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Avaya Solution & Interoperability Test Lab

Application Notes for Configuring IntelePeer™ SIP Trunk Service with Avaya Aura® Communication Manager Evolution Server 6.0.1 and Avaya Session Border Controller for Enterprise 4.0.5 – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between the IntelePeer™ SIP Trunk Service and an Avaya SIP-enabled enterprise solution. IntelePeer™ can offer the SIP Trunk Service using several different platform technologies in the IntelePeer™ network. These Application Notes correspond to the SIP Trunk Service offered using a Sonus GSX9000 SBC in the network. The Avaya solution consists of Avaya Session Border Controller for Enterprise, Avaya Aura® Communication Manager Evolution Server, and various Avaya endpoints.

IntelePeer™ 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 the IntelePeer™ SIP Trunk Service and an Avaya SIP-enabled enterprise solution. IntelePeer™ can offer the SIP Trunk Service using several different platform technologies in the IntelePeer™ network. These Application Notes correspond to the SIP Trunk Service offered using a Sonus GSX9000 SBC in the network. The Avaya solution consists of Avaya Session Border Controller for Enterprise, Avaya Aura® Communication Manager Evolution Server and various Avaya endpoints.

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

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Communication Manager and the Avaya Session Border Controller for Enterprise to connect to the public Internet using a broadband connection. The enterprise site was configured to connect to the SIP Trunk Service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising standards-based interfaces pertinent to the interoperability of the tested products and their functionality. 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.

The SIP Trunk Service passed compliance testing.

2.1. Interoperability Compliance Testing

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

- Incoming PSTN calls to various phone types. Phone types included H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator (soft client).

- Avaya one-X® Communicator supports two modes (Road Warrior and Telecommuter). Each supported mode was tested. Avaya one-X Communicator also supports two Voice over IP (VoIP) protocols: H.323 and SIP. The H.323 protocol was the only protocol tested.
- Various call types including: local, long distance, international, outbound toll-free.
- Codec G.711MU
- DTMF transmission using RFC 2833.
- G.711 Faxing.
- Caller ID presentation and Caller ID restriction.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and mobility (extension to cellular).

Items not supported or not tested included the following:
- Inbound toll-free, 411 and emergency calls (911) are supported but were not tested as part of the compliance test.

2.2. Test Results

Interoperability testing of the SIP Trunk Service was completed with successful results for all test cases.

2.3. Support

For technical support with the SIP Trunk Service, contact IntelePeer™ using the Customer Care links at **www.IntelePeer™.com**

3. Reference Configuration

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

The Avaya components used to create the simulated customer site included:
- Communication Manager
- Communication Manager Messaging
- Avaya Session Border Controller for Enterprise
- Avaya G450 Media Gateway
- Avaya 9600-Series IP telephones (H.323)
- Avaya one-X® Communicator (H.323)
- Avaya digital and analog telephones

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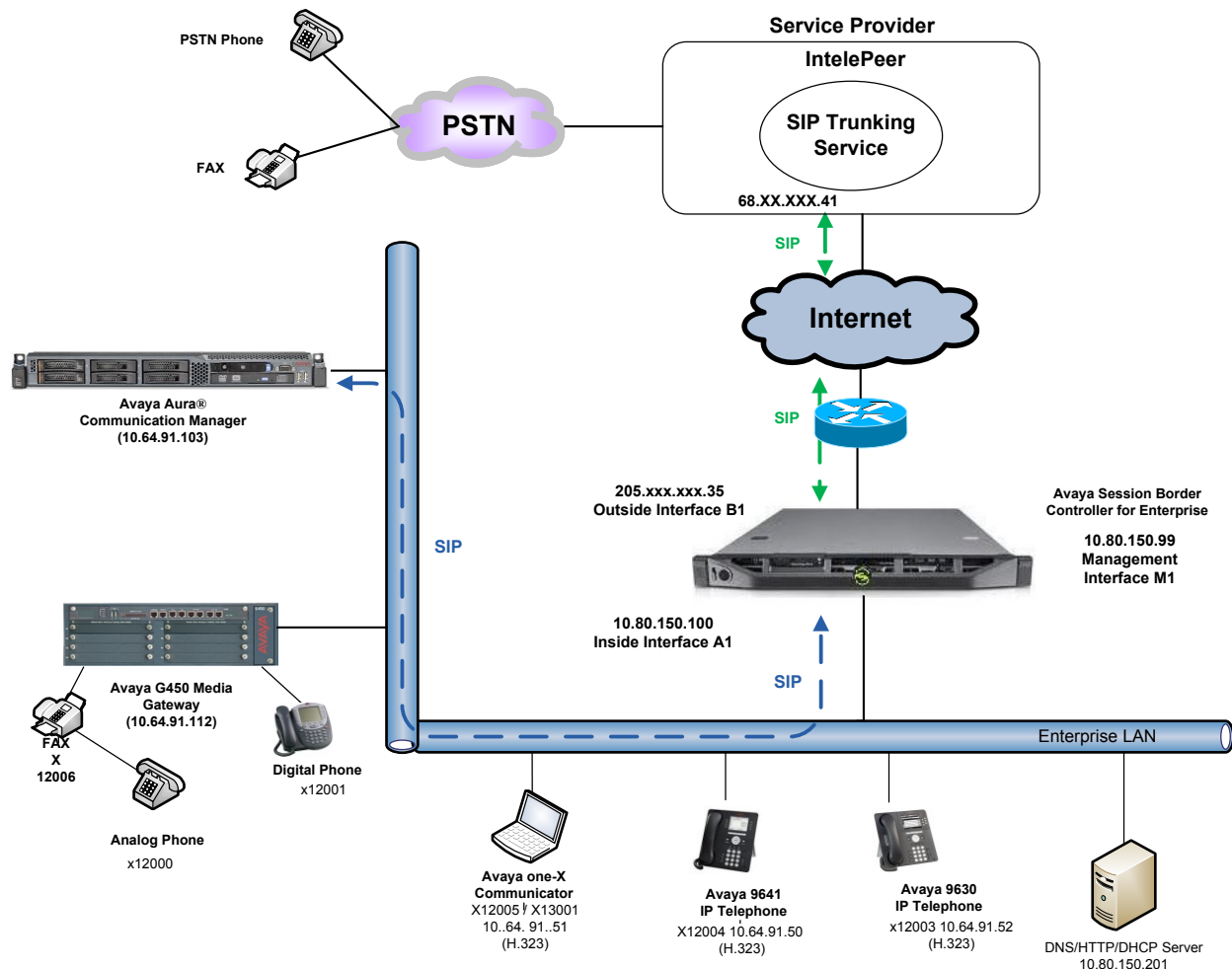

Figure 1: Avaya IP Telephony Network using the SIP Trunk Service

For inbound calls, the calls flow from the service provider to the Avaya SBCE and then to Communication Manager. Once the call arrives at Communication Manager, 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

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restrictions. Once Communication Manager selects the proper SIP trunk, the call is routed to Avaya SBCE. From Avaya SBCE, the call is sent to the SIP Trunk Service.

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

4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Component	Release
Avaya Aura® Communication Manager	R016x.00.1.510.1-19350 (SP 6)
Avaya Aura® Communication Manager Messaging	N6.0.1-8.0
Avaya Session Border Controller for Enterprise	4.0.5.Q02
Avaya G450 Media Gateway	31.20.1
Avaya 9630 IP Telephone (H.323)	Avaya one-X® Deskphone Edition 3.103S
Avaya 9641 IP Telephone (H.323)	Avaya one-X® Deskphone SIP Edition 6.0.3
Avaya one-X® Communicator (H.323)	6.1.2.06 SP2
Avaya 2420 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
IntelePeer™ SIP Trunking Solution Components	
Component	Release
Sonus GSX9000 SBC	V07.03.04S003

Table 1: Equipment and Software Tested

The specific configuration above was used for the compliance testing.

Note: This solution will be compatible with other Avaya Server and Media Gateway platforms running similar versions of Communication Manager.

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager for the SIP Trunk Service. A SIP trunk is established between Communication Manager and the Avaya SBCE for use by signaling traffic to and from IntelePeer™. It is assumed the general installation of Communication Manager and the Avaya G450 Media Gateway 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: IP addresses and phone numbers shown throughout these Application Notes have been edited so that the actual IP addresses of the network elements and public PSTN numbers are not revealed.

5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to the service provider. The example below shows that **12000** licenses are available and **280** 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 2
               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 1
                       Maximum Administered SIP Trunks: 12000 280
     Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
     Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 10    0
                   Maximum Media Gateway VAL Sources: 250   1
            Maximum TN2602 Boards with 80 VoIP Channels: 128   0
            Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0


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

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

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

```
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: 1
            International Access Code: 011

ENBLOC DIALING PARAMETERS
   Enable Enbloc Dialing without ARS FAC? n

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

5.3. IP Node Names

Use the **change node-names ip** command to verify the node name defined for the IP address of Communication Manager **(procr)** created during installation. Add a node name and IP address for the Avaya SBCE's internal interface (e.g., **ASBCE**). These node names will be needed for defining the service provider signaling group in **Section 5.7**.

```
change node-names ip                                          Page   1 of   2
                              IP NODE NAMES
Name              IP Address
ASBCE             10.80.150.100
Messaging         10.64.91.103
SM                10.64.91.109
acme              10.64.91.115
default           0.0.0.0
procr             10.64.91.103
```

5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, **ip-codec-set 2** was used for this purpose. The SIP Trunk Service supports G.729A, and G.711MU. During compliance testing only G.711MU was tested. The true order of preference is defined by the end customer. In the example below, **G.711MU** and **G.729A** was entered in the **Audio Codec** column of the table. Default values can be used for all other fields.

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

IP Codec Set

    Codec Set: 2

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

Set the **Fax Mode** to **t.38-standard** on **Page 2**.

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

                            IP Codec Set

                        Allow Direct-IP Multimedia? n



                   Mode              Redundancy
          FAX      t.38-standard         0
          Modem    off                   0
          TDD/TTY  US                    3
```

5.5. IP Interface for procr

The **add ip-interface procr** or **change ip-interface procr** command can be used to configure the Processor Ethernet (PE) parameters. The following screen shows the parameters used in the sample configuration. While the focus here is the use of the Processor Ethernet (Procr) for SIP Trunk Signaling, observe that the Processor Ethernet will also be used for registrations from H.323 IP Telephones and H.248 gateways in the sample configuration.

```
change ip-interface proc                                        Page   1 of   2
                            IP INTERFACES


                   Type: PROCR
                                                    Target socket load: 1700

          Enable Interface? y                    Allow H.323 Endpoints? y
                                                   Allow H.248 Gateways? y
          Network Region: 1                       Gatekeeper Priority: 5


                            IPV4 PARAMETERS
          Node Name: procr                     IP Address: 10.64.91.103


          Subnet Mask: /24
```

5.6. IP Network Region

Create a separate IP network region for the service provider trunk. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the service provider versus calls within the enterprise or elsewhere. For the compliance test, **IP-network-region 2** was chosen for the service provider trunk. **IP network region 1** is the default IP network region and encompasses all the elements within the enterprise. Use the **change ip-network-region 2** command to configure IP network region 2 with the following parameters:

- Set the **Location** field to match the enterprise location for this SIP trunk.

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is **avayalab.com**. This name appears in the "From" header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. To enable shuffling, set both **Intra-region** and **Inter-region IP-IP Direct Audio** fields to **yes.** This is the default setting. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 2                                        Page   1 of  20
                                IP NETWORK REGION
  Region: 2
Location: 1                      Authoritative Domain: avayalab.com
    Name: SIP Trunks
MEDIA PARAMETERS                      Intra-region IP-IP Direct Audio: yes
      Codec Set: 2                    Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 5000                             IP Audio Hairpinning? n
  UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
       Audio PHB Value: 46
       Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
       Audio 802.1p Priority: 6
       Video 802.1p Priority: 5     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
           Keep-Alive Count: 5
```

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

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

 Source Region: 2     Inter Network Region Connection Management      I       M
                                                                 G  A    t
 dst codec direct   WAN-BW-limits   Video        Intervening    Dyn A  G    c
 rgn set   WAN  Units    Total Norm  Prio Shr Regions           CAC R  L    e
 1   2     y    NoLimit                                             n       t
 2   2
 3
 4
```

5.7. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and the Avaya SBCE 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 2** was used for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- Set the **IMS Enabled** field to **n**. This specifies Communication Manager will serve as an Evolution Server.
- Set the **Transport Method** to **tcp** (Transmission Control Protocol). Set the **Near-end Listen Port** and **Far-end Listen Port** to **5060**.
- Set the **Peer Detection Enabled** field to **n**.
- Set the **Peer Server** to **Others**.
- 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 **ASBCE**. This node name maps to the IP address of the Avaya SBCE's internal interface as defined in **Section 5.3**.
- Set the **Far-end Network Region** to the IP network region defined for the service provider in **Section 5.6**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set **Direct IP-IP Audio Connections** to **y**. This field will enable media shuffling on the SIP trunk.
- Set the **DTMF over IP** field to **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Default values may be used for all other fields.

```
add signaling-group 2                                          Page   1 of   1
                              SIGNALING GROUP

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


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

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

5.8. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.7**. For the compliance test, **Trunk Group 2** 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 **Direction** to **two-way**.
- 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 2                                           Page    1 of  21
                              TRUNK GROUP

Group Number: 2                      Group Type: sip          CDR Reports: y
  Group Name: SIP SP 2                         COR: 1      TN: 1     TAC: *02
   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: 2
                                                  Number of Members: 10
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. For the compliance test, the value of **600** seconds was used.

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

TRUNK PARAMETERS

     Unicode Name: auto

                                          Redirect On OPTIM Failure: 5000

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

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

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

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to **y**. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.2**, if the inbound call enabled CPN block. For outbound calls, these same settings request that CPN block be activated on the far-end destination if a local user requests CPN block on a particular call routed out this trunk. Default values were used for all other fields.

