Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Session Border Controller for Enterprise, Avaya Aura® Session Manager R6.2 and Avaya Aura® Communication Manager R6.2 to interoperate with Semafone in a Southbound Configuration – Issue 1.0

Abstract

These Application Notes describe the steps to configure Avaya Session Border Controller for Enterprise, Avaya Aura® Session Manager and Avaya Aura® Communication Manager with Semafone. Semafone extracts DTMF tones entered by the caller from SIP signaling and replaces them with a generic tone for a call center agent to hear. The extracted DTMF tones can then be sent to a payment platform for processing.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

The Semafone solution in conjunction with Avaya Session Border Controller for Enterprise (ASBCE) enables DTMF tones delivered over a SIP trunk provided by a 3rd party service provider to be extracted and replaced with a generic DTMF tone. The DTMF tones captured can then be sent to a payment platform for processing; the agent hears only the replaced generic tone. In a 'Southbound' configuration, the Semafone solution sits on the private or 'south' side of the Session Border Controller as shown in **Figure 1** below. This is in contrast with the "Northbound" configuration, where the Semafone solution sits on the public or 'north' side of the Session Border Controller. The "Northbound" configuration is described in a separate application note.

The Semafone solution addresses the traditional situation whereby an agent may ask a caller for payment details, which can be spoken by the caller and entered manually by the agent, or the agent would transfer the caller to a separate IVR for capture of the payment details. The Semafone solution enables the caller and the agent to remain connected for the entire duration of the call, and eradicates the inherent risk and overhead of traditional methods used. As the DTMF tones captured are replaced with a generic tone, the agent is never made aware of the payment information. In addition, call recording playback cannot be used to surreptitiously collect the DTMF tones as the Semafone solution sits on the trunk side, in this instance, between private or 'enterprise' interface of the ASBCE and Session Manager. When used in this way, the Semafone solution typically enables a Call Center to more easily gain Payment Card Industry (PCI) compliance.

2. General Test Approach and Test Results

The interoperability compliance testing evaluated the ability of the Semafone solution to establish a SIP trunk connection between the Avaya Session Border Controller for Enterprise and Session Manager and collect and clean DTMF tones when placed into SecureMode.

SecureMode is the feature by which a call routed via Semafone is tagged by a Call Center agent using a unique code and the subsequent DTMF tones entered by the customer are masked. The unique code is generated by Semafone and displayed to the agent through the payment page web interface. The Call Center agent enters the unique code on their telephone keypad, and an icon demonstrating that the call has entered SecureMode is shown on the web interface. The customer is then able to enter their card details using their telephone keypad without the Call Center agent having to perform any additional actions.

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

The following tests were performed as part of the compliance testing.

- DTMF tones delivered by 3rd party service provider per RFC2833/in-band/SIP INFO method
- Collection and cleaning of DTMF tones received by the Semafone solution using the above methods
- Entering SecureMode and verification that DTMF tones collected and passed to the payment engine are as entered
- Verification that none of the original DTMF tones entered are audible by the agent
- Various call routing and agent call handling scenarios

2.2. Test Results

All functionality and serviceability test cases were completed successfully.

2.3. Support

Support is available via www.semafone.com

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The solution consists of a simulated SIP service provider with a SIP trunk to the Public Interface of ASBCE. A separate SIP trunk is established from the Private interface of the ASBCE to the “dirty” interface of the Semafone server SIP Interworking Gateway (SIG). The “dirty” interface is the interface on which the true DTMF tones entered by the customer are received. A further SIP trunk is established between the “clean” side of the Semafone server and the Session Manager SIP Interface. The “clean” interface is the interface on which the DTMF tones entered by the customer have been removed and replaced with a generic tone. Session Manager has a further SIP trunk to Communication Manager. Communication Manager is connected to a G450 which provides DSP resources and services 9630 H.323 Deskphones.

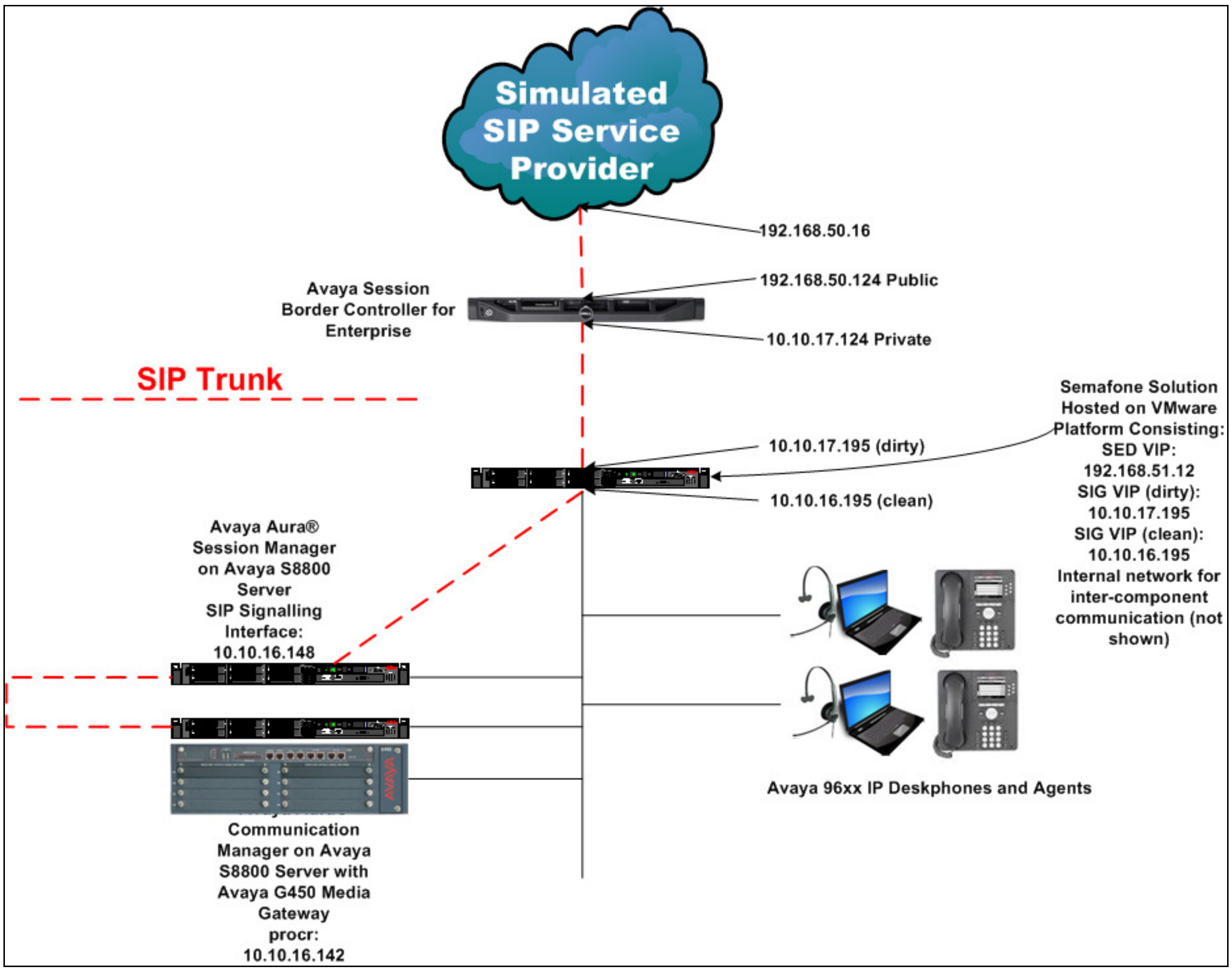


Figure 1: Avaya Session Border Controller for Enterprise, Avaya Aura® Session Manager and Avaya Aura® Communication Manager with Semafone Solution

Figure 2 illustrates the payment network topology used during compliance testing. The solution consists of the same Semafone server hardware and agents shown in **Figure 1** with the addition of a payment page hosted on a webserver residing on the Semafone server.

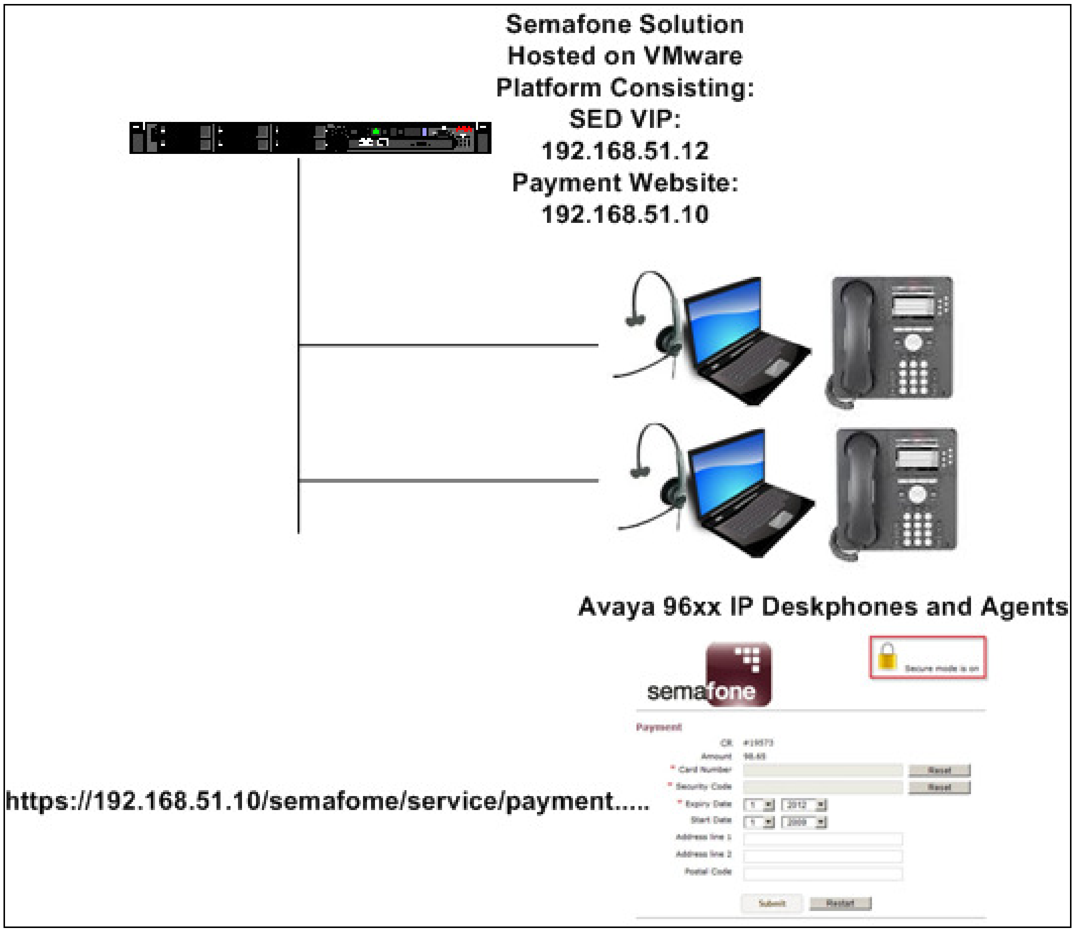


Figure 2: Avaya Agents and workstations with Semafone payment network

Note: The Semafone SED Virtual Machine is a high availability firewall and router appliance. It helps Semafone achieve Network Segmentation in the customer environment by creating a secure Semafone PCI Zone.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Avaya S8800 Server	R6.2 SP5 build R016x.02.0.823.0-20396
Avaya Aura® System Manager running on Avaya S8800 Server	R6.2 SP4
Avaya Aura® Session Manager running on Avaya S8800 Server	R6.2 SP4
Avaya Session Border Controller for Enterprise	4.0 Q19
Avaya G450 Media Gateway	32.24.0
Avaya 9630 IP Deskphone	H323 3.2
Semafone hosted on VMware ESXi 5.0 infrastructure	v3

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section are performed using Communication Manager System Access Terminal (SAT). The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation as referenced in **Section 11**. The configuration operations described in this section can be summarized as follows:

- Configure Routing
- Configure DTMF option

5.1. Configure Routing

The AAR table must be configured with the relevant routing entry for calls to the simulated SIP Service Provider. In this instance trunk-group 1 is already configured as the SIP trunk to Session Manager and route-pattern 1 is configured to route calls over this trunk group. Enter the command **change aar analysis 0**. In the **Dialed String** column enter the digits which will be routed to the SIP Service Provider, in this case **20**. Set the **Total Min** and **Max** value to **4** and the **Route Pattern** value to **1**. When a 4-digit string is dialed beginning with 20, the call will route to Session Manager. The Session Manager configuration later in this document explains how the call is then routed to the SIP Service Provider via the Semafone gateway and the Avaya Session Border Controller for Enterprise.

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

          Dialed              Total     Route     Call   Node  ANI
          String             Min  Max  Pattern    Type   Num   Reqd
    13                       4    4     4         aar           n
    20                       4    4     1         unku          n
    3                        11   11    1         unku          n
    4                        4    4     1         aar           n
    402                      4    4     4         aar           n
    57                       4    4     1         aar           n
    5999                     4    4     1         unku          n
    6                        4    4     1         unku          n
```

5.2. Configure DTMF option

For the purposes of the compliance test, Communication Manager was configured to send DTMF using either the **in-band** (as part of the RTP stream), **out-of-band** (as a SIP INFO message) or **rtp-payload** (RFC2833 DTMF event) method. Enter the command **change signaling-group x** where **x** is the signaling group in relation to the SIP trunk-group connecting to Session Manager, and configure the **DTMF over IP** field as appropriate.

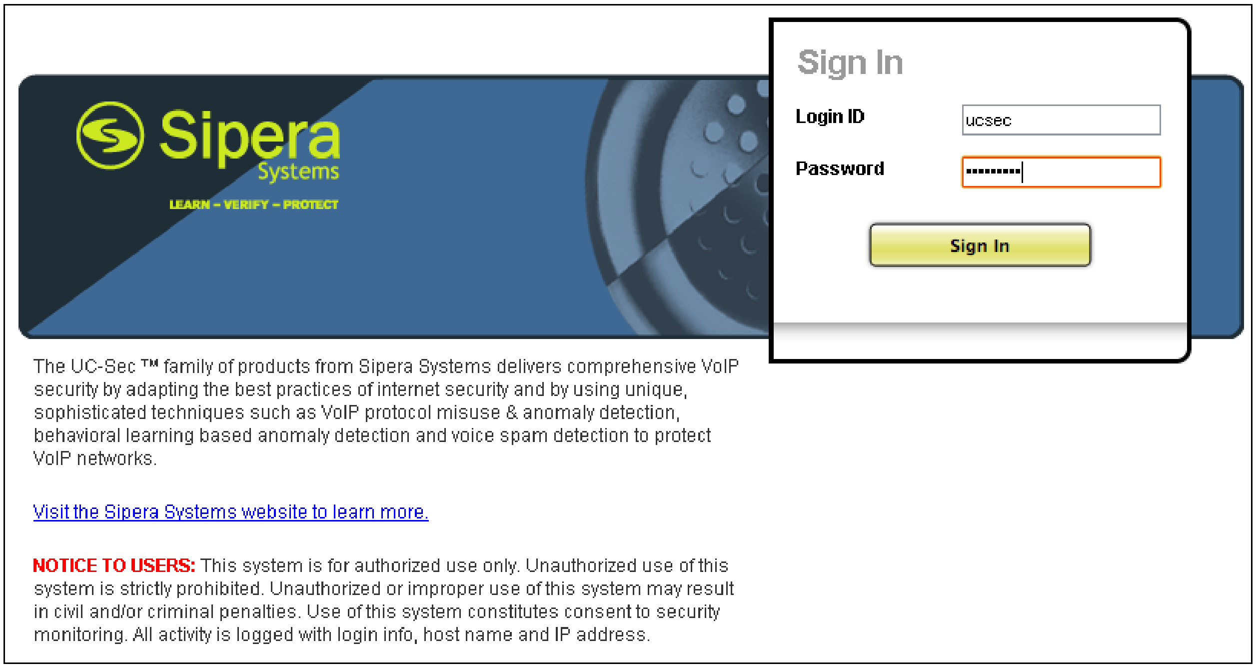
```
change signaling-group 1                                     Page   1 of   2
                                SIGNALING GROUP

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



   Near-end Node Name: procr                 Far-end Node Name: sm62sigint
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain:
                                           Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                RFC 3389 Comfort Noise? n
         DTMF over IP: in-band                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? y
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

6. Configure Avaya Session Border Controller for Enterprise

These Application Notes assume that the installation of ASBCE and the assignment of a Management IP Address has already been completed.

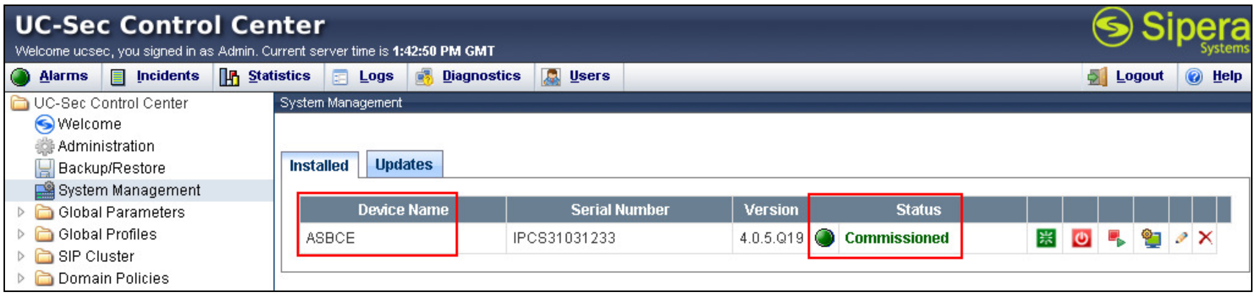
6.1. Access Management Interface

Use a WEB browser to access the web management interface by entering URL https://<ip-addr>, where **<ip-addr>** is the management LAN IP address assigned during installation. Select **UCSec Control Center** (not shown) on the displayed web page, and log in using proper login credentials.

