Avaya Solution & Interoperability Test Lab

Application Notes for Configuring EarthLink Complete SIP Trunking with Avaya Aura® Communication Manager Evolution Server 6.2, Avaya Aura® Session Manager 6.2 and Acme Packet Net-Net 3800 Session Border Controller – Issue 1.0

Abstract

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

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1. Introduction

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

Customers using this Avaya SIP-enabled enterprise solution with EarthLink Complete 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 the EarthLink Complete SIP Trunking service 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 Acme SBC. Communication Manager and Session Manager were running on a single server as part of the Avaya Aura® Solution for Midsize Enterprise. However, these compliance test results are applicable to other server and media gateway platforms running similar versions of Communication Manager and Session Manager.

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:

- Response to SIP OPTIONS queries
- Incoming PSTN calls to various phone types including Avaya 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 including 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 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.

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- Various call types including: local, long distance, international, outbound toll-free, and local directory assistance (411).
- Codecs G.711MU and G.729A
- DTMF transmission using RFC 2833
- Caller ID presentation and Caller ID restriction
- Response to incomplete call attempts and trunk errors
- Voicemail navigation for inbound and outbound calls
- Voicemail Message Waiting Indicator (MWI)
- User features such as hold and resume, internal call forwarding, transfer, and conference
- Off-net call forwarding and enterprise mobility (extension to cellular)

Items not supported or not tested included the following:

- EarthLink Complete SIP Trunking was not configured to send SIP OPTIONS messages during the compliance test but will respond to the OPTIONS messages sent by the Acme SBC.
- Inbound toll-free, operator, operator services (0 + 10 digits) and emergency calls (911) are supported but were not tested as part of the compliance test.
- The SIP REFER method is not supported for network redirection.
- A “302 Moved Temporarily” response with new Contact header is not supported for network redirection.
- T.38 Fax is not supported

2.2. Test Results

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

- **SilenceSupp versus annexb for silence suppression:** EarthLink uses the SIP SDP parameter SilenceSupp to signal support for silence suppression. Communication Manager uses the annexb parameter for this purpose. As a result, even though both sides are configured to enable silence suppression, neither side is able to signal this properly to the other. During the compliance test no user perceived problems were observed though silence suppression was most likely not achieved in all cases.
- **G.729 codec and Avaya 96x0 SIP Telephones**: It was observed that in order for this service to interwork with Avaya 96x0 SIP phones using G.729, it was necessary to enable use of the G.729B codec (enabling silence suppression) on the internal trunk between SIP endpoints and Session Manager. See **Section 5.4** for configuration details.
- **Avaya one-X® Communicator and “Other Phone” Mode**: During the compliance test, dropped calls or no audio were observed during call transfer/conferencing with Avaya one-X® Communicator (H.323 and SIP) in “Other Phone” Mode. This is under investigation by Avaya. Use of Avaya one-X® Communicator in “Other Phone” Mode with Communication Manager 6.2 and this solution is not recommended.
- **Unexpected 127 RTP payload header**: During calls established using the G.711Mu codec, EarthLink sends some unexpected RTP packets with a payload type of 127

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interspersed with the valid G.711MU RTP packets with payload type 0. No user perceived problems were observed as a result of these unexpected RTP packets.

- **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 a re-INVITE message. EarthLink does not use the updated Contact header for displaying calling party information.

2.3. Support

For technical support on the EarthLink Complete SIP Trunking Service, contact EarthLink Business Customer Care by using the support links provided at www.earthlinkbusiness.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.

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3. Reference Configuration

Figure 1 illustrates a sample Avaya SIP-enabled enterprise solution connected to EarthLink Complete SIP Trunking. This is the configuration used for compliance testing.

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

- Communication Manager
- System Manager
- Session Manager
- Avaya G450 Media Gateway
- Avaya 1600-Series IP Telephones (H.323)
- Avaya 9600-Series IP Telephones (H.323 and SIP)
- Avaya one-X® Communicator (H.323 and SIP)
- Avaya A175 Desktop Video Device
- Avaya digital and analog telephones

Located at the edge of the enterprise is the Acme SBC. The Acme SBC 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 Acme SBC. In this way, the Acme SBC can protect the enterprise against any SIP-based attacks. The Acme 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.

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.

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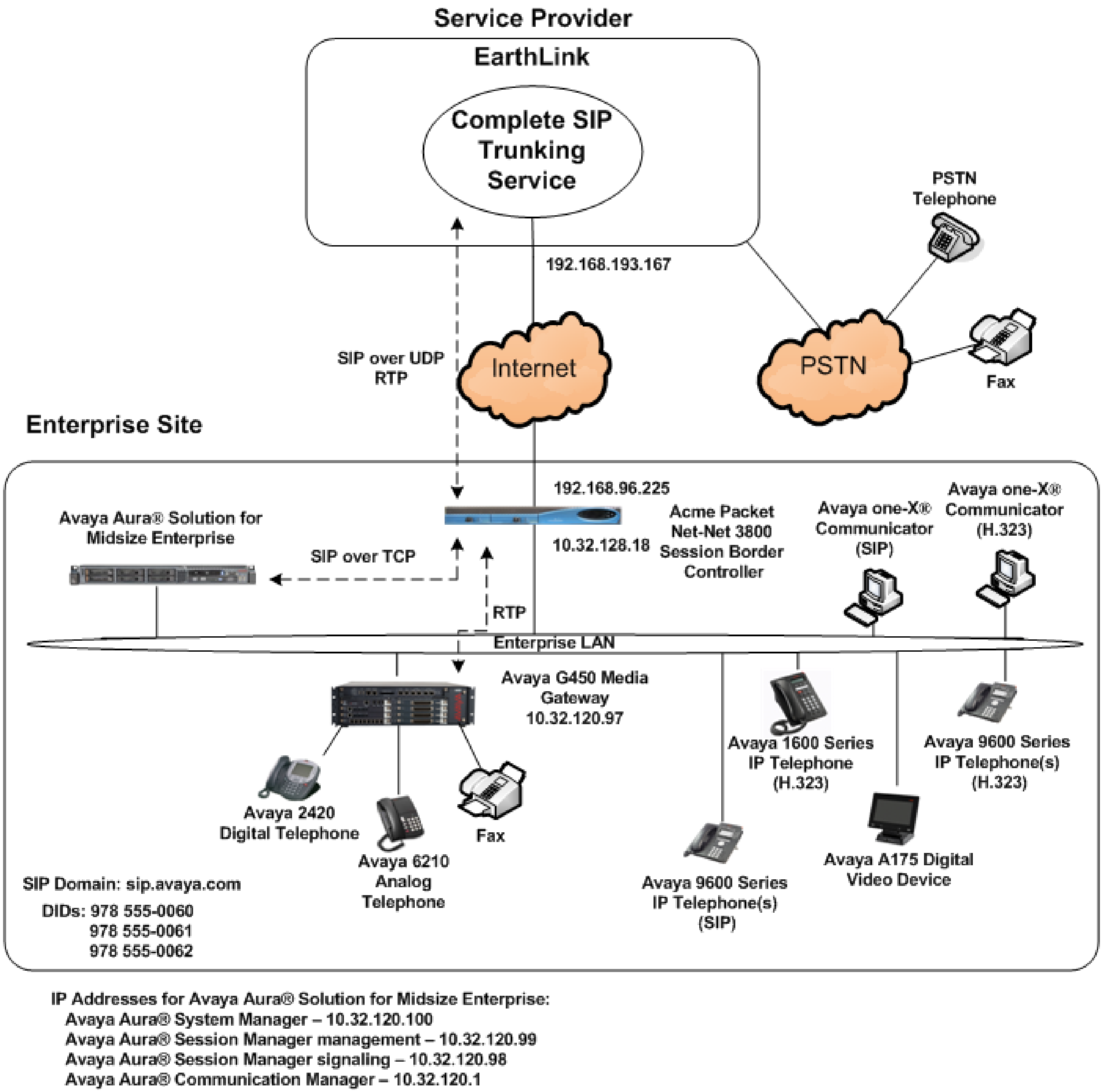

Figure 1: Avaya IP Telephony Network using EarthLink Complete SIP Trunking

For inbound calls, the calls flow from the service provider to the Acme 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 Acme SBC. From the Acme SBC, the call is sent to EarthLink Complete SIP Trunking.

For the compliance test, the enterprise sent 11 digits in the destination headers (e.g., Request-URI and To) and sent 10 digits in the source headers (e.g., From, Contact, and P-Asserted-Identity (PAI)) of the SIP messaging. EarthLink sent 10 digits in both the source 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	
Equipment/Software	Release/Version
Avaya Aura® Solution For Midsize Enterprise running on an HP Proliant DL360 Server - Avaya Aura® System Manager - Avaya Aura® Session Manager - Avaya Aura® Communication Manager - Avaya Aura® Communication Manager Messaging - System Platform	6.2 6.2 SP2 (Build 6.2.0.0.15669-6.2.12.202) (Software Update Revision 6.2.14.1.1925) 6.2 SP2 (Build 6.2.2.0.622005) 6.2 SP2 (Build R016x.02.0.823.0-19883) 6.2 SP0 (Build CMM-02.0.823.0-0002) 6.2.1.0.9
Avaya G450 Media Gateway	31.22.0
Avaya 1608 IP Telephone (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP1
Avaya 9640G IP Telephone (H.323) running Avaya one-X® Deskphone Edition	3.1 SP4 (3.1.04S)
Avaya 9641G IP Telephone (H.323) running Avaya one-X® Deskphone Edition	6.2 SP1 (S6.2119)
Avaya 9630 IP Telephone (SIP) running Avaya one-X® Deskphone SIP Edition	2.6 SP6 (2.6.6)
Avaya 9611 IP Telephone (SIP) running Avaya one-X® Deskphone SIP Edition	6.0 SP3 (6.0.3)
Avaya A175 Desktop Video Device with Avaya Flare® Experience	1.1
Avaya one-X® Communicator (H.323 or SIP)	6.1 SP5 (Build 6.1.5.07-SP5-37495)

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Avaya 2420 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Acme Packet Net-Net 3800 Session Border Controller	SC6.2.0 MR-3 GA (Build 619)
EarthLink Complete SIP Trunking Solution Components	
Component	Release
Metaswitch Softswitch	7.4
Acme Packet Net-Net 4500 Session Border Controller	6.1.0 M7P4

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.