```
add trunk-group 2                                            Page   3 of  21
TRUNK FEATURES
         ACA Assignment? n            Measured: none
                              Maintenance Tests? y


                     Numbering Format: public
                                          UUI Treatment: service-provider

                                          Replace Restricted Numbers? y
                                          Replace Unavailable Numbers? y


                              Modify Tandem Calling Number: no


  Show ANSWERED BY on Display? y
```

On **Page 4**, set the **Network Call Redirection** field to **y** if using the SIP REFER method. This allows inbound calls transferred back to the PSTN to use the SIP REFER method, see **Reference [15]**. Set the **Send Diversion Header** field to **y** as well. This field provides additional information to the network if the call has been re-directed. This is necessary to support call forwarding of inbound calls back to the PSTN and some Extension to Cellular (EC500) call scenarios. Set the **Support Request History** field to **n**. Set the **Telephone Event Payload Type** to **101**, the value preferred by IntelePeer™.

Note: IntelePeer™'s SIP Trunking Feature **does not require** the use of the Diversion header on re-directed calls. When using IntelePeer™'s SIP Trunking Feature, set the **Send Diversion Header** and **Network Call Redirection** field to **n** but the **Support Request History** field to **y.**

```
add trunk-group 2                                          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? n
                 Support Request History? y
            Telephone Event Payload Type: 101



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

5.9. Inbound Routing

In general, the incoming call handling treatment for a trunk group can be used to manipulate the digits received for an incoming call if necessary. The DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. Use the **change inc-call-handling-trmt trunk-group 2** command to create an entry for each DID. As an example, the following screen illustrates the conversion of DID number **3033289130** to extension **12001** and so forth.

```
change inc-call-handling-trmt trunk-group 2                Page    1 of  30
         INCOMING CALL HANDLING TREATMENT

Service/        Number    Number       Del Insert
 Feature        Len       Digits
public-ntwrk    10 3033289130          10  12001
public-ntwrk    10 3033289131          10  12002
public-ntwrk    10 3033289132          10  12003
public-ntwrk    10 3033289133          10  12004
public-ntwrk    10 3033289134          10  12005
```

5.10. Calling Party Information

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

In the sample configuration, five DID numbers were assigned for testing. These five numbers were assigned to the five extensions **12001**, **12002**, **12003**, **12004** and **12005**. 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 five extensions at the Enterprise.

```
change public-unknown-numbering 1                            Page   1 of   2
                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext            Trk     CPN             CPN
Len Code           Grp(s)  Prefix          Len
                                                     Total Administered: 13
                           5               5            Maximum Entries: 9999
  5  5                                     5
  5  3                                     5         Note: If an entry applies to
  5  4                                     5         a SIP connection to Avaya
  5  5                                     5         Aura(tm) Session Manager,
  5  6                                     5         the resulting number must
  5  7                                     5         be a complete E.164 number.
  5  8                                     5
  5  12000         2       3033289130      10
  5  12001         2       3032899131      10
  5  12003         2       3033289132      10
  5  12004         2       3033289133      10
  5  12005         2       3033289134      10
```

5.11. Outbound Routing

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

```
change dialplan analysis                                          Page   1 of  12
                            DIAL PLAN ANALYSIS TABLE
                              Location: all         Percent Full: 2

   Dialed    Total  Call     Dialed   Total  Call     Dialed   Total  Call
   String   Length  Type     String  Length  Type     String  Length  Type
   0          1     attd
   1          5     ext
   2          5     ext
   3          5     ext
   4          5     ext
   5          5     ext
   6          5     ext
   7          5     ext
   8          5     ext
   9          1     fac
   *          3     dac
   #          3     dac
```

Use the **change feature-access-codes** command to configure **9** as the **Auto Route Selection (ARS) – Access Code 1**.

```
change feature-access-codes                                       Page   1 of  10
                         FEATURE ACCESS CODE (FAC)
          Abbreviated Dialing List1 Access Code: *10
          Abbreviated Dialing List2 Access Code: *12
          Abbreviated Dialing List3 Access Code: *13
 Abbreviated Dial - Prgm Group List Access Code: *14
                   Announcement Access Code: *19
                 Answer Back Access Code:


     Auto Alternate Routing (AAR) Access Code: *00
     Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
               Automatic Callback Activation: *33      Deactivation: #33
 Call Forwarding Activation Busy/DA: *30     All: *31   Deactivation: #30
   Call Forwarding Enhanced Status:        Act:         Deactivation:
                     Call Park Access Code: *40
                   Call Pickup Access Code: *41
 CAS Remote Hold/Answer Hold-Unhold Access Code: *42
               CDR Account Code Access Code:
                    Change COR Access Code:
               Change Coverage Access Code:
         Conditional Call Extend Activation:        Deactivation:
               Contact Closure   Open Code: *80     Close Code: #80
```

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9.

- **Dialed String:** enter the leading digits (e.g., **1303**) necessary to uniquely select the desired route pattern.
- **Total Min:** enter the minimum number of digits (e.g., **11**) expected for this PSTN number.
- **Total Max:** enter the maximum number of digits (e.g., **11**) expected for this PSTN number.
- **Route Pattern:** enter the route pattern number (e.g., **1**) to be used. The route pattern (to be defined next) will specify the trunk group(s) to be used for calls matching the dialed number.
- **Call Type:** enter **fnpa**, the call type for North American 1+10 digit calls. For local 7 or 10 digit calls enter **hnpa**. For 411 and 911 calls use **svcl** and **emer** respectively. The call type tells Communication Manager what kind of call is made to help decide how to handle the dialed string and whether or not to include a preceding 1. For more information and a complete list of Communication Manager call types, see **Reference [3]** and **[4].**

The example below shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to route pattern 1 which contains the SIP trunk to the service provider (as defined next).

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

           Dialed         Total      Route    Call   Node  ANI
           String         Min  Max   Pattern  Type   Num   Reqd
     1303                 11   11    1         fnpa         n
     1502                 11   11    1         fnpa         n
     1720                 11   11    1         fnpa         n
     1800                 11   11    1         fnpa         n
     1866                 11   11    1         fnpa         n
     1877                 11   11    1         fnpa         n
     1888                 11   11    1         fnpa         n
     1908                 11   11    1         fnpa         n
     2                    10   10    1         hnpa         n
     3                    10   10    1         hnpa         n
     4                    10   10    1         hnpa         n
     411                  3    3     1         svcl         n
     5                    10   10    1         hnpa         n
     555                  7    7     deny      hnpa         n
     6                    10   10    1         hnpa         n
```

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the service provider trunk route pattern in the following manner. The example below shows the values used for route pattern 1 during the compliance test.

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the SIP service provider. For the compliance test, trunk group 2 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**: The prefix mark (**Pfx Mrk**) of **1** will prefix any FNPA 10-digit number with a 1 and leave numbers of any other length unchanged. This will ensure 1 + 10 digits are sent to the service provider for long distance North American Numbering Plan (NANP) numbers. All HNPA 10 digit numbers are left unchanged.

```
change route-pattern 1                                           Page   1 of   3
                       Pattern Number: 1   Pattern Name: SIP TRK
                              SCCAN? n     Secure SIP? n

 Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
   No        Mrk Lmt List Del  Digits                              QSIG
                         Dgts                                      Intw
 1: 2    0    1                                                     n   user
 2:                                                                 n   user
 3:                                                                 n   user
 4:                                                                 n   user
 5:                                                                 n   user
 6:                                                                 n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                   Dgts Format
                                                            Subaddress
 1: y y y y y n  n            rest                                      none
 2: y y y y y n  n            rest                                      none
 3: y y y y y n  n            rest                                      none
 4: y y y y y n  n            rest                                      none
```

Use the **change ars digit-conversion** command to manipulate the routing of dialed digits that match the DIDs to prevent these calls from going out the PSTN and using unnecessary SIP trunk resources. The example below shows the DID numbers assigned by IntelePeer™ being converted to 5 digit extensions.

```
change ars digit-conversion 0                              Page   1 of   2
                      ARS DIGIT CONVERSION TABLE
                            Location: all             Percent Full: 0

  Matching Pattern      Min   Max   Del   Replacement String   Net  Conv ANI Req

    3033289130          10    10    10    12001                ext   y       n
    3033289131          10    10    10    12002                ext   y       n
    3033289132          10    10    10    12003                ext   y       n
    3033289133          10    10    10    12004                ext   y       n
    3033289134          10    10    10    12005                ext   y       n
```

5.12. Saving Communication Manager Configuration Changes

The command **save translation all** can be used to save the configuration.