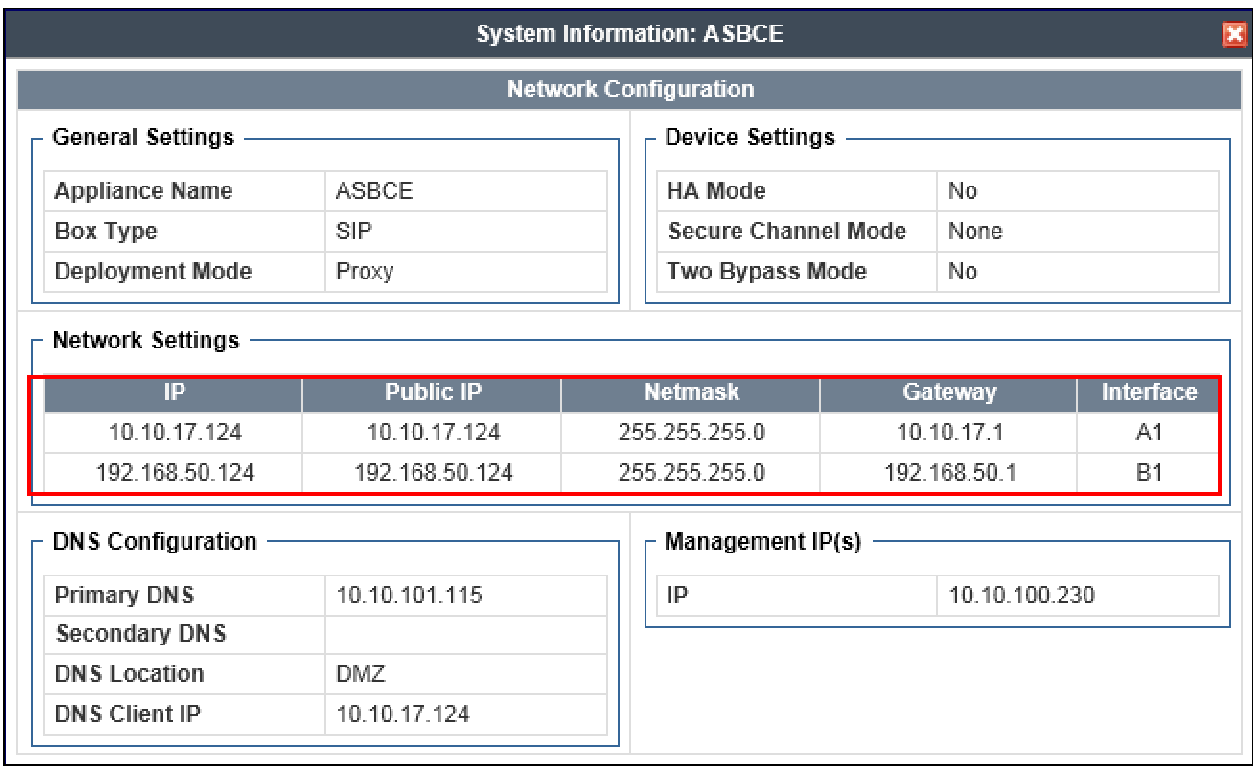


6.2. System Status

Navigate to **UC-Sec Control Center → System Management**. A list of installed devices is shown in the right pane. For the sample configuration, a single device named **ASBCE** is shown. The **Status** will appear as **Commissioned** as shown below.

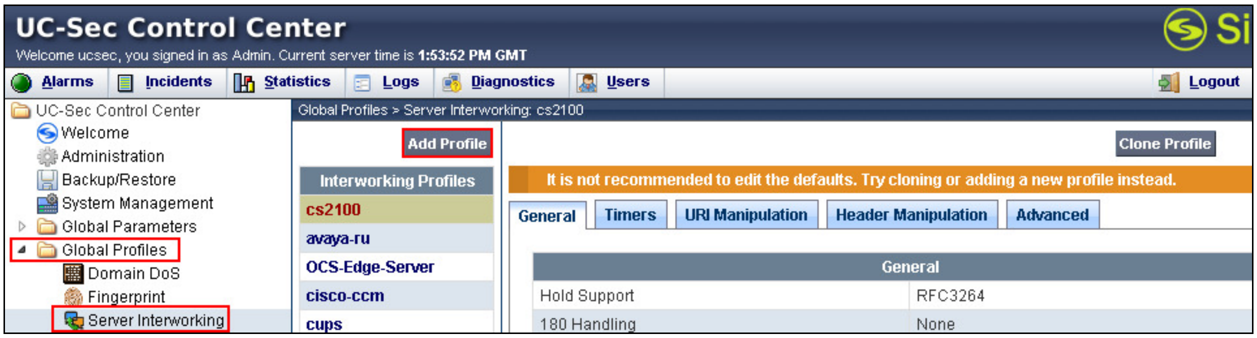


To view the network information of this device, which was assigned during installation, click the **View Config** icon button (the third icon from the right). A **Network Configuration** window is displayed as shown below. Note that the **A1** and **B1** interface IP addresses correspond to the Private and Public interfaces, respectively, for the ASBCE as shown in **Figure 1** in **Section 3**.



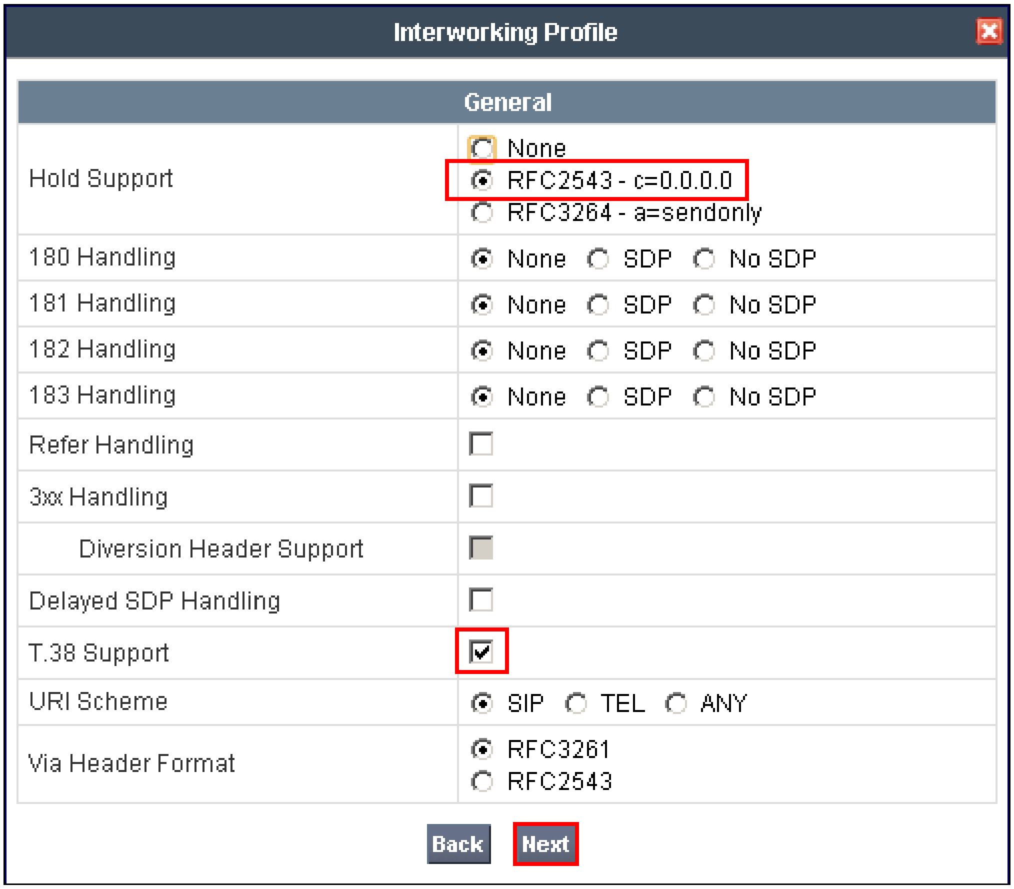
6.3. Global Profiles – Add Server Interworking Profiles

An interworking profile must be administered to define the features supported by the relevant server configured in **Section 6.4**. Server interworking is defined for each server connected to ASBCE. For the compliance test, the Simulated SIP Service Provider serves as the Trunk Server and the Semafone SIG interface serves as the Call Server. Navigate to **Global Profiles → Server Interworking** from the left-side menu and click **Add Profile** to configure new server interworking profiles.

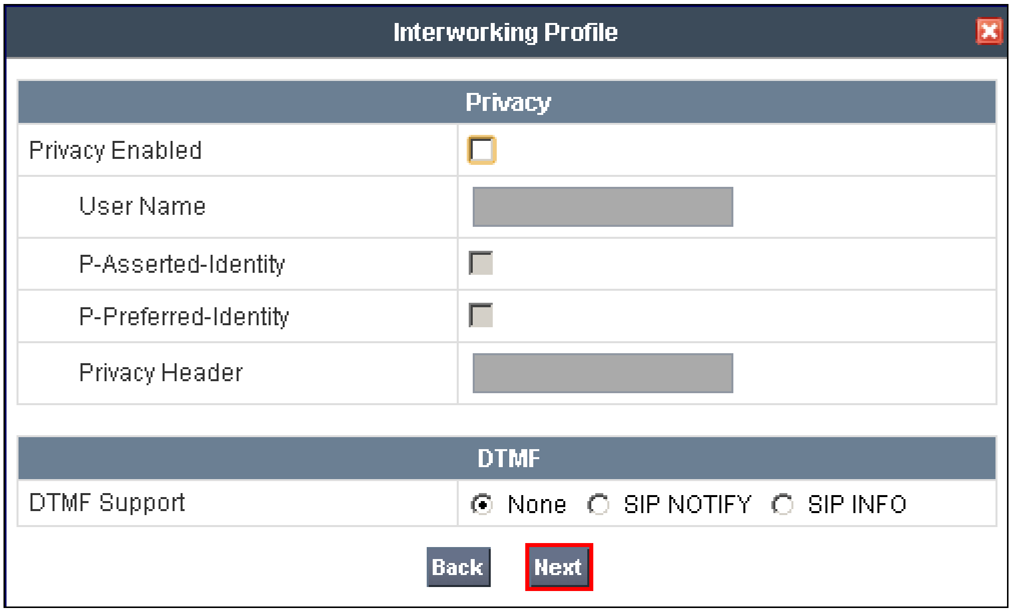


Enter an appropriate **Profile Name** such as **Avaya SM** shown below and click **Next**.

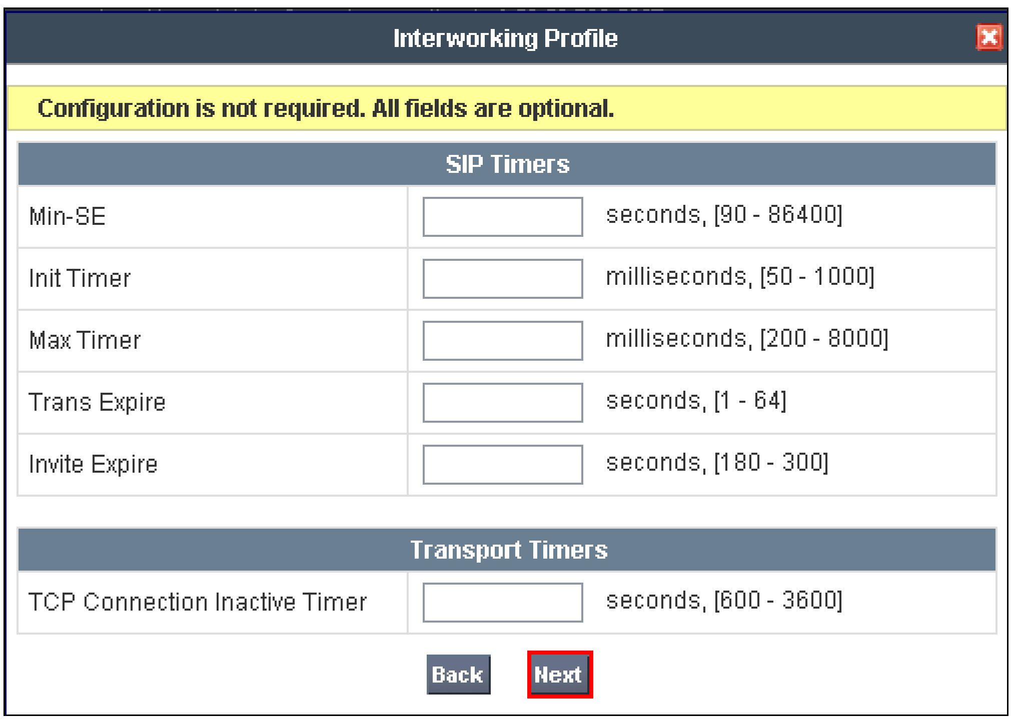
The **General** page is displayed. In the sample configuration, **T.38 Support** was checked to enable T.38 faxing (though not relevant to the configuration), and **Hold Support** was set for **RFC2543**. Click **Next**.



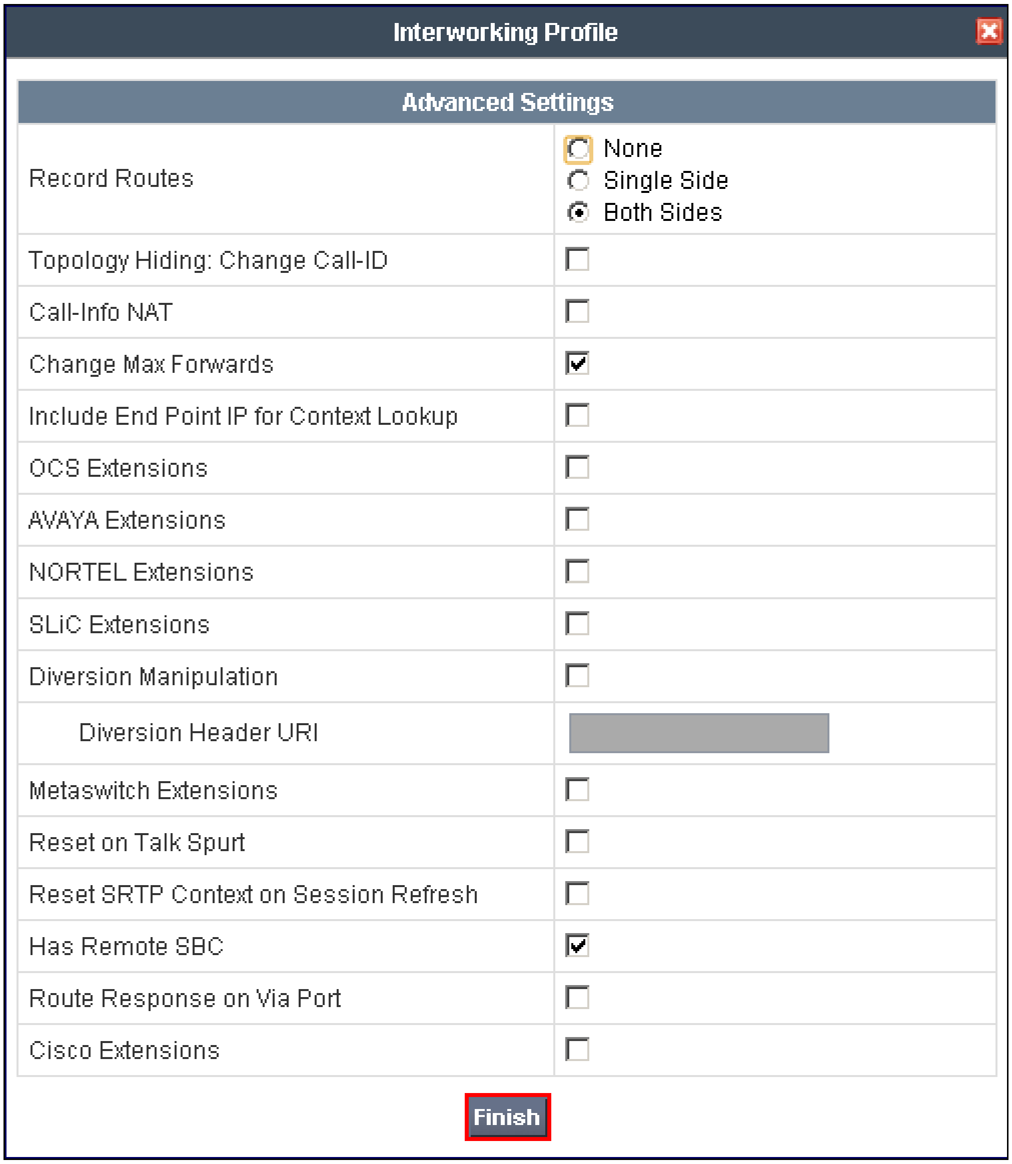
Leave the **Privacy** and **DTMF** fields at their default values and click **Next**.



Leave the **SIP Timers** and **Transport Timers** fields at their default values and click **Next**.



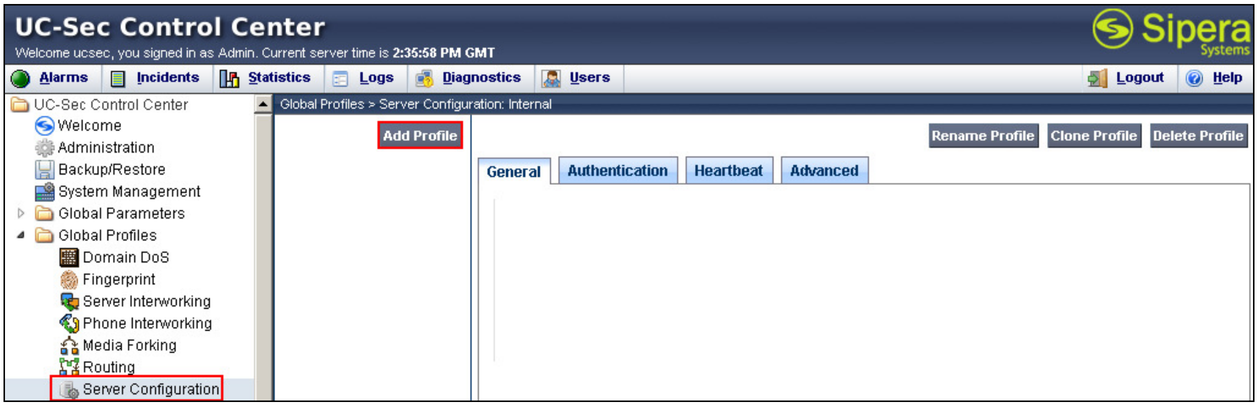
Leave the **Advanced Settings** fields at their default values and click **Finish**.



