Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the PAETEC Dynamic IP SIP Trunk Service (BroadSoft Platform) with Avaya Aura® Solution for Midsize Enterprise 6.1 – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between the PAETEC Dynamic IP SIP Trunk Service and an Avaya SIP-enabled enterprise solution. PAETEC can offer the Dynamic IP SIP Trunk Service using several different platform technologies in the PAETEC network. These Application Notes correspond to the Dynamic IP SIP Trunk Service offered using a Broadsoft platform in the network. The Avaya solution consists of Avaya Aura® Solution for Midsize Enterprise, and various Avaya endpoints.

PAETEC 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 the PAETEC Dynamic IP SIP Trunk Service and an Avaya SIP-enabled enterprise solution. PAETEC can offer the Dynamic IP SIP Trunk Service using several different platform technologies in the PAETEC network. These Application Notes correspond to the Dynamic IP SIP Trunk Service offered using a Broadsoft platform in the network. The Avaya solution consists of Avaya Aura® Solution for Midsize Enterprise using the Avaya Aura® Session Boarder Controller, Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server along with various Avaya endpoints.

Customers using this Avaya SIP-enabled enterprise solution with the PAETEC Dynamic IP SIP Trunk Service 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 configure a simulated enterprise site using Midsize Enterprise's Communication Manager, Session Manager and the Session Border Controller to connect to the public Internet using a broadband connection. The enterprise site was configured to connect to the Dynamic IP SIP Trunk Service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

The Dynamic IP SIP Trunk Service passed compliance testing.

2.1. Interoperability Compliance Testing

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

- Incoming PSTN calls to various phone types. Phone types included H.323, SIP, 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 phone types. Phone types included H.323, SIP, 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 supports two modes (Road Warrior and Telecommuter). Each supported mode was tested. Avaya one-X Communicator also supports two Voice Over IP (VoIP) protocols: H.323 and SIP. Each supported protocol was tested.
- Various call types including: local, long distance, international, outbound toll-free, operator assisted calls and local directory assistance (411).
- Codecs G.729A, G.711MU and G.711A.

- DTMF transmission using RFC 2833.
- Caller ID presentation and Caller ID restriction.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Network Call Redirection using the SIP REFER method or a 302 response.
- Off-net call forwarding and mobility (extension to cellular).

Items not supported or not tested included the following:

- Inbound toll-free and emergency calls (911) are supported but were not tested as part of the compliance test.
- T.38 Fax not supported.

2.2. Test Results

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

- **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.
- **Network Call Redirection**: When PAETEC's Enterprise Trunking feature is active and Communication Manager is programmed to redirect an inbound call to a PSTN number before answering the call in a vector, PAETEC will send an ACK to the "302 Moved Temporarily" SIP message from the enterprise but will not redirect the call to the new party in the Contact header of the 302 message. The inbound call initiator hears a recording from PAETEC in this failure scenario. A workaround is to use the REFER method to redirect the call by having Communication Manager answer the call first with an announcement in the vector. When PAETEC's Enterprise Trunking feature is NOT active, Network Call Redirection works as expected.

2.3. Support

For technical support on the Dynamic IP SIP Trunk Service, contact PAETEC using the Customer Care links at www.paetec.com.

3. Reference Configuration

Figure 1 illustrates a sample Avaya SIP-enabled enterprise solution connected to the Dynamic IP SIP Trunk Service. This is the configuration used for compliance testing.

Avaya Aura® Solution for Midsize Enterprise packages several Avaya Aura® applications onto one server using System Platform technology. The following applications are included in the Midsize Enterprise template:

- Communication Manager
- Application Enablement Services
- Communication Manager Messaging

- Presence Services
- Avaya Aura® Session Border Controller
- Session Manager
- System Manager
- Utility Services

Along with the Midsize Enterprise the other Avaya components used to create the simulated customer site included:

- Avaya 9600-Series IP telephones (H.323 and SIP)
- Avaya 4600-Series IP telephones (H.323)
- Avaya 1600-Series IP telephones (H.323)
- Avaya one-X Communicator (H.323 and SIP)
- Avaya digital and analog telephones

Note: Application Enablement Services and Presence Services were installed as part of the Midsize Enterprise solution but were not used during compliance testing. Configuration of these services is not covered in these Application Notes.

Located at the edge of the enterprise is the Session Border Controller (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 SBC. In this way, the SBC can protect the enterprise against any SIP-based attacks. The 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 cannot be routed by the PSTN.

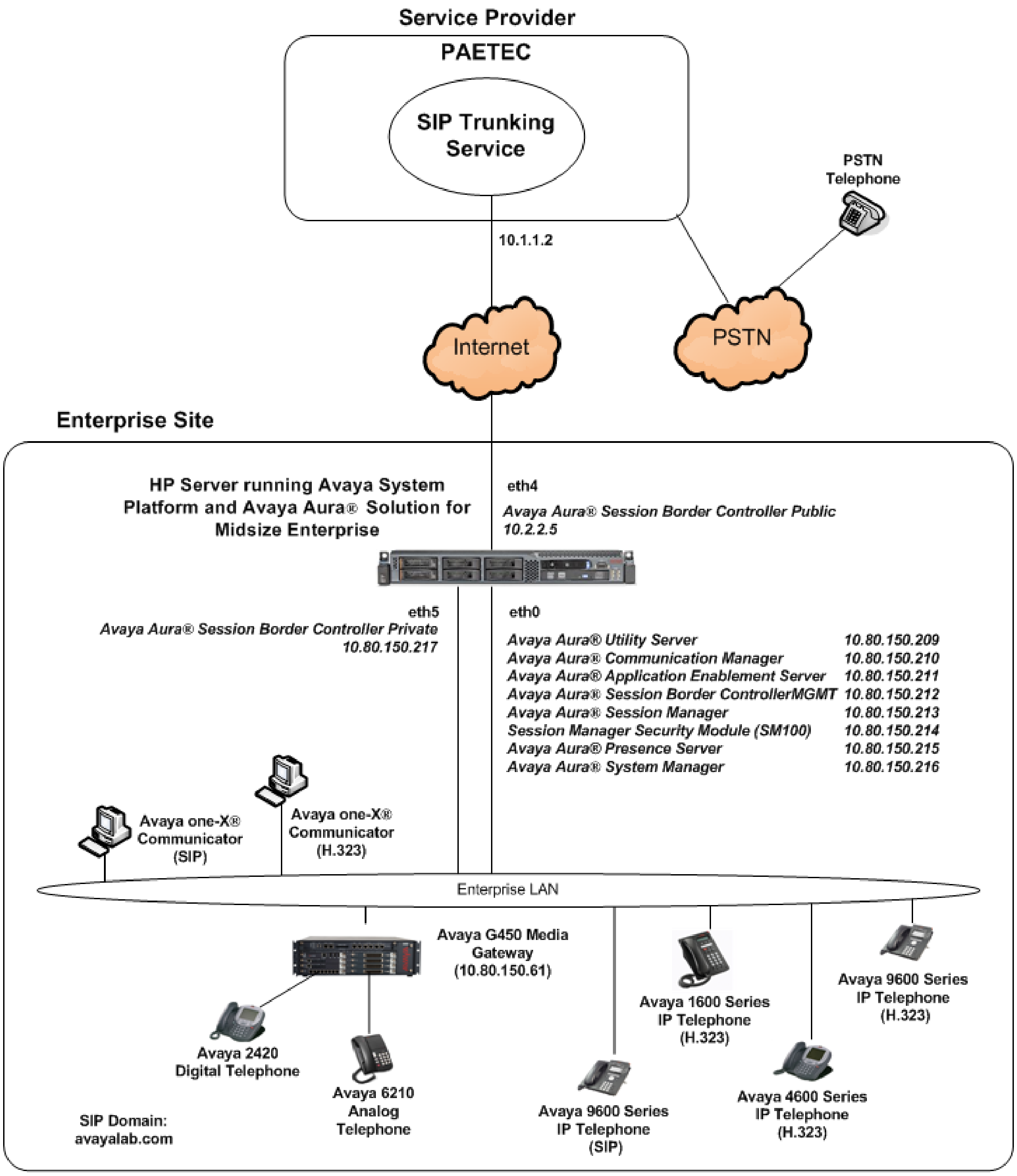


Figure 1: Avaya IP Telephony Network using the Dynamic IP SIP Trunk Service

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

For inbound calls, the calls flow from the service provider to the 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 SBC. From the SBC, the call is sent to the Dynamic IP SIP Trunk Service.

PAETEC allows all North American Numbering Plan (NANP) numbers to be dialed with either 10 digits or 11 digits (1 + 10).

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® Solution for Midsize Enterprise.	6.1.0.0.2580
Avaya Aura® Communication Manger	R016x.00.1.510.1
Avaya Aura® Communication Manger Messaging	vcm-016-00.1.510.1
Avaya Aura® System Manager	6.1.0.0.7345-6.1.5.112
Avaya Aura® Session Manager	6.1.1.0.611023
Avaya 1608 IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.2.2
Avaya Aura® Session Border Controller	E362P4
Avaya G450	31.18.1
Avaya 4625SW IP Telephone (H.323)	2.9010
Avaya 9641 IP Telephone (H.323)	Avaya one-X Deskphone Edition 6.0
Avaya 9621 IP Telephone (SIP)	Avaya one-X Deskphone SIP Edition 6.0
Avaya one-X Communicator (H.323 and SIP)	6.1.0.12
Avaya 2420 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
PAETEC SIP Trunking Solution Components	
Component	Release
BroadSoft Platform	14sp9

Table 1: Equipment and Software Tested

The specific configuration above was used for the compatibility 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 Communication Manager

This section describes the procedure for configuring Communication Manager for the Dynamic IP SIP Trunk Service. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from PAETEC. It is assumed the general installation of Communication Manager, Avaya G450 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 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 **12000** licenses are available and **259** are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 12000 259
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

5.2. 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? y
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y
```

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

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

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

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

INTERNATIONAL CALL ROUTING PARAMETERS
                 Local Country Code: 1
          International Access Code: 011

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
```

5.3. 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 Communication Manager **(procr)** and for Session Manager (***SM***). These node names will be needed for defining the service provider signaling group in **Section 5.6**.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
Name              IP Address
SM                  10.80.150.214
default             0.0.0.0
mes-aes             10.80.150.211
procr               10.80.150.210
procr6              ::
```

5.4. 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, ip-codec-set 2 was used for this purpose. The Dynamic IP SIP Trunk Service supports G.729A and G.711MU. Thus, these codecs were included in this set, in order of preference. The order of preference is defined by the end customer. Enter **G.729A** and **G.711MU** in the **Audio Codec** column of the table. Default values can be used for all other fields.

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

                          IP Codec Set

    Codec Set: 2

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

Since T.38 fax is not supported, set the **Fax Mode** to **off**.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

                    Mode                 Redundancy
    FAX             off                   0
    Modem           off                   0
    TDD/TTY         US                    3
```

5.5. 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 2 was chosen for the service provider trunk. IP network region 1 is the default IP network region and encompasses the rest of the enterprise. Use the **change ip-network-region 2** command to configure region 2 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is **avayalab.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 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location: 1       Authoritative Domain: avayalab.com
    Name: PAETEC SIP TRUNK
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

On **Page 4**, define the IP codec set to be used for traffic between region 2 and region 1 (the rest of the enterprise). 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 2 will be used for calls between region 2 (the service provider region) and region 1 (the rest of the enterprise).

```
change ip-network-region 2                                   Page   4 of  20

 Source Region: 2     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    2     y    NoLimit                                             n       t
 2    2
 3
 4
```

5.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and the 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 1 was used for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- Set the **IMS Enabled** field to **n**. This specifies the Communication Manager will serve as an Evolution Server for the Session Manager.
- Set the **Transport Method** to the recommended default value of **tls** (Transport Layer Security). Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061). This is necessary so the SM can distinguish this trunk from the trunk used for other enterprise SIP traffic. For ease of troubleshooting, the compliance test was conducted with the **Transport Method** set to **tcp** and the **Near-end Listen Port** and **Far-end Listen Port** set to **5070**. (For TCP, the well-known port value is 5060).
- Set the **Peer Detection Enabled** field to **n**.
- Set the **Peer Server** to **Others**. When the Peer Server is detected or set to SM, Communication Manager precedes a + sign to the From, Contact and P-Asserted Identity headers. The addition of the + sign impacted interoperability with PAETEC.
- Set the **Near-end Node Name** to **procr**. This node name maps to the IP address of the Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to **SM**. This node name maps to the IP address of Session Manager as defined in **Section 5.3**.
- Set the **Far-end Network Region** to the IP network region defined for the service provider in **Section 5.5**.
- 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.
- Set the **DTMF over IP** field to **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Default values may be used for all other fields.

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                               SIP Enabled LSP? n
     IP Video? n                                     Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? n  Peer Server: Others


   Near-end Node Name: procr                  Far-end Node Name: SM
 Near-end Listen Port: 5070                 Far-end Listen Port: 5060
                                         Far-end Network Region: 2

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

5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.6**. For the compliance test, trunk group 1 was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Service Type** field to **public-ntwrk**.
- Set **Member Assignment Method** to **auto**.
- Set the **Signaling Group** to the signaling group shown in the previous step.
- 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 1                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: SIP trunk to PAETEC          COR: 1        TN: 1        TAC: *01
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 4
```

On **Page 2**, 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 1                                                 Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                             Redirect On OPTIM Failure: 5000

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

 Disconnect Supervision - In? y  Out? y
```

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 5.2**, if the inbound call enabled CPN block. For outbound calls, these same settings request that CPN block be activated on the far-end destination if a local user requests CPN block on a particular call routed out this trunk. Default values were used for all other fields.

```
add trunk-group 1                                                 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



                                Modify Tandem Calling Number: no



 Show ANSWERED BY on Display? y
```

On **Page 4**, set the **Network Call Redirection** field to **y**. This allows inbound calls transferred back to the PSTN to use the SIP REFER method, see [17]. Set the **Send Diversion Header** field to **y**. This field provides additional information to the network if the call has been re-directed. This is only needed when Enterprise Trunking from PAETEC is not being used to support call forwarding of inbound calls back to the PSTN and some Extension to Cellular (EC500) call scenarios. If Enterprise Trunking from PAETEC is used set this value to **n**. Set the **Support Request History** field to **n**.

Set the **Telephone Event Payload Type** to **101**, the value preferred by PAETEC.

```
add trunk-group 1                                                 Page   4 of  21
                              PROTOCOL VARIATIONS

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


       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: P-Asserted-Identity
```

5.8. Inbound Routing

In general, the incoming call handling treatment for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can be used to perform digit conversion using an Adaptation, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. If the DID number sent by PAETEC is unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. Use the **change inc-call-handling-trmt trunk-group** command to create an entry for each DID. As an example, the following screen illustrates a conversion of DID number **7135551234** to extension **12001**.

```
change inc-call-handling-trmt trunk-group 1                      Page   1 of  30
                      INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 7135551234       10  12001
 public-ntwrk    10 7135551235       10  12002
 public-ntwrk    10 7135551236       10  12003
 public-ntwrk    10 7135551237       10  12004
 public-ntwrk
```

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 5.7**), 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, four DID numbers were assigned for testing. These four numbers were assigned to the four extensions **12001**, **12002**, **12003** and **12004**. 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 four extensions.

```
change public-unknown-numbering 1                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                 Total Administered: 13
 5  1                                        5      Maximum Entries: 9999
 5  2                                        5
 5  3                                        5     Note: If an entry applies to
 5  4                                        5     a SIP connection to Avaya
 5  5                                        5     Aura(tm) Session Manager,
 5  6                                        5     the resulting number must
 5  7                                        5     be a complete E.164 number.
 5  8                                        5
 5  12001         1         7135551234       10
 5  12002         1         7135551235       10
 5  12003         1         7135551236       10
 5  12004         1         7135551237       10
```

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. 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
   0           1   attd
   1           5   ext
   2           5   ext
   3           5   ext
   4           5   ext
   5           5   ext
   6           5   ext
   7           5   ext
   8           5   ext
   9           1   fac
   *           3   dac
   #           3   dac
```

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  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *10
         Abbreviated Dialing List2 Access Code: *12
         Abbreviated Dialing List3 Access Code: *13
Abbreviated Dial - Prgm Group List Access Code: *14
                      Announcement Access Code: *19
                       Answer Back Access Code:

      Auto Alternate Routing (AAR) Access Code: *00
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *33    Deactivation: #33
Call Forwarding Activation Busy/DA: *30    All: *31    Deactivation: #30
   Call Forwarding Enhanced Status:        Act:        Deactivation:
```

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 1 which contains the SIP trunk to the service provider (as defined next).