```
save translation all

                              SAVE TRANSLATION

        Command Completion Status                          Error Code

        Success                                            0
```

6. Configure Avaya Session Border Controller for Enterprise

This section covers the configuration of Avaya Session Border Controller for Enterprise (Avaya SBCE). It is assumed that the software has already been installed. For additional information on these configuration tasks, see **Reference [12]** and [**13**].

A pictorial view of this configuration is shown below. It shows the components needed for the compliance test. Each of these components is defined in the Avaya SBCE web configuration as described in the following sections.
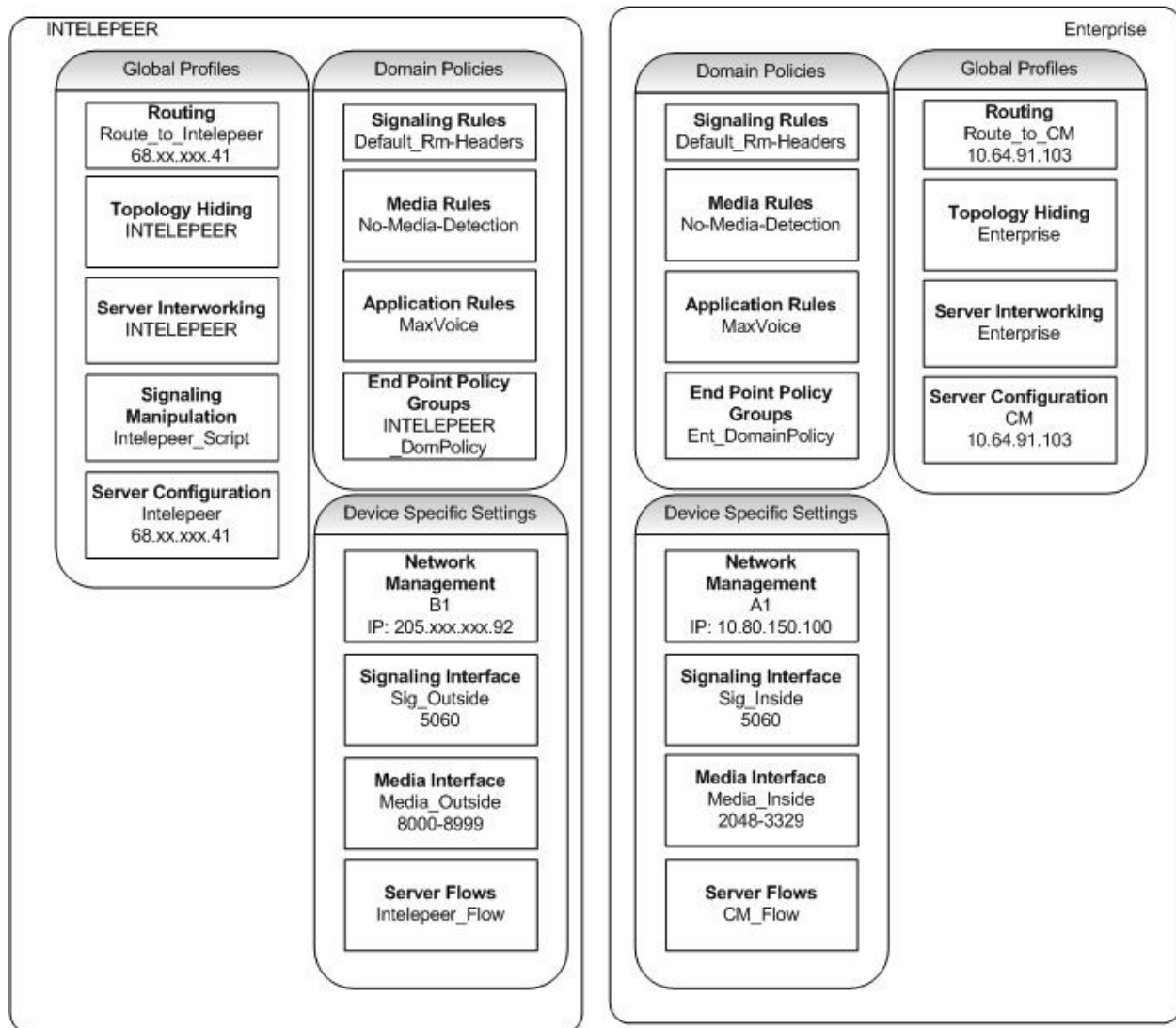
Use a web browser to access the Avaya SBCE UC-Sec web interface, enter https://<ip-addr>/ucsec in the address field of the web browser, where <ip-addr> is the management LAN IP address of UC-Sec.

Log in with the appropriate credentials. Click **Sign In**.

The main page of the Avaya SBCE UC-Sec Control Center will appear.

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To view system information that was configured during installation, navigate to **UC-Sec Control Center → System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named ASBCE is shown. To view the configuration of this device, click the monitor icon (the third icon from the right).
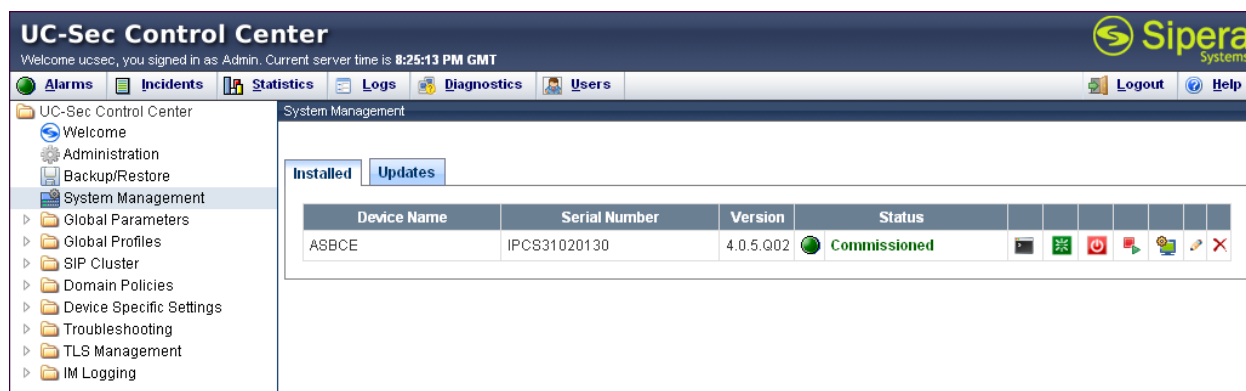
The **System Information** screen shows the **Network Settings, DNS Configuration** and **Management IP** information provided during installation, and corresponds to **Figure 1**. The **Box Type** was set to **SIP** and the **Deployment Mode** was set to **Proxy**. Default values were used for all other fields.
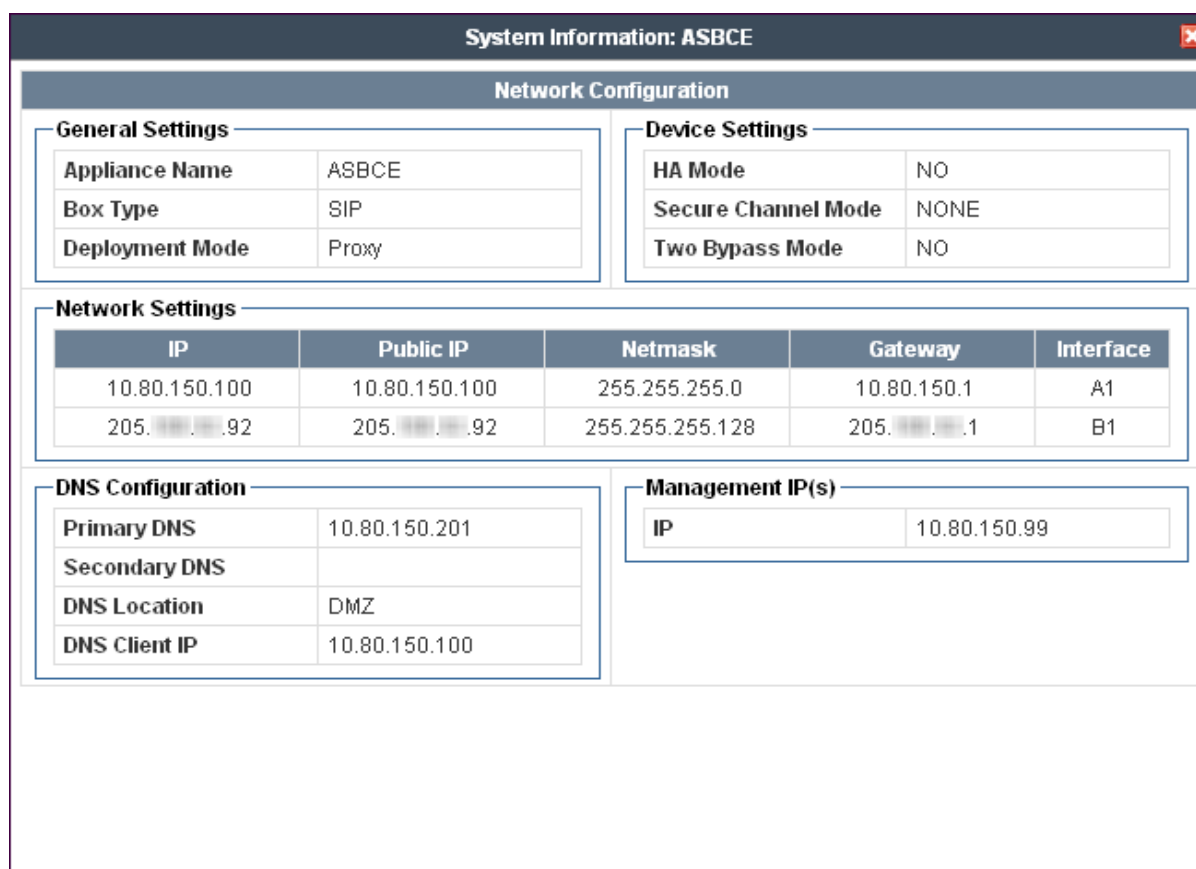
6.1. Global Profiles

Global Profiles allow for configuration of parameters across all UC-Sec appliances.

6.1.1. Routing Profile

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types.

Create the Routing Profile for Communication Manager and IntelePeer™ SIP Trunk. To add a routing profile, navigate to **UC-Sec Control Center** → **Global Profiles** → **Routing** and select **Add Profile**. Enter a **Profile Name** and click **Next** to continue.
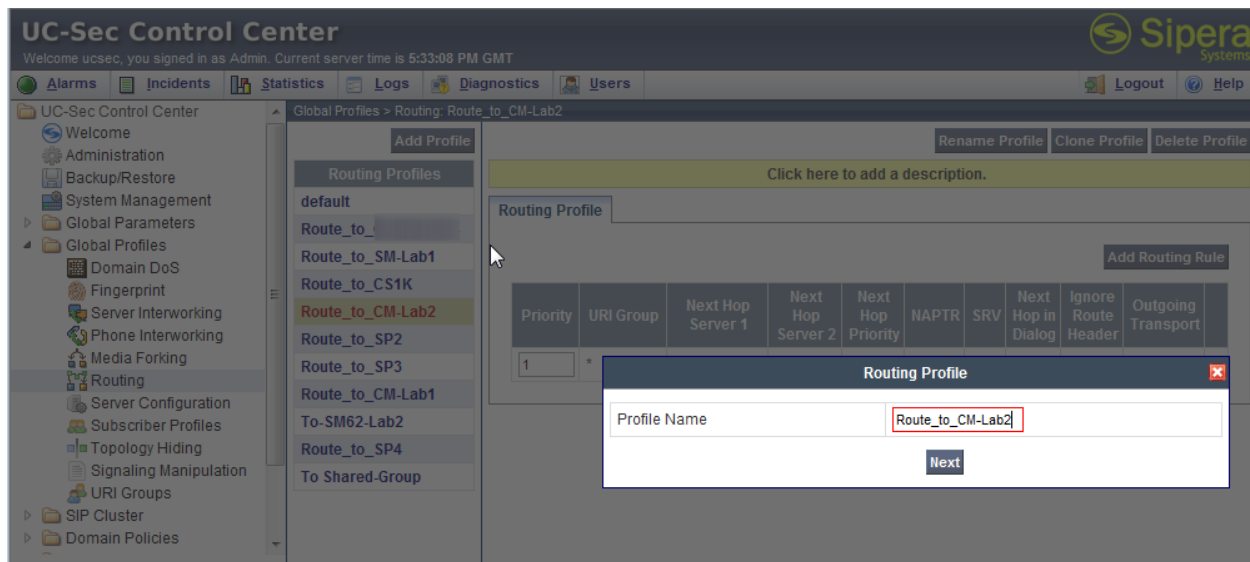
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In the new window that appears, enter the following values. Use default values for all remaining fields:

- **URI Group:**   Select "*****" from the drop down box.
- **Next Hop Server 1:**   Enter the Domain Name or IP address of the Primary Next Hop server.

- **Next Hop Server 2:**   (Optional) Enter the Domain Name or IP address of the secondary Next Hop server.

- **Routing Priority Based on Next Hop Server**: Checked.
- **Outgoing Transport:**   Choose the protocol used for transporting outgoing signaling packets.

Click **Finish** (not shown).

The following screen shows the Routing Profile to Communication Manager. The **Next Hop Server 1** is the IP address of the Communication Manager Processor Ethernet as defined in **Section 5.3**. The Outgoing Transport is set to **TCP** and matches the **Transport Method** set in the Communication Manager Signaling Group in **Section 5.7**.

The following screen shows the Routing Profile to IntelePeer™. In the **Next Hop Server 1** field enter the IP address that IntelePeer™ uses to listen for SIP traffic and set the **Outgoing Transport** to **UDP**.
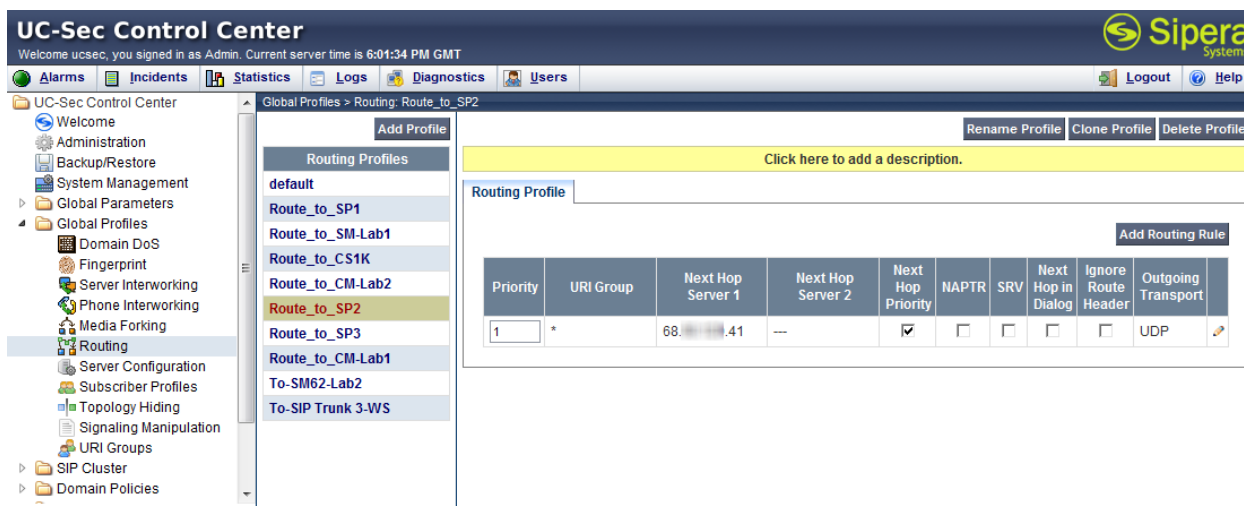
6.1.2. Topology Hiding Profile

The Topology Hiding profile manages how various source and destination addresses, and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks.

Create a Topology Hiding Profile for the enterprise and IntelePeer™ SIP Trunk. In the sample configuration, the **Enterprise** and **IntelePeer™** profiles were cloned from the default profile. To clone a default profile, navigate to **UC-Sec Control Center → Global Profiles → Topology Hiding**. Select the **default** profile and click on **Clone Profile** as shown below.

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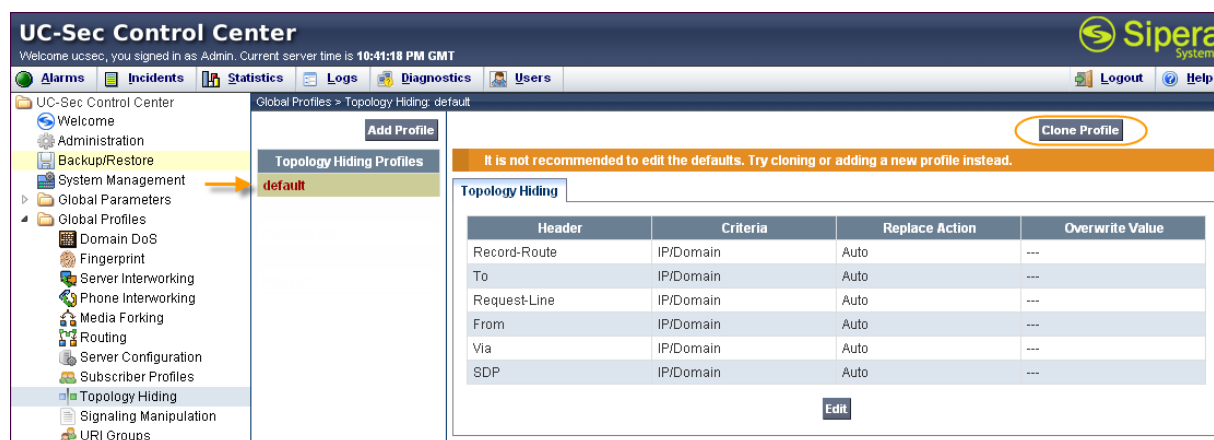
Enter a descriptive name for the new profile and click **Finish**.

Clone Profile	
Profile Name	default
Clone Name	Enterprise

Finish

Edit the **Enterprise** profile to overwrite the **To**, **Request-Line** and **From** headers with the enterprise domain as shown below. The **Overwrite Value** should match the Far-end Domain set in the Communication Manager Signaling Group (**Section 5.7**). Click **Finish** to save the changes.

Edit Topology Hiding Profile

Header	Criteria	Replace Action	Overwrite Value	
Record-Route	IP/Domain	Auto		✕
To	IP/Domain	Overwrite	avayalab.com	✕
Request-Line	IP/Domain	Overwrite	avayalab.com	✕
From	IP/Domain	Overwrite	avayalab.com	✕
Via	IP/Domain	Auto		✕
SDP	IP/Domain	Auto		✕

Finish

It is not necessary to modify the **IntelePeer™** profile from the default values. The following screen shows the Topology Hiding Policy created for IntelePeer™.
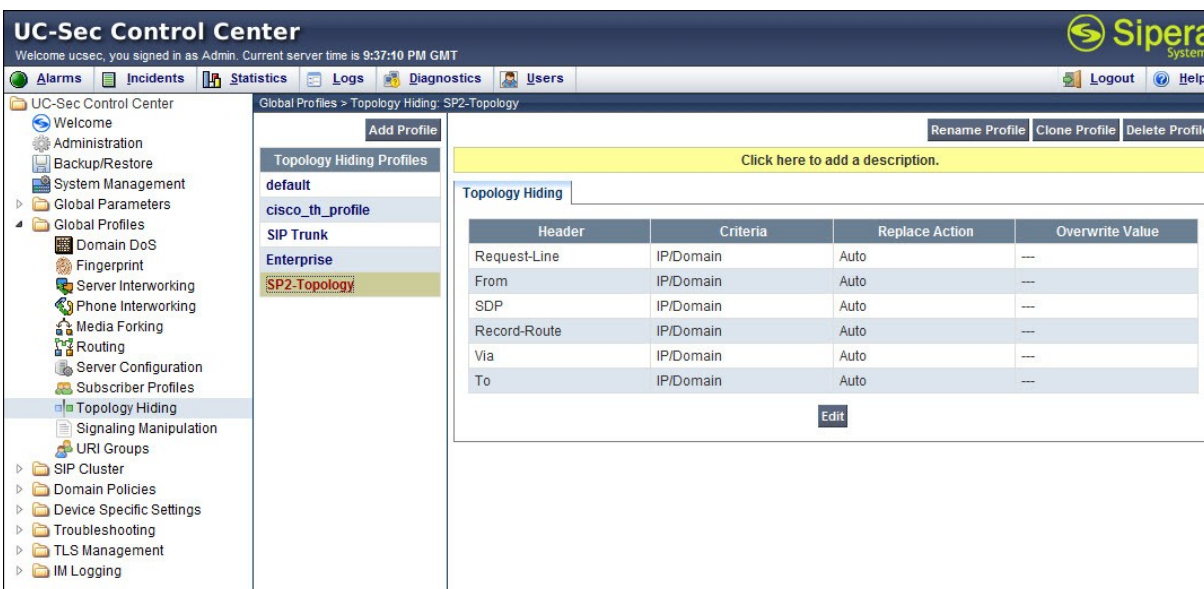
When creating or editing Topology Hiding Profiles, there are six types of headers available for selection in the Header drop-down list to choose from. In addition to the six headers, there are additional headers not listed that are affected when either of two types of listed headers (e.g., **To Header** and **From Header**) are selected in the **Header** drop-down list. **Table 2** lists the six headers along with all of the other affected headers in three header categories (e.g., **Source Headers, Destination Headers, and SDP Headers**).

Topology Hiding Headers	
Main Header Names	**Header(s) Affected by Main Header**
Source Headers	
Record-Route	
From	(1) Referred-By (2) P-Asserted Identity
Via	
Destination Headers	
To	(1) ReferTo
Request-Line	
