Avaya Solution & Interoperability Test Lab

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

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

Table of Contents

1. Introduction 4
2. General Test Approach and Test Results 4
2.1. Interoperability Compliance Testing 4
2.2. Test Results 5
2.3. Support 6
3. Reference Configuration 6
4. Equipment and Software Validated 8
5. Configure Avaya Aura® Communication Manager 9
5.1. Licensing and Capacity 9
5.2. System Features 10
5.3. IP Node Names 11
5.4. Codecs 11
5.5. IP Interface for procr 12
5.6. IP Network Region 12
5.7. Signaling Group 13
5.8. Trunk Group 15
5.9. Inbound Routing 17
5.10. Calling Party Information 18
5.11. Outbound Routing 19
5.12. Saving Communication Manager Configuration Changes 22
6. Configure Avaya Session Border Controller for Enterprise 23
6.1. Global Profiles 27
6.1.1. Routing Profile 27
6.1.2. Topology Hiding Profile 29
6.1.3. Server Interworking Profile 32
6.1.4. Signaling Manipulation 39
6.1.5. Server Configuration 42
6.2. Domain Policies 49
6.2.1. Media Rules 49
6.2.2. Signaling Rules 51
6.2.3. Application Rules 54
6.2.4. Endpoint Policy Group 55
6.3. Device Specific Settings 57
6.3.1. Network Management 57
6.3.2. Signaling Interface 58
6.3.3. Media Interface 59
6.3.4. End Point Flows - Server Flow 60
7. Dynamic IP SIP Trunk Service Configuration 63
8. Verification and Troubleshooting 64
8.1. Verification 64

8.2. Troubleshooting 65
9. Conclusion 67
10. References 68

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between the PAETEC Dynamic IP SIP Trunk Service and an Avaya SIP-enabled enterprise solution. PAETEC can offer the Dynamic IP SIP Trunk Service using several different platform technologies in the PAETEC network. These Application Notes correspond to the Dynamic IP SIP Trunk Service offered using a BroadSoft platform in the network. The Avaya solution consists of Avaya 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 PAETEC Dynamic IP SIP Trunk Service are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using 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 Dynamic IP 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 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:

- 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, operator assisted calls and local directory assistance (411).
- Codecs G.729A, G.711MU and G.711A.
- 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.
- Network Call Redirection using the SIP REFER method or a 302 response.
- Off-net call forwarding and mobility (extension to cellular).

Items not supported or not tested included the following:

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

2.2. Test Results

The Dynamic IP SIP Trunk Service passed compliance testing.

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

- **Calling Party Number (PSTN transfers)**: The calling party number displayed on the PSTN phone is not updated to reflect the true connected party on calls that are transferred to the PSTN. After the call transfer is complete, the calling party number displays the number of the transferring party and not the actual connected party.
- **Network Call Redirection**: When PAETEC's Enterprise Trunking feature is active and Communication Manager is programmed to redirect an inbound call to a PSTN number before answering the call in a vector, PAETEC will send an ACK to the "302 Moved Temporarily" SIP message from the enterprise but will not redirect the call to the new party in the Contact header of the 302 message. The inbound call initiator hears a recording from PAETEC in this failure scenario. A workaround is to use the REFER method to redirect the call by having Communication Manager answer the call first with an announcement. When PAETEC's Enterprise Trunking feature is NOT active, Network Call Redirection works as expected.
- **SendOnly SIP Parameter:** With the Network Call Redirection feature enabled, Communication Manager will use the SIP parameter *SendOnly* to signal any hold call conditions. The *SendOnly* SIP parameter is currently not supported by PAETEC Dynamic IP service and will respond with an inactive media when it receives *SendOnly* instead of responding with *RecvOnly*. As a result, the originating side does not hear anything until the re-INVITE comes in with *SendRecv*. The Avaya Session Border Controller for Enterprise is used to remove the *SendOnly* parameter to allow hold music to be received by PAETEC properly. See **Section 6.1.4**.

2.3. Support

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

3. Reference Configuration

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

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

Located at the edge of the enterprise is the 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.

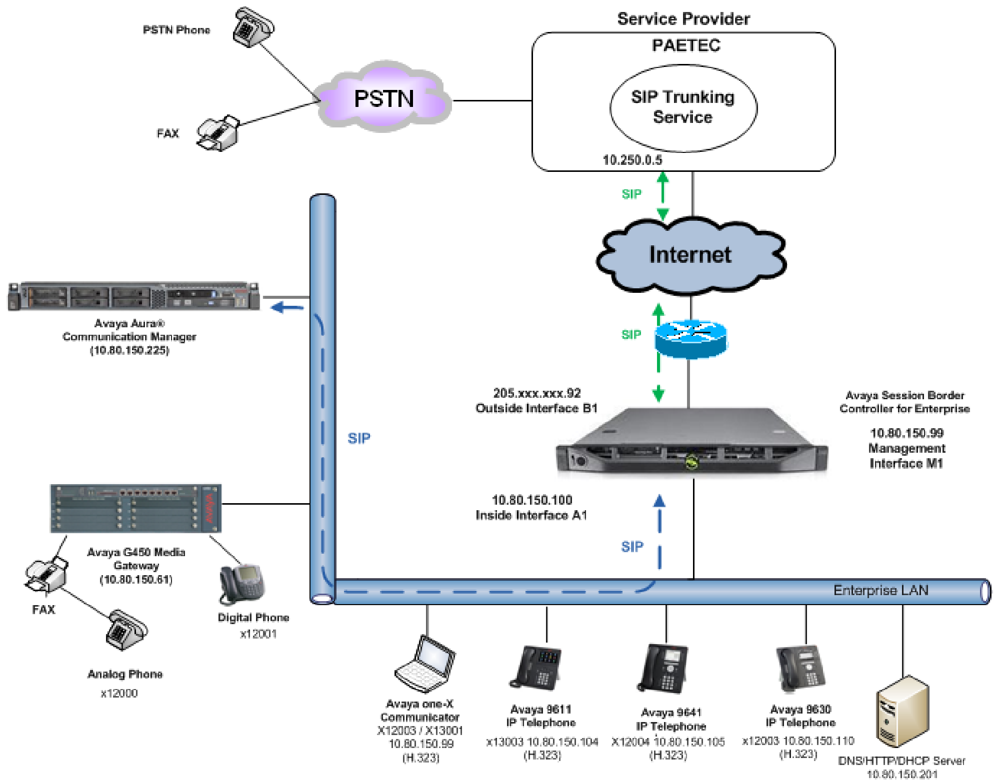


Figure 1: Avaya IP Telephony Network using the Dynamic IP SIP Trunk Service

For inbound calls, the calls flow from the service provider to the Avaya SBCE 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 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 Dynamic IP SIP Trunk Service.

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

4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Component	Release
Avaya Aura® 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 9611 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
PAETEC SIP Trunking Solution Components	
Component	Release
BroadSoft Platform	14sp9

Table 1: Equipment and Software Tested

The specific configuration above was used for the compatibility testing.

Note: 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 Dynamic IP SIP Trunk Service. A SIP trunk is established between Communication Manager and Avaya SBCE for use by signaling traffic to and from PAETEC. It is assumed the general installation of Communication Manager and 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 shows that **12000** licenses are available and **290** 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 2
                   Maximum Video Capable IP Softphones: 18000 1
                       Maximum Administered SIP Trunks: 12000 290
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

5.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to **all** to allow calls 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.

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

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

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

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

ENBLOC DIALING PARAMETERS
   Enable Enbloc Dialing without ARS FAC? n

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

5.3. IP Node Names

Use the **change node-names ip** command to verify the node name defined for the IP address of Communication Manager **(procr)** created during installation. Add a node name and IP address for 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
CMMessaging         10.80.150.225
ASBCE               10.80.150.100
default             0.0.0.0
procr               10.80.150.225
procr6              ::
```

5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, ip-codec-set 2 was used for this purpose. The Dynamic IP SIP Trunk Service supports G.729A, G.711A and G.711MU. During compliance testing each of the supported codecs were tested independently by changing the order of preference to list the codec being tested as the first choice. The true order of preference is defined by the end customer. In the example below, **G.729A** and **G.711MU** 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.729A           n          2        20
 2: G.711MU          n          2        20
 3:
```

Since T.38 fax is not supported, set the **Fax Mode** to **off** on **Page 2**.

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

                           IP Codec Set

                               Allow Direct-IP Multimedia? n

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

5.5. IP 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 PE 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 procr                                          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: 10.80.150.225


           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 the rest of the enterprise. Use the **change ip-network-region 2** command to configure 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: PAETEC SIP TRUNK
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

On **Page 4**, define the IP codec set to be used for traffic between region 2 and region 1 (the rest of the enterprise). Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The example below shows the settings used for the compliance test. It indicates that codec set 2 will be used for calls between region 2 (the service provider region) and region 1 (the rest of the enterprise).

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

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

5.7. Signaling Group

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

- Set the **Group Type** field to **sip**.
- Set the **IMS Enabled** field to **n**. This specifies 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 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 1                                              Page   1 of   1
                                SIGNALING GROUP

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


   Near-end Node Name: procr                    Far-end Node Name: 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 1 was configured using the parameters highlighted below.

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

```
change trunk-group 1                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: SIP Trunk to SP            COR: 1        TN: 1        TAC: *01
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 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 1                                                 Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

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

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

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

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

```
add trunk-group 1                                                  Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                              Maintenance Tests? y


                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y



                                 Modify Tandem Calling Number: no



 Show ANSWERED BY on Display? y
```

On **Page 4**, set the **Network Call Redirection** field to **y**. This allows inbound calls transferred back to the PSTN to use the SIP REFER method, see **Reference [15]**. Set the **Send Diversion Header** field to **y**. This field provides additional information to the network if the call has been re-directed. This is 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 PAETEC.

Note: PAETEC's Enterprise Trunking Feature does not require the use of the Diversion header on re-directed calls. When using PAETEC's Enterprise Trunking Feature, set the **Send Diversion Header** field to **n**.

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

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


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

5.9. 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** command to create an entry for each DID. As an example, the following screen illustrates a conversion of DID number **7135553761** to extension **12001**.

```
change inc-call-handling-trmt trunk-group 1                    Page   1 of  30
                         INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 7135553761       10  12001
 public-ntwrk    10 7135553762       10  12002
 public-ntwrk    10 7135553763       10  12003
 public-ntwrk    10 7135553764       10  12004
 public-ntwrk
```

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 used to authenticate the caller.

In the sample configuration, four DID numbers were assigned for testing. These four numbers were assigned to the four extensions **12001**, **12002**, **12003** and **12004**. Thus, these same 10-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these four extensions.

```
change public-unknown-numbering 1                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                             Total
Ext Ext            Trk       CPN              CPN
Len Code           Grp(s)    Prefix           Len
                                                    Total Administered: 13
 5  1                                          5       Maximum Entries: 9999
 5  2                                          5
 5  3                                          5     Note: If an entry applies to
 5  4                                          5     a SIP connection to Avaya
 5  5                                          5     Aura(tm) Session Manager,
 5  6                                          5     the resulting number must
 5  7                                          5     be a complete E.164 number.
 5  8                                          5
 5  12001          1         7135553761       10
 5  12002          1         7135553762       10
 5  12003          1         7135553763       10
 5  12004          1         7135553764       10
```

Use the **change tandem-calling-party-num** command, to define the calling party number to send to the PSTN for tandem calls from SIP users.

In the example shown below, all calls originating from a 5-digit extension beginning with 13 and routed to trunk group 1 will result in a 10-digit calling number. For **Number Format**, use an applicable format, in this case **pub-unk**.

```
change tandem-calling-party-num                                 Page   1 of   8
                  CALLING PARTY NUMBER CONVERSION
                          FOR TANDEM CALLS
    CPN              Trk                                   Number
 Len Prefix          Grp(s)     Delete  Insert             Format

 5   13              1          5       7135553761         pub-unk
```

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:
```

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 **1** was used.
- **FRL**: Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **Pfx Mrk**: **1** The prefix mark (**Pfx Mrk**) of **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: PAETEC SIP TRK
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 1    0       1                                                     n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

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

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 PAETEC 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

  7135553761           10    10    10    12001                ext   y       n
  7135553762           10    10    10    12002                ext   y       n
  7135553763           10    10    10    12003                ext   y       n
  7135553764           10    10    10    12004                ext   y       n
                                                                            n
                                                                            n
                                                                            n
                                                                            n
                                                                            n
                                                                            n
                                                                            n
                                                                            n
                                                                            n
```

5.12. Saving Communication Manager Configuration Changes

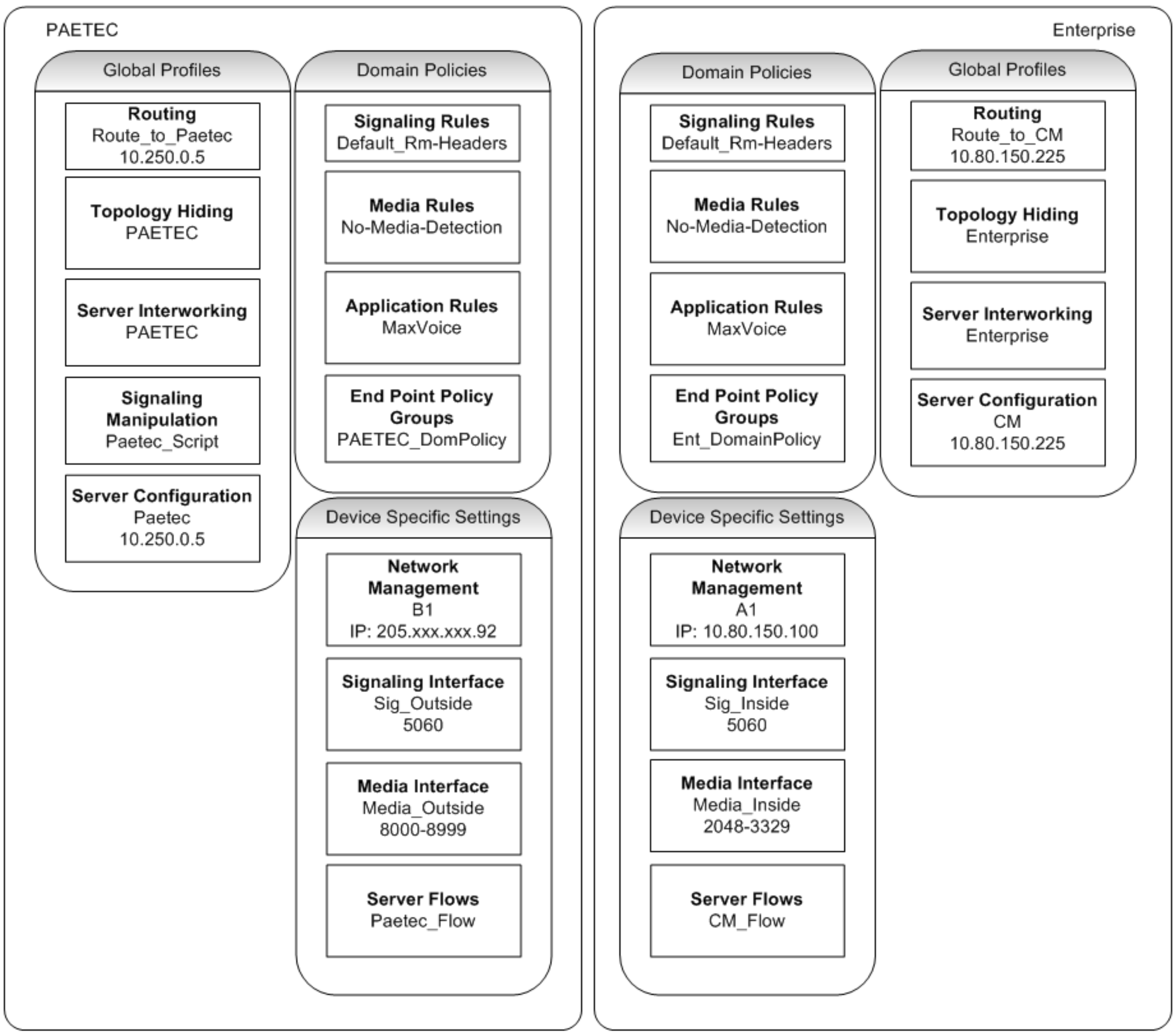
The command **save translation all** can be used to save the configuration.