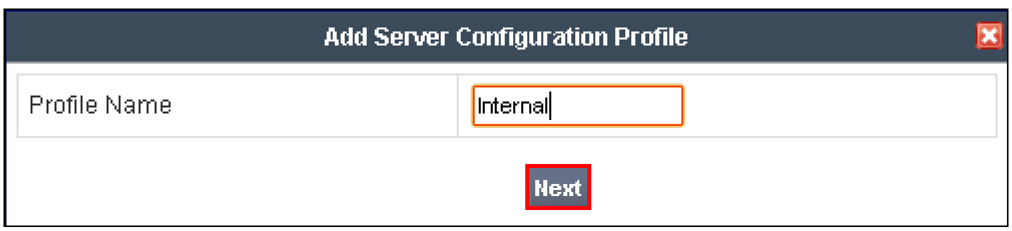
Similarly add an identical interworking profile for the public side of the ASBCE named **PUBLIC**. The screenshot below displays the two newly administered profiles.

6.4. Global Profiles – Server Configuration

In the compliance test, the Simulated SIP Service Provider is administered and connected as the Trunk Server and the Semafone SIG interface is administered and connected as the Call Server. Navigate to **Global Profiles → Server Configuration** from the left-side menu and click **Add Profile** to configure the first of the two servers.



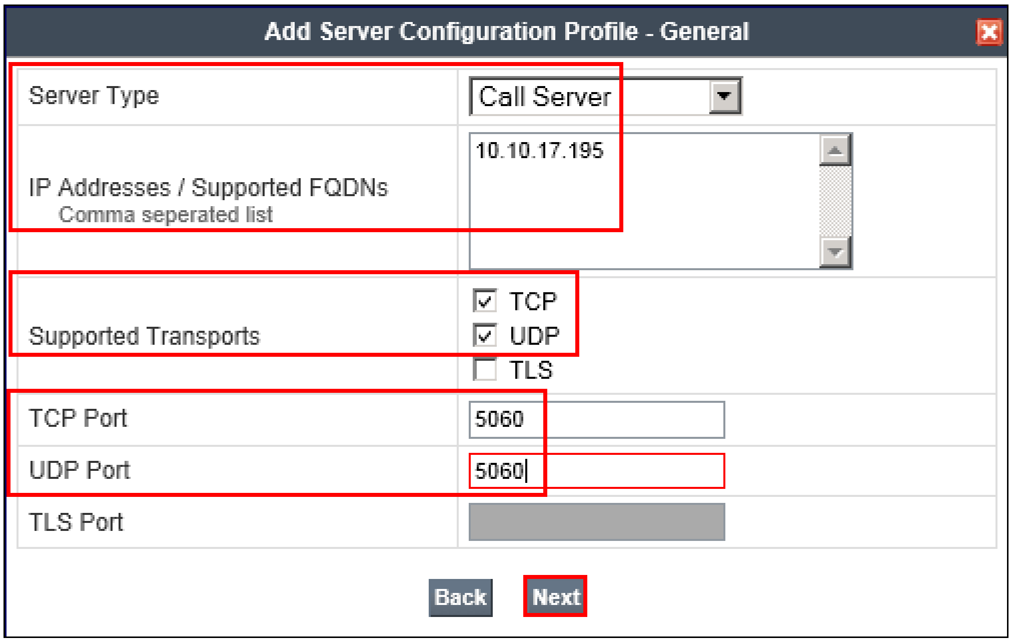
Enter an appropriate **Profile Name** such as **Internal** shown below. Click **Next**.



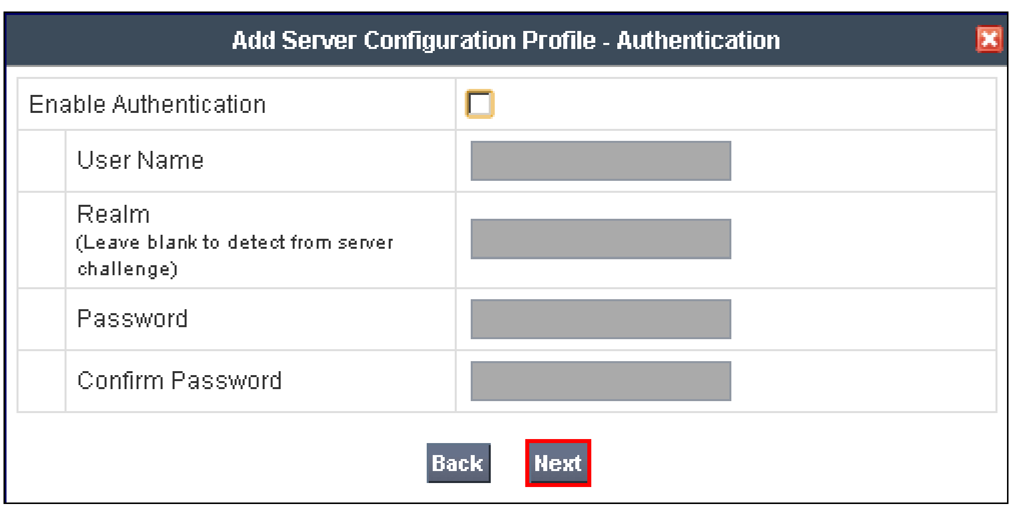
Configure the **General** fields as follows:

- **Server Type** – select **Call Server** from the drop down box
- **IP Addresses / Supported FQDNs** - enter the IP address assigned to the Semafone SIG interface, in this case **10.10.17.195**
- **Supported Transports** –place a check in the **TCP** and **UDP** boxes
- **TCP Port** and **UDP** Port – set to **5060**

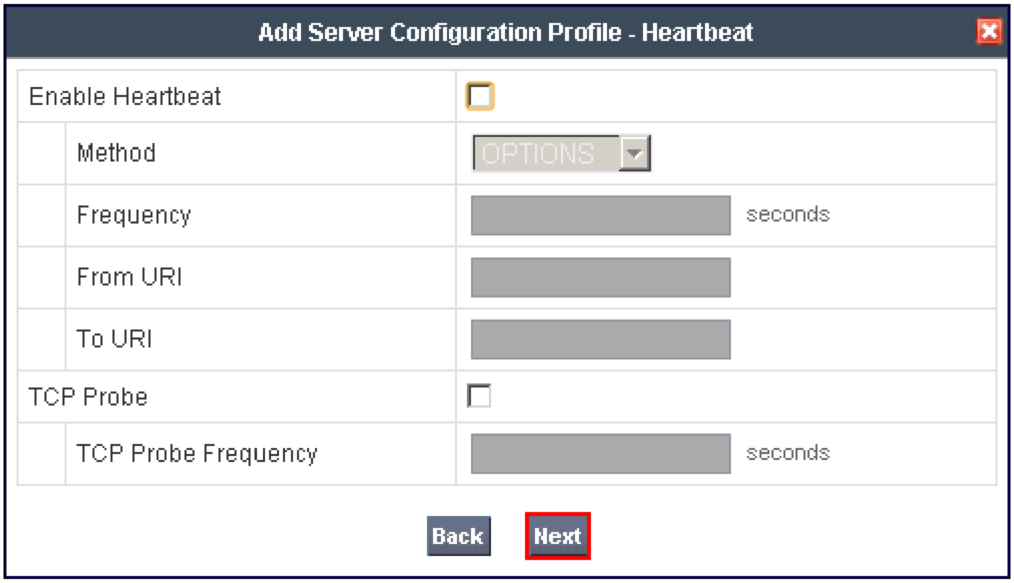
This configuration relates to the connection between the ASBCE private interface and the Semafone SIG interface. Click **Next**.



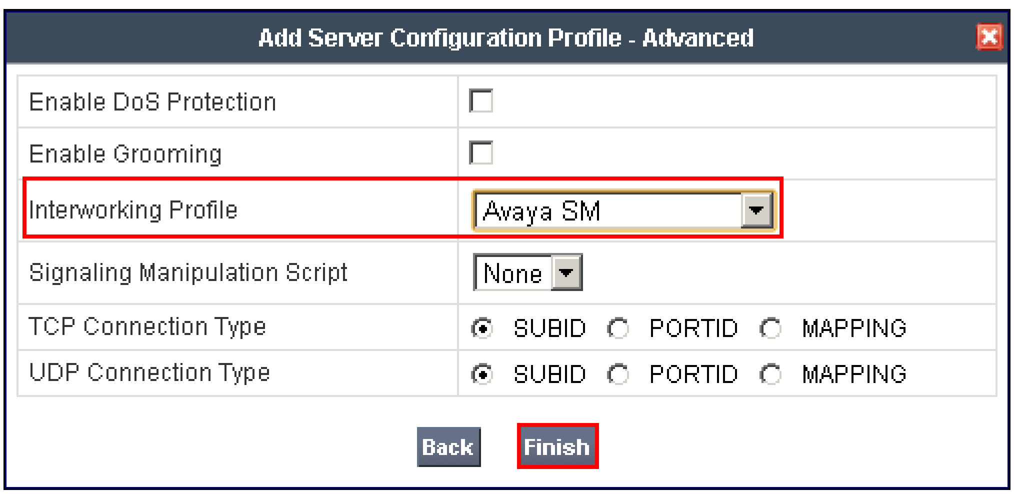
Leave the **Authentication** screen fields at their default values and click **Next**.



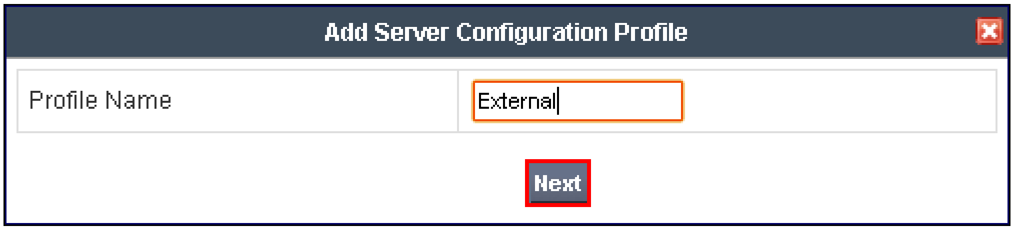
Leave the **Heartbeat** fields at their default values and click **Next**.



From the drop down box select the **Interworking Profile** configured in the previous section for the internal side of the ASBCE, in this case **Avaya SM**. Leave all other fields at their default values and click **Finish**.



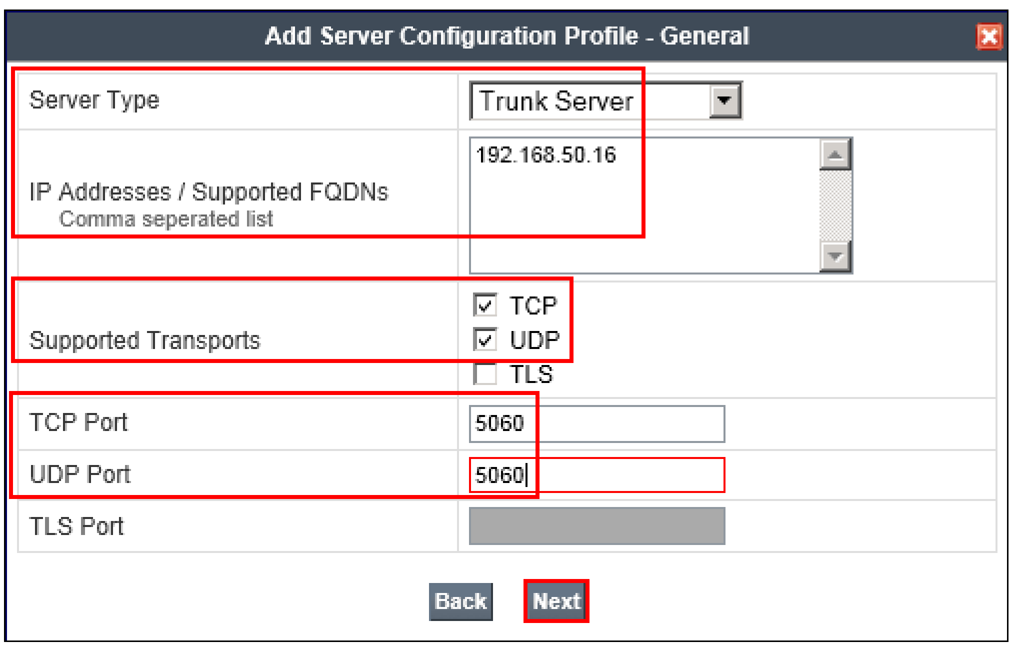
Repeat the steps above to configure the Server for the Public side of the ASBCE. Enter an appropriate **Profile Name** such as **External** shown below. Click **Next**.



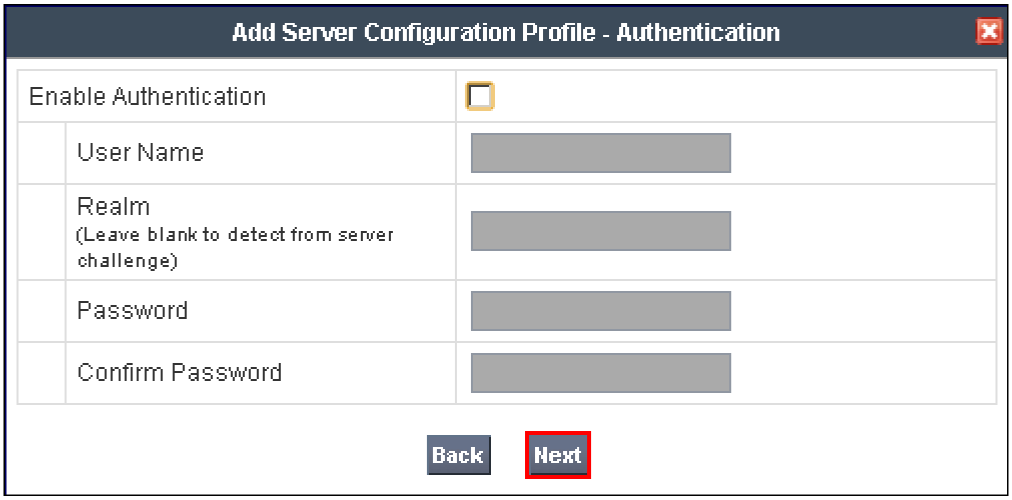
Configure the **General** fields as follows:

- **Server Type** – select **Trunk Server** from the drop down box
- **IP Addresses / Supported FQDNs** - enter the IP address assigned to the Simulated SIP Service Provider, in this case **192.168.50.16**
- **Supported Transports** –place a check in the **TCP** and **UDP** boxes.
- **TCP Port** and **UDP Port** – set to **5060**

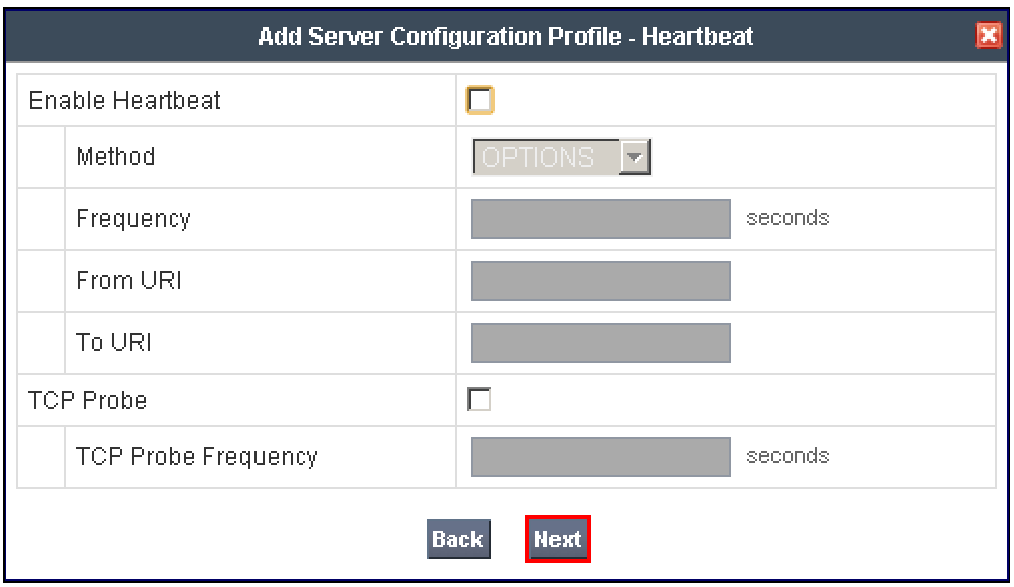
This configuration relates to the connection between the ASBCE public interface and the Simulated SIP Service Provider.



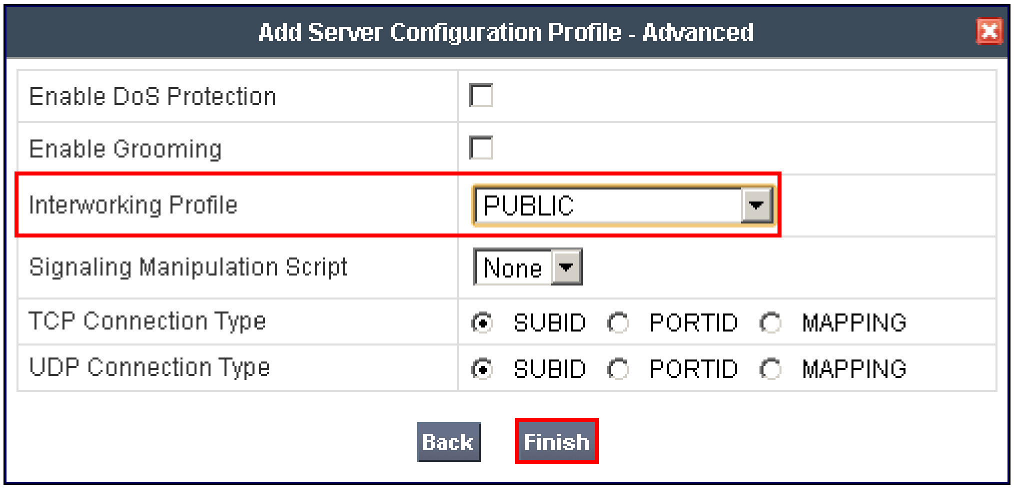
Leave the **Authentication** screen fields at their default values and click **Next**.



