Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Intermedia SIP Trunking Using TLS/SRTP with Avaya IP Office 9.1 and Avaya Session Border Controller for Enterprise Release 7.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Intermedia Session Initiation Protocol (SIP) Trunking using TLS/SRTP with Avaya IP Office Release 9.1 and Avaya Session Border Controller for Enterprise Release 7.0.

Intermedia SIP Trunking provides PSTN access via a SIP trunk between the enterprise and the Intermedia network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Intermedia 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.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Intermedia and an Avaya IP Office solution using TLS/SRTP. In the sample configuration, Avaya IP Office solution consists of an Avaya IP Office 500v2 release 9.1.6, Avaya Session Border Controller for Enterprise release 7.0.1 (Avaya SBCE), Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, SIP, digital, and analog endpoints.

The Intermedia SIP Trunking service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to Intermedia SIP Trunking service in TLS/SRTP via Avaya SBCE. This configuration (shown in **Figure 1**) was used to exercise the features 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

A simulated enterprise site with Avaya IP Office was connected to Intermedia SIP Trunking service via Avaya SBCE. To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Response to SIP OPTIONS queries.
- SIP trunk registration with service provider.
- Incoming PSTN calls to various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya IP Office Softphone.
- Inbound and outbound long holding time call stability.
- Various call types including: local, long distance, international, outbound toll-free, operator service and directory assistance.
- Codec G.711MU and G.729.
- Caller number/ID presentation.

- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- Telephony features such as hold and resume, transfer, and conference.
- Fax G.711 Pass Through modes.
- Off-net call forwarding.
- Twinning to mobile phones on inbound calls.
- Avaya Communicator for Web (WebRTC).
- Remote Worker which allows Avaya SIP endpoints to connect directly to the public Internet as enterprise phones.

Note:
1. Remote Worker and Avaya Communicator for Web (WebRTC) were tested as part of this solution. The configuration necessary to support remote worker and Avaya Communicator for Web is beyond the scope of these Application Notes and are not included in these Application Notes. For these configuration details, see **Reference [8]** and **Reference [9]** respectively.

2.2. Test Results

Intermedia SIP Trunking passed compliance testing.

Items not supported or not tested included the following:

- Inbound toll-free is supported but was not tested as part of the compliance test.
- T.38 Fax is not supported.
- Operator Call (dial 0) and Operator Assisted (dial 0+10digits) are not supported.
- SIP OPTIONS sent by Intermedia is not supported.
- Intermedia does not support SIP Diversion Header.
- Call Redirection using SIP REFER is not supported by Intermedia.

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

- Intermedia does not send SIP OPTIONS message but responded to SIP OPTIONS message.

2.3. Support

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

For technical support on Intermedia SIP Trunking, contact Intermedia at https://www.intermedia.net

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to the Intermedia SIP Trunking service via Avaya SBCE through the public IP network. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

Located at the enterprise site is an Avaya IP Office Server Edition with IP Office 500v2 as expansion which provides connections for 16 digital stations and the extension PHONE 8 card which provides connections for 8 analog stations to the PSTN as well as 64-channel VCM (Voice Compression Module) for supporting VoIP codecs. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network. Endpoints include an Avaya 9600 Series IP Telephone (with H.323 firmware), Avaya 11x0 Series IP Telephone (with SIP firmware), an Avaya 9508 Digital Telephones, an Avaya Symphony 2000 Analog Telephone and an Avaya IP Office Softphone. A separate Windows OS PC runs Avaya IP Office Manager to configure and administer Avaya IP Office.

Mobility Twinning is configured for some of Avaya IP Office users so that calls to these user phones will also ring and can be answered at the configured mobile phones.

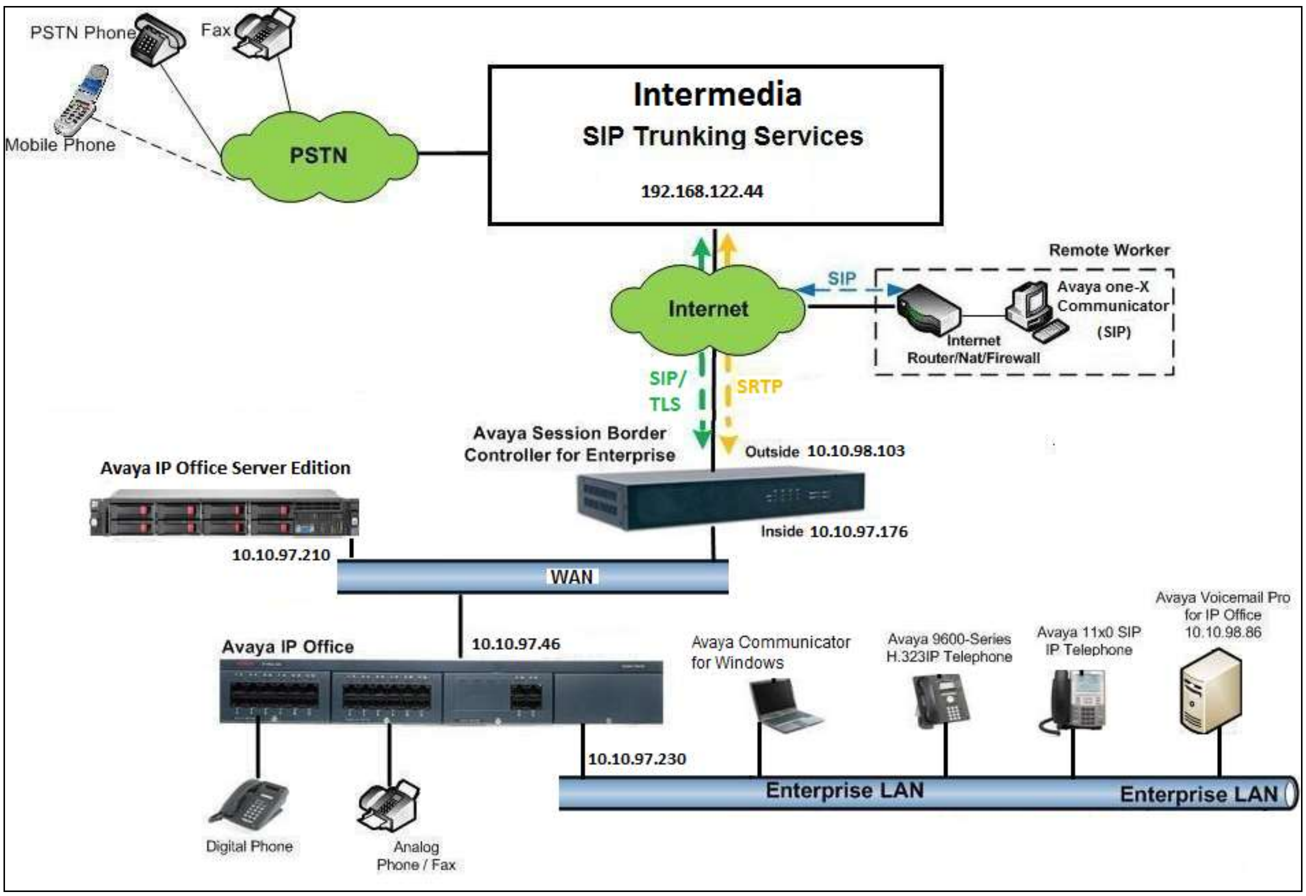


Figure 1: Test Configuration for Avaya IP Office with Intermedia SIP Trunking Service

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 6 + N digits to send digits across the SIP trunk to Intermedia. The short code of 6 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Intermedia. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office would send 11 digits in the Request URI and the To field of an outbound SIP INVITE message. It was configured to send 10 digits in the From field. For inbound calls, Intermedia SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

4. Equipment and Software Validated

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

Avaya Telephony Components	
Equipment	**Release**
Avaya IP Office Server Edition	9.1.600.153
Avaya IP Office 500v2 (Expansion)	9.1.600.153
Avaya IP Office Manager	9.1.600.153
Avaya Voicemail Pro for IP Office	9.1.600.153
Avaya Application Server	9.1.600.153
Avaya Session Border Controller for Enterprise (running on Portwell CAD-0208 platform)	7.0.1-03-8739
Avaya 11x0 IP Telephone (SIP)	SIP11x0e04.04.18.00
Avaya 9621G IP Telephone (H.323)	Avaya one-X® Deskphone Edition S9621
Avaya Communicator for Windows	2.0.3.40
Avaya Communicator for Web (WebRTC)	1.0.16.1217
Avaya Digital Telephone (9508)	0.45
Avaya Symphony 2000 Analog Telephone	N/A
Intermedia SIP Trunking Service Components	
Component	Release
Intermedia SBC	16.14.2
Intermedia Softswitch	16.14.2

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition without T.38 Fax Service.