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    1303                  11   11    1        fnpa         n
    1502                  11   11    1        fnpa         n
    1720                  11   11    1        fnpa         n
    1800                  11   11    1        fnpa         n
    1866                  11   11    1        fnpa         n
    1877                  11   11    1        fnpa         n
    1888                  11   11    1        fnpa         n
    1908                  11   11    1        fnpa         n
    2                     10   10    1        hnpa         n
    3                     10   10    1        hnpa         n
    4                     10   10    1        hnpa         n
    411                   3    3     1        svcl         n
    5                     10   10    1        hnpa         n
    555                   7    7     deny     hnpa         n
    6                     10   10    1        hnpa         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 1 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 **1** was used.
- **FRL**: Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **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. All HNPA 10 digit numbers are left unchanged.

```
change route-pattern 1                                            Page   1 of   3
                    Pattern Number: 1   Pattern Name: PAETEC SIP TRK
                             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: 1    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                                          none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

6. Configure Avaya Aura® Session Manager

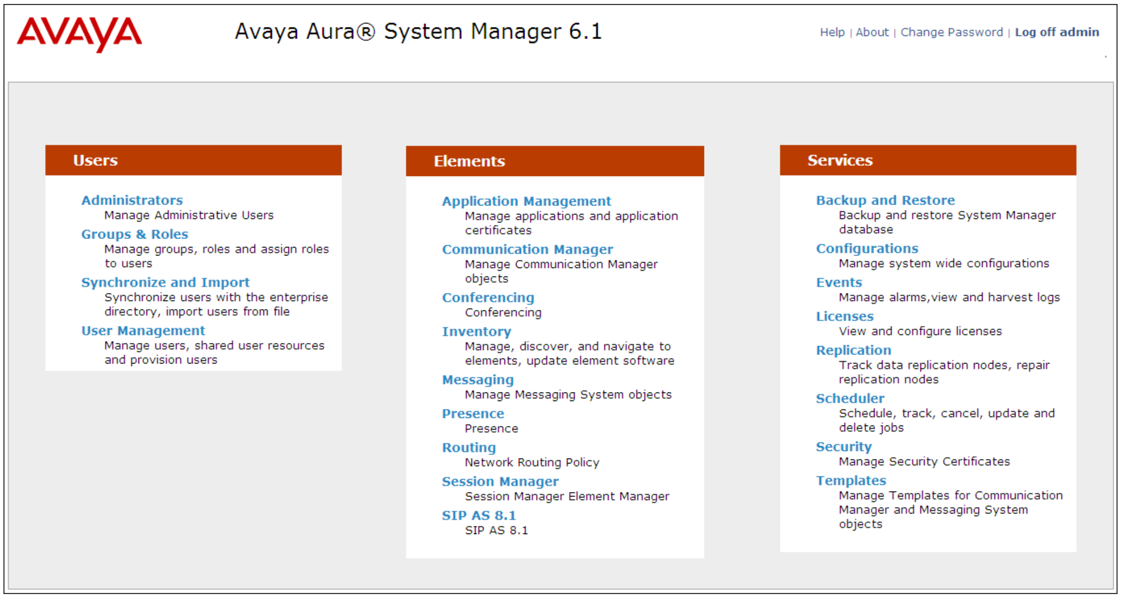
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- 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 govern to which SIP Entity a call is routed
- Session Manager Instance, corresponding to the Session Manager Server to be administered in System Manager.

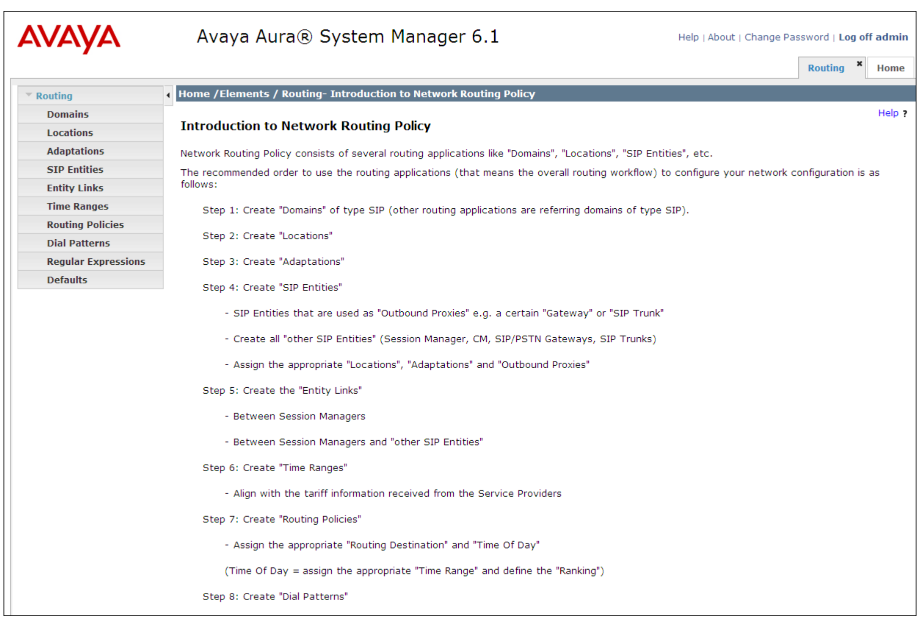
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session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

6.1. 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 **Log On** (not shown). The screen shown below is then displayed.



Most of the configuration items are performed in the Routing Element. Click on **Routing** in the Elements column shown above to bring up the Introduction to Network Routing Policy screen.

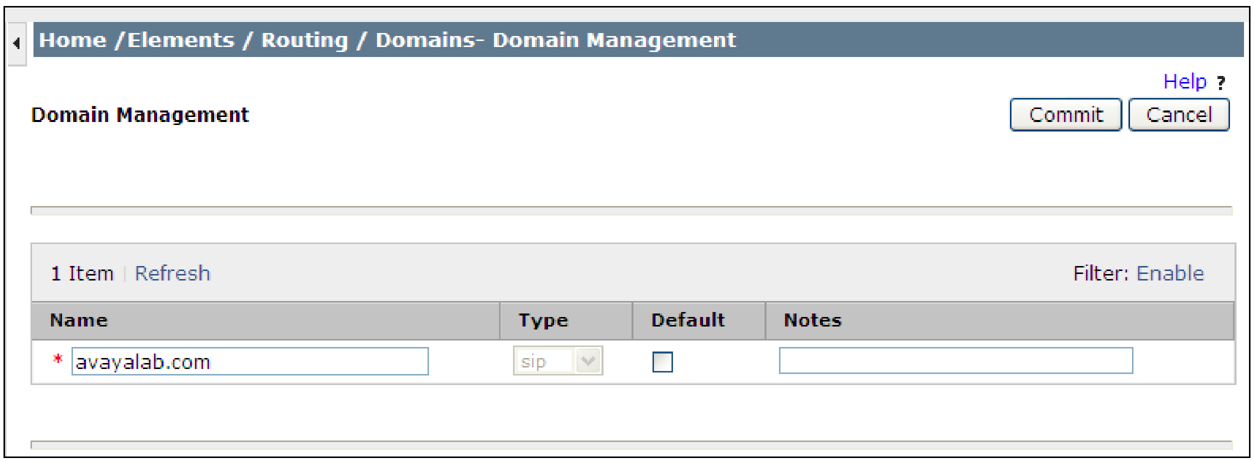


6.2. Specify SIP Domain

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 (**avayalab.com**). Navigate to **Routing → Domains** and click the **New** button in the right pane (not shown). In the new right pane that appears, 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 **avayalab.com** 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. To add a location, navigate to **Routing →Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown).

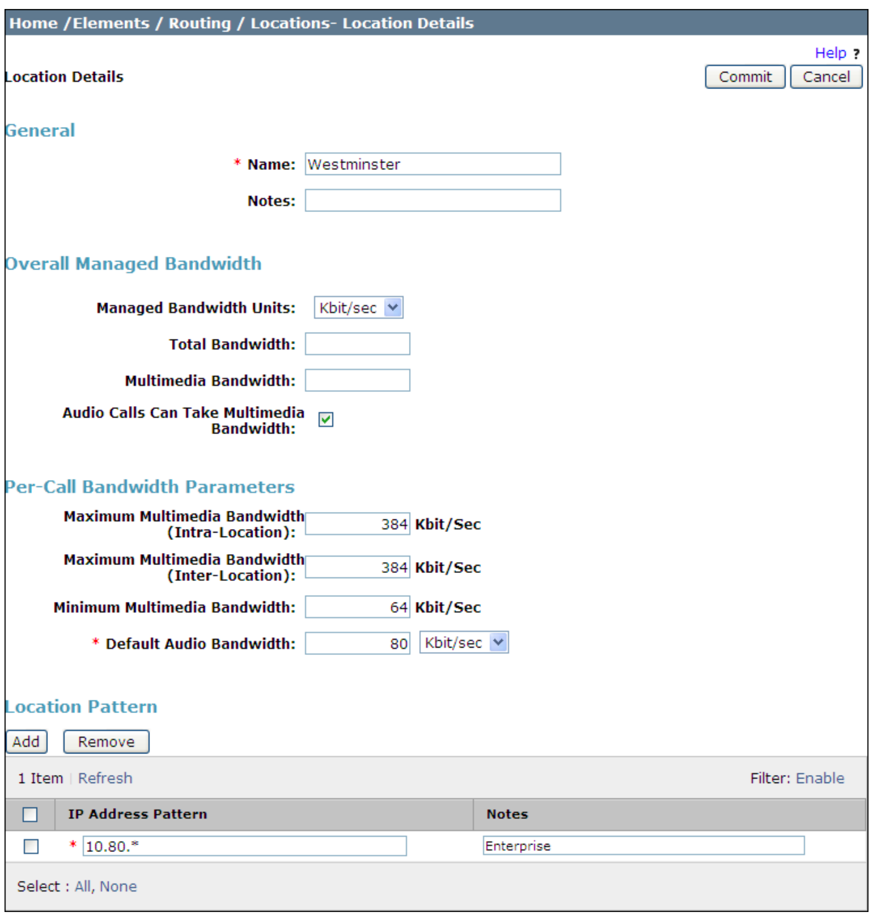
In the **General** section, enter the following values. Use default values for all remaining fields:

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

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

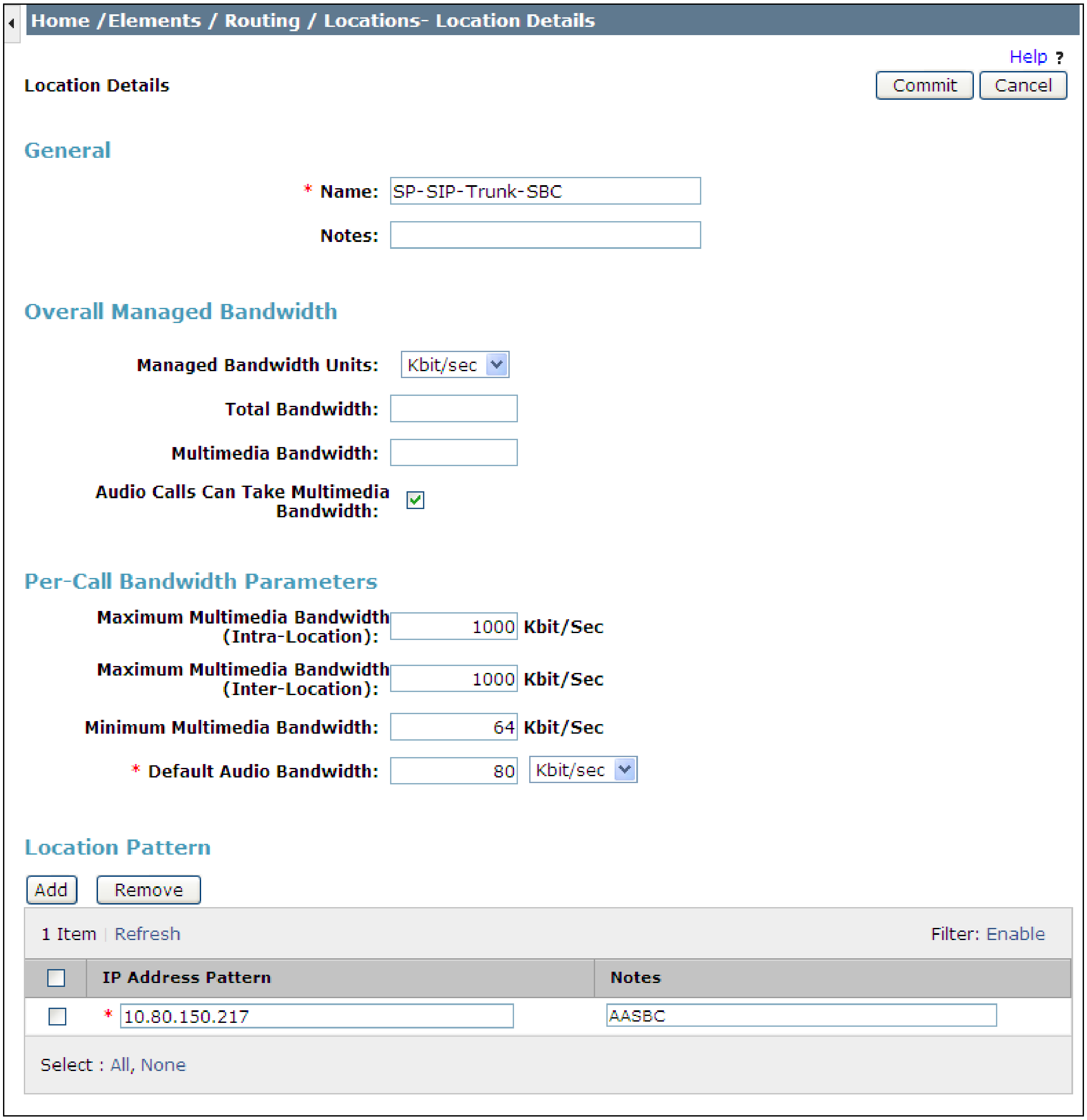
- **IP Address Pattern:** An IP address pattern used to identify the location.
- **Notes:** Add a brief description (optional).

The screen below shows the addition of **Westminster**, which includes all equipment on the **10.80.x.x** subnet including Communication Manager, Session Manager and SIP clients. Click **Commit** to save.



Note: that call bandwidth management parameters should be set per customer requirement.

Repeat the preceding procedure to create a separate Location for the SBC. Displayed below is the screen for addition of the **SP-SIP-TRUNK-SBC** Location, which specifies the specific IP address for the AA-SBC. Click **Commit** to save.



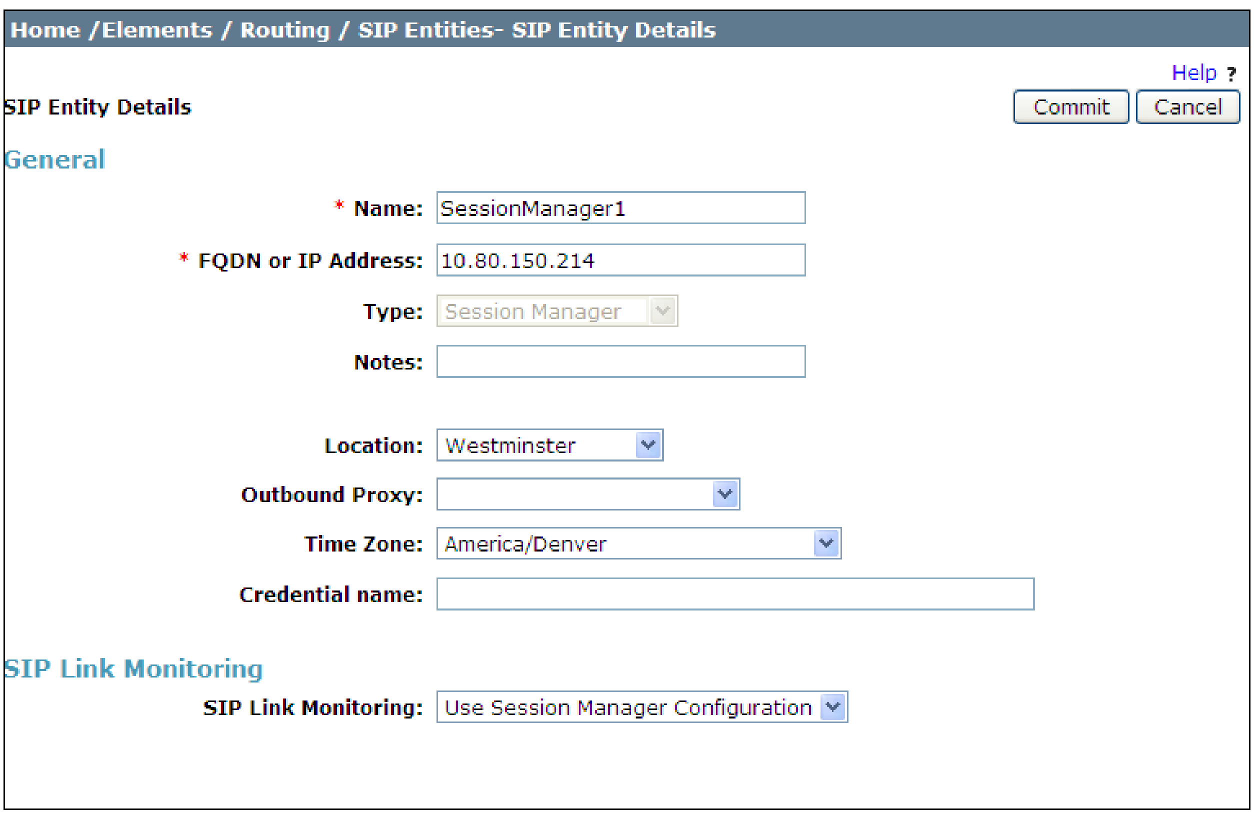
6.4. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it which includes Communication Manager and the SBC. Navigate to **Routing → SIP Entities** in the left-hand navigation pane and click on the **New** button in the right pane (not shown).