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5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager for EarthLink Complete SIP Trunking. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from EarthLink. 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 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 **12000** SIP trunks are available and **275** 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 3
                       Maximum Administered SIP Trunks: 12000 275
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
```


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5.2. System Features

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

```
change system-parameters features                               Page   1 of  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 **anonymous** 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: anonymous
  CPN/ANI/ICLID Replacement for Unavailable Calls: anonymous

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

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

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 server running 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                  10.32.120.98
default             0.0.0.0
nwk-aes1            10.32.120.3
procr               10.32.120.1
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. The list should include the codecs and preferred order defined by EarthLink. For the compliance test, codecs G.729B and G.711MU were tested using ip-codec-set 4. To configure the codecs, enter the codecs in the **Audio Codec** column of the table in the order of preference. Default values can be used for all other fields.

In order to use G.729 with Avaya SIP endpoints and this solution, G.729B must also be enabled on the internal SIP trunk used by the SIP phones with Session Manager. Typically, G.711MU is already enabled on this trunk. For the compliance test this was ip-codec-set 3 which is not shown but it is similar to ip-codec-set 4 shown below.

```
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.729B           n          2        20
 2: G.711MU          n          2        20
 3:
```


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On **Page 2**, set the **Fax Mode** to **t.38-standard**.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

                    Mode                Redundancy
    FAX             t.38-stand           0
    Modem           off                  0
    TDD/TTY         US                   3
    Clear-channel   n                    0
```


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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 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 **sip.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 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 4                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 4
Location:         Authoritative Domain: sip.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
 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
```


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On **Page 4**, 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** (not shown).

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

 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
```

5.6. 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 4 was used for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** to the recommended default value of **tls** (Transport Layer Security). For ease of troubleshooting during testing, part of the compliance test was conducted with the **Transport Method** set to **tcp**. The transport method specified here is used between Communication Manager and Session Manager.
- Set the **IMS Enabled** field to **n**. This specifies the Communication Manager will serve as an Evolution Server for Session Manager.
- Set the **Peer Detection Enabled** field to **y**. The **Peer-Server** field will initially be set to **Others** and can not be changed via administration. Later, the **Peer-Server** field will automatically change to **SM** once Communication Manager detects its peer as a Session Manager.
- Set the **Near-end Node Name** to **procr**. This node name maps to the IP address of Communication Manager as defined in **Section 5.3**.
- 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 **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061 and for TCP the well-known port value is 5060). At the time of Session Manager installation, a

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SIP connection between Communication Manager and Session Manager would have been established for use by all Communication Manager SIP traffic using the well-known port value for TLS or TCP. By creating a new signaling group with a separate port value, a separate SIP connection is created between Communication Manager and Session Manager for SIP traffic to the service provider. As a result, any signaling group or trunk group settings (**Section 5.7**) 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 **5260**.

- 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 allowing Communication Manager to redirect media traffic directly between the SIP trunk and the enterprise endpoint.
- 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 4                                           Page   1 of   2
                                SIGNALING GROUP

 Group Number: 4              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                     Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                    Far-end Node Name: SM
 Near-end Listen Port: 5260                   Far-end Listen Port: 5260
                                           Far-end Network Region: 4
                                     Far-end Secondary Node Name:
Far-end Domain: sip.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? y                      Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 15
```


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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 4 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 4                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 4                   Group Type: sip          CDR Reports: y
  Group Name: SP Trunk                   COR: 1        TN: 1        TAC: *04
   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: 4
                                                    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 5.6**.

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

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```
change trunk-group 4                                            Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                             Redirect On OPTIM Failure: 15000

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

 Disconnect Supervision - In? y  Out? y

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

On **Page 3**, set the **Numbering Format** field to **private**. This field specifies the format of the calling party number (CPN) sent to the far-end. Beginning with Communication Manager 6.0, public numbers are automatically preceded with a + sign (E.164 numbering format) when passed in the SIP From, Contact and P-Asserted Identity headers. To remove the + sign, the **Numbering Format** was set to **private** and the **Numbering Format** in the route pattern was set to **unk-unk** (see **Section 5.9**).

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 4                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y


                     Numbering Format: private
                                                  UUI Treatment: service-provider

                                                   Replace Restricted Numbers? y
                                                  Replace Unavailable Numbers? y

                                  Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y

 DSN Term? n
```


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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** field provides additional information to the network if the call has been re-directed. These settings are needed by EarthLink 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 EarthLink.

```
add trunk-group 4                                                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


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

5.8. Calling Party Information

The calling party number is sent in the SIP "From", "Contact" and "PAI" headers. Since private numbering was selected to define the format of this number (**Section 5.7**), use the **change private-numbering** command to create an entry for each extension which has a DID assigned. The DID number will be assigned by the SIP service provider. It is used to authenticate the caller.

In the sample configuration, three DID numbers were assigned for testing. These three numbers were assigned to the three extensions 50003, 50006 and 50015. 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 three extensions.

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

Ext Ext              Trk        Private       Total
Len Code             Grp(s)     Prefix         Len
                                                    Total Administered: 4
 5  5                                          5       Maximum Entries: 240
 5  50003           4         9785550060      10
 5  50006           4         9785550061      10
 5  50015           4         9785550062      10
```

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

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

Ext Ext              Trk        Private       Total
Len Code             Grp(s)     Prefix         Len
                                                    Total Administered: 2
 5  5                                          5      Maximum Entries: 240
 5  5               4         97855            10
```


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5.9. 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
   0           1   attd
   1           5   ext
   5           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  11
                               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:
```


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

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    0                     1    1      4       op           n
    0                     11   11     4       op           n
    011                   10   18     4       intl         n
    1732                  11   11     4       fnpa         n
    1800                  11   11     4       fnpa         n
    1877                  11   11     4       fnpa         n
    1908                  11   11     4       fnpa         n
    411                   3    3      4       svcl         n
    978555                10   10     4       natl         n
```


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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 route pattern in the following manner. The example below shows the values used for route pattern 4 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 **4** 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.
- **Numbering Format**: **unk-unk** All calls using this route pattern will use the private numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.7**.
- **LAR**: **next**

```
change route-pattern 4                                          Page   1 of   3
                    Pattern Number: 4   Pattern Name: TM 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: 4    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                               unk-unk   next
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```


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5.10. Save Translation

Use the **save translation** command to save the changes.