5. Configure IP Office

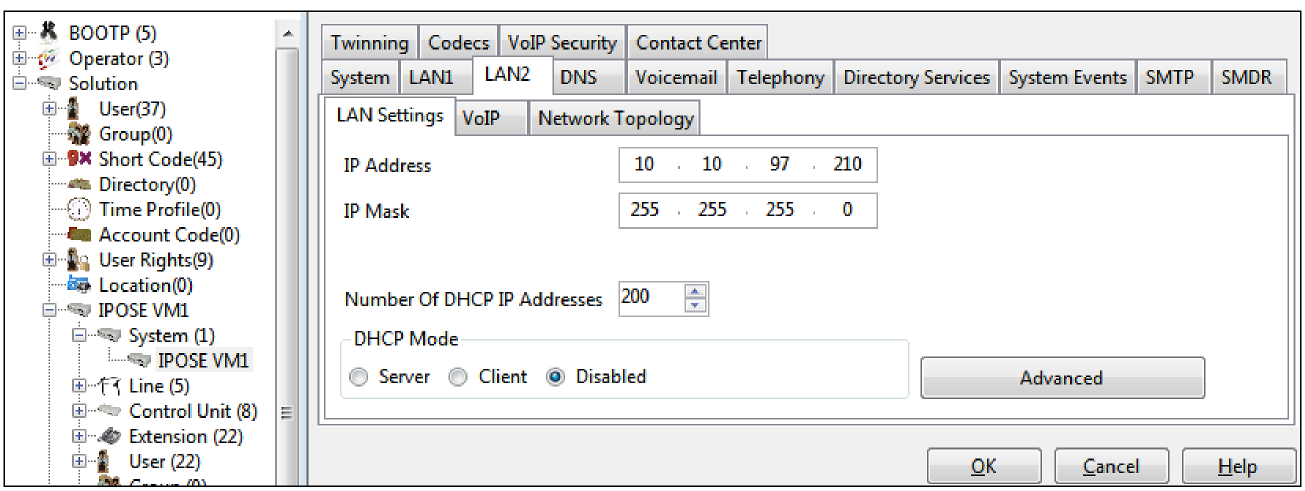
This section describes the Avaya IP Office configuration to support connectivity to Intermedia SIP Trunking service through Avaya SBCE. Avaya IP Office is configured through Avaya IP Office Manager PC application. From a PC running Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one shown in the next section. The appearance of IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interface with the service provider (such as LAN interface to the enterprise site and IP Office Softphone support) is assumed to be already in place.

5.1. LAN Settings

In the sample configuration, **IPO SP** was used as the system name and the WAN port was used to connect Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on Avaya IP Office.

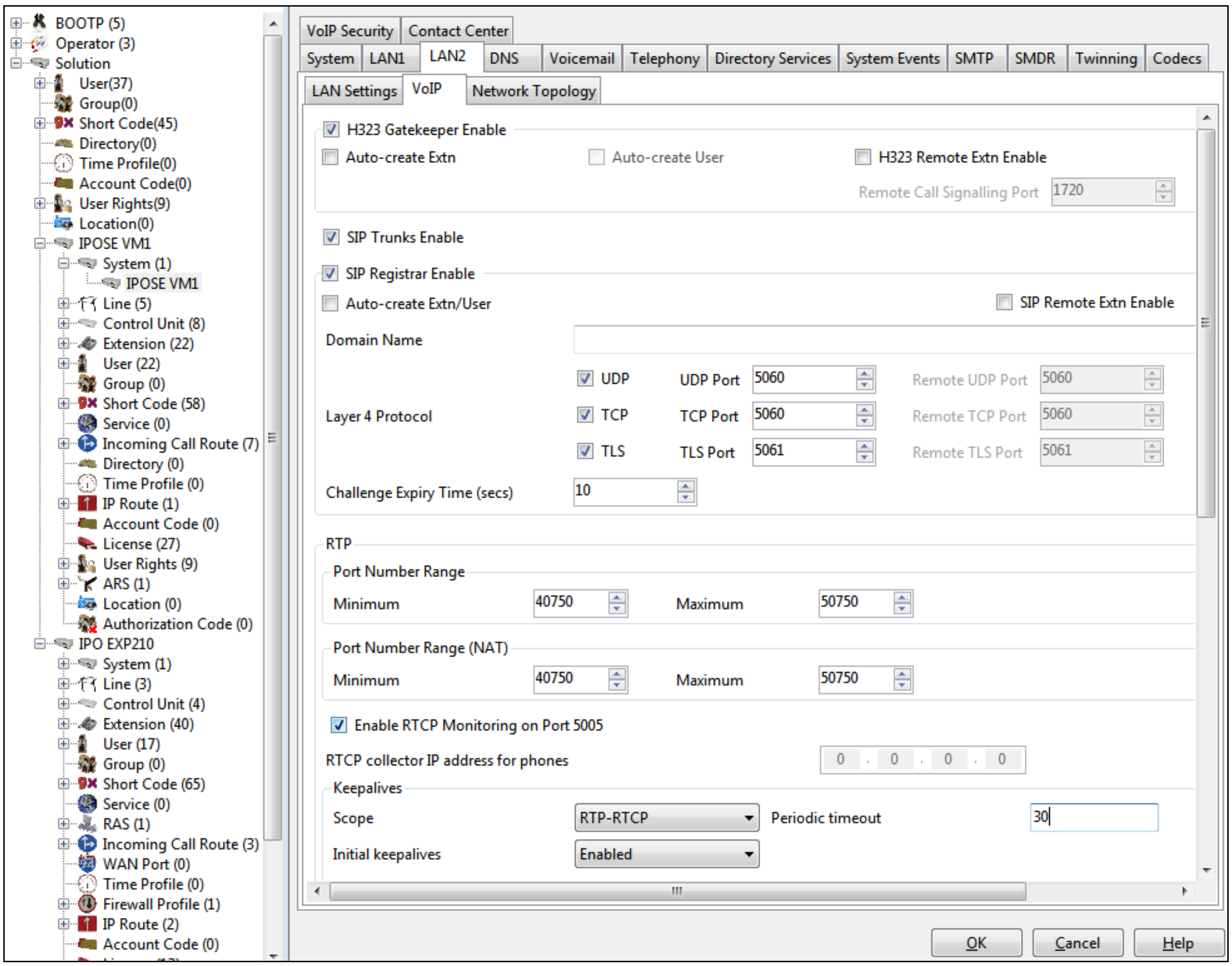
To access the LAN settings, first navigate to **System (1) → IPO SP** in the Navigation and Group Panes and then navigate to the **LAN2 → LAN Settings** tab in the Details Pane.

- Set the **IP Address** field to the IP address assigned to the IP Office WAN port.
- Set the **IP Mask** field to the mask used on the public network.
- All other parameters should be set according to customer requirements.
- Click **OK**.



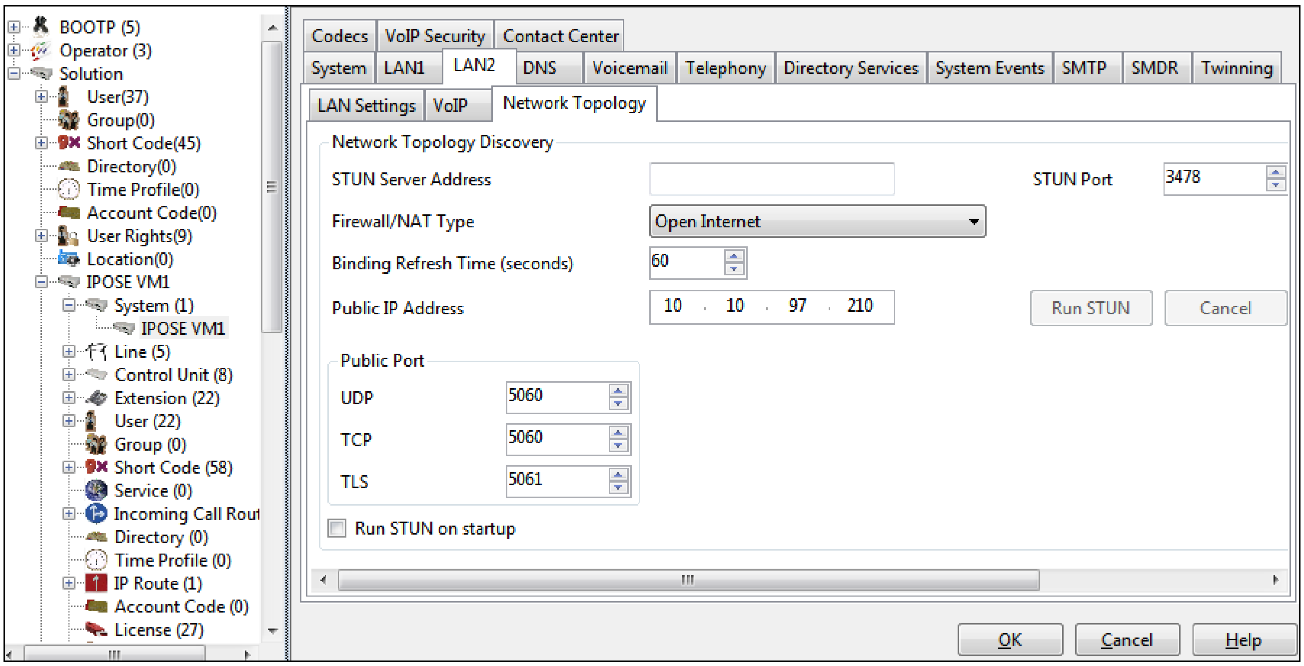
Select the **VoIP** tab as shown in the following screen.