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 SBC.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If applicable, select the **Adaptation Name** that will be applied to this entity.
- **Location:** Select one of the locations defined previously.
- **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 Session Manager signaling interface is entered for **FQDN or IP Address**.



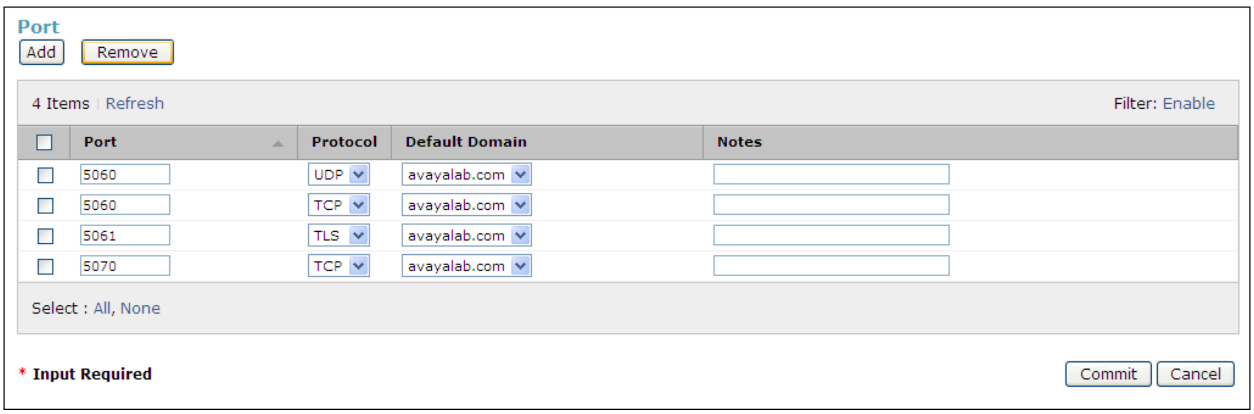
To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP entities. This section defines a default set of ports that the Session Manager will use to listen for SIP requests, typically from registered SIP endpoints. The Session Manager can also listen on additional ports defined elsewhere such as the ports specified in the SIP Entity Link definition in **Section 6.5**.

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

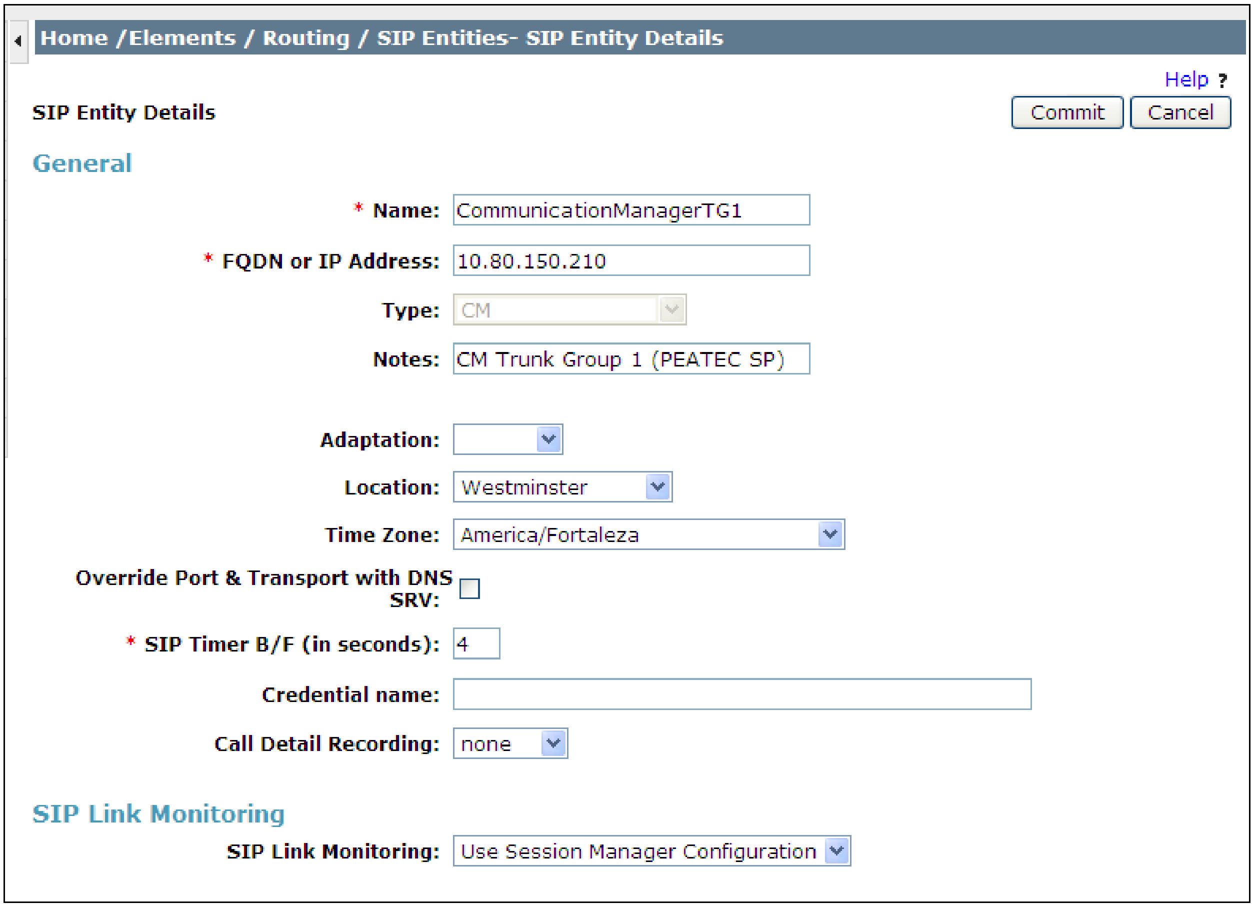
- **Port:** Port number on which the Session Manager can listen for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain:** The domain used for the enterprise.

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

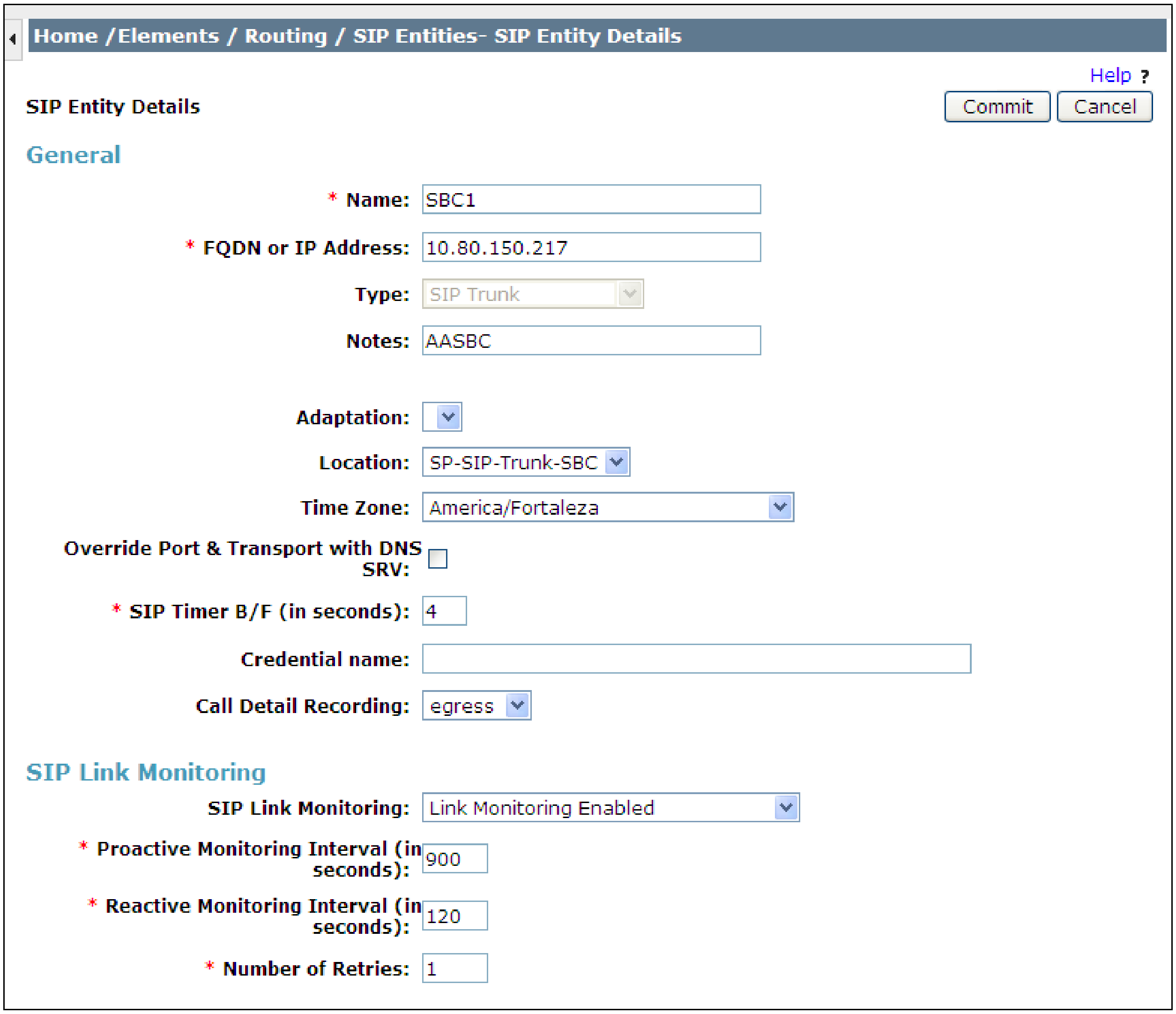
For the compliance test, four **Port** entries were added.



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, a new SIP entity is created separate from 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 Communication Manager.



The following screen shows the addition of the SBC SIP Entity. The **FQDN or IP Address** field is set to the IP address of its private network interface (see **Figure 1**). The Location is set to the one defined for the SBC in **Section 6.3**. **Link Monitoring Enabled** was selected for **SIP Link Monitoring** using the specific time settings for **Proactive Monitoring Interval (in seconds)** and **Reactive Monitoring Interval (in seconds)** for the compliance test. These time settings should be adjusted or left at their default values per customer needs and requirements.



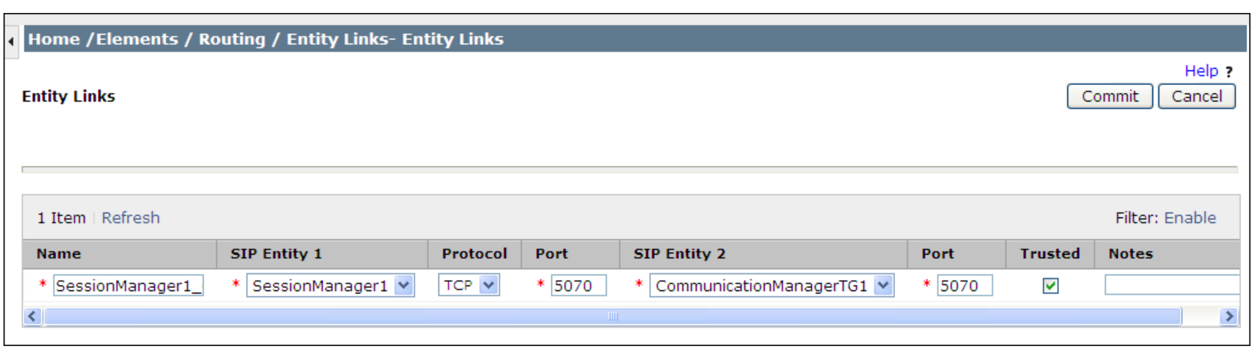
6.5. Add Entity Links

A SIP trunk between Session Manager and a telephony system is described as an Entity Link. Two Entity Links were created; one to the Communication Manager for use only by service provider traffic and one to the SBC. To add an Entity Link, navigate to **Routing → Entity Links** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

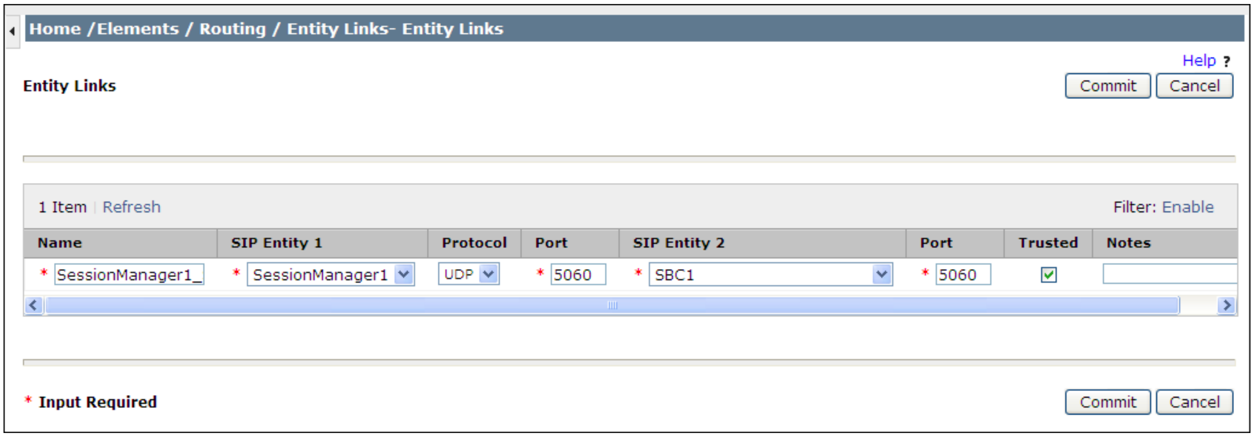
- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the transport protocol used for this link.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For the Communication Manager, this must match the **Far-end Listen Port** defined on the Communication Manager signaling group in **Section 5.6**.
- **SIP Entity 2:** Select the name of the other system. For the Communication Manager, select the Communication Manager SIP Entity defined in **Section 6.4**.
- **Port:** Port number on which the other system receives SIP requests from the Session Manager. For the Communication Manager, this must match the **Near-end Listen Port** defined on the Communication Manager signaling group in **Section 5.6**.
- **Trusted:** Check this box. ***Note**: If this box is not checked, calls from the associated SIP Entity specified in **Section 6.4** will be denied.*

Click **Commit** to save. The following screens illustrate the Entity Links to Communication Manager and the SBC. It should be noted that in a customer environment the Entity Link to Communication Manager would normally use TLS. For the compliance test, TCP was used to aid in troubleshooting since the signaling traffic would not be encrypted. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.6**.

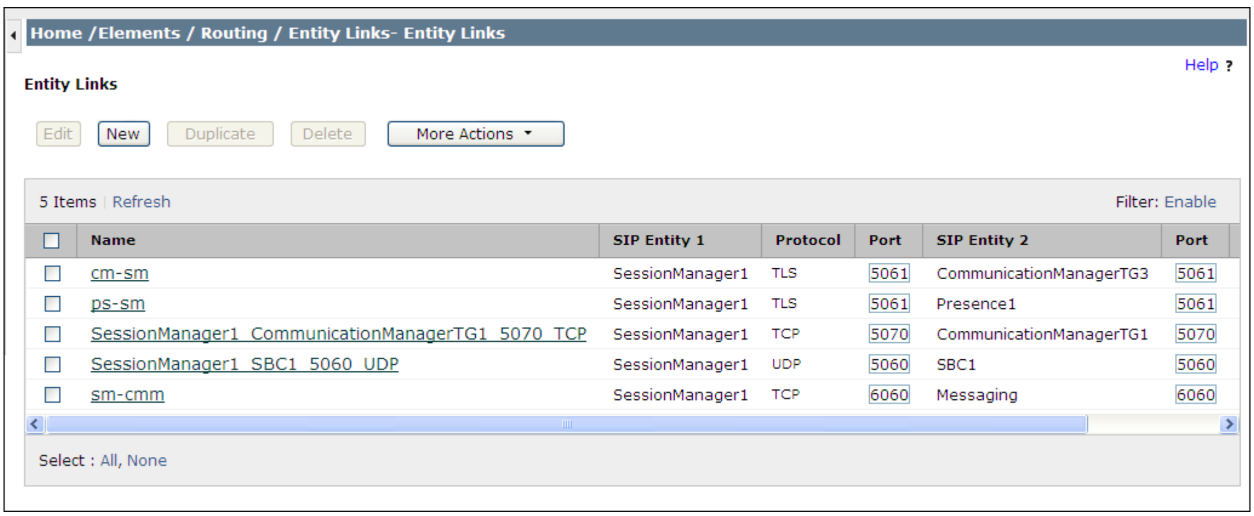
Entity Link to Communication Manager:



Entity Link to the SBC:



The following screen shows the complete list of Entity Links.



6.6. Add Routing Policies

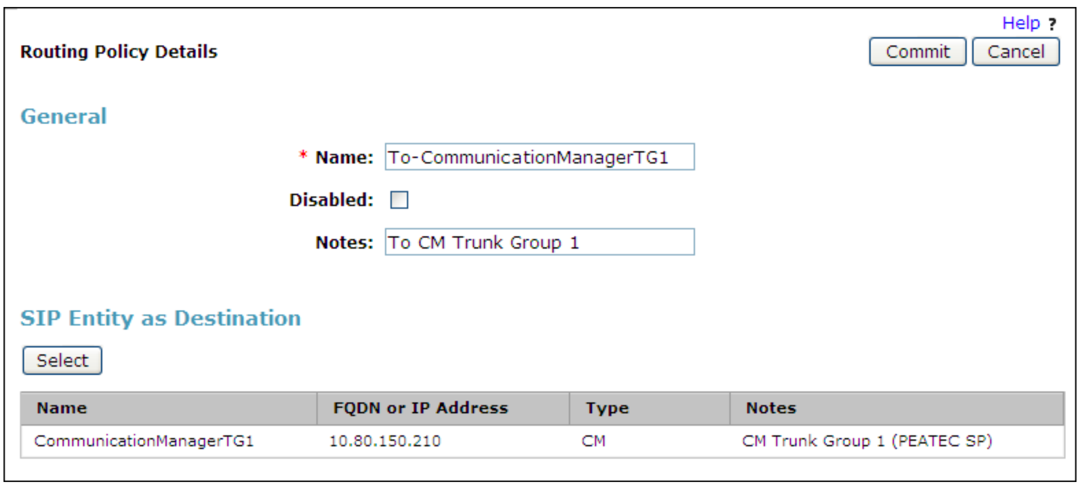
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.4**. Two routing policies must be added: one for Communication Manager and one for the SBC. To add a routing policy, navigate to **Routing → Routing Policies** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). The screen below is displayed. Fill in the following:

