Avaya Solution & Interoperability Test Lab

Application Notes for Telecommunications Services of Trinidad and Tobago SIP Trunking Service with Avaya IP Office Release 9.0 and Avaya Session Border Controller for Enterprise Release 6.2 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Telecommunications Services of Trinidad and Tobago Session Initiation Protocol (SIP) Trunking Service with Avaya IP Office Release 9.0 and Avaya Session Border Controller for Enterprise Release 6.2.

Telecommunications Services of Trinidad and Tobago SIP Trunking Service provides PSTN access via a SIP Trunk between the enterprise and Telecommunications Services of Trinidad and Tobago network as an alternative to legacy analog or ISDN-PRI trunks. This approach generally results in lower cost for the enterprise.

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

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1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Telecommunications Services of Trinidad and Tobago and Avaya IP Office solution.

In the sample configuration, Avaya IP Office solution consists of Avaya IP Office (IP Office) 500v2 Release 9.0, Avaya Session Border Controller for Enterprise (Avaya SBCE) Release 6.2, Avaya IP Office Softphones and Avaya Deskphones, including SIP, H.323, digital, and analog endpoints. The Avaya SBCE provides security for the Avaya IP Office solution, as well as interoperability features for the SIP trunk.

Telecommunications Services of Trinidad and Tobago (TSTT) SIP Trunking Service referenced within these Application Notes is designed for business customers. Customers using this service with the Avaya IP Office solution are able to place and receive PSTN calls via a broadband WAN connection using SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI trunks. This approach generally results in lower cost for the enterprise

Telecommunications Services of Trinidad and Tobago will be referred to as **TSTT** here after.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using IP Office to connect to TSTT via the Avaya SBCE. This configuration (shown in **Figure 1**) was used to exercise the feature and functionality tests 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 TSTT SIP Trunking interoperability, the following features and functionalities were exercised during the compliance testing:

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various Avaya endpoints including SIP, H.323, digital and analog at the enterprise. All incoming calls from PSTN were routed to the enterprise across the SIP Trunk from the service provider networks.
- Outgoing PSTN calls from Avaya endpoints including SIP, H.323, digital and analog telephone at the enterprise. All outgoing calls to PSTN were routed from the enterprise across the SIP trunk to the service provider networks.
- Incoming and outgoing PSTN calls to/from Avaya IP Office Softphone.
- Incoming and outgoing PSTN calls to/from IP Office Flare® Experience for Windows.

- Dialing plans including long distance, international, outbound toll-free, etc.
- Caller ID presentation and Caller ID restriction.
- Codec's G.711MU and G.729A (For Codec G.729A Test Results refer to **Section 2.2**).
- Proper early media transmissions using G.711MU codec.
- DTMF tone transmissions per RFC 2833.
- Voicemail navigation for incoming and outgoing calls.
- Telephony features such as hold and resume, call transfer, call forward and conferencing.
- Off-net call forwards and transfers.
- Mobility Twinning of incoming calls to mobile phones.
- Response to incomplete call attempts and trunk errors.

2.2 Test Results

Interoperability testing with TSTT with was successfully completed with the exception of observations/limitations described below:

- **SIP REFER** – On PSTN calls to or from IP Office that are transferred back to the PSTN on the SIP trunk, TSTT responds with a "202 Accepted" to the REFER message sent by IP Office, but the call between the two PSTN endpoints drops, the PSTN phone receives re-order tone. REFER needs to be disabled in IP Office for the Call Transfer to complete successfully, otherwise the call transfer will fail. The implication is that IP Office SIP trunk channels are not released after the call transfer is completed, two (2) trunk channels will remain connected/busy for the duration of the call..
- **T.38 or G.711 Pass-Through fax calls** – With IP Office **Fax Transport Support** set as **T.38** or **T.38 Fallback** on the **SIP Line/VoIP**, on outbound calls (IPO→PSTN) TSTT did not send a re-INVITE to switch from G.711 to T.38. TSTT's recommendation is **not** to use T.38 fax transport, only G.711 fax Pass-through. With IP Office **Fax Transport Support** set as **G.711** on the **SIP Line/VoIP**, fax calls were unsuccessful, thus **T.38 or G.711** fax transports **are not** recommended for this solution.
- **Codec G.729A** – TSTT supports codec's G.711MU and G.729A, but during the testing, TSTT was rejecting calls with G.729A codec offer with **488 Invalid Media Type**. This issue is under investigation by TSTT.
- **Direct Media** – With Direct Media enabled in IP office, when calling IVR systems (or any recorded messaging system) from IP Office, a noticeable clipping of the recorded message is heard when IP Office sends the re-Invite to establish the direct media connection to the IP Phone. Testing was done with Direct Media disabled in IP Office. This issue is being investigated by Avaya.
- **Outbound Calling Party Number (CPN) Blocking** – On outbound calls from the enterprise to the PSTN with Calling Party Number Block (CPN) enabled on the IP Office station, TSTT responds with a **503 Service Unavailable**.
- **Call Forward Off-Net** – When inbound calls from the PSTN to IP Office are forwarded back out to another PSTN endpoint, TSTT responds with **503 Service Unavailable**, the reason is that TSTT is looking at the **Contact Header** instead of the **Diversion Header**. The work around for this issue is to set the **Send Caller ID** field under **SIP Line** to **None** instead **Diversion Header**.

2.3 Support

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

For technical support on TSTT SIP Trunking Service visit http://tstt.co.tt/

3. Reference Configuration

Figure 1 below illustrates the test configuration. It shows an enterprise site connected to the TSTT network through the public internet.

For confidentiality and privacy purposes, actual public IP addresses and PSTN routable phone numbers (DIDs) used during the compliance testing have been replaced with fictitious IP addresses and PSTN routable phone numbers throughout the Application Notes.

The Avaya components used to create the simulated enterprise customer site includes:

- Avaya IP Office 500v2.
- Avaya Session Border Controller for Enterprise.
- Avaya Voicemail Pro for IP Office.
- Avaya 9600 Series H.323 IP Telephones.
- Avaya 11x0 Series SIP IP Telephones.
- Avaya IP Office Softphone.
- IP Office Flare® Experience for Windows.
- Avaya 1408 Digital Telephones.
- Avaya 9508 Digital Telephones.

Located at the enterprise site is Avaya IP Office 500v2 with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codec's. The IP Office has **LAN1** port connects to the inside interface of the Avaya SBCE across the enterprise LAN (private) network. The outside interface of the Avaya SBCE connects to TSTT networks via the public internet.

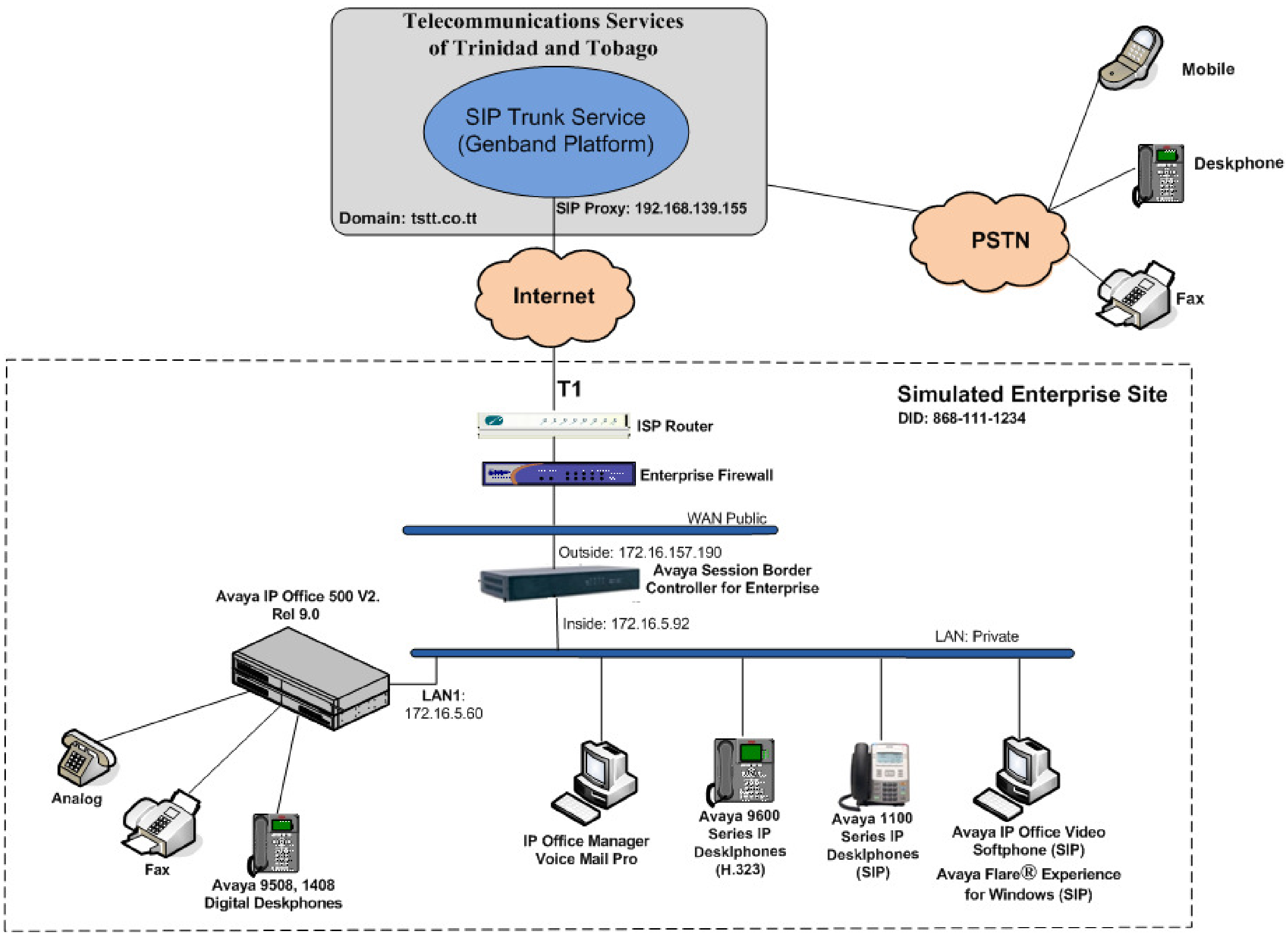


Figure 1: Avaya IP Telephony Network Connecting to TSTT SIP Trunking Service.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to TSTT. The short code 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to the network. Since Trinidad and Tobago is a country member of the North American Numbering Plan (NANP), the users dialed 10 digits for local calls, including the area code, and 11 (1 + 10) digits for other calls between the NANP.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the enterprise such as a Firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the service provider and the enterprise must be allowed to pass through these devices.

4. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration.

Avaya Telephony Components	
Equipment/Software	**Release/Version**
Avaya IP Office 500v2	9.0 (829)
Avaya IP Office DIG DCPx16 V2	9.0 (829)
Avaya IP Office Manager	9.0 (829)
Avaya Session Border Controller for Enterprise (running on Portwell CAD-0208 platform)	6.2 (6.2.0.Q48)
Avaya Voicemail Pro for IP Office	9.0 Built 311
Avaya 9620 IP Telephone (H.323)	Avaya one-X® Deskphone Edition S3.2
Avaya 1140 IP Telephone (SIP)	SIP1140 Ver. 04.03.18.00
Avaya IP Office Softphone	3.2.3.49 68975
IP Office Flare® Experience for Windows	1.1.4.23
Avaya Digital Telephones 1408	32
Avaya Digital Telephones 9508	0.45

Telecommunications Services of Trinidad and Tobago SIP Trunk Service	
Equipment/Software	**Release/Version**
Genband Softswitch	CVM13

Testing was performed with IP Office 500v2 R9.0, but it also applies to IP Office Server Edition R9.0. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0 to support analog or digital endpoints or trunks.

5. Configure IP Office

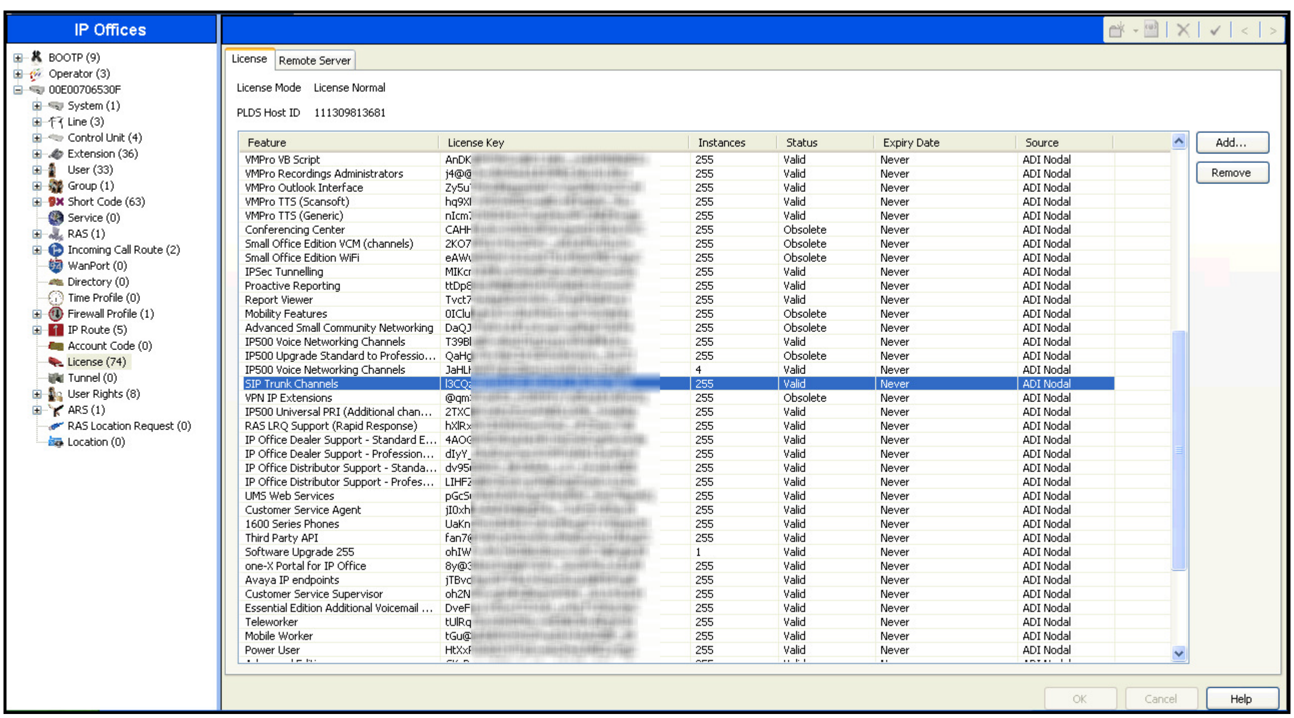
This section describes the IP Office configuration required to interwork with TSTT. IP Office is configured through Avaya IP Office Manager (IP Office Manager) which is a PC application. On the PC, select **Start → Programs → IP Office → Manager** to launch IP Office Manager. Navigate to **File → Open Configuration**, select the proper IP Office from the pop-up window, and log in with the appropriate credentials. A management window will appear as shown in the next sections. The appearance of IP Office Manager can be customized using the **View** menu (not shown). In the screenshots presented in this section, the **View** menu was configured to show the **Navigation Pane** on the left side and the **Details Pane** on the right side. These panes will be referenced throughout these Application Notes.

These Application Notes assume the basic installation and configuration have already been completed and are not discussed here. For further information on IP Office, please consult References in **Section 10**.

5.1 Licensing

The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

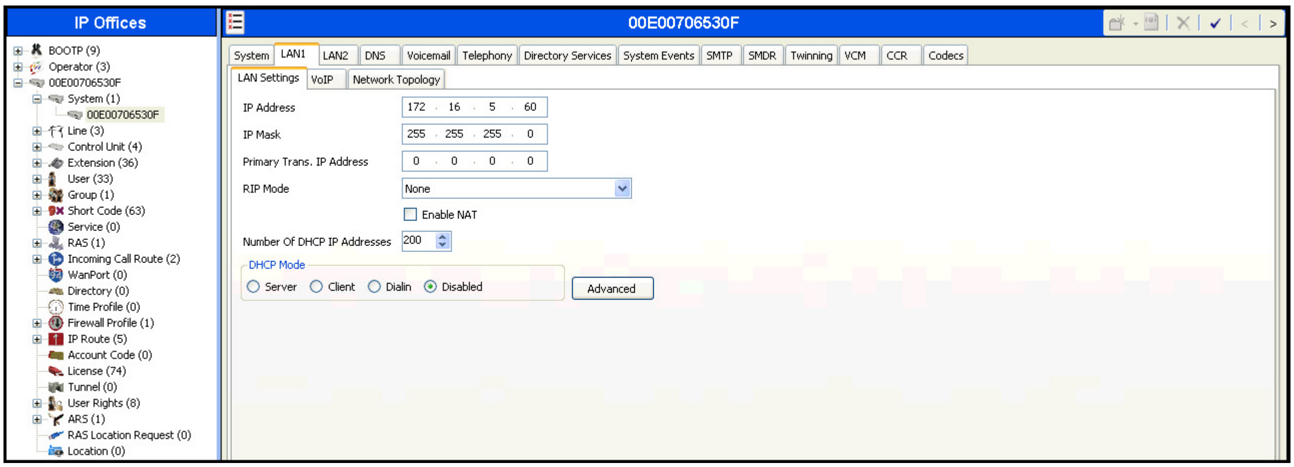
To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm that there is a valid license with sufficient "Instances" (trunk channels) in the Details pane. Note that the actual License Keys in the screen below were edited for security purposes.



5.2 LAN1 Settings

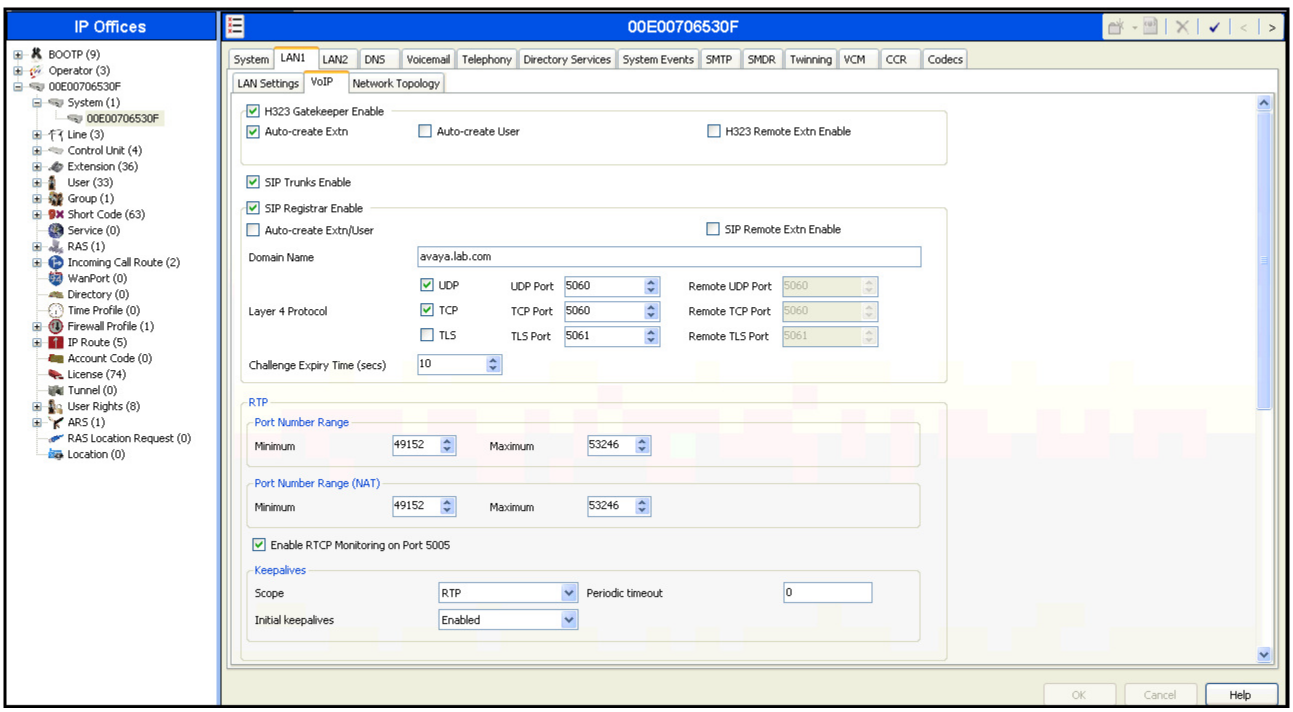
In the sample configuration, the MAC address **00E00706530F** was used as the system name and the **LAN** port connects to the inside interface of the Avaya SBCE across the enterprise LAN (private) network. The outside interface of the Avaya SBCE connects to TSTT networks via the public internet. The **LAN1** settings correspond to the **LAN** port in IP Office. To access the **LAN1** settings, navigate to **System (1) → 00E00706530F** in the Navigation Pane then in the Details Pane navigate to the **LAN1→ LAN Settings** tab. The **LAN1** settings for the compliance testing were configured with following parameters.

- Set the **IP Address** field to the LAN IP address, e.g. **172.16.5.60**.
- Set the **IP Mask** field to the subnet mask of the public network, e.g. **255.255.255.0**.
- All other parameters should be set according to customer requirements.
- Click OK to commit (not shown).



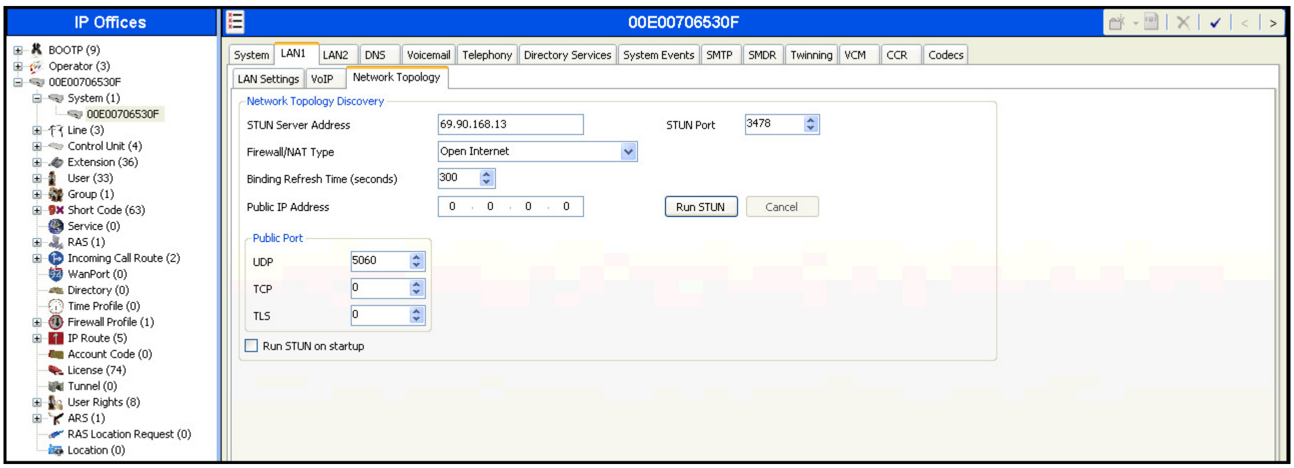
The **VoIP** tab as shown in the screenshot below was configured with following settings.