- The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as 9600-Series IP Telephones used in the sample configuration.
- The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Intermedia.
- The **SIP Registrar Enable** box is checked to allow IP Office Softphone usage.
- The **Layer 4 Protocol,** check the **TLS** box and set **TLS Port** to ***5061***.
- The **Keepalives**, select ***RTP-RTCP*** for **Scope** and ***Enabled*** for **Initial keepalives** from pull down menus respectively.
- **Periodic timeout** is *30*.
- All other parameters should be set according to customer requirements**.**
- Click **OK**.



On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***. With this configuration, **STUN** will not be used.
- Set **Binding Refresh Time (seconds)** to ***60***. This value is used as one input to determine the frequency at which IP Office will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the IP address of IP Office WAN port. **Public Port TLS** is set to ***5061***.
- All other parameters should be set according to customer requirements.
- Click **OK**.

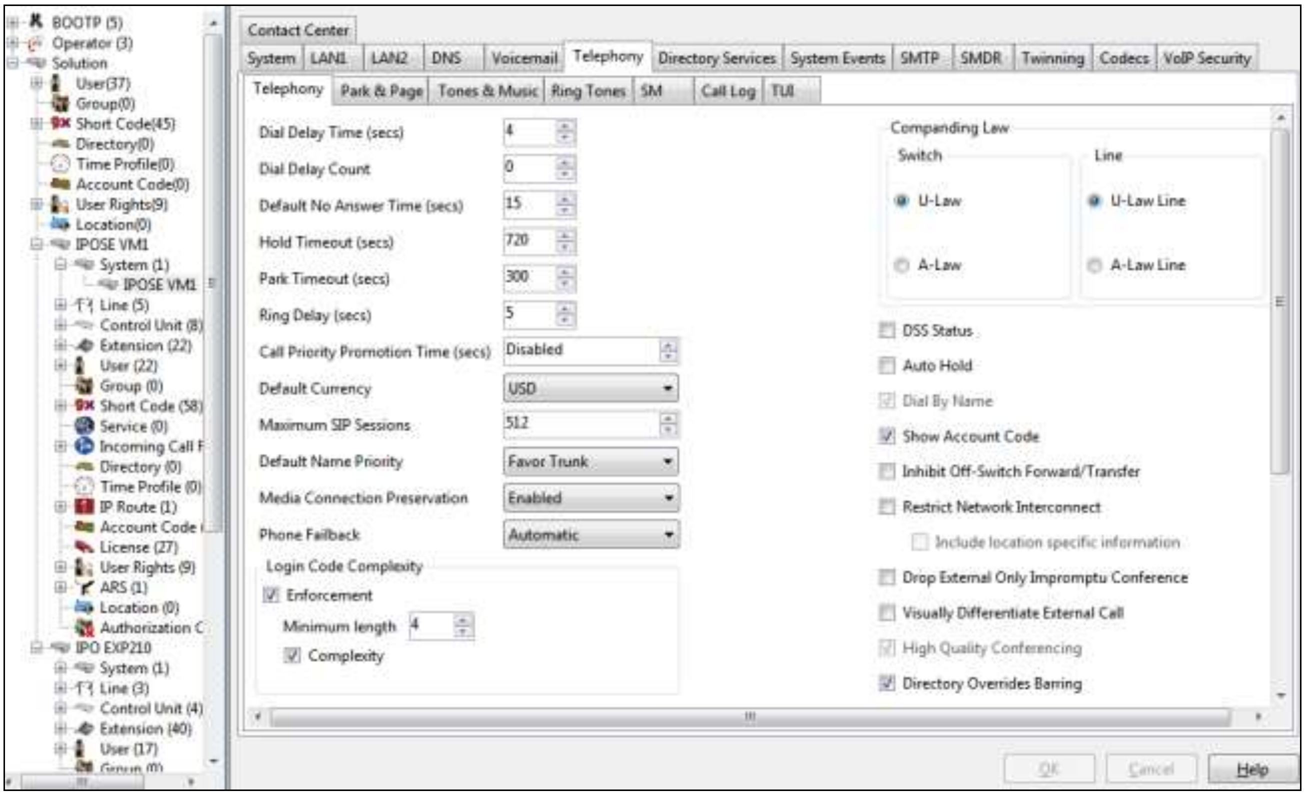


In the compliance test, the LAN1 interface was used to connect IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with Intermedia SIP Trunking service, and therefore is not described in these Application Notes.

5.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** Tab in the Details Pane.

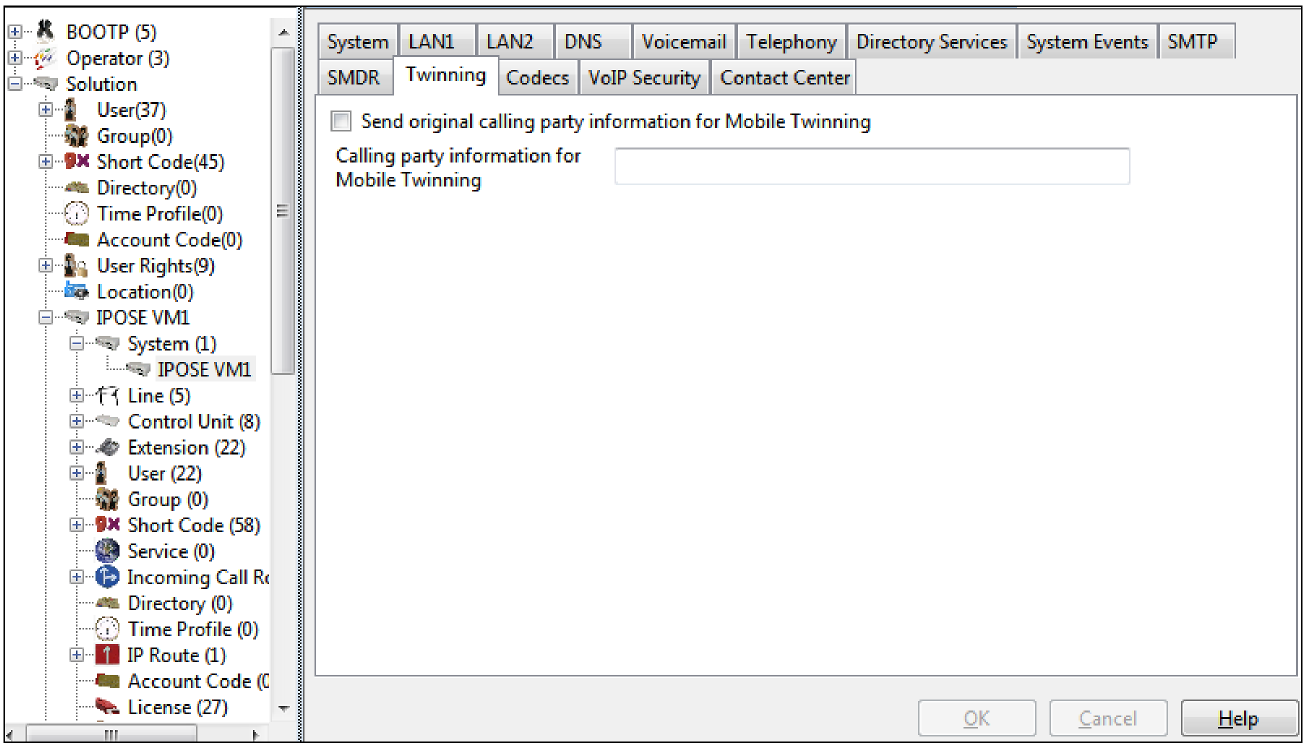
- Choose the **Companding Law** typical for the enterprise location. For North America, ***ULAW*** is used.
- Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk.
- Uncheck the **Drop External Only Impromptu Conference** box to allow the host of 3 way conference leaving the active call without forcing all the parties off the conference.
- Other parameters are left at default.
- Click **OK**.