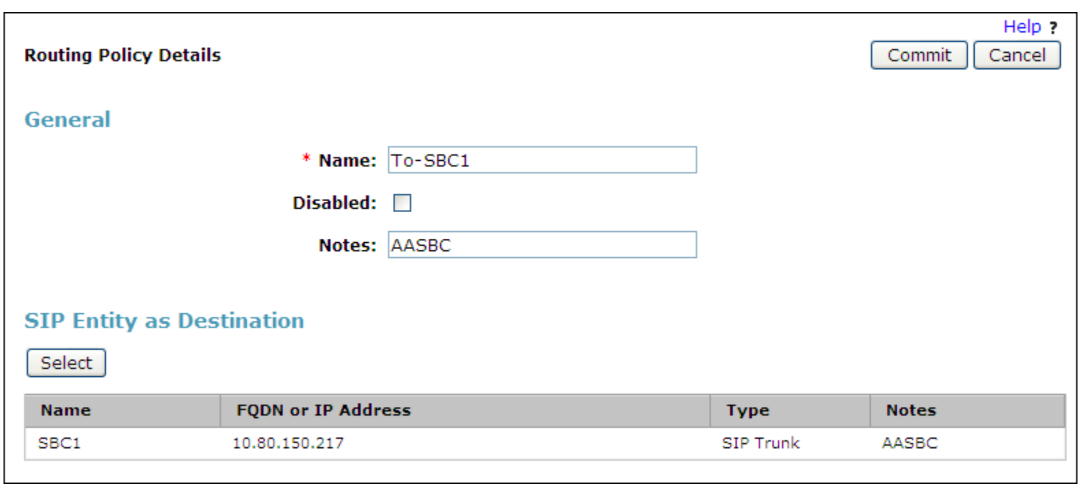
In the **General** section, enter the following values. Use default values for all remaining fields:

- **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** (not shown)**.** 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 SBC.





6.7. 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 PAETEC 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 and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

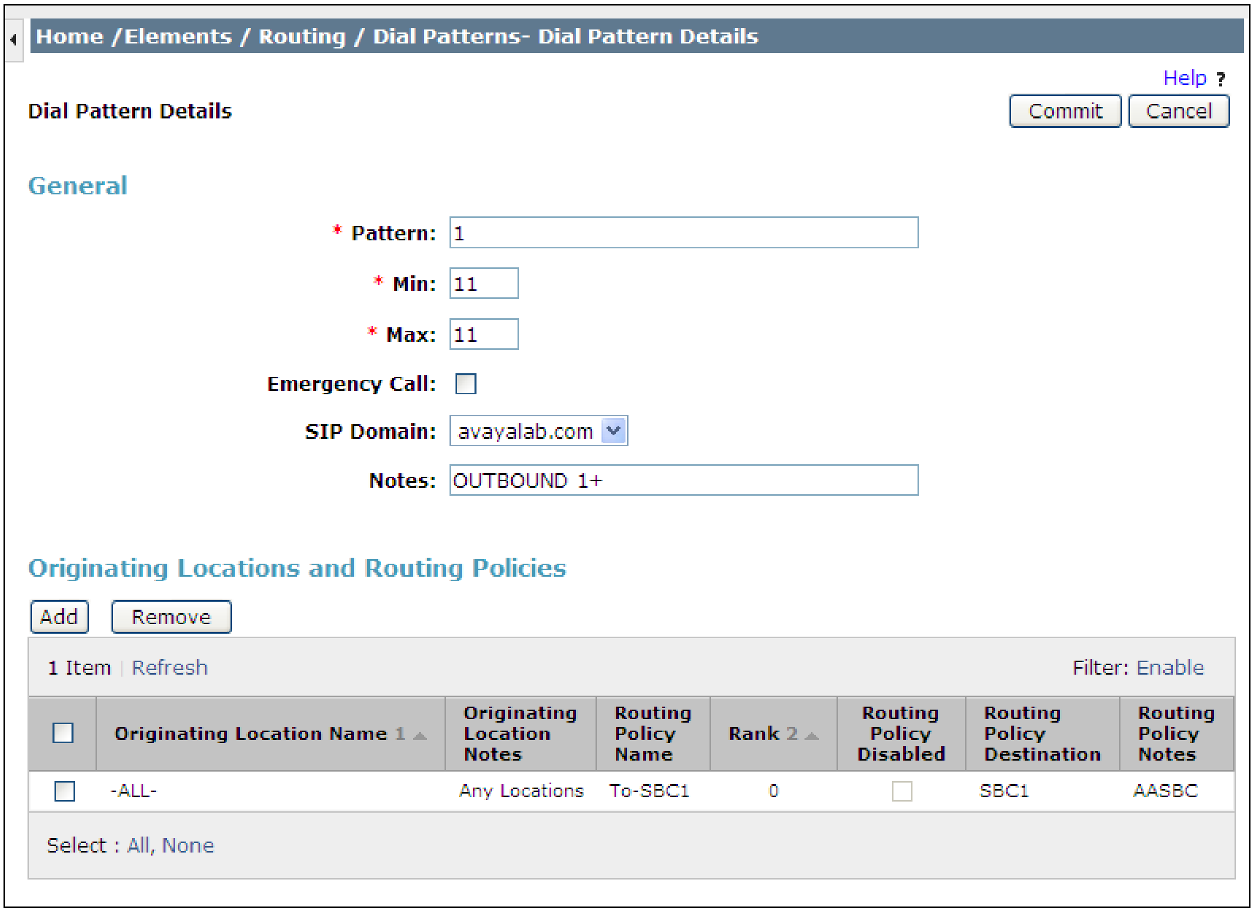
In the **General** section, enter the following values. Use default values for all remaining fields:

- **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).

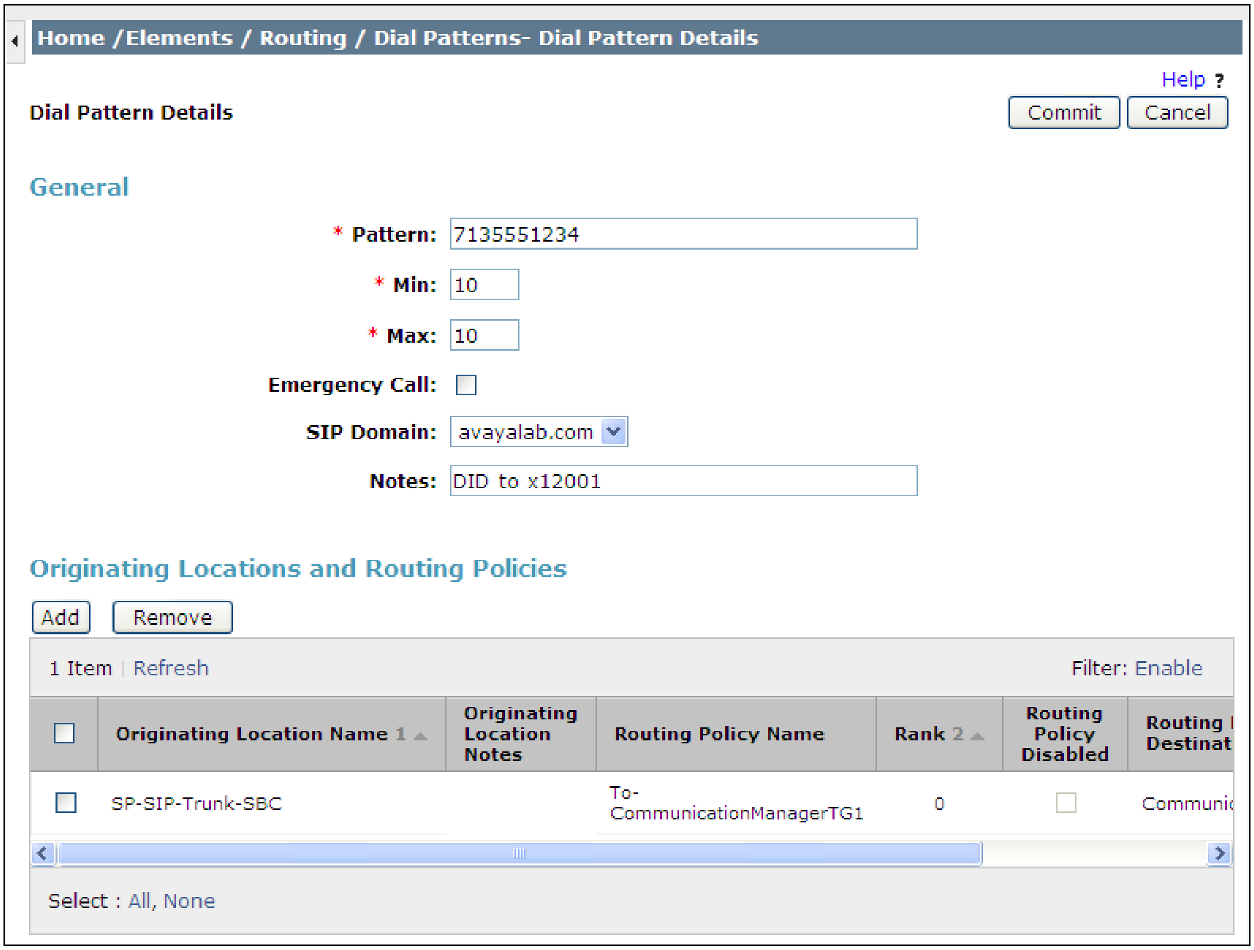
In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used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

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

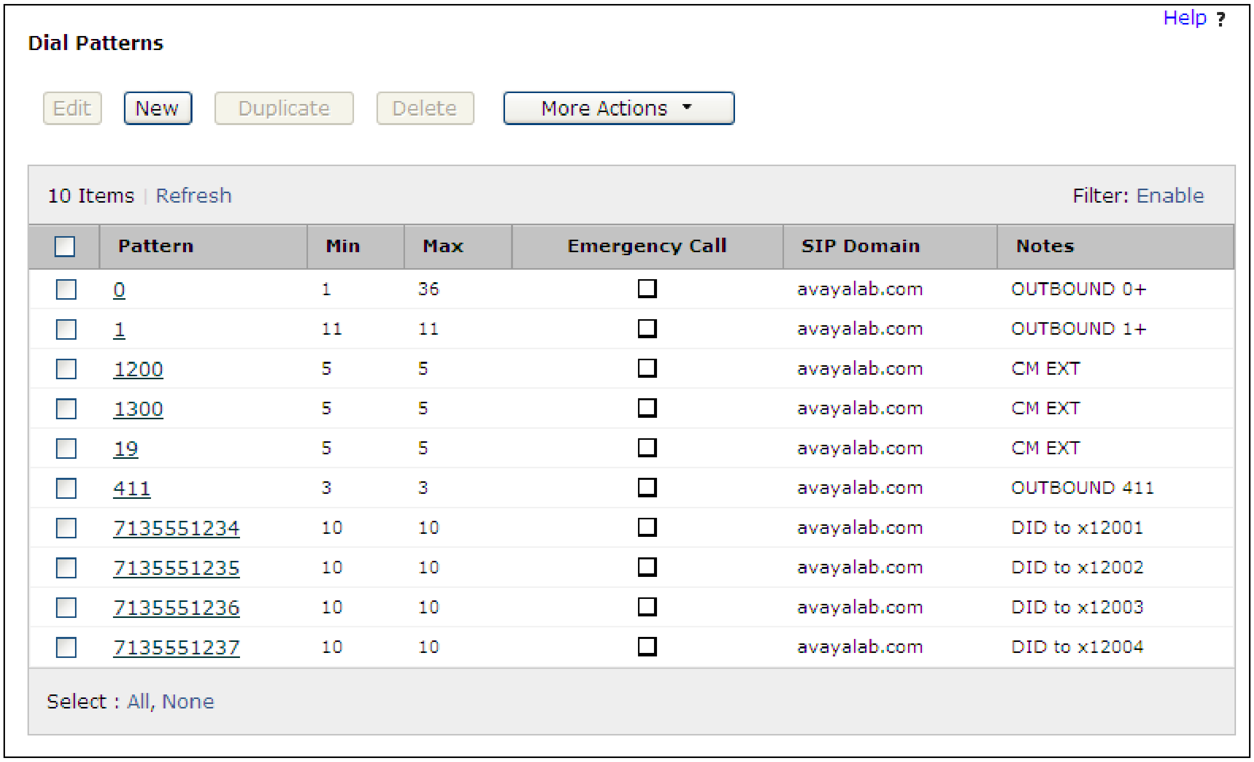
Two examples of the dial patterns used for the compliance test are shown below. The first example shows that **11** digit dialed numbers that begin with **1** uses route policy **To-SBC1**.



The second example shows that a 10 digit number **7135551234** to domain **avayalab.com** and originating from **SP-SIP-TRUNK-SBC** uses route policy **To-CommunicationManagerTG1**. This is a DID number assigned to the enterprise from PAETEC. SP-SIP-TRUNK-SBC is selected because these calls come from the SBC which resides in that location.



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



Help ?

Dial Patterns

Edit | New | Duplicate | Delete | More Actions

10 Items | Refresh   Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Notes
0	1	36	☐	avayalab.com	OUTBOUND 0+
1	11	11	☐	avayalab.com	OUTBOUND 1+
1200	5	5	☐	avayalab.com	CM EXT
1300	5	5	☐	avayalab.com	CM EXT
19	5	5	☐	avayalab.com	CM EXT
411	3	3	☐	avayalab.com	OUTBOUND 411
7135551234	10	10	☐	avayalab.com	DID to x12001
7135551235	10	10	☐	avayalab.com	DID to x12002
7135551236	10	10	☐	avayalab.com	DID to x12003
7135551237	10	10	☐	avayalab.com	DID to x12004

Select : All, None

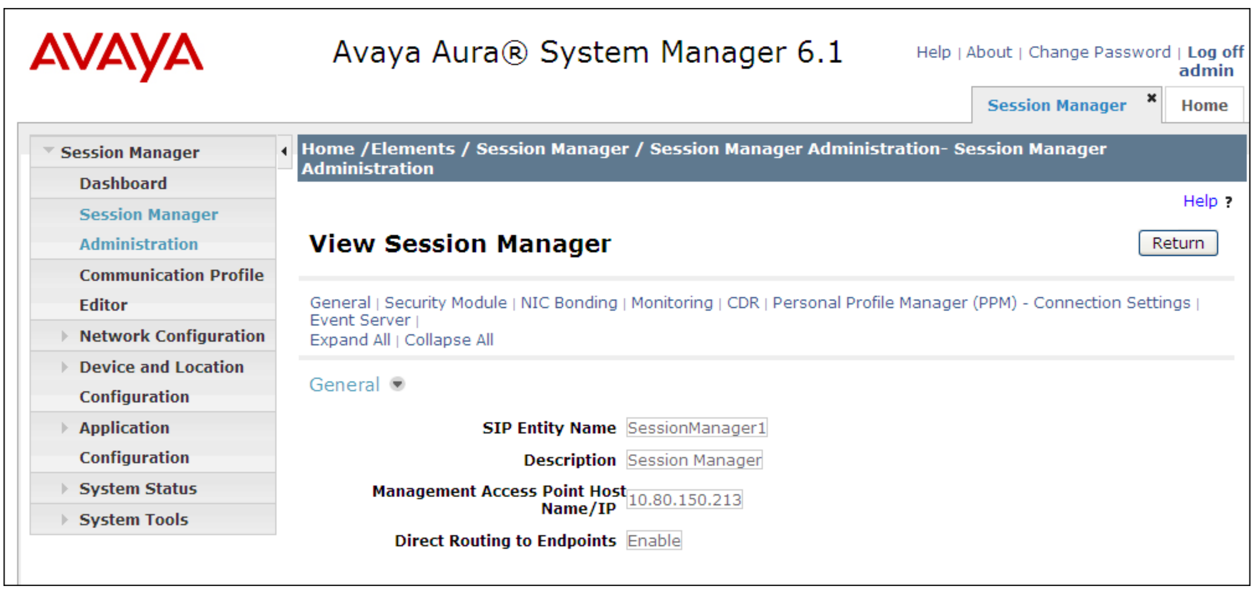
6.8. Add/View Session Manager Instance

The creation of a Session Manager Instance 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, navigate to **Elements → Session Manager → Session Manager Administration** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). If the Session Manager already exists, click **View** (not shown) to view the configuration. Enter/verify the data as described below and shown in the screen below:

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 Session Border Controller

This section describes the configuration of the Session Border Controller (SBC). This configuration is done in two parts. The first part is done during the Midsize Enterprise installation via the installation wizard. These Application Notes will not cover the Midsize Enterprise installation in its entirety but will include the SBC portion of the installation wizard. For information on installing the Avaya System Platform and the loading of the Solution for Midsize Enterprise template see [1] and [3].

The second part of the configuration is done after the installation is complete using the SBC web interface. The resulting SBC configuration file is shown in **Appendix A**.

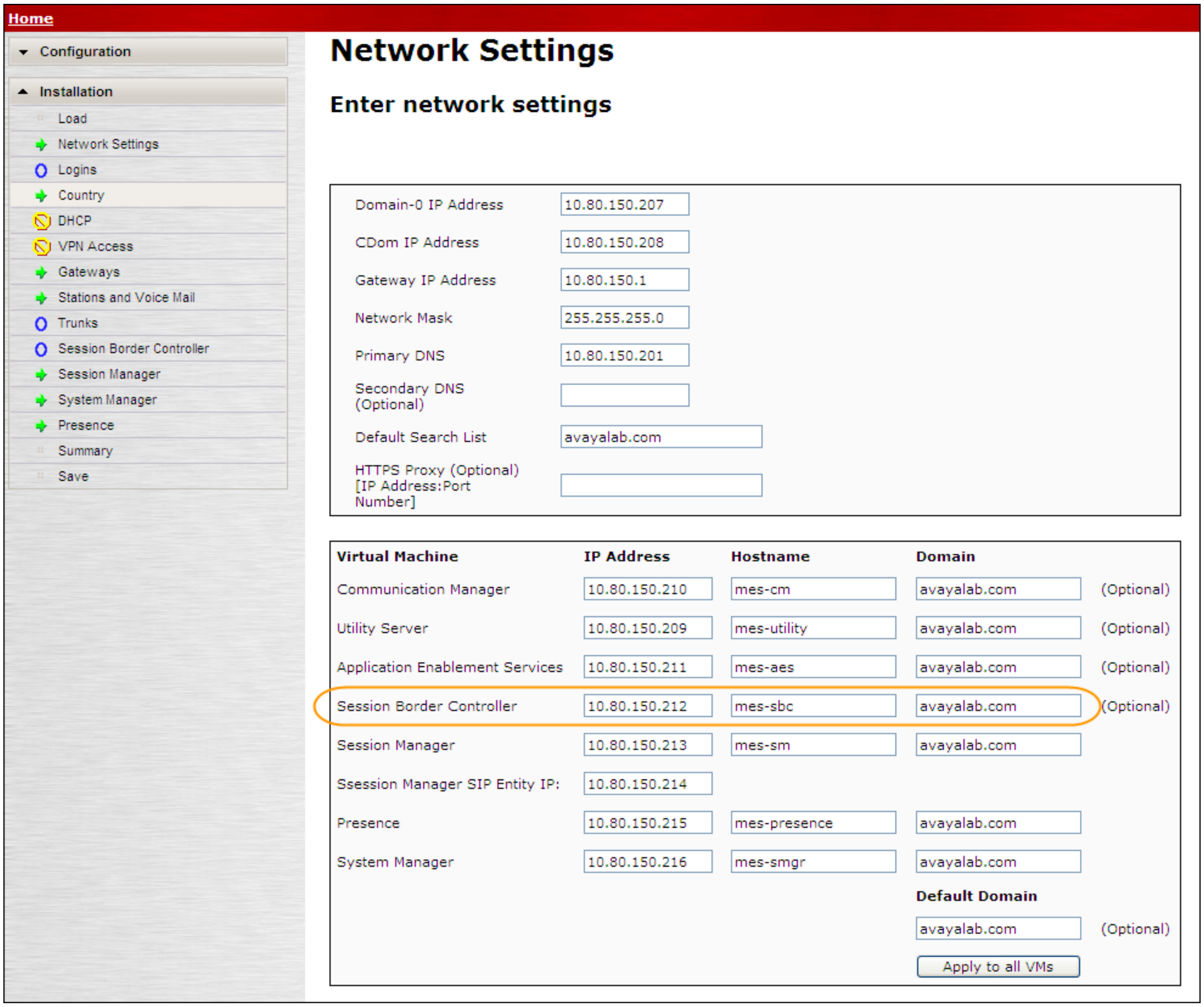
7.1. Installation Wizard

During the installation of the Solution for Midsize Enterprise template, the installation wizard will prompt the installer for information that will be used to create the initial configuration of the SBC.

7.1.1. Network Settings

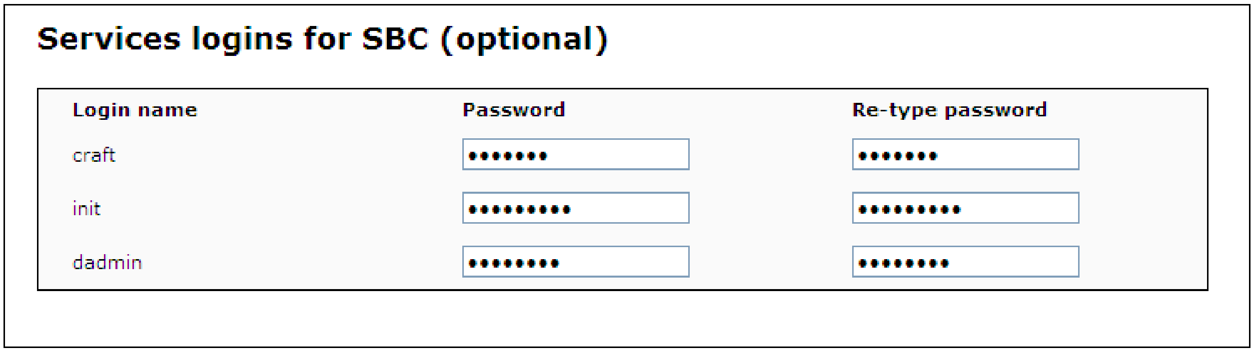
The **Network Settings** screen is where IP Addresses, Hostnames and Domains are assigned to Virtual Machines. Fill in the fields as described below and shown in the following screen:

- **IP Address**: Enter the IP address of the management side of the SBC (eth0).
- **Hostname**: Enter a host name for the SBC
- **Domain:** Enter the Enterprise Domain



7.1.2. Logins