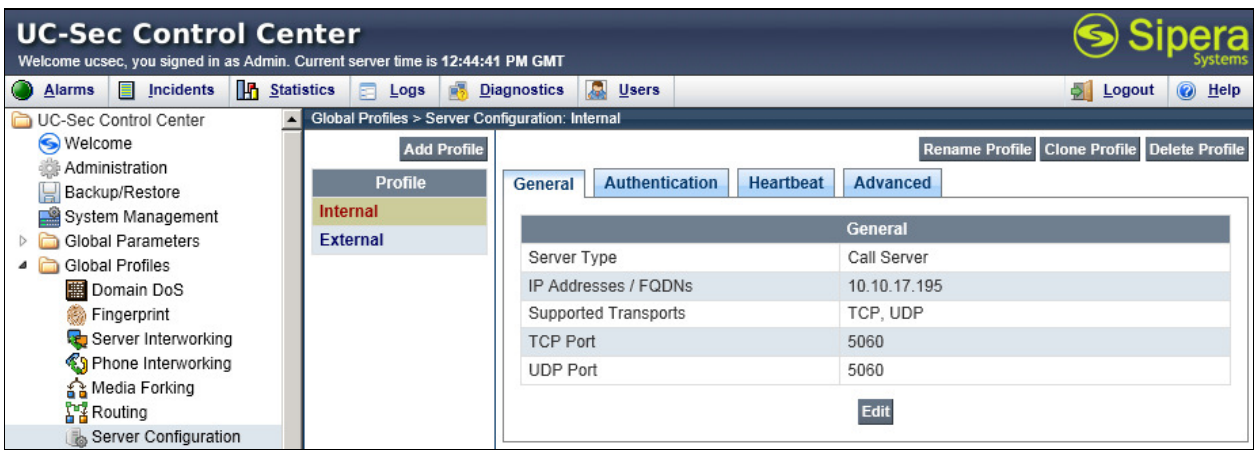
Leave the **Heartbeat** fields at their default values and click **Next**.



From the drop down box select the **Interworking Profile** configured in the previous section for the Public side of the ASBCE, in this case **PUBLIC**. Leave all other fields at their default values and click **Finish**.



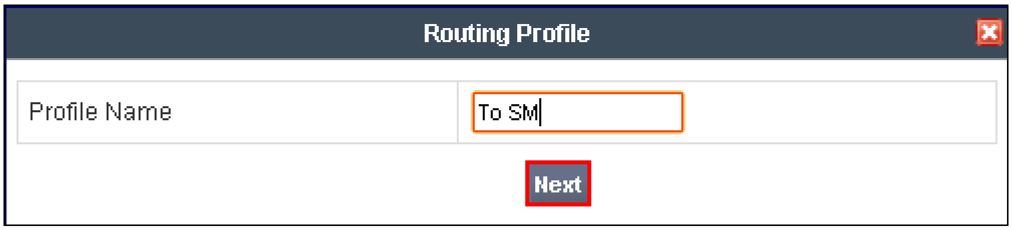
The following screen will appear displaying the newly administered Server Configuration Profiles



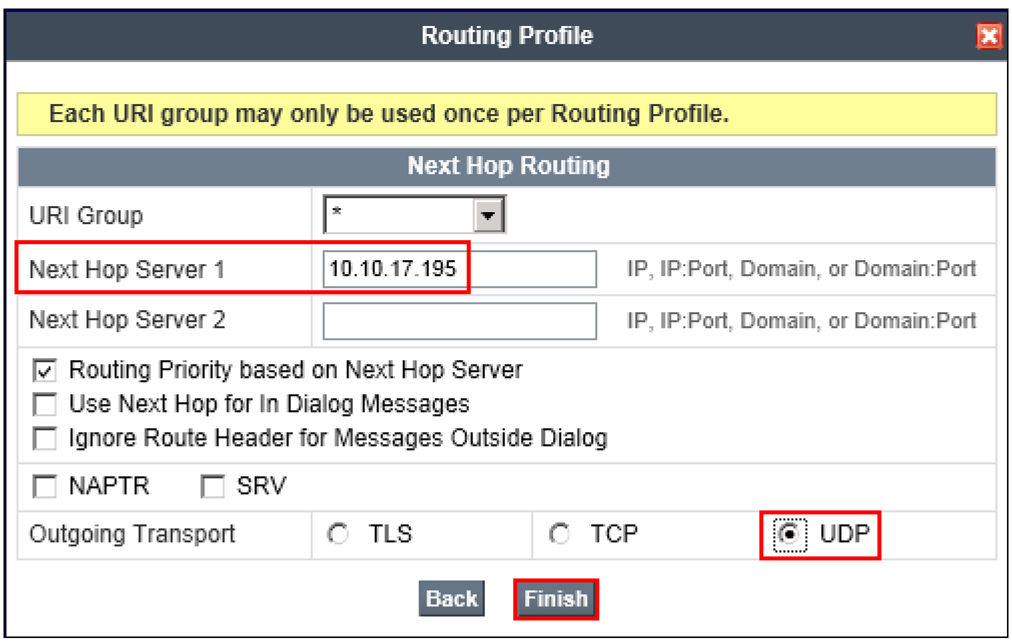
6.5. Global Profiles – Routing

Routing information is required for traffic to be routed to Session Manager via Semafone on the internal side, and to the Simulated SIP Service Provider on the external side. The IP addresses and ports defined here will be used as the destination addresses for signaling. If no port is specified, the default SIP port of 5060 is used.

Navigate to **Global Profiles → Routing → Add Profile**. Enter an appropriate **Profile Name** such as **To SM** as shown below. Click **Next**.

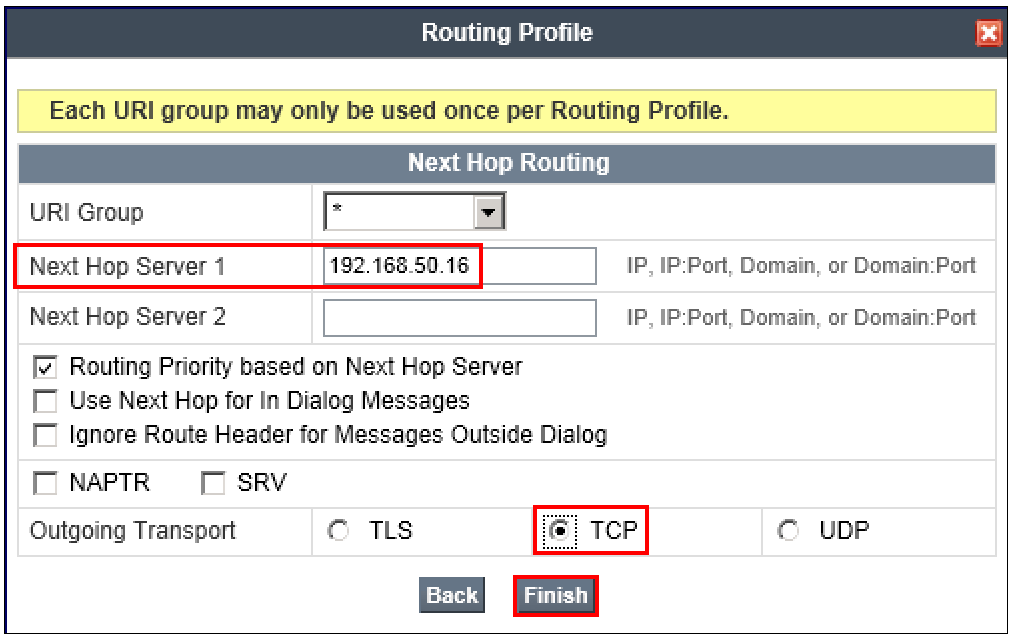


In the **Next Hop Routing** configuration, enter the IP Address of the Semafone SIG interface as **Next Hop Server 1**, as shown below. Choose **UDP** for **Outgoing Transport** and click **Finish**.

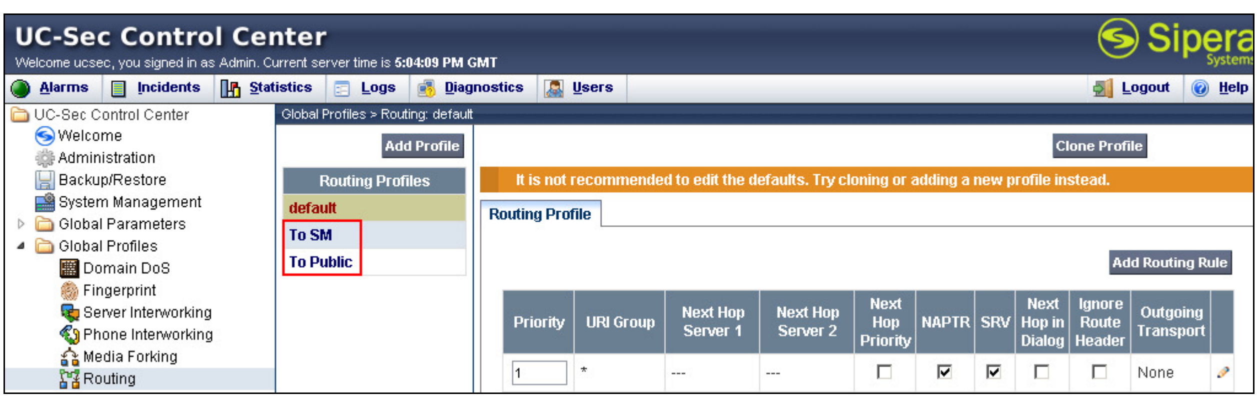


Similarly create a Routing Profile to the Simulated Service Provider interface.

Note the **Next Hop Server 1** IP address is that assigned to the Simulated SIP Service Provider and the **Outgoing Transport** is **TCP**. Click **Finish**.



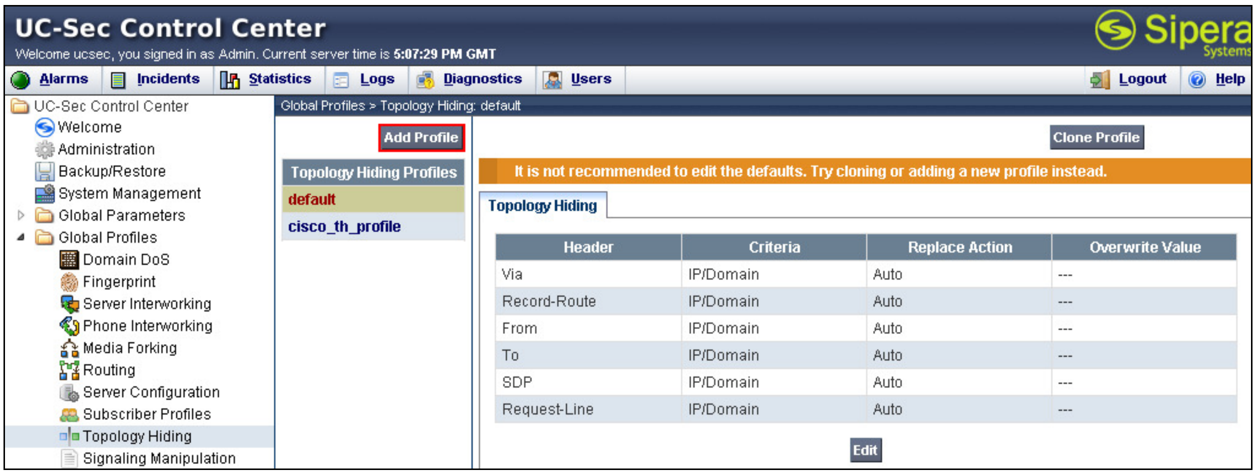
The screenshot below shows the newly configured Routing Profiles.



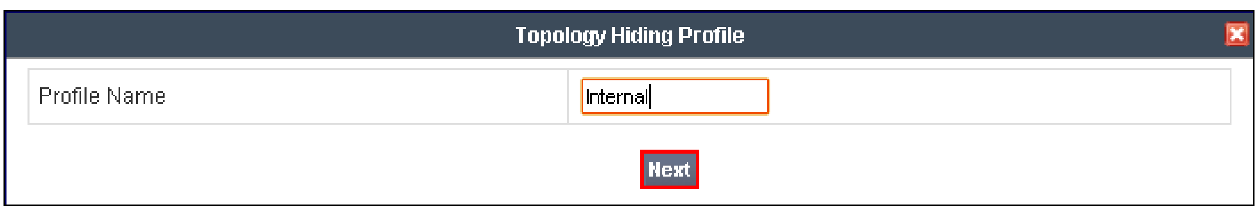
6.6. Global Profiles – Topology Hiding

Topology Hiding is a security feature which allows the changing of several parameters within SIP packets, preventing the private enterprise network information from being propagated to the un-trusted public network. Topology Hiding can also be used as an interoperability tool to adapt certain parameters in selected SIP headers to meet expectations by Session Manager and the SIP trunk service provider, allowing the call to be accepted in each case. For the compliance test, only the minimum configuration required to achieve interoperability was performed.

Navigate to **Global Profiles → Topology Hiding → Add Profile**.

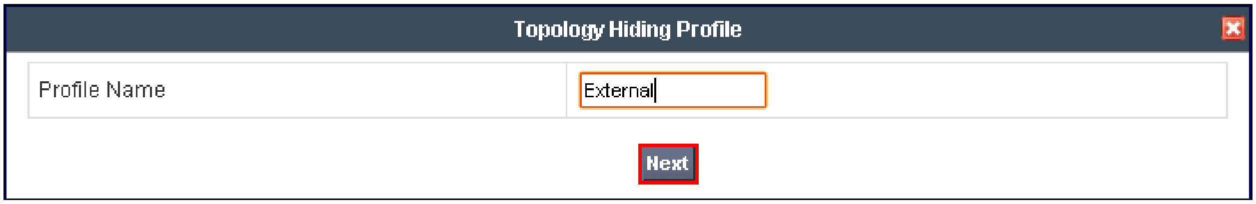


Enter a **Profile Name** such as **Internal** shown below. Click **Next**.

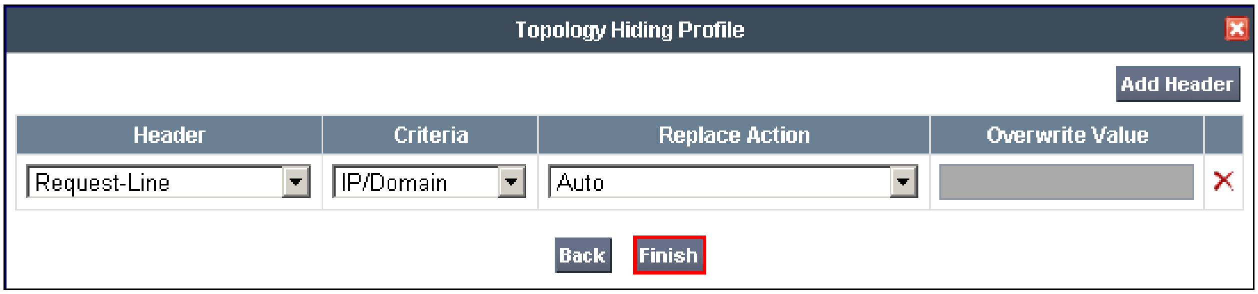


The screen below will appear. For the purposes of the compliance test it was unnecessary to make any changes or additions. Click **Finish**.

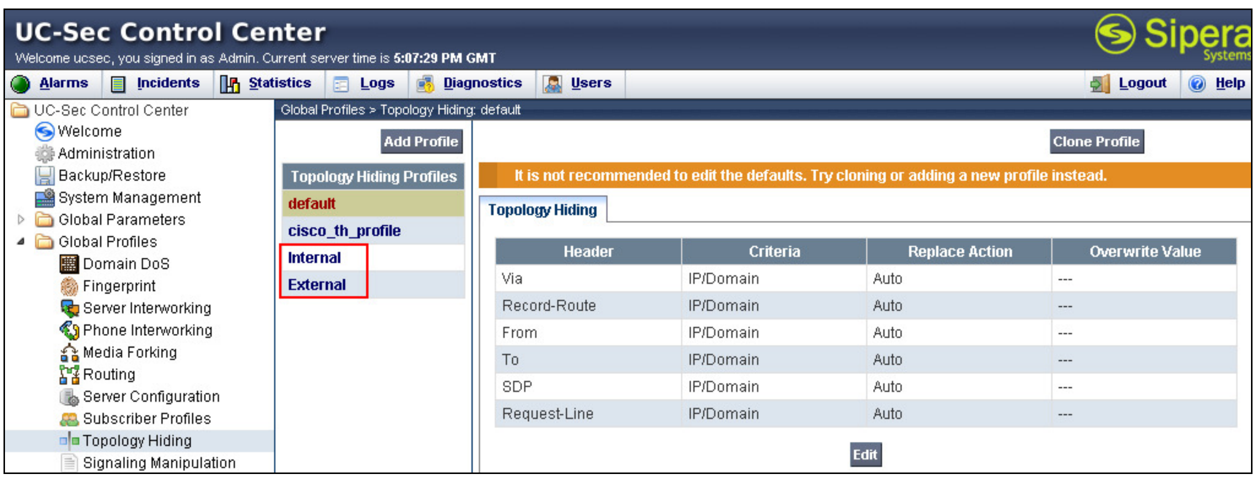
Similarly, configure a Topology Hiding Profile for the **External** interface.