```
save translation all

                               SAVE TRANSLATION

           Command Completion Status                                Error Code

           Success                                                  0
```

6. Configure Avaya Session Border Controller for Enterprise

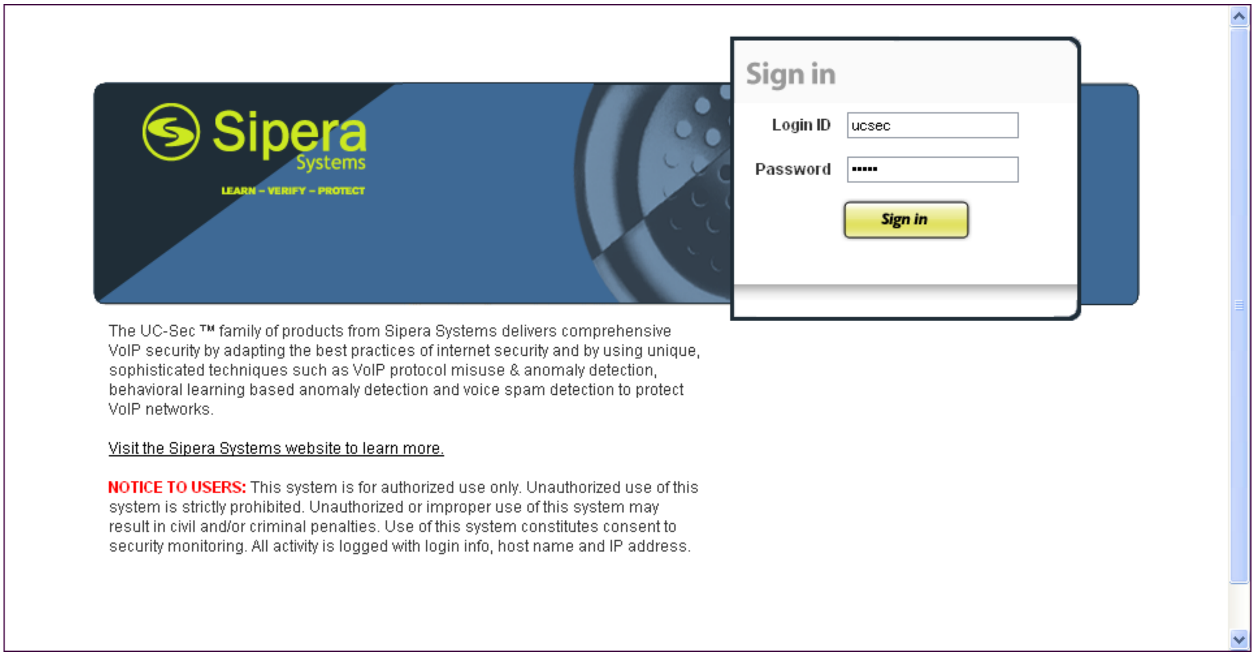
This section covers the configuration of Avaya Session Border Controller for Enterprise (Avaya SBCE). It is assumed that the software has already been installed. For additional information on these configuration tasks, see **Reference [12]** and [**13**].

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.

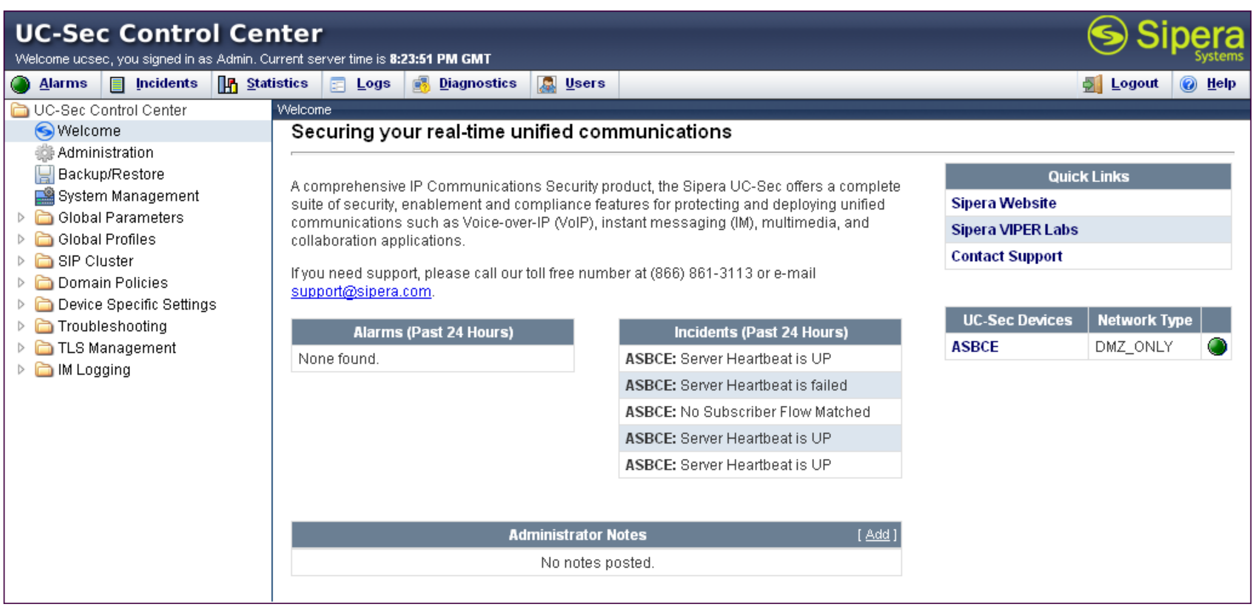


Use a WEB browser to access the UC-Sec web interface, enter https://<ip-addr>/ucsec in the address field of the web browser, where <ip-addr> is the management LAN IP address of UC-Sec.

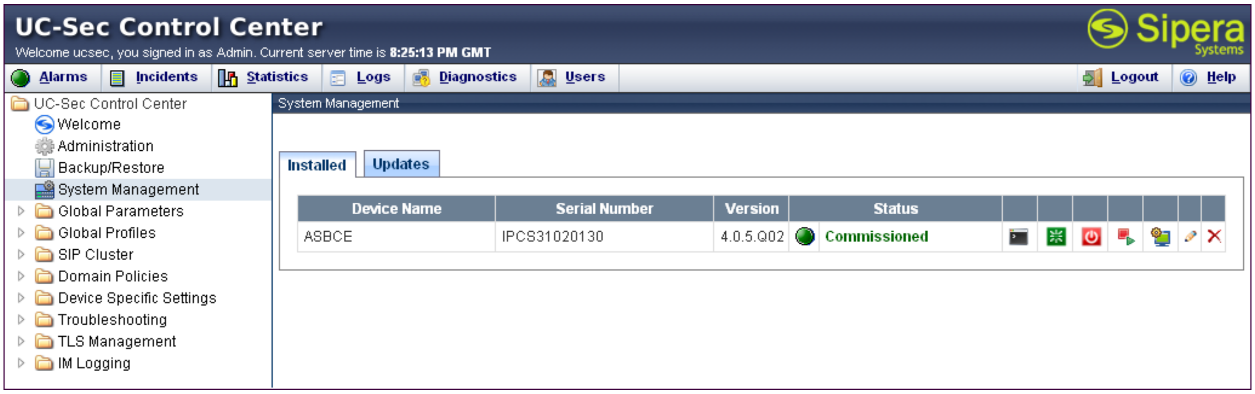
Log in with the appropriate credentials. Click **Sign In**.



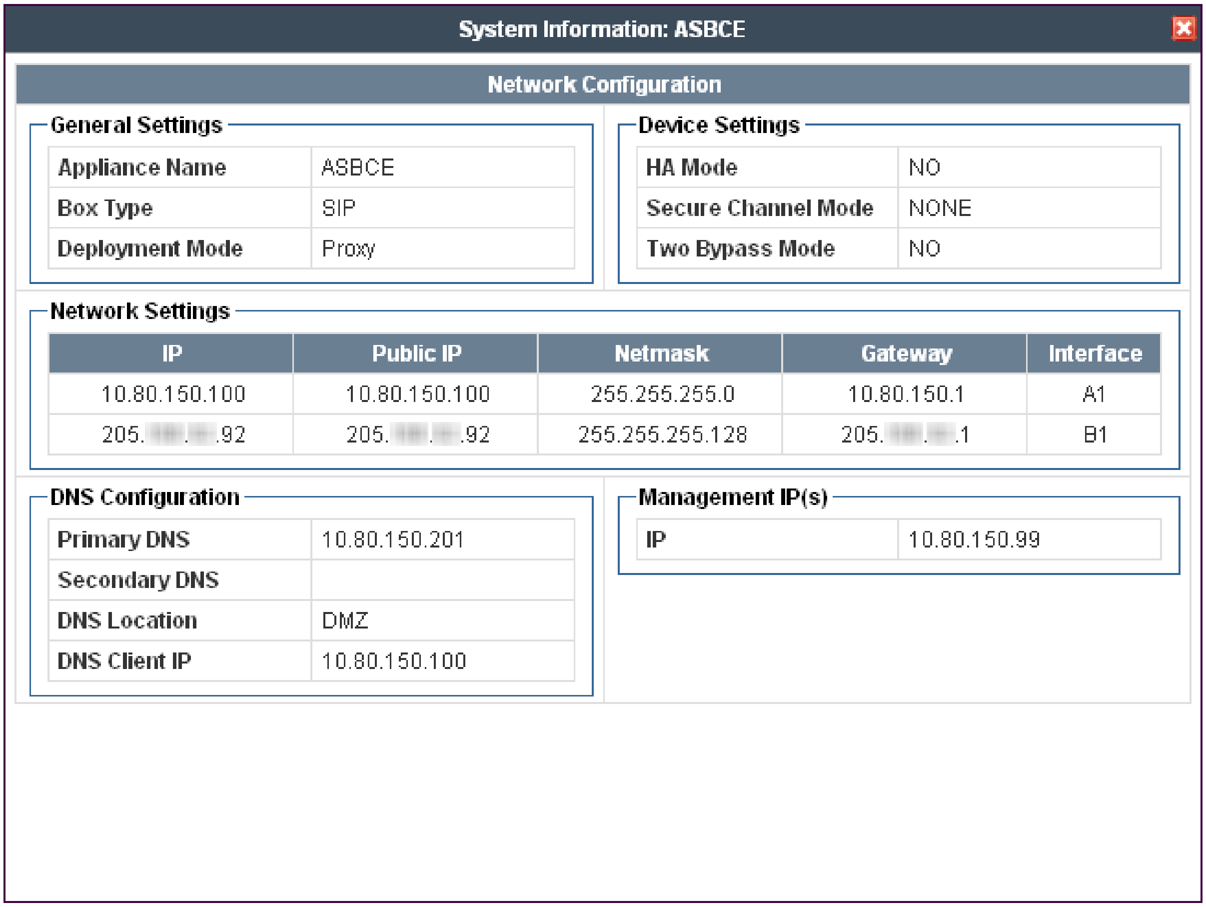
The main page of the UC-Sec Control Center will appear.



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.



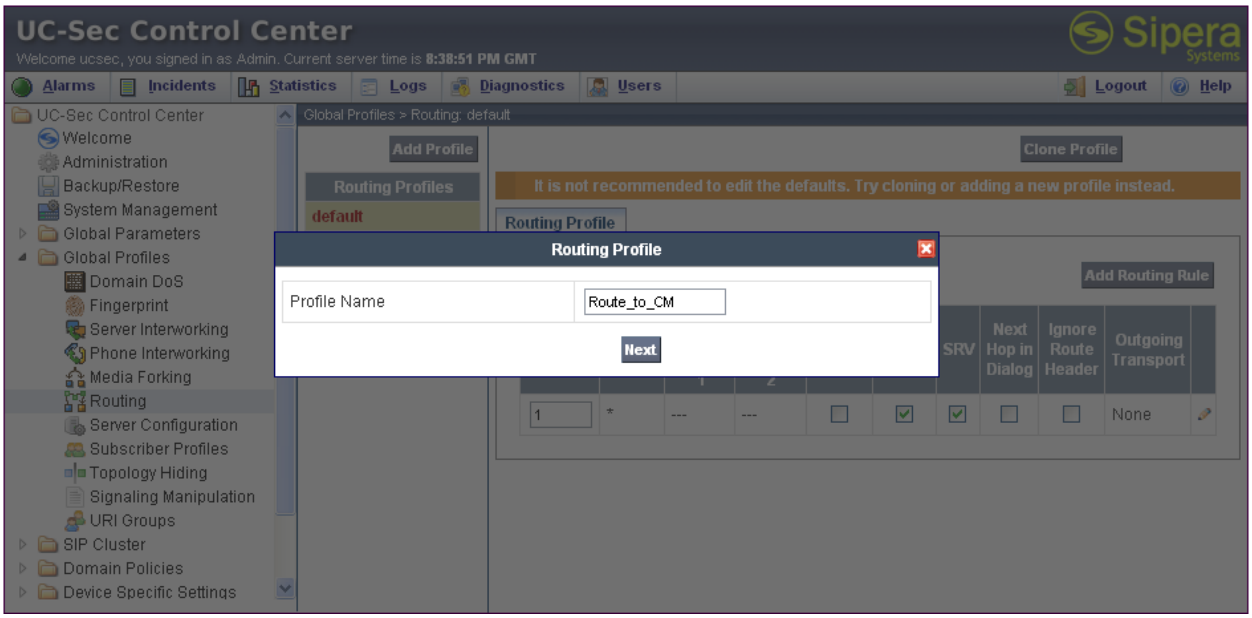
6.1. Global Profiles

Global Profiles allows 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 a Routing Profile for Communication Manager and PAETEC 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.

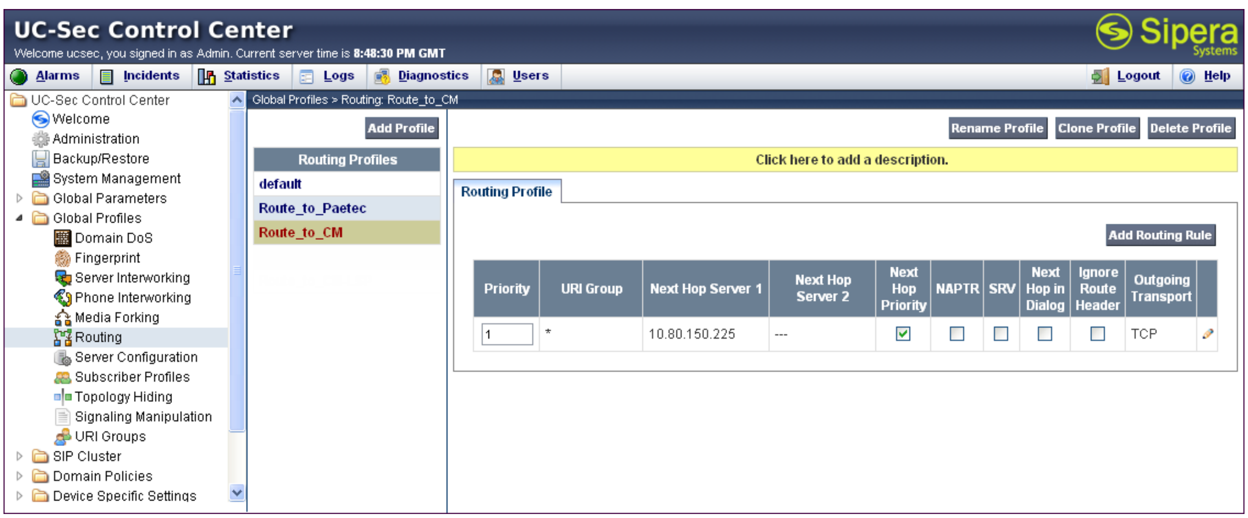


In the new window that appears, enter the following values. Use default values for all remaining fields:

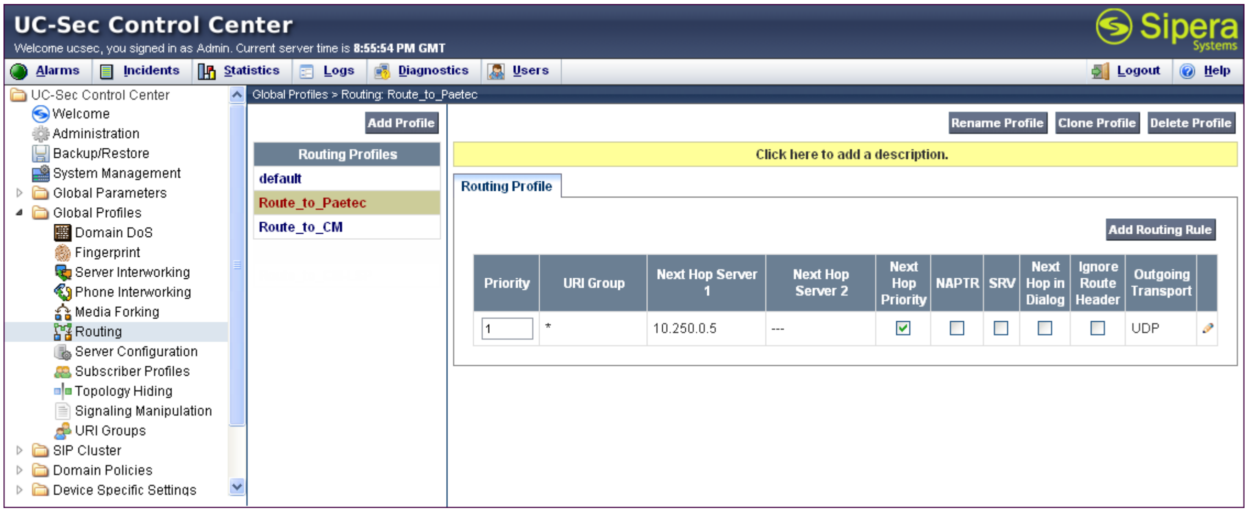
- **URI Group:** Select "*" from the drop down box.
- **Next Hop Server 1:** Enter the Domain Name or IP address of the Primary Next Hop server.
- **Next Hop Server 2:** (Optional) Enter the Domain Name or IP address of the secondary Next Hop server.
- **Routing Priority Based on Next Hop Server**: Checked.
- **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 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**.



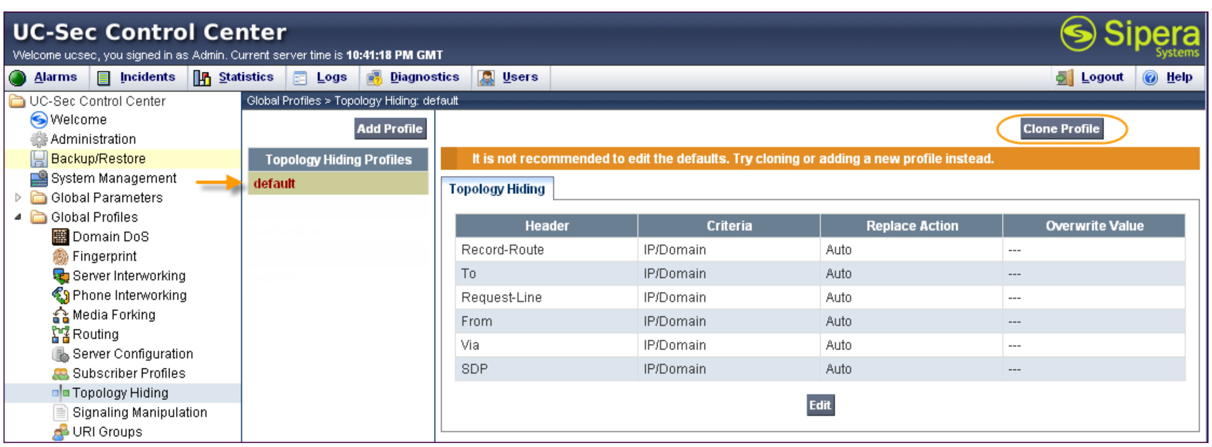
The following screen shows the Routing Profile to PAETEC. In the **Next Hop Server 1** field enter the IP address that PAETEC uses to listen for SIP traffic and the **Outgoing Transport** to **UDP**.



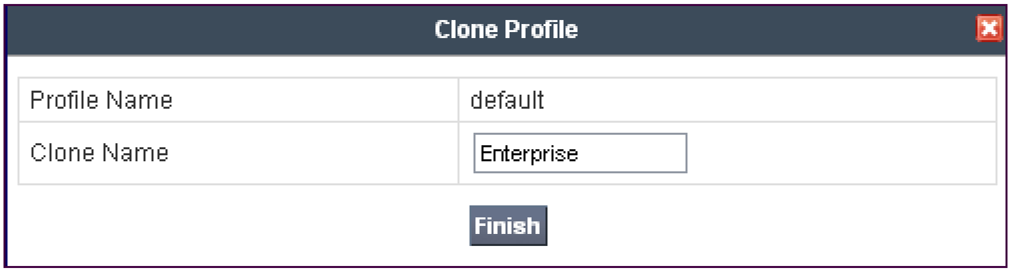
6.1.2. Topology Hiding Profile

The Topology Hiding profile manages how various source, destination 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.

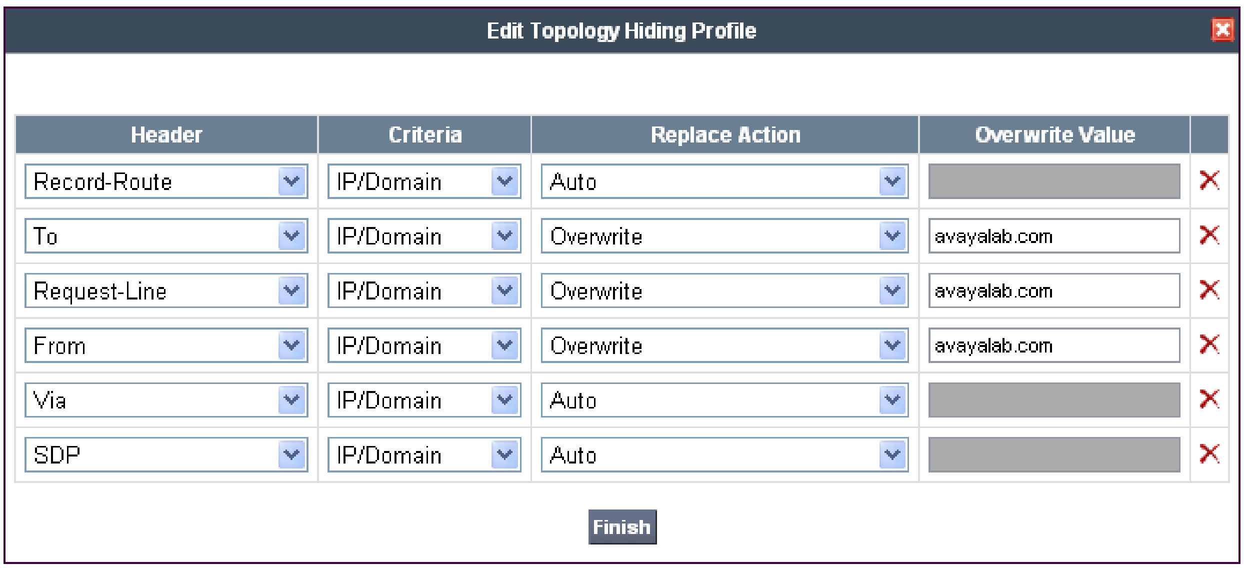
Create a Topology Hiding Profile for the enterprise and PAETEC Dynamic IP SIP Trunk. In the sample configuration, the **Enterprise** and **PAETEC** profiles were cloned from the default profile. To clone a default profile, navigate to **UC-Sec Control Center →Global Profiles → Topology Hiding**. Select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



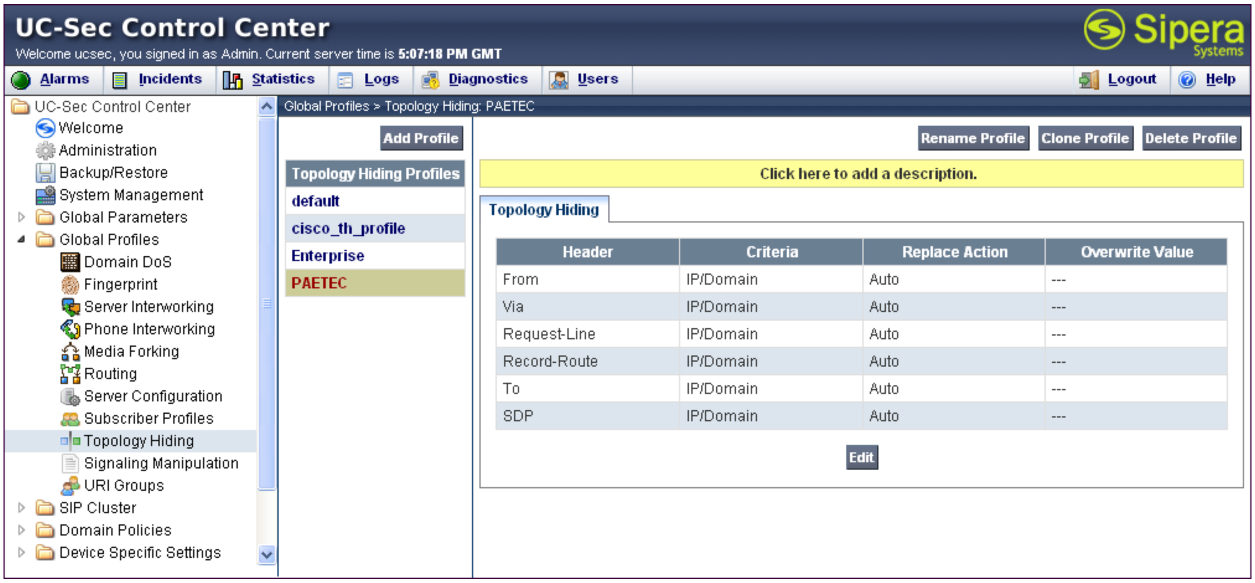