```
save translation

                              SAVE TRANSLATION

         Command Completion Status                            Error Code

         Success                                               0
```


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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 Acme 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.

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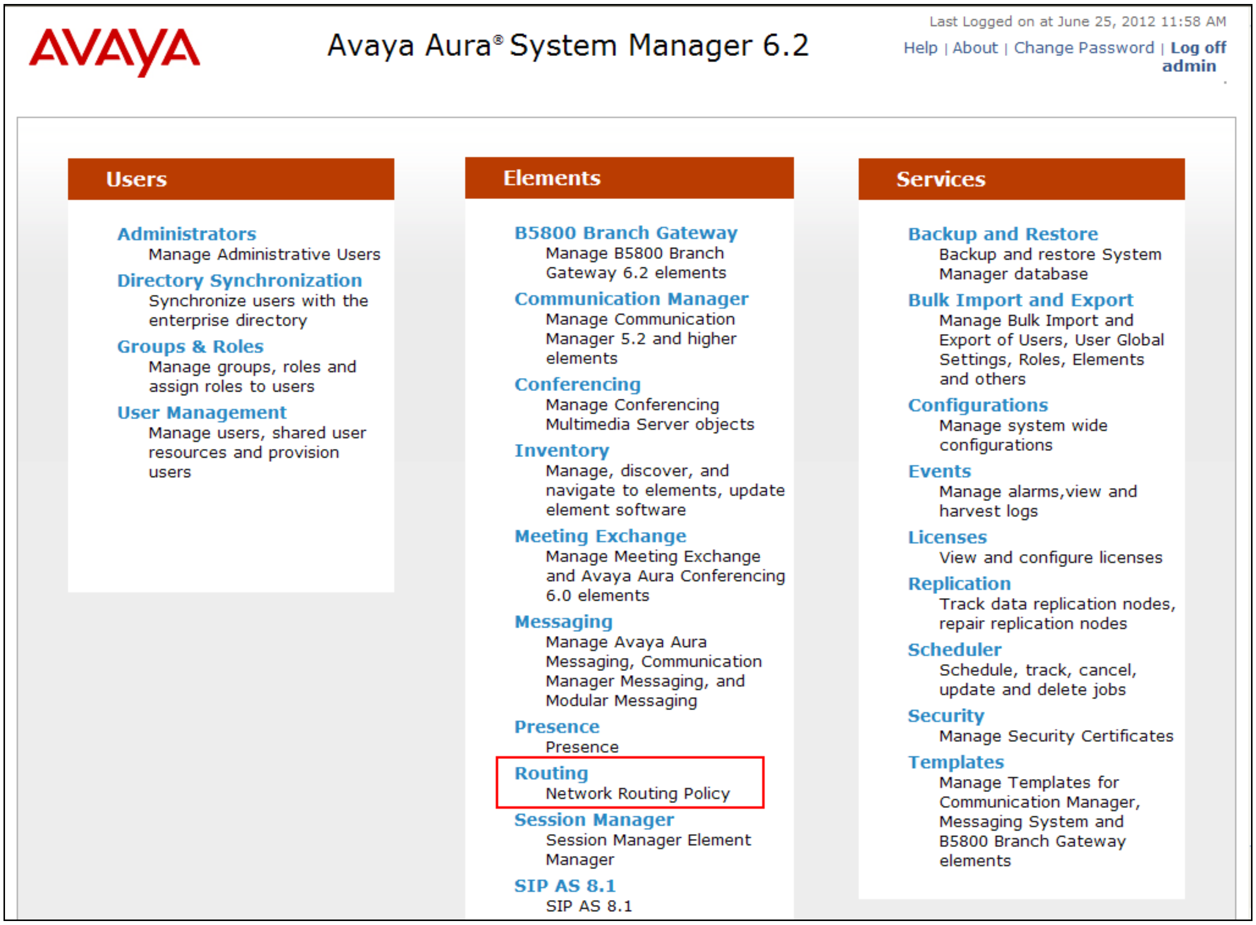
6.1. Avaya Aura® System Manager Login and Navigation

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.



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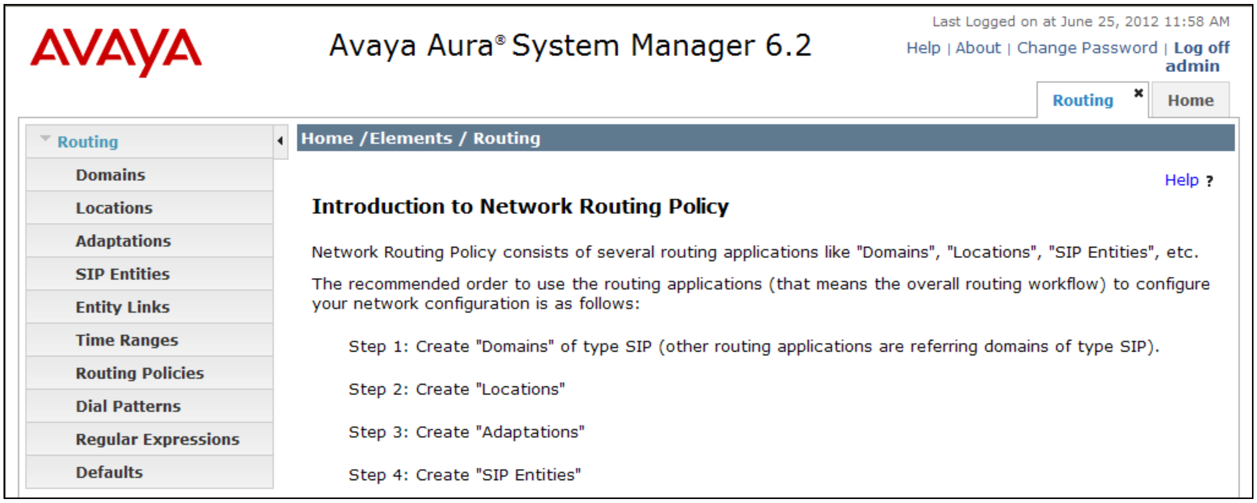
Clicking the **Elements → Routing** link, displays the **Introduction to Network Routing Policy** page. In the left-hand pane is a navigation tree containing many of the items to be configured in the following sections.



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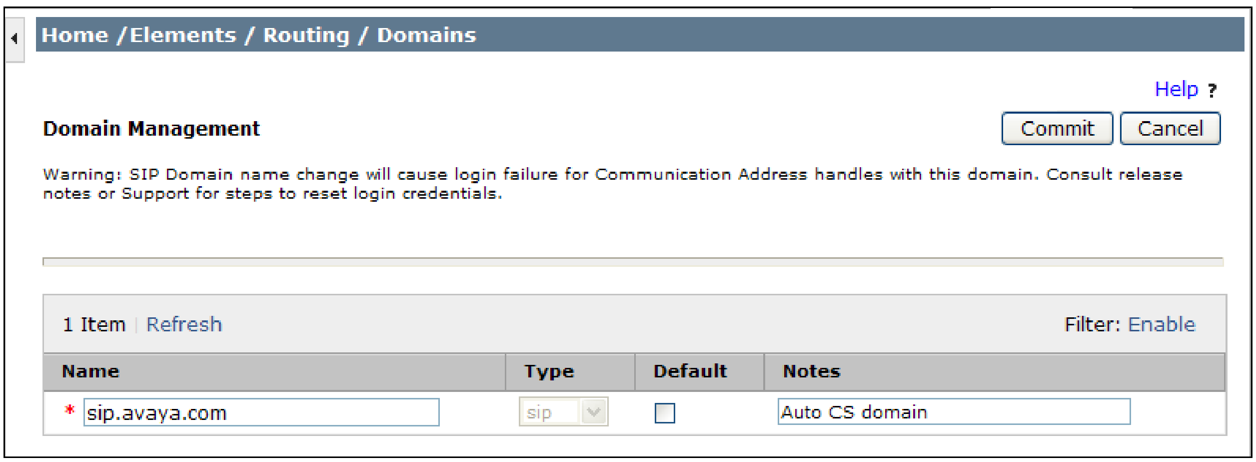
6.2. Specify SIP Domain

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 (**sip.avaya.com**). This is the domain configured on Communication Manager in **Sections 5.5** and **5.6**. Navigate to **Routing → Domains** in the left-hand navigation pane (**Section 6.1**) and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

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



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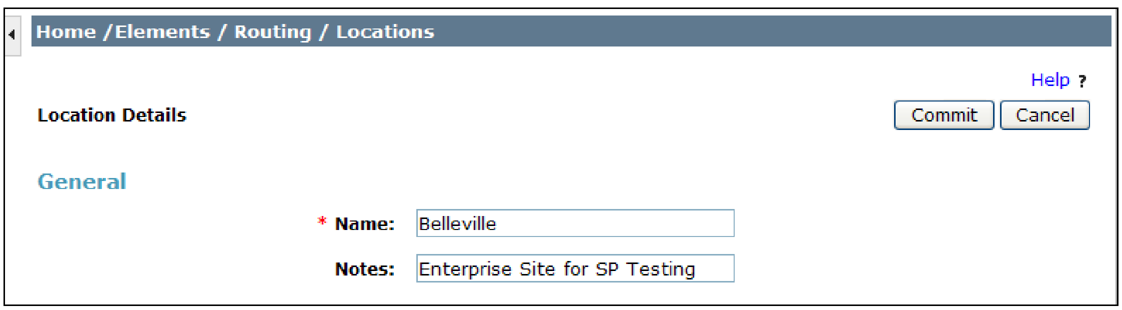
6.3. Add Location

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 **Belleville**, which includes all equipment on the enterprise including Communication Manager, Session Manager and the Acme 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.

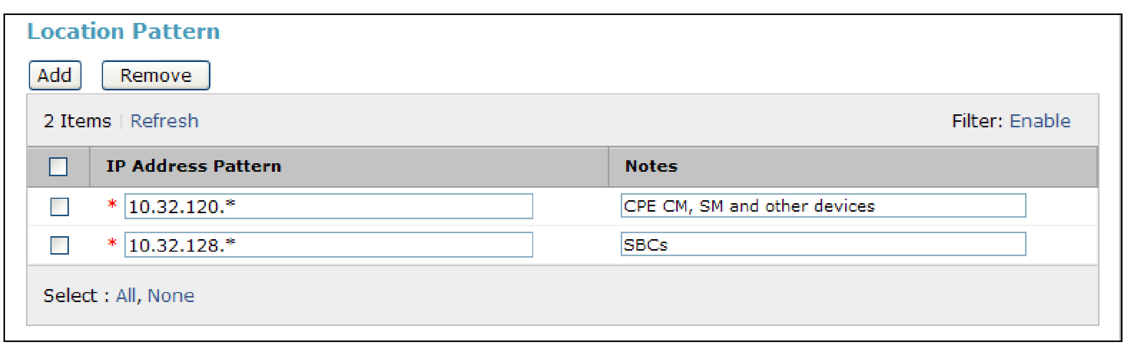
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).



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

- **IP Address Pattern:** Add all IP address patterns used to identify the location. The test environment included two subnets as shown below.
- **Notes:** Add a brief description (optional).

Click **Commit** to save.



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6.4. Add Adaptation Module

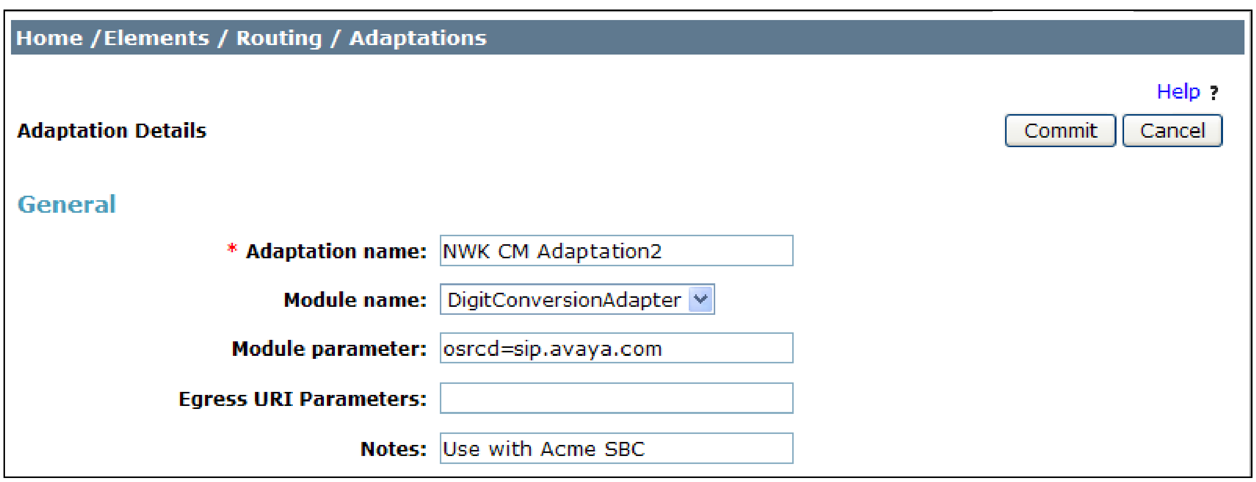
Session Manager can be configured with adaptation modules that can modify SIP messages before or after routing decisions have been made. A generic adaptation module **DigitConversionAdapter** supports digit conversion of telephone numbers in specific headers of SIP messages. Other adaptation modules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

For the compliance test one adaptation was created. The adaptation was applied to the Communication Manager SIP entity and converts the domain part of the inbound PAI header to the enterprise domain (**sip.avaya.com**). In addition, this adaptation maps inbound DID numbers from EarthLink to local Communication Manager extensions.

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=sip.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 5.6**.



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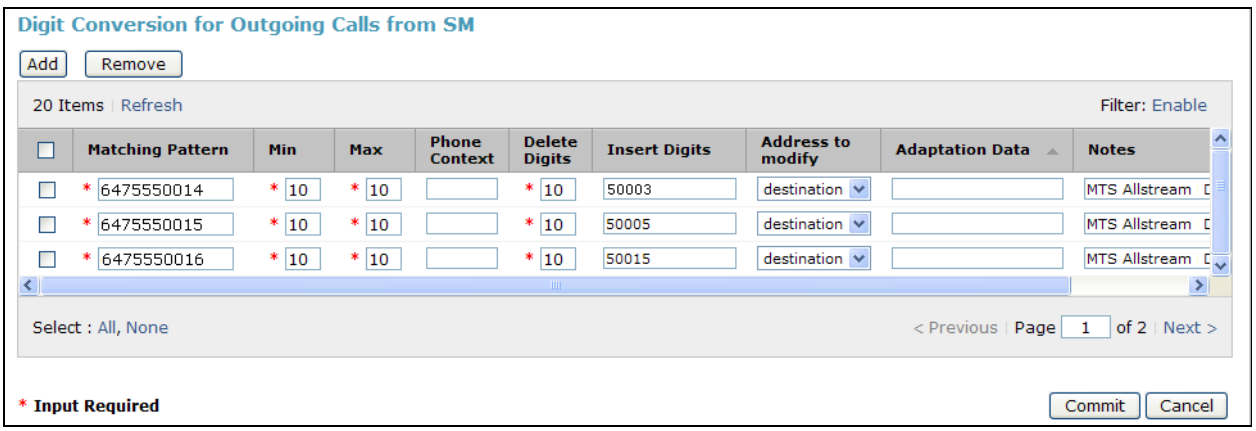
To map inbound DID numbers from EarthLink to Communication Manager extensions, scroll down to the **Digit Conversion for Outgoing Calls from SM** section. Create an entry for each DID to be mapped. Click **Add** and enter the following values for each mapping. Use default values for all remaining fields.

- **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.


Digit Conversion for Outgoing Calls from SM

Add Remove

20 Items | Refresh Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* 6475550014	* 10	* 10		* 10	50003	destination		MTS Allstream D
	* 6475550015	* 10	* 10		* 10	50005	destination		MTS Allstream D
	* 6475550016	* 10	* 10		* 10	50015	destination		MTS Allstream D

Select : All, None < Previous | Page 1 of 2 | Next >

* Input Required Commit Cancel

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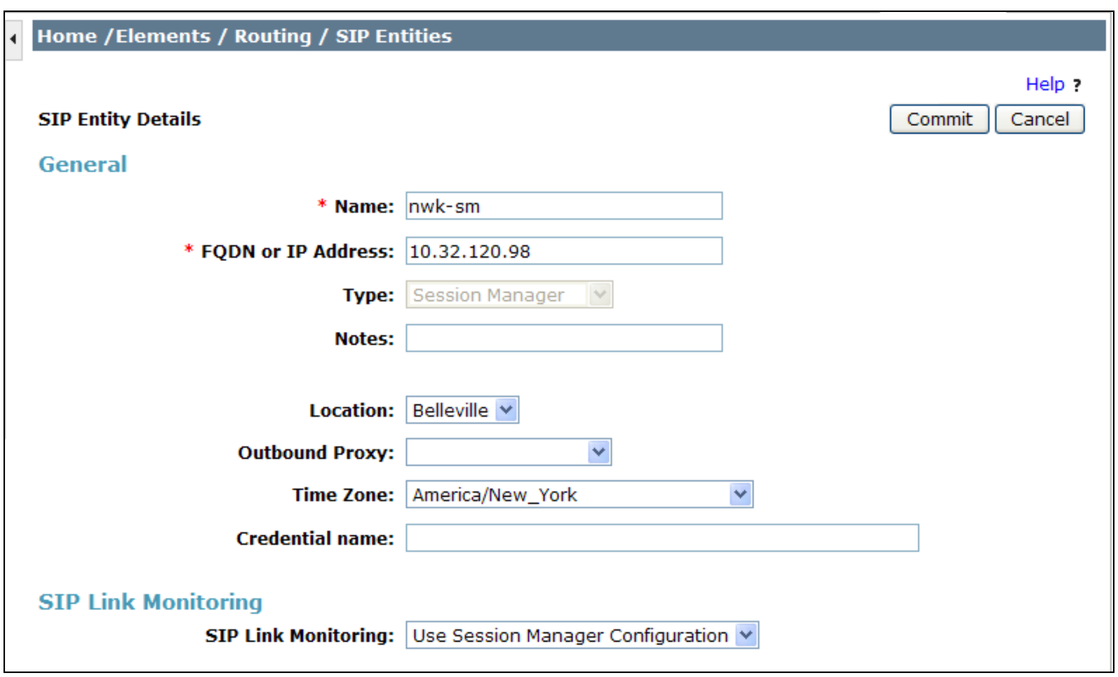
6.5. Add SIP Entities

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 Acme 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 Acme 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 **Belleville**.
