stored for the Diversion user from the previous rule. This rule is necessary to complete calls from the PSTN which are redirected back to the PSTN. See **Section 2.2** for more details.

1. Enter **header-rule**
2. Enter **name → chgPAIuser**
3. Enter **header-name → P-Asserted-Identity**
4. Enter **action → manipulate**
5. Enter **comparison-type → boolean**
6. Enter **msg-type → request**
7. Enter **methods → INVITE**
8. Enter **match-value → $strDivNum.$strval.$0**
9. Enter **element-rule**
    a. Enter **name → chgUserFromDiv**
    b. Enter **type → uri-user**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → pattern-rule**
    f. Enter **match-value → (.*)**
    g. Enter **new-value → $strDivNum.$strval.$0**
    h. Enter **done**
    i. Enter **exit**
10. Enter **element-rule**
    a. Enter **name → chgDispName**
    b. Enter **type → uri-display**
    c. Enter **action → replace**
    d. Enter **match-val-type → any**
    e. Enter **comparison-type → pattern-rule**
    f. Enter **match-value → (.*)**
    g. Enter **$strDivNum.$strDisName.$0**
    h. Enter **done**

i. Enter **exit**
11. Enter **done**
12. Enter **exit**

7.10.2.14 Delete Diversion Header

This rule deletes the Diversion header. The Diversion header is no longer needed once the PAI has been modified. It may also cause unexpected behavior if forwarded on to the TELUS network.

1. Enter **header-rule**
2. Enter **name → delDiv**
3. Enter **header-name → Diversion**
4. Enter **action → delete**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → any**
7. Enter **methods →**
8. Enter **done**
9. Enter **exit**

7.10.2.15 Modify Max-Forwards in OPTIONS message

This rule changes the value of the Max-Forward header to zero in the OPTIONS message. A Max-Forwards value of zero is a requirement of the TELUS network. See **Section 2.2** for more details.

1. Enter **header-rule**
2. Enter **name → OptZeroMaxFwd**
3. Enter **header-name → Max-Forwards**
4. Enter **action → manipulate**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → request**
7. Enter **methods → OPTIONS**
    a. Enter **element-rule**
    b. Enter **name → zeroMaxFwd**
    c. Enter **type → header-value**
    d. Enter **action → replace**
    e. Enter **match-val-type → any**
    f. Enter **comparison-type → case-sensitive**
    g. Enter **match-value →**
    h. Enter **new-value → 0**
    i. Enter **done**
    j. Enter **exit**
8. Enter **done**
9. Enter **exit**

8. TELUS SIP Trunking Configuration

TELUS is responsible for the network configuration of the TELUS SIP Trunking service. TELUS will require that the customer provide the public IP address used to reach the 3800 Net-Net SBC at the edge of the enterprise. TELUS will provide the IP address of the TELUS SIP proxy/SBC, IP addresses 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 the SBC configuration discussed in the previous sections.

The configuration between TELUS and the enterprise is a static configuration. There is no registration of the SIP trunk or enterprise users to the TELUS network.

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 the 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 trunk group information.
   - **status trunk** <trunk access code number/channel number> - Displays 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.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Acme Packet 3800 Net-Net Session Border Controller to TELUS SIP Trunking. This solution successfully passed compliance testing via the Avaya DevConnect Program. 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 6.0.3, February 2011.
[2] *Administering Avaya Aura® System Platform*, Release 6.0.3, February 2011.
[3] *Administering Avaya Aura® Communication Manager*, May 2009, Document Number 03-300509.
[4] *Avaya Aura® Communication Manager Feature Description and Implementation,* May 2009, *D*ocument Number 555-245-205.
[5] *Installing and Upgrading Avaya Aura® System Manager*, Release 6.1, November 2010.
[6] *Administering Avaya Aura® System Manager*, Release 6.1, November 2010.
[7] *Installing and Configuring Avaya Aura® Session Manager*, Release 6.1, April 2011, Document Number 03-603473.
[8] *Administering Avaya Aura® Session Manager,* Release 6.1, October 2011, Document Number 03-603324.