5.3. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affects twinning and do not impact the messaging or operation of other redirected calls such as forwarded calls. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **System → Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.5**).

- For the compliance testing, the **Send original calling party information for Mobile Twinning** as shown below was unchecked. There is no requirement to use this parameter in this testing configuration.
- Click **OK**.



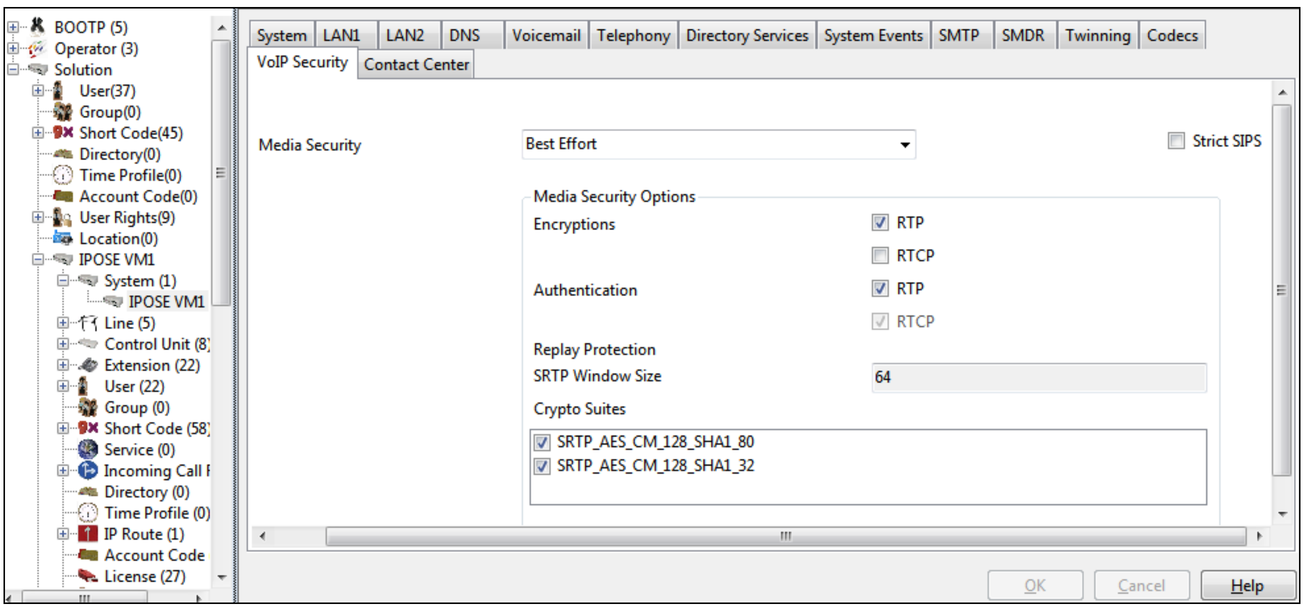
5.4. VoIP Security Settings

When enabling SRTP on the system, the recommended setting is Best Effort. In this scenario, IP Office uses SRTP if supported by the other end, and otherwise uses RTP. If the Enforced setting is used, and SRTP is not supported by the other end, the call is not established.

Individual SIP lines and extensions have media security settings that can override system level settings. This can be used for special cases where the trunk or extension setting must be different from the system settings.

In the compliance testing, Best Effort is set at system level and extensions. Enforce is set at trunk level to ensure the secured communication over the public internet using both signaling (TLS) and media (SRTP). Navigate to **System → VoIP Security** tab and configure as follow:

- Select ***Best Effort*** for **Media Security.** Media security is preferred. Attempt to use secure media first and if unsuccessful, fall back to non-secure media within Avaya IP Office system.
- Other parameters are left as default.
- Click **OK**.



5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between IP Office and Intermedia SIP Trunking service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.5.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses.
- SIP Credentials (if applicable).
- SIP URI entries.
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.5.2**.

Also, the following SIP Line settings are not supported on Basic Edition:

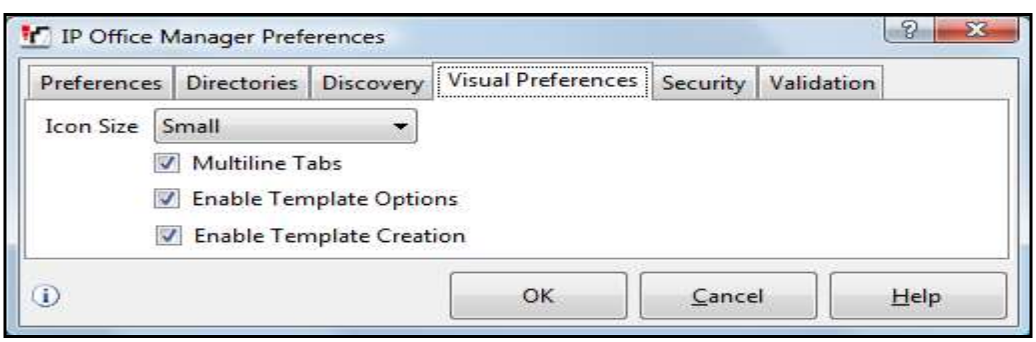
- SIP Line – Originator number for forwarded and twinning calls.
- Transport – Second Explicit DNS Server.
- SIP Credentials – Registration Required.

Alternatively, a SIP Line can be created manually. To do so right-click **Line** in the Navigation Pane and select **New → SIP Line**, then follow the steps outlined in **Sections 5.5.2**.

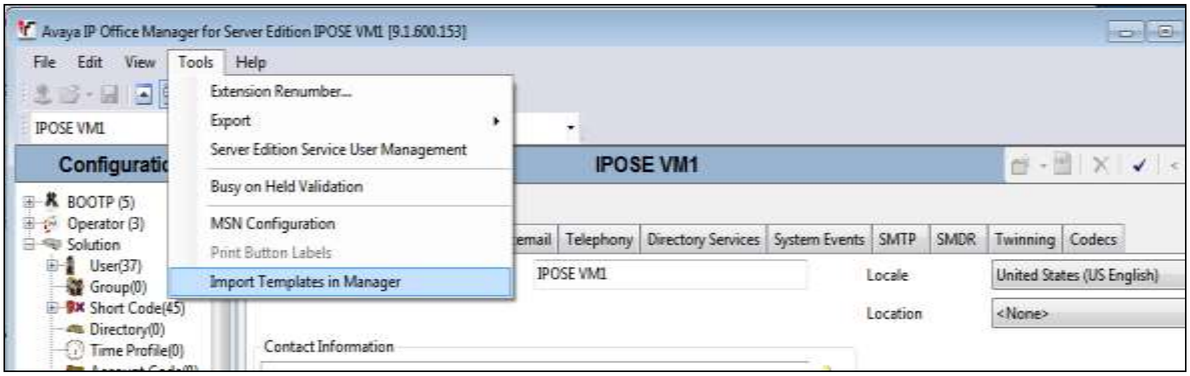
5.5.1. Create SIP line from Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **AF_Intermedia_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.

2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the Visual Preferences tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