Enter a descriptive name for the new profile and click **Finish**.



Edit the **Enterprise** profile to overwrite the **To**, **Request-Line** and **From** headers shown below to the enterprise domain. 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.



It is not necessary to modify the **PAETEC** profile from the default values. The following screen shows the Topology Hiding Policy created for PAETEC.



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

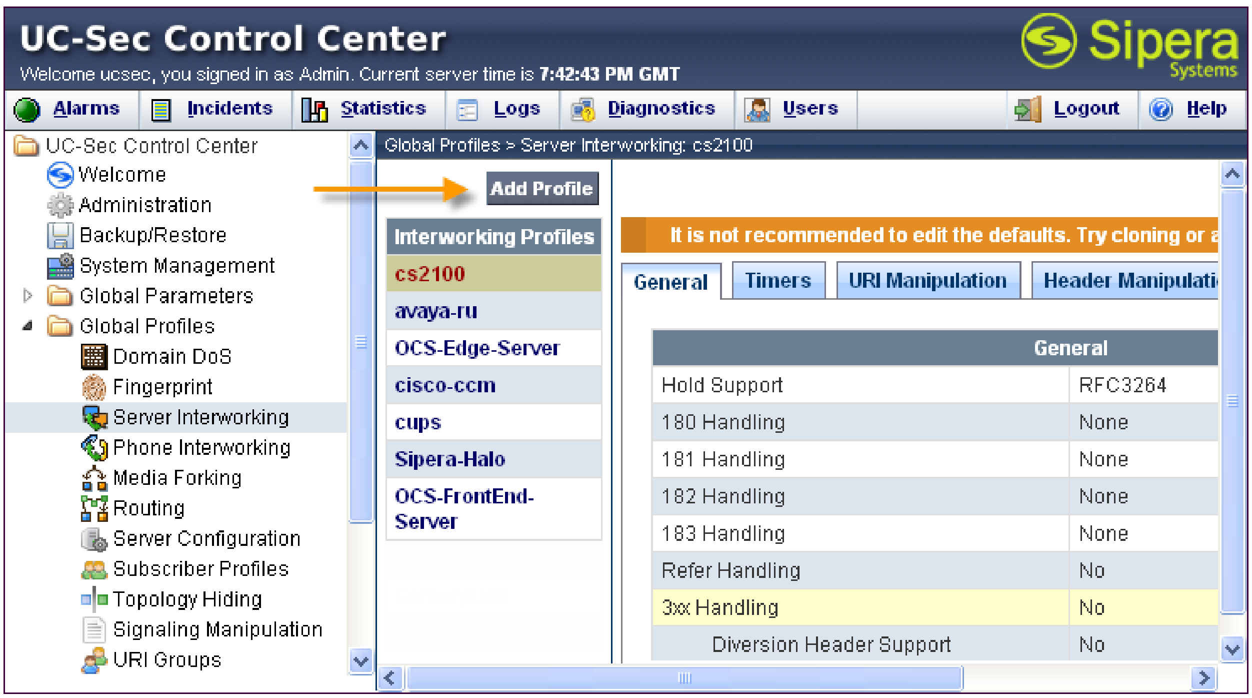
6.1.3. Server Interworking Profile

The Server Internetworking 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, or new profiles can be configured as described below.

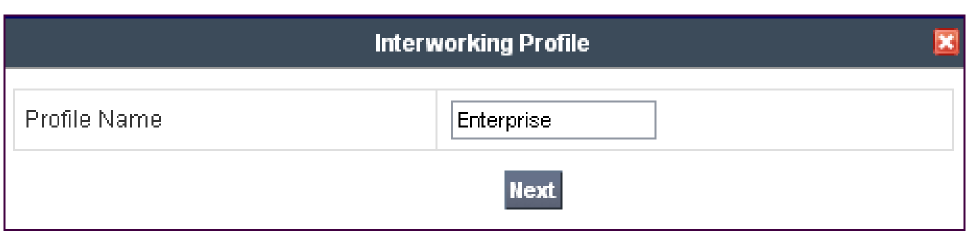
In the sample configuration, separate Server Interworking Profiles were created for **Enterprise** and **PAETEC**.

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.



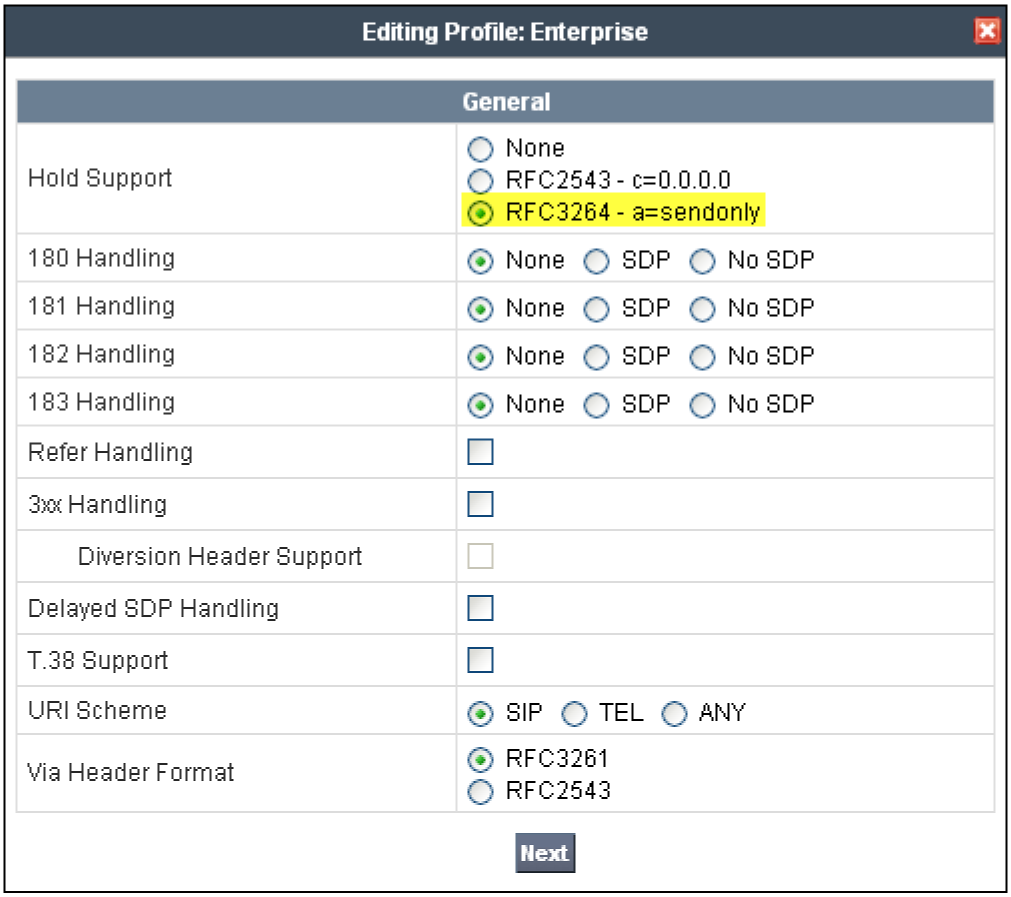
Enter a descriptive name for the new profile and click **Next** to continue.



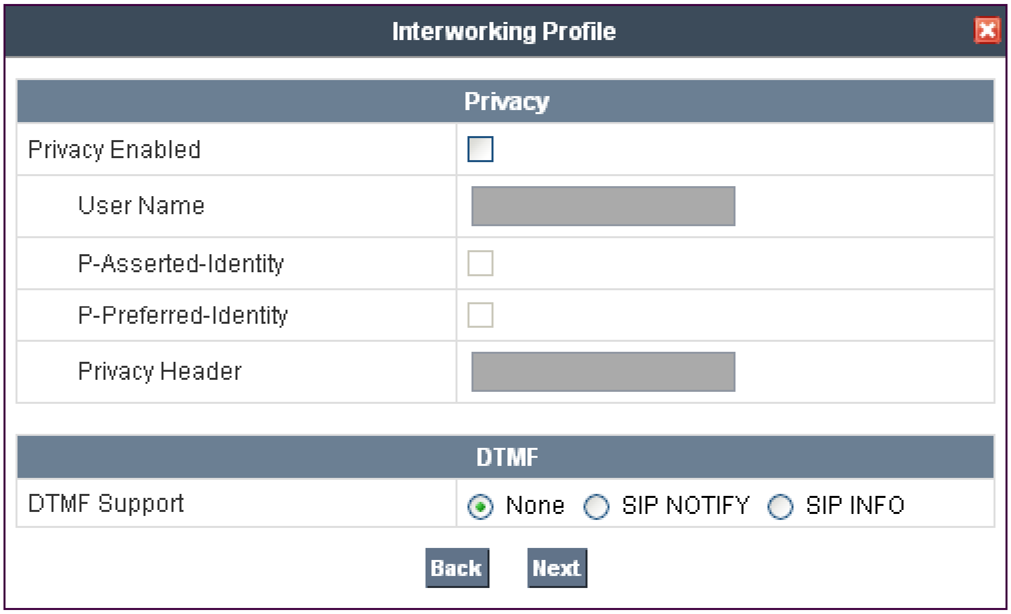
In the new window that appears, enter the following values. Use default values for all remaining fields:

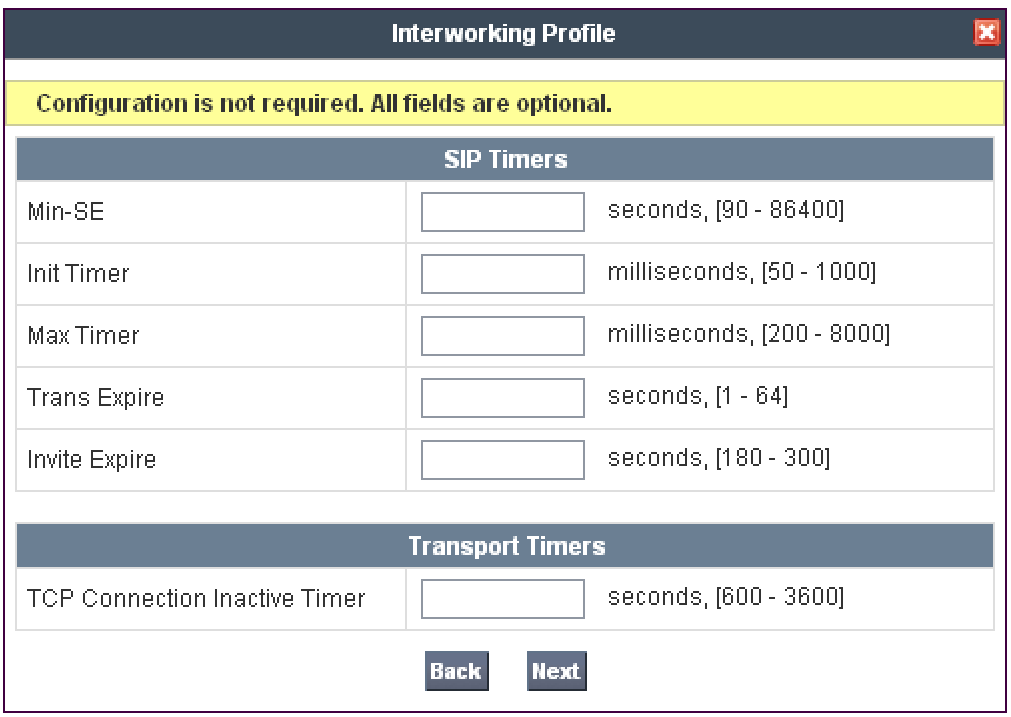
- **Hold Support:** Select **RFC3264**.

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



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

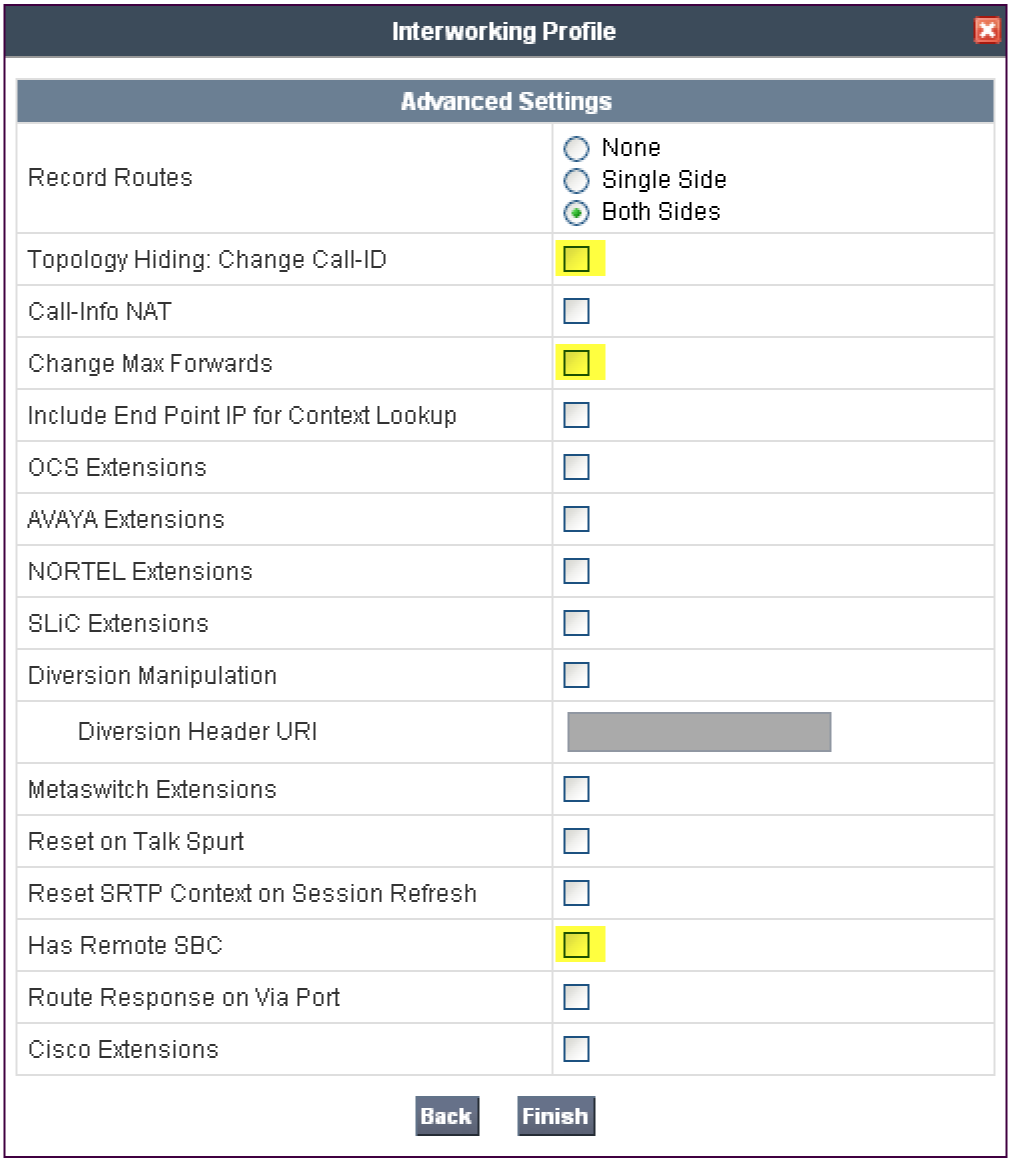




On the **Advanced Settings** window uncheck the following default settings:

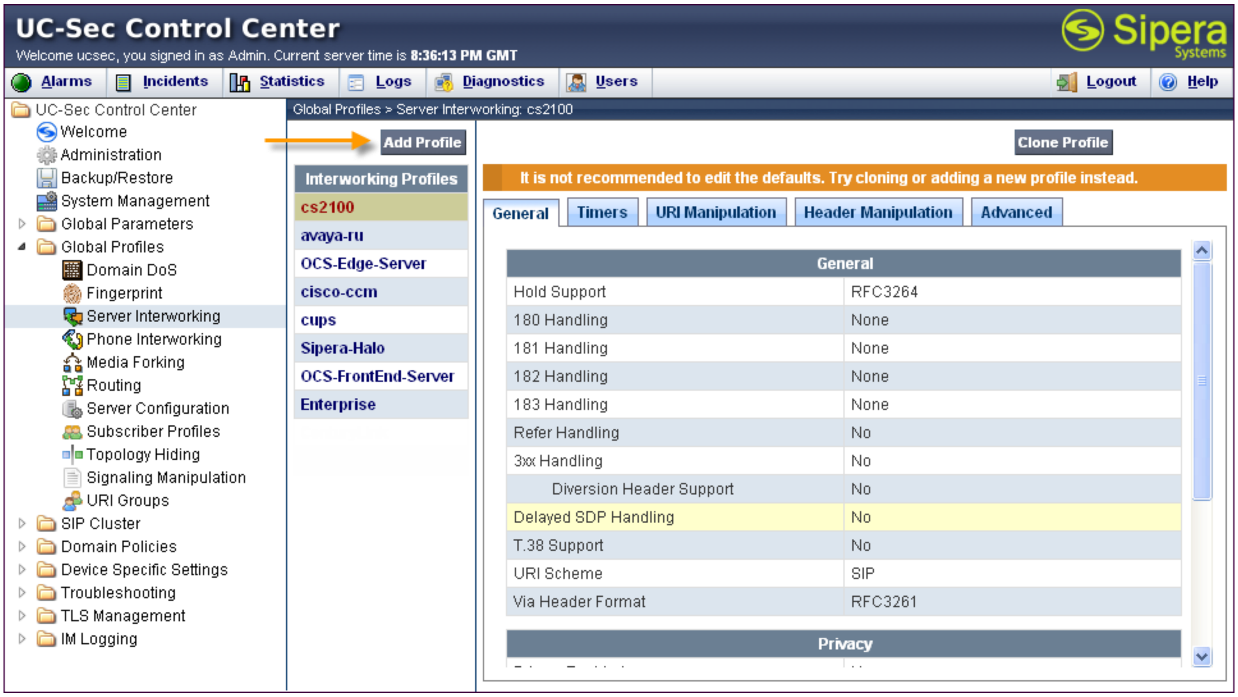
- **Topology Hiding: Change Call-ID**
- **Change Max Forwards**
- **Has Remote SBC**

Click **Finish** to save changes.

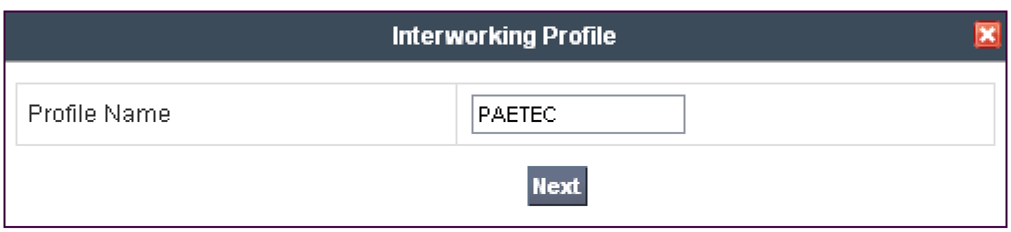


6.1.3.2 Server Interworking Profile – PAETEC

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

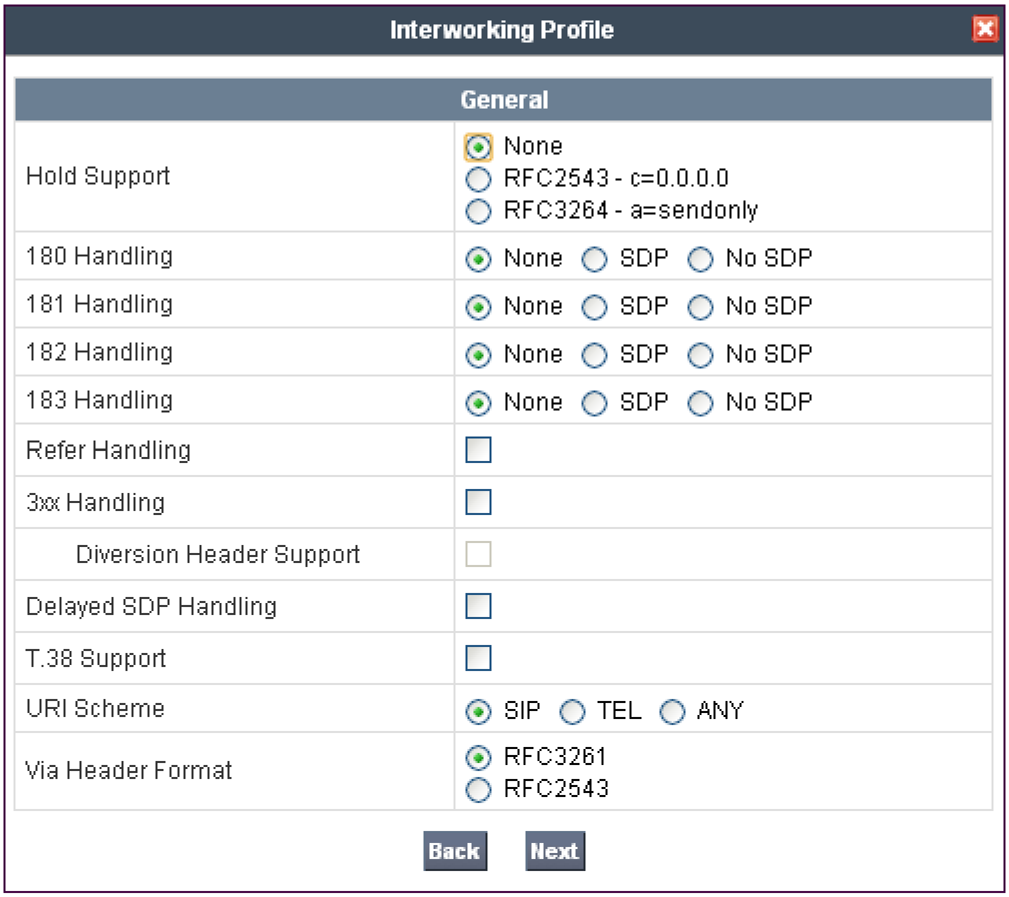


Enter a descriptive name for the new profile and click **Next** to continue.

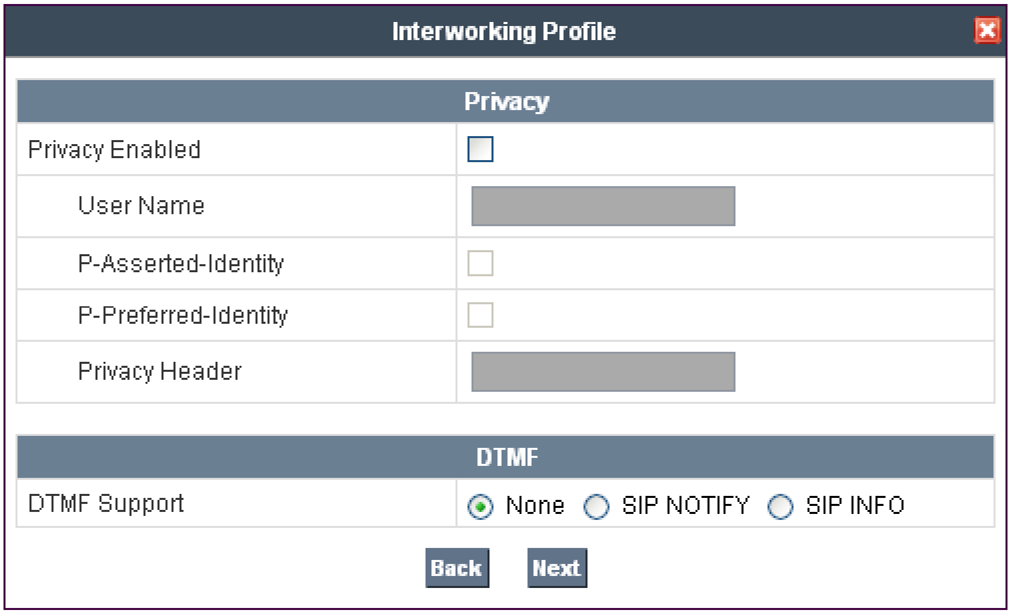


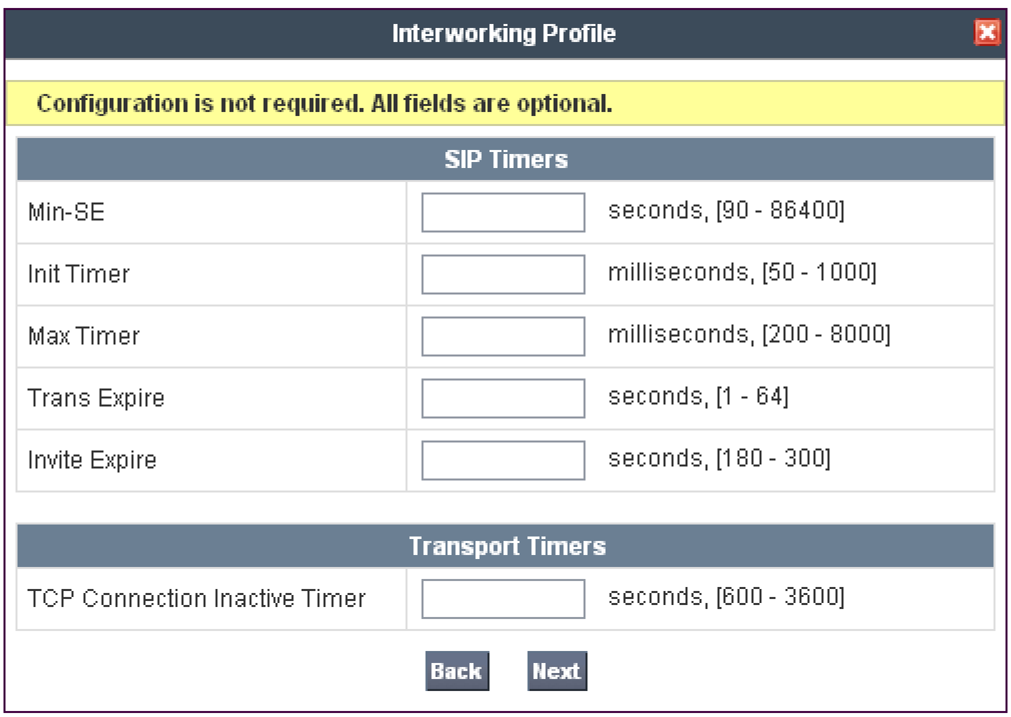
In the new window that appears, keep the default **Hold Support** value of **None**. PAETEC Dynamic IP SIP Trunk Service is not capable of supporting calls placed on hold by either the RFC 3264 method using the *a=sendonly* SDP attribute, nor the RFC 2543 method of setting the address in the *c=* SDP 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**.**

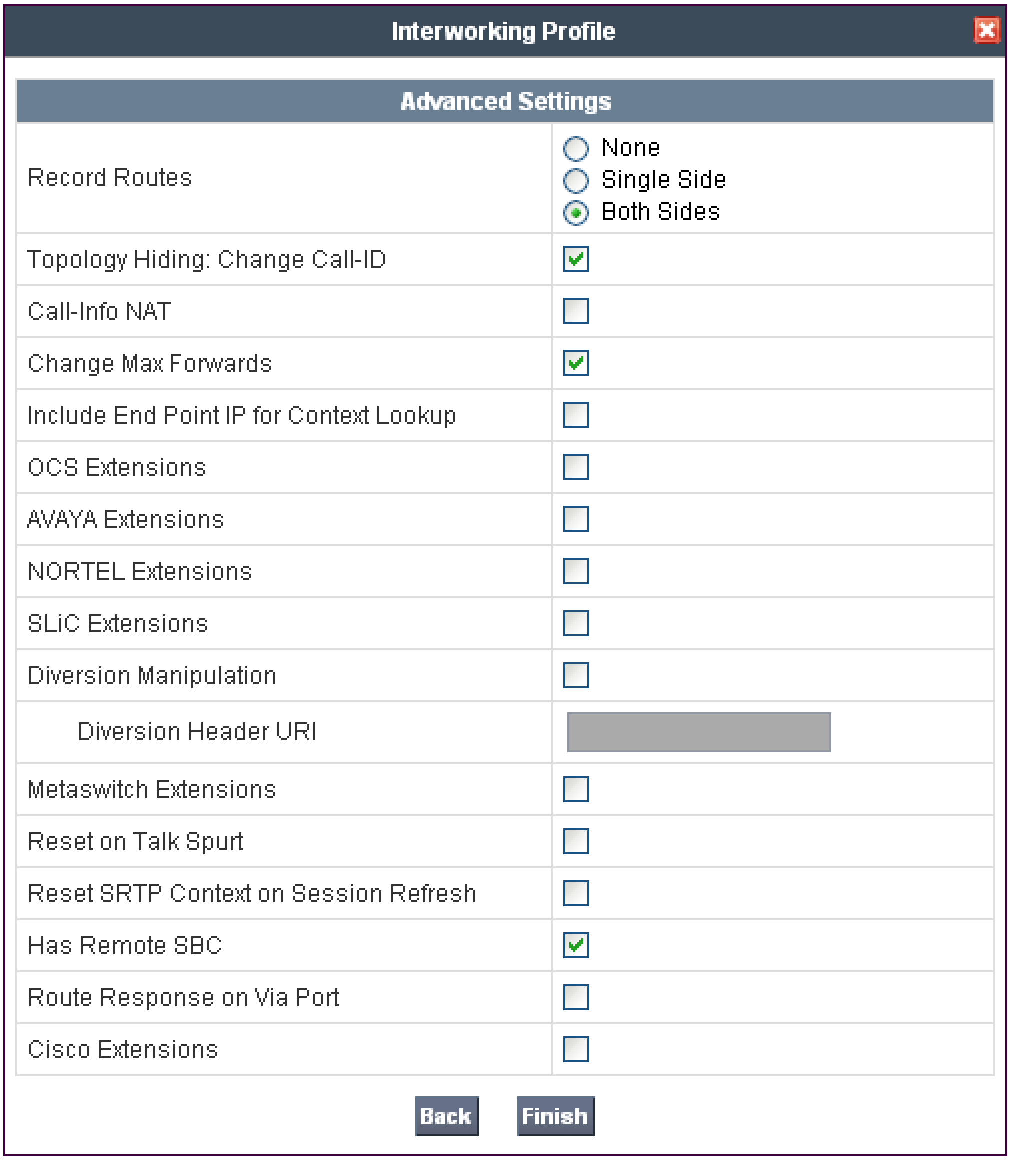


Default values can be used for the next two windows that appear. Click **Next** to continue.





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 add 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".

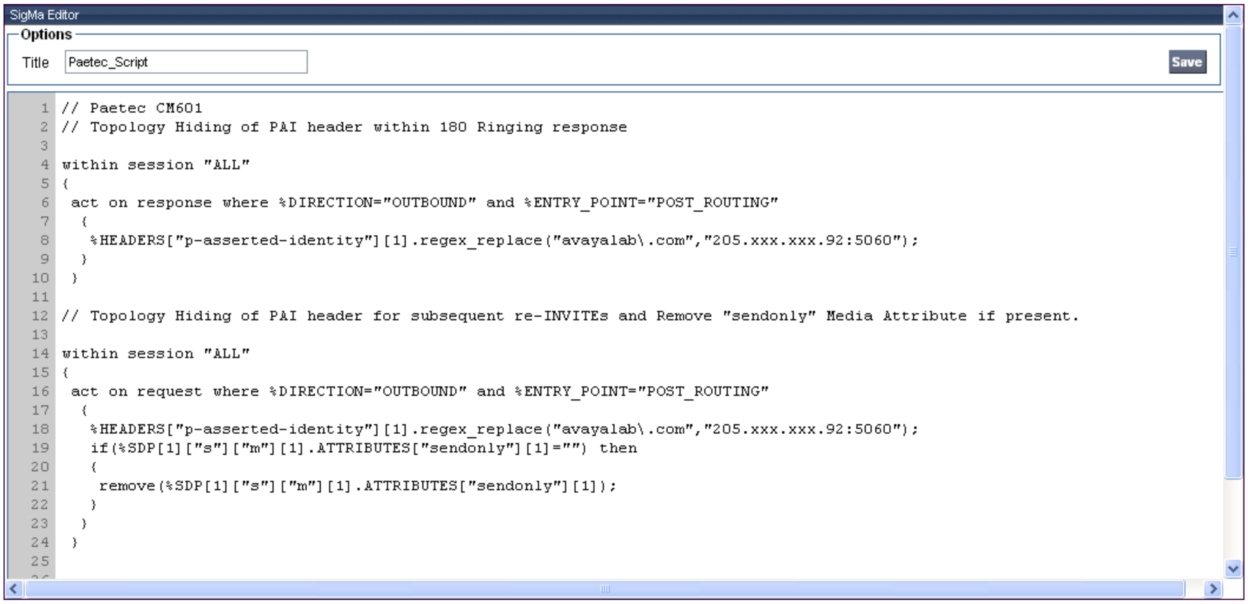