[9] *Avaya 1600 Series IP Deskphones Administrator Guide Release Release 1.3.x*, May 2010, Document Number 16-601443.
[10] *4600 Series IP Telephone LAN Administrator Guide,* October 2007, Document Number 555-233-507.
[11] *Avaya one-X® Deskphone Edition for 9600 Series IP Telephones Administrator Guide,* November 2009, Document Number 16-300698.
[12] *Administering Avaya one-X® Communicator*, July 2011.
[13] RFC 3261 *SIP: Session Initiation Protocol,* http://www.ietf.org/
[14] RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, http://www.ietf.org/

Appendix A: Acme Packet 3800 Net-Net SBC Configuration File

```
host-routes
       dest-network                  10.1.2.0
       netmask                       255.255.255.0
       gateway                       10.32.128.254
       description
       last-modified-by              admin@192.168.168.37
       last-modified-date            2011-10-27 16:57:53
host-routes
       dest-network                  10.32.0.0
       netmask                       255.255.0.0
       gateway                       10.32.128.254
       description                   DevConnectLAN
       last-modified-by              admin@135.11.141.118
       last-modified-date            2010-08-05 15:25:58
host-routes
       dest-network                  192.168.0.0
       netmask                       255.255.0.0
       gateway                       10.32.128.254
       description                   Route to remote testers
       last-modified-by              admin@192.168.168.37
       last-modified-date            2011-09-10 10:50:25
local-policy
       from-address
                                     *
       to-address
                                     *
       source-realm
                                     INTERNAL2
       description
       activate-time                 N/A
       deactivate-time               N/A
       state                         enabled
       policy-priority               none
       last-modified-by              admin@192.168.168.37
       last-modified-date            2011-12-19 13:06:16
       policy-attribute
              next-hop                      10.2.2.66
              realm                         EXTERNAL
              action                        none
              terminate-recursion           enabled
              carrier
              start-time                    0000
              end-time                      2400
              days-of-week                  U-S
              cost                          0
              app-protocol                  SIP
              state                         enabled
              methods
              media-profiles
              lookup                        single
              next-key
              eloc-str-lkup                 disabled
              eloc-str-match
local-policy
       from-address
                                     *
```

```
        to-address
                                      *
        source-realm
                                      EXTERNAL
        description
        activate-time                 N/A
        deactivate-time               N/A
        state                         enabled
        policy-priority               none
        last-modified-by              admin@192.168.168.37
        last-modified-date            2011-10-27 17:17:00
        policy-attribute
              next-hop                      10.32.24.235
              realm                         INTERNAL2
              action                        none
              terminate-recursion           enabled
              carrier
              start-time                    0000
              end-time                      2400
              days-of-week                  U-S
              cost                          0
              app-protocol                  SIP
              state                         enabled
              methods
              media-profiles
              lookup                        single
              next-key
              eloc-str-lkup                 disabled
              eloc-str-match
media-manager
        state                         enabled
        latching                      enabled
        flow-time-limit               86400
        initial-guard-timer           300
        subsq-guard-timer             300
        tcp-flow-time-limit           86400
        tcp-initial-guard-timer       300
        tcp-subsq-guard-timer         300
        tcp-number-of-ports-per-flow  2
        hnt-rtcp                      disabled
        algd-log-level                NOTICE
        mbcd-log-level                NOTICE
        red-flow-port                 1985
        red-mgcp-port                 1986
        red-max-trans                 10000
        red-sync-start-time           5000
        red-sync-comp-time            1000
        media-policing                enabled
        max-signaling-bandwidth       10000000
        max-untrusted-signaling       100
        min-untrusted-signaling       30
        app-signaling-bandwidth       0
        tolerance-window              30
        rtcp-rate-limit               0
        trap-on-demote-to-deny        enabled
        min-media-allocation          2000
        min-trusted-allocation        4000
        deny-allocation               64000
        anonymous-sdp                 disabled
        arp-msg-bandwidth             32000
        fragment-msg-bandwidth        0
        rfc2833-timestamp             disabled
```

```
       default-2833-duration          100
       rfc2833-end-pkts-only-for-non-sig enabled
       translate-non-rfc2833-event    disabled
       media-supervision-traps        disabled
       dnsalg-server-failover         disabled
       last-modified-by               admin@135.11.141.142
       last-modified-date             2010-06-16 05:40:01
network-interface
       name                           s0p0
       sub-port-id                    0
       description
       hostname
       ip-address                     10.3.3.225