- Check the **H323 Gatekeeper Enable** to allow Avaya IP Telephones/Softphone using the H.323 protocol to register.
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to TSTT.
- Check the **SIP Registrar Enable** to allow Avaya IP Telephones/Softphone to register using the SIP protocol.
- Enter the Domain Name under **Domain Name**.
- Verify the **UDP Port** and **TCP Port** numbers under **Layer 4 Protocol** are set to **5060**.
- Verify the **RTP Port Number Range** settings for a specific range for the RTP traffic. The **Port Range (Minimum)** and **Port Range (Maximum)** values were kept as default.
- In the **Keepalives** section at the bottom of the page, set the **Scope** field to **RTP**, and **Initial keepalives** to **Enabled**. This will cause the IP Office to send RTP keepalive packets at the beginning of the calls, to avoid problems of media deadlock that can occur with certain types of forwarded calls that are routed from the IP Office back to the network, over the same SIP trunk.
- All other parameters should be set according to customer requirements.
- Click OK to commit (not shown).



In the **Network Topology** tab, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. In the compliance testing, it was set to **Open Internet**. With this configuration, even the default STUN settings are populated but they will not be used.
- Set the **Binding Refresh Time (seconds)** to a desired value, the value of **300 (or every 5 minutes)** was used during the compliance testing. This value is used to determine the **frequency** that IP Office will send OPTIONS heartbeat to the service provider.
- Leave the **Public IP Address** as **0.0.0.0**
- Set the **Public Port** to **5060** for **UDP**.
- All other parameters should be set according to customer requirements.
- Click OK to commit (not shown).

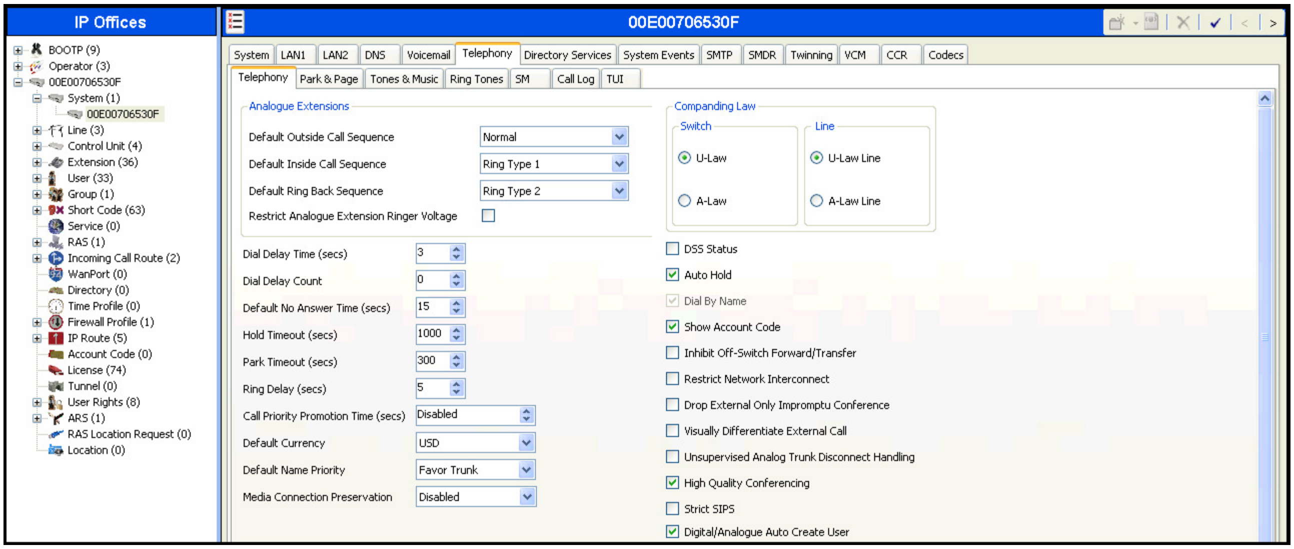


In the compliance test, the **LAN1** interface was used to connect Avaya IP Office to the enterprise private network (LAN), **LAN2** was not used.

5.3 System Telephony Settings

Navigate to the **Telephony → Telephony** Tab in the Details Pane, configure the following parameters:

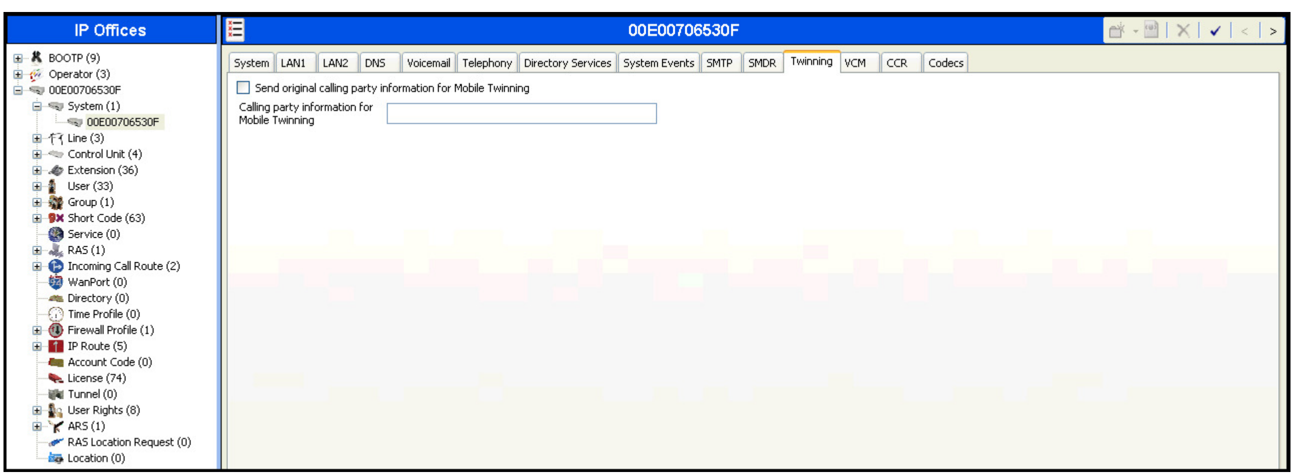
- Choose the **Companding Law** typical for the enterprise location, **U-Law** was used.
- Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the SIP trunk to the service provider.
- All other parameters should be set according to customer requirements.
- Click OK to commit (not shown).



5.4 Twinning Calling Party Settings

Navigate to the **Twinning** tab on the Details Pane, configure the following parameters:

- Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be controlled by the setting on the SIP Line (**Section 5.7**). This setting also impacts the Caller ID for call forwarding.
- Click OK to commit (not shown).

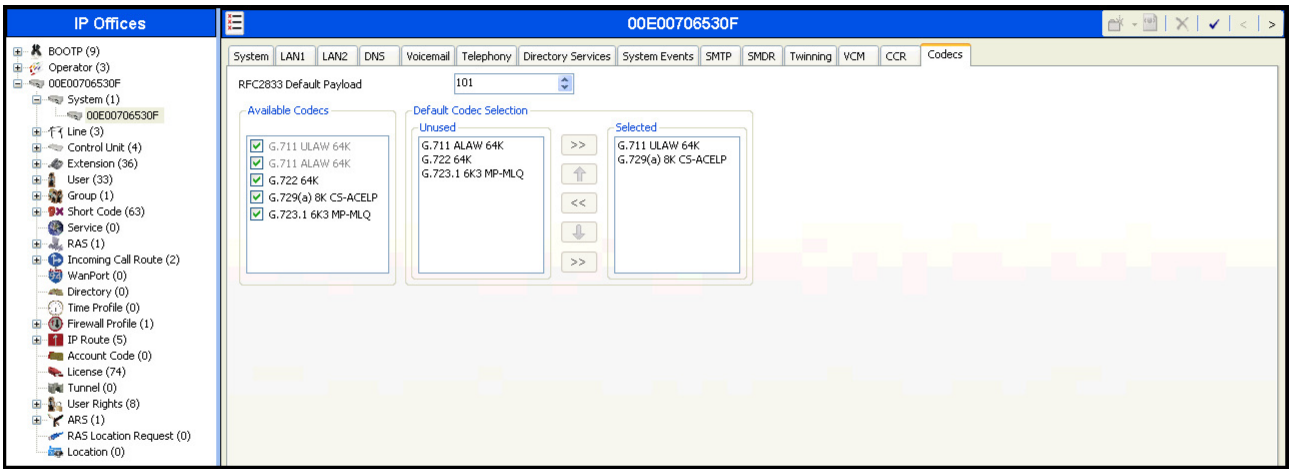


5.5 Codec's settings

For **Codec's** settings, navigate to the **System (1) → 00E00706530F** in the Navigation Pane, select the **Codecs** tab and configure the following parameters:

- Select the **Codecs**.
- Click OK to commit (not shown).

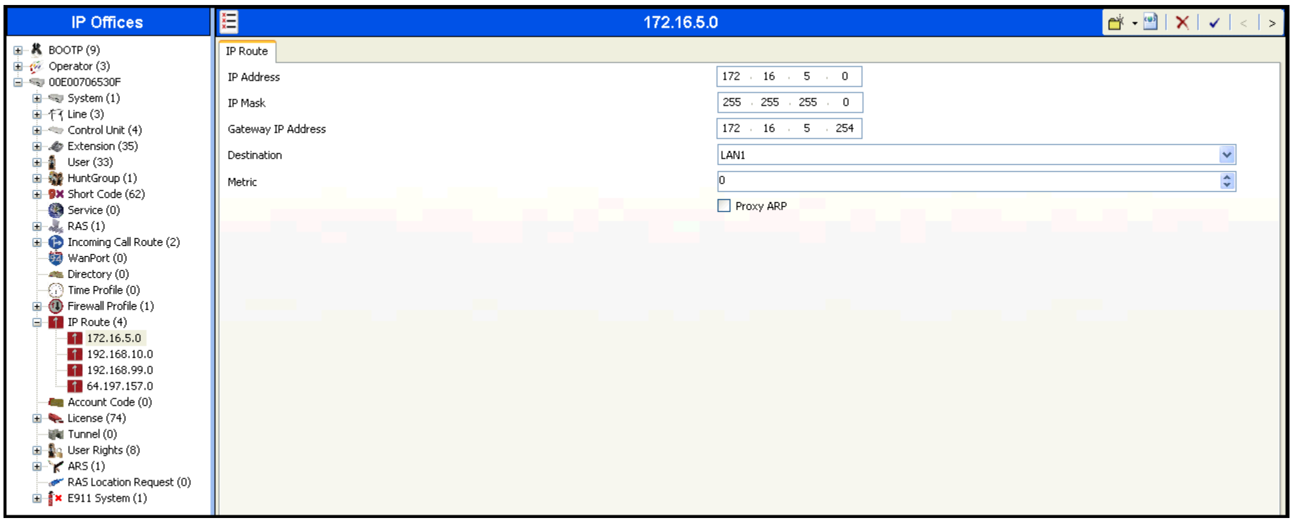
The **Codec's** settings are shown in the screenshot below with G.711ULAW and G.729(a) were selected in prioritized order. During the compliance testing, only codec G.711ULAW was tested (For Codec G.729A Test Results refer to **Section 2.2**).



5.6 IP Route

Create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets in order to reach the subnet where the SIP proxy is located on the TSTT network. On the left navigation pane, right-click on **IP Route** and select **New**.

- Set the **IP Address** and **IP Mask** of LAN1 connecting to the Avaya SBCE for SIP and RTP traffics to TSTT.
- Set **Gateway IP Address** to the IP Address of the router used to reach the external network.
- Set **Destination** to **LAN1** from the pull-down menu.
- Click OK to commit (not shown).



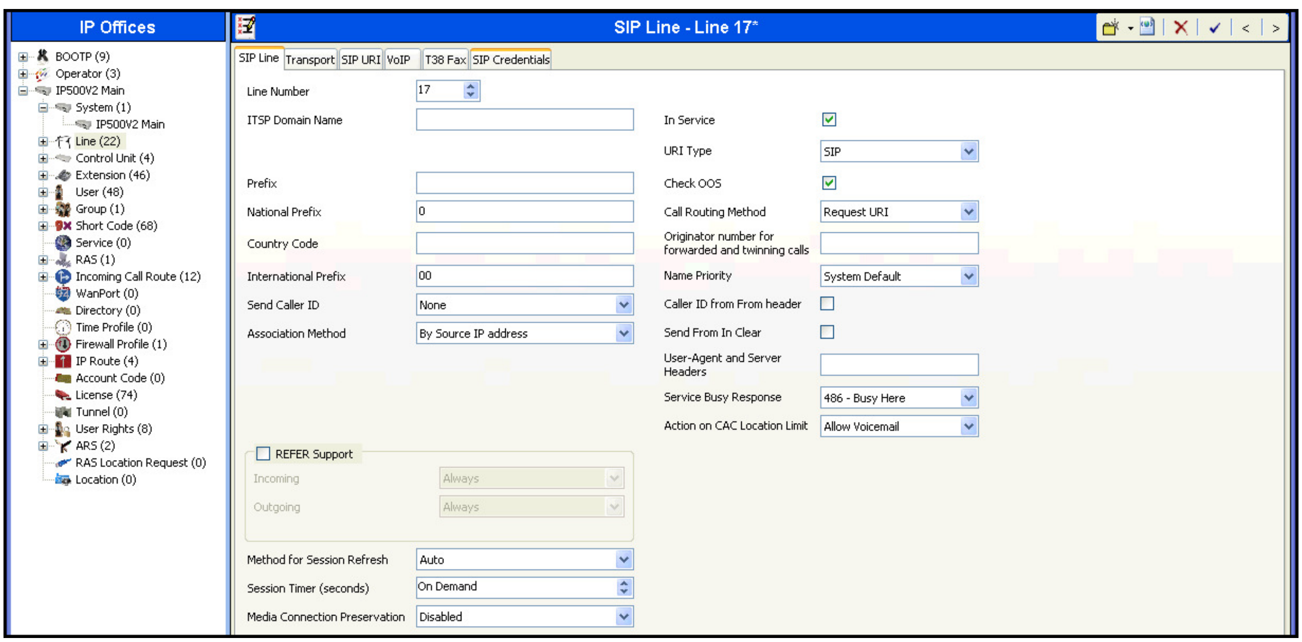
5.7 Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the TSTT SIP Trunk Service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New→ SIP Line**.

5.7.1 SIP Line Tab

On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

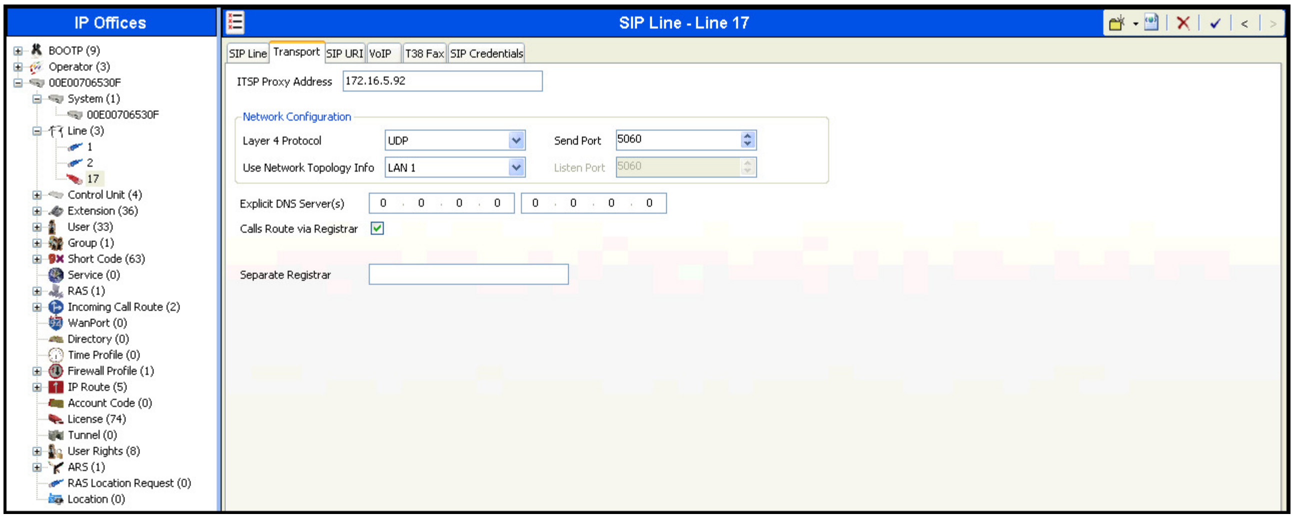
- Leave the **ITSP Domain Name** blank.
- Verify that **In Service** box is checked.
- Verify that **Check OOS** box is checked. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Verify that **Call Routing Method** is set to **Request URI**.
- Set **Send Caller ID** to **None.**
- Uncheck the **REFER support** box. IP Office will not send REFER messages for calls that are transferred back to the PSTN. See **Section 2.2** for more information.
- Set **Method for Session Refresh** to **Auto**.
- Set **Session Timer (Seconds**) to **On Demand.**
- Set **Media Connection Preservation** to **Disabled.**
- Default values may be used for all other parameters.
- Click OK to commit (not shown).



5.7.2 Transport Tab

Select the **Transport** tab; configure the parameters as shown below:

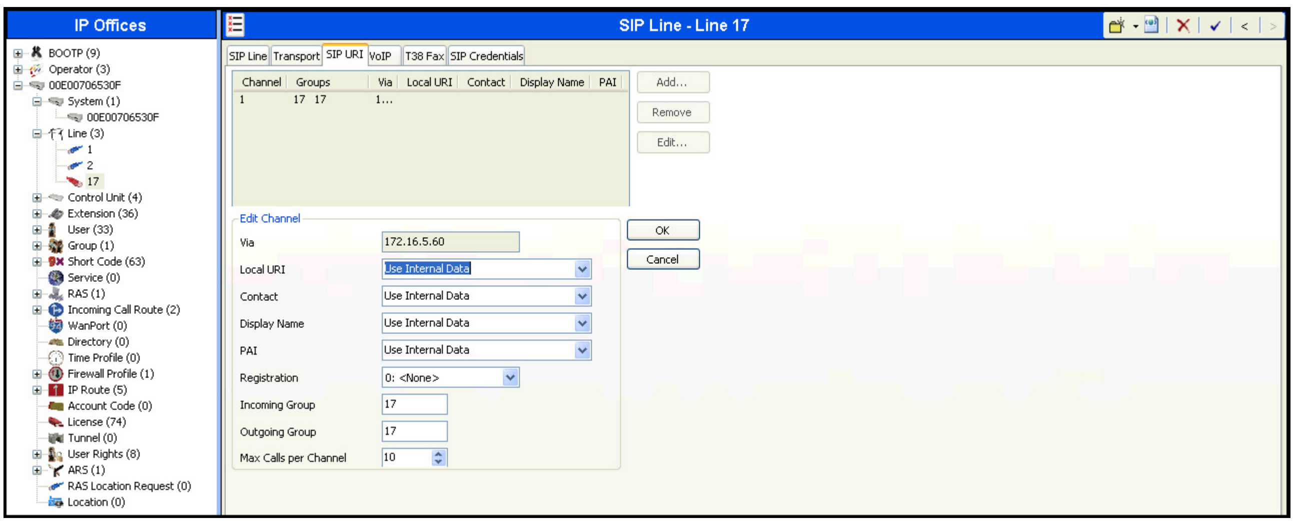
- Set the **ITSP Proxy Address** was set to the inside IP Address of the Avaya SBCE **172.16.5.92** as shown in **Figure 1**.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN1** as configured in **Section 5.2**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.
- Click OK to commit (not shown).



5.7.3 SIP URI Tab

A SIP URI entry needs to be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, and then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the example screen below, a previously configured entry was edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

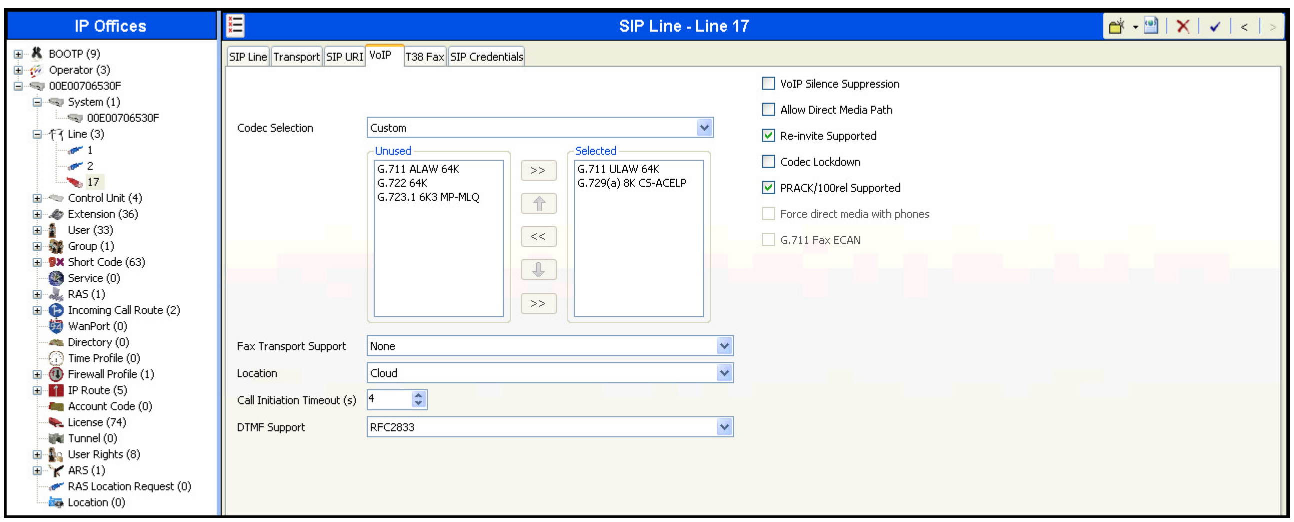
- Set **Local URI**, **Contact, Display Name and PAI** to **Use Internal Data**. This setting allows calls on this line whose SIP URI match the number set in the **SIP** tab of any User as shown in **Section 5.9**.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **17** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Click OK to commit (not shown).



5.7.4 VoIP Tab

Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- In the sample configuration, the **Codec Selection** was configured using the **Custom** option, allowing an explicit ordered list of codec's to be specified. The buttons allow setting the specific order of preference for the codec's to be used on the line, as shown. TSTT supports codec's G.711MU and G.729A, during the compliance testing, only codec G.711ULAW was tested (For Codec G.729A Test Results refer to **Section 2.2**).
- Set **Fax Transport Support** to **None**. **T.38 or G.711** fax transports **are not** recommended for this solution, as described in **Section 2.2**.
- Set the **DTMF Support** field to **RFC2833.** This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Verify that **Allow Direct Media Path** is unchecked. Testing was done with Direct Media disabled (Refer to **Section 2.2**).
- Check the **Re-invite Supported** box to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check the **PRACK/100rel Supported** box, to advertise the support for reliable provisional responses and Early Media to TSTT.
- Default values may be used for all other parameters.
- Click OK to commit (not shown).