These application notes will not discuss the full feature of the Signaling Manipulation but will show an example of a script created during compliance testing to aid in topology hiding and to remove the *sendonly* media attribute sent by Communication Manager when a call is placed on hold. The PAETEC Dynamic SIP Trunk Service will stop receiving RTP packets when the *sendonly* media attribute is received resulting in no music or message being heard when a call is placed on hold. The *sendrecv* media attribute is assumed as the default for the session when no other attribute is sent. So rather than replacing *sendonly* with *sendrecv*, the *sendonly* media attribute was simply removed.

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]**.

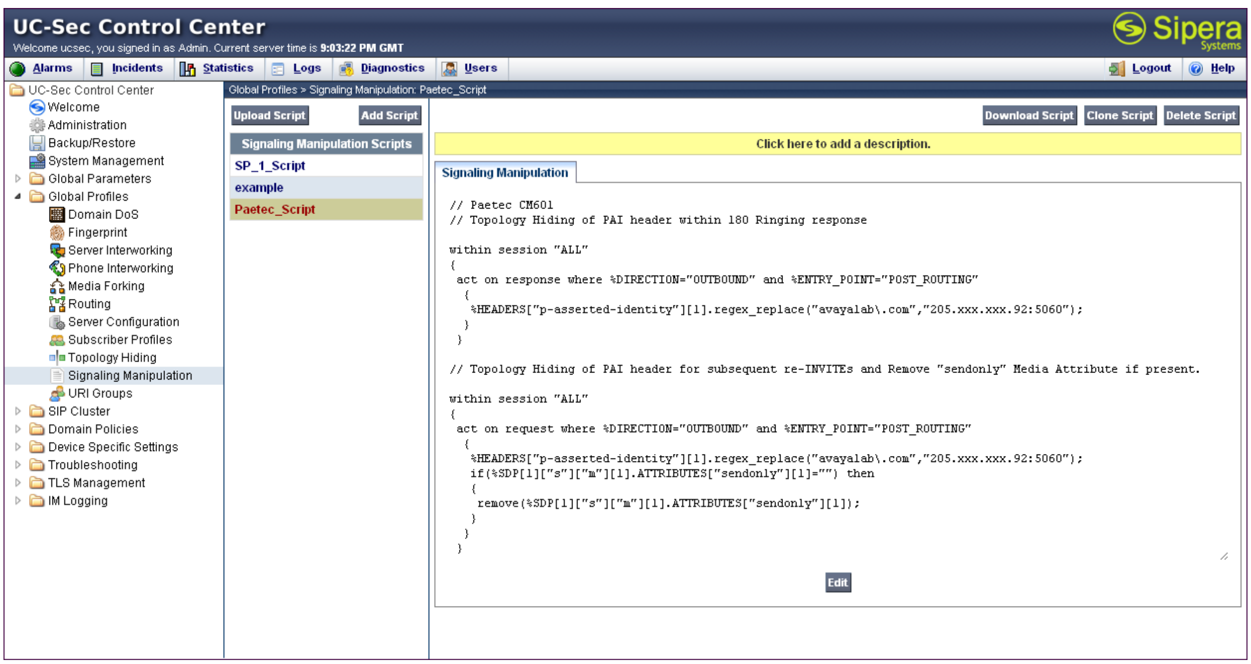
The following sample script is written in two sections. Each section begins with a comment describing what will take place in that portion of the script. The first section will act on the response of an inbound call from PAETEC (e.g., 180 Ringing and 200 OK) while the second acts on the request of an outbound call to PAETEC (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.92:5060")** — An action to replace a given match with the provide string (e.g., find "avayalab.com" and replace it with "205.xxx.xxx.92:5060".

The P-Asserted-Identity header will be modified by replacing the domain "avayalab.com" with the external IP address of Avaya SBCE and the SIP port of 5060 in both the response and request sessions. The SDP portion of the SIP message will be modified by removing the *sendonly* attribute if it is present.



The following screen shows the finished Signaling Manipulation Script **PAETEC_Script**.



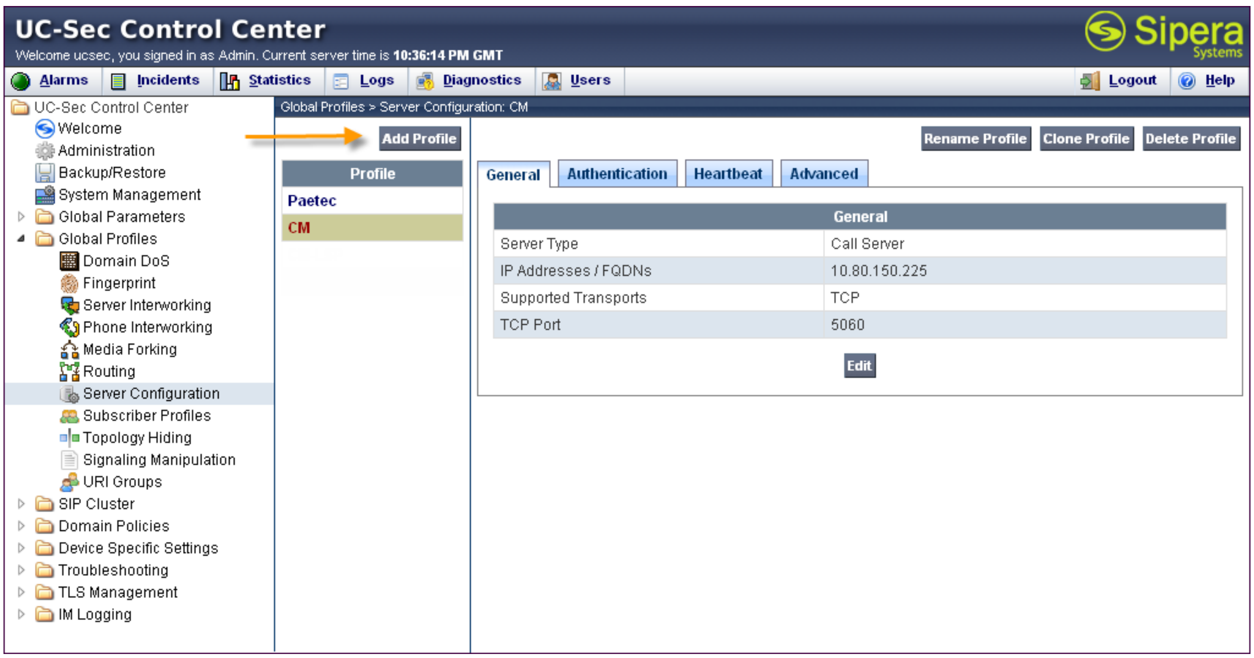
6.1.5. Server Configuration

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.

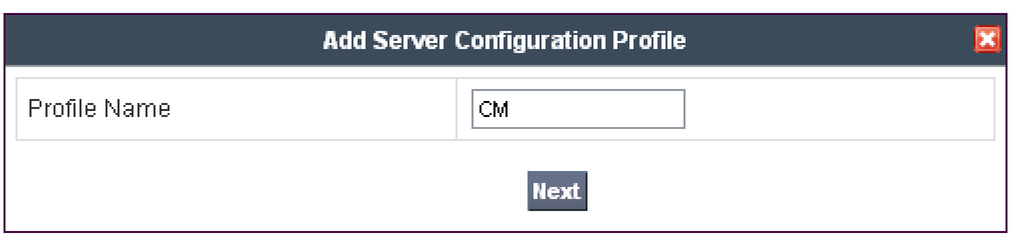
In the sample configuration, separate Server Configurations were created for **Session_Manager** and **PAETEC.**

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**.**



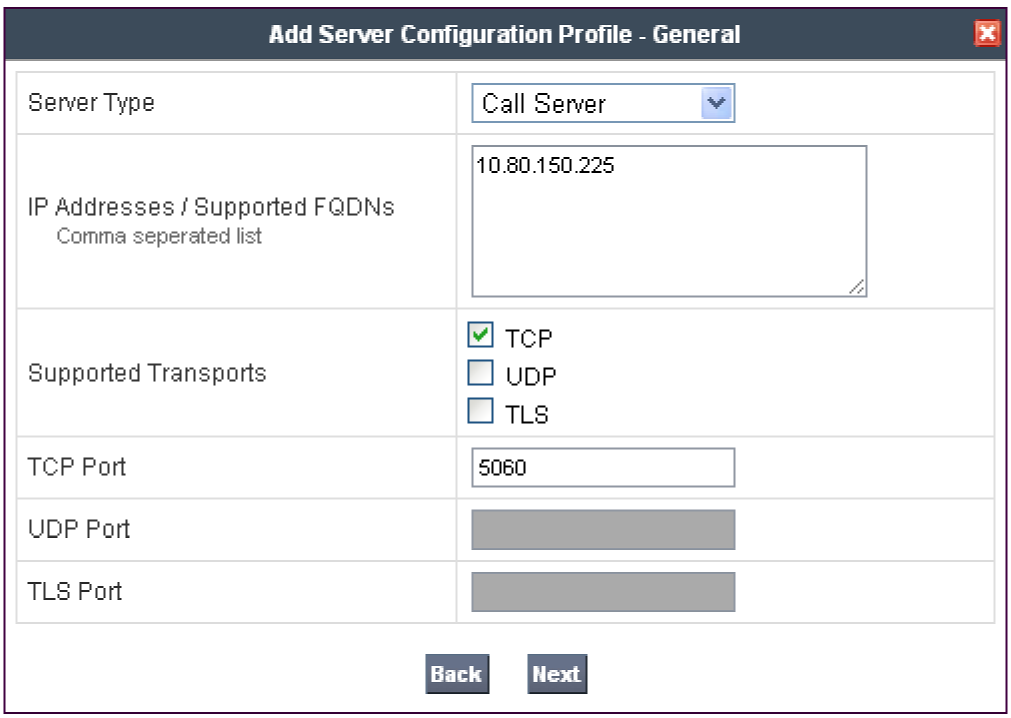
Enter a descriptive name for the new profile and click **Next**.



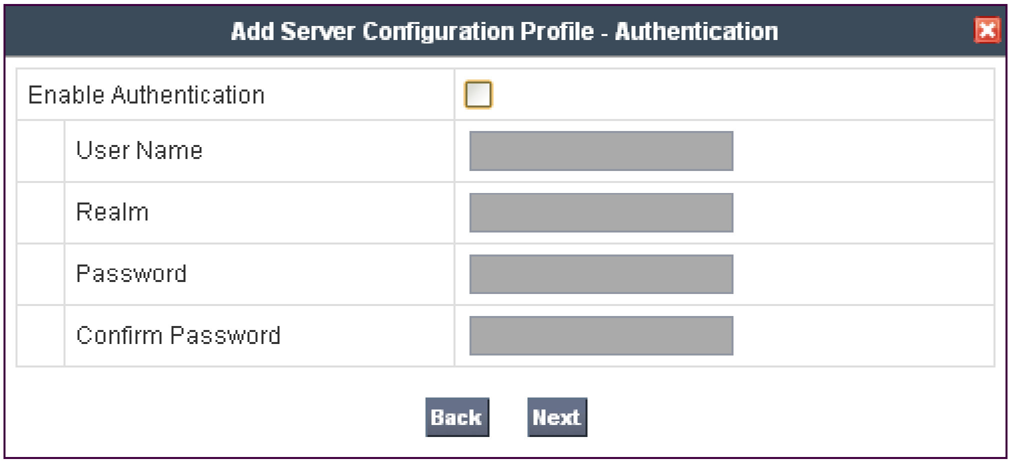
In the new window that appears, enter the following values. Use default values for all remaining fields:

- **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.



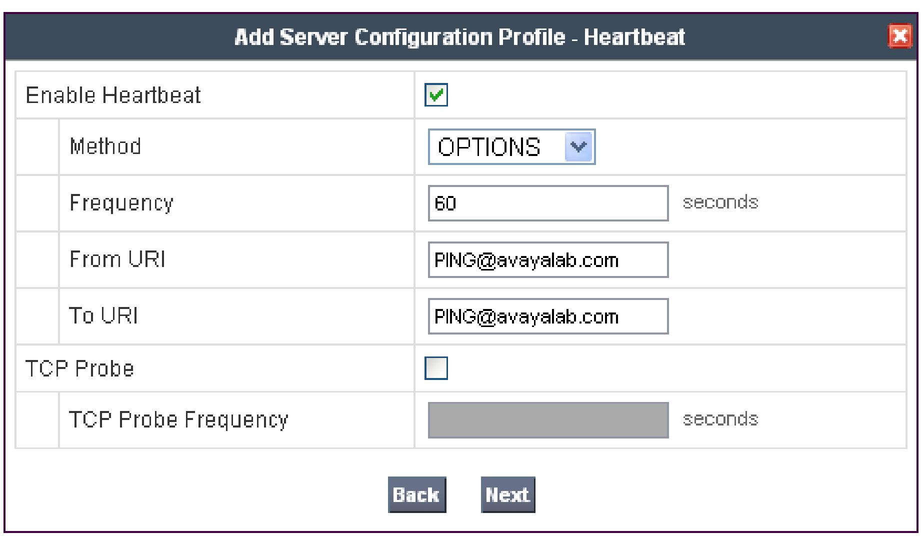
Verify **Enable Authentication** is unchecked as Communication Manager does not require authentication. Click **Next** to continue.



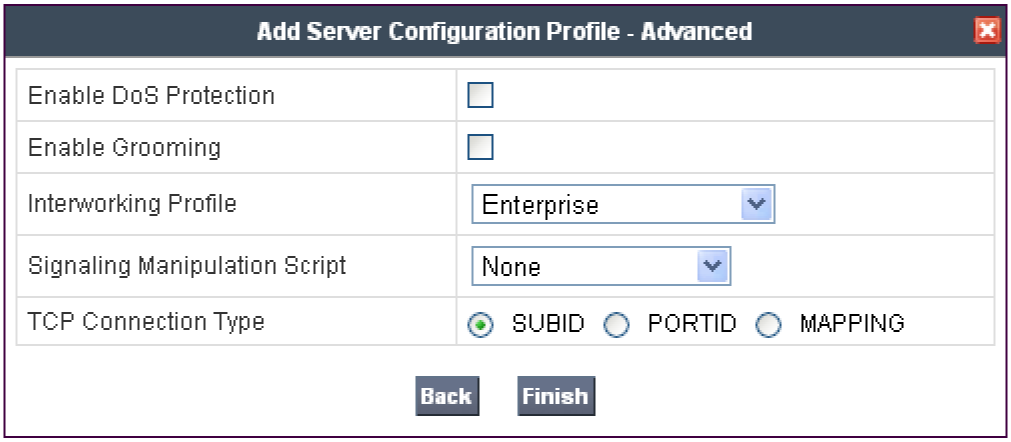
In the new window that appears, enter the following values. Use default values for all remaining fields:

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

Click **Next** to continue.

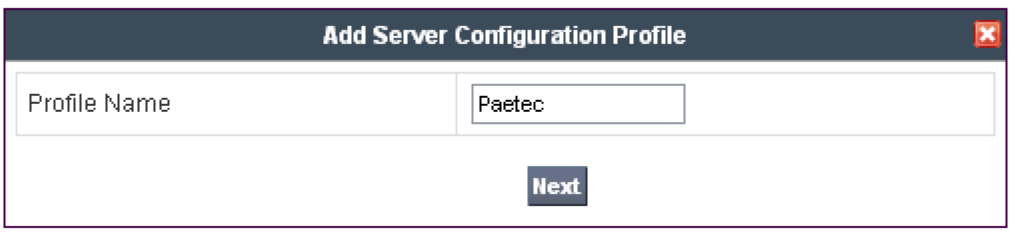


In the new window that appears, select the **Interworking Profile** created for the enterprise in **Section 6.1.3.1**. Use default values for all remaining fields. Click **Finish** to save the configuration.



6.1.5.2 Server Configuration - PAETEC

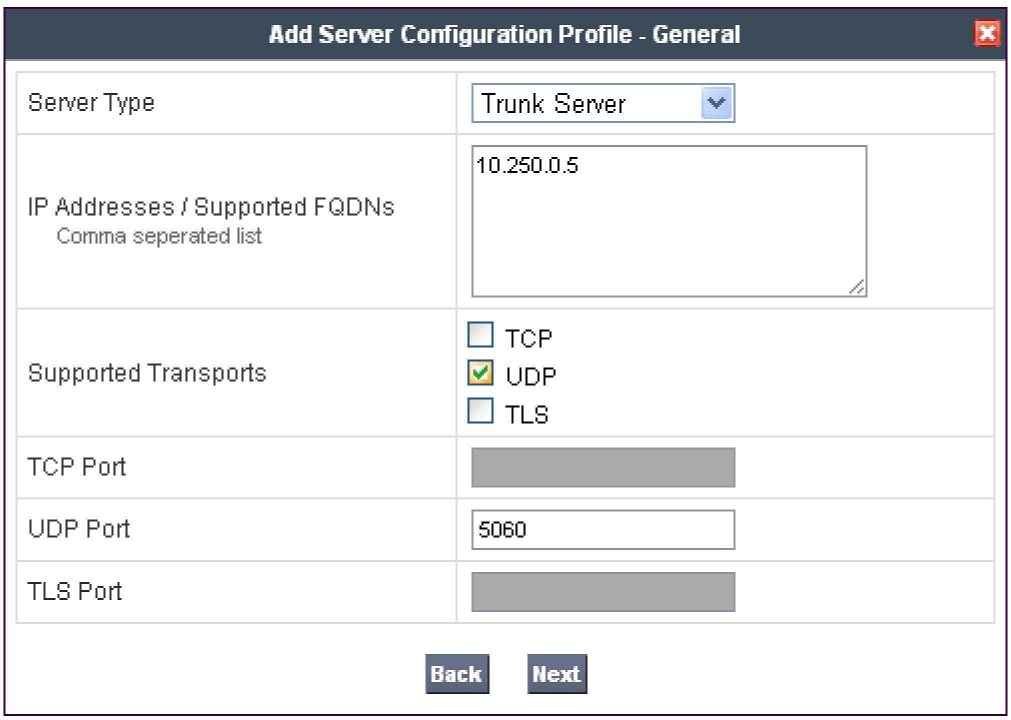