       pri-utility-addr
       sec-utility-addr
       netmask                        255.255.255.224
       gateway                        10.3.3.254
       sec-gateway
       gw-heartbeat
             state                          disabled
             heartbeat                      0
             retry-count                    0
             retry-timeout                  1
             health-score                   0
       dns-ip-primary                 10.3.184.199
       dns-ip-backup1                 10.3.184.7
       dns-ip-backup2
       dns-domain
       dns-timeout                    11
        hip-ip-list                    10.3.3.225
       ftp-address
        icmp-address                   10.3.3.225
       snmp-address
       telnet-address
       ssh-address
       last-modified-by               admin@192.168.168.37
       last-modified-date             2011-09-10 10:08:47
network-interface
       name                           s1p0
       sub-port-id                    0
       description
       hostname
       ip-address                     10.32.128.13
       pri-utility-addr
       sec-utility-addr
       netmask                        255.255.255.0
       gateway                        10.32.128.254
       sec-gateway
       gw-heartbeat
             state                          disabled
             heartbeat                      0
             retry-count                    0
             retry-timeout                  1
             health-score                   0
       dns-ip-primary
       dns-ip-backup1
       dns-ip-backup2
       dns-domain
       dns-timeout                    11
        hip-ip-list                    10.32.128.13
       ftp-address                    10.32.128.13
        icmp-address                   10.32.128.13
```

```
       snmp-address
       telnet-address                10.32.128.13
       ssh-address
       last-modified-by              admin@192.168.168.37
       last-modified-date            2011-11-03 11:42:43
phy-interface
       name                          s0p0
       operation-type                Media
       port                          0
       slot                          0
       virtual-mac
       admin-state                   enabled
       auto-negotiation              enabled
       duplex-mode
       speed
       overload-protection           disabled
       last-modified-by              admin@console
       last-modified-date            2011-09-09 19:39:05
phy-interface
       name                          s1p0
       operation-type                Media
       port                          0
       slot                          1
       virtual-mac                   00:08:25:a0:f4:8a
       admin-state                   enabled
       auto-negotiation              enabled
       duplex-mode                   FULL
       speed                         100
       overload-protection           disabled
       last-modified-by              admin@console
       last-modified-date            2011-09-09 19:38:24
realm-config
       identifier                    EXTERNAL
       description
       addr-prefix                   0.0.0.0
       network-interfaces
                                     s0p0:0
       mm-in-realm                   disabled
       mm-in-network                 enabled
       mm-same-ip                    enabled
       mm-in-system                  enabled
       bw-cac-non-mm                 disabled
       msm-release                   disabled
       generate-UDP-checksum         disabled
       max-bandwidth                 0
       fallback-bandwidth            0
       max-priority-bandwidth        0
       max-latency                   0
       max-jitter                    0
       max-packet-loss               0
       observ-window-size            0
       parent-realm
       dns-realm
       media-policy
       media-sec-policy
       in-translationid
       out-translationid
       in-manipulationid
       out-manipulationid
       manipulation-string
       manipulation-pattern
       class-profile
```

```
        average-rate-limit              0
        access-control-trust-level      none
        invalid-signal-threshold        0
        maximum-signal-threshold        0
        untrusted-signal-threshold      0
        nat-trust-threshold             0
        deny-period                     30
        ext-policy-svr
        symmetric-latching              disabled
        pai-strip                       disabled
        trunk-context
        early-media-allow
        enforcement-profile
        additional-prefixes
        restricted-latching             none
        restriction-mask                32
        accounting-enable               enabled
        user-cac-mode                   none
        user-cac-bandwidth              0
        user-cac-sessions               0
        icmp-detect-multiplier          0
        icmp-advertisement-interval     0
        icmp-target-ip
        monthly-minutes                 0
        net-management-control          disabled
        delay-media-update              disabled
        refer-call-transfer             disabled
        dyn-refer-term                  disabled
        codec-policy
        codec-manip-in-realm            disabled
        constraint-name
        call-recording-server-id
        xnq-state                       xnq-unknown
        hairpin-id                      0
        stun-enable                     disabled
        stun-server-ip                  0.0.0.0
        stun-server-port                3478
        stun-changed-ip                 0.0.0.0
        stun-changed-port               3479
        match-media-profiles
        qos-constraint
        sip-profile
        sip-isup-profile
        block-rtcp                      disabled
        hide-egress-media-update        disabled
        last-modified-by                admin@135.11.207.156
        last-modified-date              2010-11-03 08:55:21
realm-config
        identifier                      INTERNAL2
        description