Click **Finish** when completed.



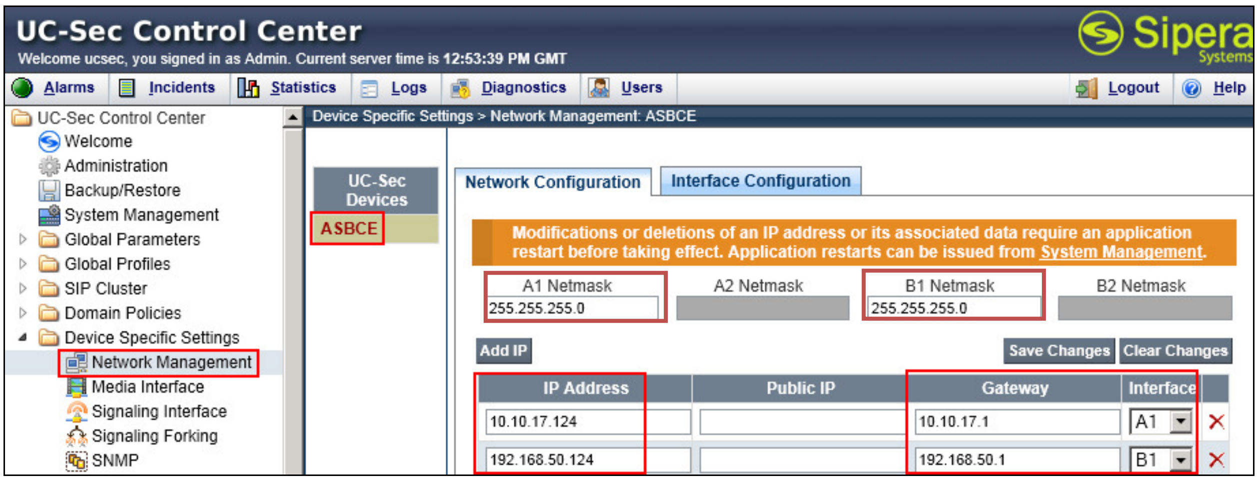
The screenshot below shows the newly configured Topology Hiding Profiles.



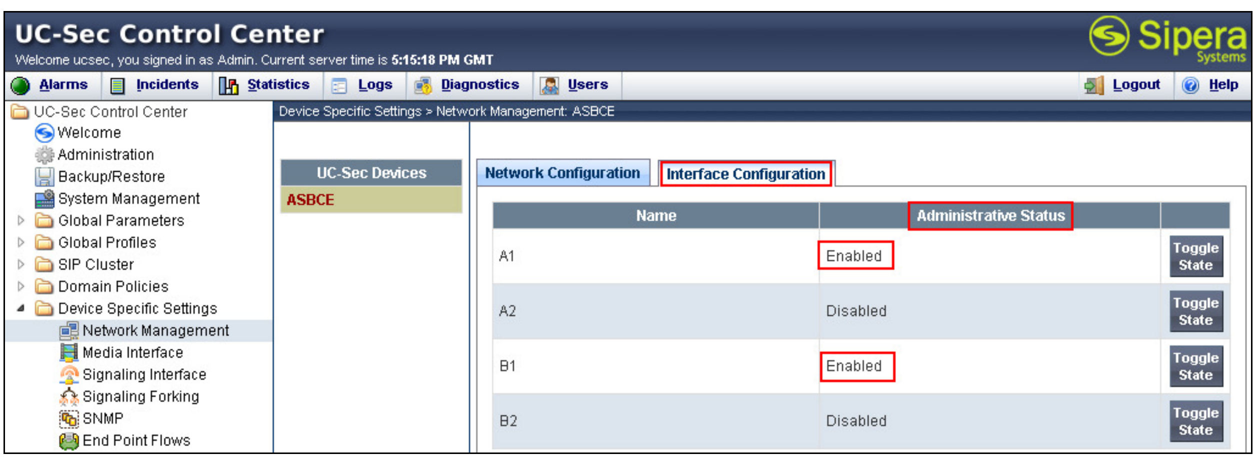
6.7. Device Specific Settings – Network Management

The network information should have been previously specified during installation of the ASBCE.

Navigate to **Device Specific Settings → Network Management** from the left-side menu. Under **UC-Sec Devices**, select the device being managed, which was named **ASBCE** in the sample configuration. The **Network Configuration** tab is shown below. Observe the **IP Address**, **Netmask**, **Gateway** and **Interface** information previously assigned. Note that only the **A1** and **B1** interfaces are used. Typically the A interfaces are used for the internal side and B interfaces are used for the external side of the ASBCE.



Select the **Interface Configuration** tab. The **Administrative Status** can be toggled between **Enabled** and **Disabled** in this screen. The following screen was captured after the interfaces had already been enabled. To enable the interface if it is disabled, click the **Toggle State** button.

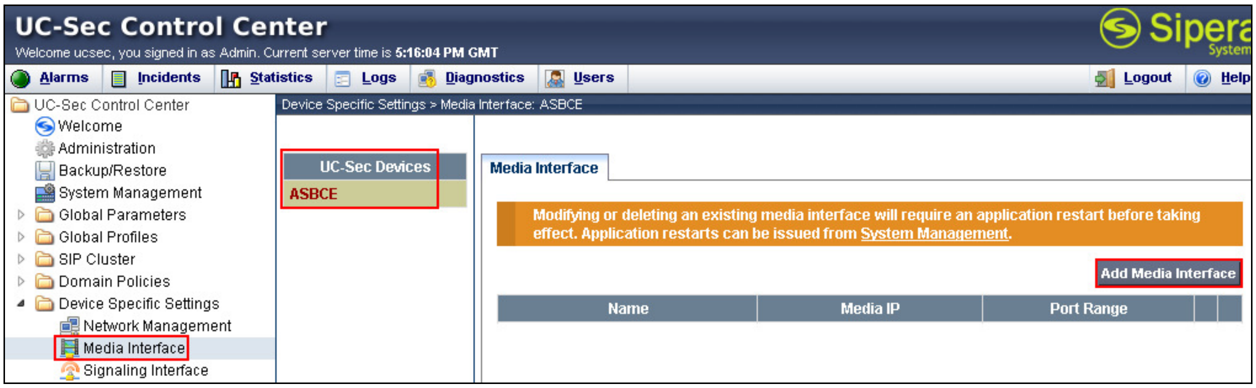


When IP addresses and network masks are assigned to interfaces, these are then configured as Signaling Interfaces and Media Interfaces.

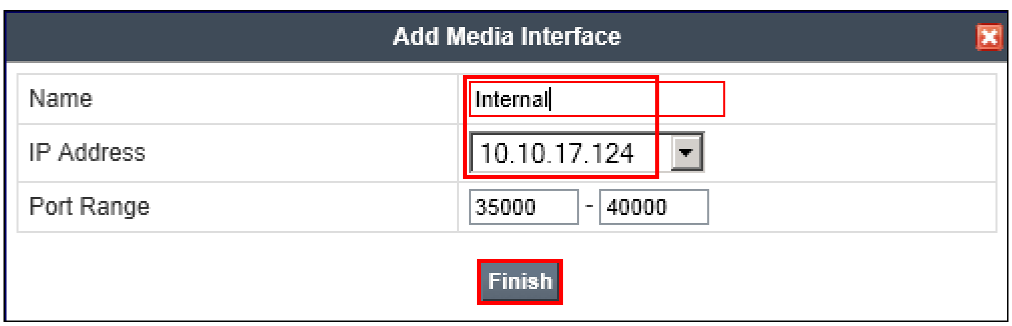
6.8. Device Specific Settings – Media Interface

Media Interfaces are created to adjust the port range assigned to media streams leaving the interfaces of ASBCE. The compliance test used the default port range of 35000 to 40000.

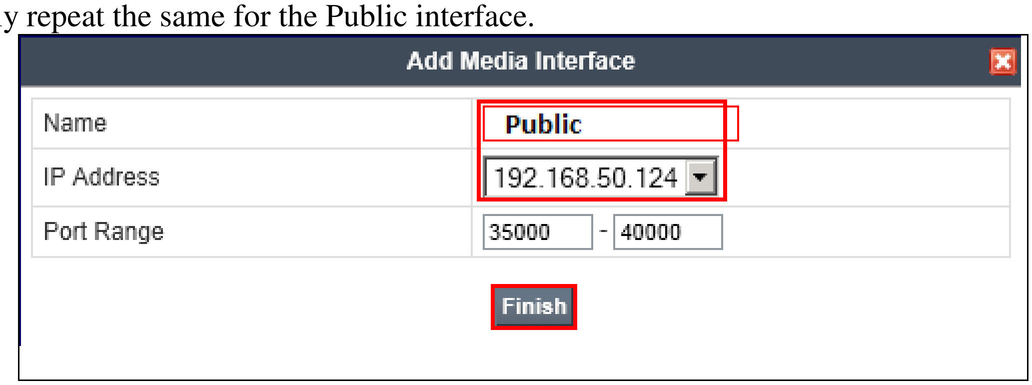
Navigate to **Device Specific Setting → Media Interface**. Under **UC-Sec Devices**, select the device being managed, which was named **ASBCE** in the sample configuration, and select **Add Media Interface**.



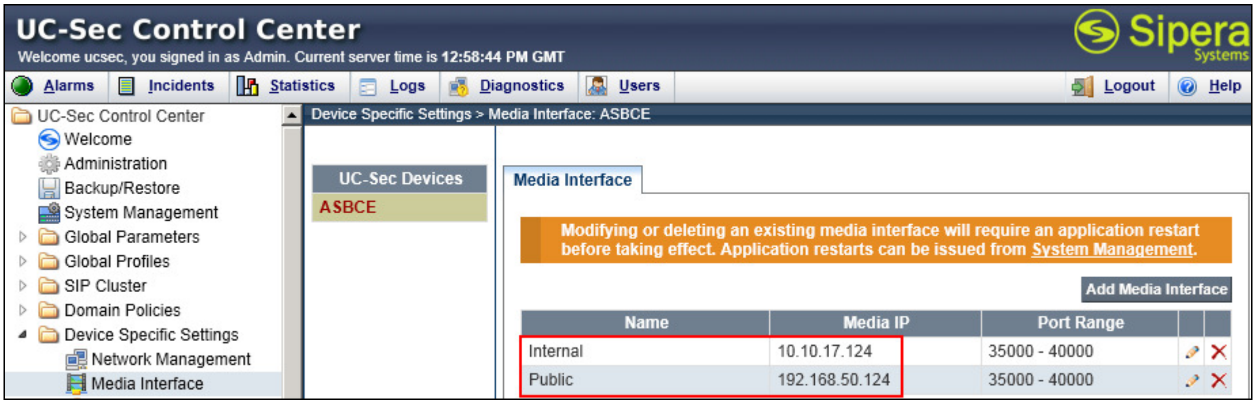
Enter an appropriate **Name** for the Media Interface facing the enterprise and select the inside private IP Address of the ASBCE from the **IP Address** drop-down menu. In the sample configuration, **Internal** was chosen as the **Name**, and the inside IP Address of the ASBCE is **10.10.17.124**. Leave the Port Range at its default value and click **Finish**.



Similarly repeat the same for the Public interface.

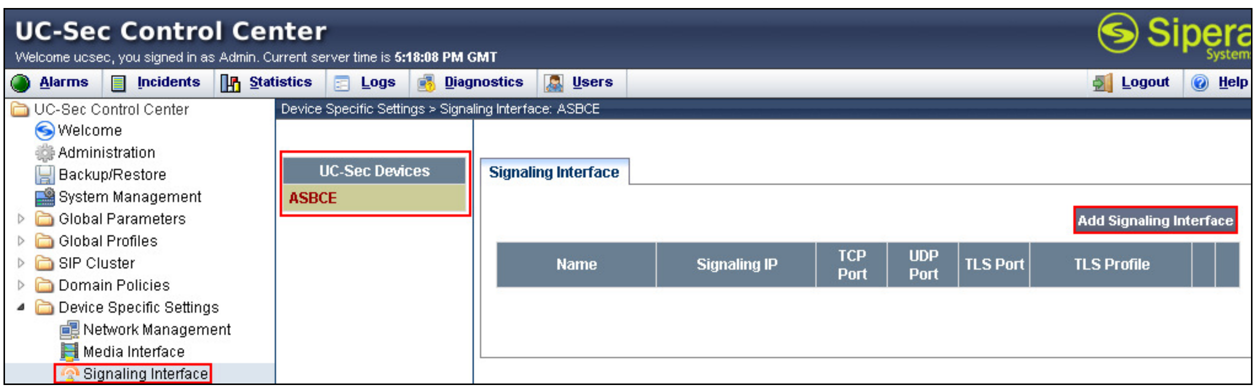


The screenshot below shows the newly configured Media Interfaces.

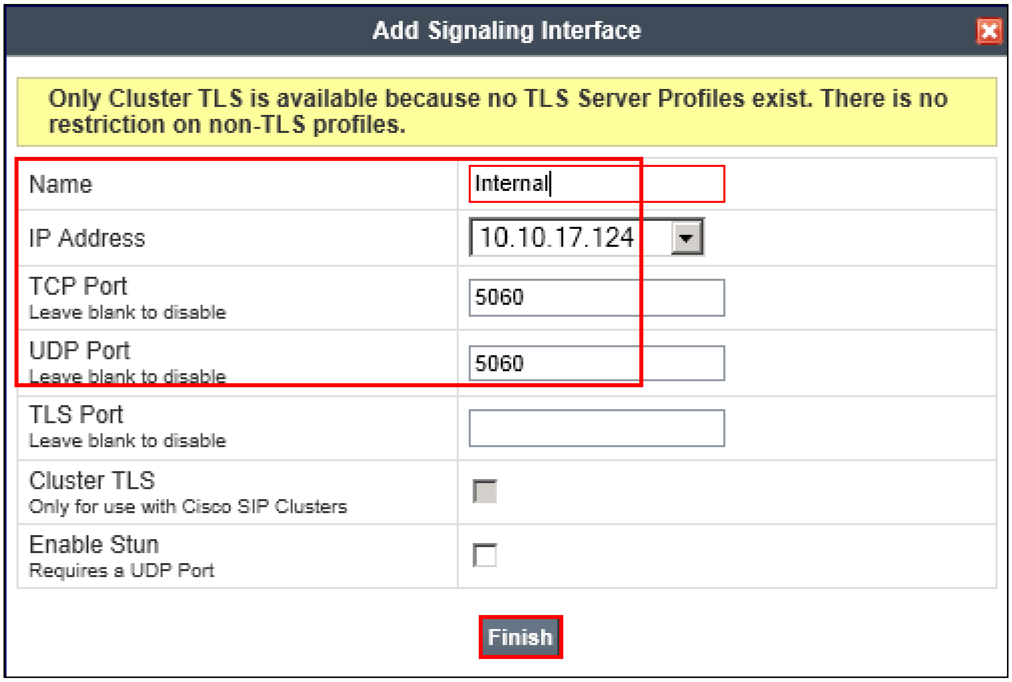


6.9. Device Specific Settings – Signaling Interface

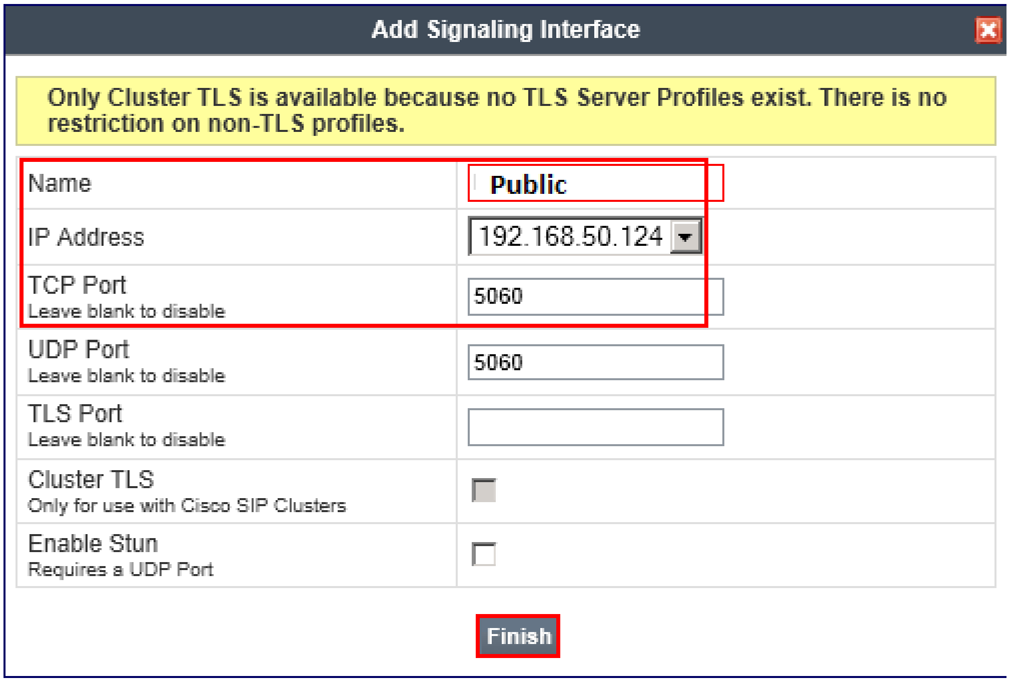
Navigate to **Device Specific Settings → Signaling Interface**. Under **UC-Sec Devices**, select the device being managed, which was named **ASBCE** in the sample configuration and select **Add Signaling Interface**.



In the **Add Signaling Interface** screen, enter an appropriate **Name** for the inside interface, and choose the private, inside IP Address of the ASBCE from the **IP Address** drop-down menu. Enter **5060** for the **UDP Port** since **UDP** port **5060** is used for the SIP connection between Semafone and ASBCE in the sample configuration. Click **Finish**.