To add a Server Configuration Profile for PAETEC navigate to **UC-Sec Control Center →Global Profiles → Server Configuration** and click on **Add Profile** (not shown). Enter a descriptive name for the new profile and click **Next**.



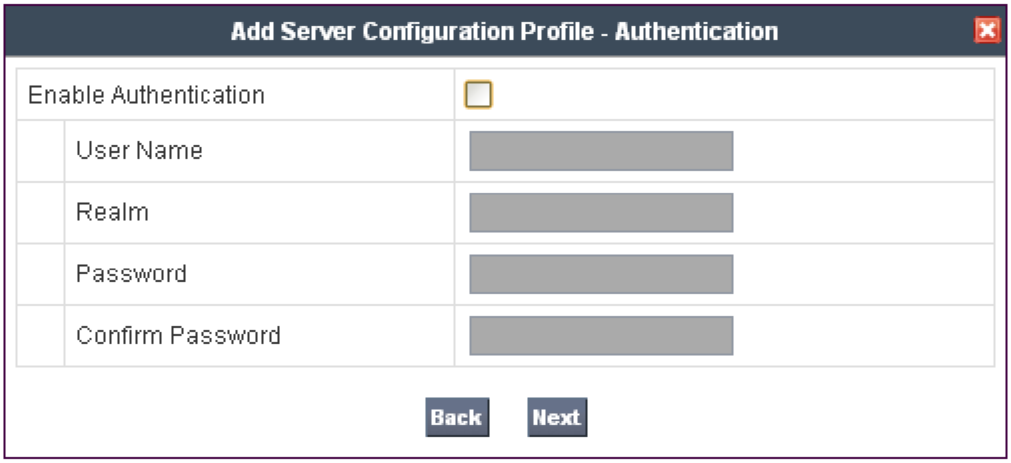
In the new window that appears, enter the following values. Use default values for all remaining fields:

- **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 PAETEC SIP Trunk IP address. This will associate the inbound SIP messages from PAETEC to this Sever Configuration.
- **Supported Transports:** Select the transport protocol to be used for SIP traffic between Avaya SBCE and PAETEC.
- **TCP Port:** Enter the port number that PAETEC uses to send SIP traffic.

Click **Next** to continue.



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

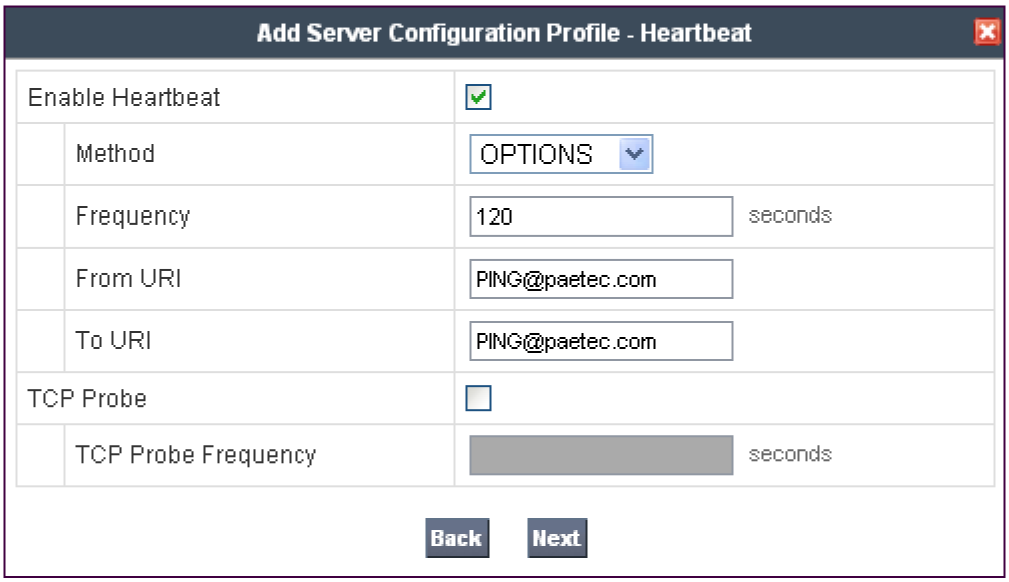


In the new window that appears, enter the following values. Use default values for all remaining fields:

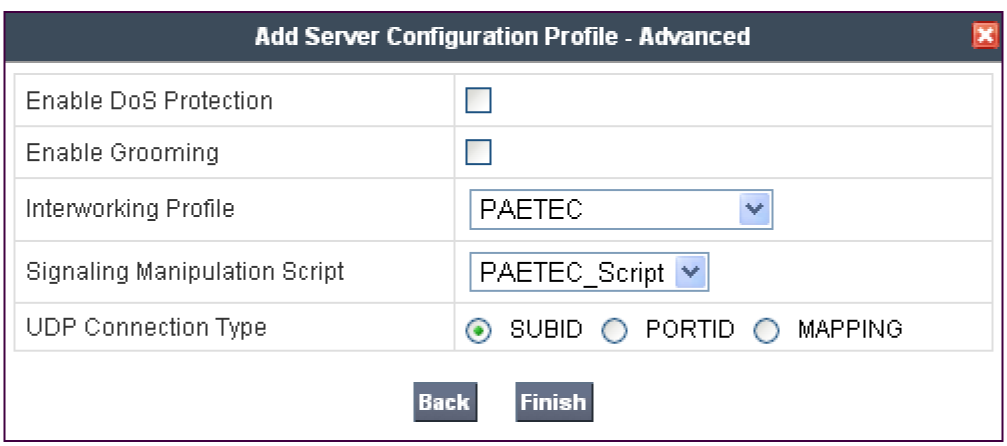
- **Enabled Heartbeat:** Checked.
- **Method:** Select **OPTIONS** from the drop-down box.
- **Frequency:** Choose the desired frequency in seconds the Avaya SBCE will send SIP OPTIONS. For compliance testing **120** seconds was chosen.
- **From URI:** Enter an URI to be sent in the FROM header for SIP OPTIONS.
- **TO URI:** Enter an 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@paetec.com was used in the sample configuration to better identify the SIP OPTIONS in the call traces. PAETEC 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).



In the new window that appears, select the **Interworking Profile** created for PAETEC 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.



6.2. Domain Policies

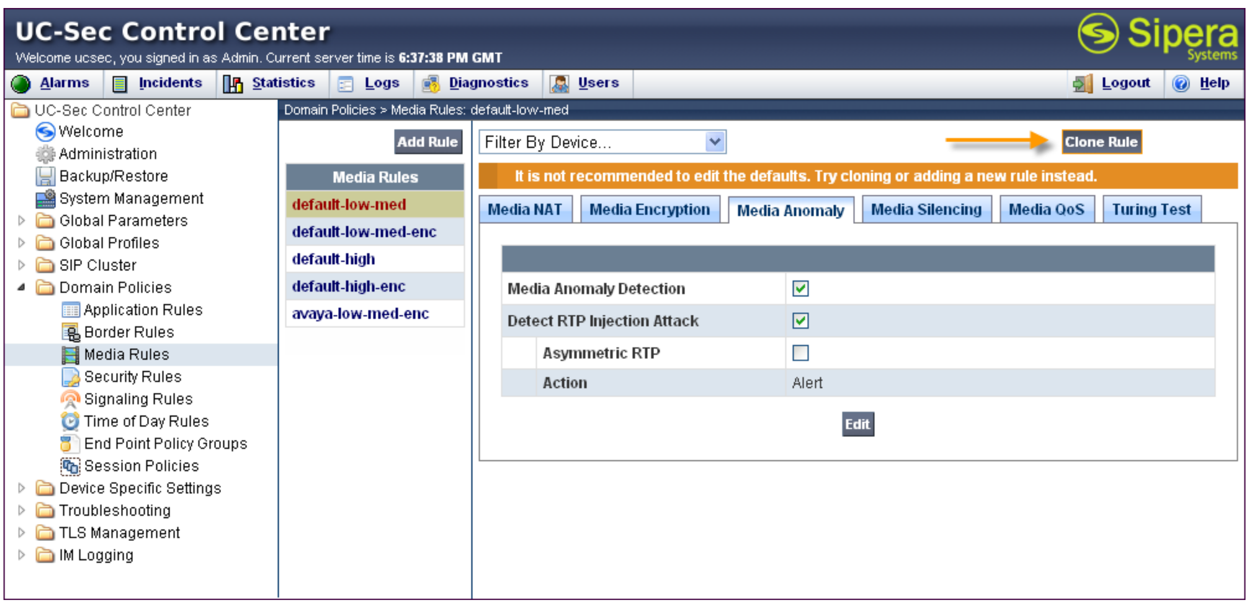
The Domain Policies feature configures, applies, and manages various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating in the enterprise. These criteria can be used to trigger policies which, in turn, activate 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

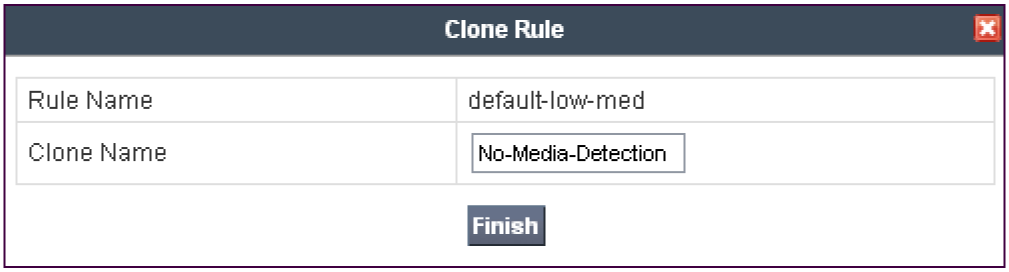
Media Rules define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be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

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.

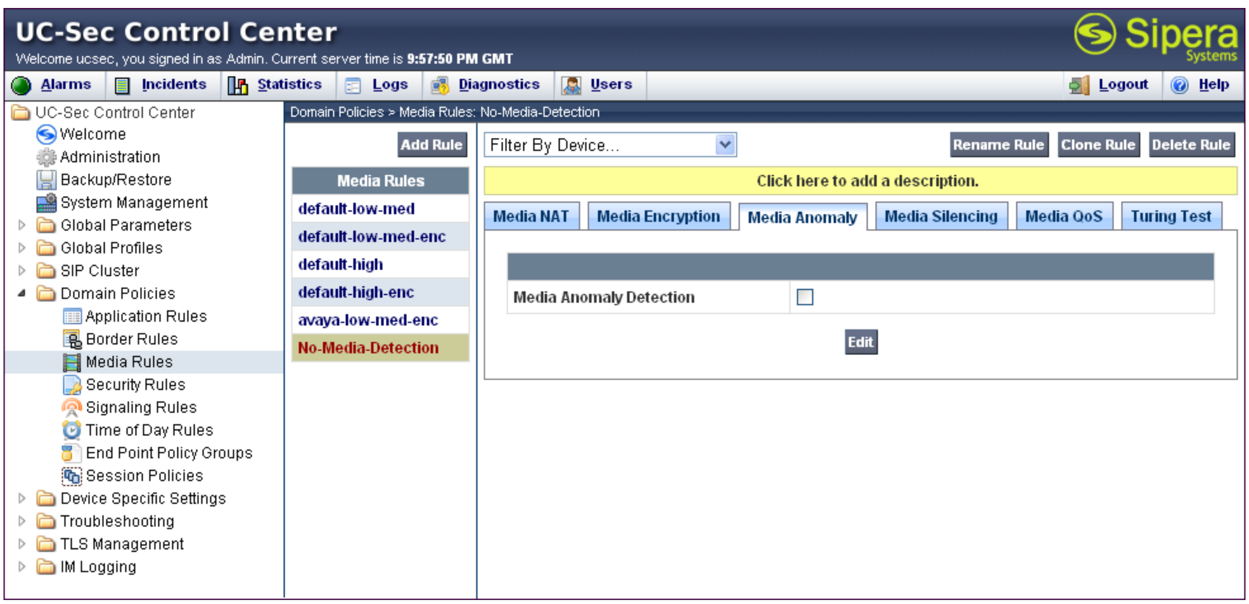


Enter a descriptive name for the new rule and click **Finish**.

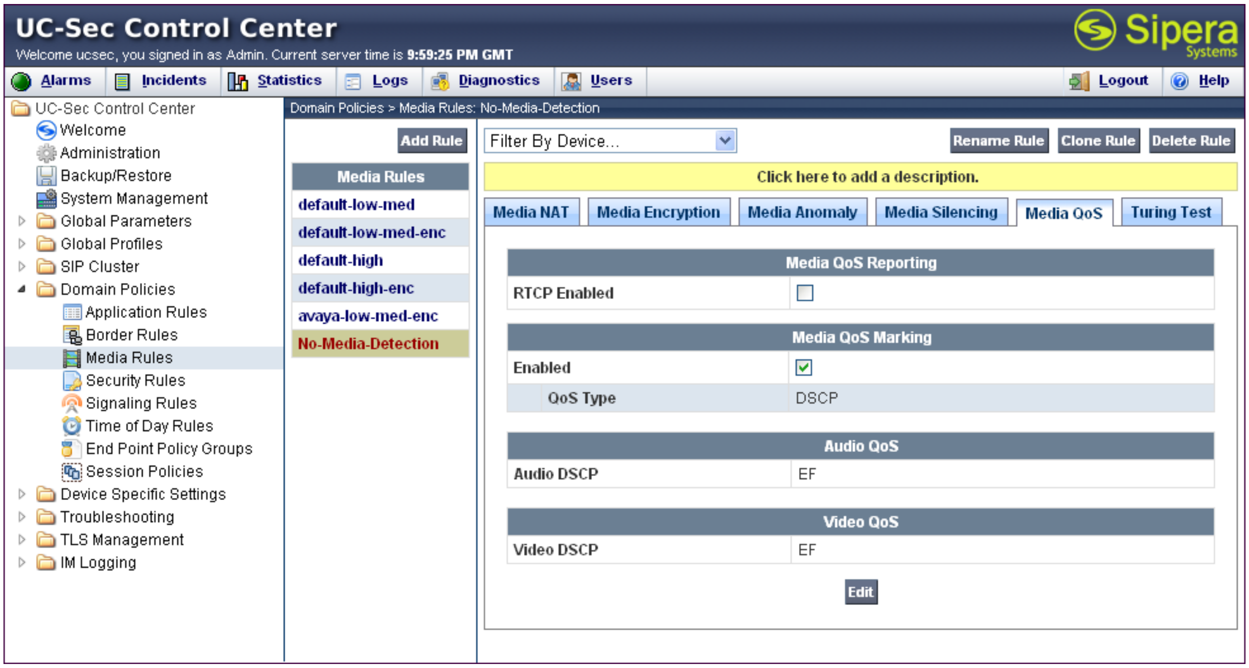


When the RTP packets of a call are shuffled from Communication Manager to an IP Phone, Avaya SBCE will interpret this as an anomaly and an alert will be created in the Incidents Log. Disabling **Media Anomaly Detection** prevents the **RTP Injection Attack** alerts from being created during an audio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



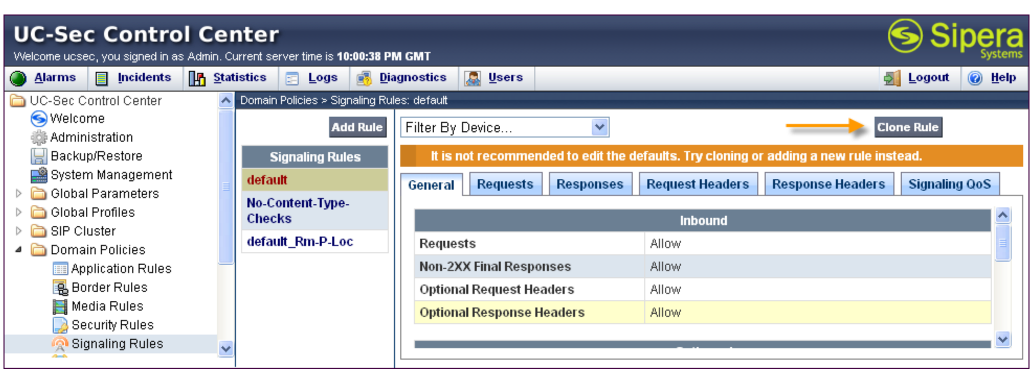