The **Services Logins for SBC (optional)** screen is where passwords for the various applications are set. Assign passwords for the different accounts.



7.1.3. SBC

On the **SBC** screen, fill in the fields as described below and shown in the screen below:

In the Configure section check **Yes** for the question **Do you wish to configure Session Border Controller?**

In the SIP Service Provider Data section:

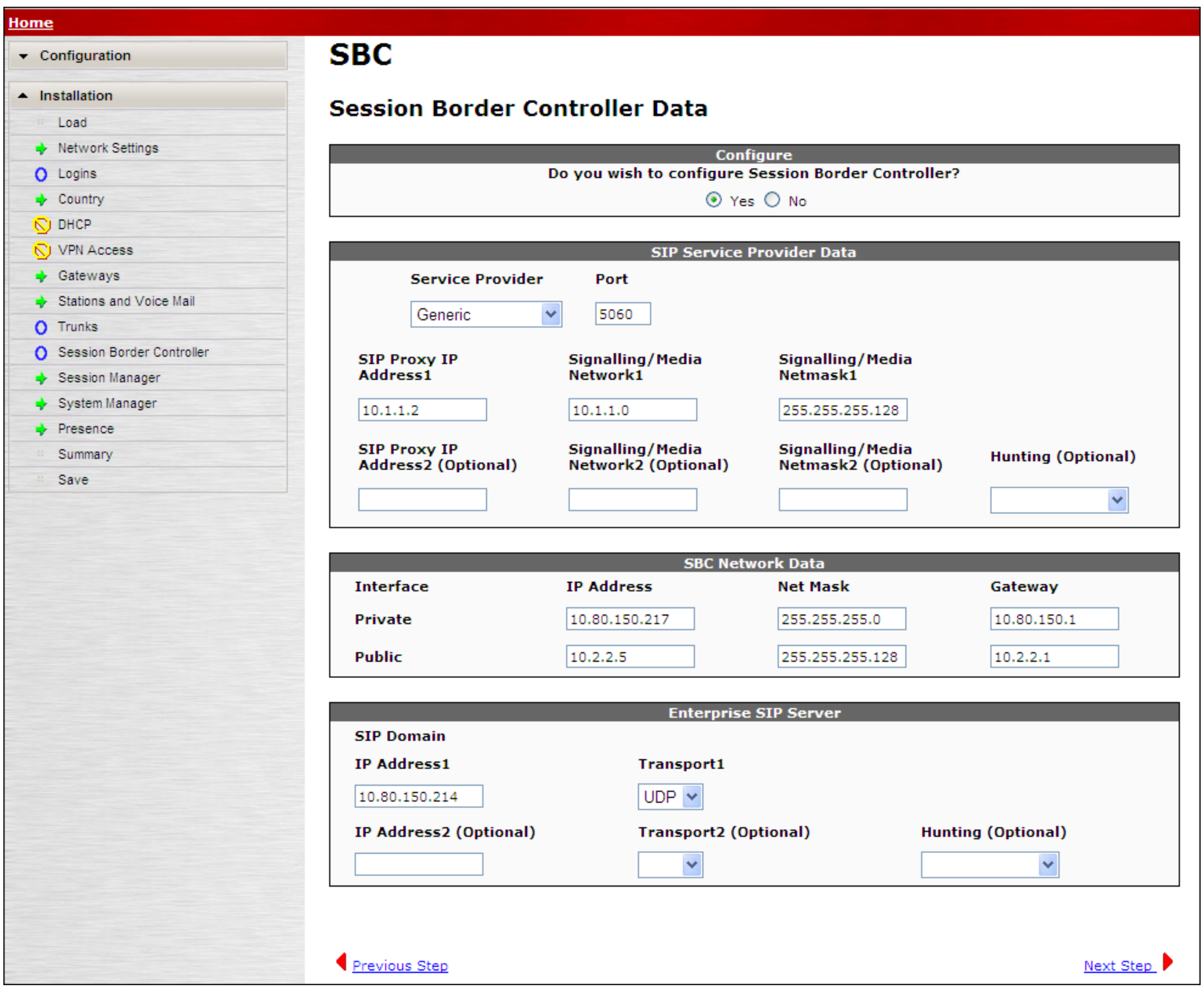
- **Service Provider**: From the pull-down menu, select the name of the service provider to which the SBC will connect. This will allow the wizard to create a configuration file customized for this service provider. At the time of the compliance test, a customized configuration file did not exist for PAETEC. Thus, **Generic** was chosen instead and further customization was done manually after the wizard was complete.
- **IP Address**: Enter the IP address of the SIP proxy of the service provider. If the service provider has multiple proxies, enter the primary proxy on this screen and additional proxies can be added after installation.
- **Port**: Enter the port number that the service provider uses to listen for SIP traffic.
- **Media Network**: Enter the network address of the network where media traffic will originate from the service provider. If media can originate from multiple networks, enter one network address on this screen and additional networks can be added after installation.
- **Media Netmask**: Enter the netmask corresponding to the **Media Network**.

In the **SBC Network Data** section:

- **Private IP Address**: Enter the IP address of the private side of the SBC (eth5).
- **Private Net Mask**: Enter the netmask associated with the private network to which the SBC connects.
- **Private Gateway**: Enter the default gateway of the private network.
- **Public IP Address**: Enter the IP address of the public side of the SBC (eth4).
- **Public Net Mask**: Enter the netmask associated with the public network to which the SBC connects.
- **Public Gateway**: Enter the default gateway of the public network.

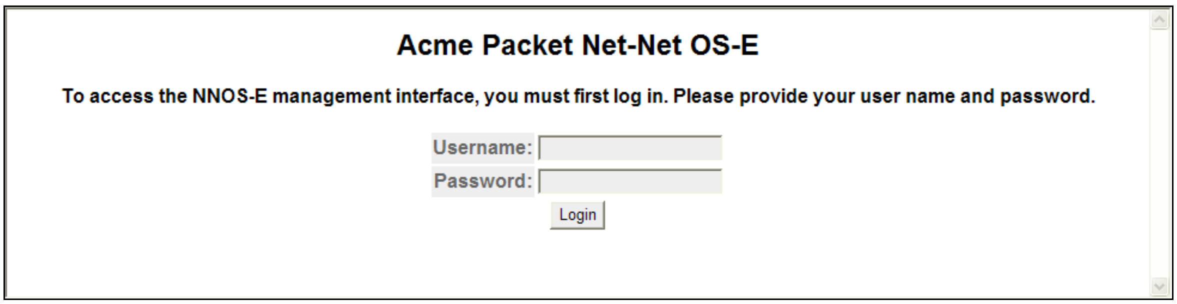
In the **Enterprise SIP Server** section:

- **IP Address**: Enter the IP address of the Enterprise SIP Server to which the SBC will connect. In the case of the compliance test, this is the IP address of the Session Manager SIP signaling interface.
- **Transport**: From the pull-down menu, select the transport protocol to be used for SIP traffic between the SBC and Session Manager.



7.2. Post Installation Configuration

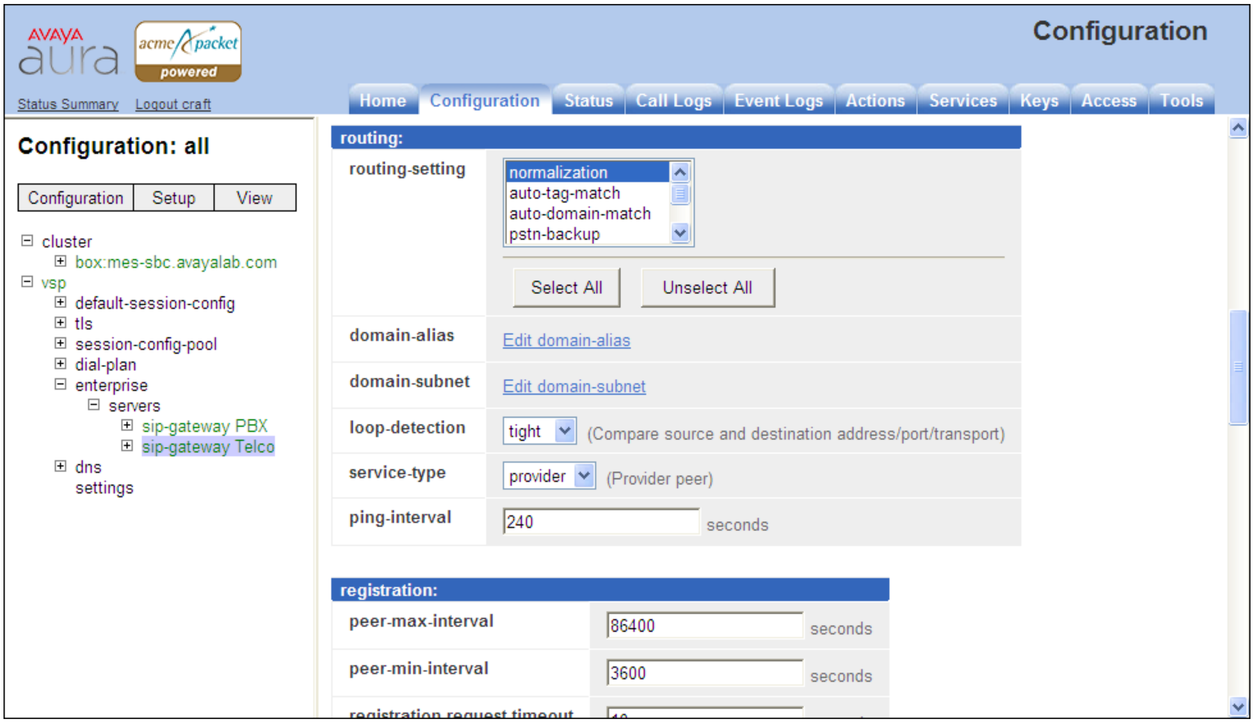
The installation wizard configures the Session Border Controller for use with the service provider chosen in **Section 7.1.3**. Since a different service provider other than PAETEC had to be selected in the installation wizard then additional manual changes must also be performed. These changes are performed by accessing the browser-based GUI of the Session Border Controller, using the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured in **Section 7.1.1**. Log in with the appropriate credentials set in **Section 7.1.2**.



7.2.1. Options Frequency

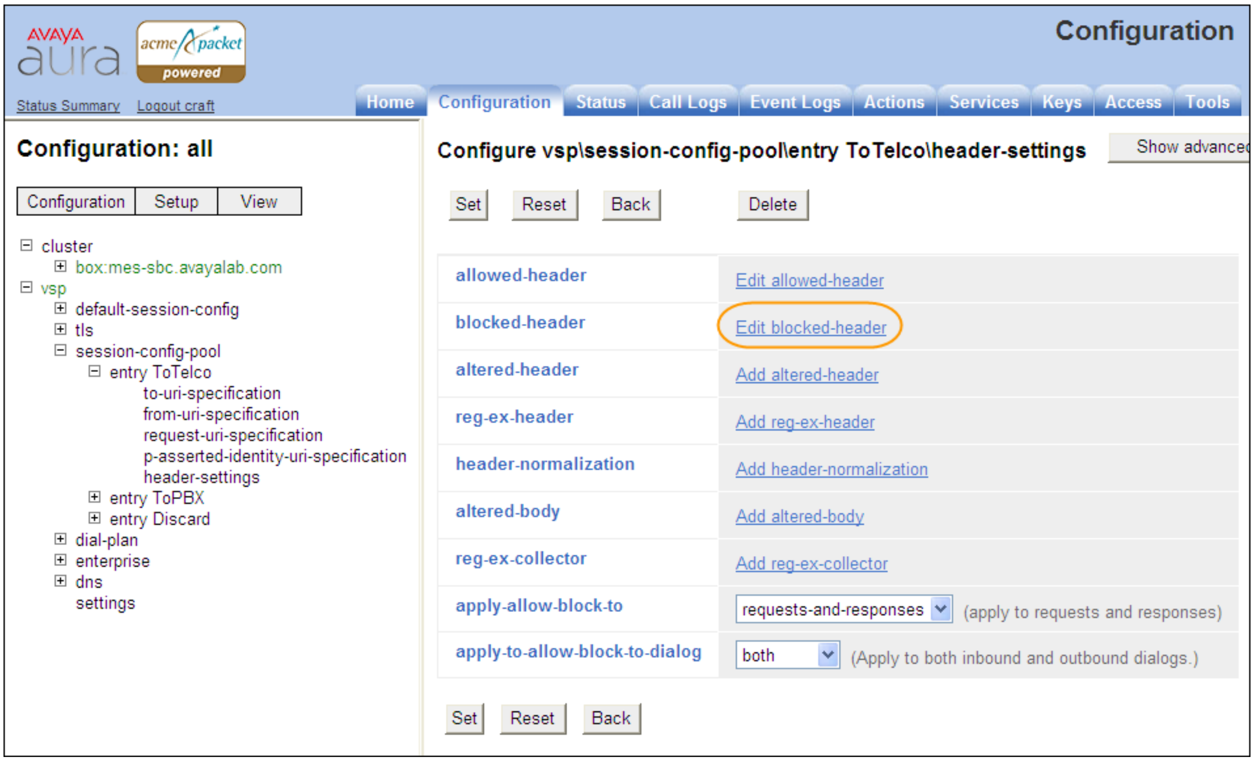
To set the frequency of the OPTIONS messages sent from the SBC to the service provider, first navigate to **vsp → enterprise → servers → sip-gateway Telco**. Click **Show Advanced.**

Scroll down to the **routing** section of the form. Enter the desired interval in the **ping-interval** field. For compliance testing ***240*** seconds was used. Click **Set** at the top of the form (shown in previous figure).

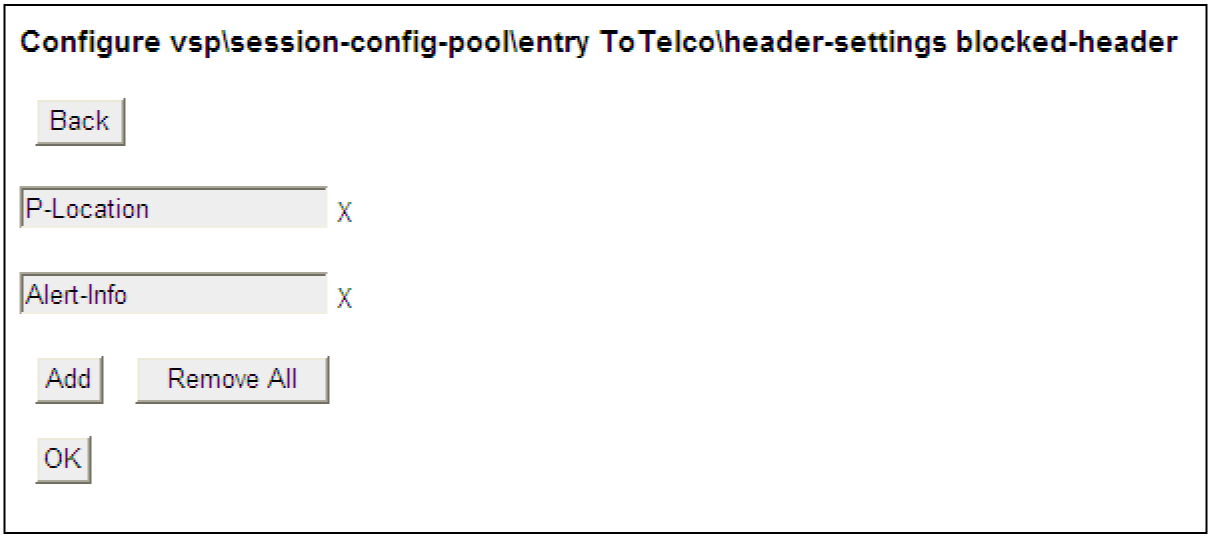


7.2.2. Blocked Headers

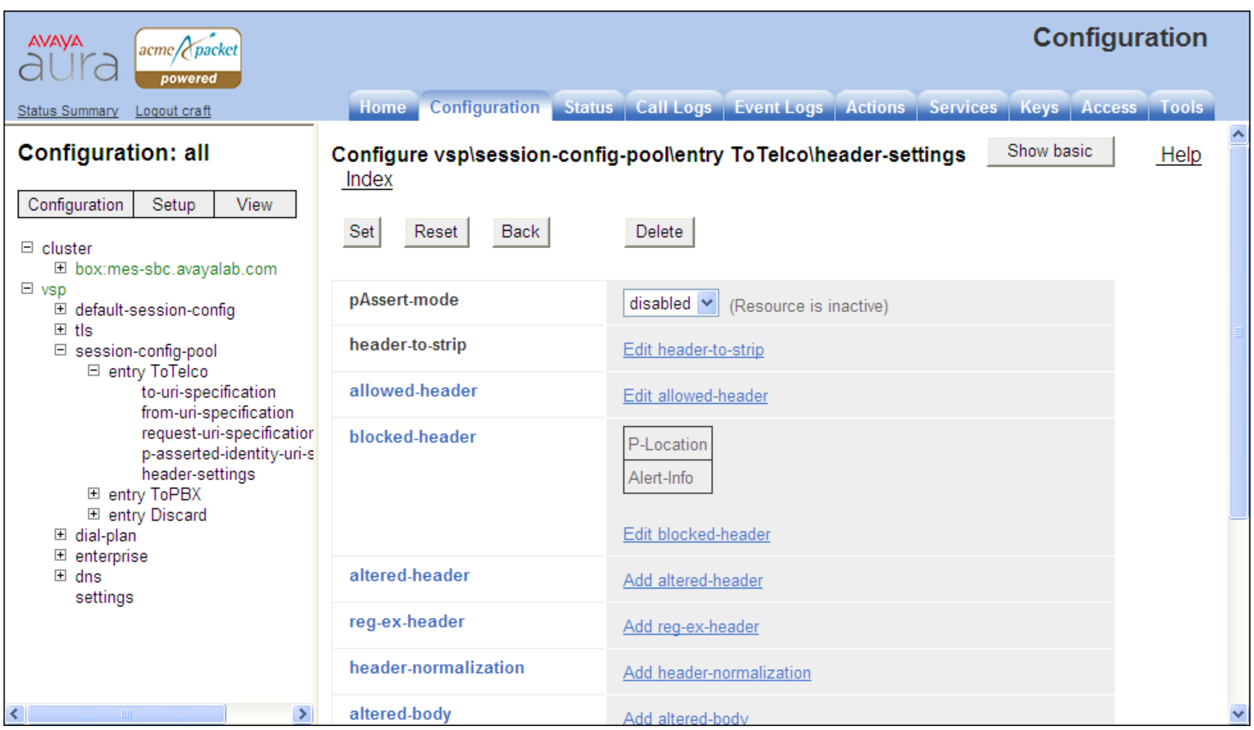
The P-Location and Alert-Info headers are sent in SIP messages from the Session Manager to the PAETEC network. These headers contain private IP addresses and SIP Domains from the enterprise. These should not be exposed external to the enterprise. These headers were simply removed (blocked) from both requests and responses for outbound calls. To create a rule for blocking a header on an outbound call, first navigate to **vsp → session-config-pool → entry ToTelco → header-settings**. Click **Edit blocked-header**.