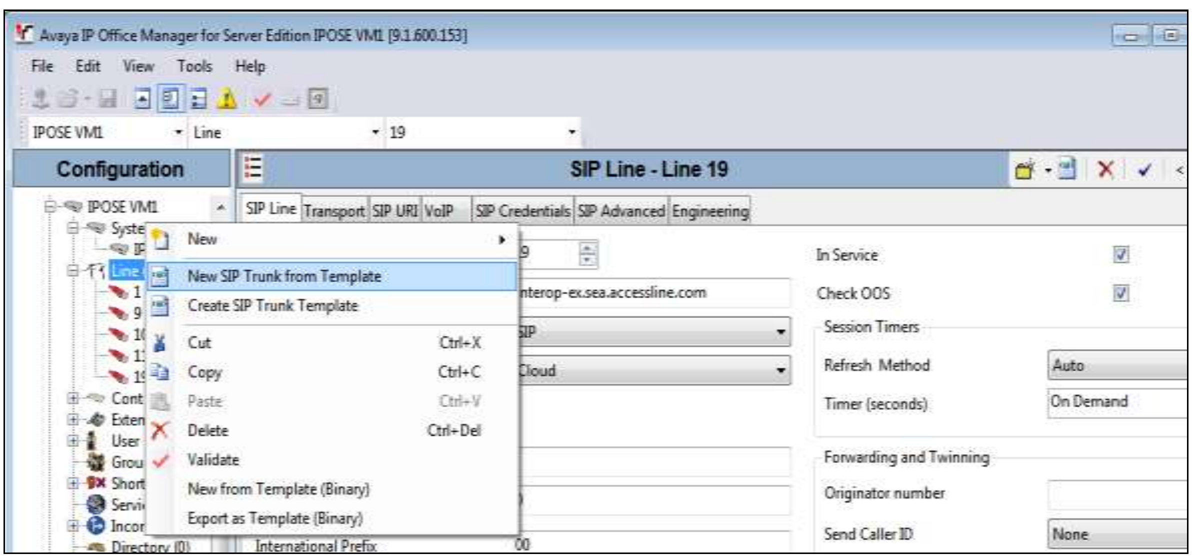


3. Import the template into IP Office Manager.
   From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.

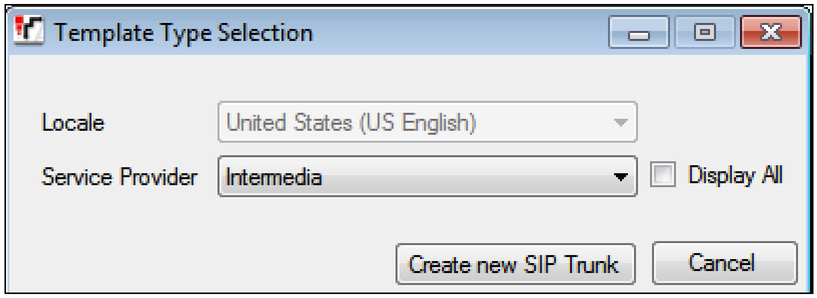


   In the pop-up window that appears (not shown), select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window will appear (not shown) stating success or failure. Then click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

4. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New SIP Trunk from Template**.



5. In the subsequent Template Type Selection pop-up window, select **Intermedia** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**AF_Intermedia_SIPTrunk.xml)** created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



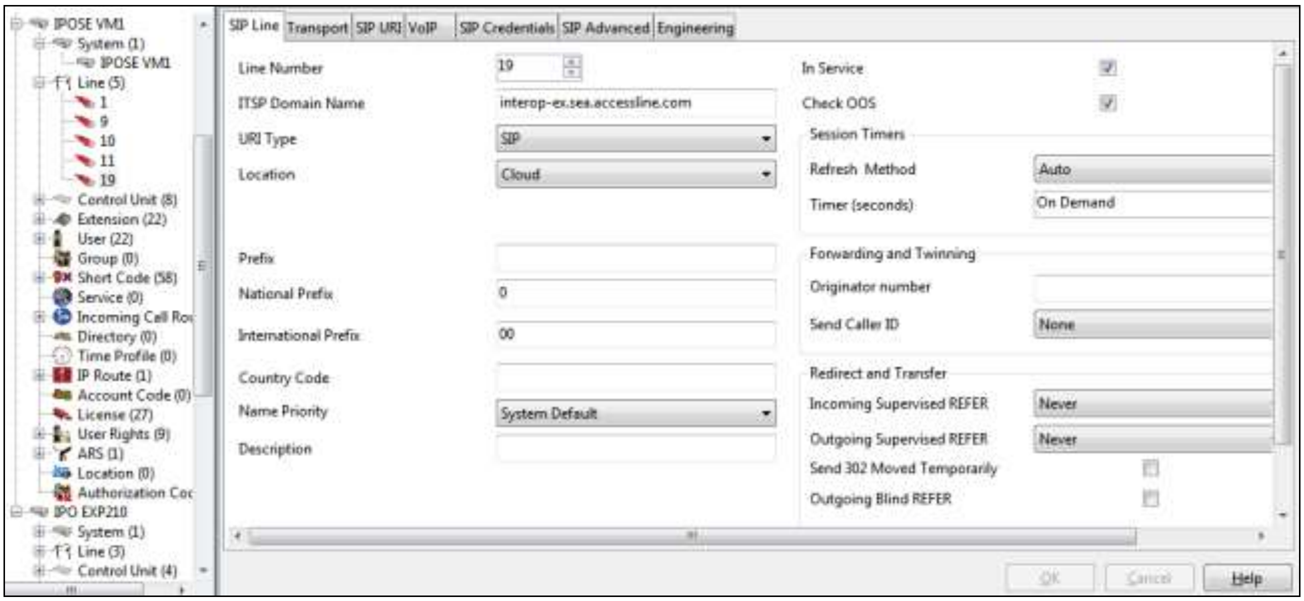
6. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.5.2.**

Note: Windows 7 (and later) locks the Avaya IP Office 9.1 **Templates** directory, and it cannot be viewed. To enable browsing of the **Templates** directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager\Templates** (or C:*Program Files (x86)*\Avaya\IP Office\Manager\Templates), and then click on the **Compatibility files** option shown below. The **Templates** directory and its contents can then be viewed.

5.5.2. Create SIP Line Manually

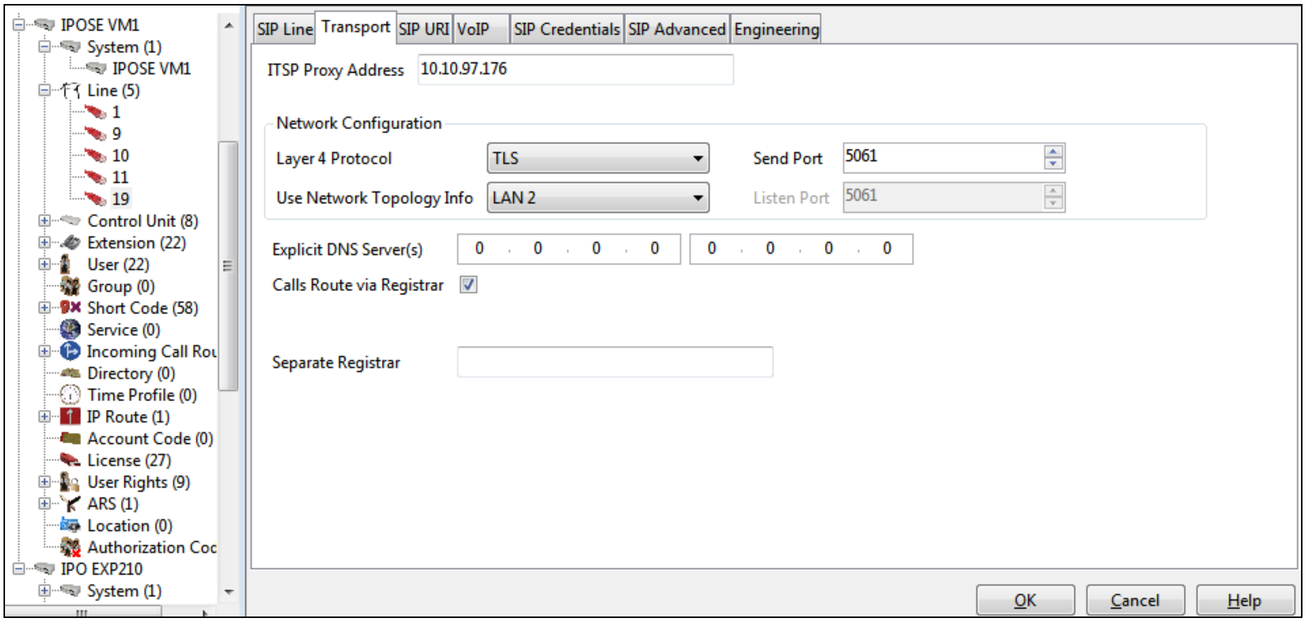
To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the enterprise domain so that IP Office uses this domain as the host portion of SIP URI in SIP headers such as the From header.
- Set **Send Caller ID** to ***None.***
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- **Incoming Supervised REFER** was set to ***Never*** as Intermedia does not support REFER.
- **Outgoing Supervised REFER** was set to ***Never*** as Intermedia does not support REFER.
- Other parameters are set as default values.
- Click **OK**.