SDP Headers	
Origin Header	

Table 2: Topology Hiding Headers

6.1.3. Server Interworking Profile

The Server Interworking profile configures and manages various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters (for HA deployments), DoS security statistics, and trusted domains. Interworking Profile features are configured based on different Trunk Servers. There are default profiles available that may be used as is, or modified. New profiles can be configured as described below.

In the sample configuration, separate Server Interworking Profiles were created for the **Enterprise** and **IntelePeer™**.

6.1.3.1 Server Interworking Profile – Enterprise

To create a new Server Interworking Profile for the enterprise, navigate to **UC-Sec Control Center** → **Global Profiles** → **Server Interworking** and click on **Add Profile** as shown below.
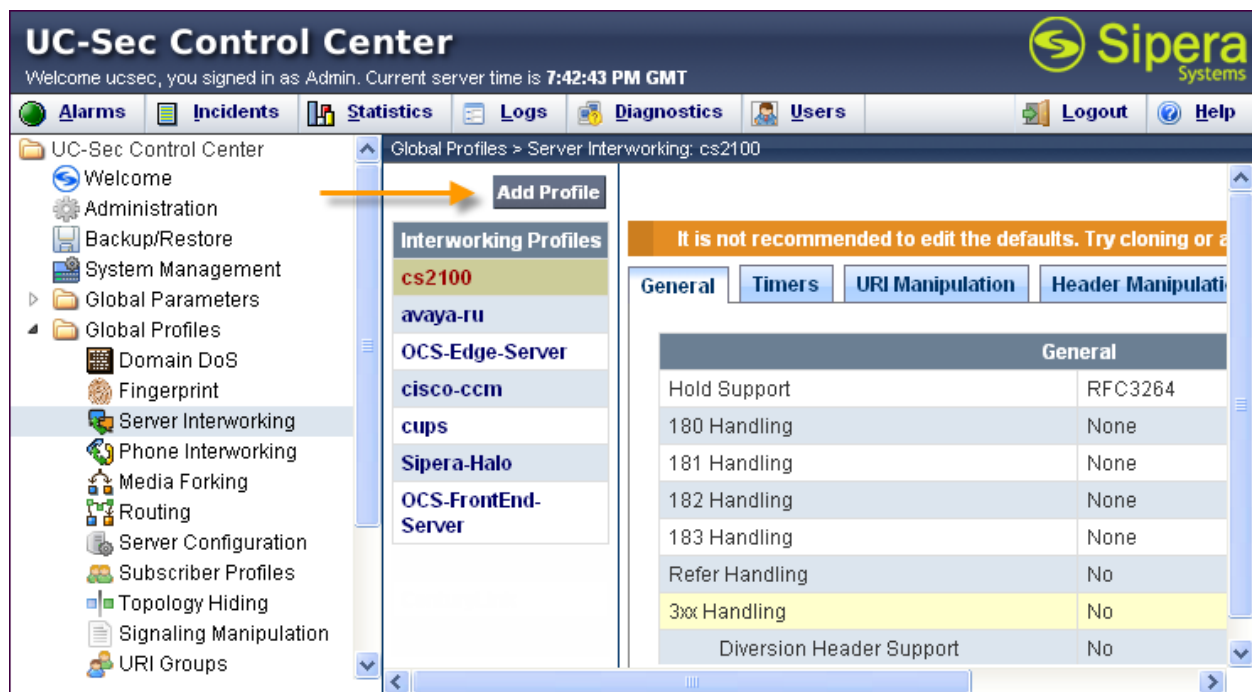
Enter a descriptive name for the new profile and click **Next** to continue.

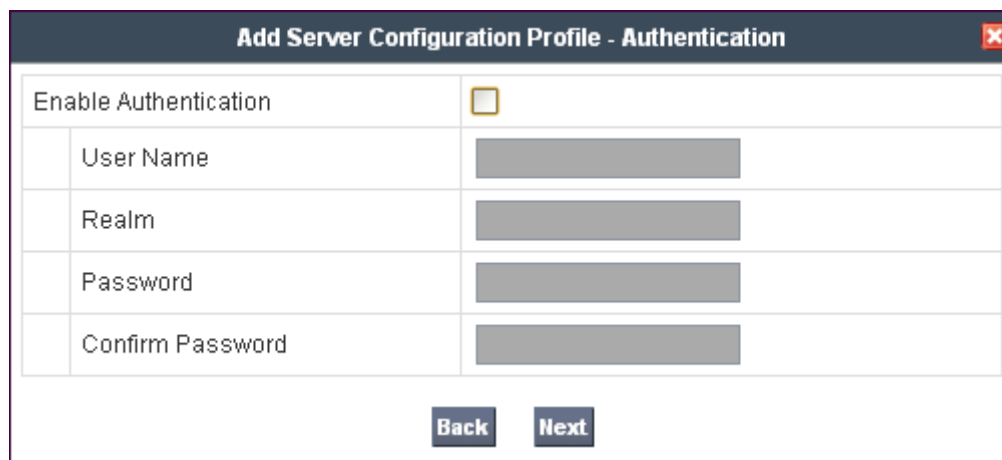In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Hold Support:** Select **RFC3264**.
- **T.38 Support:** Checked.

Click **Next** to continue**.**

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Default values can be used for the next two windows that appear. Click **Next** to continue.

On the **Advanced Settings** window uncheck the following default settings:
- **Topology Hiding: Change Call-ID**
- **Change Max Forwards**
- **Has Remote SBC**

Click **Finish** to save changes.

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6.1.3.2 Server Interworking Profile – IntelePeer™

To create a new Server Interworking Profile for IntelePeer™, navigate to **UC-Sec Control Center → Global Profiles → Server Interworking** and click on **Add Profile** as shown below.

Enter a descriptive name for the new profile and click **Next** to continue.

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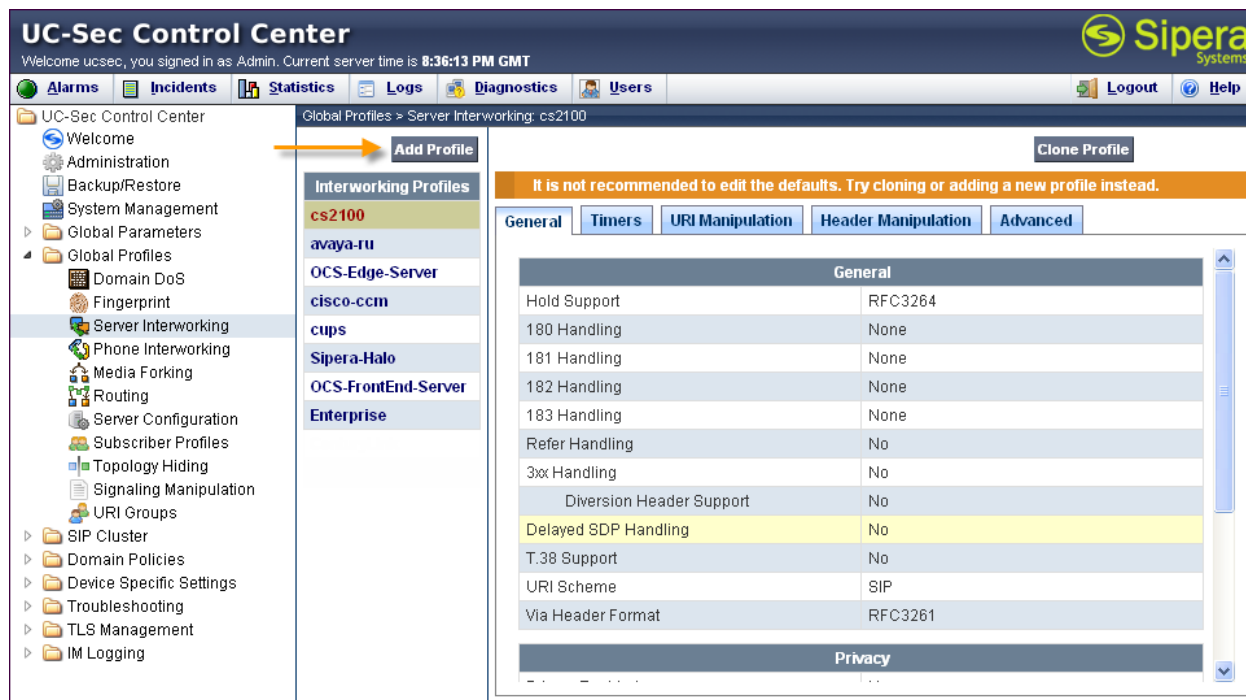
In the new window that appears, keep the default **Hold Support** value of **None**. IntelePeer™ SIP Trunk Service is not capable of supporting calls placed on hold by either the RFC 3264 method using the *a=sendonly* SDP attribute, or the RFC 2543 method of setting the address in the *c=* SDP line to *0.0.0.0*. With the Hold Support set to None, it is necessary to create a Signaling Manipulation (**Section 6.1.4**) to remove the *sendonly* media attribute sent by Communication Manager.

Click **Next** to continue**.**

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Default values can be used for the next two windows that appear. Click **Next** to continue.

Interworking Profile

Privacy

Privacy Enabled	☐
User Name	
P-Asserted-Identity	☐
P-Preferred-Identity	☐
Privacy Header	

DTMF

DTMF Support	⦿ None ◯ SIP NOTIFY ◯ SIP INFO

Back Next

Interworking Profile

Configuration is not required. All fields are optional.

SIP Timers

Min-SE		seconds, [90 - 86400]
Init Timer		milliseconds, [50 - 1000]
Max Timer		milliseconds, [200 - 8000]
Trans Expire		seconds, [1 - 64]
Invite Expire		seconds, [180 - 300]

Transport Timers

TCP Connection Inactive Timer		seconds, [600 - 3600]

Back Next

On the **Advanced Settings** window the default values can be used. Click **Finish** to save changes.

6.1.4. Signaling Manipulation

The Signaling Manipulation feature allows the ability to add, change and delete any of the headers in a SIP message. This feature will give the ability to configure such manipulation in a highly flexible manner using a proprietary scripting language called SigMa.

The SigMa scripting language is designed to express any of the SIP header manipulation operations to be done by the Avaya SBCE. Using this language, a script can be written and tied to a given flow through the EMS GUI. The Avaya SBCE appliance then interprets this script at the given entry point or "hook point".

These application notes will not discuss the full feature of the Signaling Manipulation but will show an example of a script created during compliance testing to aid in topology hiding.

To create a new Signaling Manipulation, navigate to **UC-Sec Control Center** → **Global Profiles** → **Signaling Manipulation** and click on **Add Script** (not shown). A new blank SigMa Editor window will pop up. For more information on Signaling Manipulation see **Reference** [**13**].

The following sample script is written in two sections. Each section begins with a comment describing what will take place in that portion of the script. The first section will act on the response from Communication Manager to an inbound call from IntelePeer™ (e.g., 180 Ringing and 200 OK) while the second acts on the request of an outbound call to IntelePeer™ from Communication Manager (e.g., re-INVITE messages from Communication Manager for audio shuffling). The script is further broken down as follows:

• **within session "All"**	Transformations applied to all SIP sessions.
• **act on response**	Actions to be taken to the response of an INVITE (e.g., 180 Ringing and 200 OK).
• **%DIRECTION="OUTBOUND"**	Applied to a messages leaving the Avaya SBCE.
• **%ENTRY_POINT="POST_ROUTING"**	The "hook point" to apply the script after the SIP message has routed through Avaya SBCE.
• **%HEADERS["p-asserted-identity"][1]**	Used to retrieve an entire header. The first dimension denotes which header while the second dimension denotes the 1st instance of the header in a message**.**
• **.regex_replace ("avayalab\.com","205.xxx.xxx.35:5060")**	An action to replace a given match with the provide string (e.g., find "avayalab.com" and replace it with "205.xxx.xxx.35:5060".

The P-Asserted-Identity header will be modified by replacing the domain "avayalab.com" with the external IP address of the Avaya SBCE and the SIP port of 5060 in both the response and request sessions.