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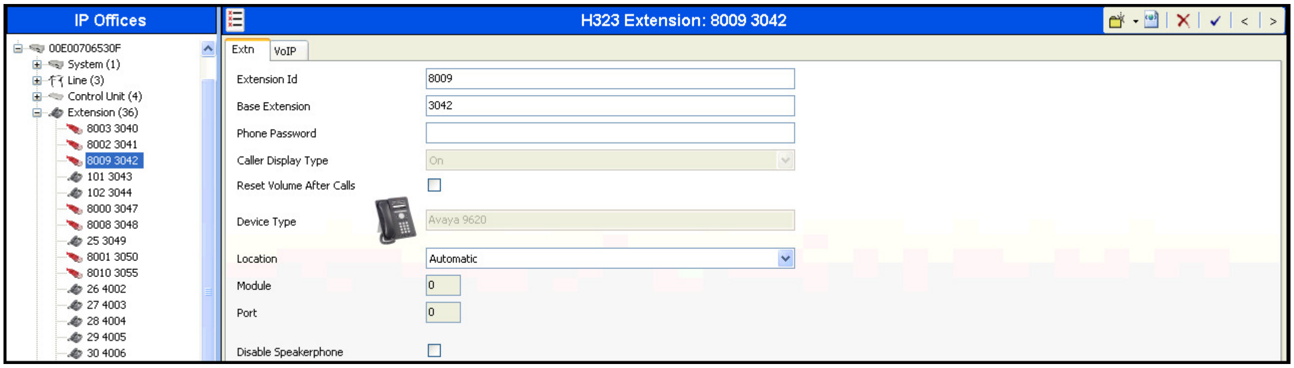
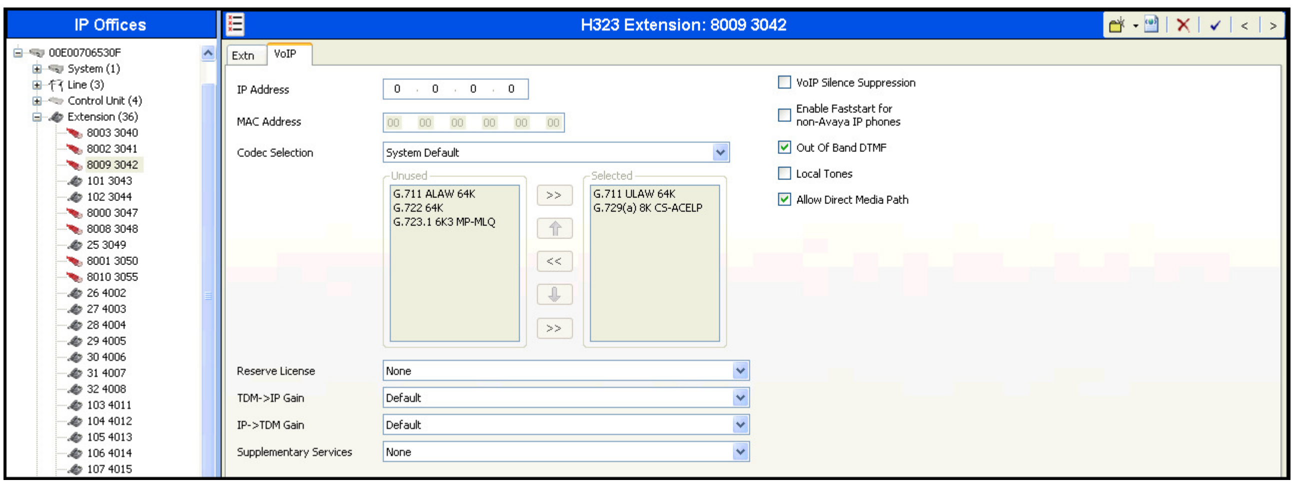
5.8 Extension

In this section, an example of an Avaya IP Office Extension will be illustrated. In the interests of brevity, not all users and extensions will be presented, since the configuration can be easily extrapolated to other users and extensions. To add an Extension, right click on **Extension** then select **New → Select H323 or SIP.**

Select the **Extn** tab. Following is an example of extension 3042; this extension corresponds to an H.323 extension.

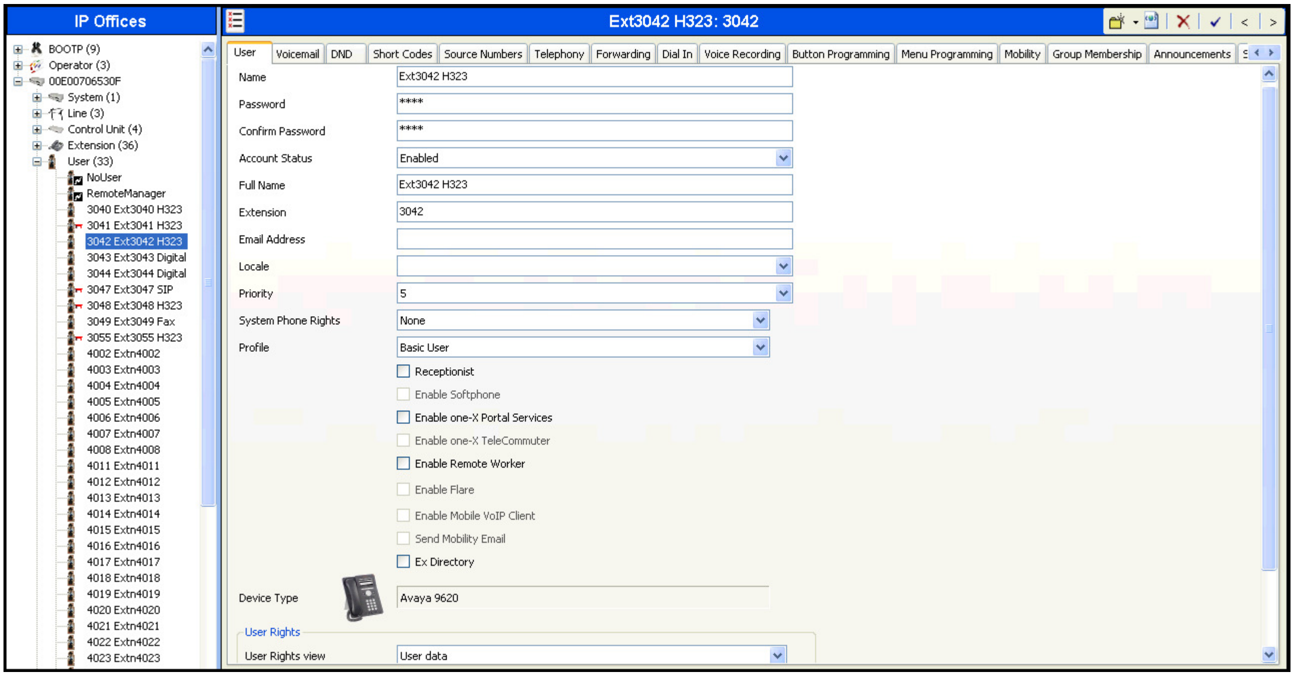


Select the **VOIP** tab. Use default values on VoIP tab. Following is an example for Extension 3042; this extension corresponds to an H.323 extension.

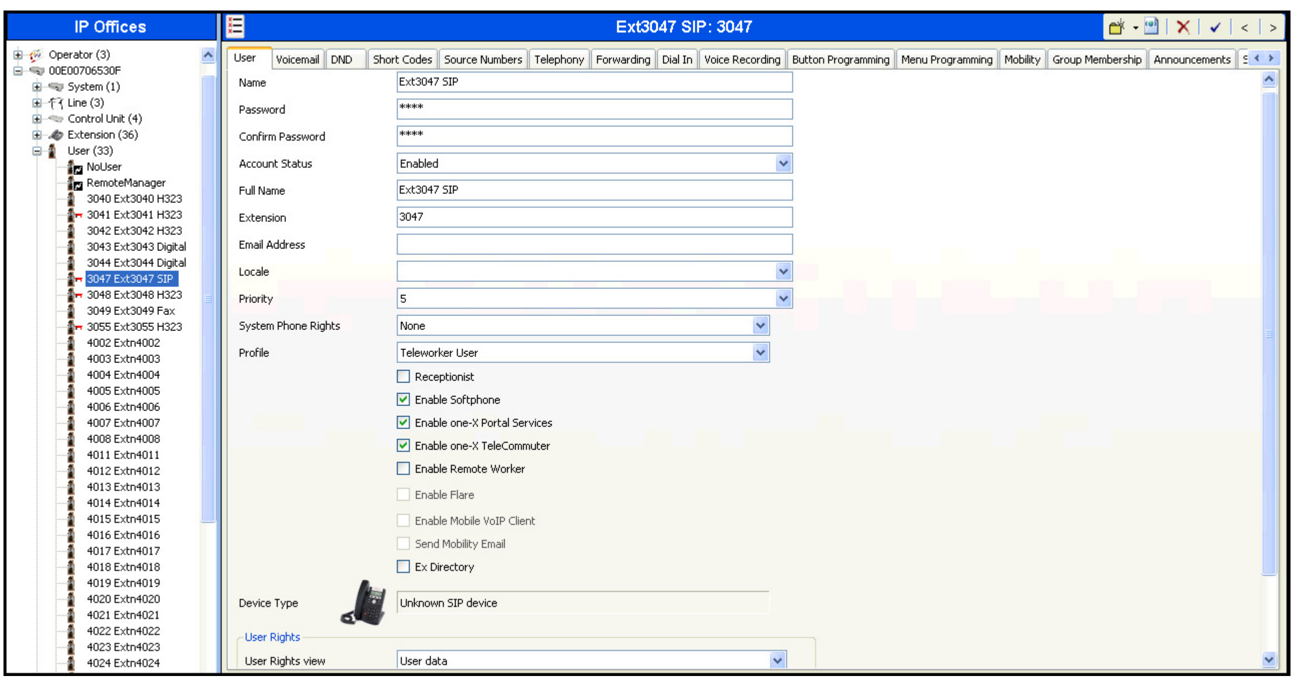


5.9 Users

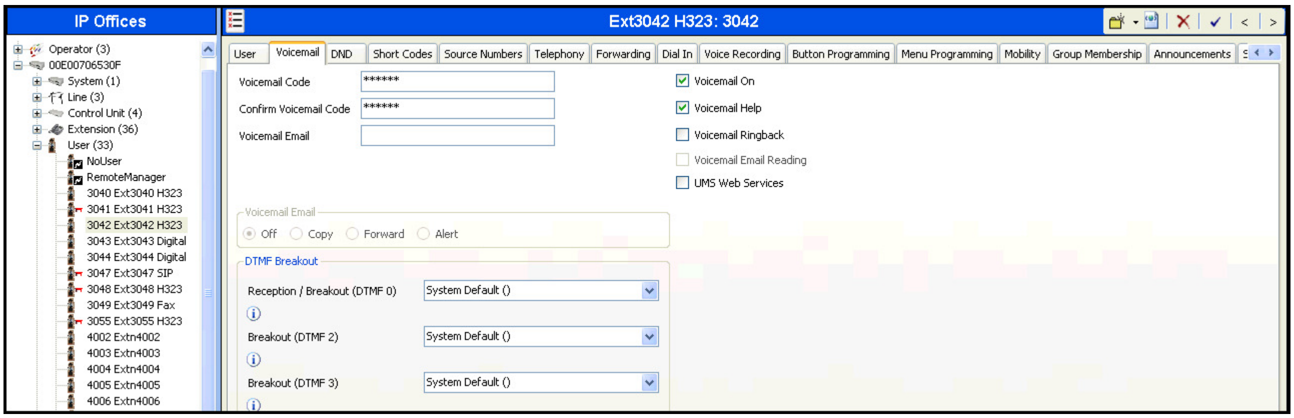
Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.7**. To configure these settings, first navigate to **User** in the left Navigation Pane, and then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **Ext3042 H323**.



In the example below, the name of the user is "Ext3047 SIP". This is an Avaya IP Office Softphone user, set the Profile to **Teleworker User** and check **Enable Softphone**.

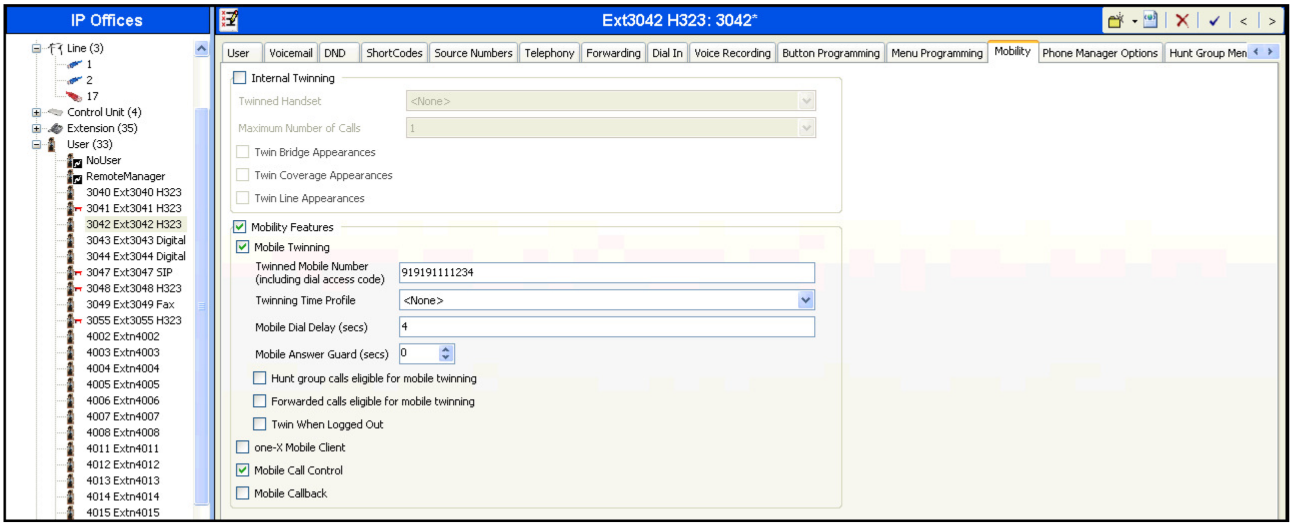


Select the **Voice Mail** tab. The following screen shows the **Voicemail** tab for the user with extension 3042. The **Voicemail On** box is checked. Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from TSTT SIP Trunk to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones to test DTMF using RFC 2833.

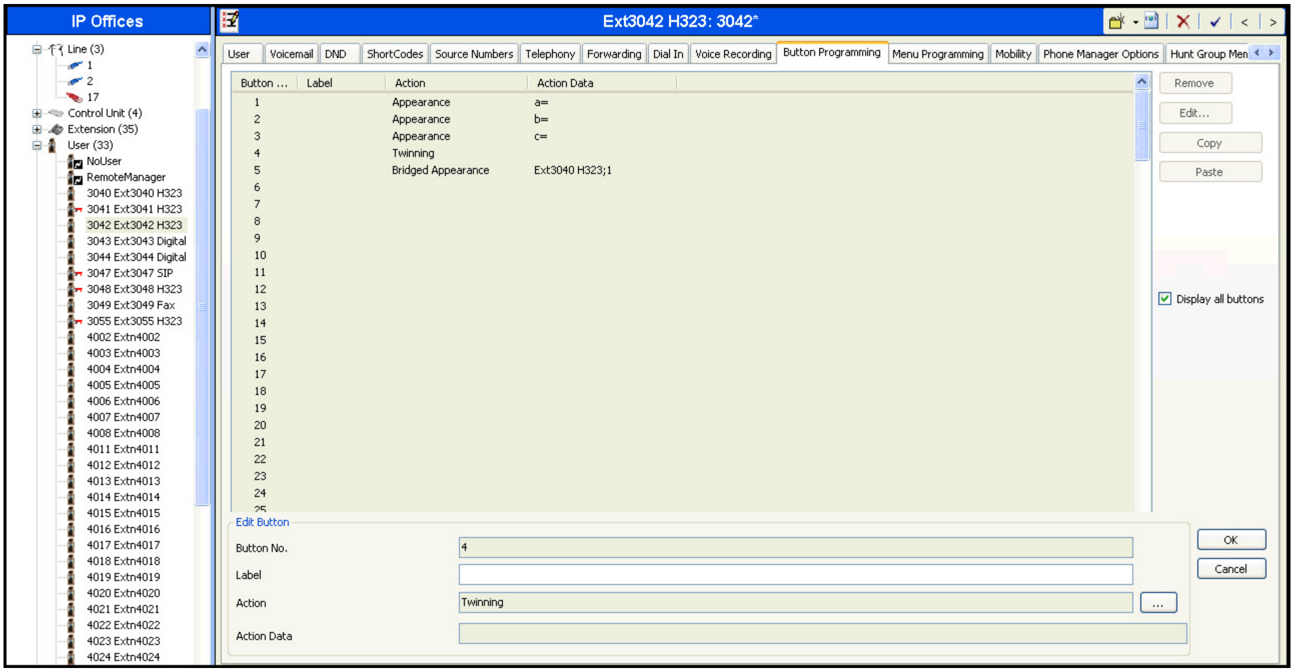


Select the **Telephony** tab, then **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an Avaya IP Office phone logged in as this extension to have multiple call appearances. Note: **Call Waiting On** is necessary for call transfer.

Select the **Mobility** tab. In the sample configuration user 3042 was one of the users configured to test the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 3042. The **Mobility Features** and **Mobile Twinning** boxes are checked**.** The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned telephone, in this case **919191111234**. Other options can be set according to customer requirements.

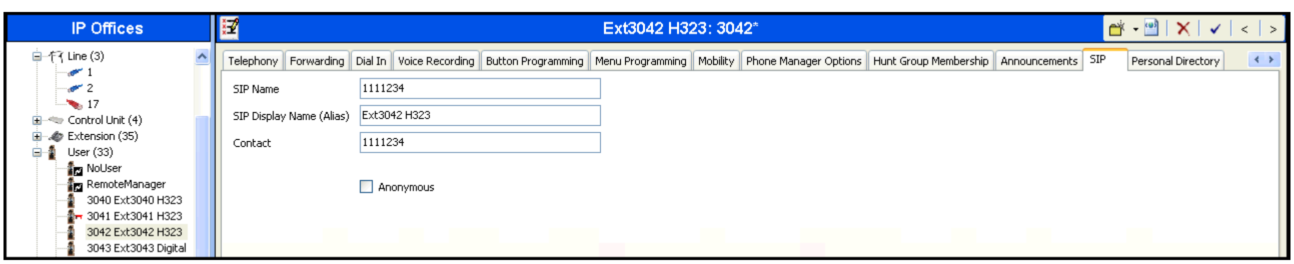


To program a key on the telephone to turn Mobil Twinning on and off, select the **Button Programming** tab on the user, then select the button to program to turn Mobil Twinning on and off, click on **Edit → Emulation → Twinning**. In the sample below, button **4** was programmed to turn Mobil Twinning on and off on user 3042.



Select the **SIP** tab, the values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.7**). The example below shows the settings for user "Ext3042 H323". The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by TSTT. In the example, DID number **1111234** was used. Only the last seven digits of the DID were assigned since TSTT only sends seven digits without the area code (868). The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.

If all calls involving this user should be considered private, then the **Anonymous** box may be checked to withhold the Caller ID information from the network.



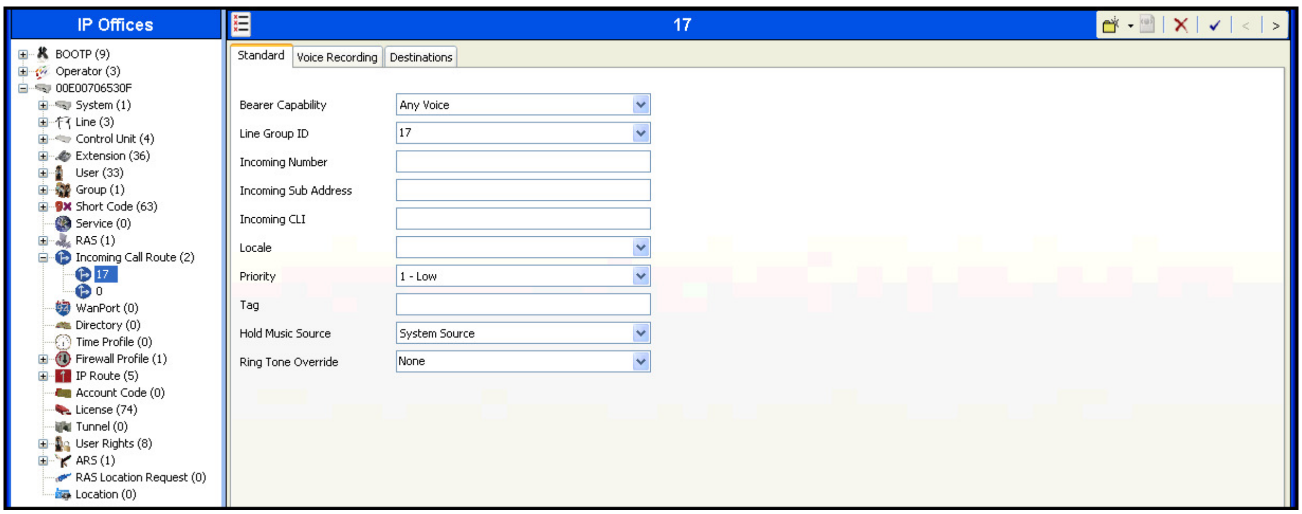
5.10 Incoming Call Route

An incoming call route maps inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc, within the IP Office system. Incoming call routes should be defined for each DID number assigned by the service provider.

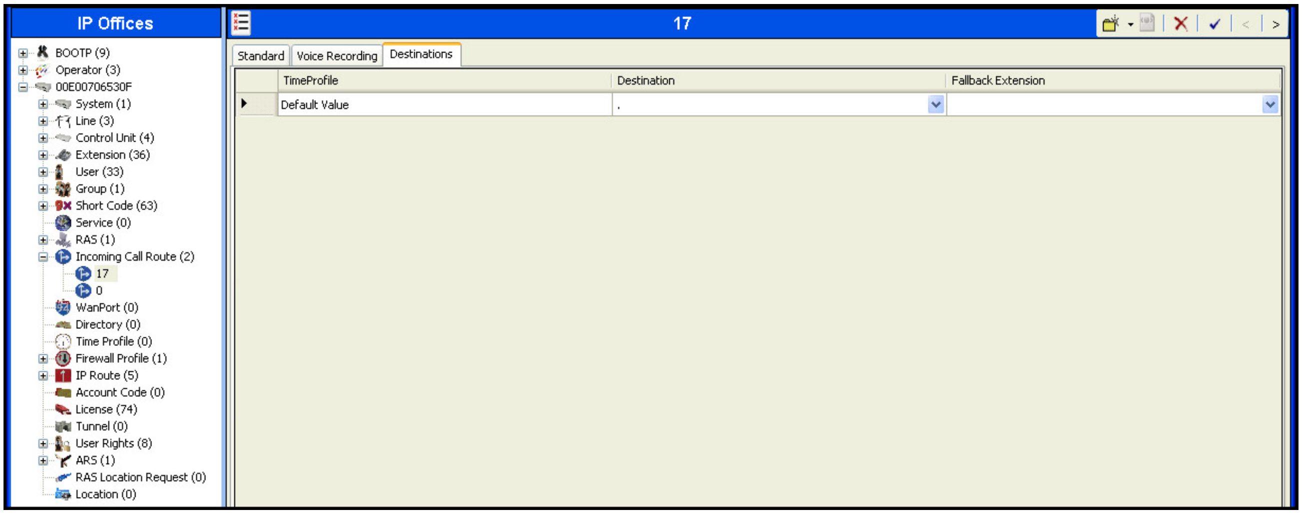
In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any predefined extension in IP Office. The routing decision for the call is based on the parameters previously configured for **Call Routing Method** and **SIP URI (Section 5.7)** and the users **SIP Name** and **Contact**, already populated with the assigned TSTT DID numbers **(Section 5.9)**

From the left Navigation Pane, right-click on **Incoming Call Route** and select **New.**
On the Details Pane, under the **Standard** tab, set the parameters as show bellow:

- Set **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.7**.
- Default values may be used for all other parameters.