        addr-prefix                     0.0.0.0
        network-interfaces
                                        s1p0:0
        mm-in-realm                     disabled
        mm-in-network                   enabled
        mm-same-ip                      enabled
        mm-in-system                    enabled
        bw-cac-non-mm                   disabled
        msm-release                     disabled
        generate-UDP-checksum           disabled
        max-bandwidth                   0
        fallback-bandwidth              0
```

```
max-priority-bandwidth          0
max-latency                     0
max-jitter                      0
max-packet-loss                 0
observ-window-size              0
parent-realm
dns-realm
media-policy
media-sec-policy
in-translationid
out-translationid
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
class-profile
average-rate-limit              0
access-control-trust-level      none
invalid-signal-threshold        0
maximum-signal-threshold        0
untrusted-signal-threshold      0
nat-trust-threshold             0
deny-period                     30
ext-policy-svr
symmetric-latching              disabled
pai-strip                       disabled
trunk-context
early-media-allow
enforcement-profile
additional-prefixes
restricted-latching             none
restriction-mask                32
accounting-enable               enabled
user-cac-mode                   none
user-cac-bandwidth              0
user-cac-sessions               0
icmp-detect-multiplier          0
icmp-advertisement-interval     0
icmp-target-ip
monthly-minutes                 0
net-management-control          disabled
delay-media-update              disabled
refer-call-transfer             disabled
dyn-refer-term                  disabled
codec-policy
codec-manip-in-realm            disabled
constraint-name
call-recording-server-id
xnq-state                       xnq-unknown
hairpin-id                      0
stun-enable                     disabled
stun-server-ip                  0.0.0.0
stun-server-port                3478
stun-changed-ip                 0.0.0.0
stun-changed-port               3479
match-media-profiles
qos-constraint
sip-profile
sip-isup-profile
block-rtcp                      disabled
hide-egress-media-update        disabled
last-modified-by                admin@135.11.207.156
```

```
        last-modified-date              2010-12-16 17:25:01
session-agent
        hostname                        10.32.24.235
        ip-address                      10.32.24.235
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                StaticTCP
        realm-id                        INTERNAL2
        egress-realm-id
        description                     SM_SPenv
        carriers
        allow-next-hop-lp               enabled
        constraints                     disabled
        max-sessions                    0
        max-inbound-sessions            0
        max-outbound-sessions           0
        max-burst-rate                  0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
        max-sustain-rate                0
        max-inbound-sustain-rate        0
        max-outbound-sustain-rate       0
        min-seizures                    5
        min-asr                         0
        time-to-resume                  0
        ttr-no-response                 0
        in-service-period               0
        burst-rate-window               0
        sustain-rate-window             0
        req-uri-carrier-mode            None
        proxy-mode
        redirect-action
        loose-routing                   enabled
        send-media-session              enabled
        response-map
        ping-method
        ping-interval                   0
        ping-send-mode                  keep-alive
        ping-all-addresses              disabled
        ping-in-service-response-codes
        out-service-response-codes
        media-profiles
        in-translationid
        out-translationid
        trust-me                        disabled
        request-uri-headers
        stop-recurse
        local-response-map
        ping-to-user-part
        ping-from-user-part
        li-trust-me                     disabled
        in-manipulationid               inManFromSM
        out-manipulationid
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
```

```
        rfc2833-mode                    none
        rfc2833-payload                 0
        codec-policy
        enforcement-profile
        refer-call-transfer             disabled
        reuse-connections               NONE
        tcp-keepalive                   none
        tcp-reconn-interval             0
        max-register-burst-rate         0
        register-burst-window           0
        sip-profile
        sip-isup-profile
        last-modified-by                admin@192.168.168.37
        last-modified-date              2011-09-20 22:39:03
session-agent
        hostname                        10.2.2.66
        ip-address                      10.2.2.66
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                UDP
        realm-id                        EXTERNAL
        egress-realm-id
        description                     TELUS
        carriers
        allow-next-hop-lp               enabled
        constraints                     disabled
        max-sessions                    0
        max-inbound-sessions            0
        max-outbound-sessions           0
        max-burst-rate                  0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
        max-sustain-rate                0
        max-inbound-sustain-rate        0
        max-outbound-sustain-rate       0
        min-seizures                    5
        min-asr                         0
        time-to-resume                  0
        ttr-no-response                 0
        in-service-period               0
        burst-rate-window               0
        sustain-rate-window             0
        req-uri-carrier-mode            None