- **Time Zone:** Select the time zone for the location above.

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



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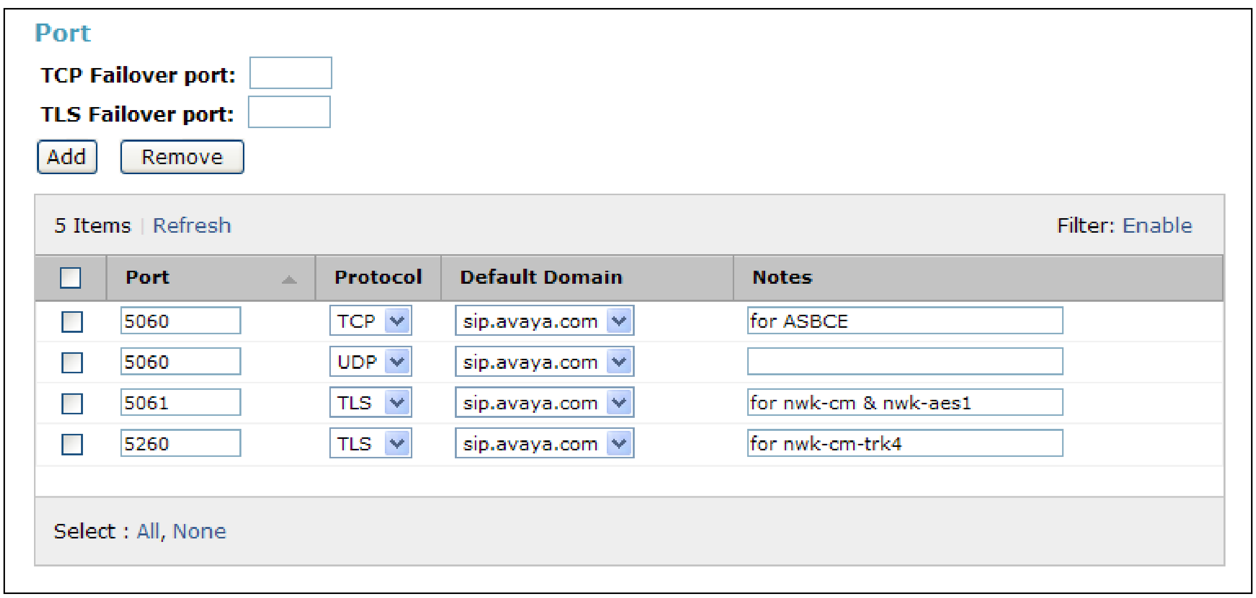
To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP entities.

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.
- **Note** Optional note relating to the entry.

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 5260 defined in **Section 5.6** for use with service provider SIP traffic between Communication Manager and Session Manager was added to the list.



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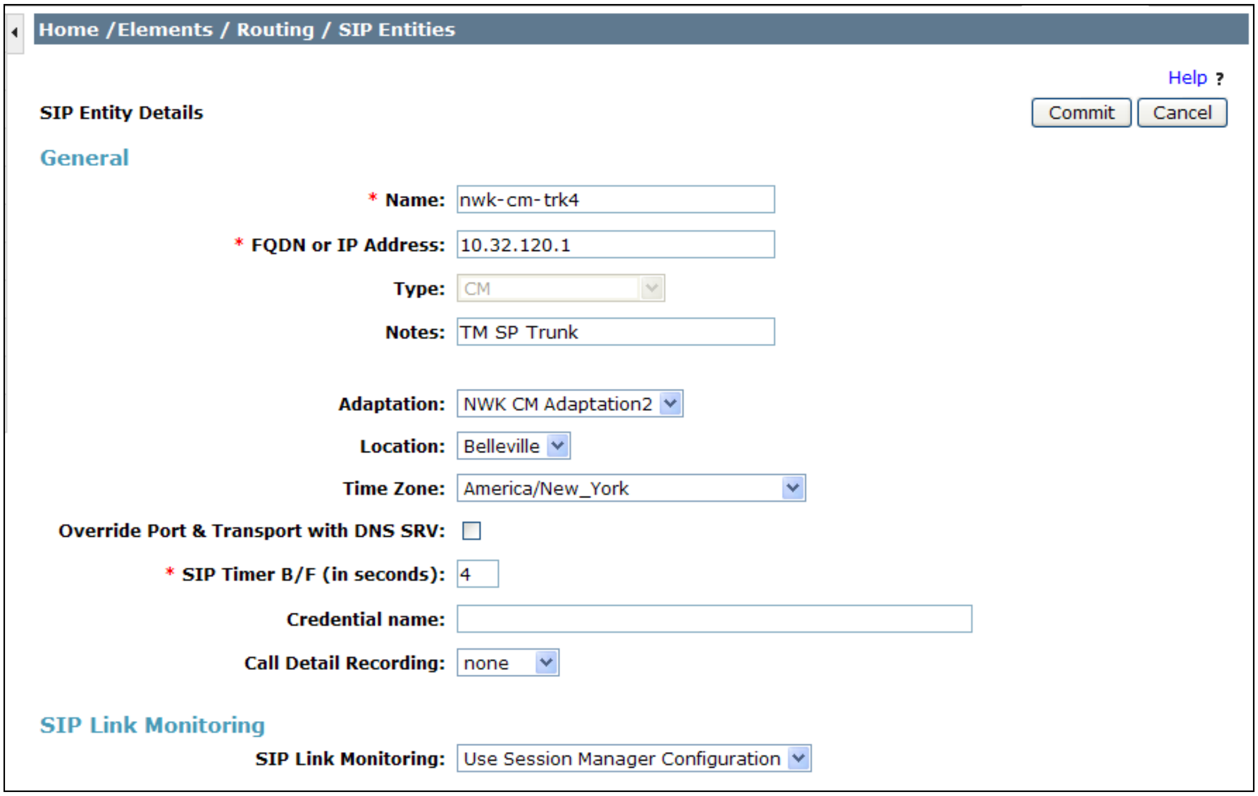
The following screen shows the addition of the Communication Manager SIP Entity. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, this requires the creation of a separate SIP entity for Communication Manager other than the one created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



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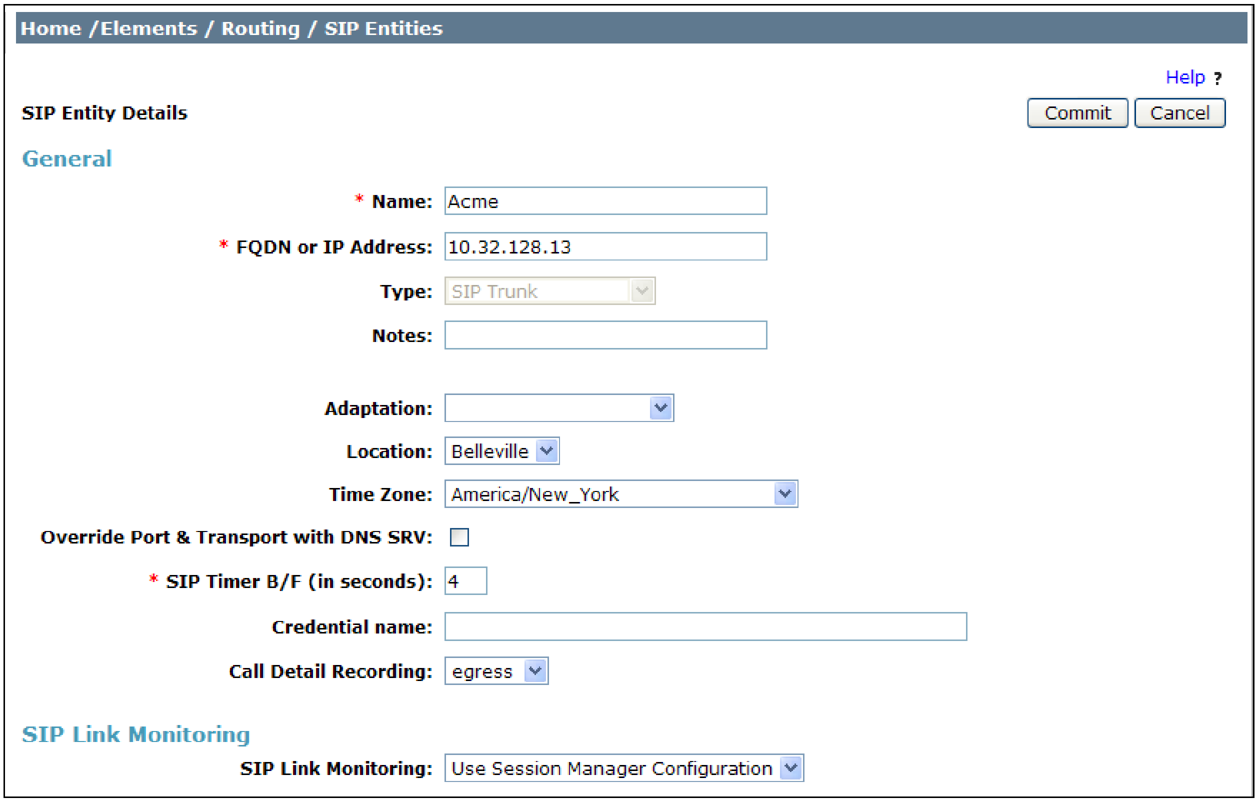
The following screen shows the addition of the Acme SBC SIP Entity. The **FQDN or IP Address** field is set to the IP address of its private network interface (see **Figure 1**). The **Location** field is set to **Belleville** which is the location defined for the subnet where the Acme SBC resides.



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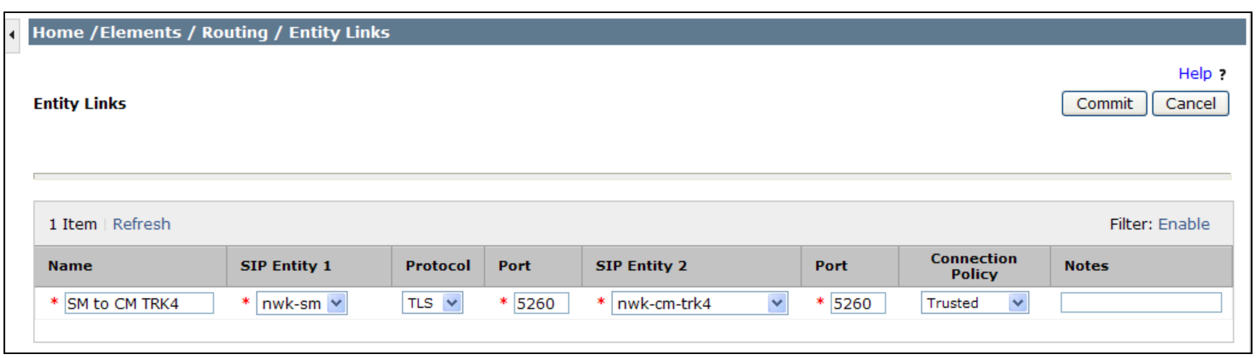
6.6. Add Entity Links

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 Acme 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 to create the Communication Manager Entity Link:

- **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 5.6**.
- **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 5.6**.
- **Connection Policy:** Select **Trusted** from pull-down menu.

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 5.6**.


Home /Elements / Routing / Entity Links

Help ?

Entity Links

Commit Cancel

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM to CM TRK4	* nwk-sm	TLS	* 5260	* nwk-cm-trk4	* 5260	Trusted	

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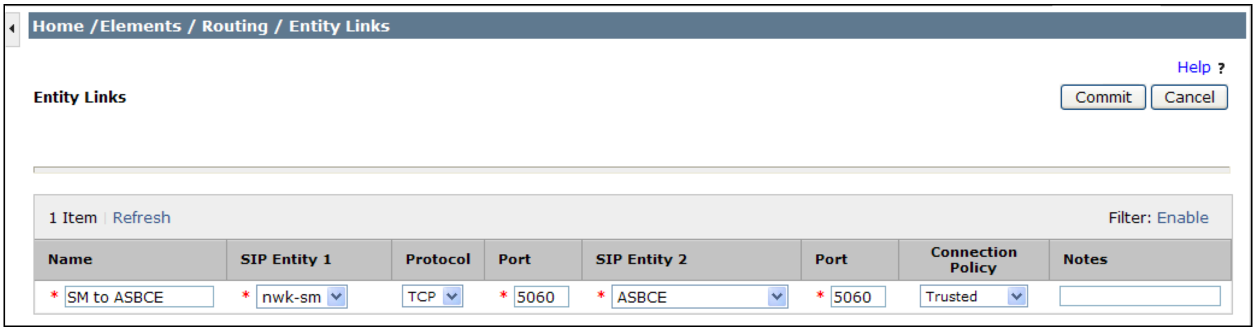
The following screen illustrates the Entity Link to the Acme SBC.



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6.7. Add Routing Policies

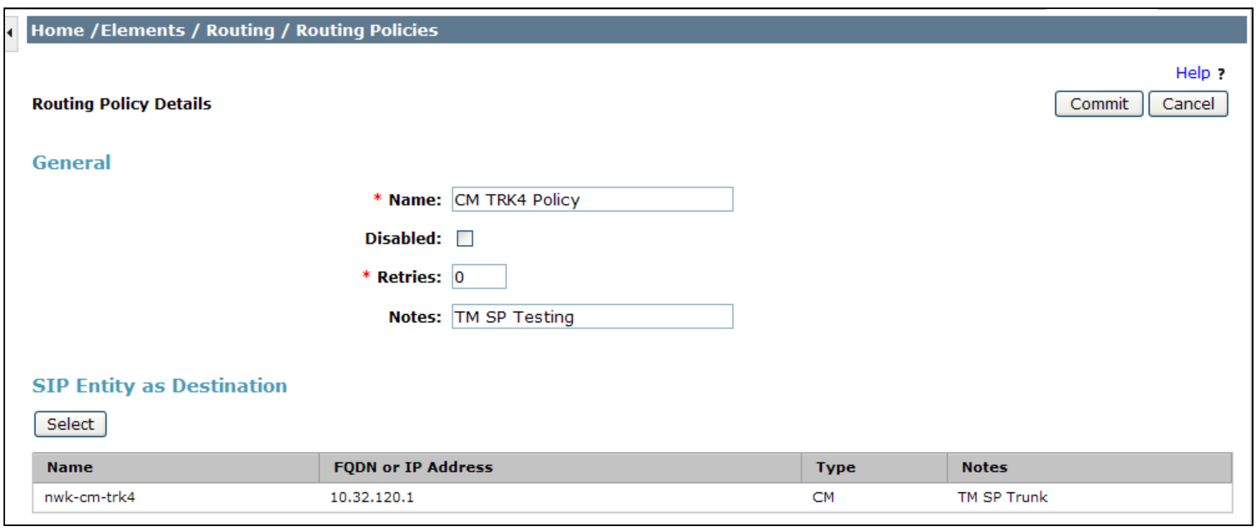
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Two routing policies must be added: one for Communication Manager and one for the Acme SBC. To add a routing policy, navigate to **Routing → Routing Policies** in the left-hand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



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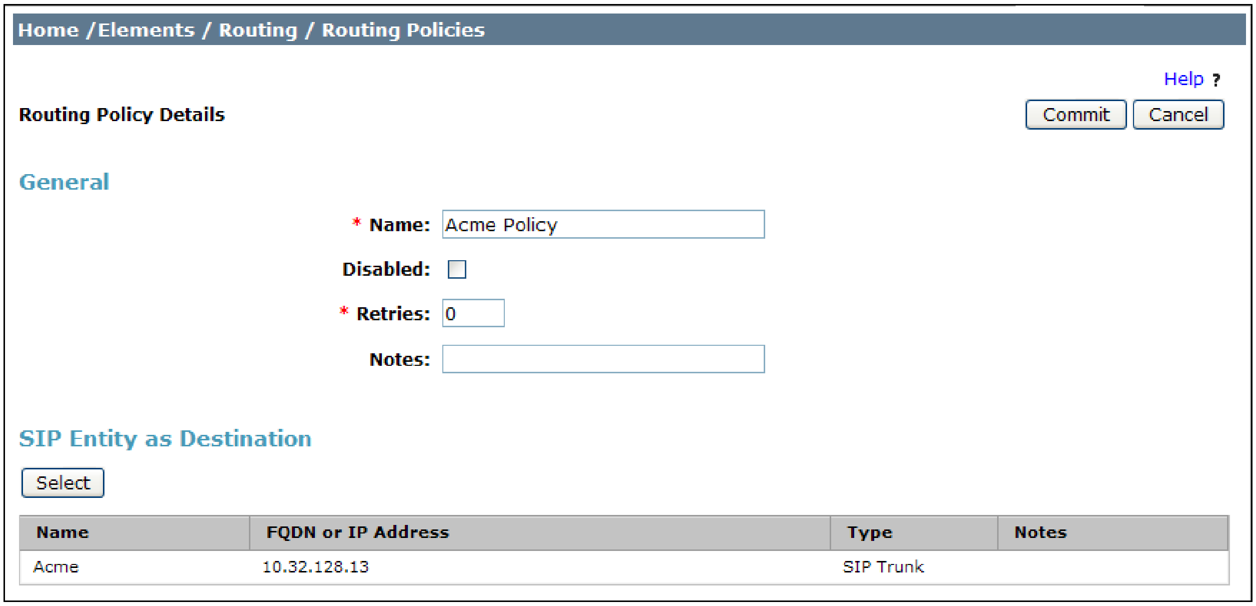

6.8. Add Dial Patterns

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 EarthLink 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.

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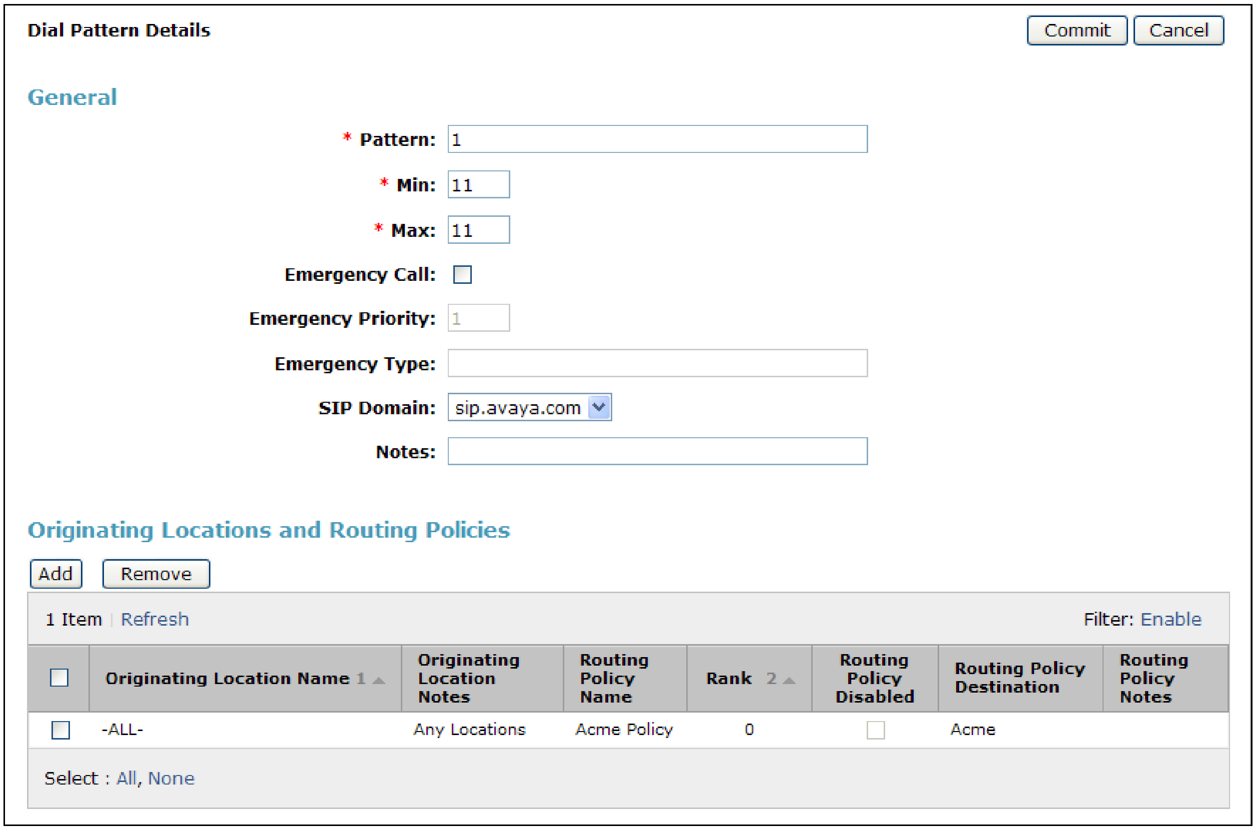
Two examples of the dial patterns used for the compliance test are shown below. The first example shows that numbers that begin with **1** and have a destination domain of **sip.avaya.com** from **ALL** locations use route policy **Acme Policy**.



The second example shows that 10 digit numbers that start with **978555** to domain **sip.avaya.com** and originating from **ALL** locations use route policy **CM TRK4 Policy**. These are the DID numbers assigned to the enterprise from EarthLink.

Dial Pattern Details Commit Cancel

General

* **Pattern:** 978555

* **Min:** 10

* **Max:** 10

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: sip.avaya.com

Notes: EarthLink DID Numbers

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	CM TRK4 Policy	0	☐	nwk-cm-trk4	TM SP Testing

Select : All, None

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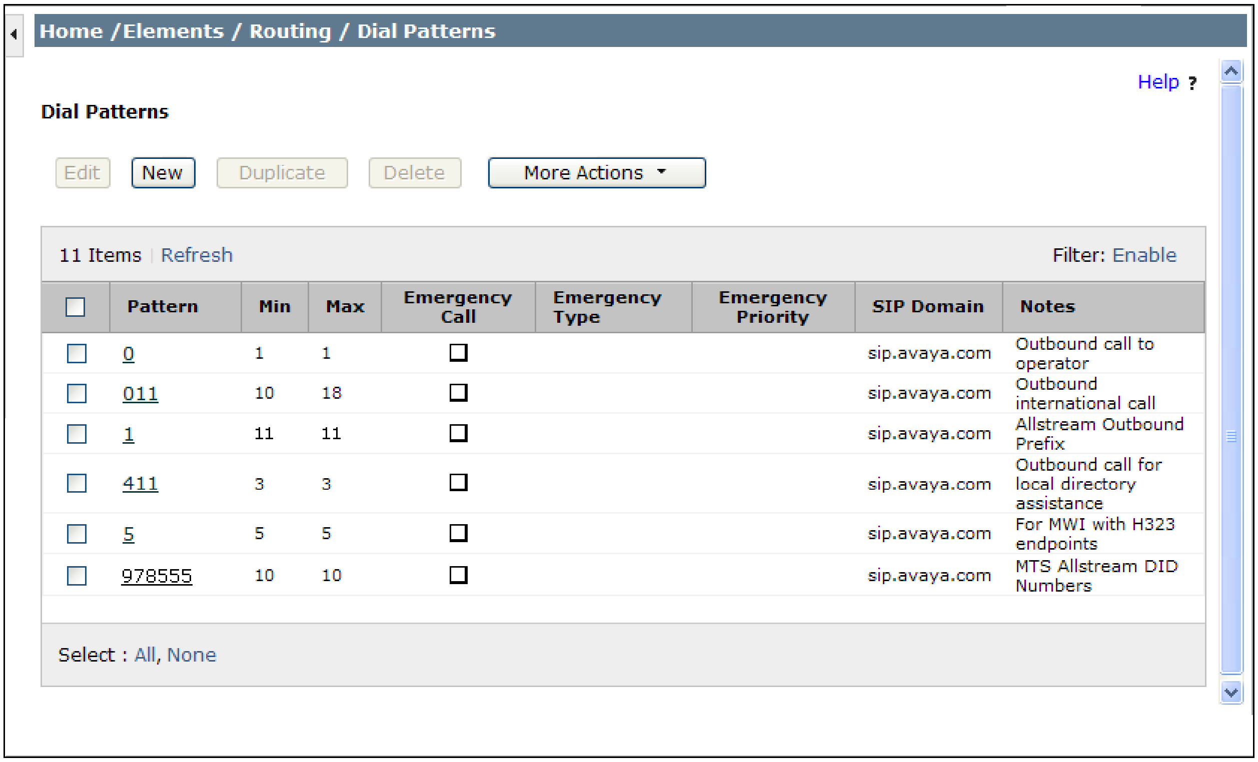
The complete list of dial patterns defined for the compliance test is shown below.



Home / Elements / Routing / Dial Patterns

Help ?

Dial Patterns

Edit New Duplicate Delete More Actions

11 Items | Refresh Filter: Enable

	Pattern	Min	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