On the **Media QoS** tab, select the proper Quality of Service (QoS). Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for the media. The following screen shows the QoS values used for compliance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



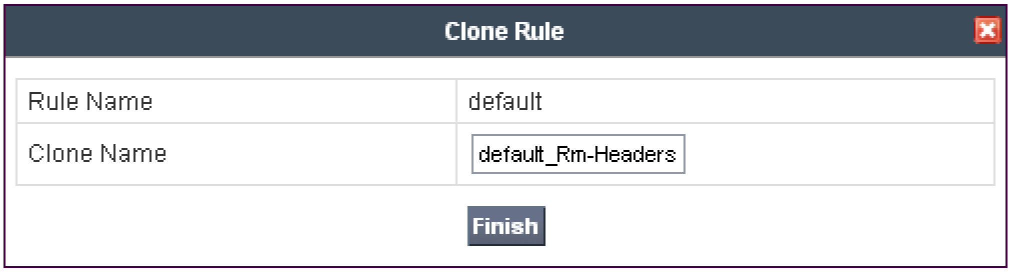
6.2.2. Signaling Rules

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 before it is sent to PAETEC. 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.



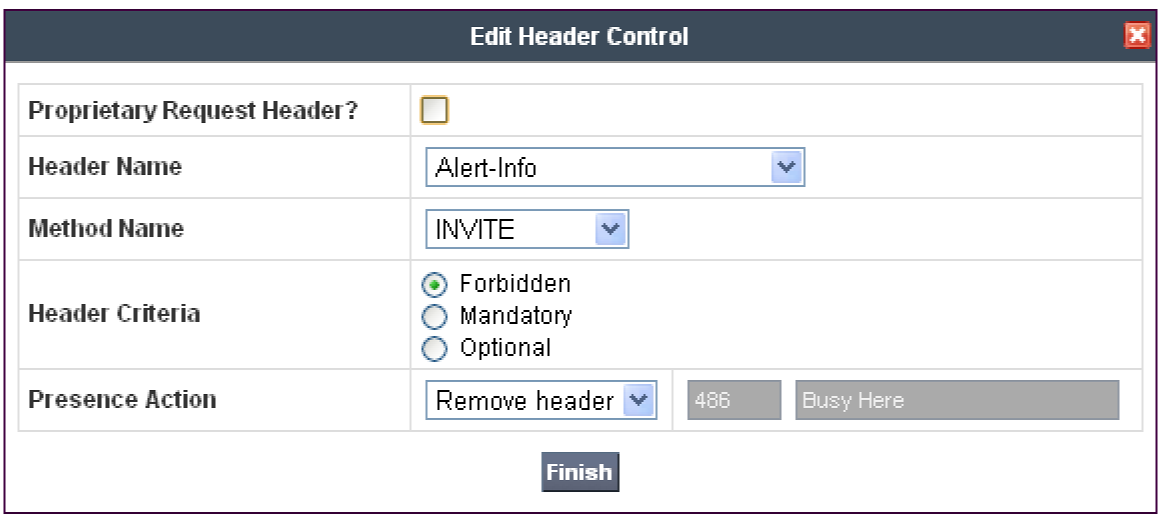
Enter a descriptive name for the new rule and click **Finish**.



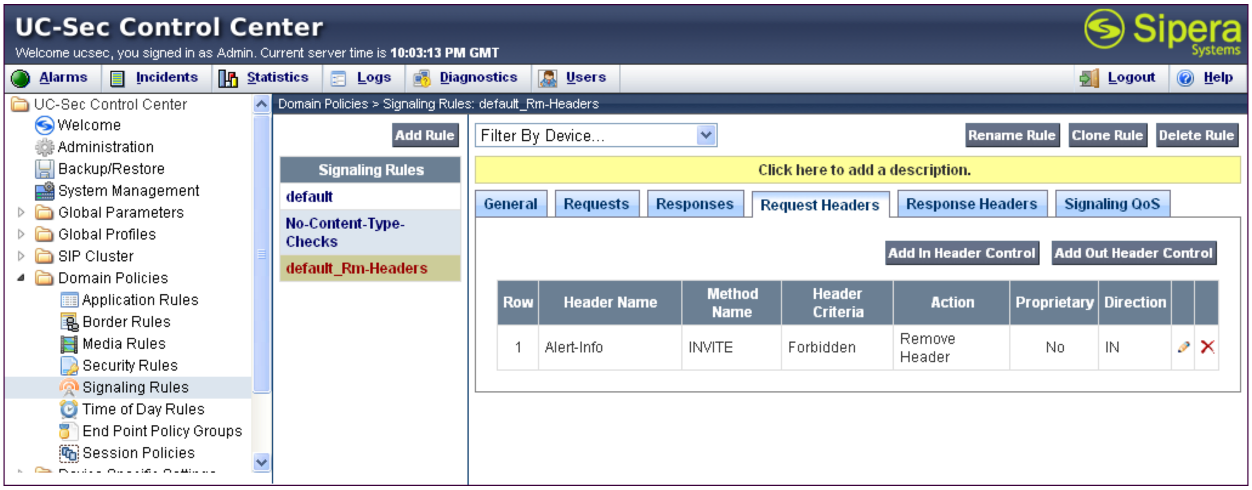
Select the **Request Headers tab** and click **Add In Header Control** (not shown). In the new window that appears, enter the following values. Use default values for all remaining fields:

- **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.

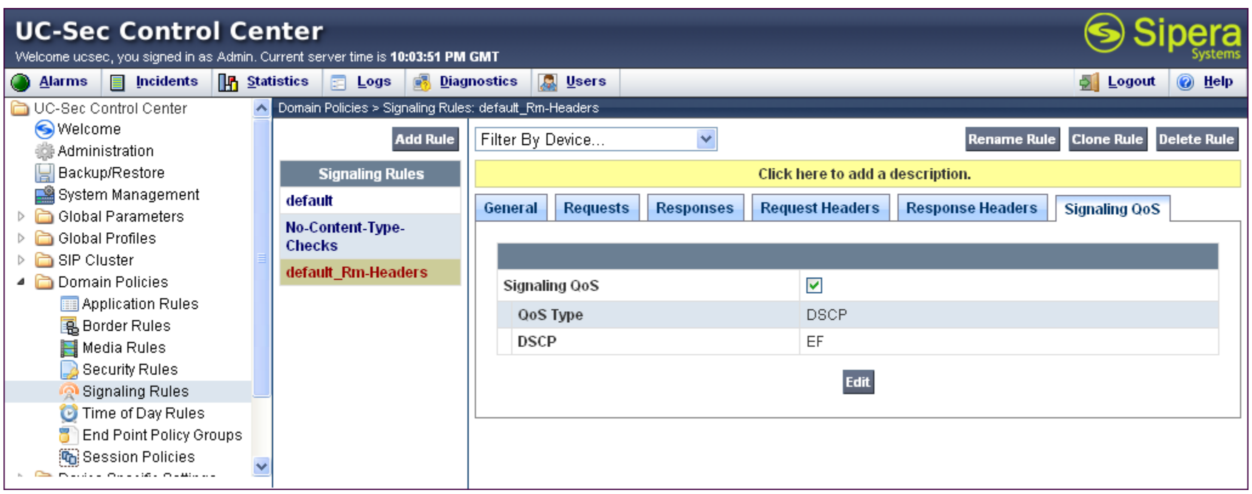
Click **Finish** to save the configuration



Repeat these steps for any other headers wished to 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.



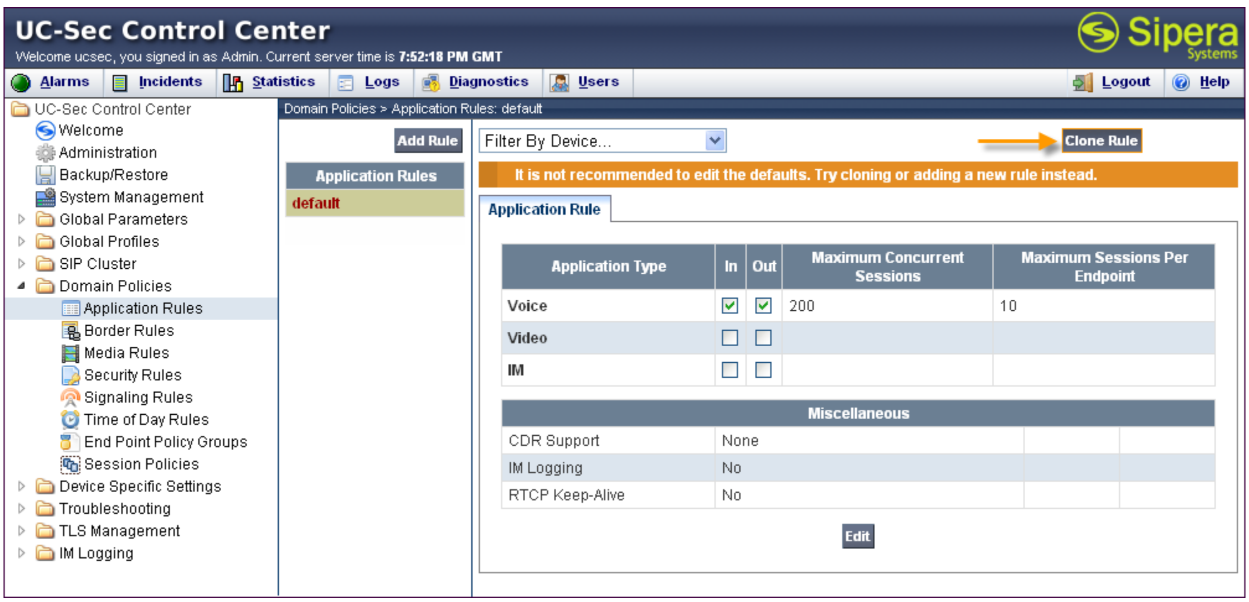
On the **Signaling QoS** tab, select the proper Quality of Service (QoS). Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for signaling. The following screen shows the QoS values used for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



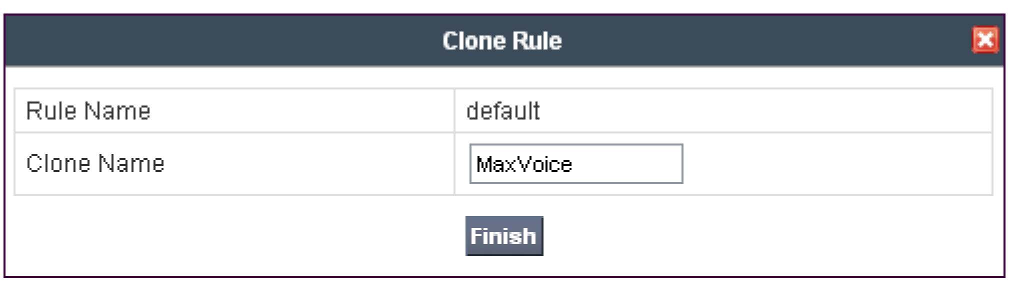
6.2.3. Application Rules

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.

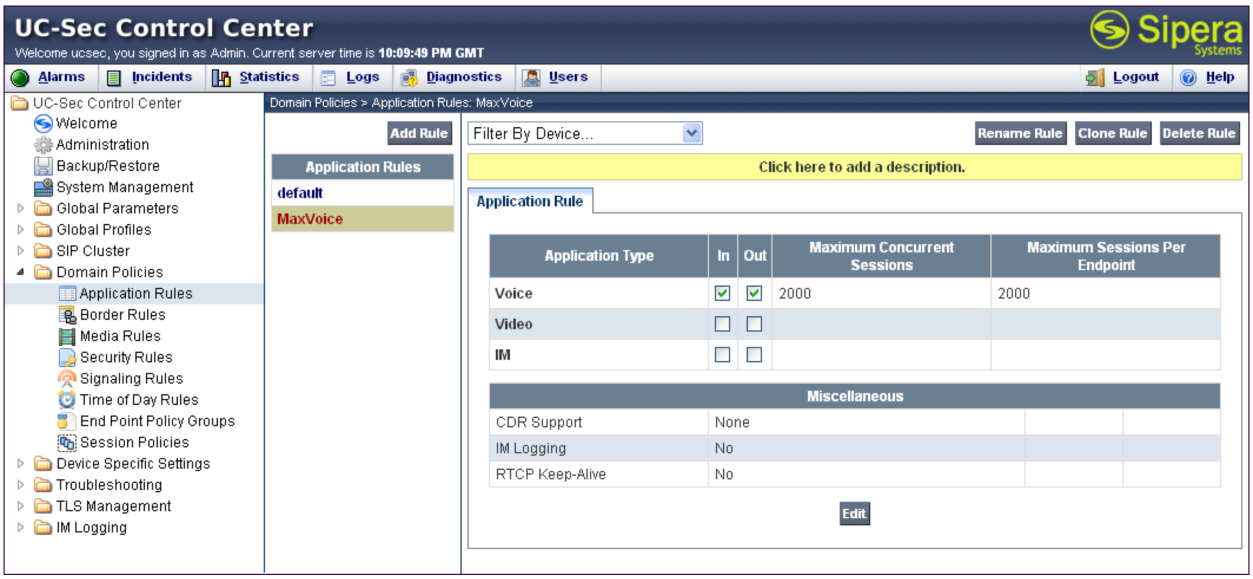
Create an Application Rule to set the number of concurrent voice traffic. The sample configuration cloned and modified the default application rule to increase the number of **Maximum Concurrent Session** and **Maximum Sessions Per Endpoint**. To clone an application rule, navigate to **UC-Sec Control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



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



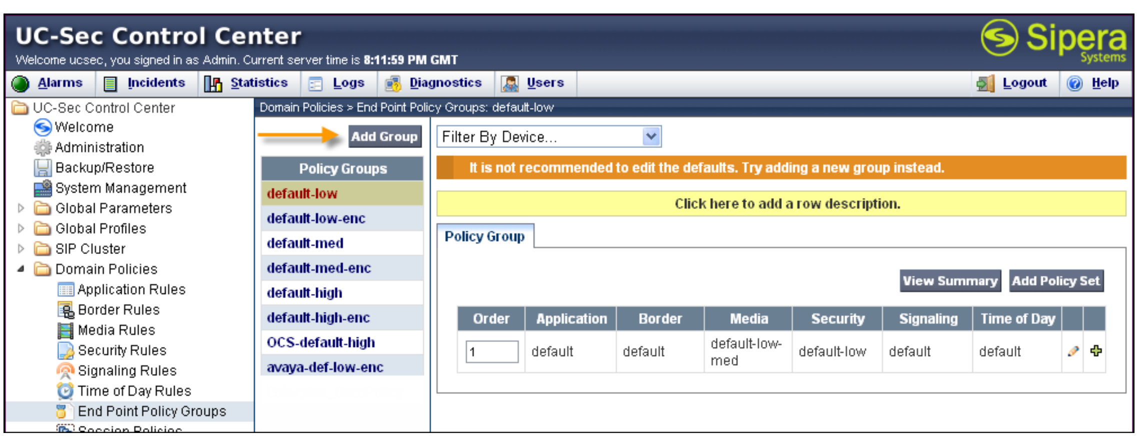
Modify the rule by clicking the **Edit** button. Set the **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** for the **Voice** application to a value high enough for the amount of traffic the network is able process. Keep in mind Avaya SBCE takes 30 seconds for sessions to be cleared after disconnect. The following screen shows the modified Application Rule with the **Maximum Concurrent Sessions** and **Maximum Session 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.



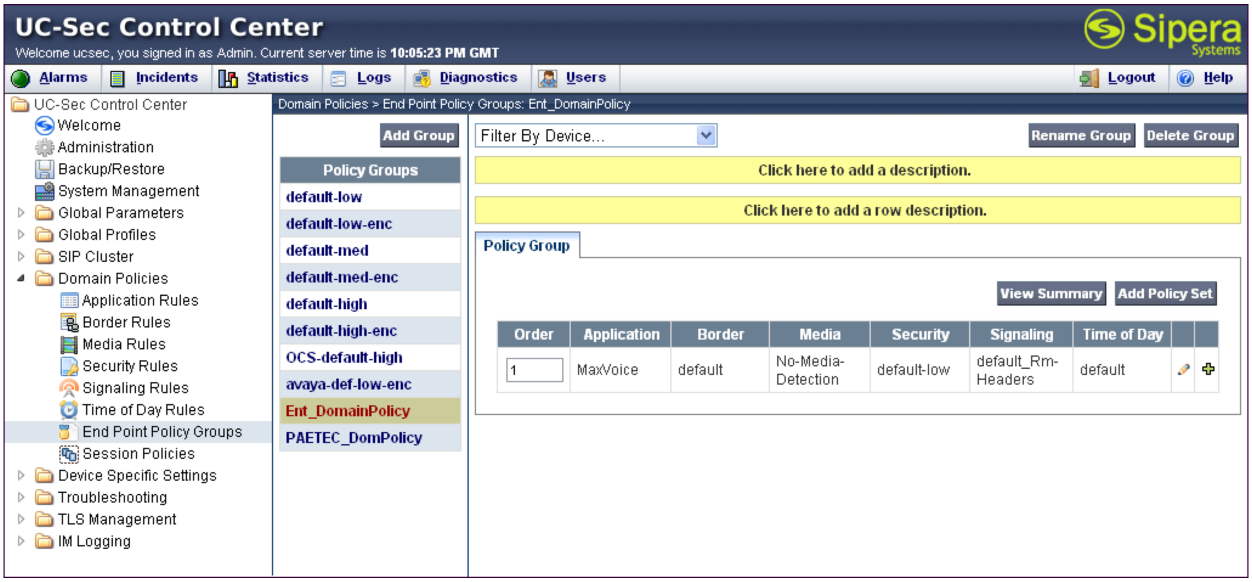
6.2.4. Endpoint Policy Group

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 the enterprise and the PAETEC Dynamic IP SIP Trunk Service.

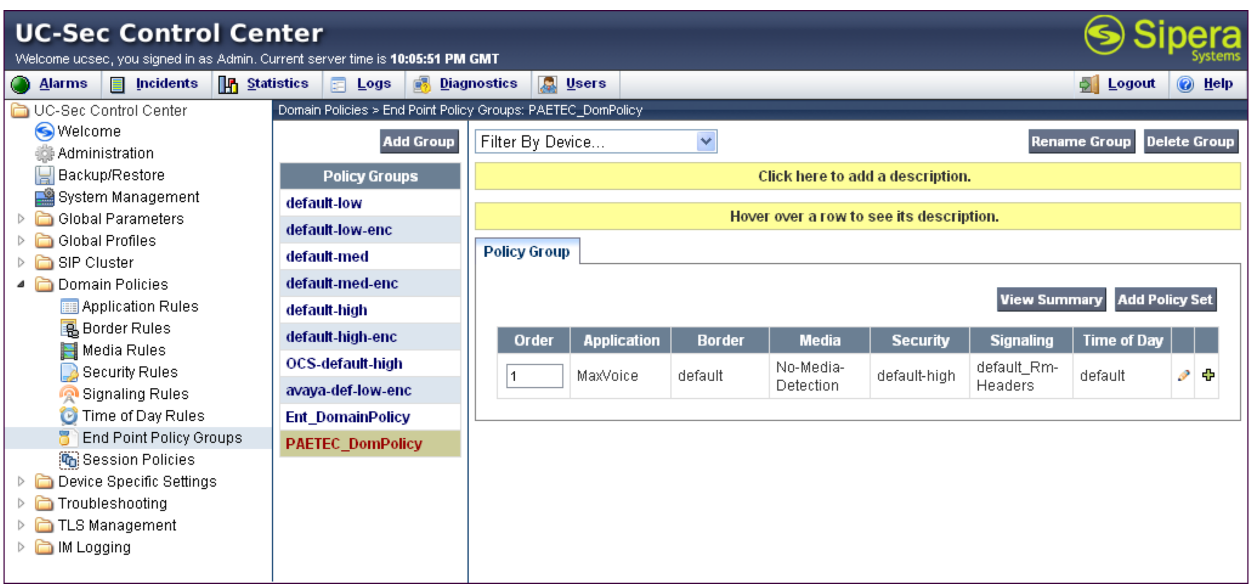
To create a new policy group, navigate to **UC-Sec Control Center →Domain Policies → Endpoint Policy Groups** and click on **Add Group** as shown below.



The following screen shows **Ent_DomainPolicy** 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**.