- Under the **Destinations** tab, enter "**.**" for the **Default Value**. This setting will allow the call to be routed to any destination with a value on its **SIP Name** field, entered on the **SIP** tab of that **User,** which matches the number present on the user part of the incoming Request URI.

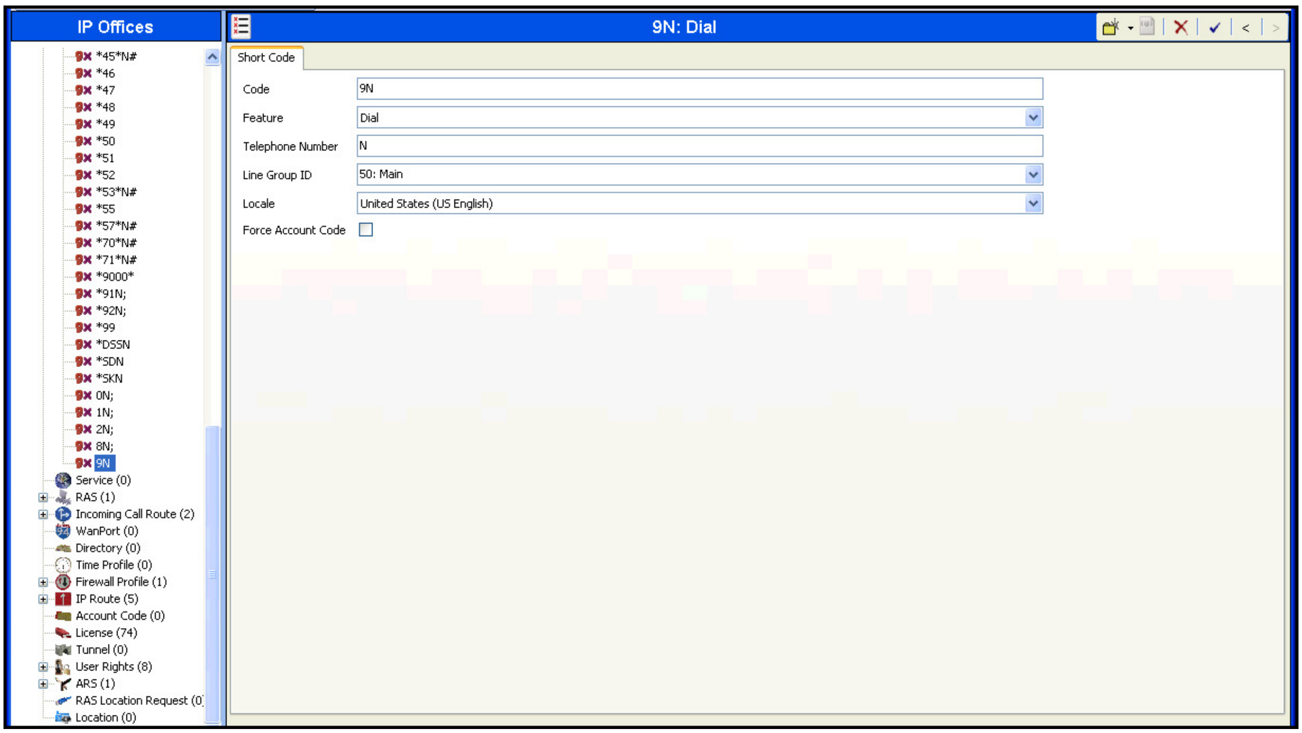


5.11 Outbound Call Routing

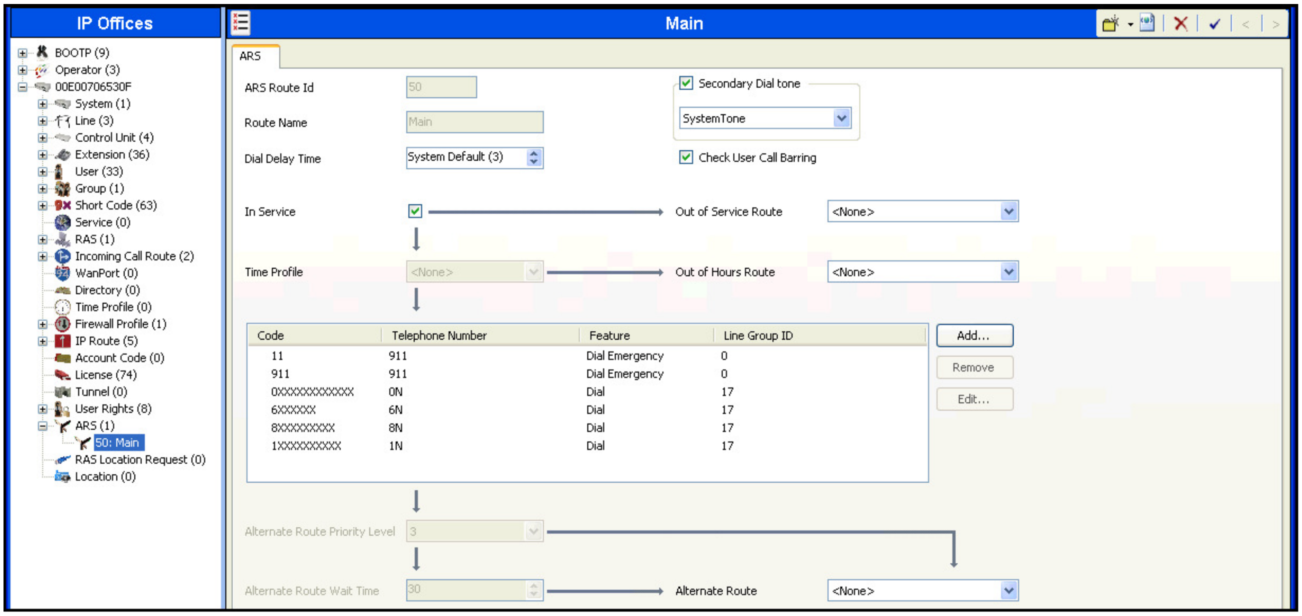
For outbound call routing, a combination of system short codes and Automatic Route Selection (ARS) entries are used. With ARS, features like time-based routing criteria and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. While detailed coverage of ARS is beyond the scope of these Application Notes and alternate routing was not used in the reference configuration, this section includes some basic screen illustrations of the ARS settings used during the compliance test

5.11.1 Short Codes and Automatic Route Selection

To create a short code to be used for ARS, right-click on **Short Code** in the Navigation Pane and select **New**. The screen below shows the short code **9N** created. Note that the semi-colon is not used here. In this case, when the Avaya IP Office user dials 9 plus any number **N**, instead of being directed to a specific Line Group ID, the call is directed to **Line Group 50: Main**, which is configurable via ARS.



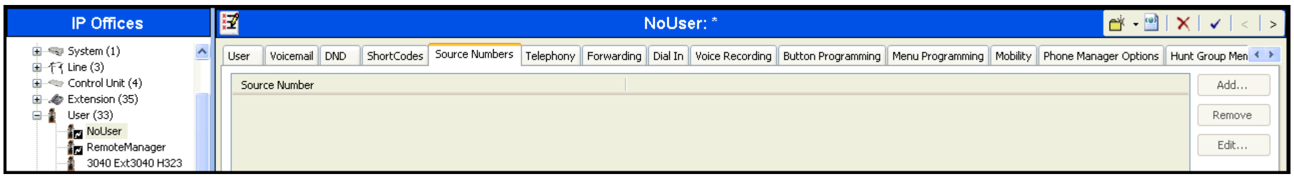
The following screen shows the example ARS configuration for the route **Main**. Note the sequence of **X**s used in the **Code** column of the entries to specify the exact number of digits to be expected, following the access code and the first digit on the string. This type of setting results in a much quicker response in the delivery of the call by IP Office. The example below shows that for local calls, the user dialed 9, then 10 digit numbers starting with an 8. For calls to other area codes in the North American Numbering Plan, the user dialed 9, followed by 11 digits, starting with a 1.



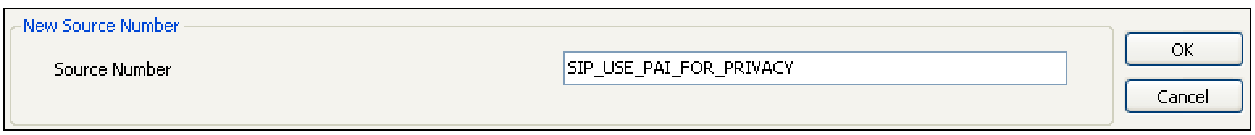
5.12 Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "restricted" and "anonymous" respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. By default, Avaya IP Office will use PPI for privacy. For the compliance test, PAI was used for the purposes of privacy.

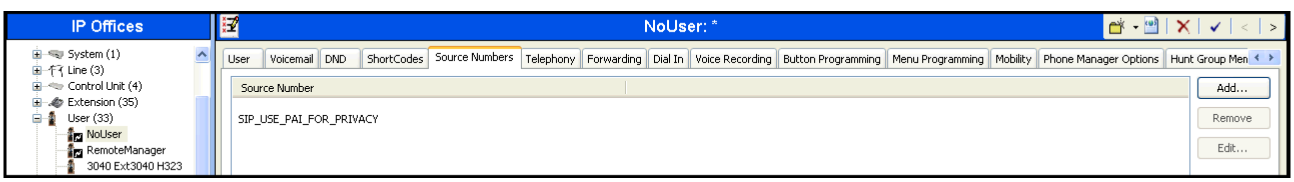
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_USE_PAI_FOR_PRIVACY**. Click **OK**.



The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.

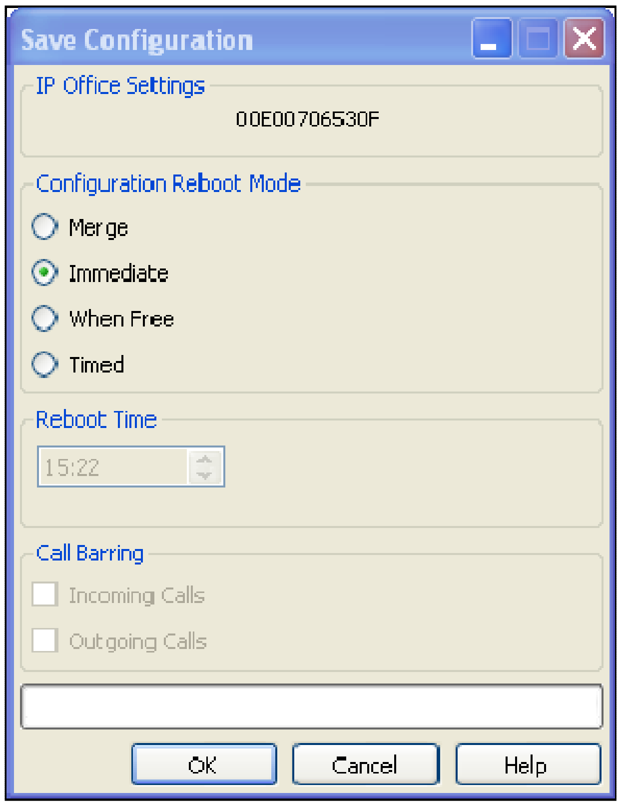


5.13 Save Configuration

When desired, send the configuration changes made in Avaya IP Office Manager to the Avaya IP Office server in order for the changes to take effect.

Navigate to **File→Save Configuration** in the menu bar at the top left of the screen to save the configuration performed in the preceding sections.

Once the configuration is validated, a screen similar to the following will appear, with either the **Merge** or the **Immediate** radio button chosen based on the nature of the configuration changes made since the last save. Note that clicking OK may cause a service disruption due to system reboot. Click OK if desired.



6. Configure the Avaya Session Border Controller for Enterprise

This section covers the configuration of the Avaya SBCE. It is assumed that the software has already been installed. For additional information on these configuration tasks, see **References** [**6**], [**7**] and [**8**] in **Section 10**.

The configuration of the Avaya SBCE covers two major components, the Trunk Server for the service provider and Call Server for the enterprise. Each component consists of a set of Global Profiles, Domain Policies and Device Specific Settings. The configuration was defined using the Avaya SBCE web user interface as described in the following sections.

Trunk Server configuration elements for the service provider - TSTT:

- Global Profiles:
    - URI Groups
    - Routing
    - Topology Hiding
    - Server Interworking
    - Signaling Manipulation
    - Server Configuration
- Domain Policies:
    - Application Rules
    - Media Rules
    - Signaling Rules
    - Endpoint Policy Group
    - Session Policy
- Device Specific Settings:
    - Network Management
    - Media Interface
    - Signaling Interface
    - End Point Flows → Server Flows
    - Session Flows

Call Server configuration elements for the enterprise - IP Office:

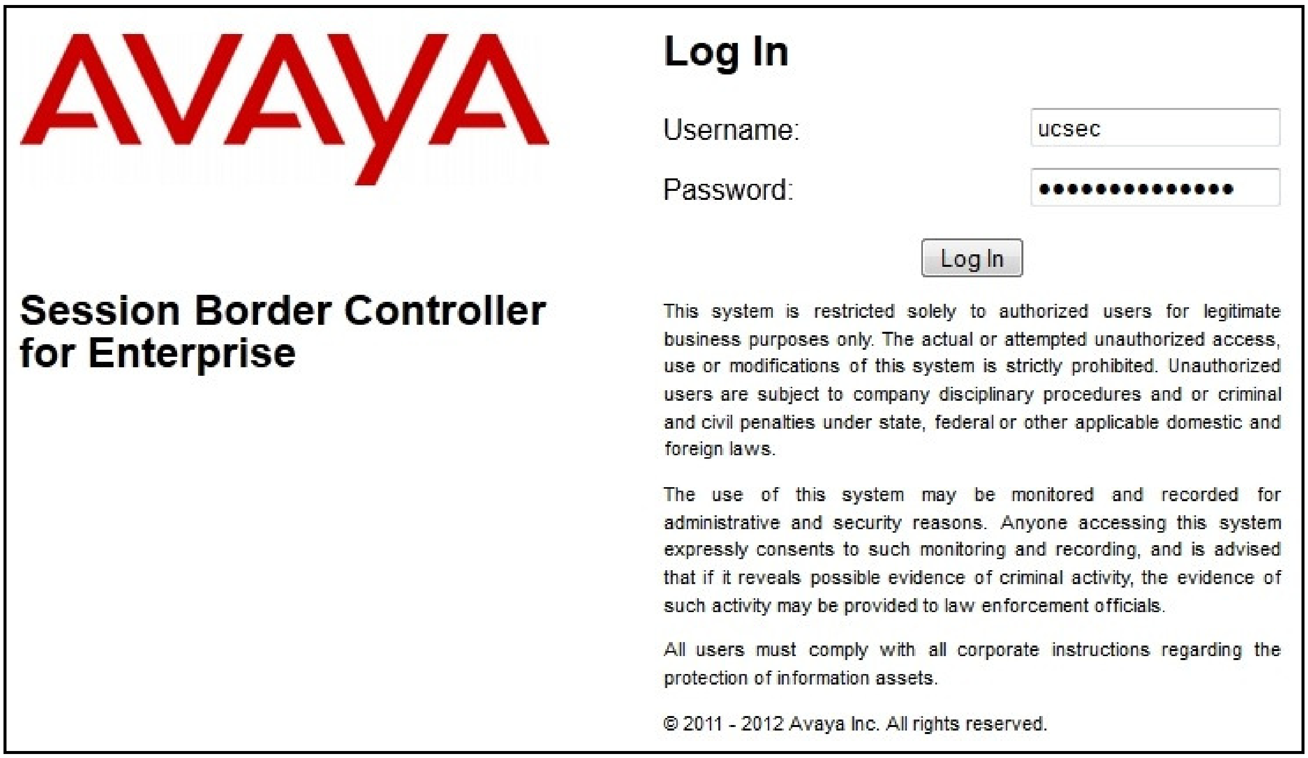
- Global Profiles:
    - URI Groups
    - Routing
    - Topology Hiding
    - Server Interworking
    - Server Configuration
- Domain Policies:
    - Application Rules
    - Media Rules
    - Signaling Rules
    - Endpoint Policy Group
    - Session Policy
- Device Specific Settings:

- Network Management
- Media Interface
- Signaling Interface
- End Point Flows → Server Flows
- Session Flows

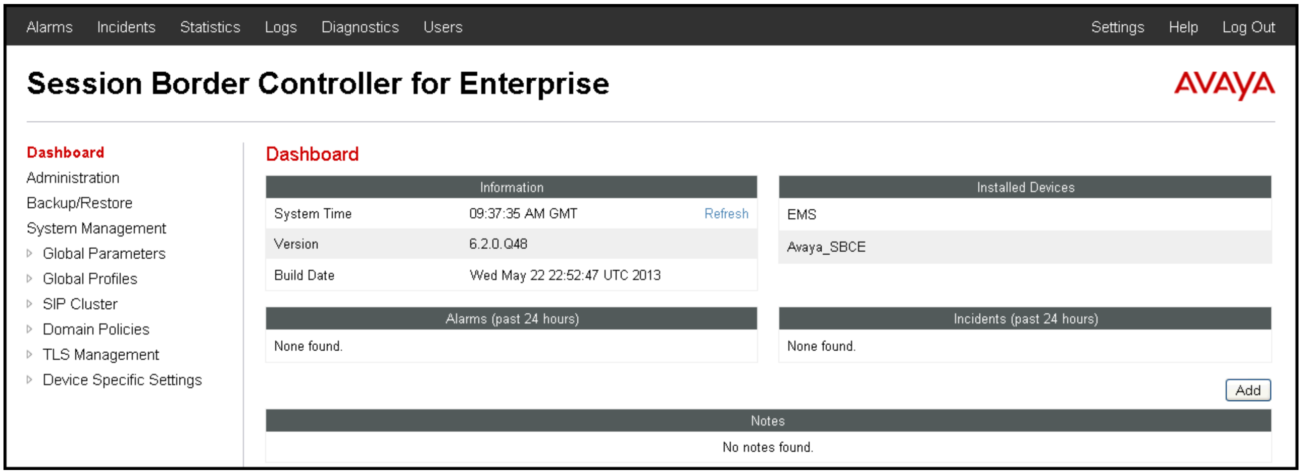
6.1 Log into the Avaya Session Border Controller for Enterprise

Use a Web browser to access the Avaya SBCE Web interface, enter https://<ip-addr>/sbc in the address field of the web browser, where <ip-addr> is the management IP address.

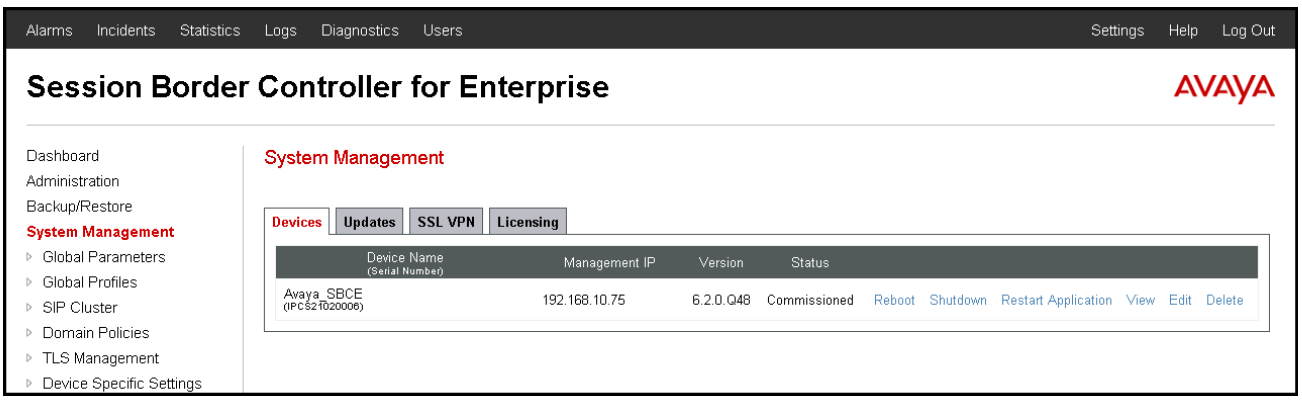
Enter the appropriate credentials then click **Log In**.



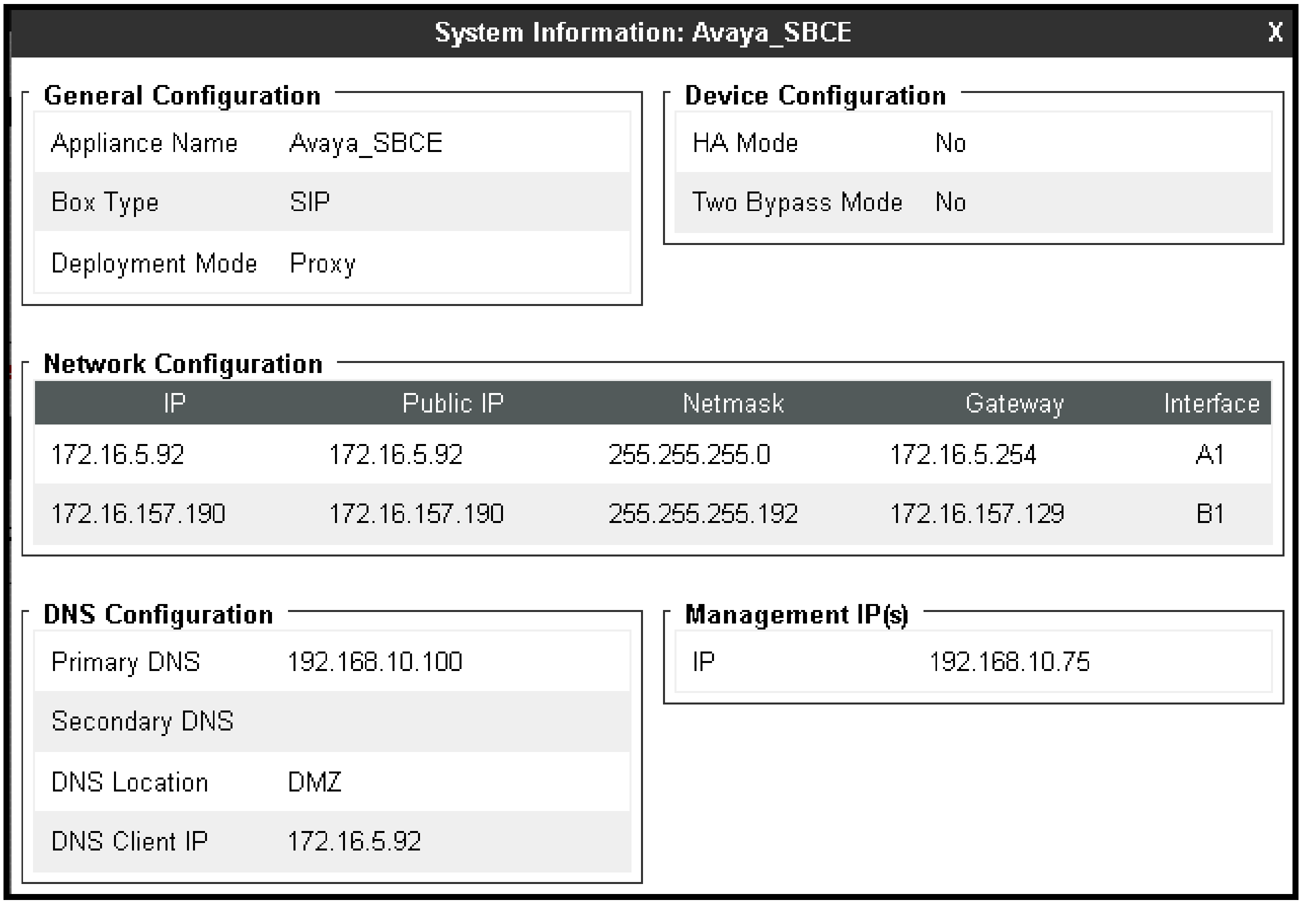
The **Dashboard** main page will appear as shown below.