☐	0	1	1	☐			sip.avaya.com	Outbound call to operator
☐	011	10	18	☐			sip.avaya.com	Outbound international call
☐	1	11	11	☐			sip.avaya.com	Allstream Outbound Prefix
☐	411	3	3	☐			sip.avaya.com	Outbound call for local directory assistance
☐	5	5	5	☐			sip.avaya.com	For MWI with H323 endpoints
☐	978555	10	10	☐			sip.avaya.com	MTS Allstream DID Numbers

Select : All, None

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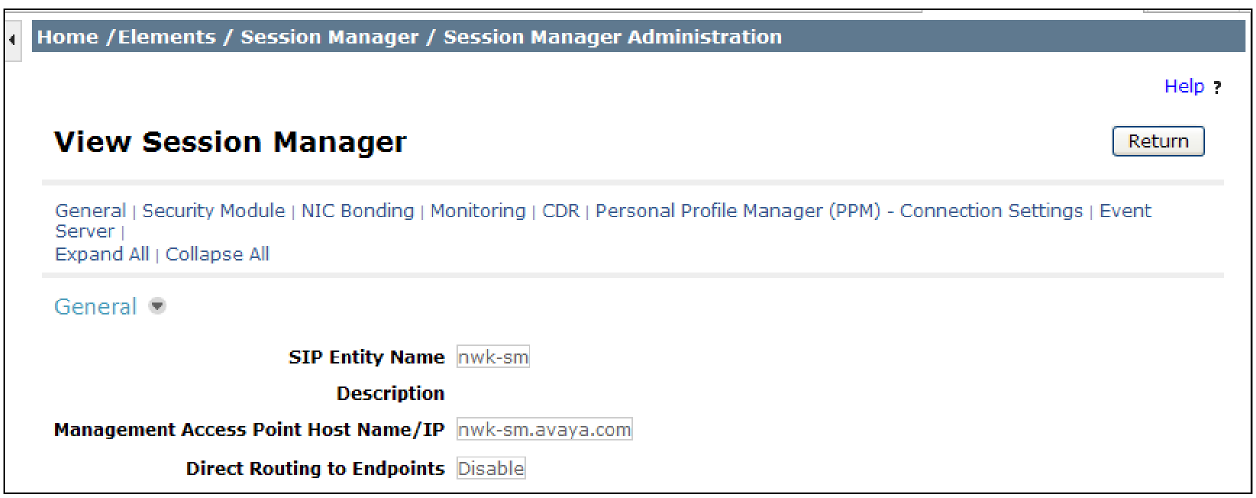
6.9. Add/View Session Manager

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

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

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

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



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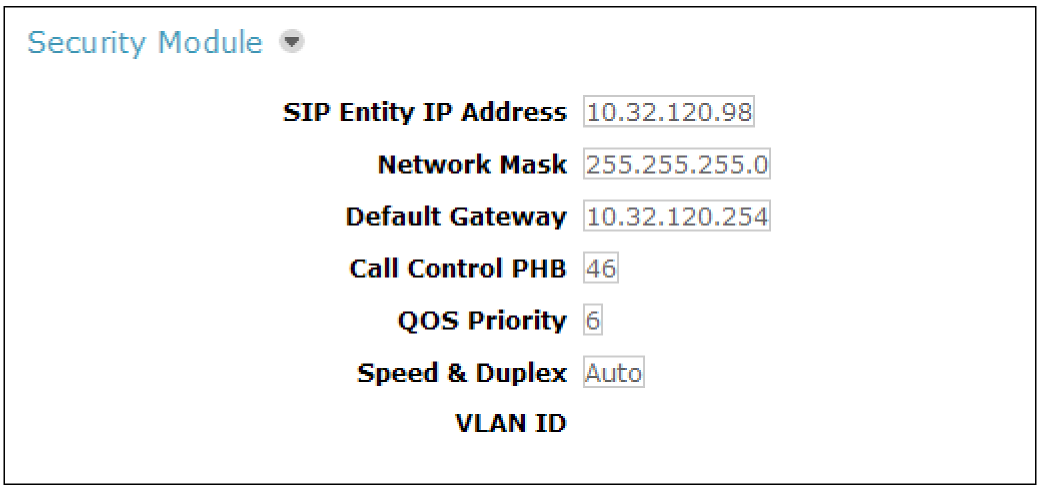
In the **Security Module** section, enter the following values:

- **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.



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7. Configure Acme Packet Net-Net 3800 Session Border Controller

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

The Acme 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 reviewed by entering the **show running-config** command. The **verify-config** command may be used to check the configuration for syntax errors.

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 SBC.

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**

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9. Enter **exit**

7.1.2. Private Interface

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

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 SBC. 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 SBC. 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 → 192.168.96.225**
4. Enter **netmask → 255.255.255.224**
5. Enter **gateway → 192.168.96.254**
6. Enter **dns-ip-primary → 192.168.96.199**
7. Enter **hip-ip-list → 192.168.96.225**
8. Enter **icmp-ip-list → 192.168.96.225**
9. Enter **done**
10. Enter **exit**

7.2.2. Private Interface

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

1. Enter **system → network-interface**
2. Enter **name → s1p0**
3. Enter **ip-address → 10.32.128.13**
4. Enter **netmask → 255.255.255.0**

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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 → 192.168.96.225**
3. Enter **start-port → 49152**
4. Enter **end-port → 65535**
5. Enter **realm-id → EXTERNAL**
6. Enter **done**
7. Enter **exit**

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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 5.5**.

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.

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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 → 192.168.96.225**
    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 → 401,403,407**
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 → 401,403,407**
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. The set of SIP header manipulation rules specified in the **out-manipulationid** parameter below are defined in **Section 7.10**.

1. Enter **session-router → session-agent**

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2. Enter **hostname → 192.168.193.167**
3. Enter **ip-address → 192.168.193.167**
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 → EarthLink**
10. Enter **ping-method →**
11. Enter **ping-interval → 0**
12. Enter **ping-send-mode → keep-alive**
13. Enter **in-manipulationid →**
14. Enter **out-manipulationid → outManToSP2**
15. Enter **done**
16. Enter **exit**

7.8.2. Inside Session-Agent

Create a session-agent for the inside network. The set of SIP header manipulation rules specified in the **in-manipulationid** and **out-manipulationid** parameters below are defined in **Section 7.10**.

1. Enter **session-router → session-agent**
2. Enter **hostname → 10.32.120.98**
3. Enter **ip-address → 10.32.120.98**
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 → NWK_SM**
10. Enter **ping-method →**
11. Enter **ping-interval → 0**
12. Enter **ping-send-mode → keep-alive**
13. Enter **in-manipulationid → inManFromSM**