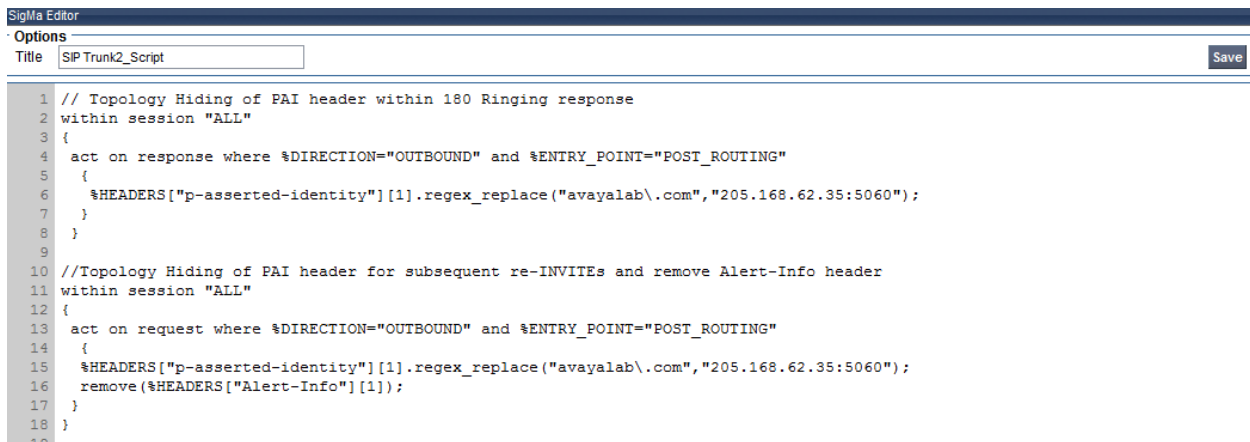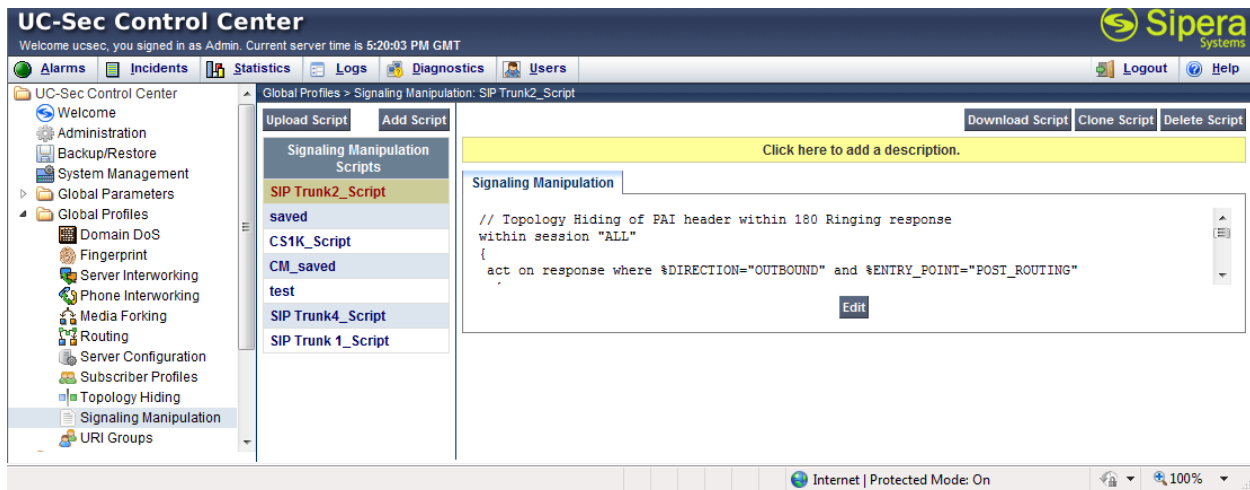
The screen above shows the finished Signaling Manipulation Script **IntelePeer™ Script** (SIP Trunk2_Script).

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6.1.5. Server Configuration

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics, and trusted domains.

In the sample configuration, separate Server Configurations were created for **Communication _Manager** and **IntelePeer™.**

6.1.5.1 Server Configuration – Communication Manager

To add a Server Configuration Profile for Communication Manager, navigate to **UC-Sec Control Center** → **Global Profiles** → **Server Configuration** and click on **Add Profile** as shown below.
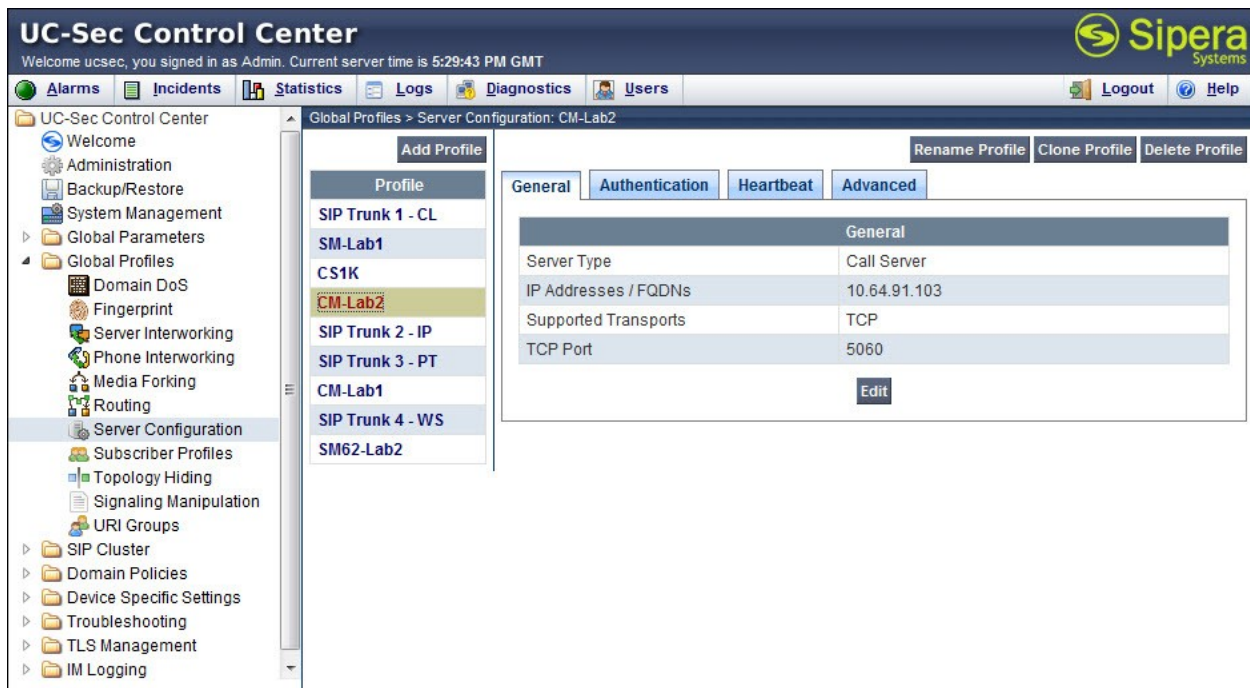
Enter a descriptive name for the new profile and click **Next**.
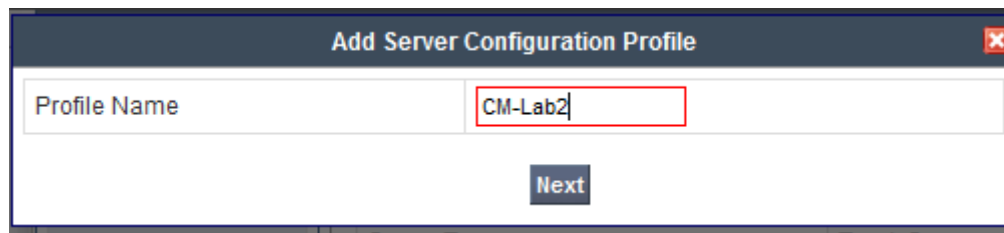
In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Server Type:** Select **Call Server** from the drop-down box.
- **IP Addresses / Supported FQDNs:** Enter the IP address of the Communication Manager Processor Ethernet as defined in **Section 5.3**.
- **Supported Transports:** Select **TCP**.
- **TCP Port:** Port number on which to send SIP requests to Communication Manager. This should match the port number used in the **Far-end Listen Port** in the Communication Manager Signaling Group as defined **Section 5.7.**

Click **Next** to continue.

Add Server Configuration Profile - General	
Server Type	Call Server
IP Addresses / Supported FQDNs Comma seperated list	10.64.91.103
Supported Transports	☑ TCP ☐ UDP ☐ TLS
TCP Port	5060
UDP Port	
TLS Port	

Back   Next

Verify **Enable Authentication** is unchecked as Communication Manager does not require authentication. Click **Next** to continue.

In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Enable Heartbeat:**   Checked.
- **Method:**   Select **OPTIONS** from the drop-down box.
- **Frequency:**   Choose the desired frequency in seconds the Avaya SBCE will send SIP OPTIONS. For compliance testing **60** seconds was chosen.
- **From URI:**   Enter the URI to be sent in the FROM header for SIP OPTIONS.
- **To URI:**   Enter the URI to be sent in the TO header for SIP OPTIONS.

Click **Next** to continue.

In the new window that appears, select the **Interworking Profile** created for the enterprise in **Section 6.1.3.1**. Use default values for all remaining fields. Click **Finish** to save the configuration (not shown).

Advanced	
Enable DoS Protection	☐
Enable Grooming	☐
Interworking Profile	
Signaling Manipulation Script	SIP Trunk2_Script
UDP Connection Type	SUBID

Edit

6.1.5.2 Server Configuration - IntelePeer™

To add a Server Configuration Profile for IntelePeer™ navigate to **UC-Sec Control Center** →**Global Profiles** → **Server Configuration** and click on **Add Profile** (not shown). Enter a descriptive name for the new profile and click **Next**.

Add Server Configuration Profile	
Profile Name	Intelepeer

Next

In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Server Type:**   Select **Trunk Server** from the drop-down box.
- **IP Addresses / Supported FQDNs:**   Enter the IP address(es) of the SIP proxy(ies) of the service provider. In the case of the compliance test, this is the IntelePeer™ SIP Trunk IP address. This will associate the inbound SIP messages from IntelePeer™ to this Sever Configuration.
- **Supported Transports:**   Select the transport protocol to be used for SIP traffic between Avaya SBCE and IntelePeer™.
- **UDP Port:**   Enter the port number that IntelePeer™ uses to send SIP traffic.

Click **Next** to continue.

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Verify **Enable Authentication** is unchecked as IntelePeer™ does not require authentication. Click **Next** to continue.

In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Enable Heartbeat:**   Checked.
- **Method:**   Select **OPTIONS** from the drop-down box.
- **Frequency:**   Choose the desired frequency in seconds the Avaya SBCE will send SIP OPTIONS. For compliance testing **60** seconds was chosen.

- **From URI:**   Enter the URI to be sent in the FROM header for
                           SIP OPTIONS.
- **To URI:**   Enter the URI to be sent in the TO header for SIP
                           OPTIONS.

Click **Next** to continue.

The SIP OPTIONS are sent to the SIP proxy(ies) entered in the **IP Addresses /Supported FQDNs** in the **Server Configuration Profile.** The URI of PING@ IntelePeer.com was used in the sample configuration to better identify the SIP OPTIONS in the call traces. IntelePeer™ does not look at the From and To headers when replying to SIP OPTIONS so any URI can be used as long as it is in the proper format (USER@DOMAIN).

Heartbeat	
Enable Heartbeat	☑
Method	OPTIONS
Frequency	60 seconds
From URI	PING@Intelepeer.com
To URI	PING@Intelepeer.com
TCP Probe	☐

In the new window that appears, select the **Interworking Profile** created for IntelePeer™ in **Section 6.1.3.2**. Select the **Signaling Manipulation Script** created in **Section 6.1.4**. Use default values for all remaining fields. Click **Finish** to save the configuration (not shown).

Advanced	
Enable DoS Protection	☐
Enable Grooming	☐
Interworking Profile	
Signaling Manipulation Script	SIP Trunk2_Script
UDP Connection Type	SUBID

6.2. Domain Policies

The Domain Policies feature configures, applies, and manages various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating to the enterprise. These criteria can be used to trigger policies which, in turn, activate various security features of the UC-Sec security device to aggregate, monitor, control, and normalize call flows. There are default policies available to use, or a custom domain policy can be created.

6.2.1. Media Rules

Media Rules define RTP media packet parameters such as packet encryption techniques and prioritizing encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the UC-Sec security product.