To view the system information that has been configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the compliance testing, a single Device Name **Avaya SBCE** was already added. To view the configuration of this device, click the **View** as shown in the screenshot below.



The **System Information** screen shows **Network Settings**, **DNS Configuration** and **Management IP** information provided during installation and corresponded 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.



6.2 Global Profiles

The Global Profiles Menu, on the left navigation pane, allows the configuration of parameters that affect all the devices under the UC-Sec control Center.

6.2.1 Server Interworking Avaya

Interworking Profile features are configured to facilitate interoperability of implementations between enterprise SIP-enabled solutions and different SIP trunk service providers.

Several profiles have been already pre-defined and they populate the list under **Interworking Profiles** on the screen below. If a different profile is needed, a new Interworking Profile can be created, or an existing default profile can be modified or "cloned". Since modifying a default profile is generally not recommended, for the test configuration the default **avaya-ru** profile was duplicated, or "cloned", and then modified to meet specific requirements for the enterprise SIP-enabled solution.

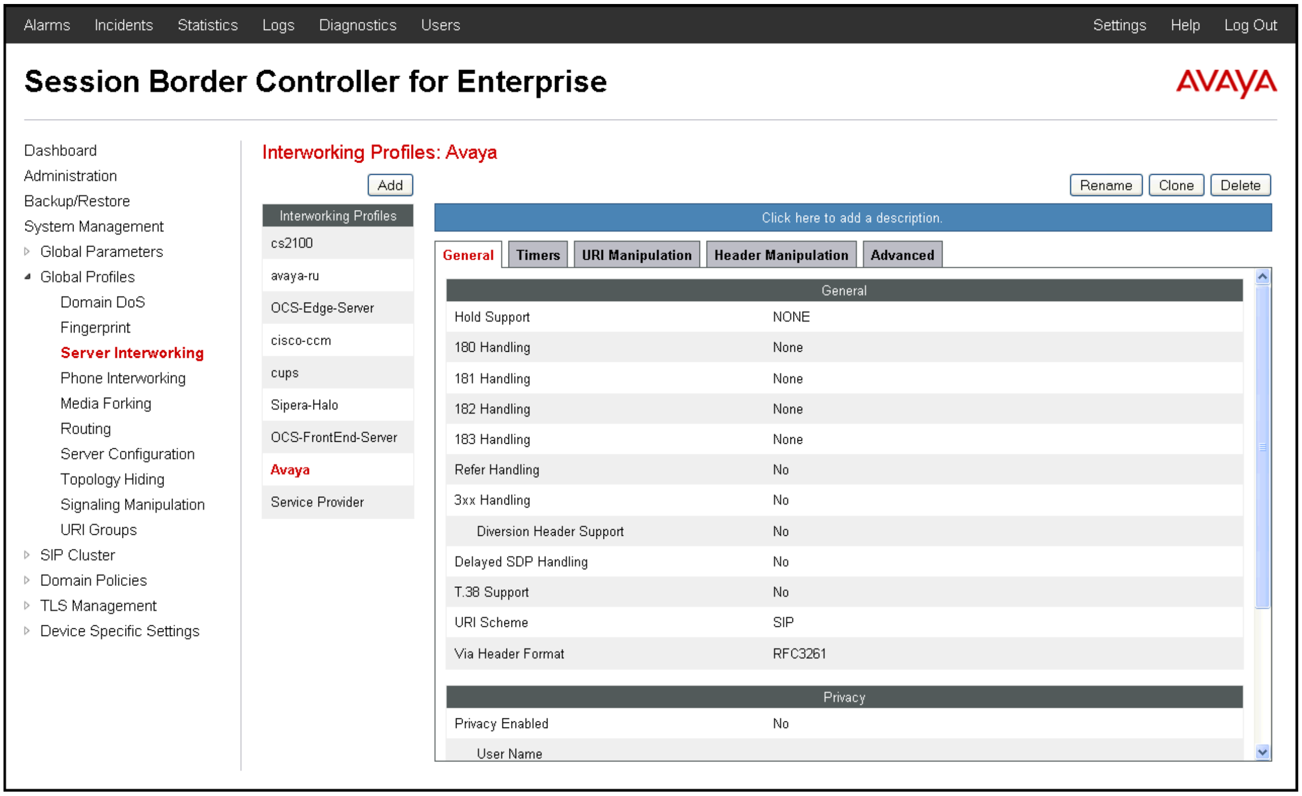
On the left navigation pane, select **Global Profiles → Server Interworking**. From the **Interworking Profiles** list, select **avaya-ru.** Click **Clone Profile.**

Enter the new profile name in the **Clone Name** field, the name of **Avaya** was chosen in this example. Click **Finish**.

For the newly created **Avaya** profile, click **Edit** (not shown) at the bottom of the General tab

- Click **Next**.
- Click **Finish** on the **Privacy and DTMF** tab.
- Leave other fields with their default values.

The following screen capture shows the newly added **Avaya** Profile.



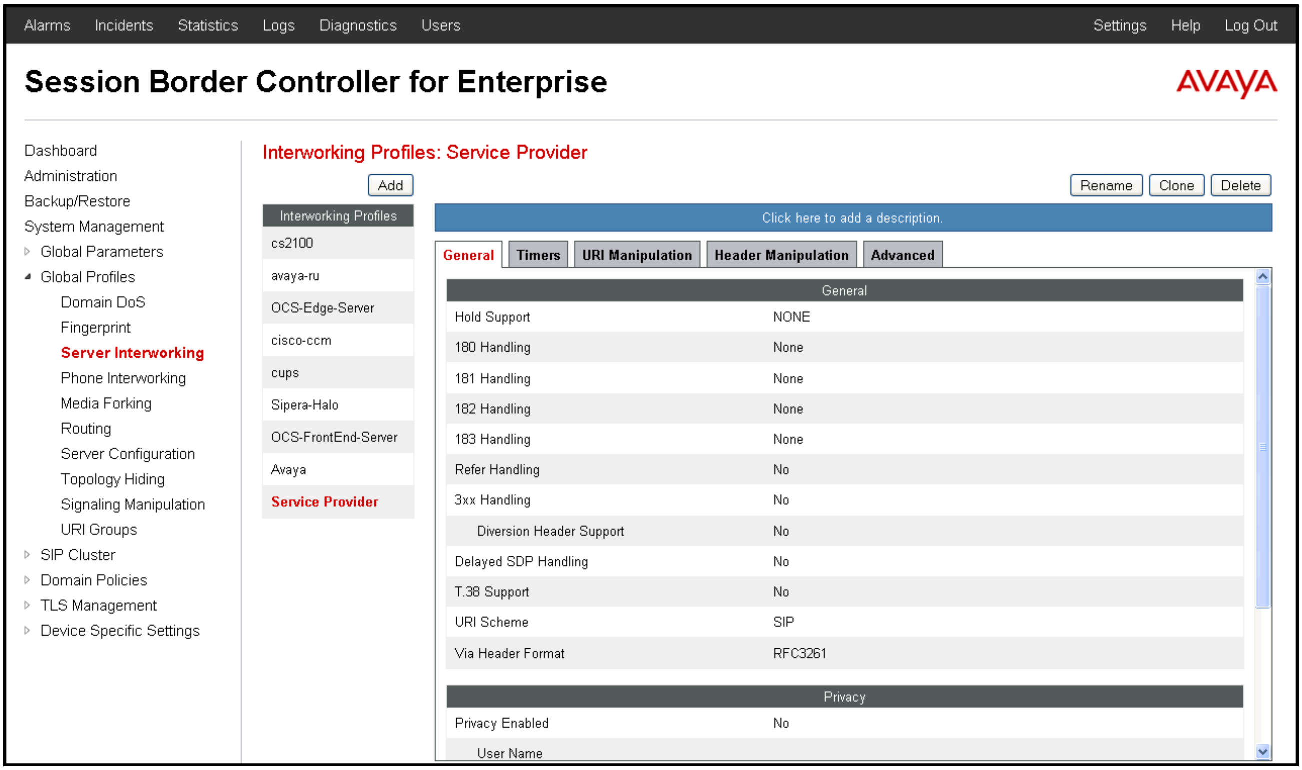
6.2.2 Server Interworking Service Provider

A second Server Interworking profile named **Service Provider** was created for the Service Provider.

On the left navigation pane, select **Global Profiles → Server Interworking**. From the **Interworking Profiles** list, select **Add.**

Enter the new profile name (not shown), the name of **Service Provider** was chosen in this example. Accept the default values for all fields by clicking **Next** and then Click **Finish.**

The following screen capture shows the newly added **Service Provider** Profile.



6.2.3 Routing Profiles

Routing profiles define a specific set of routing criteria that are used, in conjunction with other types of domain policies, to determine the route that SIP packets should follow to arrive at their intended destination.

Two Routing Profiles were created in the test configuration, one for inbound calls, with IP Office as the destination, and the second one for outbound calls, which are sent to the Service Provider SIP trunk.

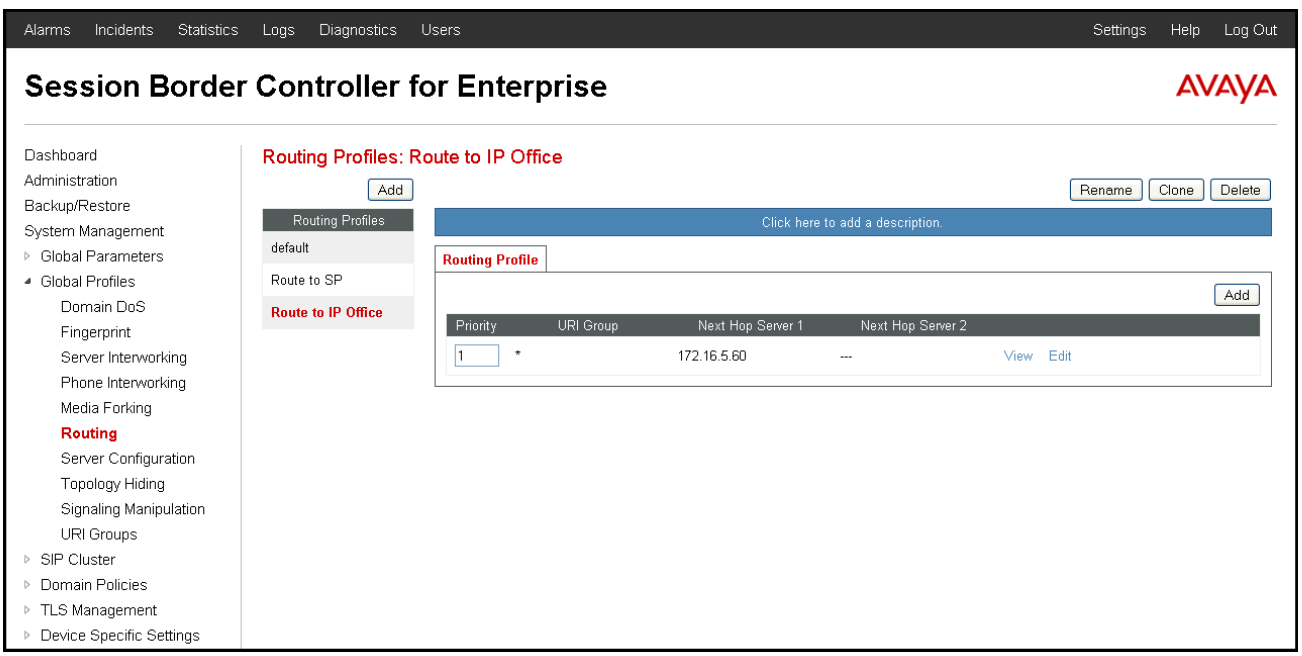
To create the inbound route, from the **Global Profiles** menu on the left-hand side:

- Select the **Routing** tab (not shown).
- Select **Add.**
- Enter Profile Name: **Route to IP Office.**
- Click **Next** (not shown)**.**

On the next screen, complete the following:

- **Next Hop Server 1: 172.16.5.60** (IP Office IP address).
- Check **Routing Priority Based on Next Hop Server** (not shown).
- Check **Outgoing Transport: UDP** (not shown).
- Click **Finish.**

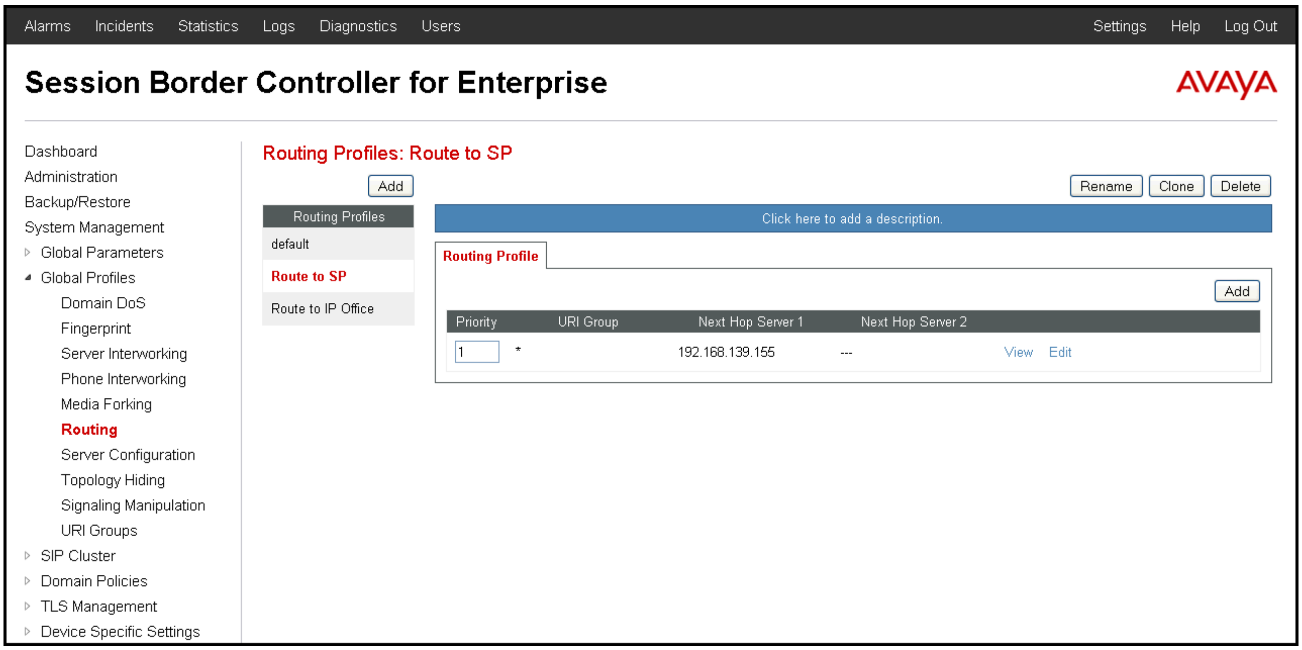
The following screen shows the newly added **Route to IP Office** Profile.



Similarly, for the outbound route:

- Select **Add.**
- Enter Profile Name: **Route to SP**
- Click **Next.**
- **Next Hop Server 1: 192.168.139.155** (IP address for Service Provider's proxy server)
- Check **Routing Priority Based on Next Hop Server** (not shown).
- Check **Outgoing Transport: UDP** (not shown).
- Click **Finish.**

The following screen capture shows the newly added **Route_to_SP** Profile.



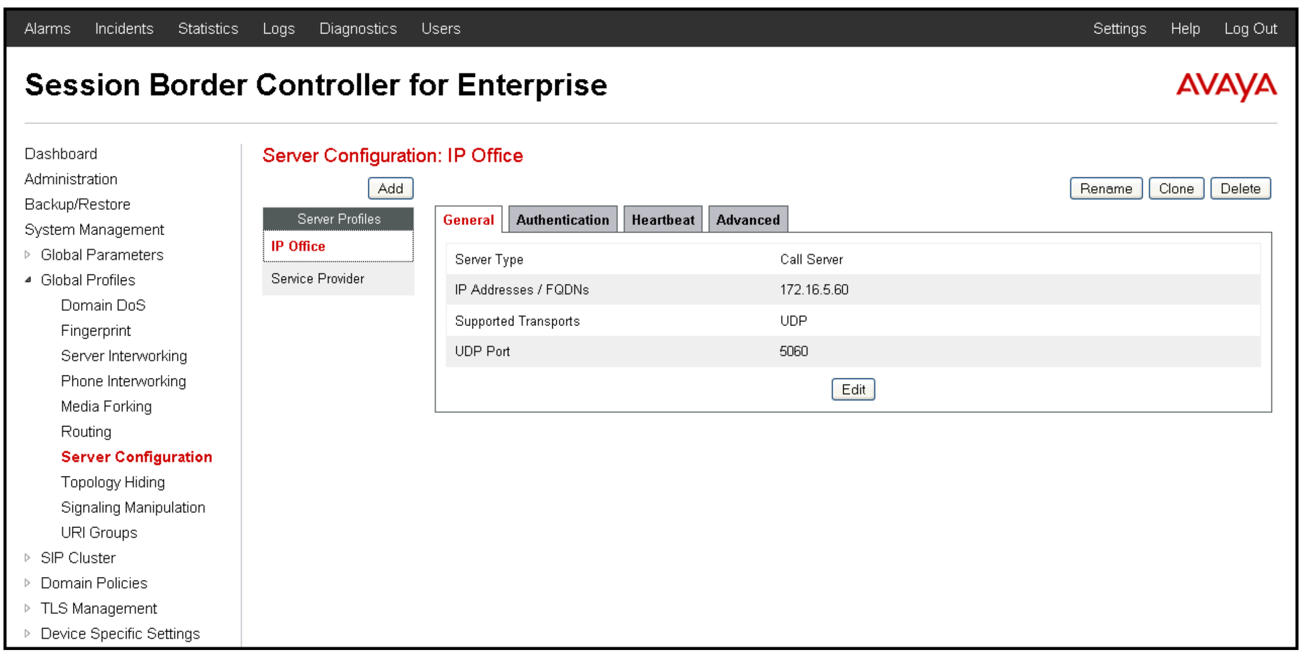
6.2.4 Server Configuration

Server Profiles should be created for the Avaya SBCE's two peers, the Call Server (IP Office) and the Trunk Server or SIP Proxy at the service provider's network.

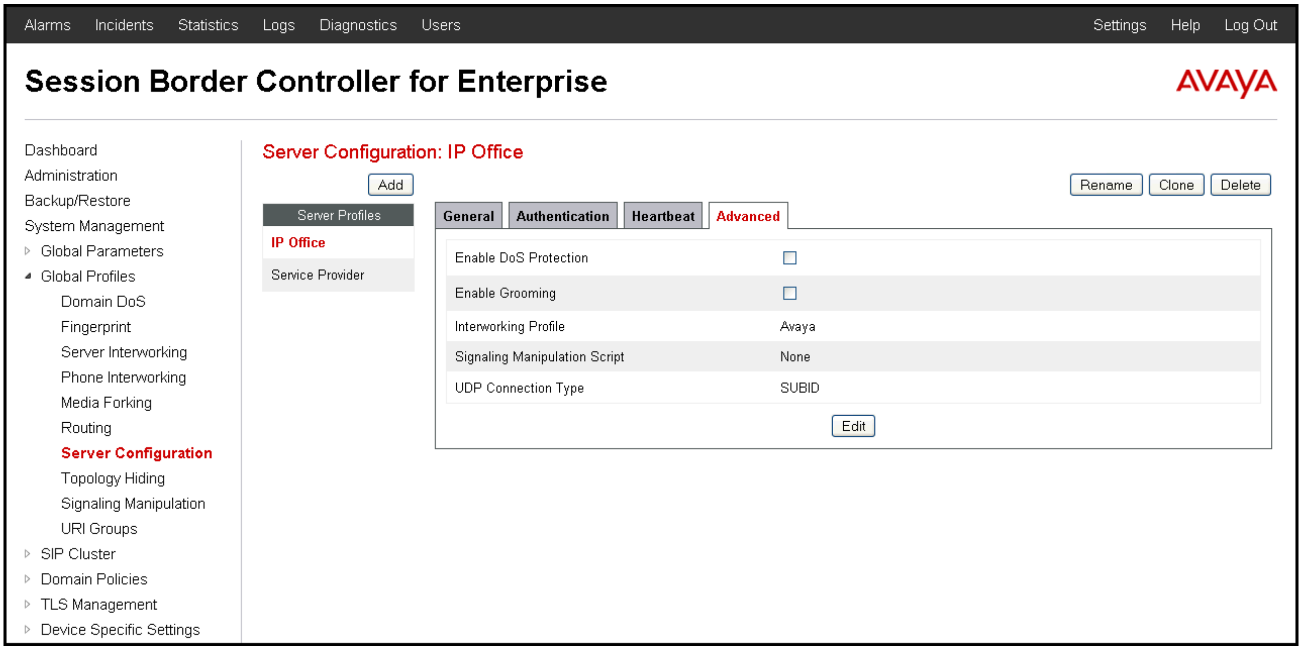
To add the profile for the Call Server, from the **Global Profiles** menu on the left-hand navigation pane, select **Server Configuration**. Click **Add** and enter the profile name: **Session Manager**.
On the **Add Server Configuration Profile** Tab (not shown):

- Select Server Type**: Call Server.**
- **IP Address: 172.16.5.60** (IP Address of IP Office).
- **Supported Transports**: Check **UDP**.
- **TCP Port: 5060.**
- Click **Next.**
- Click **Next** on the **Authentication** tab.
- Click **Next** on the **Heartbeat** tab.
- On the **Advanced** tab, select **Avaya** from the **Interworking Profile** drop down menu. Leave the **Signaling Manipulation Script** at the default **None**.
- Click **Finish.**

The following screen capture shows the **General** tab of the newly added **IP Office** Profile.



The following screen capture shows the **Advanced** tab of the added **IP Office** Profile.