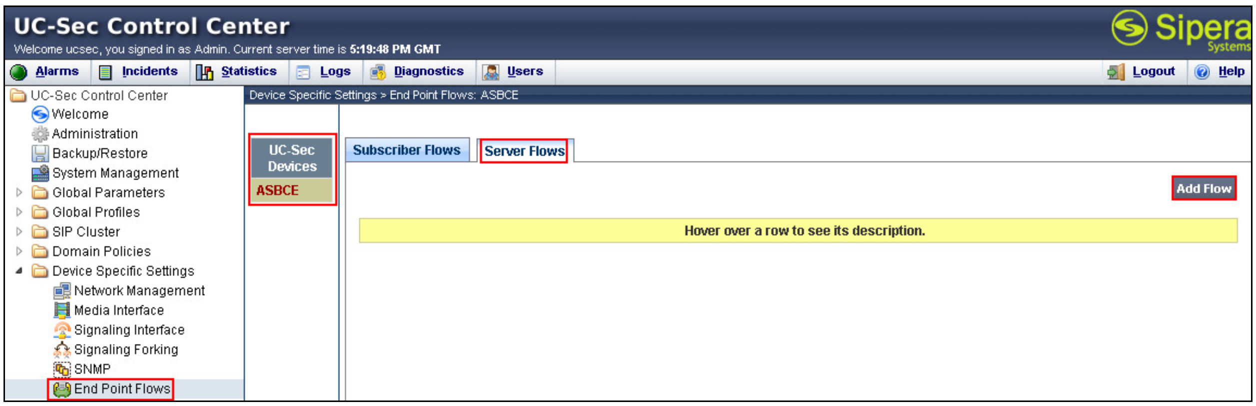
Similarly enter an appropriate **Name** for the external interface, and choose the public, external IP Address of the ASBCE from the **IP Address** drop-down menu. Enter **5060** for the **TCP Port since TCP** port **5060** is used for the SIP Trunk between ASBCE and the Simulated SIP Service Provider in the sample configuration. Click **Finish**.



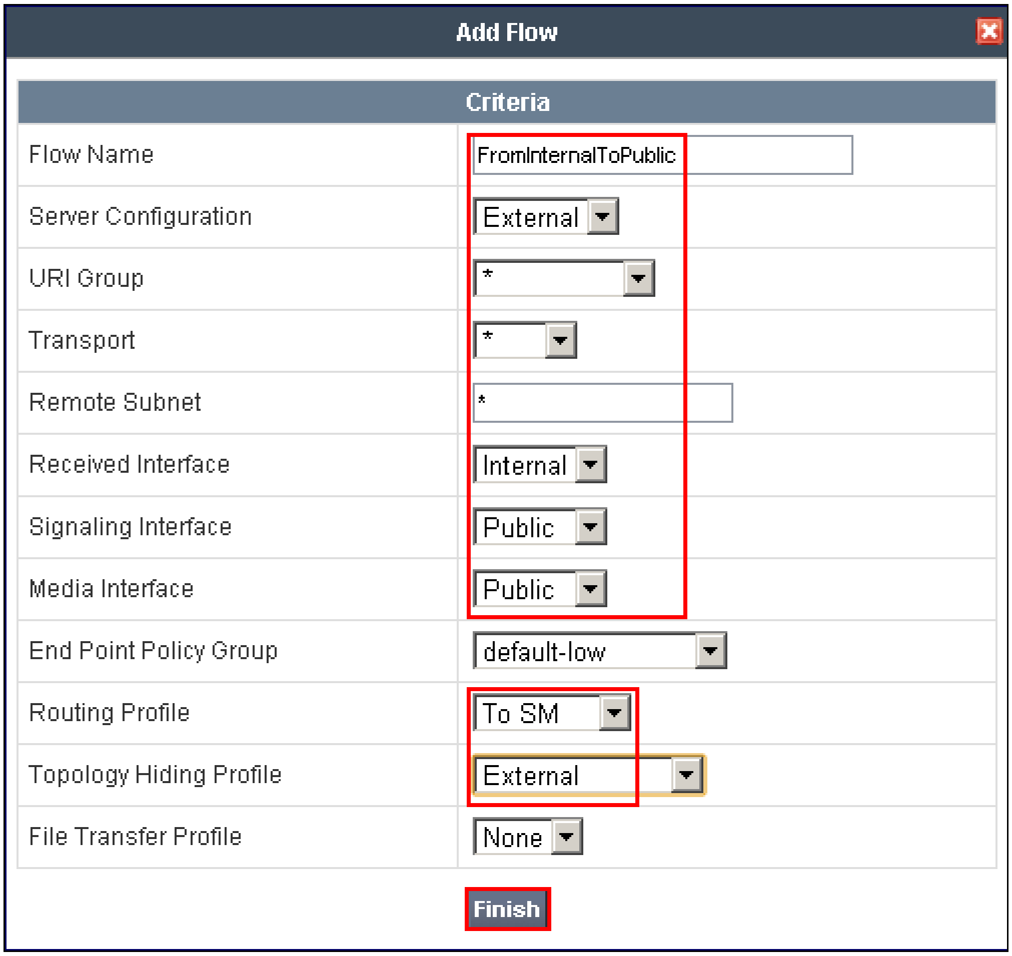
6.10. Device Specific Settings – End Point Flows

End Point Flows combine the previously defined profiles into an outgoing flow from the Call Server (Semafone) to the Trunk Server (Simulated SIP Service Provide) and an incoming flow from the Trunk Server to the Call Server. This configuration ties all the previously entered information together so that calls can be routed from Semafone to the Simulated SIP Service Provider and vice versa.

Select **Device Specific Settings → End Point Flows**. Under **UC-Sec Devices**, select the device being managed, which was named **ASBCE** in the sample configuration, and select the **Server Flows** tab. Select **Add Flow**.

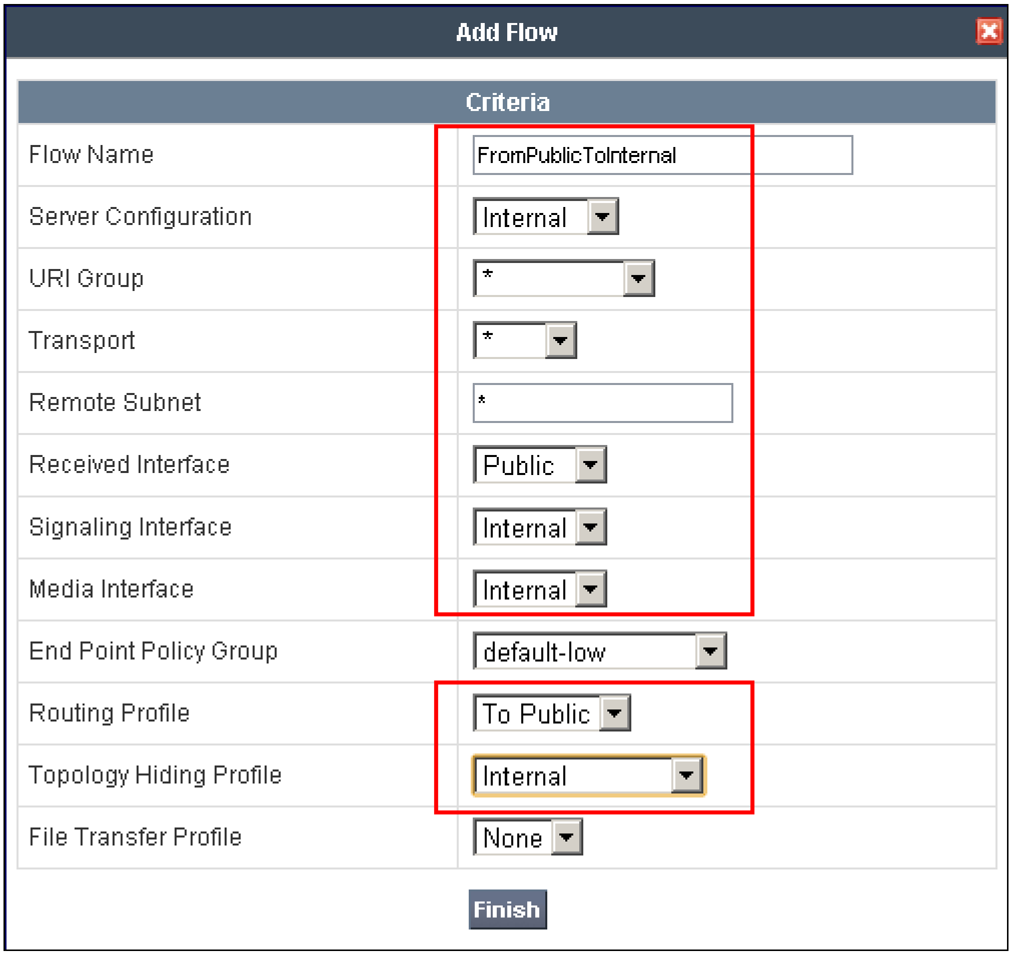


Configure a server flow from the internal network to the public network.



- Flow Name FromInternalToPublic [described as "Semafone to the Simulated SIP Service Provider and vice versa"]
- Server Configuration: **External:** Simulated SIP Service Provider **192.168.50.16**
- Received Interface: **Internal:**
- Signaling Interface: **Public**: SBCE Public Interface **192.168.50.124**
- Media Interface: **Public:** SBCE Public Interface **192.168.50.124**
- Routing profile: **To SM:** Semafone 'Dirty' Interface **10.10.17.195**
- Topology Hiding Profile: **External**

Similarly, configure a server flow from the public to the internal network.

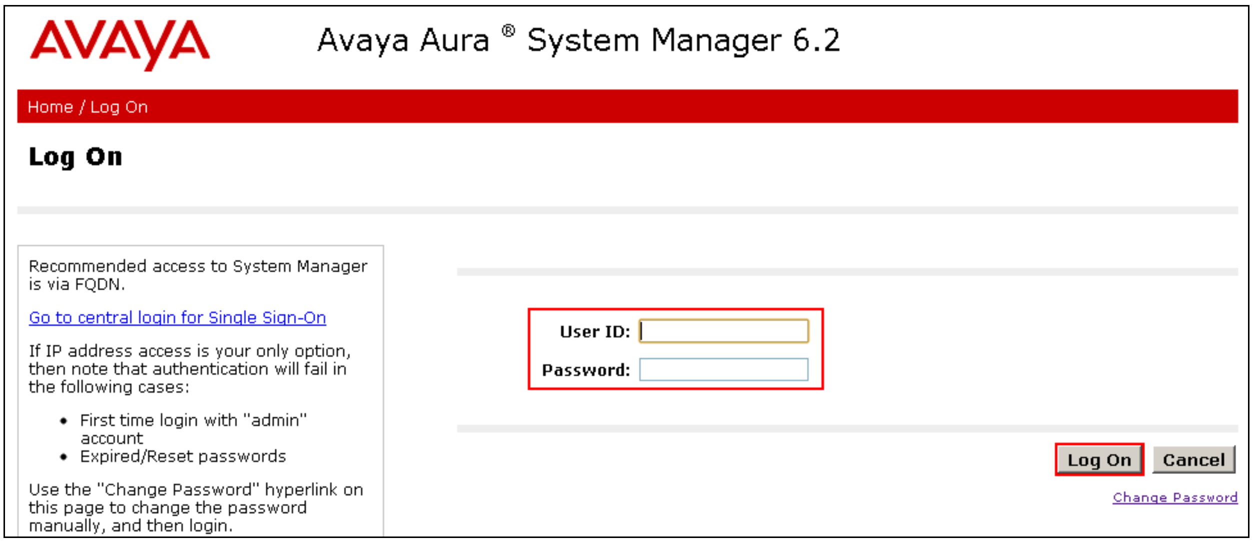


- Flow Name FromPublicToInternal [described as "Semafone to the Simulated SIP Service Provider and **vice versa**"]
- Server Configuration: **Internal:** Semafone SIG interface **10.10.17.195**
- Received Interface: **Public:**
- Signaling Interface: **Internal**: inside IP Address of the ASBCE **10.10.7.124**
- Media Interface: **Internal:** inside IP Address of the ASBCE **10.10.7.124**
- Routing profile: **To Public:** Simulated SIP Service Provider **192.168.50.16**
- Topology Hiding Profile: **Internal**
-

7. Configure Avaya Aura® Session Manager

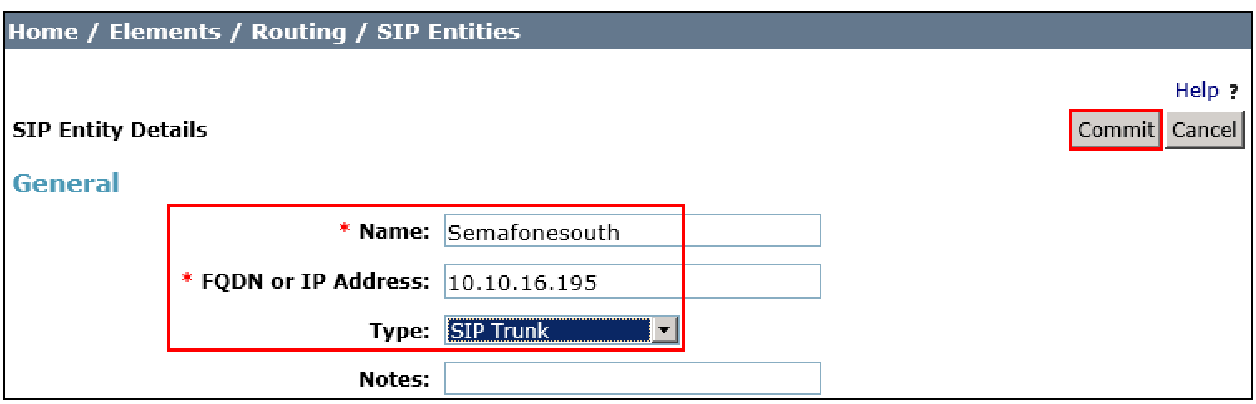
This section illustrates relevant aspects of the Session Manager configuration required for interoperating with Semafone.

Session Manager is managed via System Manager. Using a web browser, access **https://<ip-addr of System Manager>/SMGR**. In the **Log On** screen, enter appropriate **User ID** and **Password** and press the **Log On** button.



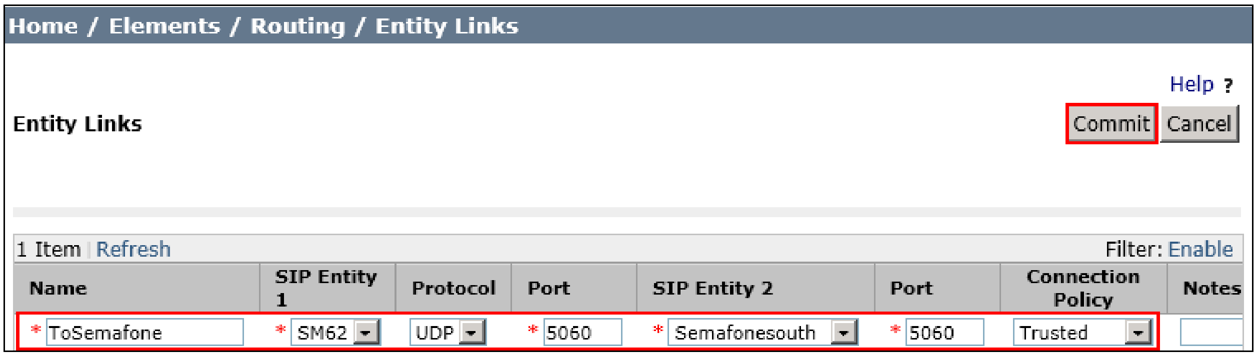
7.1. Configure Semafone SIP Entity

A SIP Entity must be created for the Semafone SIG interface. Click **Routing → SIP Entities → New**. Assign an identifying **Name** and the **FQDN or IP Address** for the Semafone clean interface, set the **Type** field to **SIP Trunk**, and click **Commit** when done.



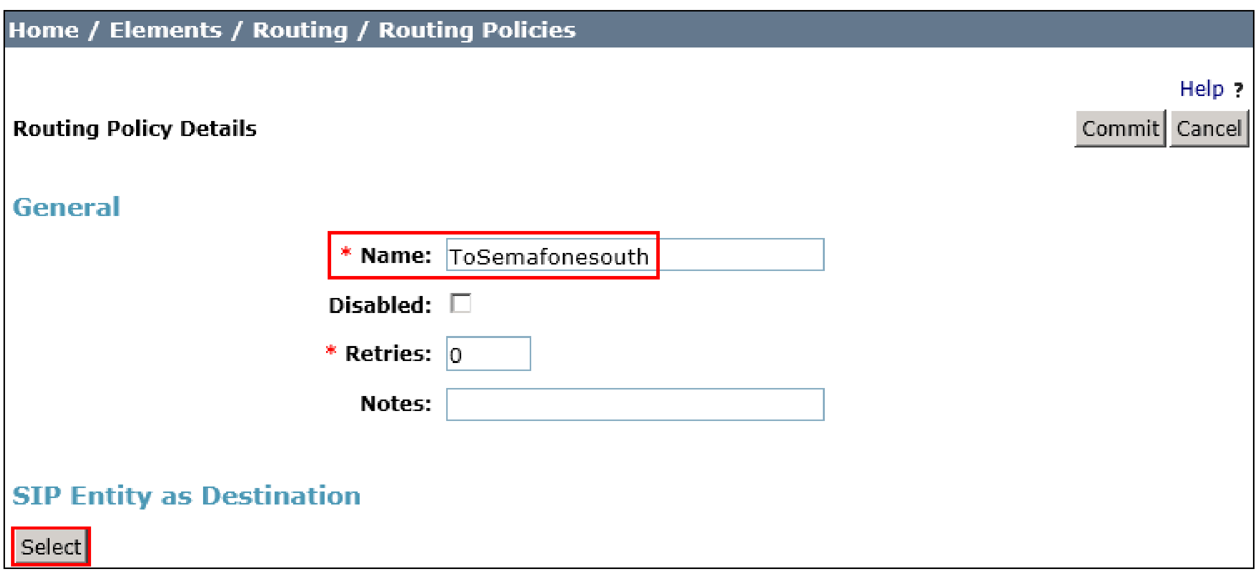
7.2. Configure Entity Link

The configuration of an Entity Link connects the Session Manager SIP Entity with the Semafone SIP Entity. Click **Routing → Entity Links → New**. Assign an identifying **Name**, choose the entity assigned to the preconfigured Session Manager SIP Signaling Interface as **SIP Entity 1**, set the **Protocol** as **UDP**, enter **5060** for the **Port**, choose the Semafone SIP entity as **SIP Entity 2**, set the **Port** to **5060**, and select **Trusted** from the **Connection Policy** drop down box. Click **Commit** when done. This establishes the Session Manager end of the SIP Trunk to Semafone.