        proxy-mode
        redirect-action
        loose-routing                   enabled
        send-media-session              enabled
        response-map
        ping-method                     OPTIONS;hops=0
        ping-interval                   60
        ping-send-mode                  keep-alive
        ping-all-addresses              disabled
        ping-in-service-response-codes
        out-service-response-codes
        media-profiles
        in-translationid
        out-translationid
        trust-me                        disabled
        request-uri-headers
        stop-recurse
```

```
        local-response-map
        ping-to-user-part
        ping-from-user-part
        li-trust-me                     disabled
        in-manipulationid
        out-manipulationid              outManToSP
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
        rfc2833-mode                    none
        rfc2833-payload                 0
        codec-policy
        enforcement-profile
        refer-call-transfer             disabled
        reuse-connections               NONE
        tcp-keepalive                   none
        tcp-reconn-interval             0
        max-register-burst-rate         0
        register-burst-window           0
        sip-profile
        sip-isup-profile
        last-modified-by                admin@192.168.168.37
        last-modified-date              2011-10-10 12:21:24
sip-config
        state                           enabled
        operation-mode                  dialog
        dialog-transparency             enabled
        home-realm-id                   INTERNAL2
        egress-realm-id
        nat-mode                        Public
        registrar-domain                *
        registrar-host                  *
        registrar-port                  5060
        register-service-route          always
        init-timer                      500
        max-timer                       4000
        trans-expire                    32
        invite-expire                   180
        inactive-dynamic-conn           32
        enforcement-profile
        pac-method
        pac-interval                    10
        pac-strategy                    PropDist
        pac-load-weight                 1
        pac-session-weight              1
        pac-route-weight                1
        pac-callid-lifetime             600
        pac-user-lifetime               3600
        red-sip-port                    1988
        red-max-trans                   10000
        red-sync-start-time             5000
        red-sync-comp-time              1000
        add-reason-header               disabled
        sip-message-len                 4096
        enum-sag-match                  disabled
        extra-method-stats              enabled
        registration-cache-limit        0
        register-use-to-for-lp          disabled
```

```
      options                         max-udp-length=0
      refer-src-routing               disabled
      add-ucid-header                 disabled
      proxy-sub-events
      pass-gruu-contact               disabled
      sag-lookup-on-redirect          disabled
      last-modified-by                admin@135.11.207.156
      last-modified-date              2010-11-02 16:18:33
sip-interface
      state                           enabled
      realm-id                        EXTERNAL
      description
      sip-port
            address                         10.3.3.225
            port                            5060
            transport-protocol              UDP
            tls-profile
            allow-anonymous                 agents-only
            ims-aka-profile
      carriers
      trans-expire                    0
      invite-expire                   0
      max-redirect-contacts           0
      proxy-mode
      redirect-action
      contact-mode                    none
      nat-traversal                   none
      nat-interval                    30
      tcp-nat-interval                90
      registration-caching            disabled
      min-reg-expire                  300
      registration-interval           3600
      route-to-registrar              disabled
      secured-network                 disabled
      teluri-scheme                   disabled
      uri-fqdn-domain
      trust-mode                      all
      max-nat-interval                3600
      nat-int-increment               10
      nat-test-increment              30
      sip-dynamic-hnt                 disabled
      stop-recurse                    401,407
      port-map-start                  0
      port-map-end                    0
      in-manipulationid
      out-manipulationid
      manipulation-string
      manipulation-pattern
      sip-ims-feature                 disabled
      operator-identifier
      anonymous-priority              none
      max-incoming-conns              0
      per-src-ip-max-incoming-conns   0
      inactive-conn-timeout           0
      untrusted-conn-timeout          0
      network-id
      ext-policy-server
      default-location-string
      charging-vector-mode            pass
      charging-function-address-mode  pass
      ccf-address
      ecf-address
```

```
      term-tgrp-mode                 none
      implicit-service-route         disabled
      rfc2833-payload                101
      rfc2833-mode                   transparent
      constraint-name
      response-map
      local-response-map
      ims-aka-feature                disabled
      enforcement-profile
      route-unauthorized-calls
      tcp-keepalive                  none
      add-sdp-invite                 disabled
      add-sdp-profiles
      sip-profile
      sip-isup-profile
      last-modified-by               admin@192.168.168.37
      last-modified-date             2011-11-18 10:38:42
sip-interface
      state                          enabled
      realm-id                       INTERNAL2
      description
      sip-port
            address                        10.32.128.13