14. Enter **out-manipulationid → outManToSM**
15. Enter **done**
16. 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.

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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 → 192.168.193.167**
    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 → 10.32.120.98**
    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. Three separate sets of SIP manipulations were required for the compliance test listed below. These rules are applied to a specific session agent in **Section 7.8**.

- **inManFromSM** – A set of SIP header manipulation rules (HMRs) on traffic from Session Manager to the SBC.
- **outManToSM** - A set of SIP header manipulation rules (HMRs) on traffic from the SBC to the Session Manager.

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- **outManToSP2** - A set of SIP header manipulation rules (HMRs) on traffic from the SBC to service provider (EarthLink).

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**

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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. This temporary header value is used as input to the rules defined **in Section 7.10.3.1** and **7.10.3.9**.

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 Session Manager

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

To create this set of SIP HMRs:

1. Enter **session-router → sip-manipulation**
2. Enter **name → outManFromSM**
3. Enter **description → "Outbound 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**

7.10.2.1 Change Host of Request-URI Header

This rule replaces the host part of the Request-URI header with the enterprise SIP domain.

1. Enter **header-rule**
2. Enter **name → chgRURI**
3. Enter **header-name → Request-URI**
4. Enter **action → manipulate**

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5. Enter **comparison-type → pattern-rule**
6. Enter **msg-type → request**
7. Enter **element-rule**
    a. Enter **name → chgRuriHost**
    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 → sip.avaya.com**
    g. Enter **done**
    h. Enter **exit**
8. Enter **done**
9. Enter **exit**

7.10.3. SBC to EarthLink

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

To create this set of SIP HMRs:

1. Enter **session-router → sip-manipulation**
2. Enter **name → outManFromSP2**
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**

7.10.3.1 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**

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   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**

7.10.3.2 Change Host of the Request-URI Header

This rule replaces the host part of the Request-URI header with the service provider's IP address. The Request-URI could have also been manipulated by the Session Manager.

1. Enter **header-rule**
2. Enter **name → manipRURI**
3. Enter **header-name → Request-URI**
4. Enter **action → manipulate**
5. Enter **comparison-type → pattern-rule**
6. Enter **msg-type → request**
7. Enter **element-rule**
   a. Enter **name → chgRuriHost**
   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**

7.10.3.3 Change Host of the To Header

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

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**

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    e. Enter **comparison-type → case-sensitive**
    f. Enter **new-value → $REMOTE_IP**
    g. Enter **done**
    h. Enter **exit**
8. Enter **done**
9. Enter **exit**

7.10.3.4 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**

7.10.3.5 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**

7.10.3.6 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**

7.10.3.7 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**

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9. Enter **exit**

7.10.3.8 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**

7.10.3.9 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**

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10. Enter **exit**

7.10.3.10 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**

7.10.3.11 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**

7.10.3.12 Delete X-Contact Header