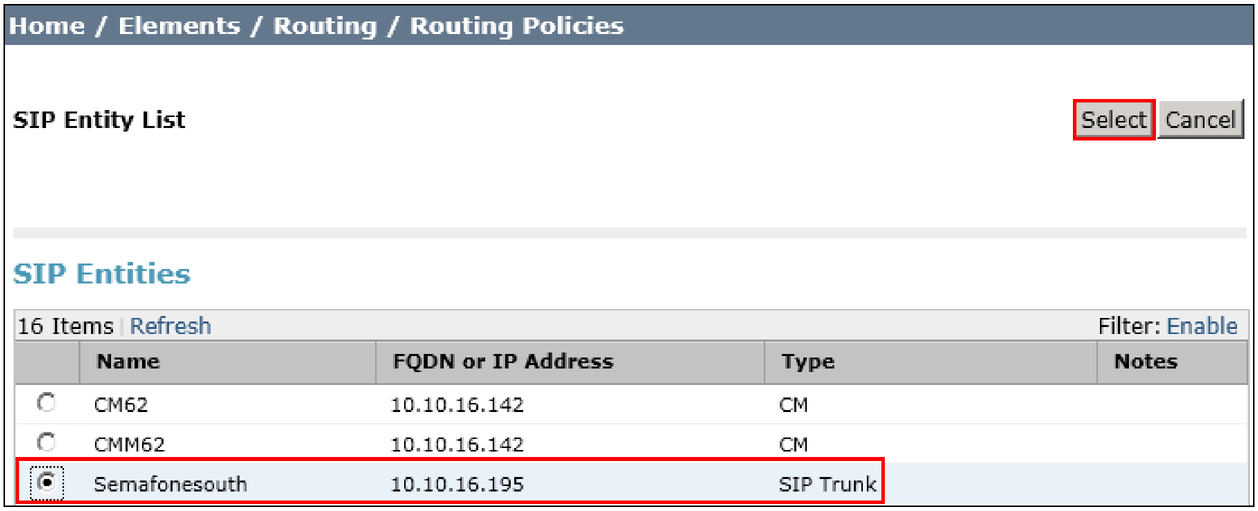


7.3. Create Routing Policy

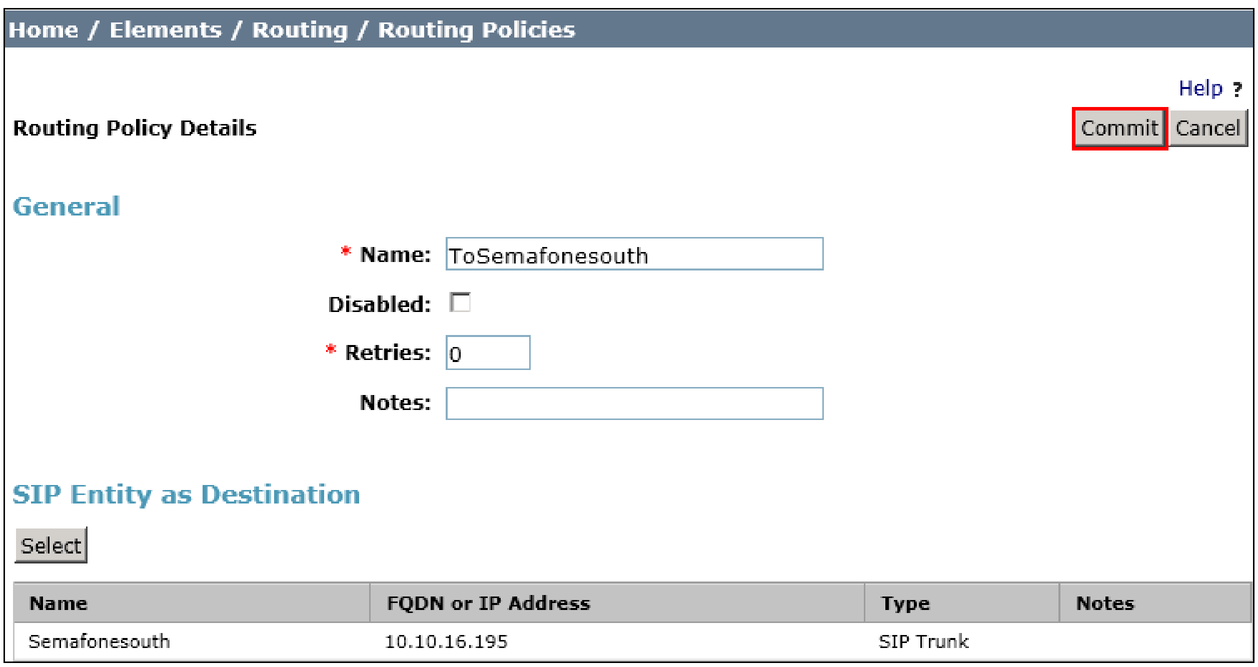
Click **Routing → Routing Polices → New**. Assign an indentifying **Name** for the route. Under the **SIP Entity as Destination** section, click on **Select**.



Choose the Semafone Entity configured in **Section 7.1** and click **Select**.

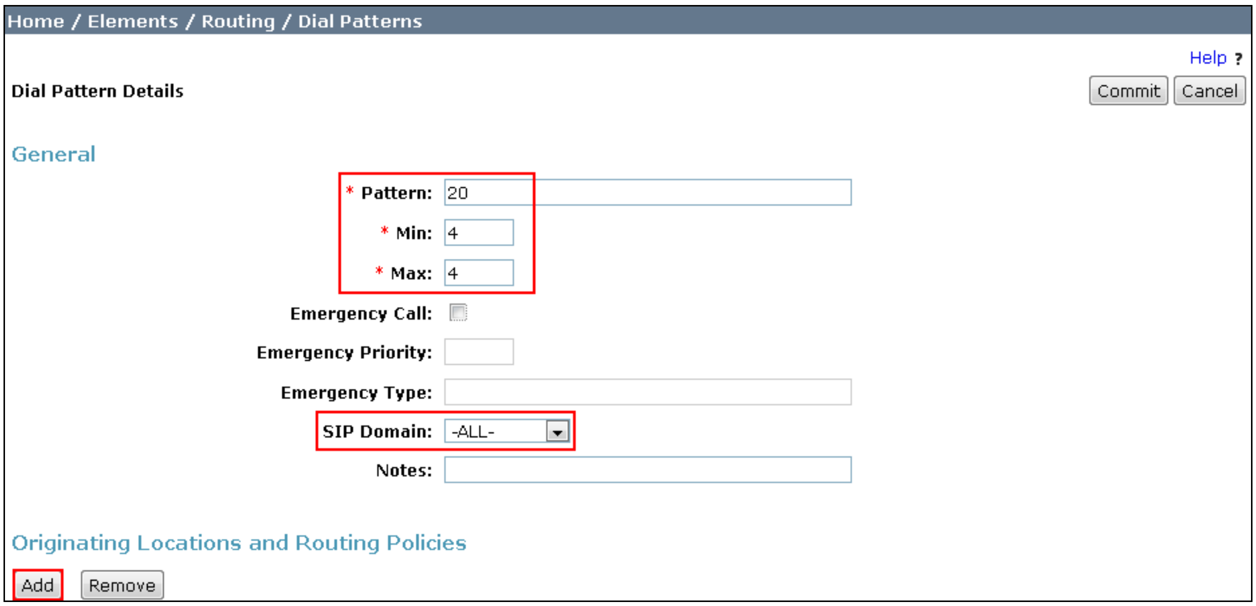


Review the configuration and click **Commit** when done.

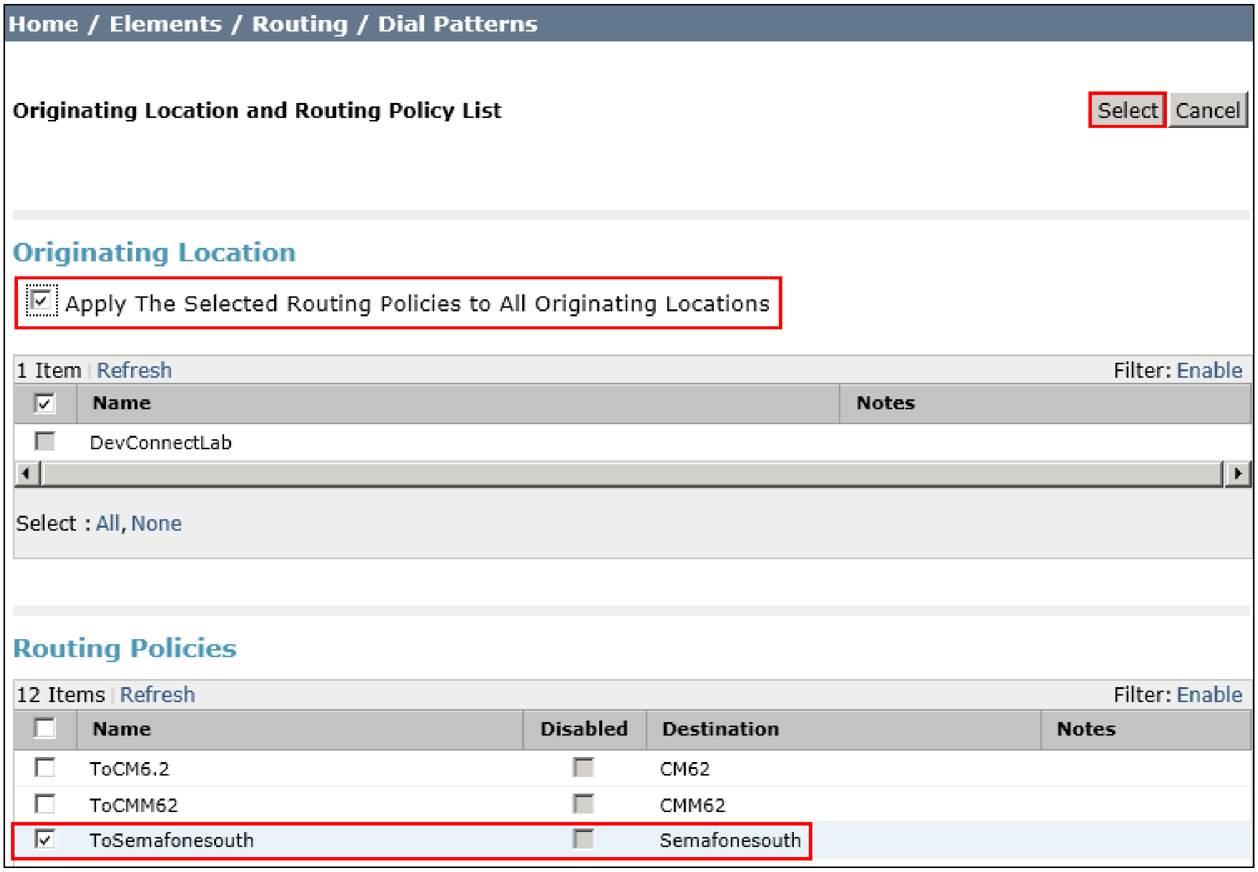


7.4. Administer Dial Patterns

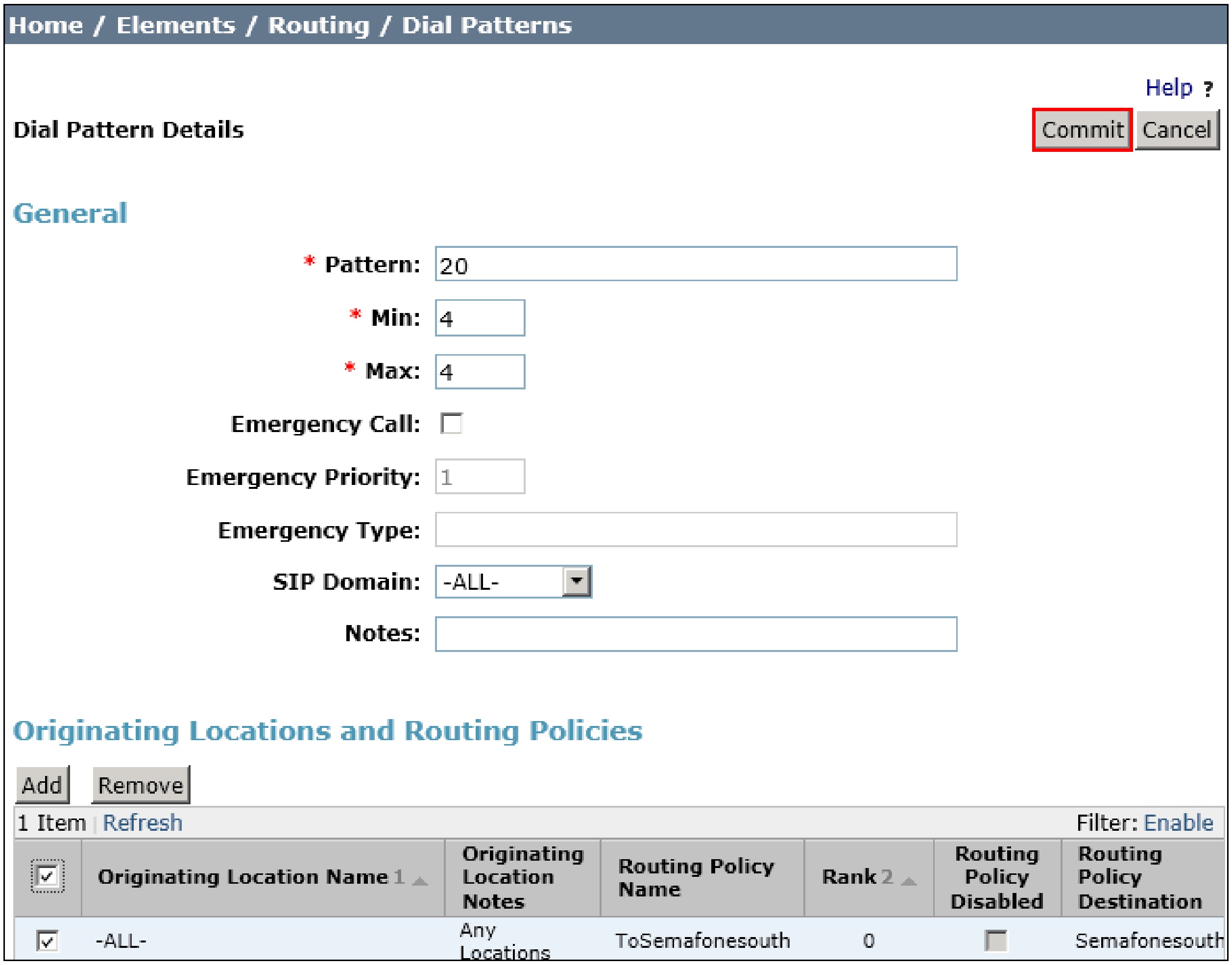
Session Manager routes SIP traffic between connected devices. Dial Patterns are created as part of the configuration to manage SIP traffic routing, which will direct calls based on the number dialed to the appropriate destination. In **Section 5.1** Communication Manager is configured to route 4-digit strings beginning with 20 to Session Manager. To create a Dial Pattern to route these digits from Session Manager to Semafone click **Routing → Dial Patterns → New**. Under **General** enter the number presented to Session Manager by Communication Manager in the **Pattern** box. Set the **Min** and **Max** digit string length, and set **SIP Domain** to **-ALL-**. In the **Originating Locations and Routing Policies** section of the web page, click **Add**.



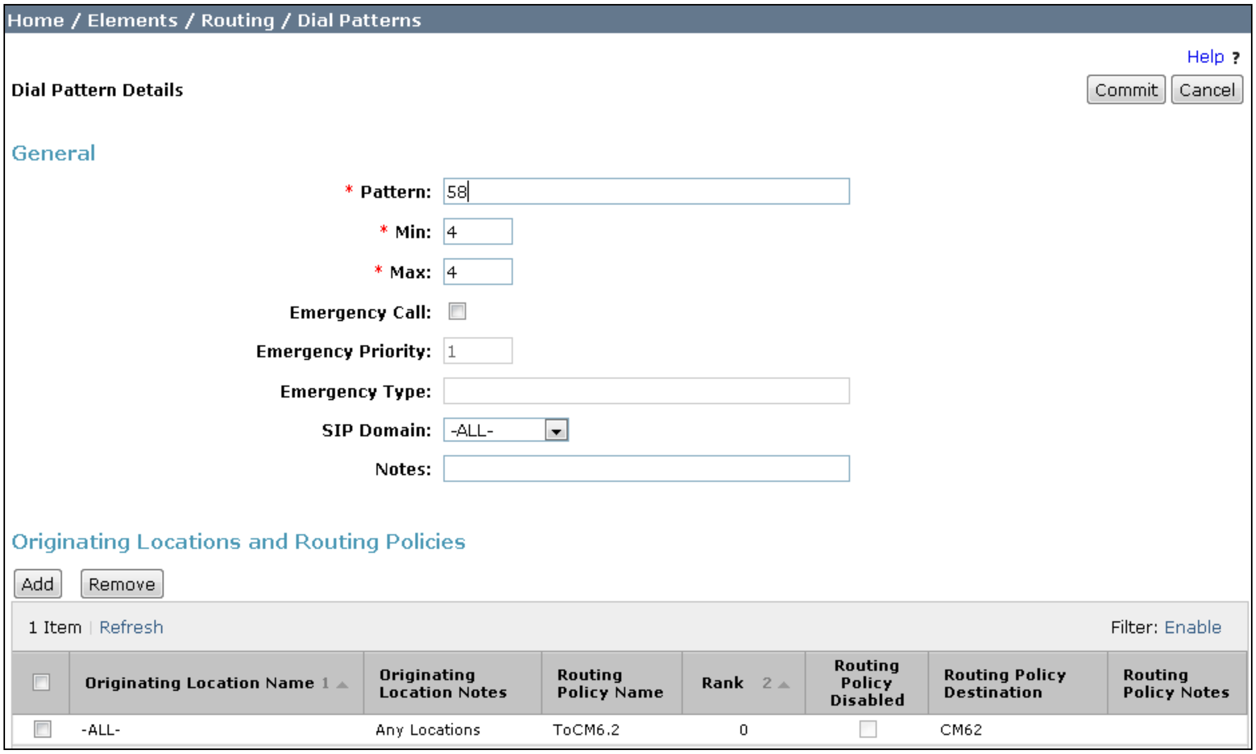
Place a tick in the **Apply The Selected Routing Policies to All Originating Locations** tick box, and select the **Routing Policy** created in **Section 7.3**. Click **Select** when done



Review the configuration and click **Commit** when done.