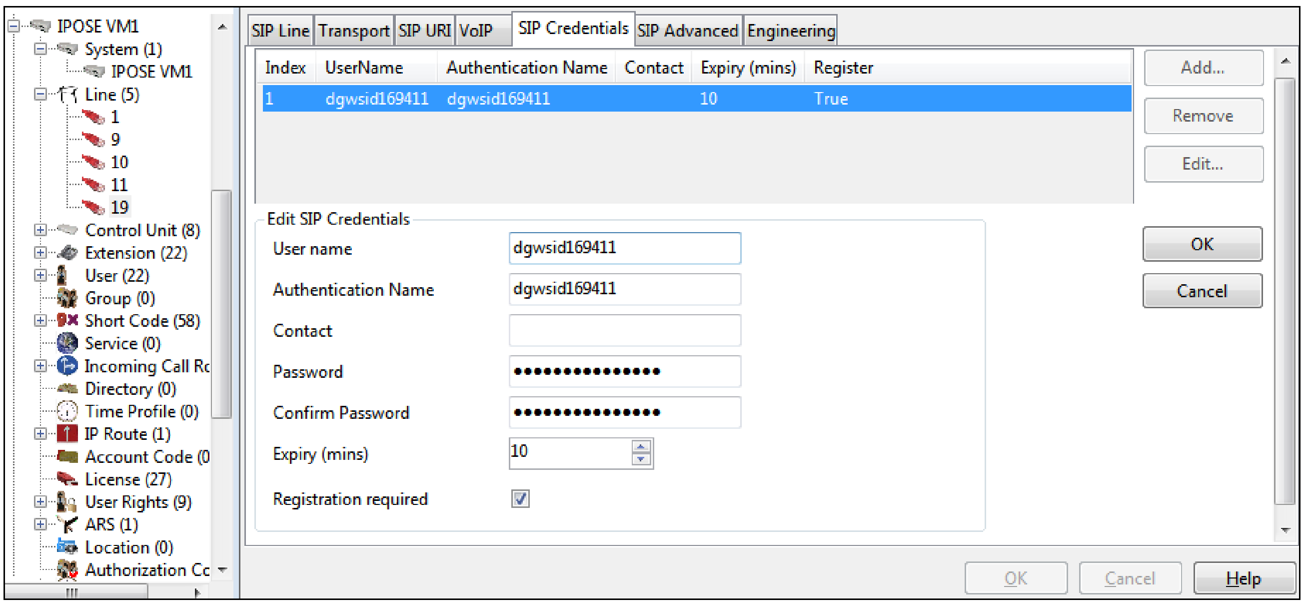
Select the **Transport** tab and enter the following information.

- The **ITSP Proxy Address** is set to the internal interface of Avaya SBCE.
- **Layer 4 Protocol** is set to ***TLS***.
- **Send Port** is set to the port number of IP Office, ***5061***.
- **Use Network Topology Info** parameter is set to ***LAN 2***. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab.
- Other parameters retain default values in the screen below.
- Click **OK**.



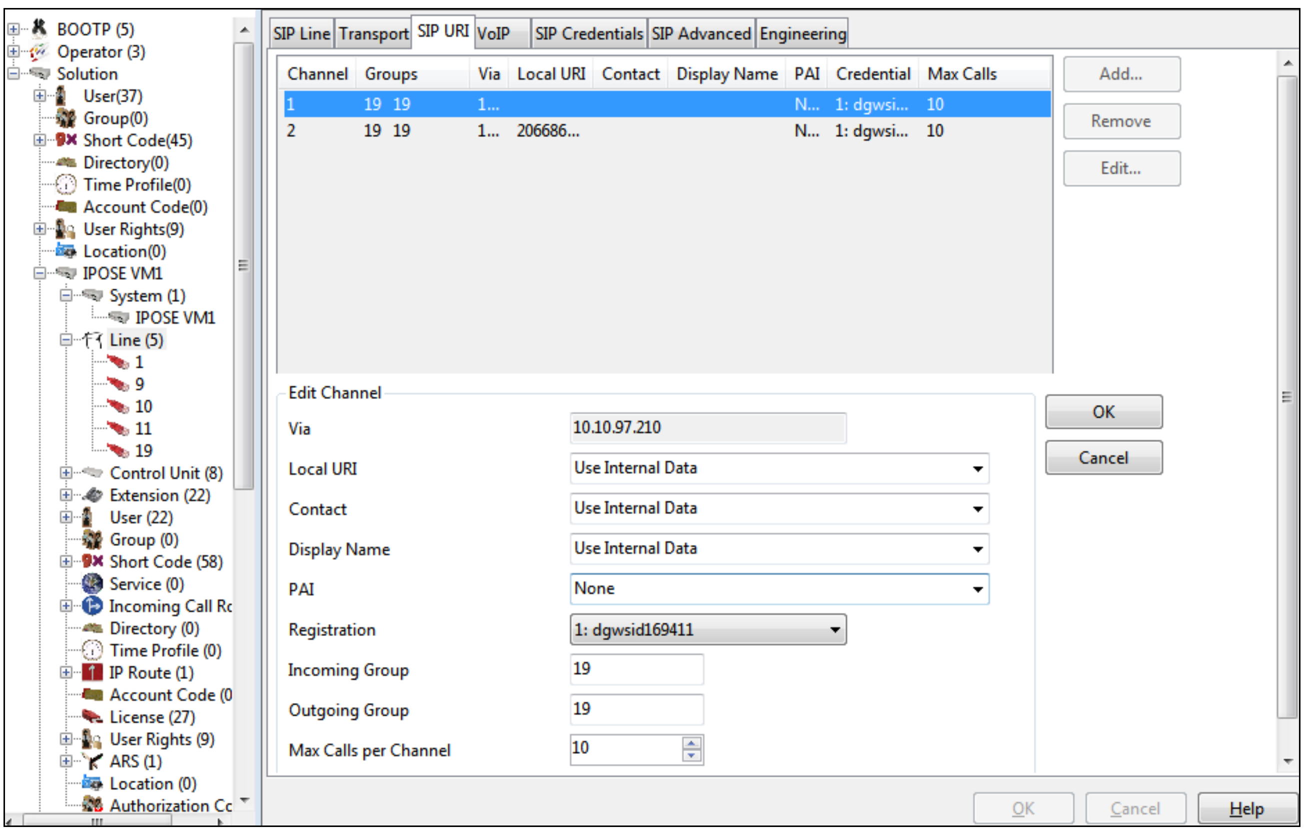
A SIP Credentials entry must be created for SIP trunking registration and Digest Authentication that are used by Intermedia SIP trunking service to authenticate calls from the enterprise to the PSTN. To create a SIP Credentials entry, first select the **SIP Credentials** tab. Click the **Add** button 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 bottom of the screen, the Edit Channel area will be opened. In the example screen below, a previously configured entry is edited. The entry was created with the parameters shown below:

- Set **User name** and **Authentication Name** to the value provided by the service provider.
- Set **Password** and **Confirmed Password** to the value provided by the service provider.
- The **Expiry (mins)** is set to ***10*** minutes in this case as service provider relies on registration from enterprise to keep the trunk alive.
- Check the **Registration required** option. Service provider requires registration for Digest Authentication.
- Click **OK.**



A SIP URI entry **Channel 1** is created to match incoming numbers that IP Office will accept on this line. Select the **SIP URI** tab, then click the **Add** button and then **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 is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an IP Office user. The entry was created with the parameters shown below:

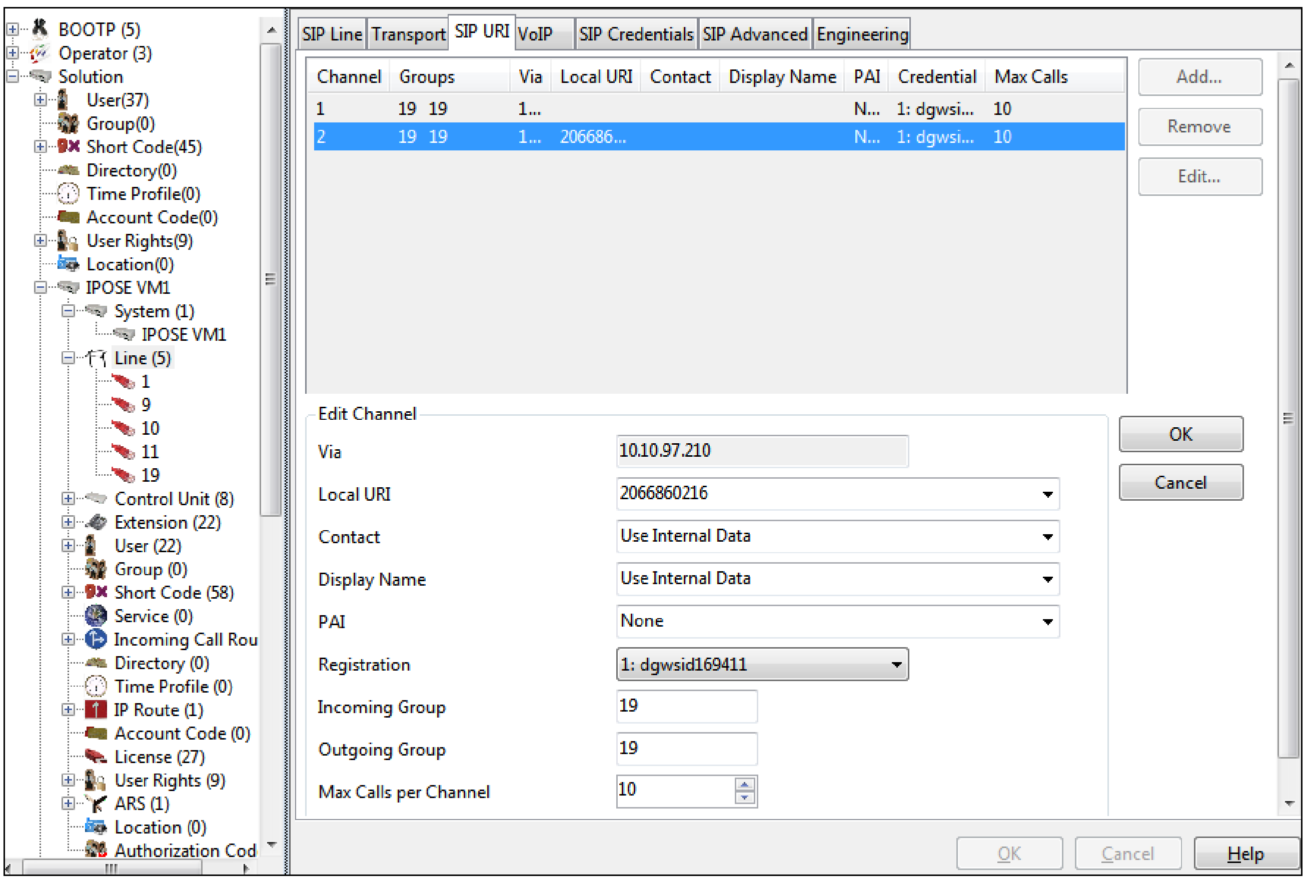
- **Via** field is pre-populated by IP Office.
- Set **Local URI**, **Contact** and **Display Name** to ***Use Internal Data***. This setting allows calls on this line which SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.7**.
- **PAI** field is set to ***None***.
- For **Registration**, select the account credentials previously configured on the line's **SIP Credentials** tab.
- Associate this line with an incoming line group 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. For the compliance test, a new incoming and outgoing group ***19*** was defined that only contains this line (line 19).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Other parameters retain default values and or set according customer requirements.
- Click **OK**.