In the right pane that appears, click **Add.** In the blank field that appears, enter the name of the header to be blocked. After all the blocked headers are added, click **OK**. The screen below shows the **P-Location** header and the **Alert-Info** header were configured to be blocked for the compliance test.

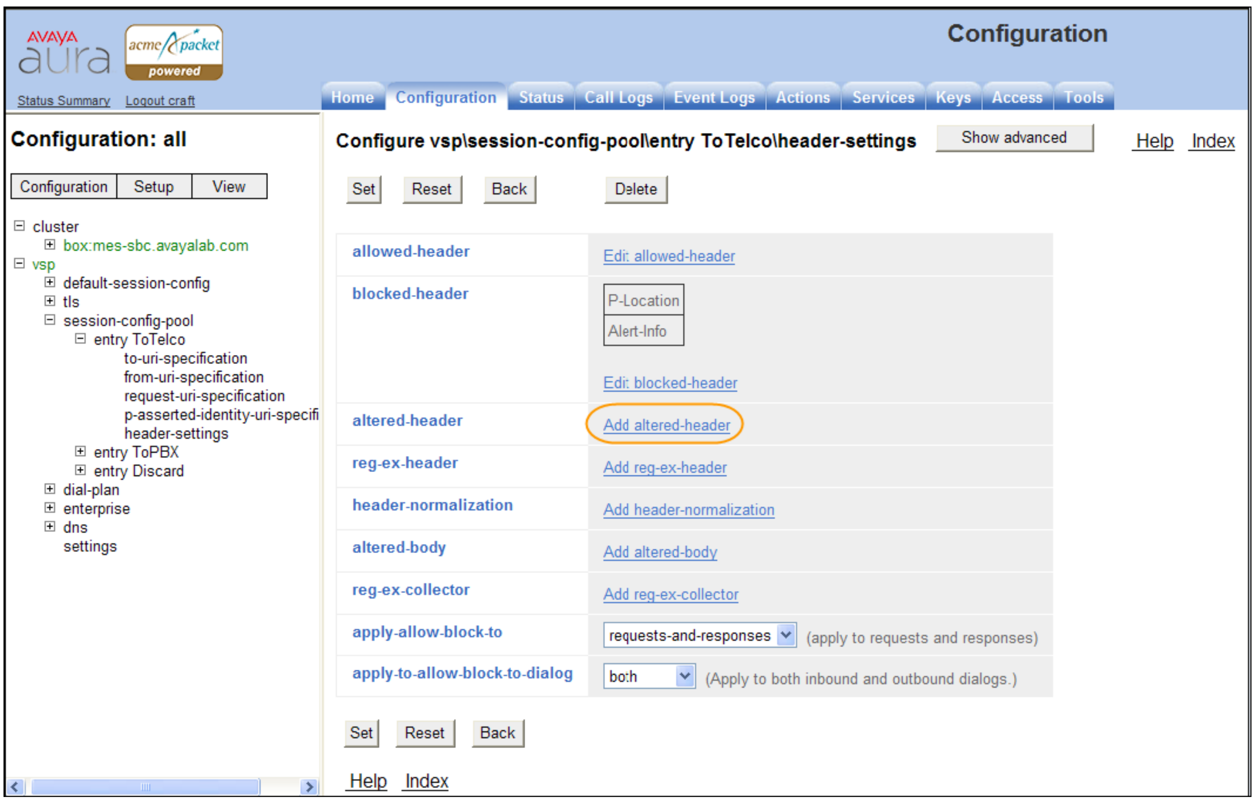


The list of blocked headers for outbound calls will appear the right pane as shown below. Click **Set** to complete the configuration.



7.2.3. Diversion Header

A Diversion Header is applied to forwarded off-net calls when the SIP trunk group on the Communication Manager has Send Diversion Header set to yes (**Section 5.7**). The Diversion Header will contain the number associated with the Enterprise user, allowing PAETEC to admit the call, and the From Header will be populated with the true calling party identity, allowing the forwarded destination to see the true caller ID. For the host portion of the header, Communication Manager sends the information entered in the signaling group Far-end Domain field **(Section 5.6).** To prevent this information from being exposed external to the enterprise, the SBC can modify the header and replace the Domain name with the IP address of the PAETEC Dynamic IP SIP Trunk. To create a rule to modify the Diversion Header first navigate to **vsp → session-config-pool → entry ToTelco → header-settings**. Click **Add altered-header**.

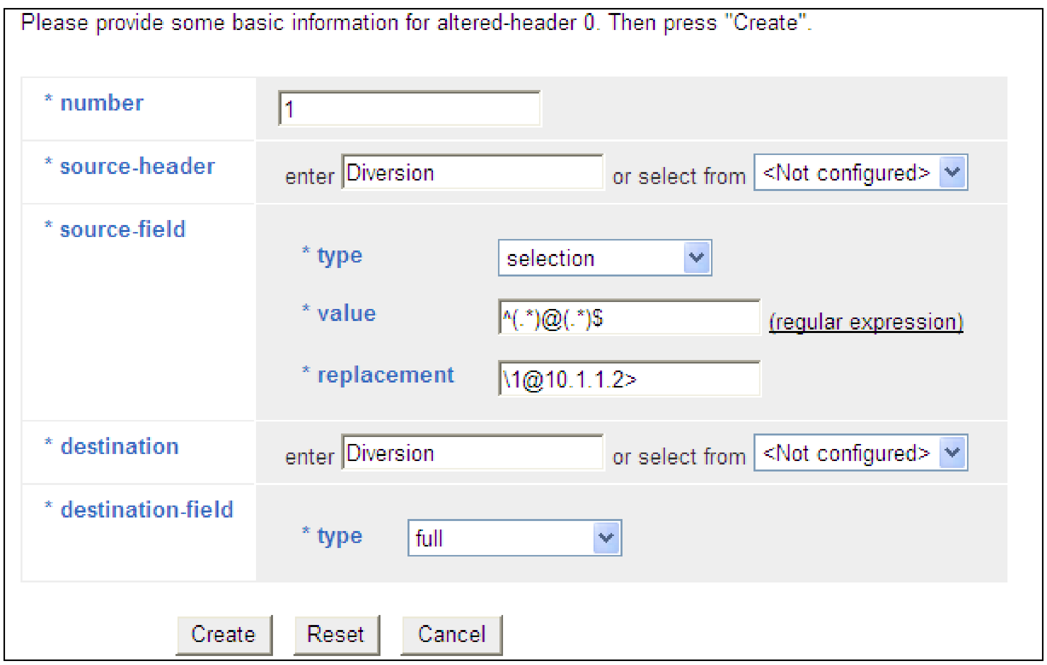


In the right pane that appears, enter any number in the **number** field and enter **“Diversion”** in the s**ource-header** field. In the **source-field** area, next to **type** choose “**selection”** from the drop-down list.

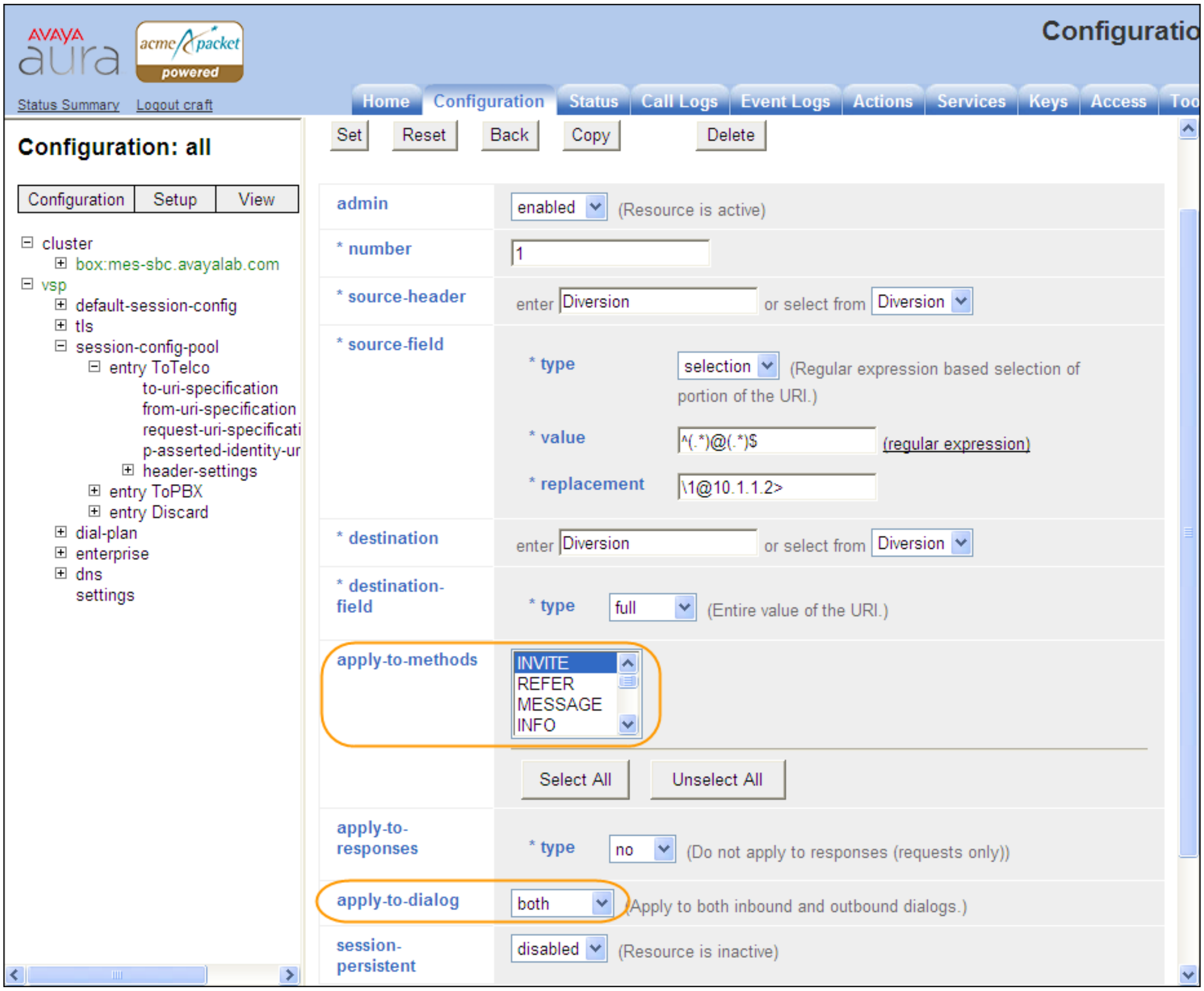
In the **value** field, enter a regular expression to match. In the sample configuration, **“^(.*)@(.*)$”** was entered. In this expression, the first (.*) will match and store any user part of the Diversion header. The second instance of (.*) matches and stores any host part of the header.

In the **replacement** field, **“\1@10.1.1.2>”** was entered in the sample configuration. The variable “\1” is the stored user part from the original Diversion header containing the number associated with the DID. The IP Address 10.1.1.2 is the IP Address of the PAETEC Dynamic IP SIP Trunk.

In the **destination** field and enter **“Diversion”** in the s**ource-header** field. Select “**full**” for **type** field in **destination-field** section and click **Create**.

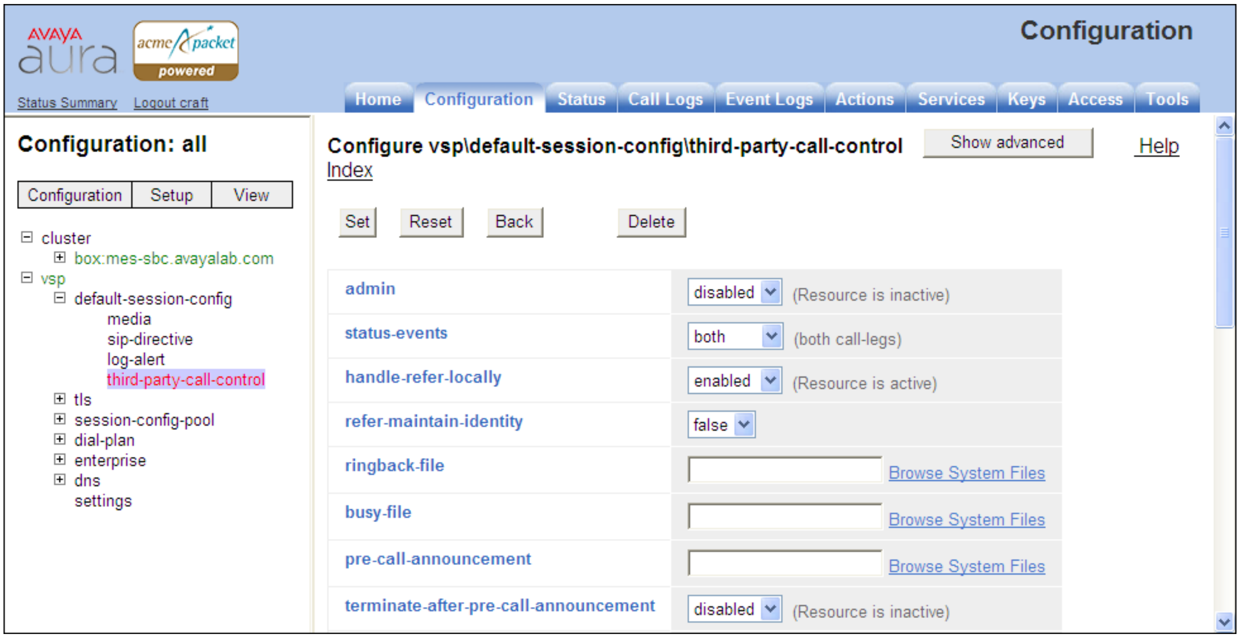


The following screen is presented, select "**INVITE**" for **apply-to-methods** and "**both**" for **type** field in **apply-to-responses** section. Click **Set** to complete the configuration.



7.2.4. Third Party Call Control

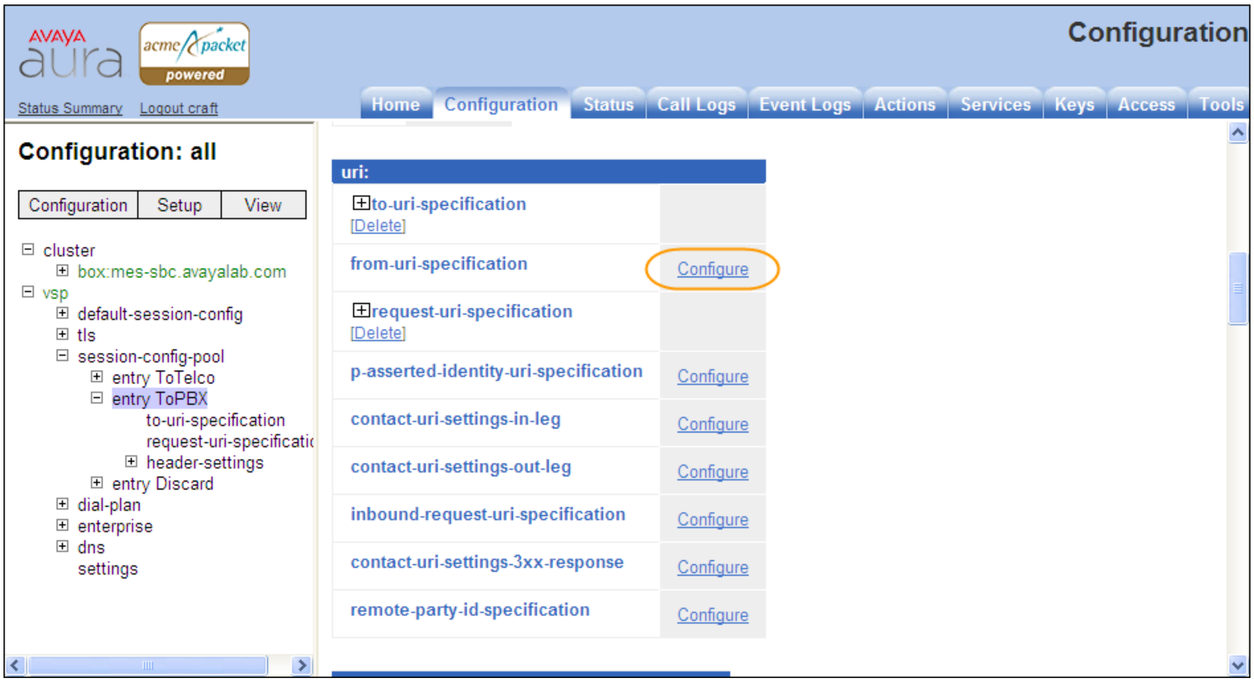
Disable third party call control. Navigate to **vsp → default-session-config → third-party-call-control**. Set the **admin** field to **disabled.**



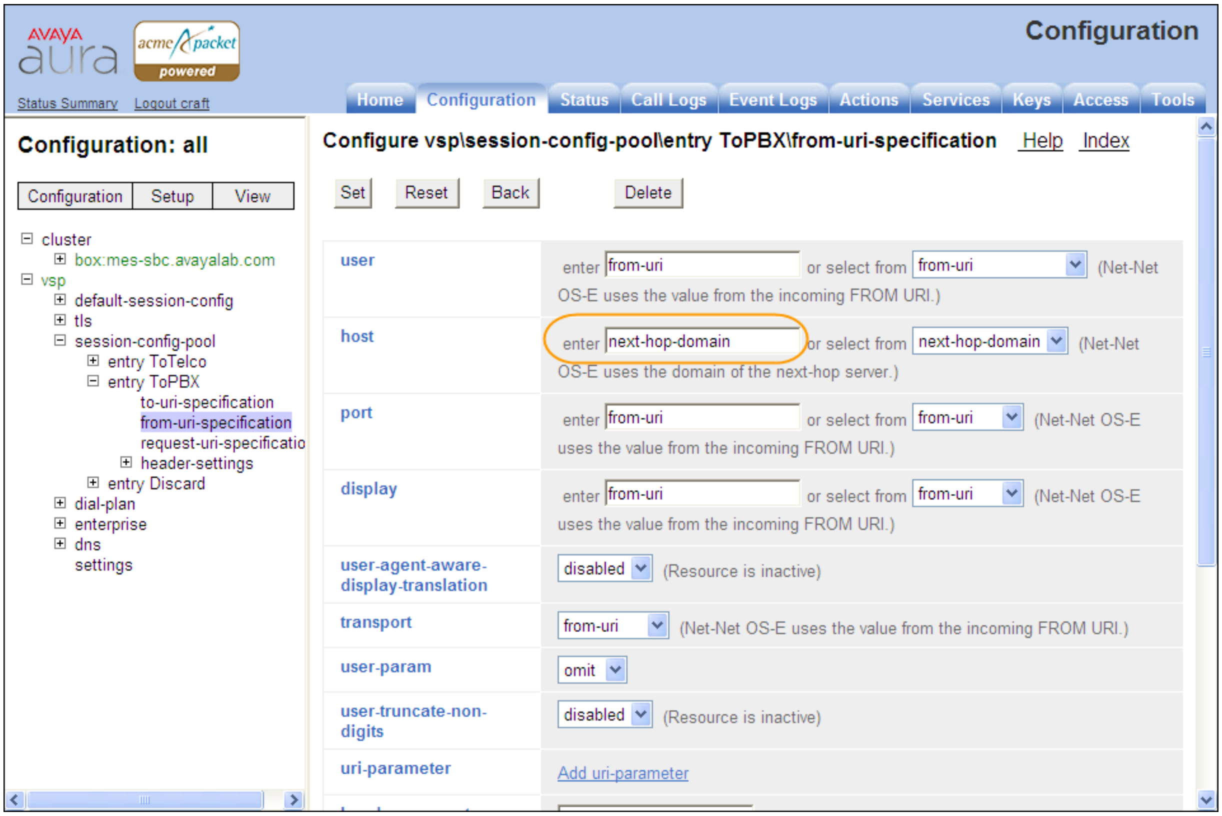
7.2.5. From URI

When calls are presented to SIP clients registered to Session Manager the Caller ID and Call Log displays the entire URI in the format user@domain (e.g. 303-555-1234@10.1.1.2). When placing a call from the Call Log it is necessary for the domain to be one that is authorized on the Session Manager for the call to route properly. Therefore it is necessary to change the host portion of the From header to the enterprise domain.

In the left side menu, navigate to **vsp → session-config-pool → entry ToPBX.** Scroll down and click on **Configure** next to **from-uri-specification.**



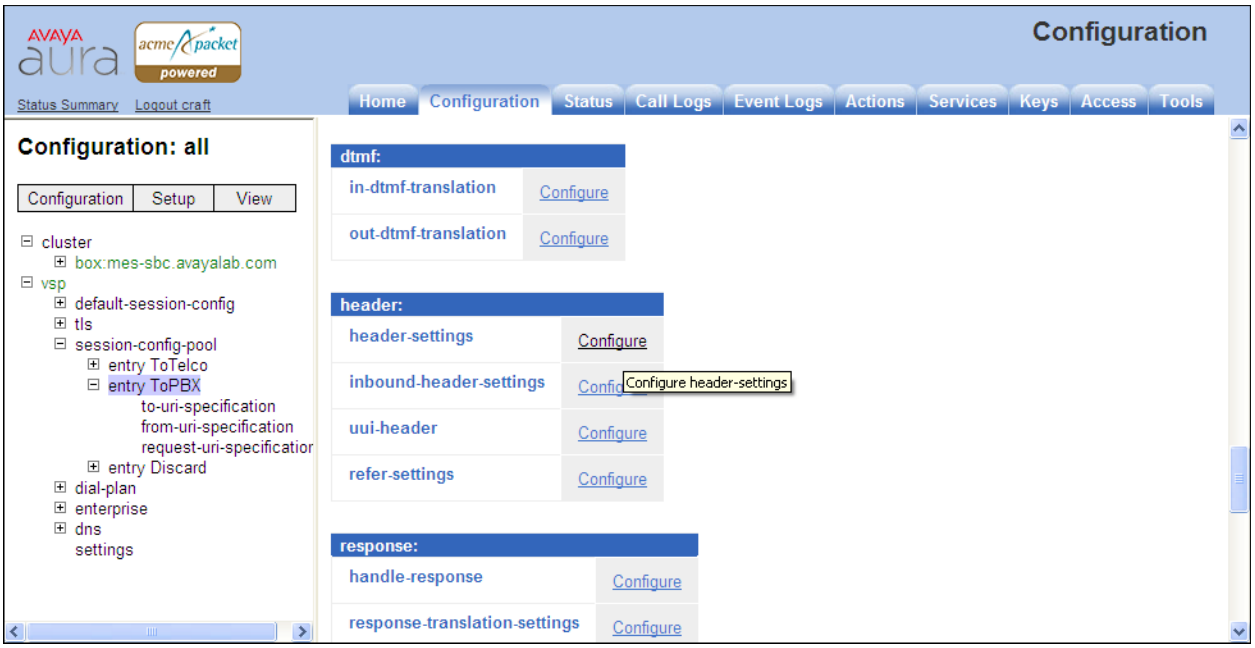
In the new right pane that appears, choose **next-hop-domain** from the drop-down list in the **host** field and click **Set**. This will set the host portion of the From Header to the enterprise domain set in **Section 7.1.1**.



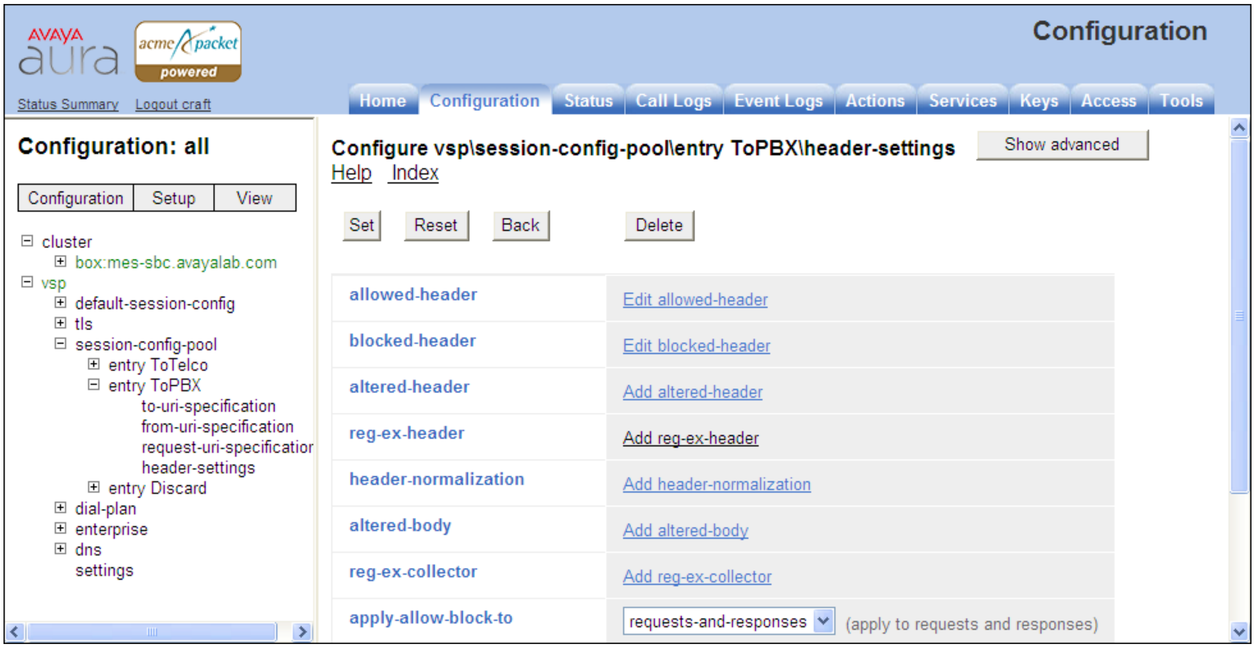
7.2.6. REFER-To Header