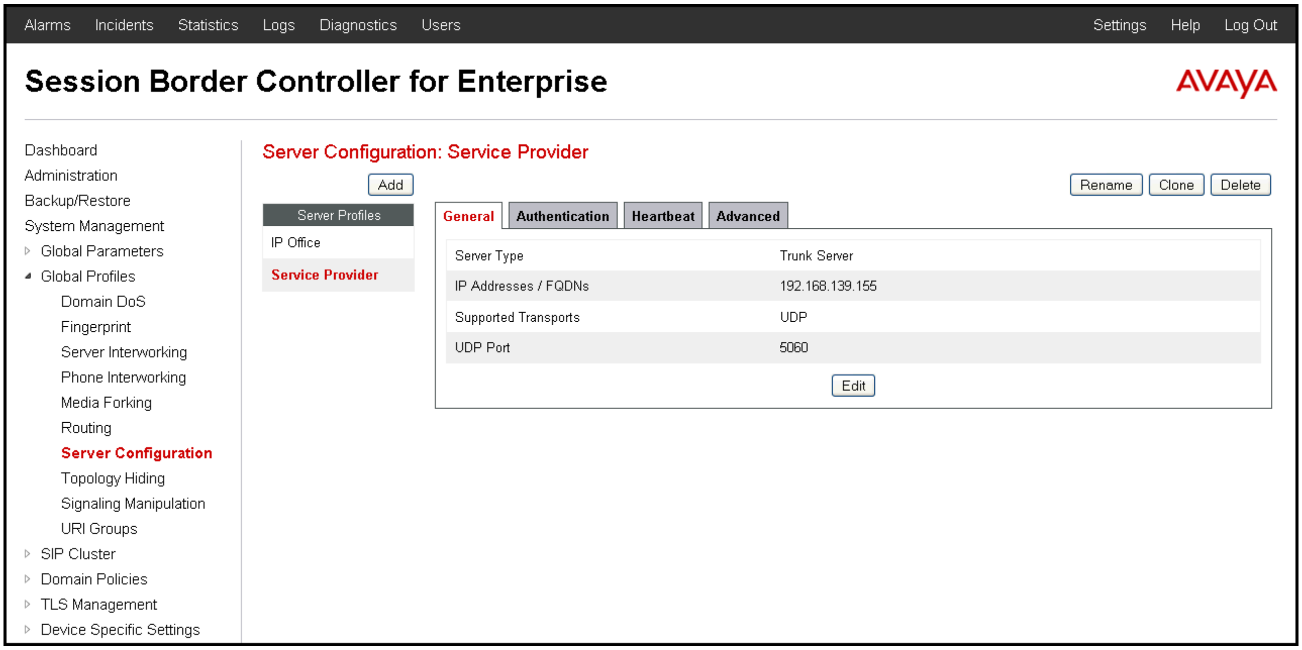


To add the profile for the Trunk Server, from the **Server Configuration** screen, click **Add** and enter the profile name: **Service Provider.**

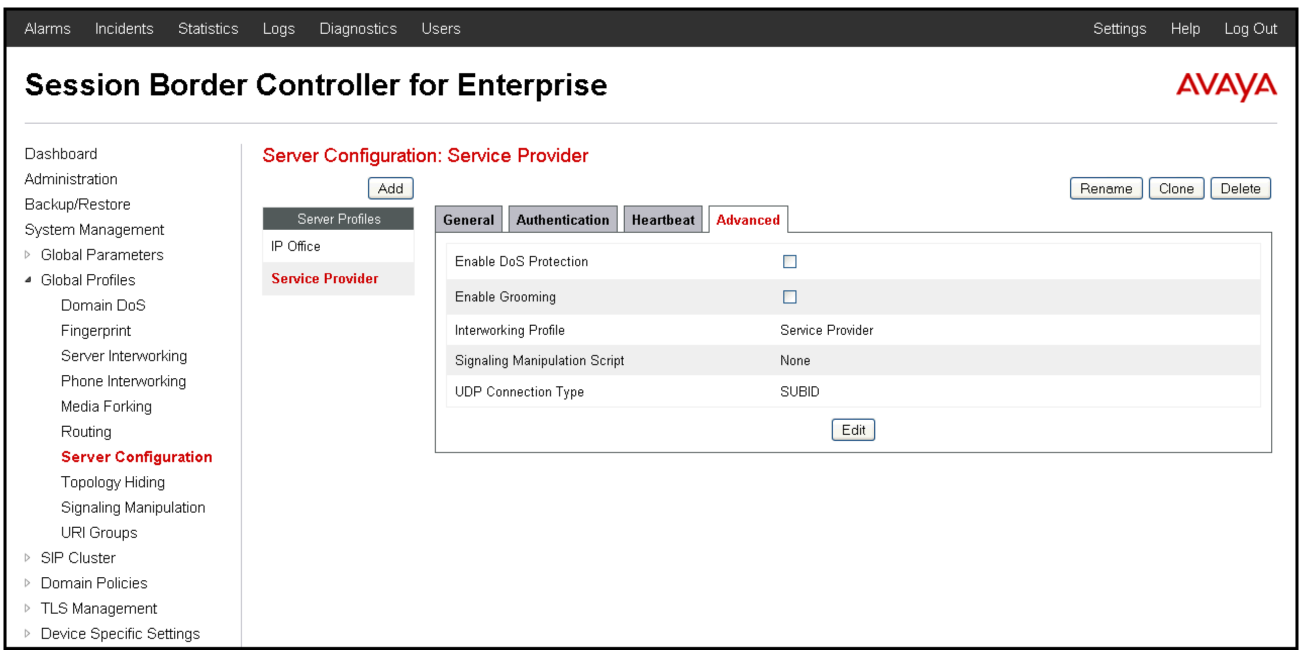
On the **Add Server Configuration Profile** Tab (not shown):

- Select Server Type**: Trunk Server.**
- **IP Address: 192.168.139.155** (service provider's SIP Proxy IP address).
- **Supported Transports**: Check **UDP**.
- **UDP Port: 5060.**
- Click **Next.**
- Click **Next** on the **Authentication** tab.
- Click **Next** on the **Heartbeat** tab.
- On the **Advanced** tab, select **Service Provider** from the **Interworking Profile** drop down menu.
  Leave the **Signaling Manipulation Script** at the default **None**.
- Click **Finish.**

The following screen capture shows the **General** tab of the **Service Provider** Profile.



The following screen capture shows the **Advanced** tab of the **Service Provider** Profile.



6.2.5 Topology Hiding

Topology Hiding is a security feature which allows changing several parameters of the SIP packets, preventing private enterprise network information from being propagated to the un-trusted public network.

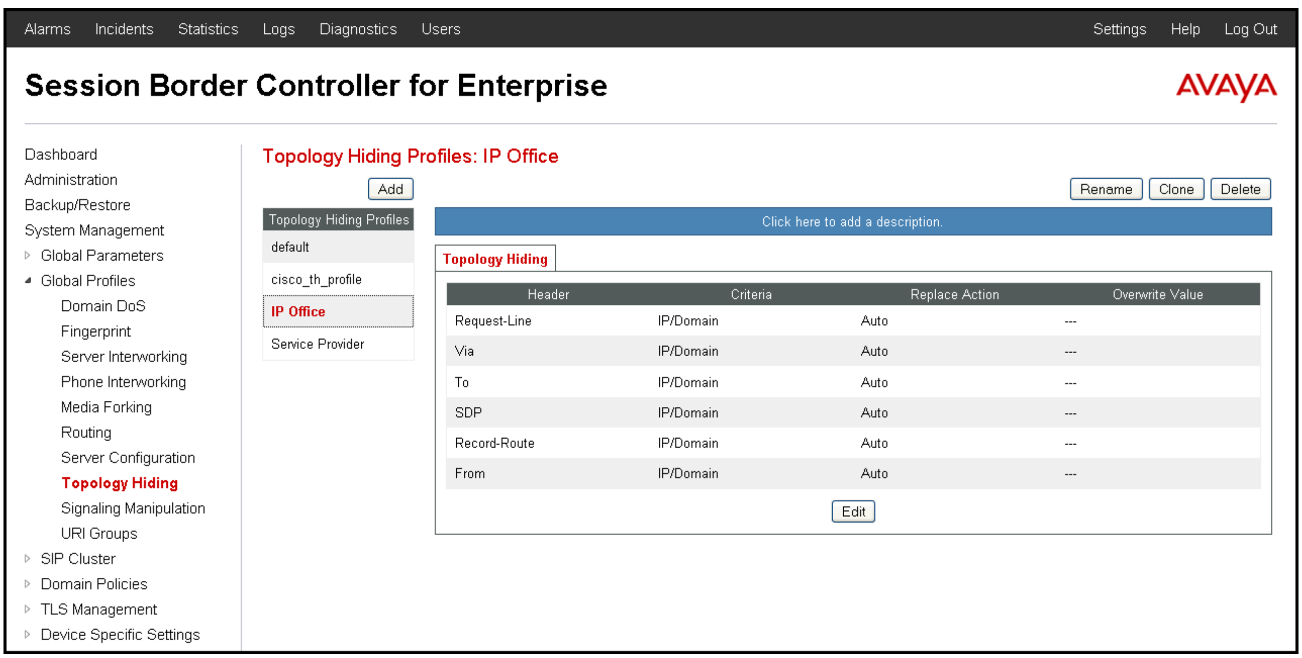
Topology Hiding can also be used as an interoperability tool to adapt the host portion in SIP headers like To, From, Request-URI, Via, Record-Route and SDP to the IP addresses or domains expected by Session Manager and the SIP trunk service provider, allowing the call to be accepted in each case.

For the compliance test, only the minimum configuration required to achieve interoperability on the SIP trunk was performed. Additional steps can be taken in this section to further mask the information that is sent from the Enterprise to the public network.

To add the Topology Hiding Profile in the Enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side:

- Click on **default** profile and select **Clone Profile.**
- Enter the **Profile Name**: **IP Office**.
- Click **Finish**.

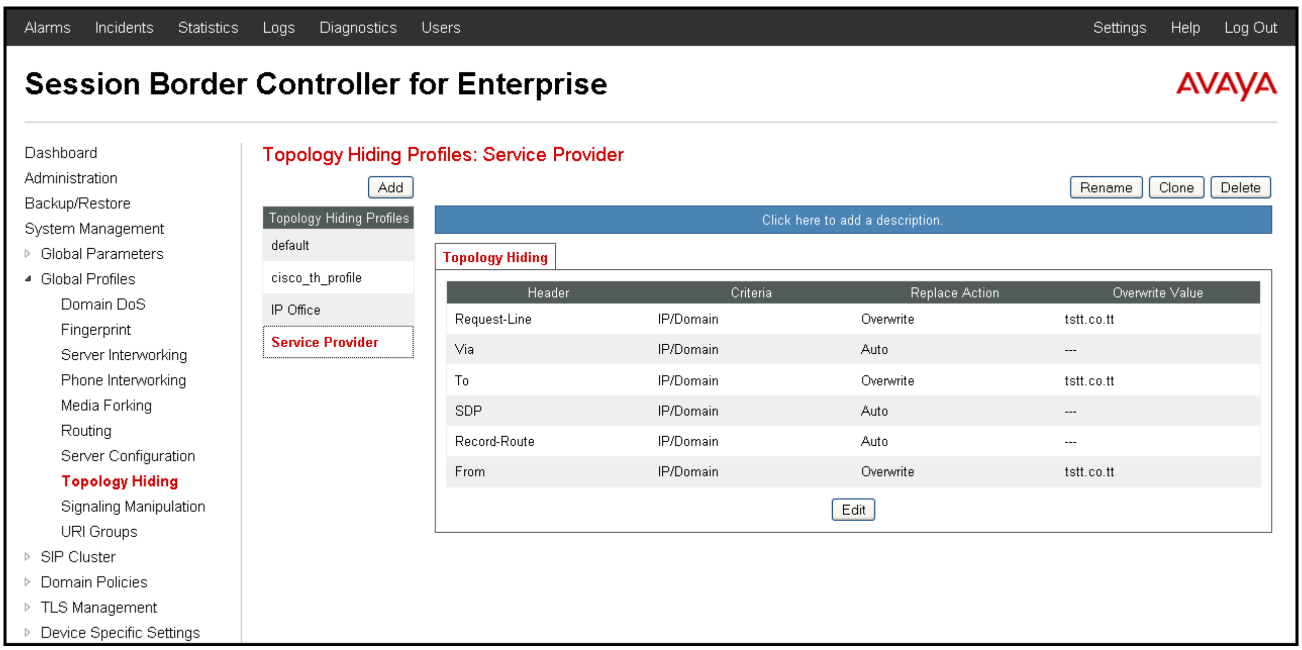
The following screen capture shows the newly added **IP Office** Profile. Note that for IP Office no values were overwritten (default).



To add the Topology Hiding Profile in the Service Provider direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side:

- Click on **default** profile and select **Clone Profile**
- Enter the **Profile Name**: **Service_Provider.**
- Click **Finish**.
- Click **Edit** on the newly added **Service Provider** Topology Hiding profile.
- In the **From** choose **Overwrite** from the pull-down menu under **Replace Action,** enter the domain name for the enterprise **(tstt.co.tt)** under **Overwrite Value**.
- In the **To** choose **Overwrite** from the pull-down menu under **Replace Action,** enter the domain name for the Enterprise **(tstt.co.tt)** under **Overwrite Value**.
- In the **Request-Line** choose **Overwrite** from the pull-down menu under **Replace Action,** enter the domain name for the Enterprise **(tstt.co.tt)** under **Overwrite Value**.

The following screen capture shows the newly added **Service_Provider** Profile.



6.2.6 Signaling Manipulation

The Signaling Manipulation feature of the Avaya SBCE allows to perform a granular header manipulation on the headers in the SIP messages, which sometimes is not possible by direct configuration on the web interface. The ability to configure header manipulation in such a highly flexible manner is achieved by the use of a proprietary scripting language called SigMa.

Signaling Manipulation was not necessary and was not used during the compliance testing.

6.3 Domain Policies

Domain Policies allow the configuration of sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise. Domain Policies include rules for Application, Media, Signaling, Security, etc.

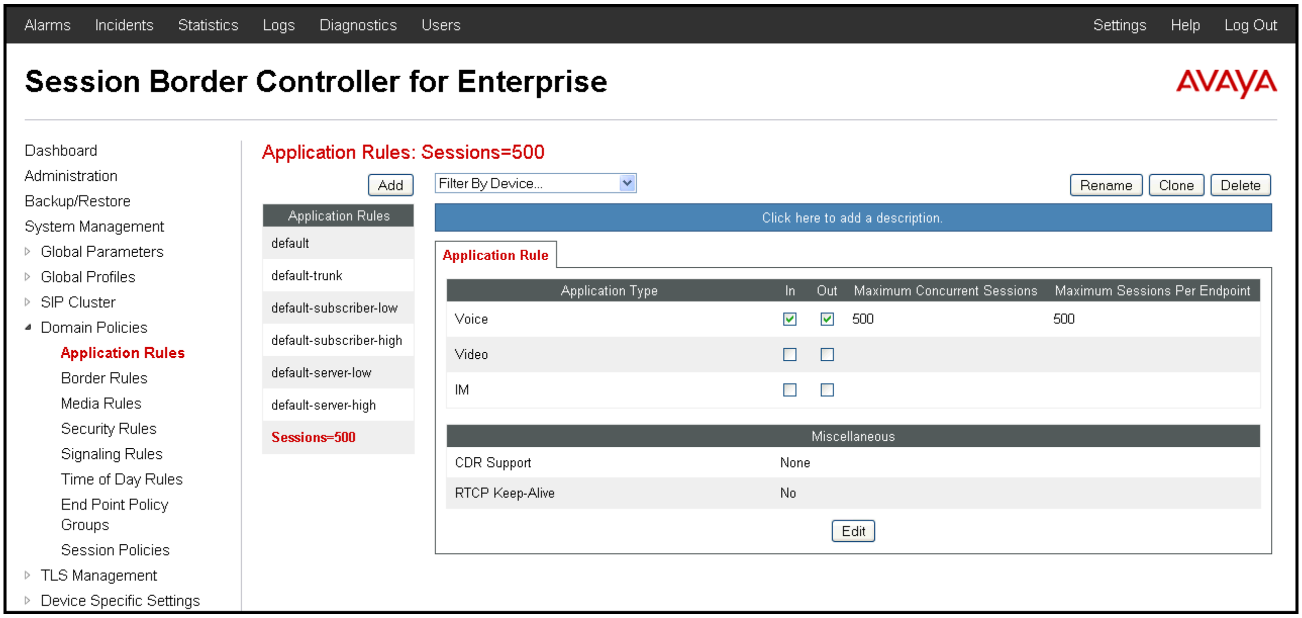
In the reference configuration, only a new Application Rule was defined. All other rules under Domain Policies, linked together on End Point Policy Groups, used one of the default sets already pre-defined in the configuration. Please note that changes should not be made to any of the defaults. If changes are needed, it is recommended to create a new rule by cloning one the defaults and then make the necessary changes to the new rule.

6.3.1 Create Application Rules

Application Rules defines 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, Application Rules defines the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion. From the menu on the left-hand side, select
Domain Policies → Application Rules
Select **default trunk** Rule (not shown)
Select **Clone Rule** button (not shown)
Name**: Sessions=500**

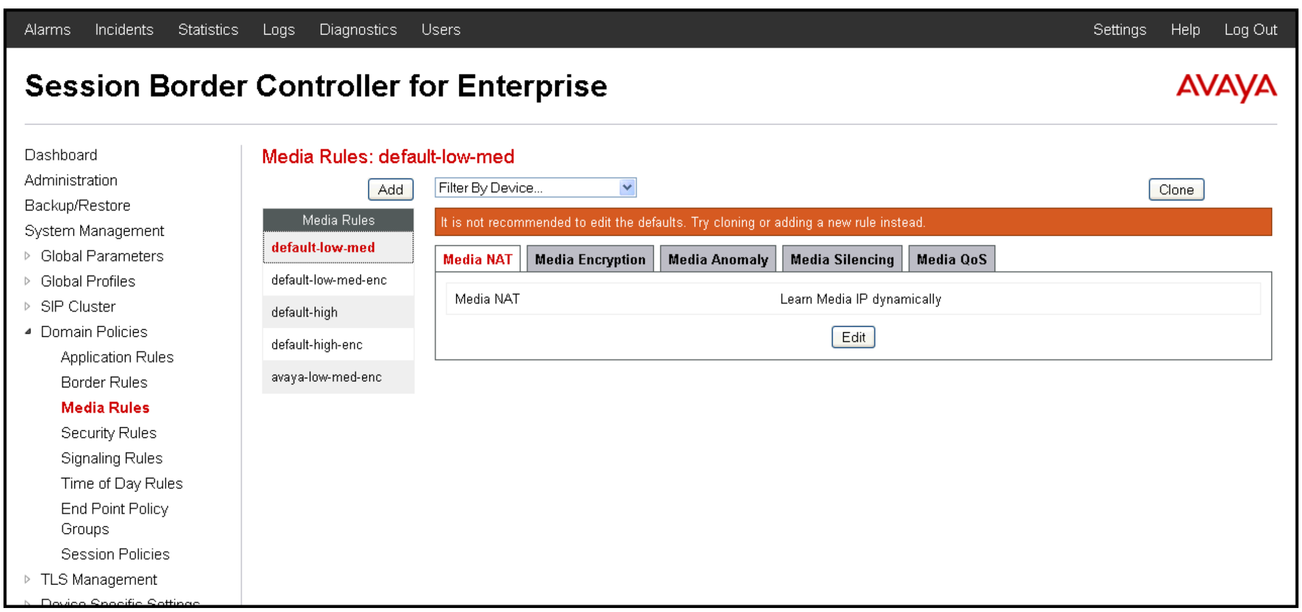
Set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** to recommended values, the value of **500** was used in the sample configuration.

Click Finish (not shown).



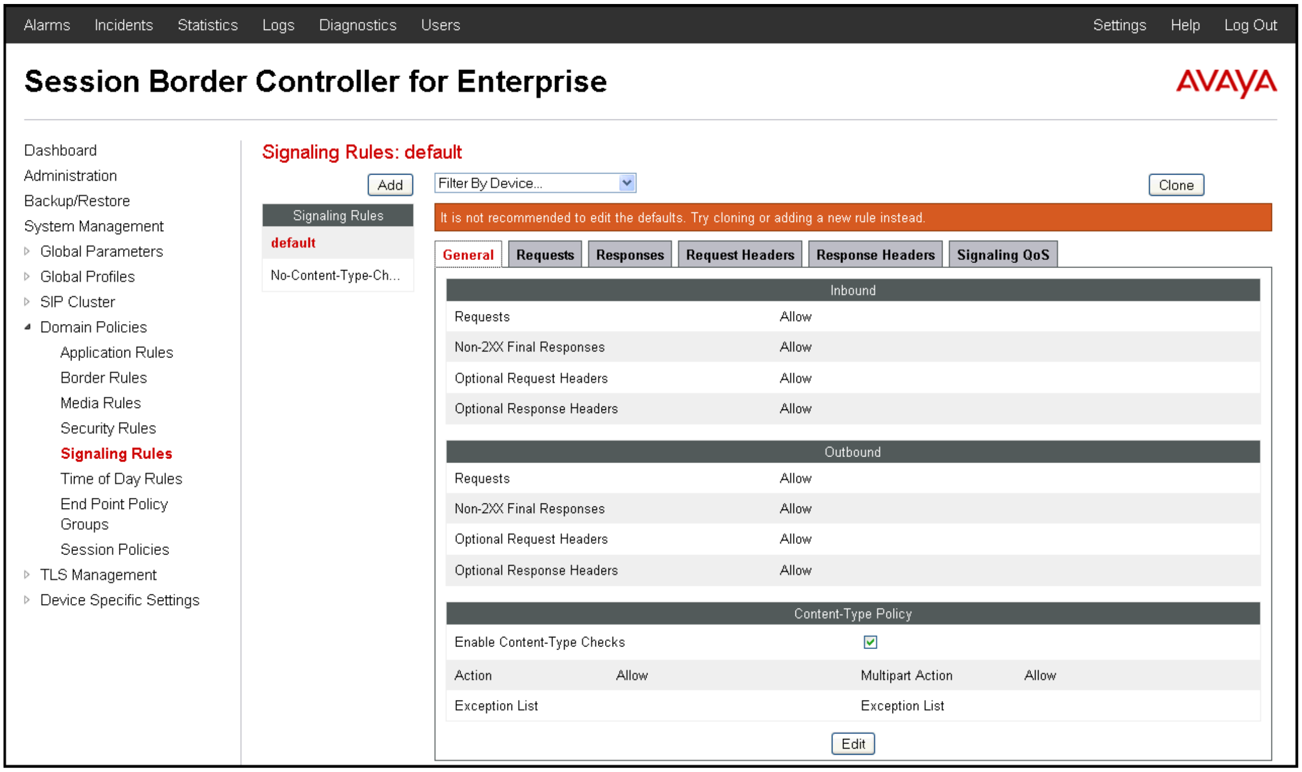
6.3.2 Media Rules

For the compliance test, the **default-low-med** Media Rule was used.



6.3.3 Signaling Rules

For the compliance test, the **default** Signaling Rule was used.