            port                           5060
            transport-protocol             TCP
            tls-profile
            allow-anonymous                all
            ims-aka-profile
      carriers
      trans-expire                   0
      invite-expire                  0
      max-redirect-contacts          0
      proxy-mode
      redirect-action
      contact-mode                   none
      nat-traversal                  none
      nat-interval                   30
      tcp-nat-interval               90
      registration-caching           disabled
      min-reg-expire                 300
      registration-interval          3600
      route-to-registrar             disabled
      secured-network                disabled
      teluri-scheme                  disabled
      uri-fqdn-domain
      trust-mode                     all
      max-nat-interval               3600
      nat-int-increment              10
      nat-test-increment             30
      sip-dynamic-hnt                disabled
      stop-recurse                   401,407
      port-map-start                 0
      port-map-end                   0
      in-manipulationid
      out-manipulationid
      manipulation-string
      manipulation-pattern
      sip-ims-feature                disabled
      operator-identifier
      anonymous-priority             none
      max-incoming-conns             0
      per-src-ip-max-incoming-conns  0
```

```
        inactive-conn-timeout           0
        untrusted-conn-timeout          0
        network-id
        ext-policy-server
        default-location-string
        charging-vector-mode            pass
        charging-function-address-mode pass
        ccf-address
        ecf-address
        term-tgrp-mode                  none
        implicit-service-route          disabled
        rfc2833-payload                 101
        rfc2833-mode                    transparent
        constraint-name
        response-map
        local-response-map
        ims-aka-feature                 disabled
        enforcement-profile
        route-unauthorized-calls
        tcp-keepalive                   none
        add-sdp-invite                  disabled
        add-sdp-profiles
        sip-profile
        sip-isup-profile
        last-modified-by                admin@135.105.224.163
        last-modified-date              2011-08-03 16:00:53
sip-manipulation
        name                            outManToSP
        description                     Outbound SIP HMRs To SP
        split-headers
        join-headers
        header-rule
                name                            manipTo
                header-name                     To
                action                          manipulate
                comparison-type                 pattern-rule
                msg-type                        request
                methods
                match-value
                new-value
                element-rule
                        name                            chgToHost
                        parameter-name
                        type                            uri-host
                        action                          replace
                        match-val-type                  any
                        comparison-type                 case-sensitive
                        match-value
                        new-value                       $REMOTE_IP
        header-rule
                name                            manipFrom
                header-name                     From
                action                          manipulate
                comparison-type                 case-sensitive
                msg-type                        request
                methods
                match-value
                new-value
                element-rule
                        name                            From
                        parameter-name
                        type                            uri-host
```

```
              action                          replace
              match-val-type                  any
              comparison-type                 case-sensitive
              match-value
              new-value                       $LOCAL_IP
header-rule
       name                             manipDiversion
       header-name                      Diversion
       action                           manipulate
       comparison-type                  case-sensitive
       msg-type                         request
       methods
       match-value
       new-value
       element-rule
              name                             Diversion
              parameter-name
              type                             uri-host
              action                           replace
              match-val-type                   any
              comparison-type                  case-sensitive
              match-value
              new-value                        $LOCAL_IP
header-rule
       name                             manipHistInfo
       header-name                      History-Info
       action                           manipulate
       comparison-type                  case-sensitive
       msg-type                         request
       methods
       match-value
       new-value
       element-rule
              name                             HistoryInfo
              parameter-name
              type                             uri-host
              action                           replace
              match-val-type                   any
              comparison-type                  case-sensitive
              match-value
              new-value                        $LOCAL_IP
header-rule
       name                             manipPAI
       header-name                      P-Asserted-Identity
       action                           manipulate
       comparison-type                  case-sensitive
       msg-type                         request
       methods