The following screen shows **PAETEC_DomPolicy** created for PAETEC Dynamic IP SIP Trunk Service. Set the **Application**, **Media** and **Signaling** rules to the ones previously created. Set the **Border**, **Signaling**, and **Time of Day** rules to **default** and set the **Security** rule to **default-high**.



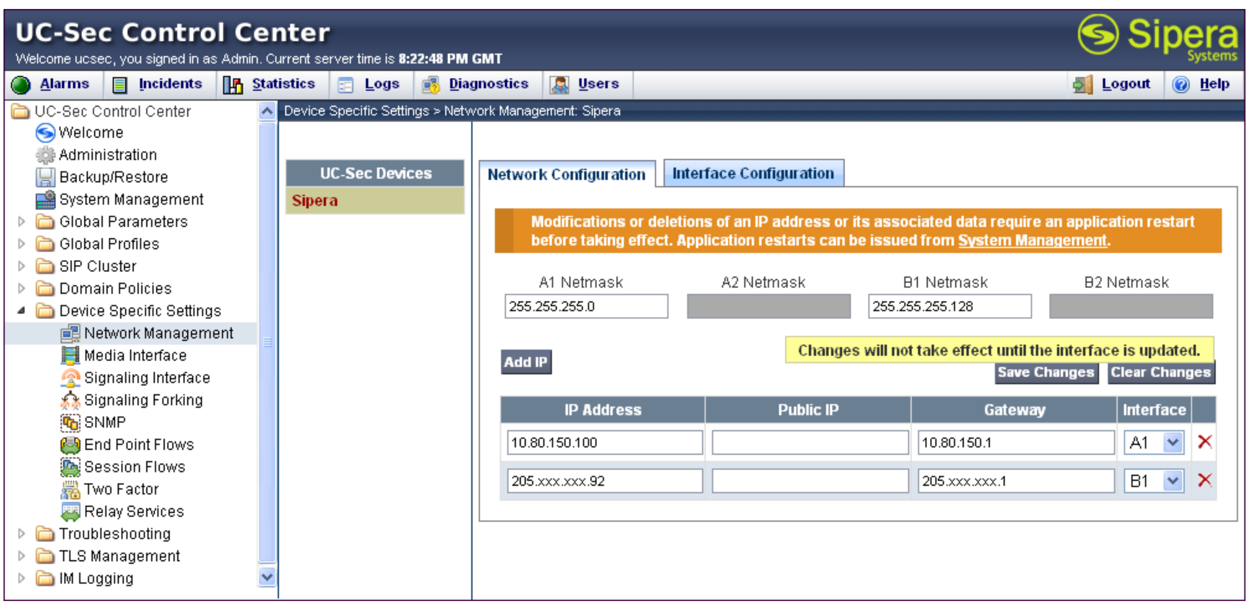
6.3. Device Specific Settings

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.

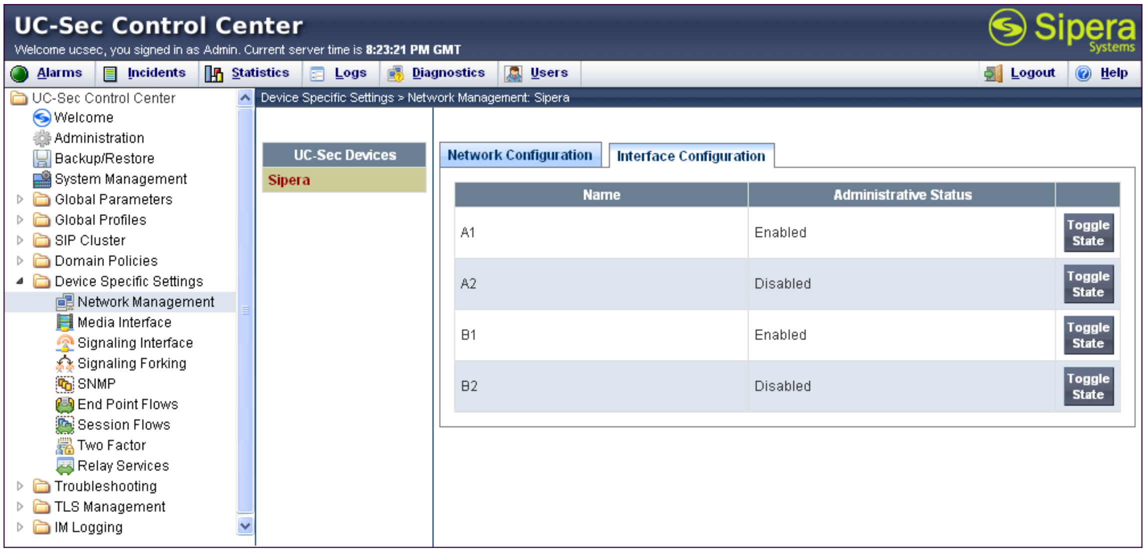
6.3.1. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of 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.

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 interface **A1** and **B1** are **Enabled**. To enable an interface click it's **Toggle State** button.

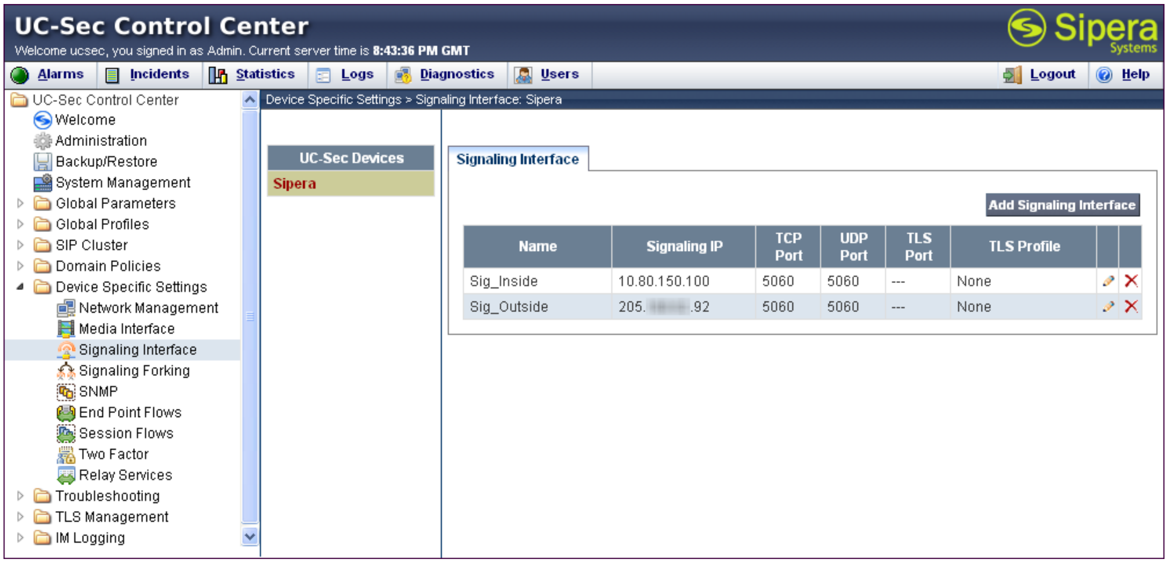


6.3.2. Signaling Interface

The Signaling Interface screen is where the SIP signaling ports are defined. 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.

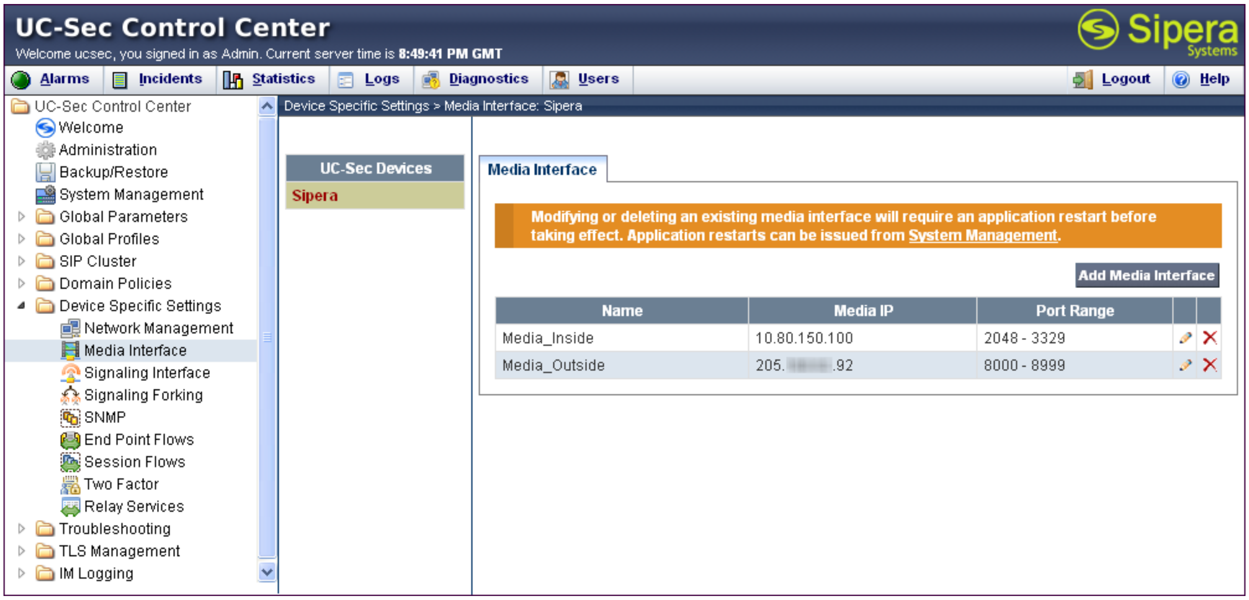


6.3.3. Media Interface

The Media Interface screen is where the SIP media ports are defined. Avaya SBCE will listen for SIP media on the defined ports. Create a SIP 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 created in **Section 5.6.** The outside port range should match the RTP port range provided by PAETEC.

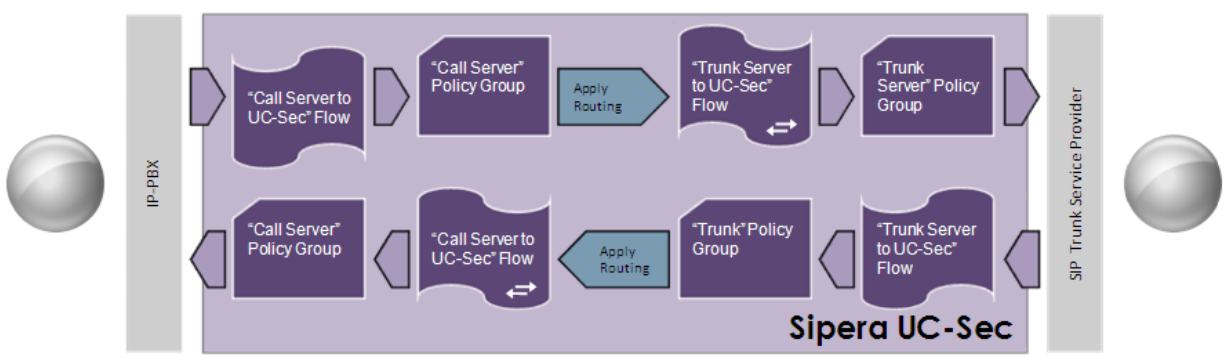
To create a new Media Interface, navigate to **UC-Sec Control Center → Device Specific Settings → Media Interface** and click **Add Media Interface.**

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.

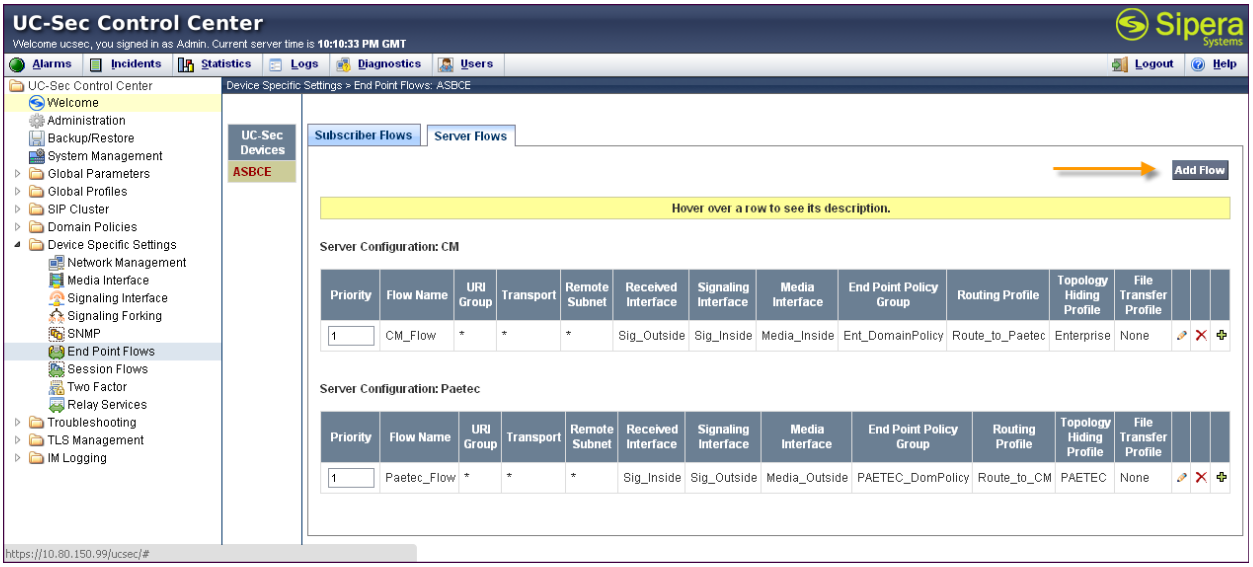


6.3.4. End Point Flows - Server Flow

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 Avaya SBCE to secure a SIP Trunk call.



Create a Server Flow for Communication Manager and the PAETEC Dynamic IP 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.

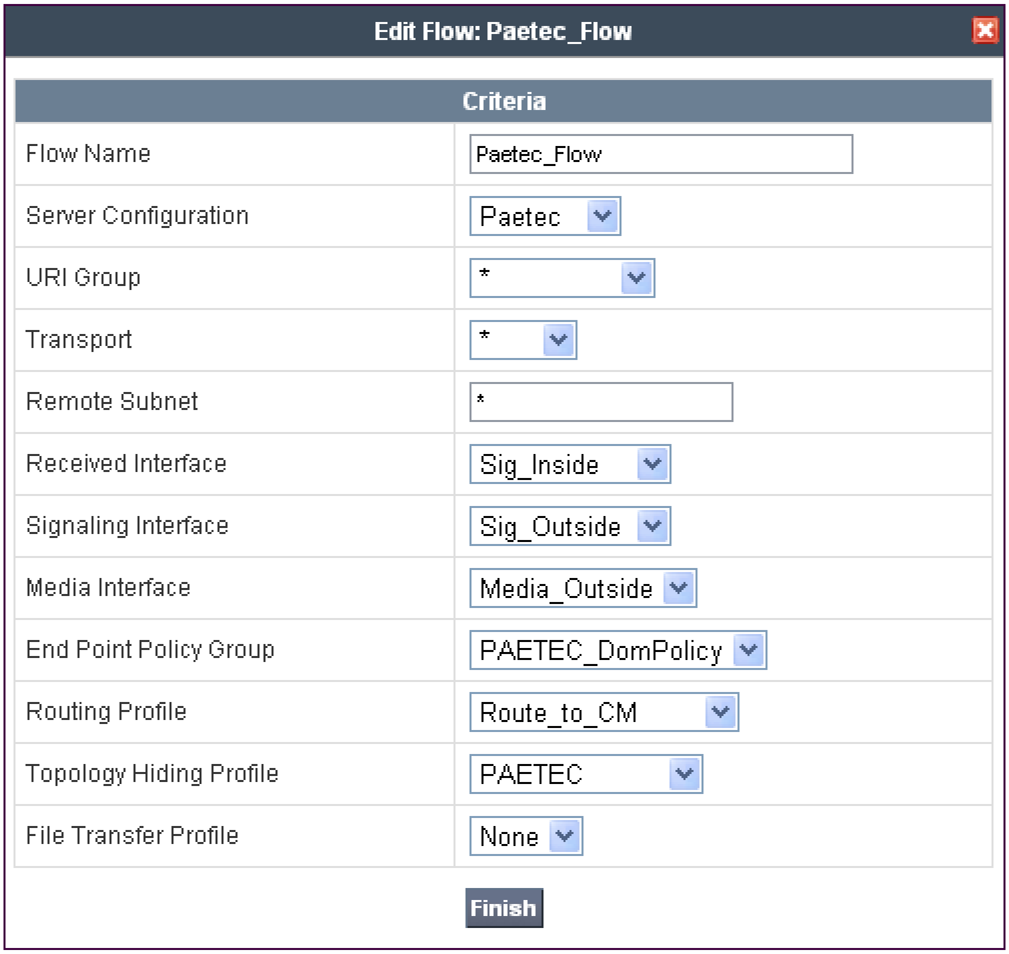


In the new window that appears, enter the following values. Use default values for all remaining fields:

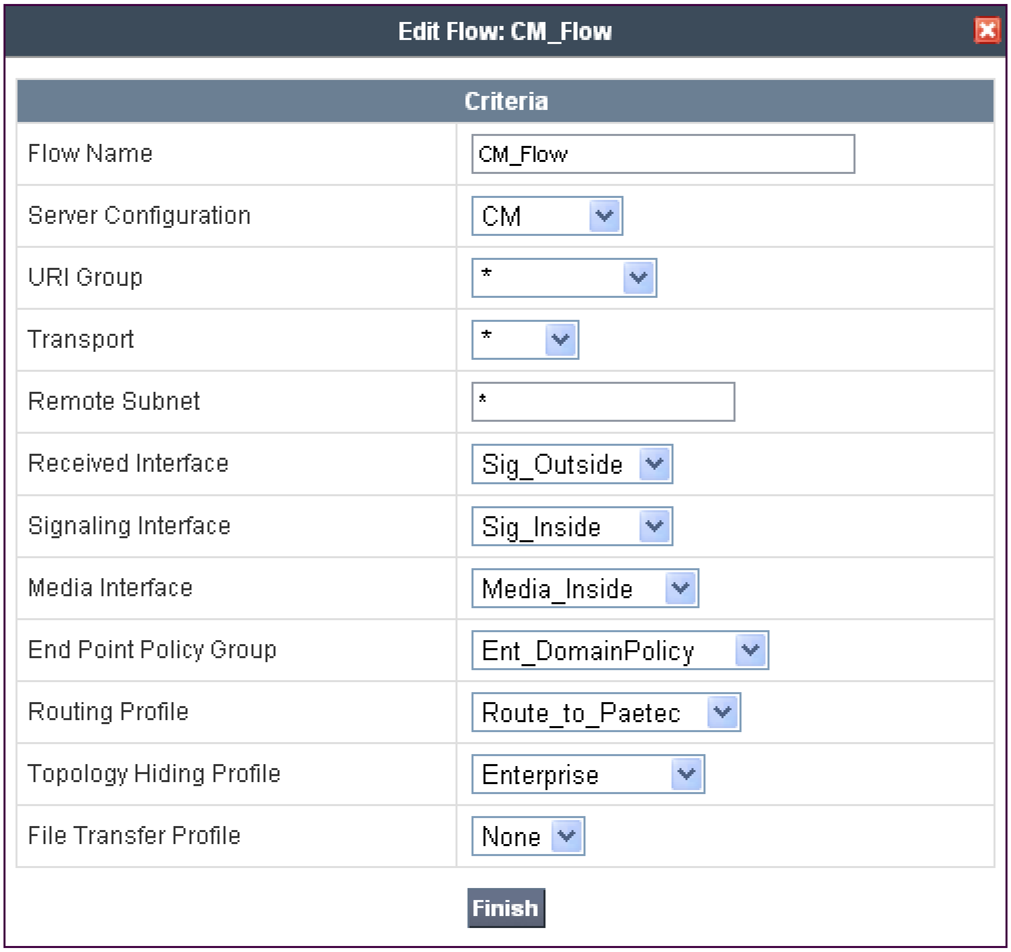
- **Flow Name:** Enter a descriptive name.
- **Server Configuration:** Select a Server Configuration created in **Section 6.1.5** to assign to the Flow.
- **Received Interface:** Select the Signaling Interface the Server Configuration is allowed to receive SIP messages from.
- **Signaling Interface:** 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.

The following screen shows the Sever Flow for PAETEC:



The following screen shows the Sever Flow for Communication Manager:



7. Dynamic IP SIP Trunk Service Configuration

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

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 PAETEC Dynamic IP SIP Trunk Service defined in **Section 5.8**.

Below is an example of an active call.

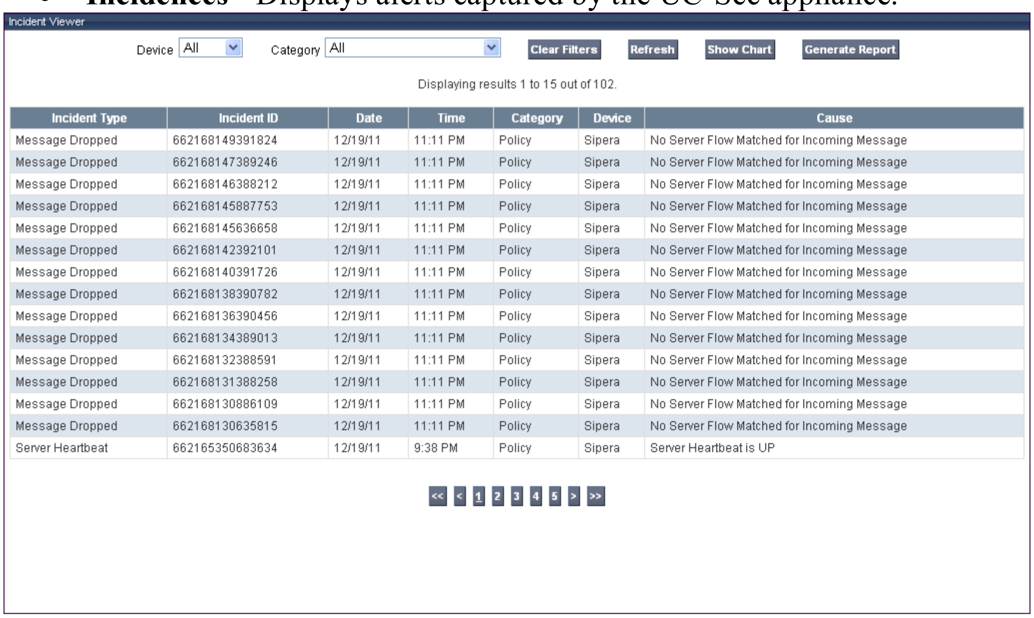
```
status trunk 1