This rule deletes the temporary X-Contact header created in **Section 7.10.1.2** 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**

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7.10.3.13 Delete Endpoint-View Header

This rule deletes the Endpoint-View 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 → delEdptView**
3. Enter **header-name → Endpoint-View**
4. Enter **action → delete**
5. Enter **comparison-type → case-sensitive**
6. Enter **msg-type → any**
7. Enter **methods →**
8. Enter **done**
9. Enter **exit**

8. Configure 9600 Series IP Telephones

For the compliance test, the DTMF payload header value for 9600 Series IP Telephones was set to 101 by adding the command **SET DTMF_PAYLOAD_TYPE=101** in the phone 46xxsettings.txt configuration file. Only the 9600 and 1600 SIP Telephones use this setting. The value of 101 is the value used by EarthLink. The purpose of this configuration was to avoid a situation where a call between EarthLink and the SIP phone could be established with a DTMF payload header value that is different in each direction of the call.

9. EarthLink Complete SIP Trunking Configuration

EarthLink is responsible for the network configuration of the EarthLink Complete SIP Trunking service. EarthLink will require that the customer provide the public IP address used to reach the Acme SBC at the edge of the enterprise. EarthLink will provide the IP address of the EarthLink 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 Acme SBC configuration discussed in the previous sections.

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

10. 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:

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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> - Traces 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.

11. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and the Acme Packet Net-Net 3800 Session Border Controller to EarthLink Complete SIP Trunking. EarthLink Complete SIP Trunking passed compliance testing. Please refer to **Section 2.2** for any exceptions or workarounds.

12. 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 6.2.1, July 2012.
[2] *Administering Avaya Aura® System Platform*, Release 6.2.1, July 2012.
[3] *Administering Avaya Aura® Communication Manager*, Issue 7.0, July 2012, Document Number 03-300509.
[4] *Avaya Aura® Communication Manager Feature Description and Implementation*, Issue 9.0, July 2012, Document Number 555-245-205.

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[5] *Upgrading Avaya Aura® System Manager to 6.2*, Release 6.2, July 2012.
[6] *Administering Avaya Aura® System Manager*, Release 6.2, July 2012.
[7] *Installing and Configuring Avaya Aura® Session Manager*, Release 6.1, April 2011, Document Number 03-603473.
[8] *Administering Avaya Aura® Session Manager*, Release 6.2, July 2012, Document Number 03-603324.
[9] *Avaya 1600 Series IP Deskphones Administrator Guide Release 1.3.x*, April 2010, Document Number 16-601443.
[10] *Avaya one-X® Deskphone Edition for 9600 Series IP Telephones Administrator Guide*, Issue 8, March 2012, Document Number 16-300698.
[11] *Avaya one-X® Deskphone Edition SIP for 9600 Series IP Telephones Administrator Guide*, Release 2.6, June 2010, Document Number 16-601944.
[12] *Avaya one-X® Deskphone SIP 9608, 9611G, 9621G, 9641G Administrator Guide*, Release 6.0.1, May 2011, Document Number 16-603813.
[13] *Administering Avaya one-X® Communicator*, October 2011.
[14] *Implementing and Administering the Avaya A175 Desktop Video Device with the Avaya Flare® Experience*, Release 1.1, March 2012, Document Number 16-603739.
[15] RFC 3261 *SIP: Session Initiation Protocol*, http://www.ietf.org/
[16] RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, http://www.ietf.org/

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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.

Appendix A: Acme Packet 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            2012-07-25 15:53:58
      policy-attribute
            next-hop                      192.168.193.167
            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
```