       match-value
       new-value
       element-rule
              name                             Pai
              parameter-name
              type                             uri-host
              action                           replace
              match-val-type                   any
              comparison-type                  case-sensitive
              match-value
              new-value                        $LOCAL_IP
header-rule
       name                             storeXcontact
       header-name                      X-Contact
```

```
        action                          manipulate
        comparison-type                 case-sensitive
        msg-type                        request
        methods                         INVITE,UPDATE
        match-value
        new-value
        element-rule
                name                            storeXcontact
                parameter-name
                type                            header-value
                action                          store
                match-val-type                  any
                comparison-type                 case-sensitive
                match-value                     (.*)
                new-value
header-rule
        name                            replacecontact
        header-name                     Contact
        action                          manipulate
        comparison-type                 pattern-rule
        msg-type                        request
        methods                         INVITE,UPDATE
        match-value
        new-value
        element-rule
                name                            replacecontact
                parameter-name
                type                            uri-user
                action                          replace
                match-val-type                  any
                comparison-type                 pattern-rule
                match-value                     (.*)
                new-value                       $storeXcontact.$storeXcontact.$0
header-rule
        name                            delXcontact
        header-name                     X-Contact
        action                          delete
        comparison-type                 pattern-rule
        msg-type                        request
        methods                         INVITE,UPDATE
        match-value
        new-value
header-rule
        name                            strDivNum
        header-name                     Diversion
        action                          manipulate
        comparison-type                 case-sensitive
        msg-type                        request
        methods                         INVITE
        match-value
        new-value
        element-rule
                name                            strval
                parameter-name
                type                            uri-user
                action                          store
                match-val-type                  any
                comparison-type                 case-sensitive
                match-value                     (.*)
                new-value
        element-rule
                name                            strDisName
```

```
              parameter-name
              type                          uri-display
              action                        store
              match-val-type                any
              comparison-type               case-sensitive
              match-value                   (.*)
              new-value
header-rule
       name                           manipRefer
       header-name                    Refer-To
       action                         manipulate
       comparison-type                case-sensitive
       msg-type                       request
       methods
       match-value
       new-value
       element-rule
              name                          chgHostRefer
              parameter-name
              type                          uri-host
              action                        replace
              match-val-type                any
              comparison-type               case-sensitive
              match-value
              new-value                     $REMOTE_IP
header-rule
       name                           delPloc
       header-name                    P-Location
       action                         delete
       comparison-type                case-sensitive
       msg-type                       any
       methods
       match-value
       new-value
header-rule
       name                           delAlert
       header-name                    Alert-Info
       action                         delete
       comparison-type                case-sensitive
       msg-type                       any
       methods
       match-value
       new-value
header-rule
       name                           chgPAIuser
       header-name                    P-Asserted-Identity
       action                         manipulate
       comparison-type                boolean
       msg-type                       request
       methods                        INVITE
       match-value                    $strDivNum.$strval.$0
       new-value
       element-rule
              name                          chgUserFromDiv
              parameter-name
              type                          uri-user
              action                        replace
              match-val-type                any
              comparison-type               pattern-rule
              match-value                   (.*)
              new-value                     $strDivNum.$strval.$0
       element-rule
```

```
                name                            chgDispName
                parameter-name
                type                            uri-display
                action                          replace
                match-val-type                  any
                comparison-type                 pattern-rule