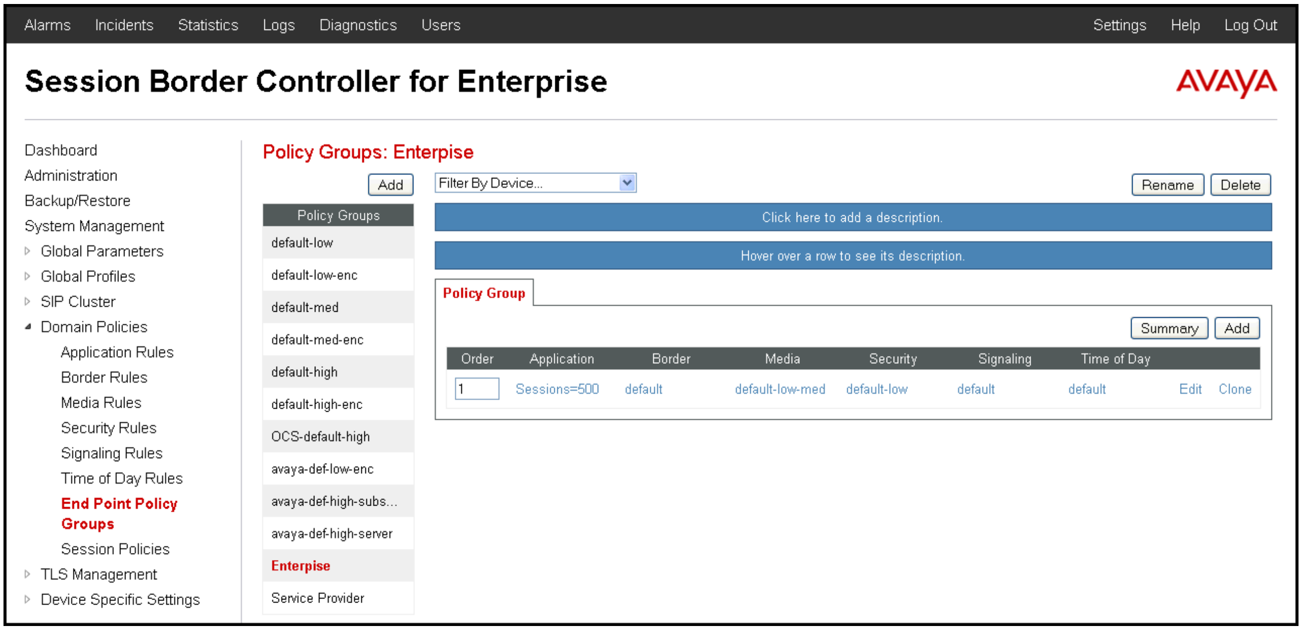
6.3.4 End Point Policy Groups

End Point Policy Groups are associations of different sets of rules (Media, Signaling, Security, etc) to be applied to specific SIP messages traversing through the Avaya SBCE.

To create an End Point Policy Group for the Enterprise, from the **Domain Policies** menu, select **End Point Policy Groups**. Select **Add**.

- **Group Name: Enterprise**.
- **Application Rule: Sessions=500.**
- **Border Rule: default.**
- **Media Rule: default-low-med.**
- **Security Rule: default-low.**
- **Signaling Rule: default.**
- **Time of Day: default.**
- Click **Finish**.

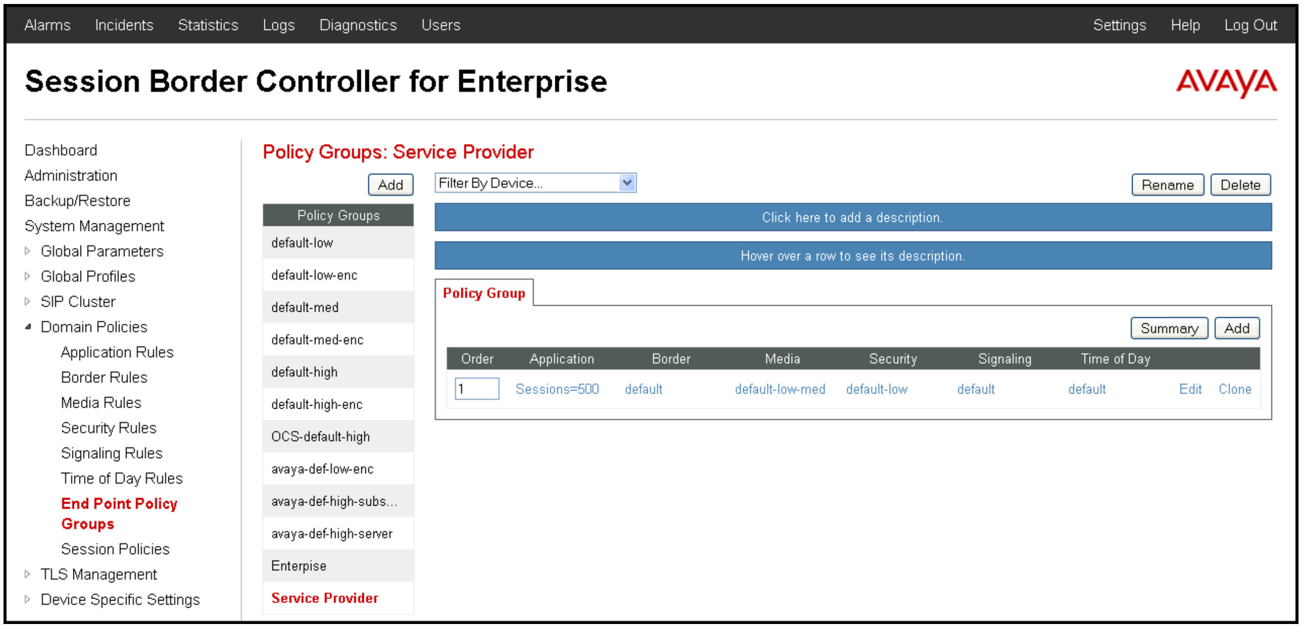
The following screen capture shows the newly added **Enterprise** End Point Policy Group.



Similarly, to create an End Point Policy Group for the Service Provider SIP Trunk, select **Add**.

- **Group Name: Service Provider**.
- **Application Rule: Sessions=500.**
- **Border Rule: default.**
- **Media Rule: default-low-med.**
- **Security Rule: default-low.**
- **Signaling Rule: default.**
- **Time of Day: default.**
- Click **Finish**.

The following screen capture shows the newly added **Service Provider** End Point Policy Group.



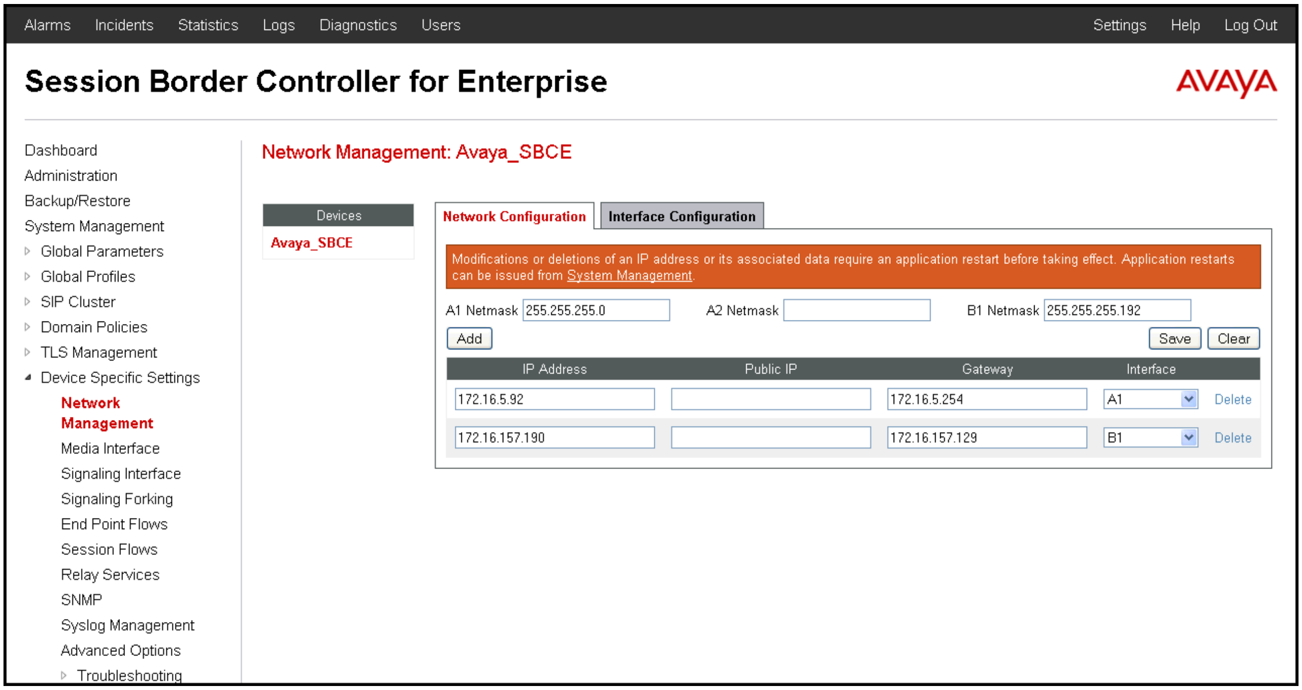
6.4 Device Specific Settings

The **Device Specific Settings** allow the management of various device-specific parameters, which determine how a particular device will function when deployed in the network. Specific server parameters, like network and interface settings, as well as call flows, etc. are defined here.

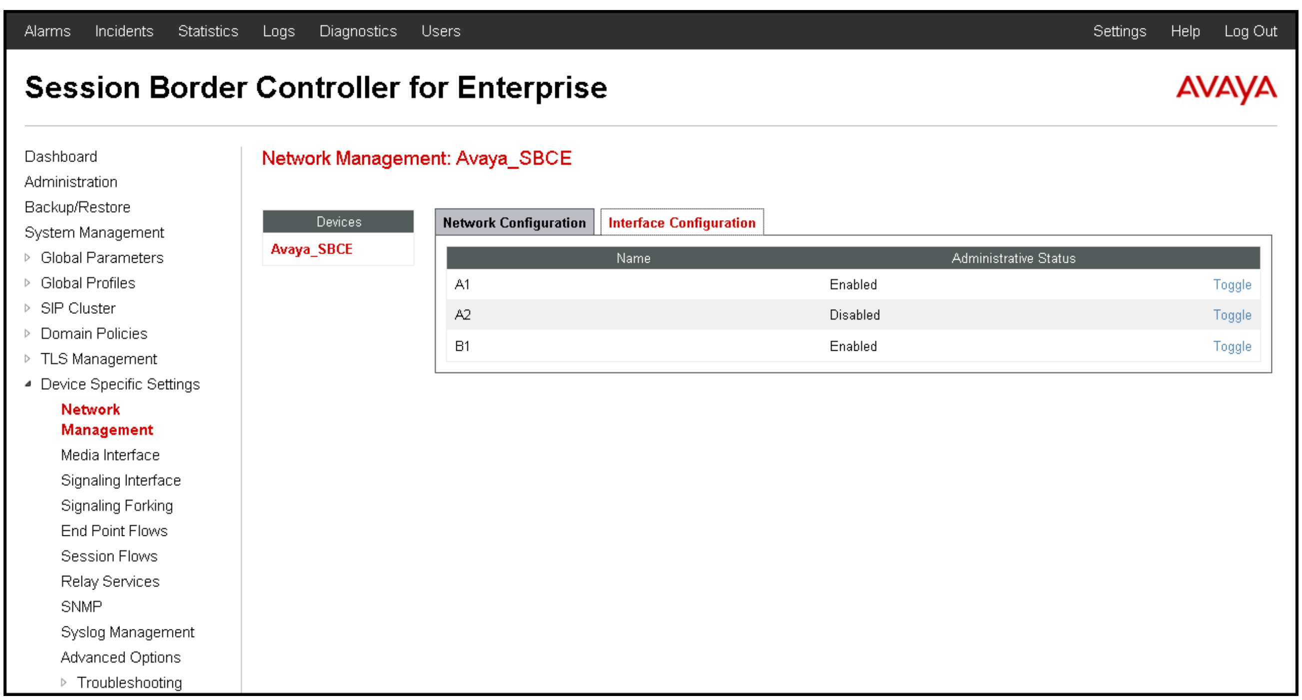
6.4.1 Network Management

The network information should have been previously completed. To verify the network configuration, from the **Device Specific Menu** on the left hand side, select **Network Management**. Select the **Network Configuration** tab.

In the event that changes need to be made to the network configuration information, they could be entered here.



On the Interface Configuration tab, click the **Toggle** control for interfaces **A1** and **B1 to** change the status to **Enabled**. It should be noted that the default state for all interfaces is **disabled**, so it is important to perform this step, or the Avaya SBCE will not be able to communicate on any of its interfaces.



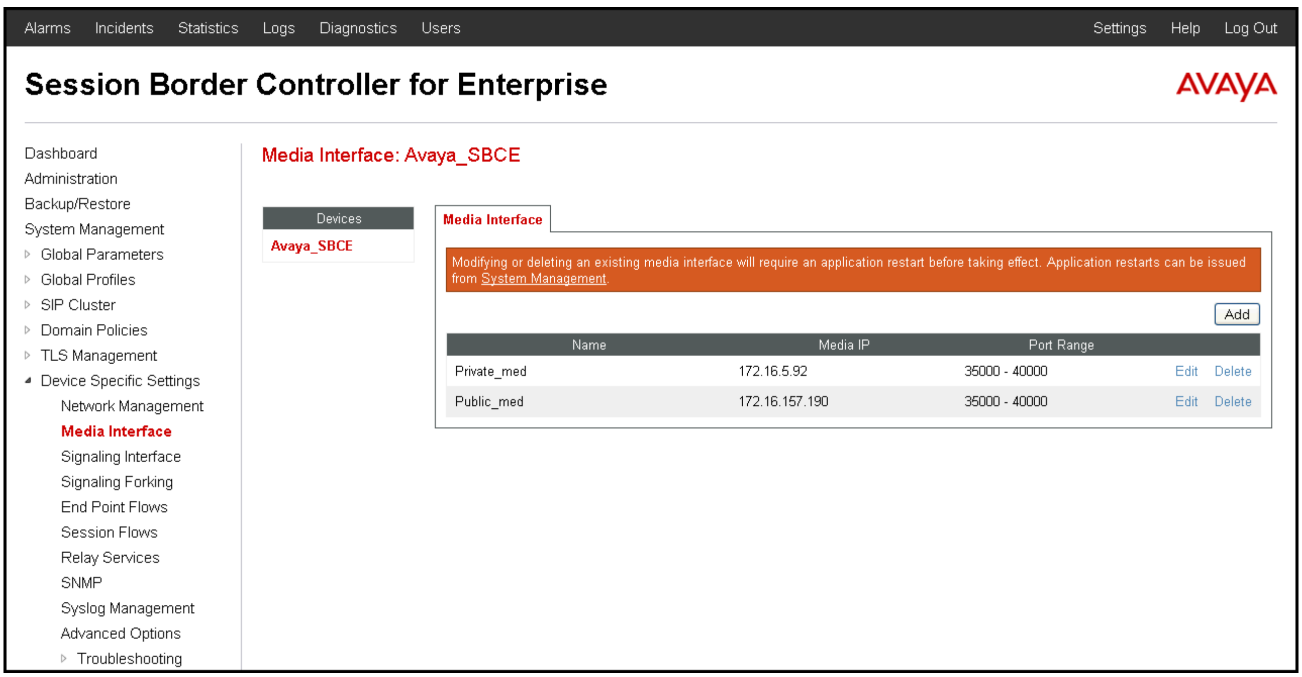
6.4.2 Media Interface

Media Interfaces were created to adjust the port range assigned to media streams leaving the interfaces of the Avaya SBCE. On the Private and Public interfaces of the Avaya SBCE ports range 35000 to 40000 was used.

From the **Device Specific Settings** menu on the left-hand side, select **Media Interface**.
Select **Add Media Interface** (not shown)

- **Name: Private.**
- Select **IP Address: 172.16.5.92** (Inside IP Address of the Avaya SBCE, toward IP Office).
- **Port Range: 35000-40000.**
- Click **Finish.**
- Select **Add Media Interface.**
- **Name: Public.**
- Select **IP Address: 172.16.157.190** (Outside IP Address of the Avaya SBCE, toward Service Provider).
- **Port Range: 35000-40000.**
- Click **Finish.**

The following screen capture shows the added **Media Interfaces**.

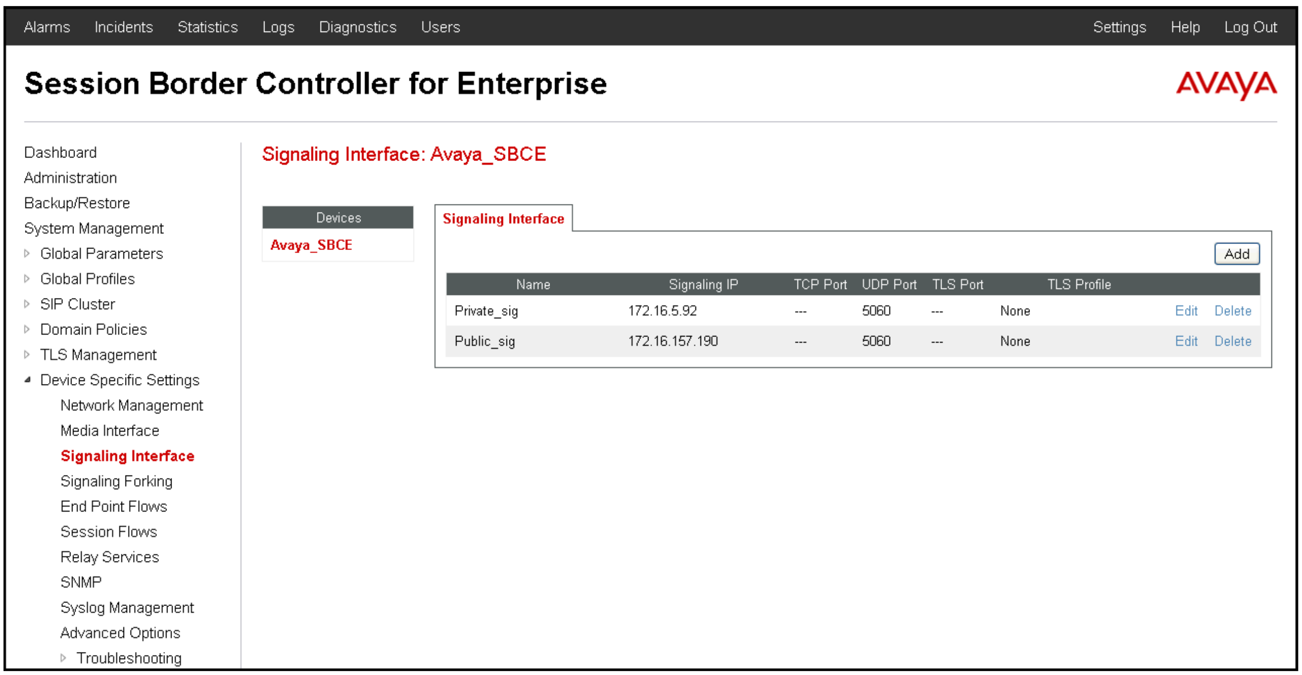


6.4.3 Signaling Interface

To create the Signaling Interface toward Session Manager, from the **Device Specific** menu on the left hand side, select **Signaling Interface**. Select **Add Signaling Interface** (not shown):

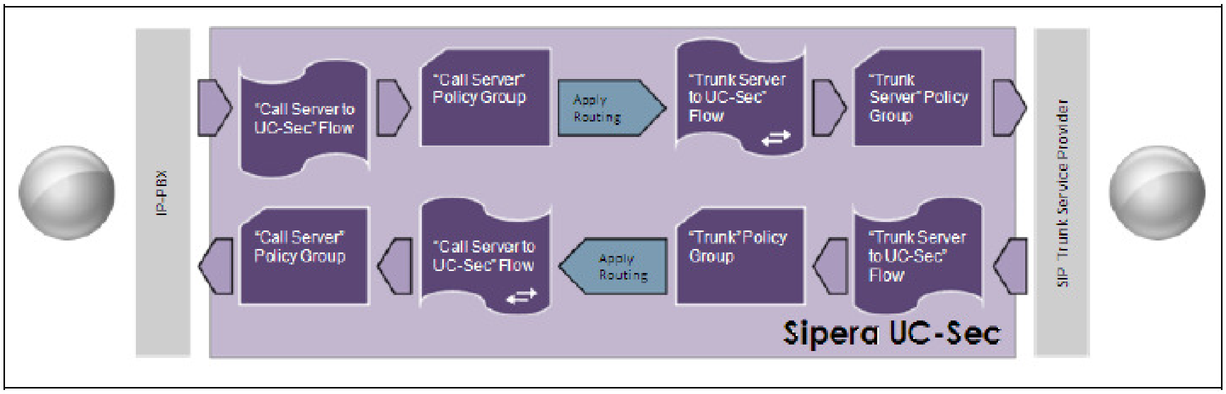
- **Name: Private.**
- Select **IP Address: 172.16.5.92** (Inside IP Address of the Avaya SBCE, toward IP Office).
- **UDP Port: 5060.**
- Click **Finish.**
- Select **Add Signaling Interface**:
- **Name: Public**
- Select **IP Address: 172.16.157.190** (Outside IP Address of the Avaya SBCE, toward the Service Provider).
- **UDP Port: 5060.**
- Click **Finish.**

The following screen capture shows the newly added **Signaling Interfaces**.



6.4.4 End Point Flows

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



The **End-Point Flows** defines certain parameters that pertain to the signaling and media portions of a call, whether it originates from within the enterprise or outside of the enterprise.

To create the call flow toward the Service Provider SIP trunk, from the **Device Specific Settings** menu, select **End Point Flows**, tab **Server Flows**. Click **Add Flow**.

- **Name: SIP Trunk Flow.**
- **Server Configuration**: **Service Provider.**
- **URI Group: ***
- **Transport: ***
- **Remote Subnet: ***
- **Received Interface**: **Private_sig.**
- **Signaling Interface: Public_sig.**
- **Media Interface**: **Public_med.**
- **End Point Policy Group: Service Provider.**
- **Routing Profile: Route to IP Office** (Note that this is the reverse route of the flow).
- **Topology Hiding Profile: Service Provider.**
- **File Transfer Profile: None.**
- Click **Finish.**

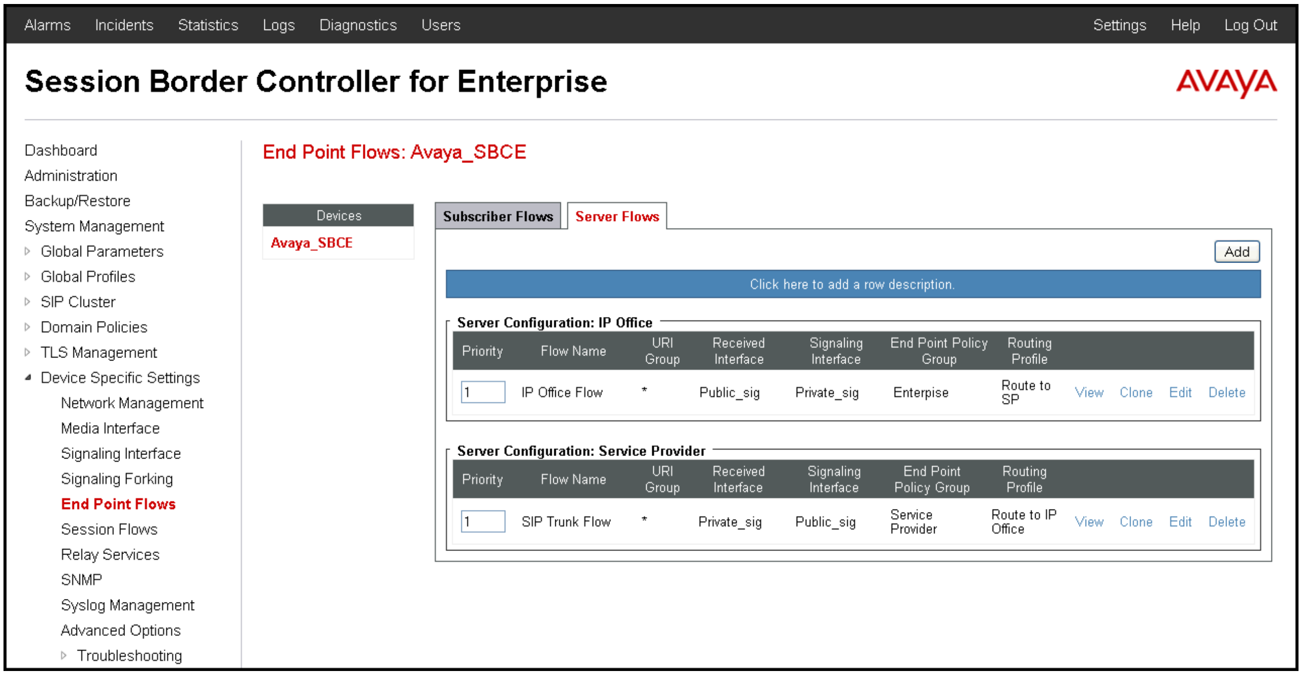
View Flow: SIP Trunk Flow			
Criteria		**Profile**	
Flow Name	SIP Trunk Flow	Signaling Interface	Public_sig
Server Configuration	Service Provider	Media Interface	Public_med
URI Group	*	End Point Policy Group	Service Provider
Transport	*	Routing Profile	Route to IP Office
Remote Subnet	*	Topology Hiding Profile	Service Provider
Received Interface	Private_sig	File Transfer Profile	None

To create the call flow toward the IP Office, click **Add Flow**.