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```
                              *
      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      2012-07-25 15:56:25
      policy-attribute
            next-hop                10.32.120.98
            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
```


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```
      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                     192.168.96.225
      pri-utility-addr
      sec-utility-addr
      netmask                        255.255.255.224
      gateway                        192.168.96.254
      sec-gateway
      gw-heartbeat
            state                          disabled
            heartbeat                      0
            retry-count                    0
            retry-timeout                  1
            health-score                   0
      dns-ip-primary                 192.168.96.199
      dns-ip-backup1
      dns-ip-backup2
      dns-domain
      dns-timeout                    11
       hip-ip-list                    192.168.96.225
      ftp-address
       icmp-address                   192.168.96.225
      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
```


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```
        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
```


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```
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
```


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```
session-agent
        hostname                        192.168.193.167
        ip-address                      192.168.193.167
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                UDP
        realm-id                        EXTERNAL
        egress-realm-id
        description                     EarthLink
        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
        out-manipulationid              outManToSP2
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
        rfc2833-mode                    none
```


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```
        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            2012-07-25 19:34:36
session-agent
        hostname                      10.32.120.98
        ip-address                    10.32.120.98
        port                          5060
        state                         enabled
        app-protocol                  SIP
        app-type
        transport-method              StaticTCP
        realm-id                      INTERNAL2
        egress-realm-id
        description                   NWK_SM
        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             outManToSM
        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             2012-08-07 18:19:39
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
```


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```
        refer-src-routing              disabled
        add-ucid-header                disabled
        proxy-sub-events
        pass-gruu-contact              disabled
        sag-lookup-on-redirect         disabled
        last-modified-by               admin@192.168.168.37
        last-modified-date             2012-02-16 13:46:26
sip-interface
        state                          enabled
        realm-id                       EXTERNAL
        description
        sip-port
                address                        192.168.96.225
                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                   401,403,407
        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
```

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```
        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             2012-07-26 09:35:27
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                   401,403,407
        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
```


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```
        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             2012-07-26 09:34:39
sip-manipulation
        name                           inManFromSM
        description                    Inbound SIP HMRs From SM
        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
```


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```
                match-val-type                  any
                comparison-type                 pattern-rule
                match-value
                new-value                       $strcon.$strval.$0
sip-manipulation
        name                            outManToSP2
        description                     Outbound SIP HMRs To SP
        split-headers
        join-headers
        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                            manipRURI
                header-name                     Request-URI
                action                          manipulate
                comparison-type                 pattern-rule
                msg-type                        request
                methods
                match-value
                new-value
                element-rule
                        name                            chgRuriHost
                        parameter-name
                        type                            uri-host
                        action                          replace
                        match-val-type                  any
                        comparison-type                 case-sensitive
                        match-value
                        new-value                       $REMOTE_IP
        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
```


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```
            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
```


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```
        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                          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                          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                          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
```


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```
            methods
            match-value
            new-value
      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                          delEdptView
            header-name                   Endpoint-View
            action                        delete
            comparison-type               case-sensitive
            msg-type                      any
            methods
            match-value
            new-value
      last-modified-by              admin@192.168.168.37
      last-modified-date            2012-08-02 15:11:14
sip-manipulation
      name                          outManToSM
      description                   Outbound SIP HMRs To SM
      split-headers
      join-headers
      header-rule
            name                          chgRURI
            header-name                   Request-URI
            action                        manipulate
            comparison-type               pattern-rule
            msg-type                      request
            methods
            match-value
            new-value
            element-rule
                  name                          chgRuriHost
                  parameter-name
                  type                          uri-host
                  action                        replace
                  match-val-type                any
                  comparison-type               case-sensitive
                  match-value
                  new-value                     sip.avaya.com
steering-pool
      ip-address                    192.168.96.225
      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
```


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```
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                192.168.96.254
      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
```


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