This section presents a sample configuration that will cause the SBC to modify the host portion of the Refer-To header in a REFER message, while preserving the user portion (containing the Refer-To destination telephone number) and any other information. In this example, the host portion was changed such that PAETEC would receive the PAETEC Dynamic SIP Trunk IP Address and port as the host portion.

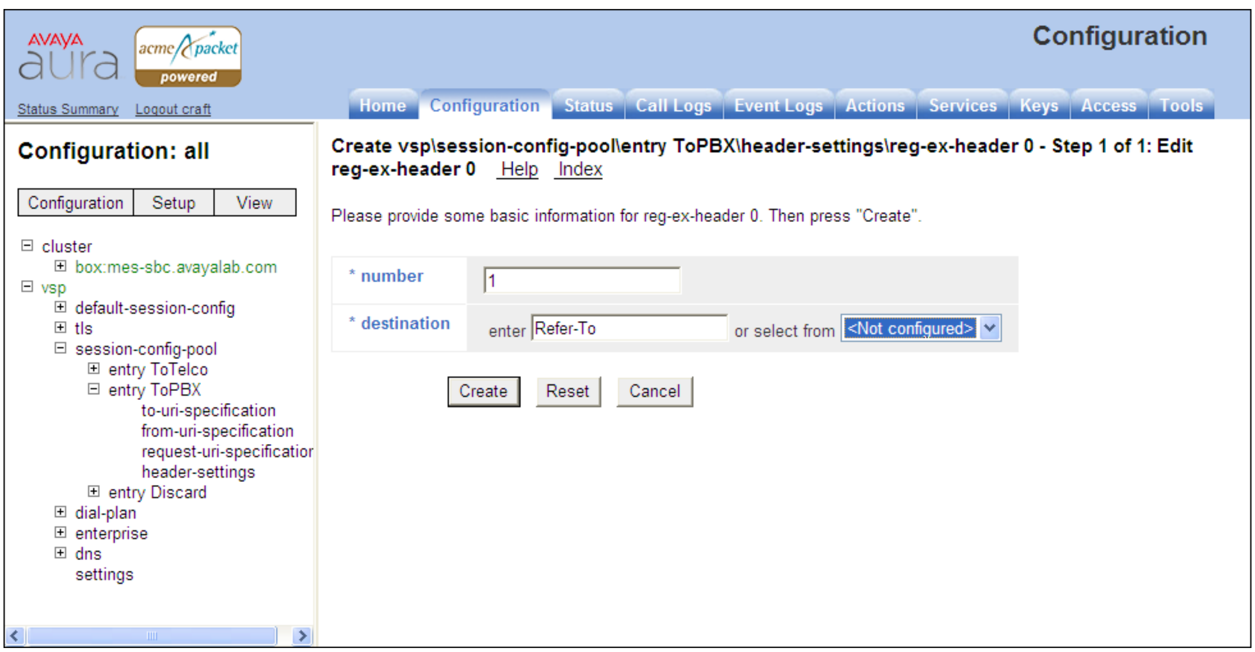
In the left side menu, navigate to **vsp → session-config-pool → entry ToPBX.** Click on **Configure** next to **header-settings**.



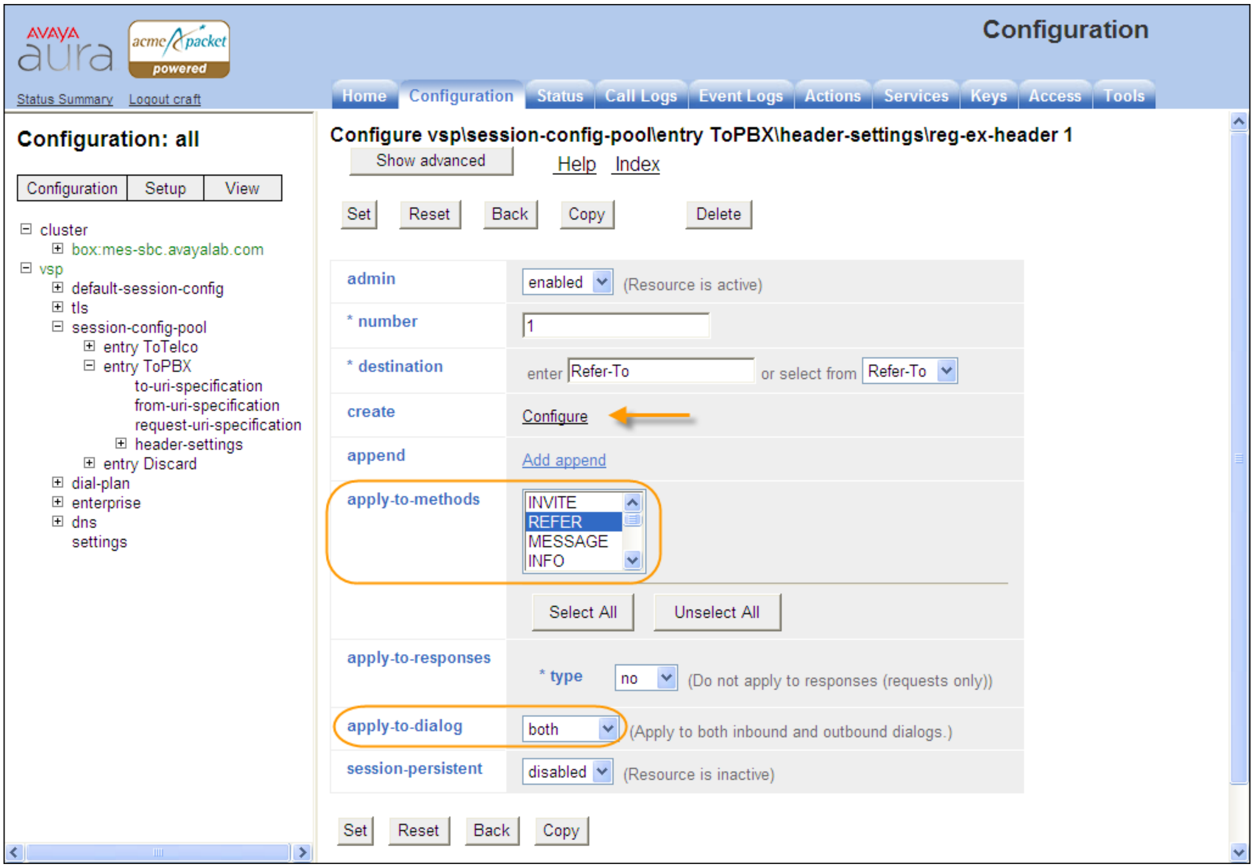
On the right panel, select **Add reg-ex-header** as shown below.



In the new right pane that appears, enter any number in the **number** field and enter "**Refer-To**" in the **destination** field and click **Create.**



The following screen is presented, select **REFER** for **apply-to-methods** and **both** for **type** field in **apply-to-responses** section. Select the **Configure** link to the right of **create**.

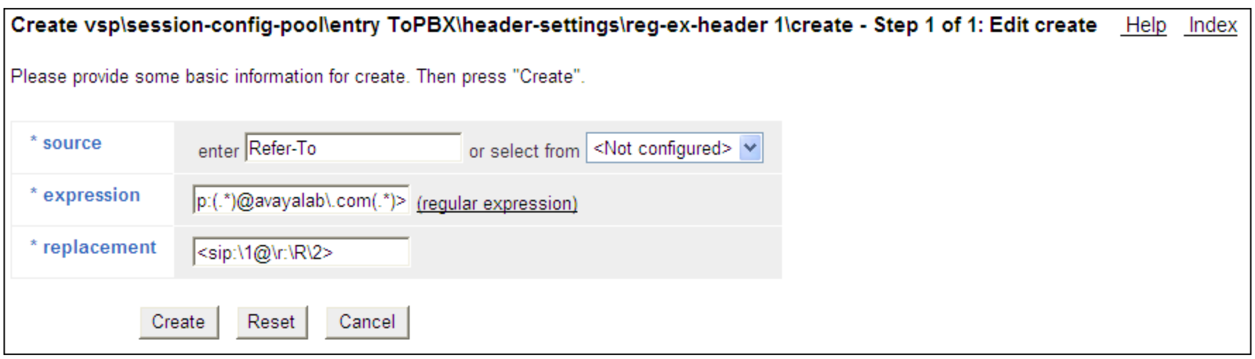


The following screen is presented. In the **source** area, select **Refer-To** from the drop-down list or type **Refer-To** in the **enter** field.

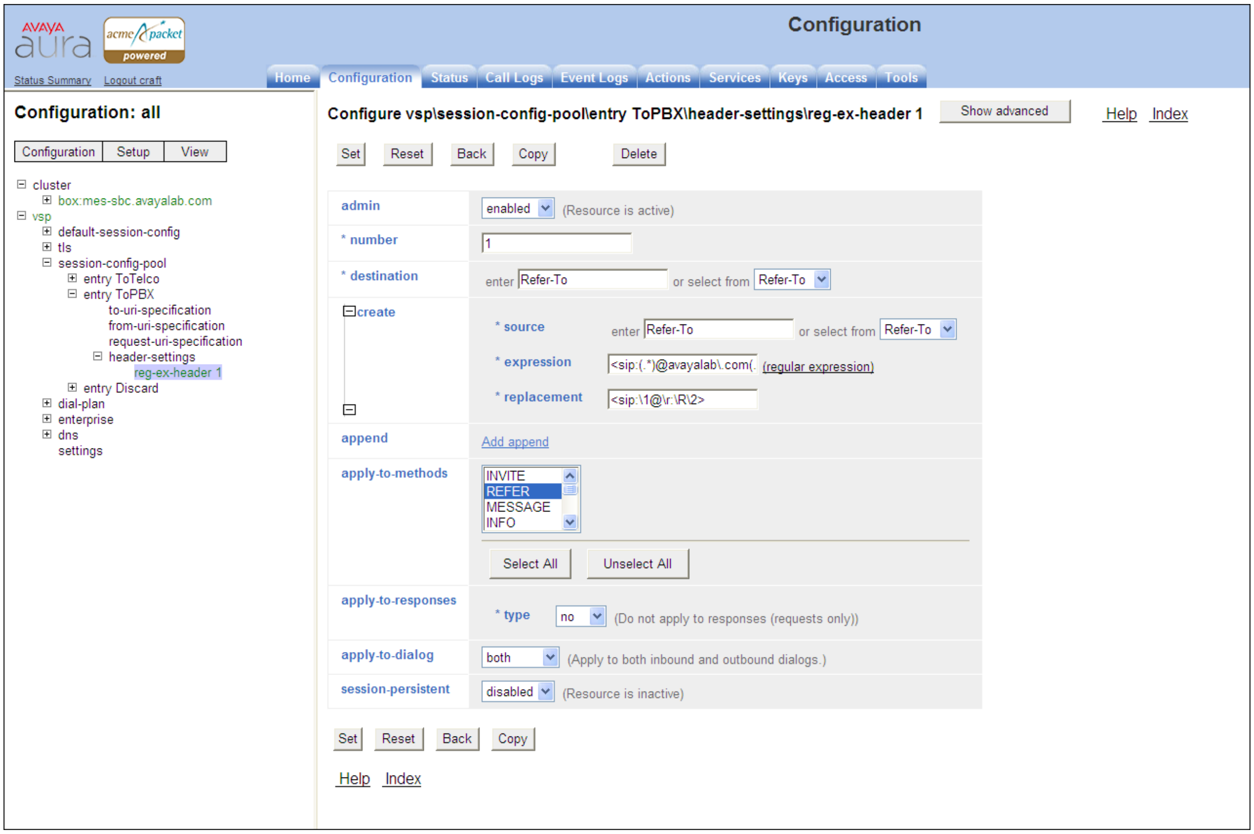
In the **expression** field, enter a regular expression to match. In the sample configuration, **<sip:(.*)@avayalab\.com(.*)>** was entered. In this expression, the first (.*) will match and store any user part of the Refer-To header. The second instance of (.*) matches and stores any UUI if present. The domain **avayalab.com** is what the SBC would otherwise put in the Refer-To header host part.

In the **replacement** field, **<sip:\1@\r:\R\2>** was entered in the sample configuration. The variable "\1" is the stored user part from the original Refer-To header containing the Refer-To number, corresponding to the first instance of **(.*)** from the **expression**. The variable **\2** is any stored UUI from the original Refer-To header, corresponding to the second instance of **(.*)** from the **expression**. The **\r** inserts the remote IP Address corresponding to the PAETEC Dynamic IP SIP Trunk IP Address. This is followed by a colon and **\R** corresponding to the PAETEC Dynamic IP SIP Trunk signaling port, which is 5060 in this case.

After completing the **source**, **expression** and **replacement** fields as appropriate, click **Create**.

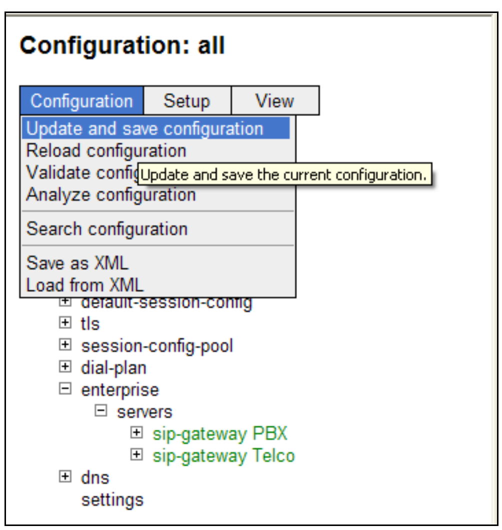


The following screen shows the completed rule. Click **Set** to complete the configuration.



7.2.7. Save the Configuration

To save the configuration, begin by clicking on **Configuration** in the left pane to display the configuration menu. Next, select **Update and save configuration**.



8. Dynamic IP SIP Trunk Service Configuration

To use the Dynamic IP SIP Trunk Service, a customer must request the service from PAETEC using their sales processes. This process can be initiated by contacting PAETEC via the corporate web site at www.paetec.com and requesting information via the online sales links or telephone numbers.

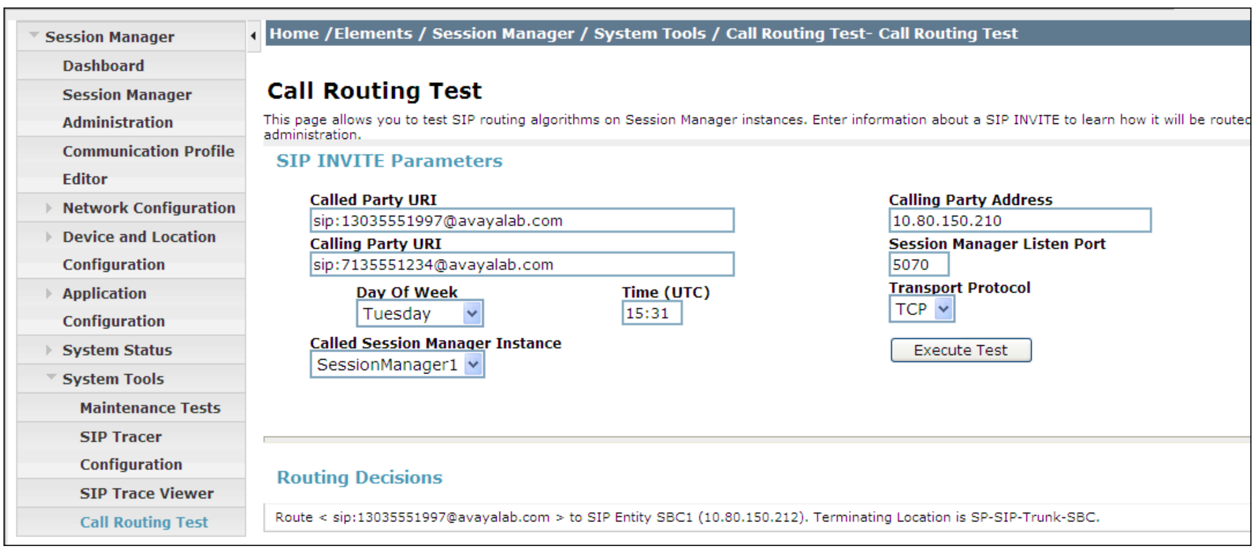
9. Verification and Troubleshooting

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.

9.1. Verification

The following steps may be used to verify the configuration:

1. Verify the call routing administration on the Session Manager by logging in to System Manger and execute the Call Routing Test. Expand **Elements → Session Manager → System Tools → Call Routing Test**. Populate the field for the call parameters of interest. For example, the following screen shows an example call routing test for an outbound call to PSTN via PAETEC. Under **Routing Decisions**, observe the call will rout via the SBC to PAETEC. Scroll down to inspect the details of the **Routing Decision Process** if desired (not shown).



2. 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.
3. 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.
4. Verify that the user on the PSTN can end an active call by hanging up.

5. Verify that an endpoint at the enterprise site can end an active call by hanging up.

Use **status trunk n** to verify the active call has ended. Where **n** is the trunk group number used for PAETEC Dynamic IP SIP Trunk Service.

Below is an example of an active call.