SIP URI entry **Channel 2** was similarly created for incoming calls appropriately to pre-defined DID numbers, which is provided by service provider, to access to Feature Name Extension 00 (FNE00). The Short Codes for FNE00 are defined in **Section 5.6** to provide Dial Tone and Mobile Callback for mobility extension.

The **Channel 2,** as shown in the screenshot below, was configured with following parameters.

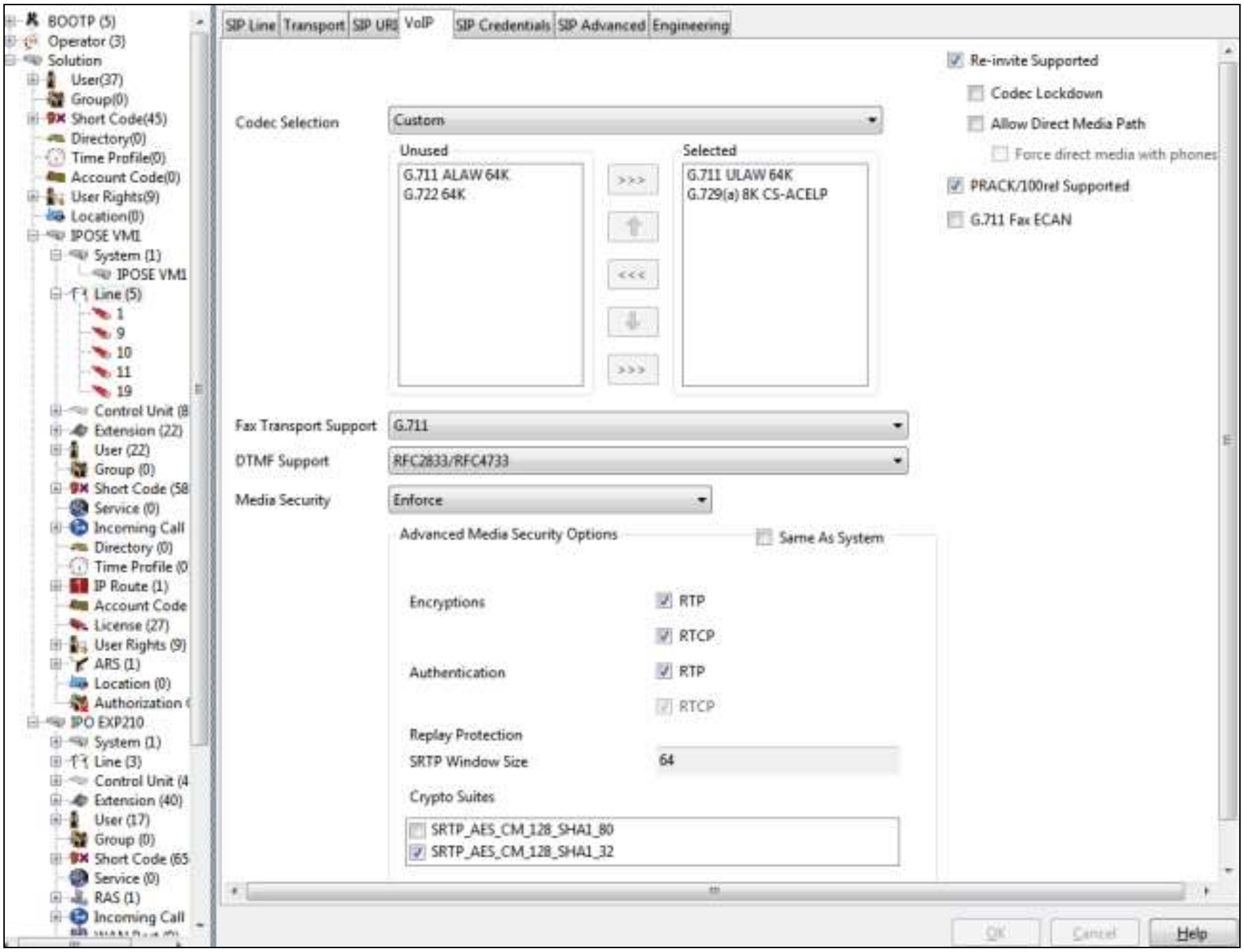
- **Via** field is pre-populated by IP Office.
- Set the **Local URI** to pre-define DID number appropriately for **Channel 2**.
- Set **Contact** and **Display Name** to ***Use Internal Data***.
- **PAI** field is set to ***None***.
- For **Registration**, select the account credentials previously configured on the lines **SIP Credentials** tab.
- Associate **Incoming Group** and **Outgoing Group** to SIP Line ***19***.
- Set the **Max Calls per Channel** field to ***10***.
- Other parameters retain default values and or set according customer requirements.
- Click **OK**.



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

- The **Codec Selection** can be selected by choosing ***Custom*** from the pull-down menu, allowing an explicit ordered list of codecs to be specified.
- Selecting ***G.711 ULAW*** and ***G.729*** codec supported by the Intermedia SIP Trunking service, in the Session Description Protocol (SDP) offer.
- Set **Fax Transport Support** to ***G.711*** from the pull-down menu (T.38 faxing is not currently supported by service provider).
- Set the **DTMF Support** field to ***RFC2833/RFT4733*** from the pull-down menu. This directs IP Office to send DTMF tones using SRTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Check the **PRACK/100rel Supported** box.
- Set **Media Security** to ***Enforce*** to ensure the use of SRTP for media communicating over the SIP trunk.
- The **Advanced Media Security Options**, uncheck the **Same As System** box to disassociate the media security settings of the SIP trunk from the media security settings of the system. Note that what is set here will over write the system and extensions settings in term of media security.
- The **Encryption**, check the **RTCP** box.
- The Crypto Suits, uncheck the **SRTP_AES_CM_128_SHA1_80** box as service provider using only **SRTP_AES_CM_128_SHA1_32**.
- Default values may be used for all other parameters.
- Click **OK**.