                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

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

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

```
status trunk 1

                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

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

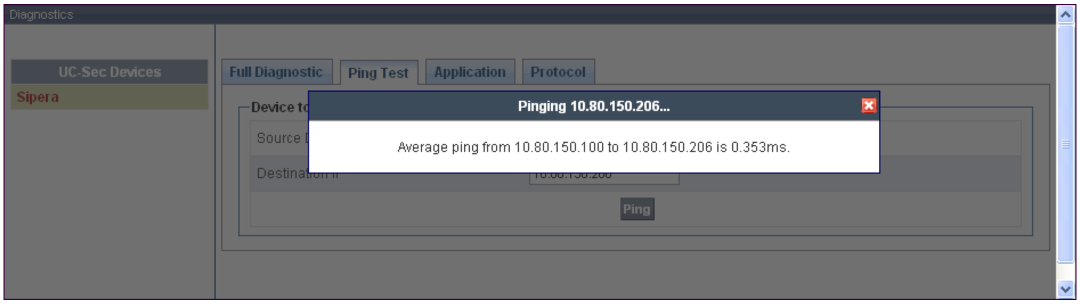
8.2. Troubleshooting

1. Communication Manager:
   - **list trace station** <extension number> - Traces calls to and from a specific station.
   - **list trace tac** <trunk access code number> - Trace calls over a specific trunk group.
   - **status station** <extension number> - Displays signaling and media information for an active call on a specific station.
   - **status trunk** <trunk access code number> - Displays trunk group information.
2. Avaya SBCE:
   - **Incidences** - Displays alerts captured by the UC-Sec appliance.

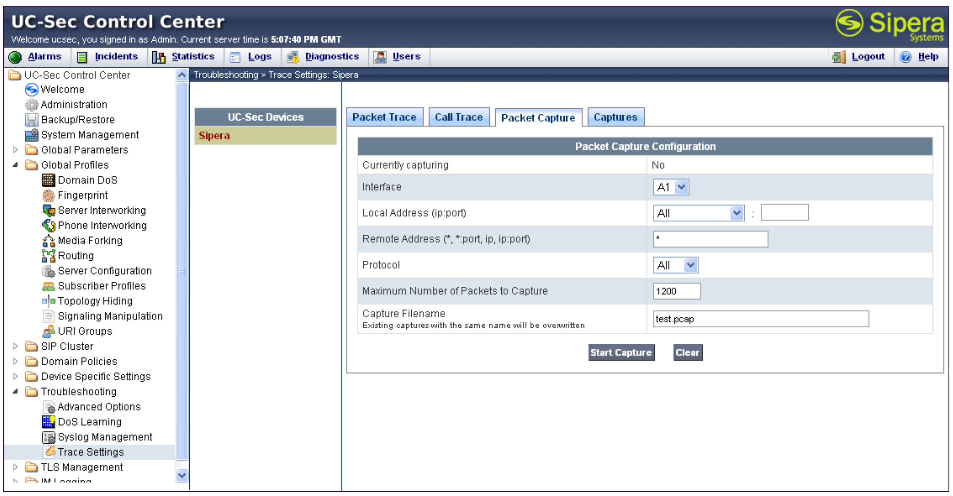


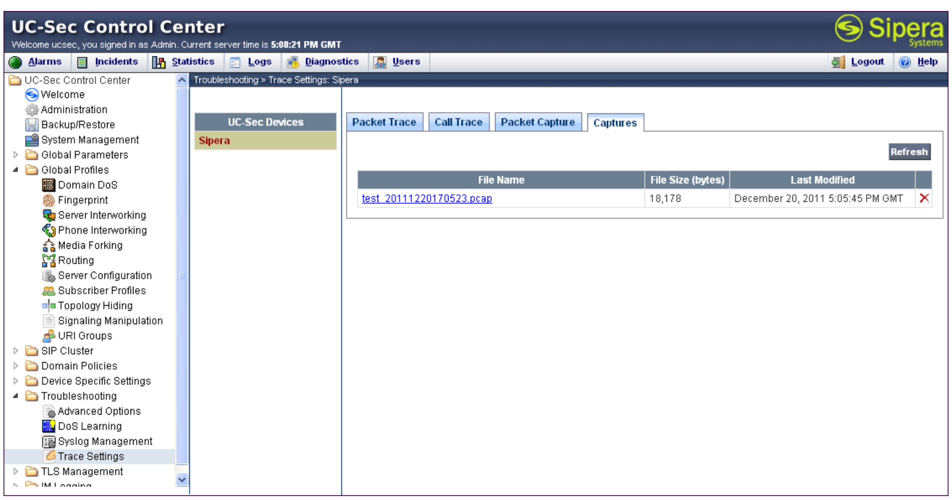
Incident Type	Incident ID	Date	Time	Category	Device	Cause
Message Dropped	662168149391824	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168147389246	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168146388212	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168145887753	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168145636658	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168142392101	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168140391726	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168138390782	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168136390456	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168134389013	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168132388591	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168131388258	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168130886109	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Message Dropped	662168130635815	12/19/11	11:11 PM	Policy	Sipera	No Server Flow Matched for Incoming Message
Server Heartbeat	662165350683634	12/19/11	9:38 PM	Policy	Sipera	Server Heartbeat is UP

   - **Diagnostics** - Allows for PING tests and displays application and protocol use.

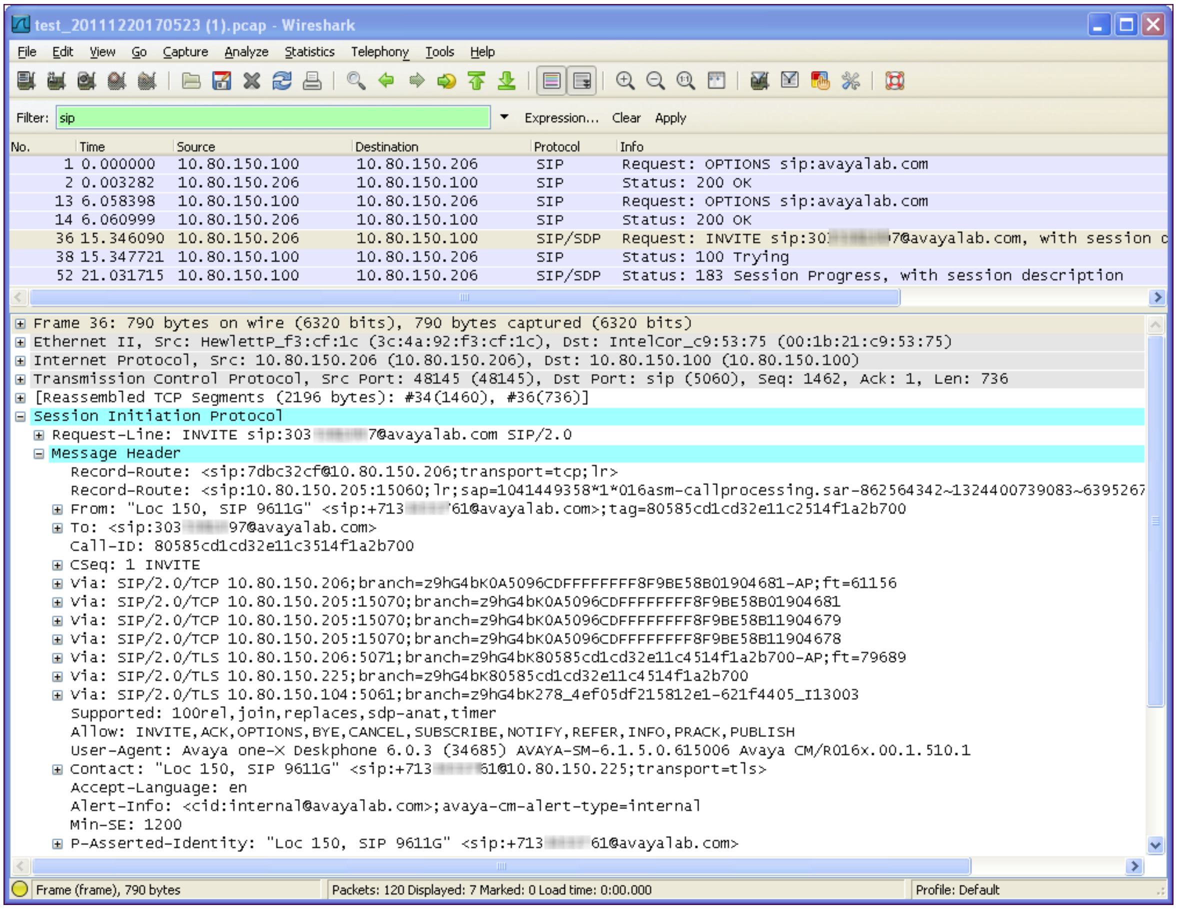


- **Troubleshooting → Trace Settings** - Configure and display call traces and packet captures for the UC-Sec appliance.





The packet capture file can be downloaded and viewed using a Network Protocol Analyzer:



9. Conclusion

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

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,* February 2011.

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

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

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

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

[6] *4600 Series IP Telephone LAN Administrator Guide,* July 2008*,* 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,* December 2010, Document Number 16-603838

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

[10] *Administrator Guide for Avaya Communication Manager,* February 2007, Issue 3, Document Number 03-300509.

[11] *Feature Description and Implementation for Avaya Communication Manager, Issue 5,* 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/*

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.