```
status trunk 1

                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0001/001 T00001   in-service/active  no   S00000
0001/002 T00002   in-service/idle    no
0001/003 T00003   in-service/idle    no
0001/004 T00004   in-service/idle    no
```

Verify the port returns to **in-service/idle** after the call has ended.

```
status trunk 1

                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0001/001 T00001   in-service/idle    no
0001/002 T00002   in-service/idle    no
0001/003 T00003   in-service/idle    no
0001/004 T00004   in-service/idle    no
```

9.2. Troubleshooting

1. Session Border Controller:
    - **Call Logs** - On the web user interface of the SBC, the **Call Logs** tab can provide useful diagnostic or troubleshooting information.
2. 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.
3. Session Manager:
    - **traceSM -x** – Session Manager command line tool for traffic analysis. Login to the Session Manager management interface to run this command.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Solution for Midsize Enterprise to the PAETEC Dynamic IP SIP Trunk Service. The PAETEC Dynamic IP SIP Trunk Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. The PAETEC Dynamic IP SIP Trunk Service provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks.

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] *Installing and Configuring Avaya Aura® Solution for Midsize Enterprise, Release 2.1, Issue 3 July 2011.*

[4] *Avaya Aura® Solution for the Midsize Enterprise (ME) 6.1 Intelligent Workbook. July 2011.*

[5] *Administering Avaya Aura™ Communication Manager*, *June2010, Document Number 03-300509.*

[6] *Avaya Aura™ Communication Manager Feature Description and Implementation, June 2010,* Document *Number 555-245-205.*

[7] *Installing and Upgrading Avaya Aura™ System Manager 6.1 GA Version*, *November 2010.*

[8] *Installing and Configuring Avaya Aura® Session Manager*, *April 2011, Document Number 03-603473*

[9] *Administering Avaya Aura® Session Manager*, *November 2010, Document Number 03-603324.*

[10] *Installing and Configuring Avaya Aura® Session Border Controller, November 2010.*

[11] *Avaya 1600 Series IP Deskphones Administrator Guide Release 1.3.x, April 2010, Document Number 16-601443.*

[12] 4600 Series IP Telephone LAN Administrator Guide, July 2008, Document Number 555-233-507.

[13] *Avaya one-X Deskphone H.323 Administrator Guide*, *May 2011, Document Number 16-300698.*

[14] *Avaya one-X Deskphone SIP Administrator Guide Release 6.1, December 2010, Document Number 16-603838*

[15] *Administering Avaya one-X Communicator, July 2011*

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

[17] *RFC 3515, The Session Initiation Protocol (SIP) Refer Method,* http://www.ietf.org/

[18] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, *http://www.ietf.org/*

[19] *RFC 4244, An Extension to the Session Initiation Protocol (SIP) for Request History Information*, *http://www.ietf.org/*

12. Appendix A: Avaya Aura® SBC Configuration File

```
#
#  Copyright (c) 2004-2011  Acme Packet Inc.
#  All Rights Reserved.
#
#  File: /cxc/cxc.cfg
#  Date: 16:15:02 Thu 2011-08-04
#
config cluster
 config box 1
  set hostname mes-sbc.avayalab.com
  set timezone America/Denver
  set name mes-sbc.avayalab.com
  set identifier 00:ca:fe:11:57:10
  config interface eth0
   config ip mgmt
    set ip-address static 10.80.150.212/24
    config ssh
     set mode ssh-2
    return
    config snmp
     set trap-target 10.80.150.208 162
     set trap-filter generic
     set trap-filter dos
     set trap-filter sip
     set trap-filter system
    return
    config web
     set ciphers
TLS_RSA_WITH_AES_128_CBC_SHA,TLS_DHE_RSA_WITH_AES_128_CBC_SHA,TLS_DHE_DSS_WIT
H_AES_128_CBC_SHA,SSL_RSA_WITH_3DES_EDE_CBC_SHA,SSL_DHE_RSA_WITH_3DES_EDE_CBC
_SHA,SSL_DHE_DSS_WITH_3DES_EDE_CBC_SHA,TLS_RSA_WITH_AES_256_CBC_SHA,TLS_DHE_R
SA_WITH_AES_256_CBC_SHA,TLS_DHE_DSS_WITH_AES_256_CBC_SHA
    return
    config web-service
     set protocol https 8443
     set authentication certificate "vsp\tls\certificate ws-cert"
    return
    config icmp
    return
    config routing
     config route Default
      set gateway 10.80.150.1
     return
     config route Static0
      set destination network 192.11.13.4/30
      set gateway 10.80.150.207
     return
     config route Static1
      set admin disabled
     return
     config route Static2
      set admin disabled
     return
     config route Static3
```

```
    set admin disabled
   return
   config route Static4
    set admin disabled
   return
   config route Static5
    set admin disabled
   return
   config route Static6
    set admin disabled
   return
   config route Static7
    set admin disabled
   return
   config route MgmtDefault
    set gateway 10.80.150.1
   return
  return
 return
return
config interface eth1
 config ip inside
  set ip-address static 10.80.150.217/24
  config sip
   set udp-port 5060 "" "" any 0
   set tcp-port 5060 "" "" any 0
   set tls-port 5061 "" "" TLS 0 "vsp\tls\certificate aasbc.p12"
  return
  config media-ports
  return
  config routing
   config route Default
    set admin disabled
   return
  return
 return
return
config interface eth2
 config ip outside
  set ip-address static 10.2.2.5/25
  config sip
   set udp-port 5060 "" "" any 0
  return
  config media-ports
  return
  config routing
   config route Default
    set admin disabled
   return
   config route external-sip-media-1
    set destination host 10.1.1.2
    set gateway 10.2.2.1
   return
  return
  config kernel-filter
   config allow-rule allow-sip-udp-from-peer-1
```

```
      set destination-port 5060
      set source-address/mask 10.1.1.2/32
      set protocol udp
     return
     config deny-rule deny-all-sip
      set destination-port 5060
     return
    return
   return
  return
  config cli
   set prompt mes-sbc.avayalab.com
  return
 return
return

config services
 config event-log
  config file access.log
   set filter access info
   set count 3
  return
  config file system.log
   set filter system info
   set count 3
  return
  config file general.log
   set filter general info
   set count 3
  return
  config file error.log
   set filter all error
   set count 3
  return
  config file db.log
   set filter db debug
   set filter dosDatabase info
   set count 3
  return
  config file management.log
   set filter management info
   set count 3
  return
  config file peer.log
   set filter sipSvr info
   set count 3
  return
  config file dos.log
   set filter dos alert
   set filter dosSip alert
   set filter dosTransport alert
   set filter dosUrl alert
   set count 3
  return
  config file krnlsys.log
   set filter krnlsys debug
```

```
   set count 3
  return
 return
return

config master-services
 config database
  set media enabled
 return
return

config vsp
 set admin enabled
 config default-session-config
  config media
   set anchor enabled
   set rtp-stats enabled
  return
  config sip-directive
   set directive allow
  return
  config log-alert
   set tracing enabled
   set apply-to-methods-for-filtered-logs INVITE+REFER
  return
  config third-party-call-control
  return
 return
 config tls
  config default-ca
   set ca-file /cxc/certs/sipca.pem
  return
  config certificate ws-cert
   set certificate-file /cxc/certs/ws.cert
  return
  config certificate aasbc.p12
   set certificate-file /cxc/certs/aasbc.p12
   set passphrase-tag aasbc-cert-tag
  return
 return
 config session-config-pool
  config entry ToTelco
   config to-uri-specification
    set host next-hop
   return
   config from-uri-specification
    set host local-ip
   return
   config request-uri-specification
    set host next-hop
   return
   config p-asserted-identity-uri-specification
    set host local-ip
   return
   config header-settings
    set blocked-header P-Location
```

```
    set blocked-header Alert-Info
    config altered-header 1
     set source-header Diversion
     set source-field selection ^(.*)@(.*)$ "\1@10.1.1.2>"
     set destination Diversion
     set destination-field full
    return
   return
  return
  config entry ToPBX
   config to-uri-specification
    set host next-hop-domain
   return
   config from-uri-specification
    set host next-hop-domain
   return
   config request-uri-specification
    set host next-hop-domain
   return
   config header-settings
    config reg-ex-header 1
     set destination Refer-To
     set create Refer-To "<sip:(.*)@avayalab\.com(.*)>" "<sip:\1@\r:\R\2>"
     set apply-to-methods REFER
    return
   return
  return
  config entry Discard
   config sip-directive
   return
  return
 return
 config dial-plan
  config route Default
   set priority 500
   set location-match-preferred exclusive
   set session-config vsp\session-config-pool\entry Discard
  return
  config source-route FromTelco
   set peer server "vsp\enterprise\servers\sip-gateway PBX"
   set source-match server "vsp\enterprise\servers\sip-gateway Telco"
  return
  config source-route FromPBX
   set peer server "vsp\enterprise\servers\sip-gateway Telco"
   set source-match server "vsp\enterprise\servers\sip-gateway PBX"
  return
 return
 config enterprise
  config servers
   config sip-gateway PBX
    set domain avayalab.com
    set failover-detection ping
    set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToPBX
    config server-pool
     config server PBX1
```

```
      set host 10.80.150.214
     return
    return
   return
   config sip-gateway Telco
    set failover-detection ping
    set ping-interval 240
    set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToTelco
    config server-pool
     config server Telco1
      set host 10.1.1.2
     return
    return
   return
  return
 return
 config dns
  config resolver
   config server 10.80.150.201
   return
  return
 return
 config settings
  set read-header-max 8191
 return
return

config external-services
return

config preferences
 config gui-preferences
  set enum-strings SIPSourceHeader Diversion
  set enum-strings SIPSourceHeader Refer-To
  set enum-strings SIPSourceHeader 1
  set show-unlicensed-features false
 return
return

config access
 config permissions superuser
  set cli advanced
 return
 config permissions read-only
  set config view
  set actions disabled
  set debug disabled
 return
 config users
  config user admin
   set password 0x00ef423a29a2107ee58ec0550339f5a61b5dba23c695975082403e542b
   set permissions access\permissions superuser
  return
  config user cust
   set password 0x00061a2062a4b3d3bc15918b0cebbe9bb5050eb8a42c63fb28a6ebd5ac
```

```
   set permissions access\permissions read-only
  return
  config user init
   set password 0x00755c995b232018224ae8f5484dce1ec3cb4dc2cb763694294c1a084d
   set permissions access\permissions superuser
  return
  config user craft
   set password 0x00c6cb901342e52636a403606bc1f3c1930bcbdcebe54fbc3a99fa171f
   set permissions access\permissions superuser
  return
  config user dadmin
   set password 0x007058db732aabf006eb0db4778db99706cb63e6e54631f7719165b568
   set permissions access\permissions read-only
  return
 return
return

config features
return
```

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.