- **Name: IP Office Flow.**
- **Server Configuration**: **IP Office.**
- **URI Group: ***
- **Transport: ***
- **Remote Subnet: ***
- **Received Interface**: **Public_sig.**
- **Signaling Interface: Private_sig.**
- **Media Interface**: **Private_med.**
- **End Point Policy Group: Enterprise.**
- **Routing Profile: Route to SP** (Note that this is the reverse route of the flow).
- **Topology Hiding Profile: IP Office.**
- **File Transfer Profile: None.**
- Click **Finish.**

View Flow: IP Office Flow			
Criteria		**Profile**	
Flow Name	IP Office Flow	Signaling Interface	Private_sig
Server Configuration	IP Office	Media Interface	Private_med
URI Group	*	End Point Policy Group	Enterpise
Transport	*	Routing Profile	Route to SP
Remote Subnet	*	Topology Hiding Profile	IP Office
Received Interface	Public_sig	File Transfer Profile	None

The following screen capture shows the added **End Point Flows.**



7. Telecommunications Services of Trinidad and Tobago SIP Trunking Configuration

TSTT is responsible for the configuration of the SIP Trunk Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. TSTT will provide the customer the necessary information to configure the Avaya IP Office SIP trunk connection, including:

- IP address of the TSTT SIP Proxy server.
- Supported codec's and order of preference.
- DID numbers.
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

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 tips that can be used to troubleshoot the solution.

8.1 Verification Steps

The following steps may be used to verify the configuration:

- Verify that endpoints at the enterprise site can place calls to PSTN and that calls remain 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.
- Verify that endpoints at the enterprise site can receive calls from PSTN and that calls can remain active for more than 35 seconds.
- Verify that the user on the PSTN side can end an active call by hanging up.
- Verify that an Avaya endpoint at the enterprise site can end an active call by hanging up.

8.2 Protocol Traces

The following SIP message headers are inspected using sniffer trace analysis tool:

- Request-URI: Verify the request number and SIP domain.
- From: Verify the display name and display number.
- To: Verify the display name and display number.
- P-Asserted-Identity: Verify the display name and display number.
- Privacy: Verify privacy masking with “user, id”.
- Diversion: Verify the display name and display number.

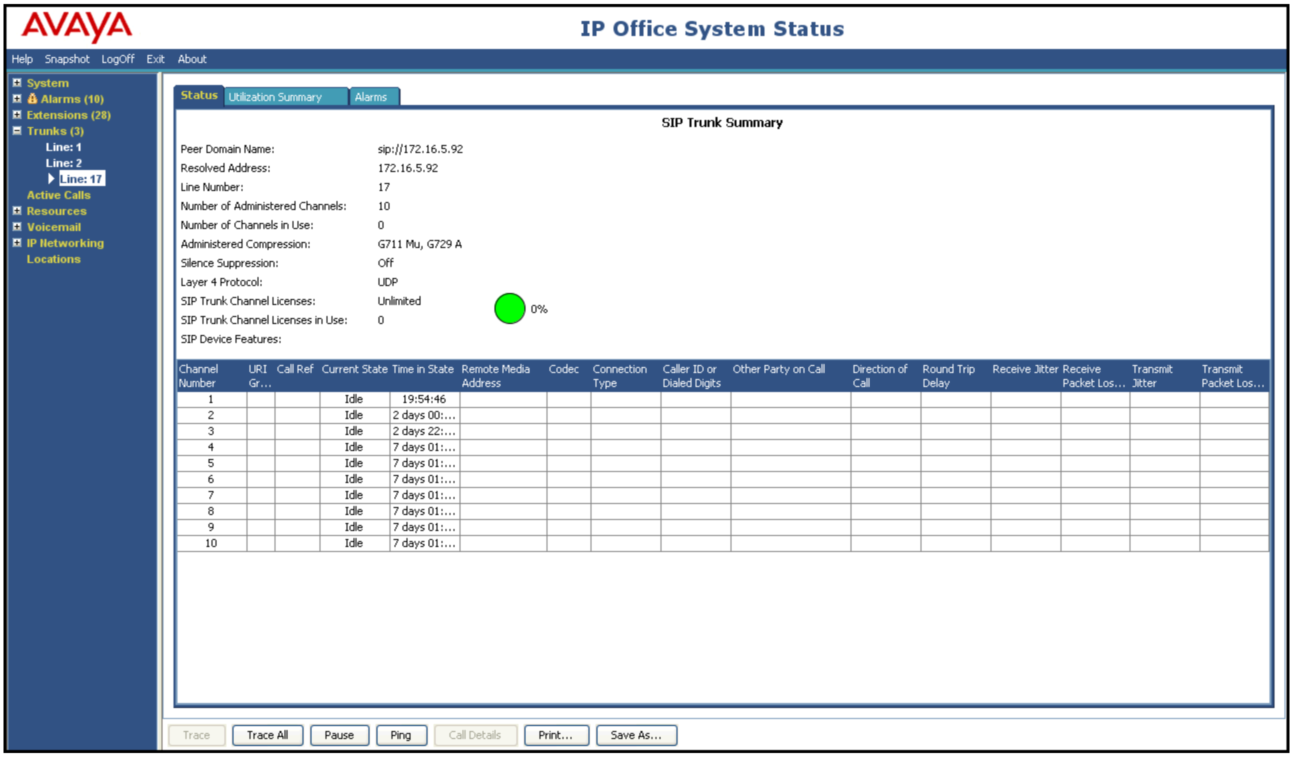
The following attributes in SIP message body are inspected using sniffer trace analysis tool:

- Connection Information (c line): Verify IP addresses of near end and far end endpoints.
- Time Description (t line): Verify session timeout value of near end and far end endpoints.
- Media Description (m line): Verify audio port, codec, DTMF event description.
- Media Attribute (a line): Verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes.

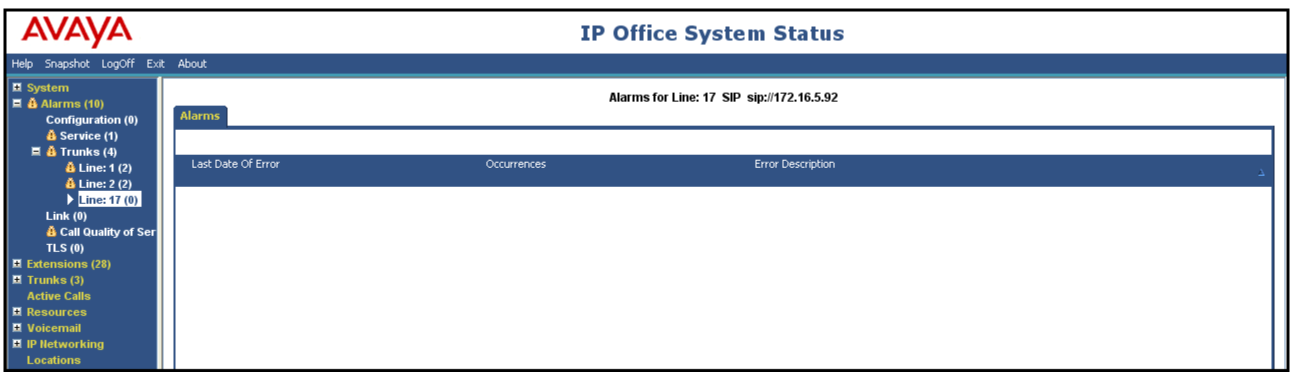
8.3 IP Office System Status

The following steps can also be used to verify the configuration.

- Use the Avaya IP Office **System Status** application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where IP Office Manager is installed. Select the SIP Line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is **Idle** for each channel (assuming no active calls at present time).



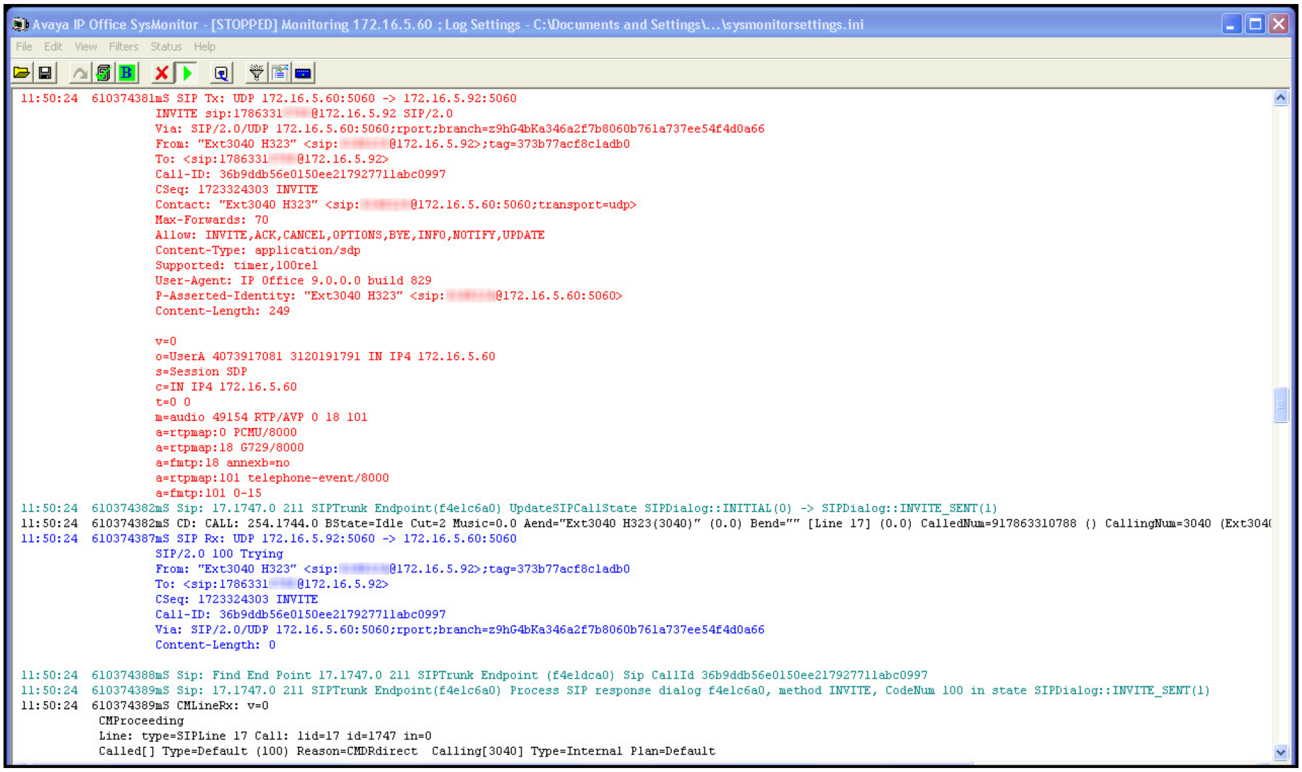
- Select the **Alarms** tab and verify that no alarms are active on the SIP Line.



8.4 IP Office Monitor

The Avaya IP Office Monitor application can also be used to monitor and troubleshoot SIP signaling messaging between TSTT and IP Office. Launch the application from **Start →Programs →IP Office →Monitor** on the PC where Avaya IP Office Manager was installed.

The sample screen below shows part of the messages on an outbound call.



9. Conclusion

These Application Notes describe the procedures required to configure SIP trunk connectivity between Avaya IP Office 9.0, Avaya Session Border Controller for Enterprise R6.2 and Telecommunications Services of Trinidad and Tobago SIP Trunk Service, as shown in **Figure 1**.

Interoperability testing was completed successfully with the observations/limitations noted in **Section 2.2**

10. References

[1] *IP Office 9.0 Installing IP500/IP500 V2*, Document Number 15-601042 Issue 28g - (11 October 2013)

[2] *IP Office Manager Release 9.0*, Document Number 15-601011 Issue 9.01 (Monday, September 09, 2013).

[3] *IP Office 9.0 Administering Voicemail Pro*, Document Number 15-601063 Issue 9.01.0 - (Tuesday, September 10, 2013)

[4] IP Office 9.0 Installing IP Office Video Softphone, Issue 4c - (21 August 2013)

[5] *Administering Avaya Flare ® Experience for IPad devices and Windows,* Release 9.0 Issue 02.01 September 2013.

[6] *Administering Avaya Session Border Controller for Enterprise,* Release 6.2, Issue 2, March 2013.

[7] *Installing Avaya Session Border Controller for Enterprise,* Release 6.2, Issue 2, March 2013.

[8] *Upgrading Avaya Session Border Controller for Enterprise,* Release 6.2, Issue 2, March 2013.

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

Product documentation for TSTT SIP Trunking Service is available from TSTT.

11. SIP Line Template

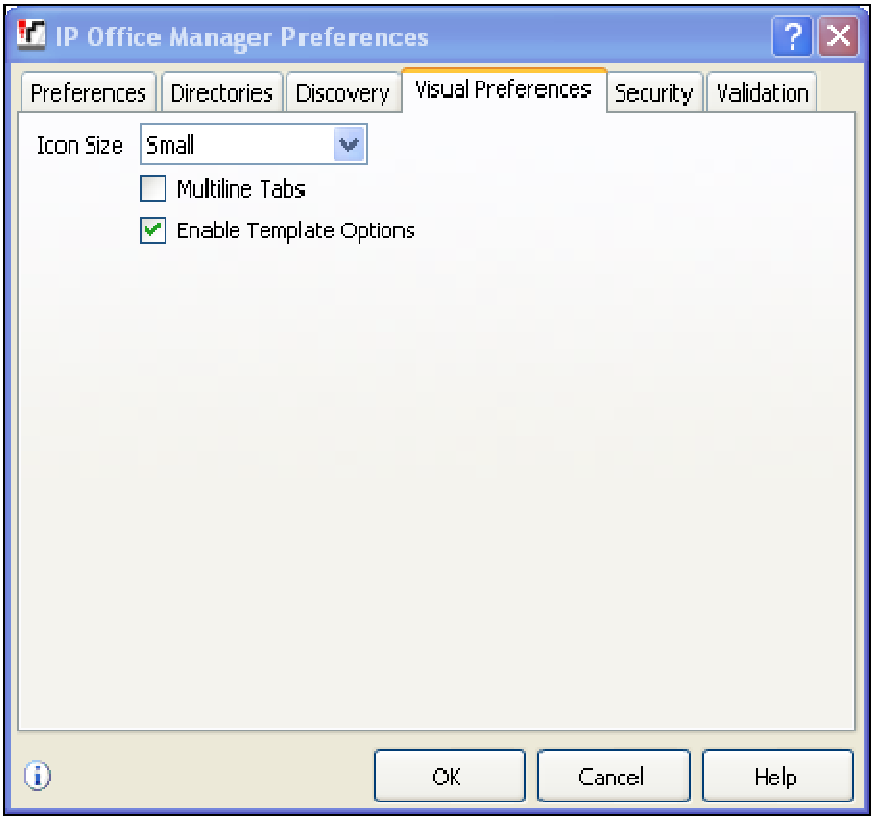
This Appendix describes how IP Office Manager Template Provisioning can be used to simplify the configuration of SIP Lines in IP Office. The Template Provisioning feature was introduced in IP Office Release 7.0.

11.1 Create a New SIP Trunk from Template

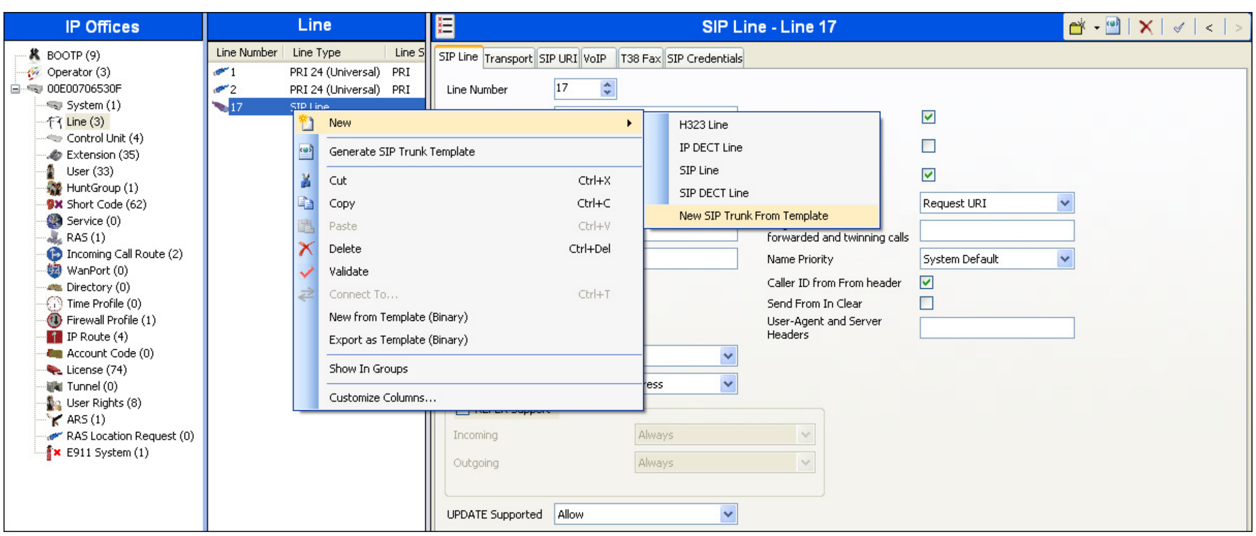
This section describes the steps performed by an IP Office system administrator to use Manager to create a new SIP Line using a previously generated template. Please follow these steps very carefully to avoid using 'New from Template (Binary)'. The binary templates ARE NOT to be used because binary templates also include IP Office system specific details to the customers IP Office including SIP line credentials and SIP line SIP URIs.

- The IP Office system administrator must place the template xml file in the Manager Templates folder. The default folder is the Templates folder under the Manager installation folder. On Windows XP the folder is C:\Program Files\Avaya\IP Office\Manager\Templates. Templates stored in a non default folder can be imported into Manager using Tools → Import Templates in Manager.
- In Manager, the administrator must ensure Manager template options are enabled. When enabled, the Manager can be used to apply trunk templates. SIP trunk templates can be used to add SIP trunks.

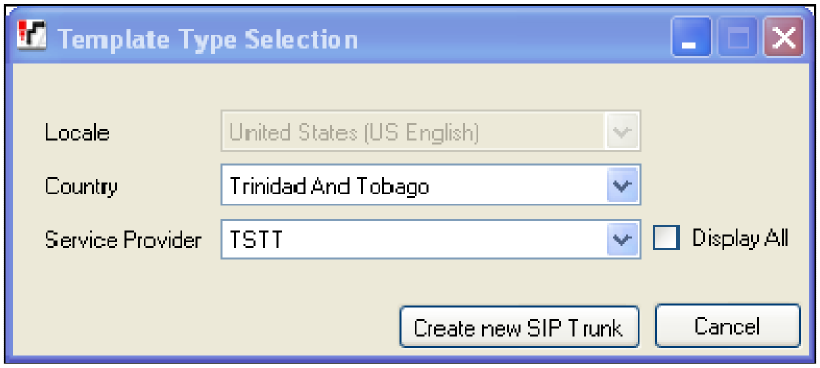
To enable template support in the IP Office Manager, select **File**, then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.



Next, import the template into the new IP Office system by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New, New SIP Trunk From Template**:



On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.



The following is the exported SIP Line Template file **TT_TSTT_SIPTrunk.xml** created after the testing was completed:


```
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
<TemplateType>SIPTrunk</TemplateType>
<Version>20131101</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>TSTT IPO 9.0</DescriptiveName>
<ITSPDomainName>tstt.co.tt</ITSPDomainName>
<SendCallerID>CallerIDDIV</SendCallerID>
<ReferSupport>false</ReferSupport>
<ReferSupportIncoming>1</ReferSupportIncoming>
<ReferSupportOutgoing>1</ReferSupportOutgoing>
<RegistrationRequired>false</RegistrationRequired>
<UseTelURI>false</UseTelURI>
<CheckOOS>true</CheckOOS>
<CallRoutingMethod>1</CallRoutingMethod>
<OriginatorNumber />
<AssociationMethod>SourceIP</AssociationMethod>
<LineNamePriority>SystemDefault</LineNamePriority>
<UpdateSupport>UpdateAuto</UpdateSupport>
<URIType>SIPURI</URIType>
<UserAgentServerHeader />
<CallerIDfromFromheader>true</CallerIDfromFromheader>
<PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
<ITSPProxy>172.16.5.92</ITSPProxy>
<LayerFourProtocol>SipUDP</LayerFourProtocol>
<SendPort>5060</SendPort>
<ListenPort>5060</ListenPort>
<DNSServerOne>0.0.0.0</DNSServerOne>
<DNSServerTwo>0.0.0.0</DNSServerTwo>
<CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
<SeparateRegistrar />
```

<CompressionMode>**AUTOSELECT**</CompressionMode>
<UseAdvVoiceCodecPrefs>**true**</UseAdvVoiceCodecPrefs>
<AdvCodecPref>**G.711 ULAW 64K,G.729(a) 8K CS-ACELP**</AdvCodecPref>
<CallInitiationTimeout>**4**</CallInitiationTimeout>
<DTMFSupport>**DTMF_SUPPORT_RFC2833**</DTMFSupport>
<VoipSilenceSupression>**false**</VoipSilenceSupression>
<ReinviteSupported>**true**</ReinviteSupported>
<FaxTransportSupport>**FOIP_NONE**</FaxTransportSupport>
<UseOffererPrefferedCodec>**false**</UseOffererPrefferedCodec>
<CodecLockdown>**false**</CodecLockdown>
<Rel100Supported>**true**</Rel100Supported>
<T38FaxVersion>**3**</T38FaxVersion>
<Transport>**UDPTL**</Transport>
<LowSpeed>**0**</LowSpeed>
<HighSpeed>**0**</HighSpeed>
<TCFMethod>**Trans_TCF**</TCFMethod>
<MaxBitRate>**FaxRate_14400**</MaxBitRate>
<EflagStartTimer>**2600**</EflagStartTimer>
<EflagStopTimer>**2300**</EflagStopTimer>
<UseDefaultValues>**true**</UseDefaultValues>
<ScanLineFixup>**true**</ScanLineFixup>
<TFOPEnhancement>**true**</TFOPEnhancement>
<DisableT30ECM>**false**</DisableT30ECM>
<DisableEflagsForFirstDIS>**false**</DisableEflagsForFirstDIS>
<DisableT30MRCompression>**false**</DisableT30MRCompression>
<NSFOverride>**false**</NSFOverride>
</Template>

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.