Create a custom Media Rule to set the Quality of Service and Media Anomaly Detection. The sample configuration shows a custom Media Rule **No-Media-Detection** created for the enterprise and IntelePeer™.

To create a custom Media Rule, navigate to **UC-Sec Control Center** →**Domain Policies** → **Media Rules**. With **default-low-med** selected, click **Clone Rule** as shown below.
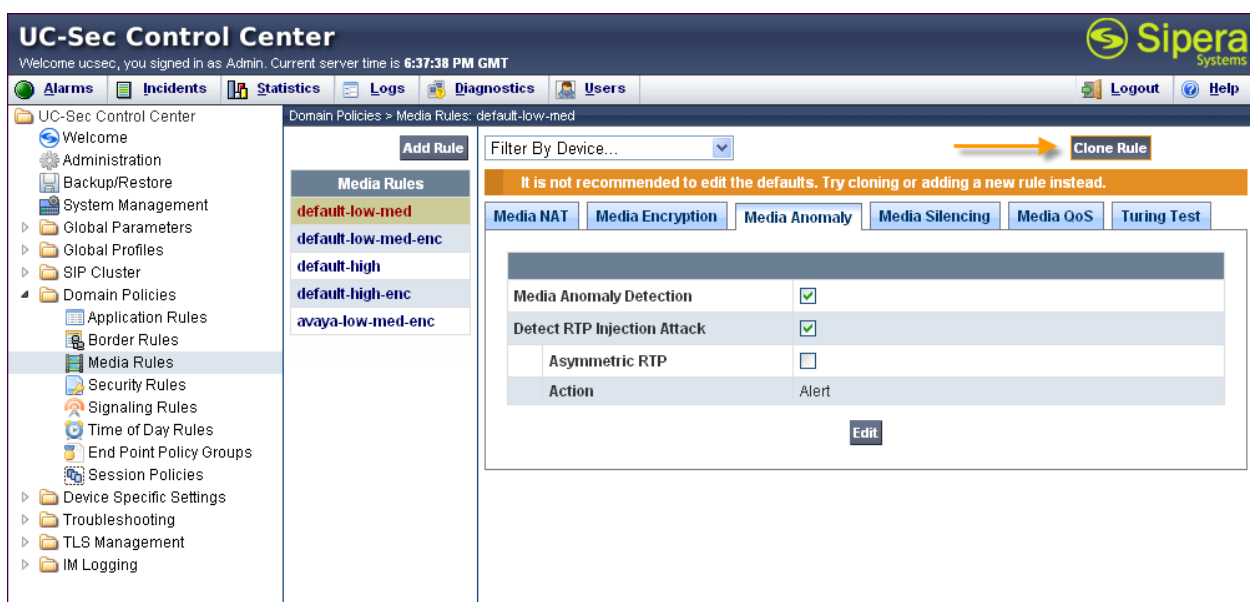
Enter a descriptive name for the new rule and click **Finish**.

When the RTP packets of a call are shuffled from Communication Manager to an IP Phone, the Avaya SBCE will interpret this as an anomaly and an alert will be created in the Incidents Log. Disabling **Media Anomaly Detection** prevents the **RTP Injection Attack** alerts from being

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created during an audio shuffle. To modify the rule, select the **Media Anomaly** tab and click **Edit**. Uncheck **Media Anomaly Detection** and click **Finish** (not shown)**.**

The following screen shows the **No-Media-Detection** rule with **Media Anomaly Detection** disabled.
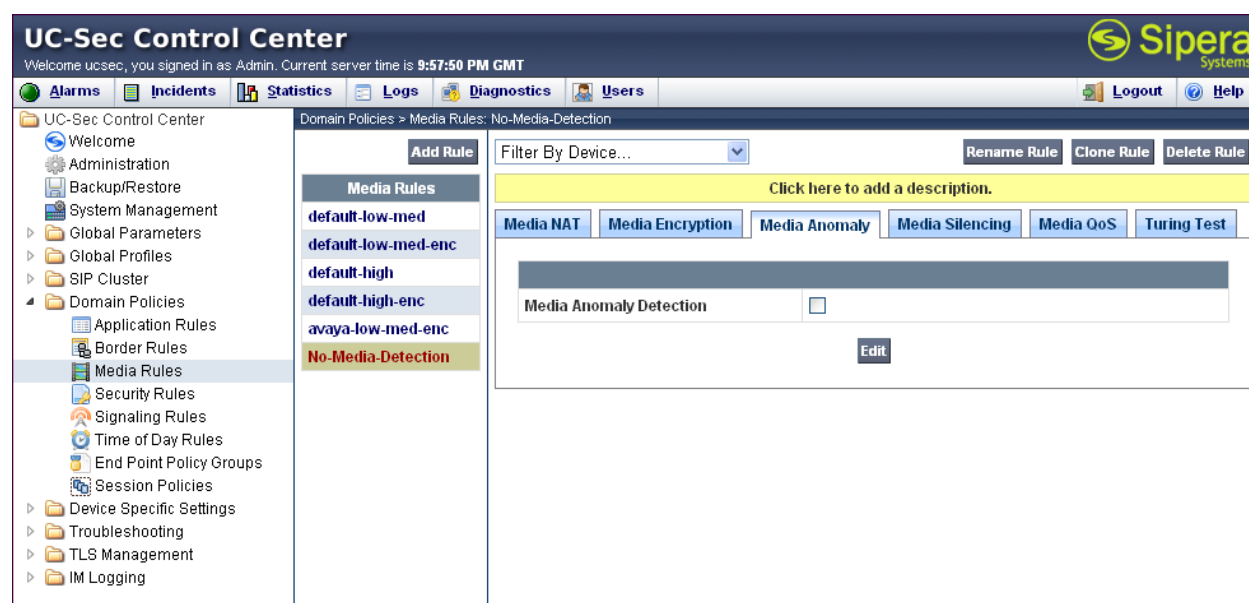
On the **Media QoS** tab, select the proper Quality of Service (QoS). The Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for the media. The following screen shows the QoS values used for compliance testing.

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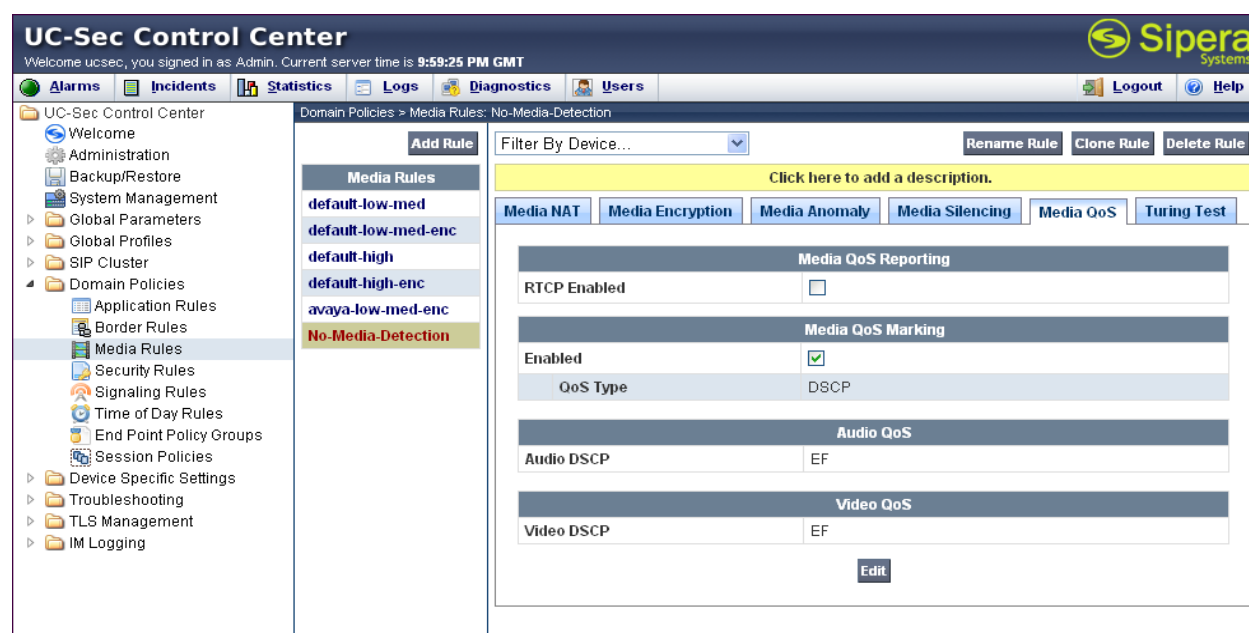
6.2.2. Signaling Rules

Signaling Rules define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by the UC-Sec, they are parsed and "pattern-matched" against the particular signaling criteria defined by these rules. Packets matching the criteria defined by the Signaling Rules are tagged for further policy matching.

Clone and modify the **default** signaling rule to strip the Alert Info header from the SIP message to hide the topology of the enterprise network from the external network before it is sent to IntelePeer™. To clone a signaling rule, navigate to **UC-Sec Control Center →Domain Policies → Signaling Rules**. With the **default** rule chosen, click on **Clone Rule** as shown below.
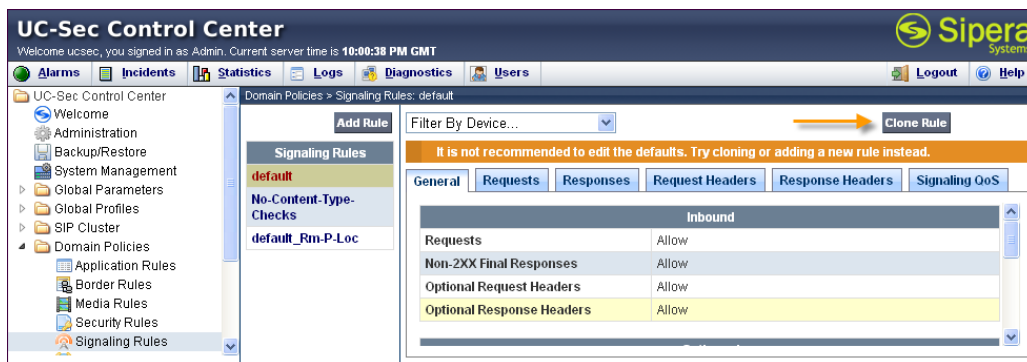
Enter a descriptive name for the new rule and click **Finish**.
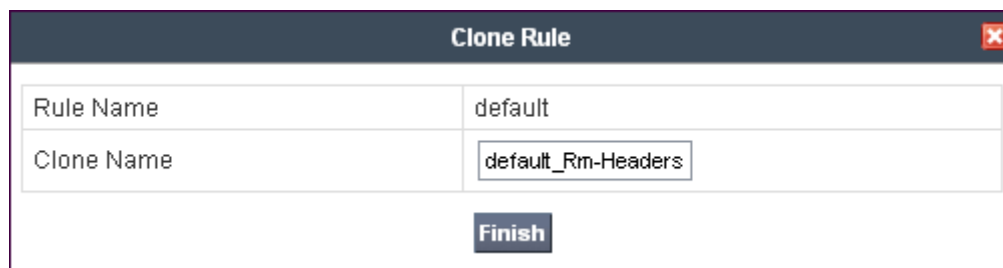
Select the **Request Headers tab** and click **Add In Header Control** (not shown). In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Header Name:**   Enter **Alert-Info**.
- **Method Name:**   Select **INVITE** from the drop-down box.
- **Header Criteria:**   Select **Forbidden**.
- **Presence Action:**   Select **Remove header** from the drop-down box.

Click **Finish** to save the configuration

Repeat these steps for any other headers that should be removed. The following screens show the **default_Rm-Headers** rule used in the sample configuration with the **Alert-Info** header configured to be removed.

On the **Signaling QoS** tab select the proper Quality of Service (QoS). The Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for signaling. The following screen shows the QoS values used for compliance testing.

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6.2.3. Application Rules

Application Rules define which types of SIP-based Unified Communications (UC) applications the UC-Sec security device will protect: voice, video, and/or Instant Messaging (IM). In addition, you can determine the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion.

Create an Application Rule to set the number of concurrent voice traffic sessions allowed. The sample configuration cloned and modified the **default** application rule to increase the number of **Maximum Concurrent Session** and **Maximum Sessions Per Endpoint**. To clone an application rule, navigate to **UC-Sec Control Center → Domain Policies → Application Rules**. With the **default** rule chosen, click on **Clone Rule** as shown below.

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Enter a descriptive name for the new rule and click **Finish**.

Modify the rule by clicking the **Edit** button. Set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** for the **Voice** application to a value high enough for the amount of traffic the network is able process. Keep in mind the Avaya SBCE takes 30 seconds for sessions to be cleared after disconnect. The following screen shows the modified Application Rule with the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** set to **2000**. In the sample configuration, Communication Manager was programmed to control the concurrent sessions by setting the number of members in the trunk group (**Section 5.8**) to the allotted amount. Therefore, the values in the Application Rule **MaxVoice** were set high enough to be considered non-blocking.

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6.2.4. Endpoint Policy Group

The rules created within the Domain Policy section are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to a Server Flow in **Section 6.3.4.** Create a separate Endpoint Policy Group for both the enterprise and the IntelePeer™ SIP Trunk Service.

To create a new policy group, navigate to **UC-Sec Control Center →Domain Policies → Endpoint Policy Groups** and click on **Add Group** as shown below.
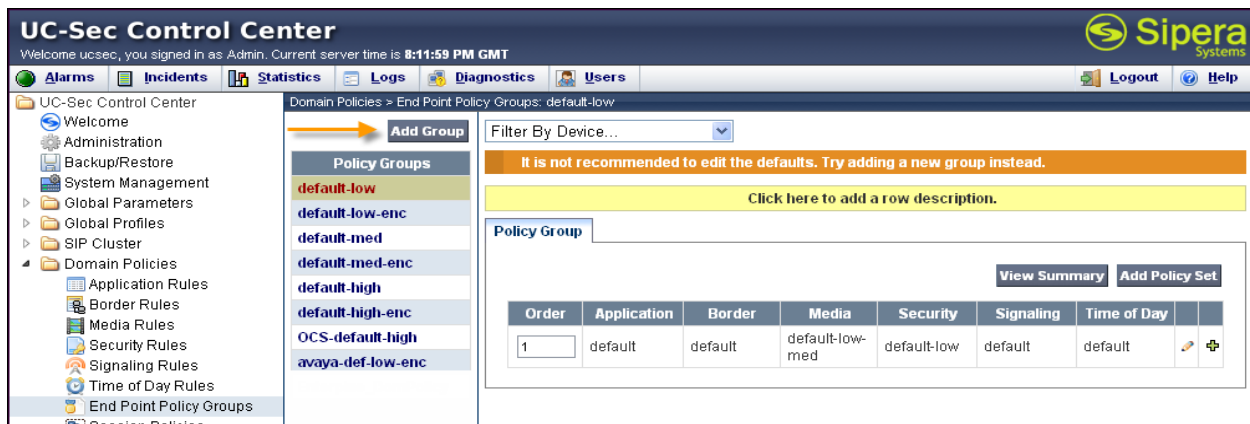
The following screen shows the **Enterprise_DomPolicy** created for the enterprise. Set the **Application**, **Media** and **Signaling** rules to the ones previously created. Set the **Border** and **Time of Day** rules to **default** and set the **Security** rule to **default-low**.

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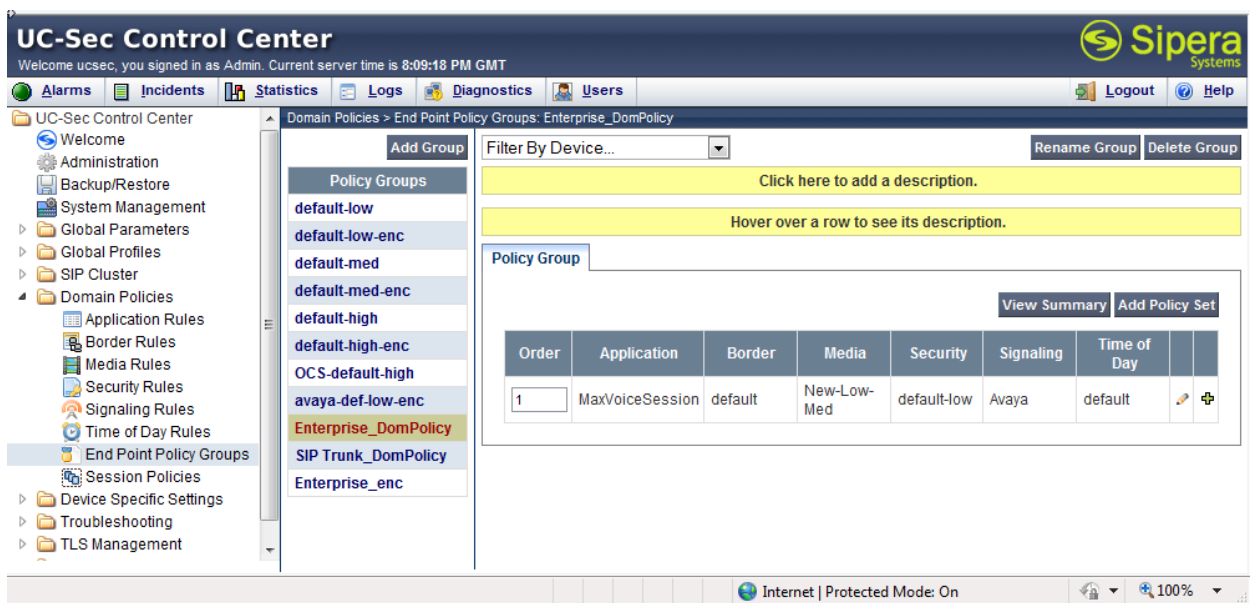