                match-value                     (.*)
                new-value                       $strDivNum.$strDisName.$0
        header-rule
                name                            delDiv
                header-name                     Diversion
                action                          delete
                comparison-type                 case-sensitive
                msg-type                        any
                methods
                match-value
                new-value
        header-rule
                name                            OptZeroMaxFwd
                header-name                     Max-Forwards
                action                          manipulate
                comparison-type                 case-sensitive
                msg-type                        request
                methods                         OPTIONS
                match-value
                new-value
                element-rule
                        name                            zeroMaxFwd
                        parameter-name
                        type                            header-value
                        action                          replace
                        match-val-type                  any
                        comparison-type                 case-sensitive
                        match-value
                        new-value                       0
        last-modified-by                admin@192.168.168.37
        last-modified-date              2012-02-06 14:06:04
sip-manipulation
        name                            inManFromSM
        description                     storecontact
        split-headers
        join-headers
        header-rule
                name                            strcon
                header-name                     Contact
                action                          manipulate
                comparison-type                 case-sensitive
                msg-type                        request
                methods                         INVITE,UPDATE
                match-value
                new-value
                element-rule
                        name                            strval
                        parameter-name
                        type                            uri-user
                        action                          store
                        match-val-type                  any
                        comparison-type                 case-sensitive
                        match-value                     (.*)
                        new-value
        header-rule
                name                            addXcontact
```

```
            header-name                     X-Contact
            action                          add
            comparison-type                 pattern-rule
            msg-type                        request
            methods                         INVITE,UPDATE
            match-value
            new-value
            element-rule
                  name                              addX
                  parameter-name
                  type                              header-value
                  action                            replace
                  match-val-type                    any
                  comparison-type                   pattern-rule
                  match-value
                  new-value                         $strcon.$strval.$0
steering-pool
      ip-address                    10.3.3.225
      start-port                    49152
      end-port                      65535
      realm-id                      EXTERNAL
      network-interface
      last-modified-by              admin@192.168.168.37
      last-modified-date            2011-09-10 10:11:31
steering-pool
      ip-address                    10.32.128.13
      start-port                    2048
      end-port                      65535
      realm-id                      INTERNAL2
      network-interface
      last-modified-by              admin@135.11.141.118
      last-modified-date            2010-10-06 11:28:26
system-config
      hostname
      description
      location
      mib-system-contact
      mib-system-name
      mib-system-location
      snmp-enabled                  enabled
      enable-snmp-auth-traps        disabled
      enable-snmp-syslog-notify     disabled
      enable-snmp-monitor-traps     disabled
      enable-env-monitor-traps      disabled
      snmp-syslog-his-table-length  1
      snmp-syslog-level             WARNING
      system-log-level              WARNING
      process-log-level             NOTICE
      process-log-ip-address        0.0.0.0
      process-log-port              0
      collect
            sample-interval               5
            push-interval                 15
            boot-state                    disabled
            start-time                    now
            end-time                      never
            red-collect-state             disabled
            red-max-trans                 1000
            red-sync-start-time           5000
            red-sync-comp-time            1000
            push-success-trap-state       disabled
      call-trace                    enabled
```

```
internal-trace                enabled
log-filter                    all
default-gateway               10.3.3.254
restart                       enabled
exceptions
telnet-timeout                0
console-timeout               0
remote-control                enabled
cli-audit-trail               enabled
link-redundancy-state         disabled
source-routing                disabled
cli-more                      disabled
terminal-height               24
debug-timeout                 0
trap-event-lifetime           0
default-v6-gateway            ::
ipv6-support                  disabled
cleanup-time-of-day           00:00
last-modified-by              admin@192.168.168.37
last-modified-date            2011-09-10 11:04:14
```

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.