Similarly, configure the appropriate Dial Patterns for routing calls arriving via ASBCE from the PSTN to Communication Manager. For the purpose of the compliance testing, Dial Patterns were administered to route 58xx, (shown below), and 6xxx (not shown) digit strings to Communication Manager.



8. Configure Semafone

The Semafone solution is installed, configured and commissioned directly by Semafone. The following items summarise the configuration options selected which are pertinent to the interworking scenarios discussed. Additional configuration, unrelated to interworking with the Avaya solution components, is required, but is not detailed here. No unusual settings are required by Semafone for successful interworking.

8.1. Configure Interfaces

The two external interfaces of the SIG must be configured to take the correct IP addresses for the networks into which they will be deployed. In the file **/etc/network/interfaces**, **eth1** represents the "clean" interface, and **eth0** represents the "dirty" interface. Standard IPv4 parameters are specified here and referenced in **Figure 1**:

```
auto eth0
iface eth0 inet static
      address 10.10.17.195
      netmask 255.255.255.0
      network 10.10.17.0
      broadcast 10.10.17.255

auto eth1
iface eth1 inet static
      address 10.10.16.195
      netmask 255.255.255.0
      network 10.10.16.0
      broadcast 10.10.16.255
```

8.2. Telephony Configuration - SIG

In addition to standard SIG configuration, the following entry must be present in the general section of the SIG's **/etc/semafone/sip.conf** configuration file in order to define the UDP port on which the SIG will receive SIP signaling (this is a global setting; if this must be changed then the CCM configuration must be adapted to match)..

```
[general]
…
bindport=5060
```

The two entities with which the SIG will communicate must also be defined in this configuration file; one for the “dirty” interface connecting to ASBCE and one for the “clean” interface” connecting to Session Manager. For both entities, the procedure is the same:

1) Define IP address of ASBCE and Session Manager in the **host** and **permit** entries – in this case **10.10.17.124** and **10.10.16.148** respectively.
2) Specify the required DTMF interworking in the **dtmfmode** entry (**inband**, **info**, or **rfc2833**)
3) Select the appropriate G.711 encoding via the **allow** entry (either **alaw** or **ulaw**)

```
[public_dirty]
insecure=invite
type=peer
allow=alaw
deny=0.0.0.0/0.0.0.0
host=10.10.17.124
permit=10.10.17.124/255.255.255.255
context=public_dirty
dtmfmode=inband

[public_clean]
insecure=invite
type=peer
allow=alaw
deny=0.0.0.0/0.0.0.0
host=10.10.16.148
permit=10.10.16.148/255.255.255.255

context=public_clean
dtmfmode=inband
```

8.3. Telephony Configuration – CCM

The Semafone CCM Virtual Machine captures all telephony events and updates the DPM Virtual Machine accordingly. The Semafone DPM Virtual Machine processes secure data and provides access to Semafone APIs. The DPM supports multiple software configurations to accommodate varying integration requirements for payment pages and payment provider integrations.

In order to support extended SIP message lengths seen in the Avaya environment, one change is required to the CCM configuration. On the CCM, in the file **/pkg/ccma/etc/ccm.cfg**, the entry:

```
    if (msg:len >= 2048) {
```

should be changed to:

```
    if (msg:len >= 4096) {
```

If the SIP port defined in the SIG **/etc/semafone/sip.conf** file above is changed from **5060**, then the following four entries in the **/pkg/ccma/etc/ccm.cfg** must be changed to reflect the new port number:

```
listen=udp:192.168.70.65:5060
listen=udp:192.168.60.65:5060
```

```
if (dst_ip==192.168.60.65) force_send_socket(192.168.70.65:5060);
if (dst_ip==192.168.70.65) force_send_socket(192.168.60.65:5060);
```

8.4. Payment Page

No specific configuration of the payment page is required for the integration with Avaya. Due to the abstraction provided by the SIG and CCM components of the Semafone solution, customization of the payment page to meet specific requirements of the deployment is permissible, and will not impact the telephony integration tested here.

9. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the Avaya and Semafone solution.

9.1. Verify Communication Manager SIP Trunk

Using the SAT, enter the **status signaling-group <n>** command, where <n> is the number of the SIP signaling group which connects to Session Manager. Verify that the signaling **Group State** is **in-service**.

```
status signaling-group 1
                         STATUS SIGNALING GROUP

        Group ID: 1
      Group Type: sip

     Group State: in-service
```

9.2. Verify Entity Link to Semafone

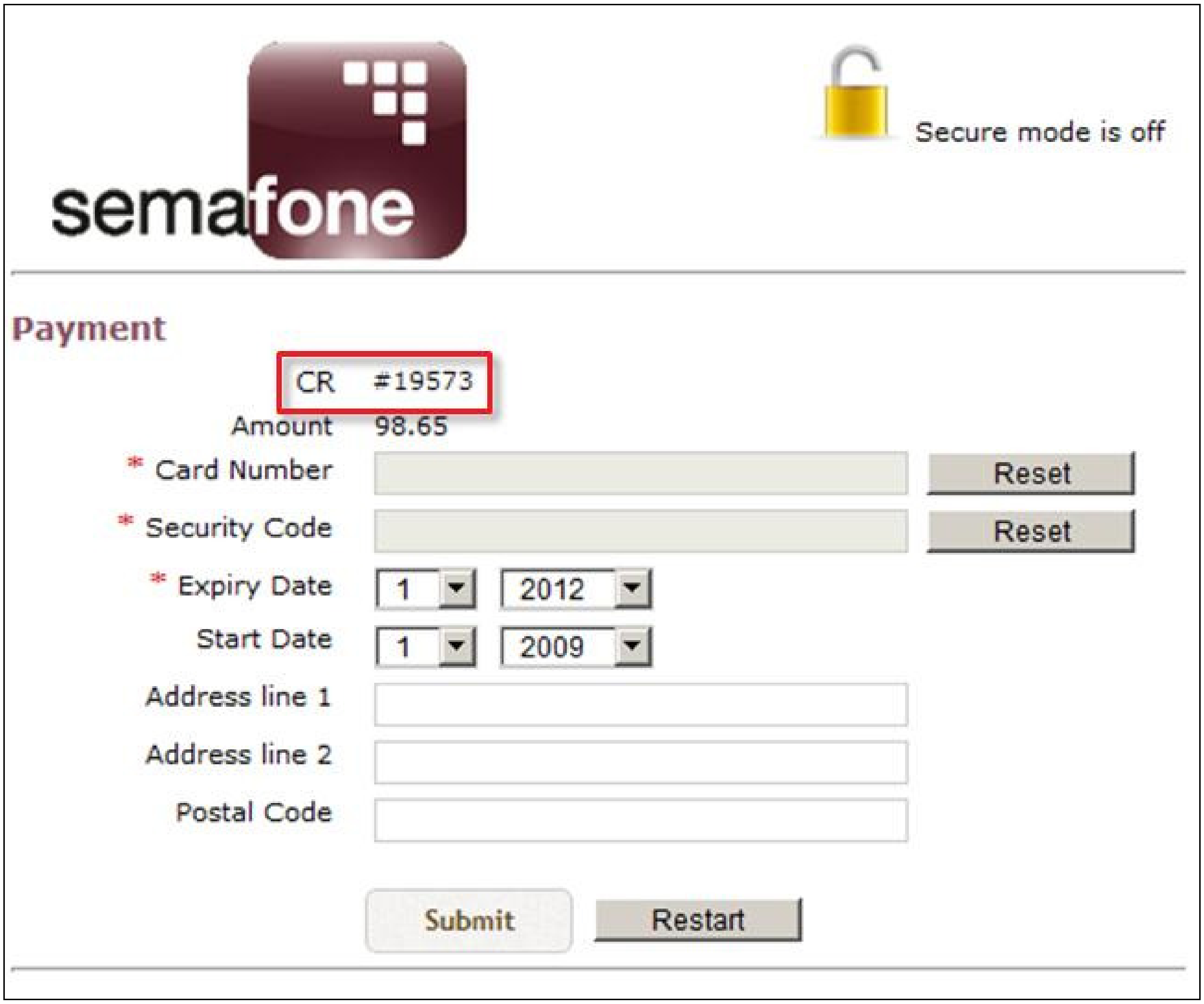
From the System Manager web interface click **Home → Session Manager → System Status → SIP Entity Monitoring**. Click on the entity configured for the Semafone SIP Entity in **Section 7.1** and confirm the **Conn. Status** is **Up**, the **Reason Code** is **200 OK** and the **Link Status** is **Up**.

1 Item | Refresh Filter: Enable

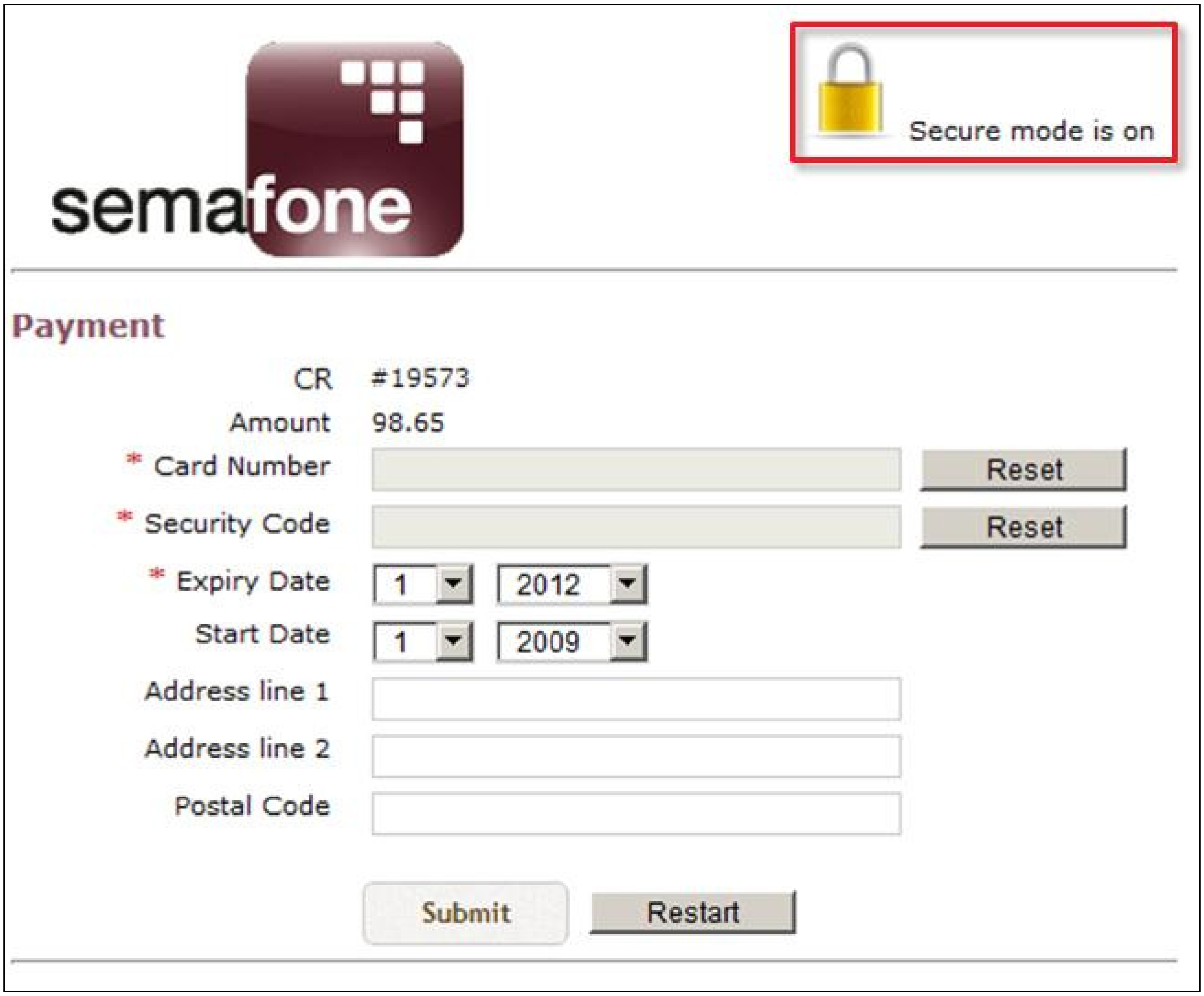
Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶ Show	SM62	10.10.16.195	5060	UDP	Up	200 OK	Up

9.3. Verify Call Routing, Semafone DTMF Manipulation and Semafone Payment Page Operation

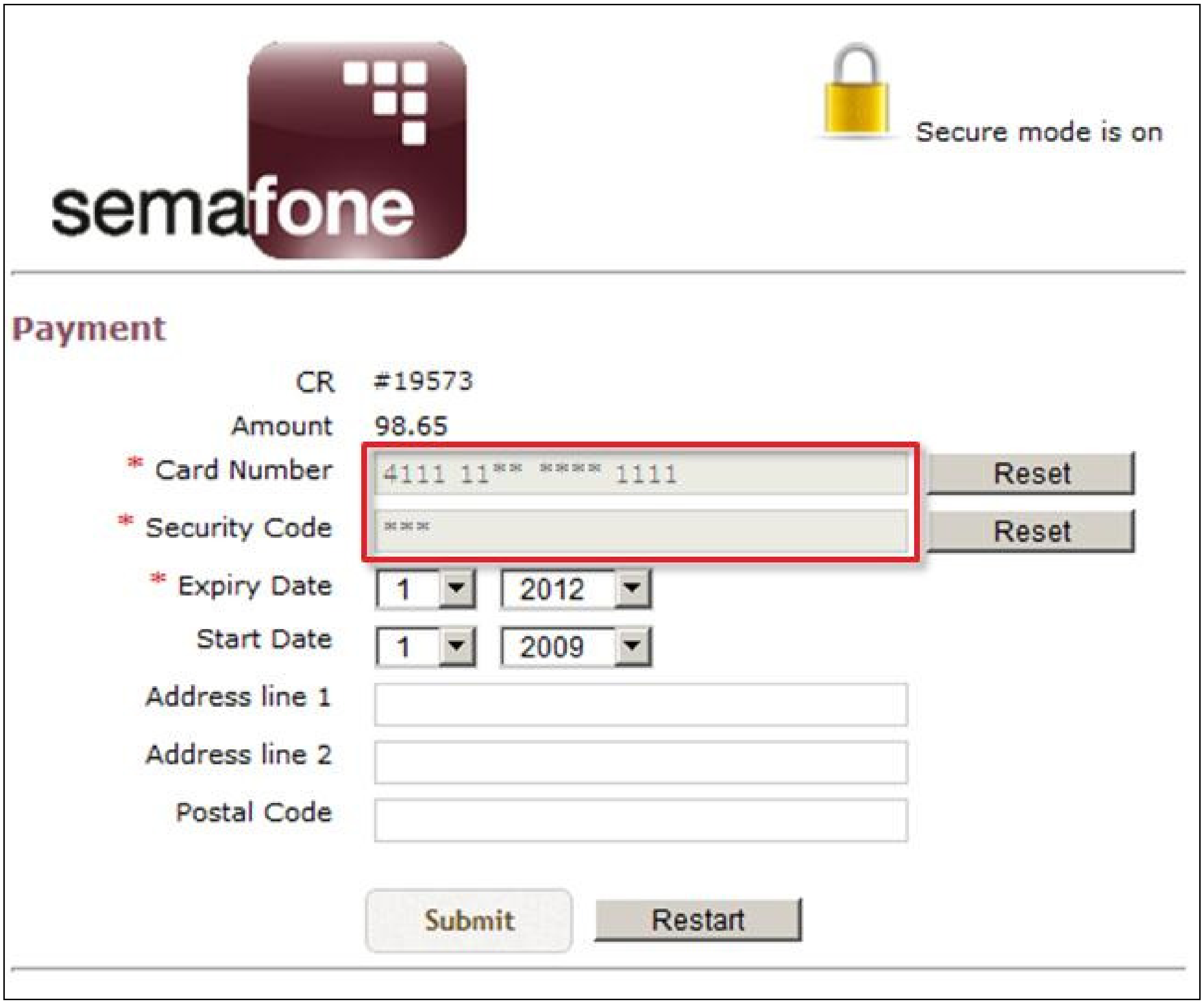
Place a call to/from the PSTN, ensure the call can be answered, controlled and terminated by a call center agent. When the agent receives a call, the agent navigates to the simulated payment page, retrieves the code displayed in the **CR** field, and enters the code on the telephone keypad.



Verify the padlock icon changes indicating the secured state has been entered.



Enter the appropriate card number using the keypad on the customer telephone and ensure the correct digits and number of digits are accurately captured on the payment page.



Verify the agent hears only a generic DTMF tone, and not that of the actual card number entered.

10. Conclusion

These Application Notes describe the configuration steps required for Semafone to successfully interoperate with Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise. All functionality cases were completed successfully with any observations noted in **Section 2.2**.

11. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com

Details for the configuration of Semafone can be obtained from the following:
http://www.semafone.com

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.