The following screen shows the **SIP Trunk_DomPolicy** created for the IntelePeer™ SIP Trunk Service. Set the **Application**, **Media** and **Signaling** (Avaya) rules to the ones previously created.   Set the **Border** and **Time of Day** rules to **default** and set the **Security** rule to **default-high**.

6.3. Device Specific Settings

The Device Specific Settings feature allows aggregate system information to be viewed, and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network. Specifically, it gives the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality and protocol scrubber rules, end-point and session call flows, as well as the ability to manage system logs and control security features.

6.3.1. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of the Avaya SBCE, certain network-specific information is defined such as device IP address(es), public IP address(es), netmask, gateway, etc. to interface the device to the network. It is this information that populates the various Network Management tab displays, which can be edited as needed to optimize device performance and network efficiency.

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Navigate to **UC-Sec Control Center** →**Device Specific Settings** → **Network Management** and verify the IP addresses assigned to the interfaces and that the interfaces are enabled. The following screen shows the private interface is assigned to **A1** and the external interface is assigned to **B1.**

Enable the interfaces used to connect to the inside and outside networks on the **Interface Configuration** tab. The following screen shows that the interfaces **A1** and **B1** are **Enabled**. To enable an interface click it's **Toggle State** button.

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6.3.2. Signaling Interface

The Signaling Interface screen is where the SIP signaling ports are defined. The Avaya SBCE will listen for SIP requests on the defined ports. Create a Signaling Interface for both the inside and outside IP interfaces.

To create a new Signaling Interface, navigate to **UC-Sec Control Center →Device Specific Settings → Signaling Interface** and click **Add Signaling Interface.**

The following screen shows the signaling interfaces created in the sample configuration with TCP and UDP ports 5060 used for the inside and outside IP interfaces respectively.

6.3.3. Media Interface

The Media Interface screen is where the media ports are defined. The Avaya SBCE will listen for RTP media on the defined ports. Create a Media Interface for both the inside and outside IP interfaces. The inside port range needs to match the **UDP Port Min** and **UDP Port Max** fields in the Communication Manager IP Network Region form created in **Section 5.6.** The outside port range should match the RTP port range provided by IntelePeer™.

To create a new Media Interface, navigate to **UC-Sec Control Center → Device Specific Settings → Media Interface** and click **Add Media Interface.**

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The following screen shows the media interfaces created in the sample configuration for the inside and outside IP interfaces. After the media interfaces are created, an application restart is necessary before the changes will take effect.

6.3.4. End Point Flows - Server Flow

When a packet is received by UC-Sec, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.

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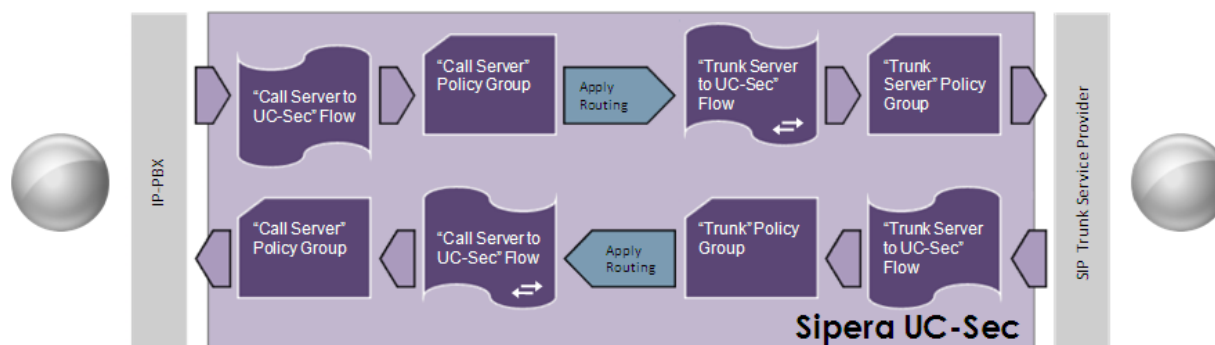
Create a Server Flow for Communication Manager and the IntelePeer™ SIP Trunk Service. To create a Server Flow, navigate to **UC-Sec Control Center → Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add Flow** as shown below.
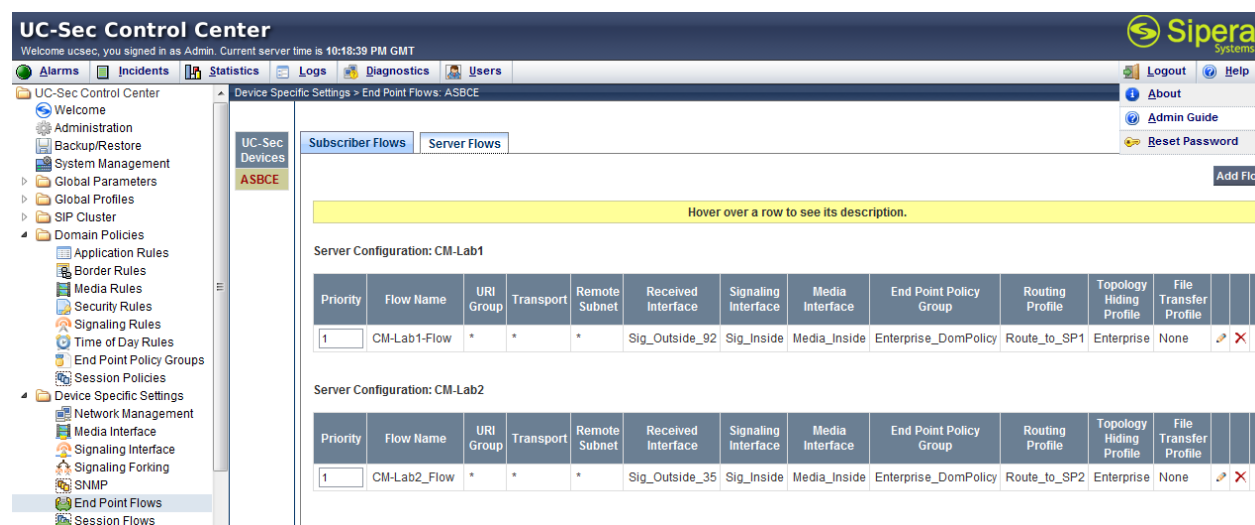
In the new window that appears, enter the following values. Use default values for all remaining fields:

- **Flow Name:** Enter a descriptive name.
- **Server Configuration:** Select a Server Configuration created in **Section 6.1.5** to assign to the Flow.
- **Received Interface:** Select the Signaling Interface the Server Configuration is allowed to receive SIP messages from.
- **Signaling Interface:** Select the Signaling Interface used to communicate with the Server Configuration.
- **Media Interface:** Select the Media Interface used to communicate with the Server Configuration.
- **End Point Policy Group:** Select the policy assigned to the Server Configuration.
- **Routing Profile:** Select the profile the Server Configuration will use to route SIP messages to.
- **Topology Hiding Profile:** Select the profile to apply toward the Server Configuration.

Click **Finish** to save and exit.

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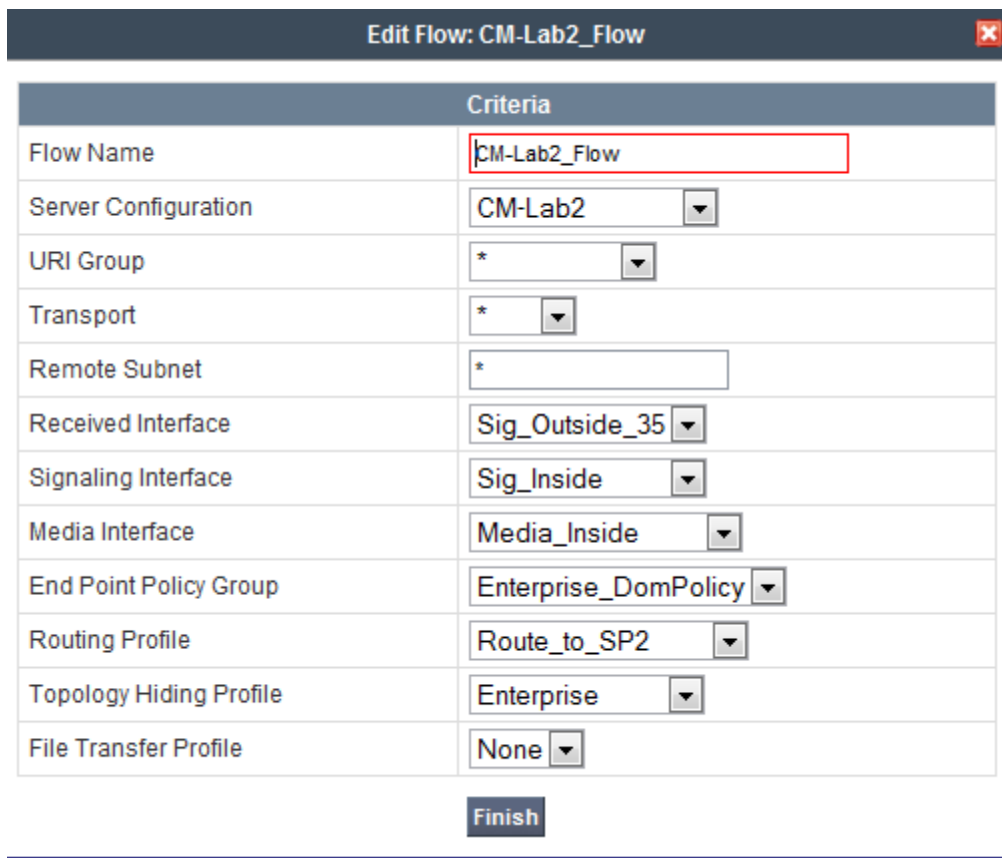
The following screen shows the Sever Flow for IntelePeer™:

Edit Flow: SP2_Flow	
Criteria	
Flow Name	SP2_Flow
Server Configuration	SIP Trunk 2 - IP
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_Inside
Signaling Interface	Sig_Outside_35
Media Interface	Media_Outside_35
End Point Policy Group	SIP Trunk_DomPolicy
Routing Profile	Route_to_CM-Lab2
Topology Hiding Profile	SP2-Topology
File Transfer Profile	None

Finish

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The following screen shows the Sever Flow for Communication Manager:

7. IP SIP Trunk Service Configuration

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

8. Verification and Troubleshooting

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

8.1. Verification

The following steps may be used to verify the configuration:

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.

Use the SAT interface on Communication Manager to verify status of SIP trunks. Specifically use the **status trunk n** command to verify the active call has ended. Where **n** is the trunk group number used for InteLePeer™ SIP Trunk Service defined in **Section 5.8**.

Below is an example of an active call.

```
status trunk 2

                          TRUNK GROUP STATUS

Member    Port       Service State       Mtce Connected Ports
                                         Busy

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

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

```
status trunk 2

                          TRUNK GROUP STATUS

Member    Port       Service State       Mtce Connected Ports
                                         Busy

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

8.2. 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.

2. Avaya SBCE:
   - **Incidences** - Displays alerts captured by the UC-Sec appliance.

   - **Diagnostics** - Allows for PING tests and displays application and protocol use.
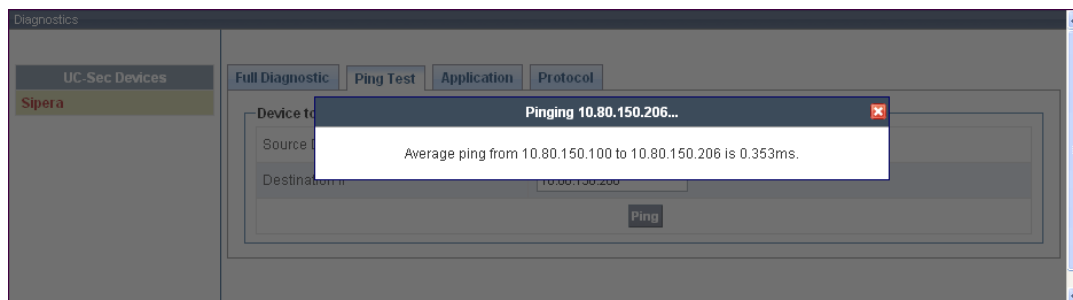
- **Troubleshooting → Trace Settings** - Configure and display call traces and packet captures for the UC-Sec appliance.

The packet capture file can be downloaded and viewed using a Network Protocol Analyzer such as Wireshark:
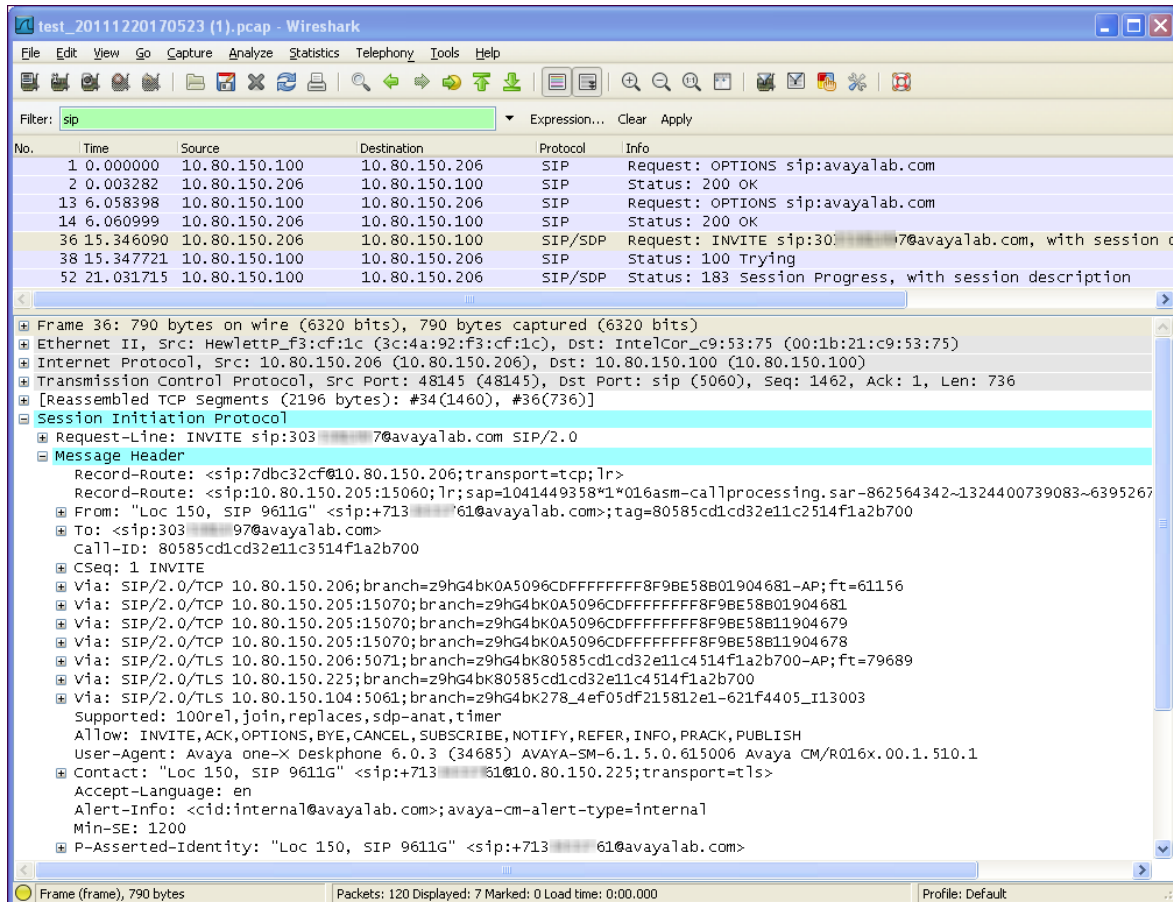
9. Conclusion

These Application Notes describe the configuration necessary to connect the Avaya Session Border Controller for Enterprise and Avaya Aura® Communication Manager Evolution Server to the IntelePeer™ SIP Trunk Service. The IntelePeer™ SIP Trunk Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. The IntelePeer™ SIP Trunk Service provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks.

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10. References

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

[1] *Installing and Configuring Avaya Aura® System Platform, Release 6.0.3,* Issue 2.1, March 2012.

[2] *Administering Avaya Aura® System Platform, Release 6.0.3,* February 2011.

[3] *Administering Avaya Aura® Communication Manager, Release 6.0 ,* Document Number 03-300509, Issue 6.0 ,June 2010

[4] *Avaya Aura® Communication Manager Feature Description and Implementation, Release 6.0,* Document Number 555-245-205, Issue 8.0, June 2010

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

[6] *4600 Series IP Telephone LAN Administrator Guide,* Document Number 555-233-507.

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

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

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

[10] *Administrator Guide for Avaya Communication Manager,* Document Number 03-300509.

[11] *Feature Description and Implementation for Avaya Communication Manager,* Document Number 555-245-205

[12] *UC-Sec Install Guide (102-5224-400v1.01)*

[13] *UC-Sec Administration Guide (010-5423-400v106)*

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

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

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

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

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