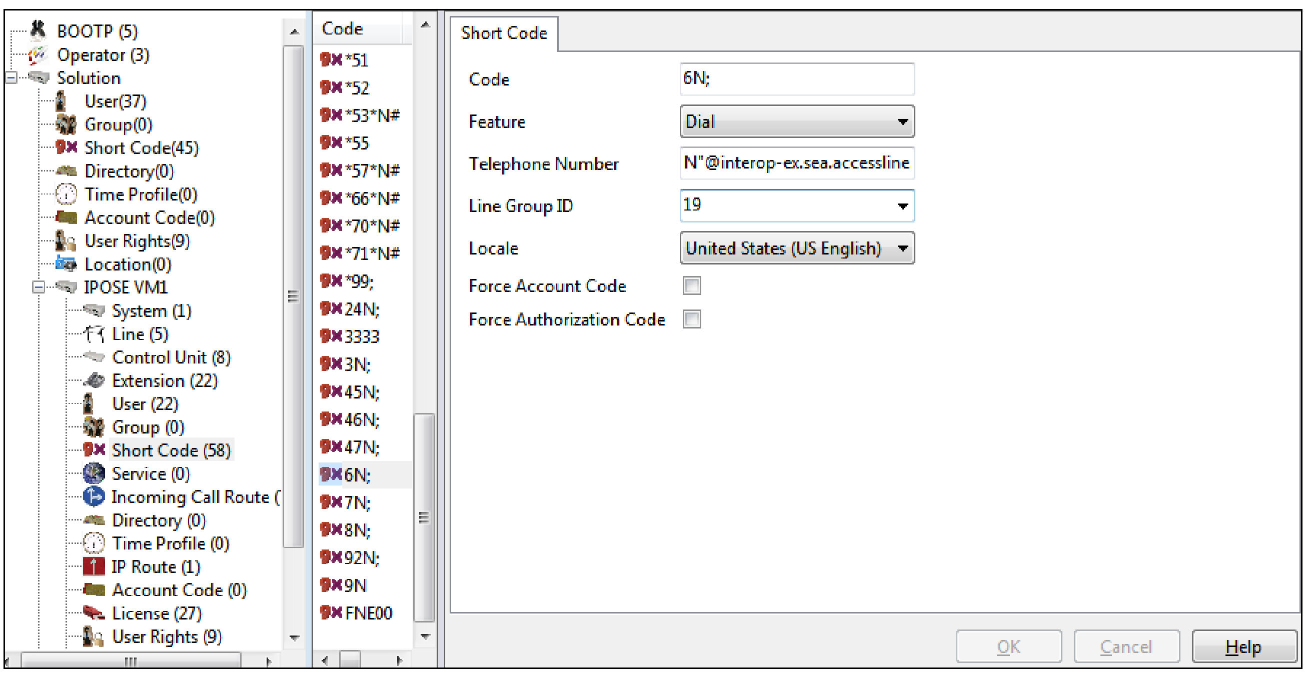
The **VoIP** tab capture is shown below.



5.6. Short Code

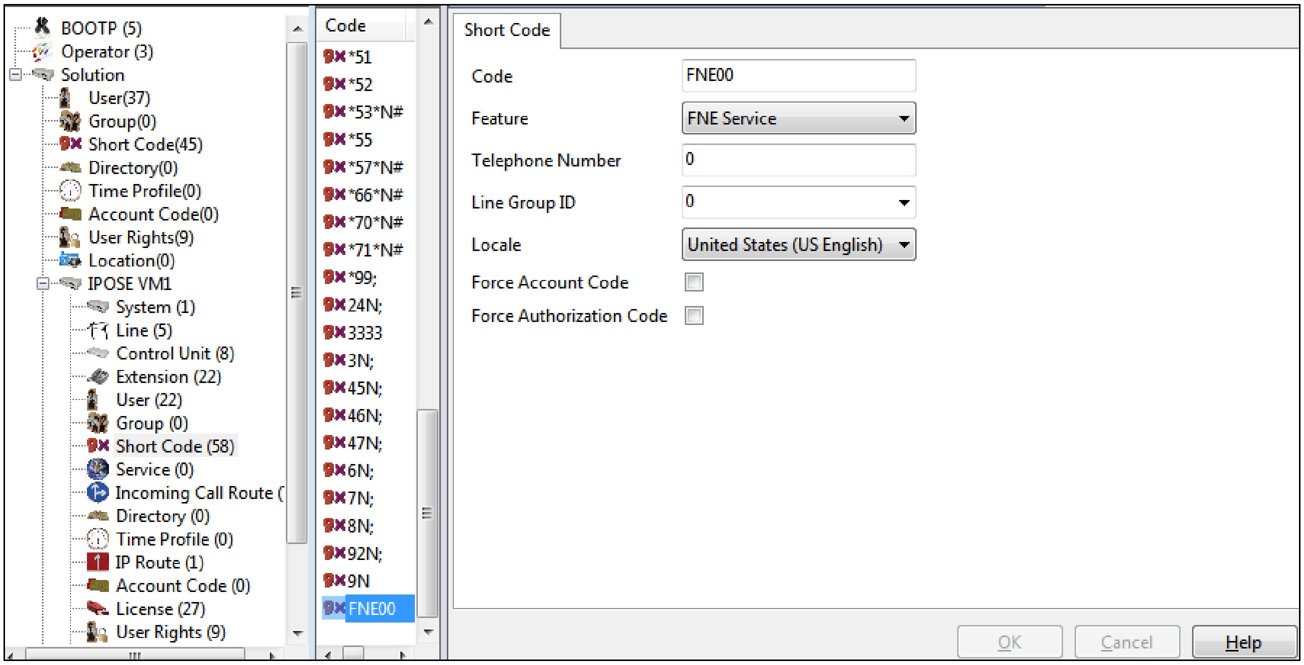
Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered "6N;" short code used in the test configuration.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, ***6N,*** this short code will be invoked when the user dials 6 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to the value shown in the capture bellow. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value ***N*** represents the number dialed by the user. The host part following the "@" is the domain of the service provider network.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.5**. This short code will use this line group when placing the outbound call.
- Select ***United State (US English)*** for **Locale**.
- Others parameters are at default values.
- Click **OK**.



For incoming calls from mobility extension to FNE features hosted by IP Office to provide **Dial Tone** functionality, Short Code **FNE00** was created. The FNE00 was configured with the following parameters.

- In the **Code** field, enter the FNE feature code as ***FNE00*** for **Dial Tone**.
- Set the **Feature** field to ***FNE Service***.
- Set the **Telephone Number** field to ***0***.
- Set the **Line Group ID** field to ***0***.
- Select ***United State (US English)*** for **Locale**.
- Retain default values for other fields.
- Click **OK**.

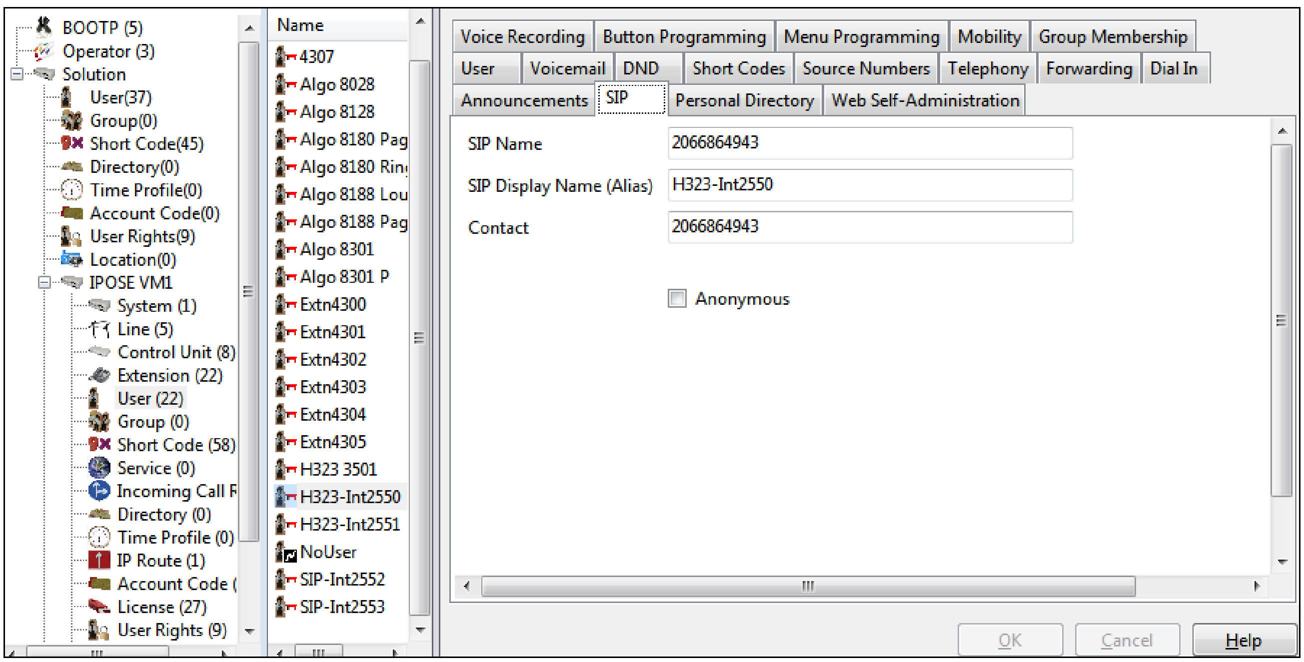


5.7. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.5**. To configure these settings, first select **User** in the left Navigation Pane, 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 "H323-Int2550". Select the **SIP** tab in the Details Pane.

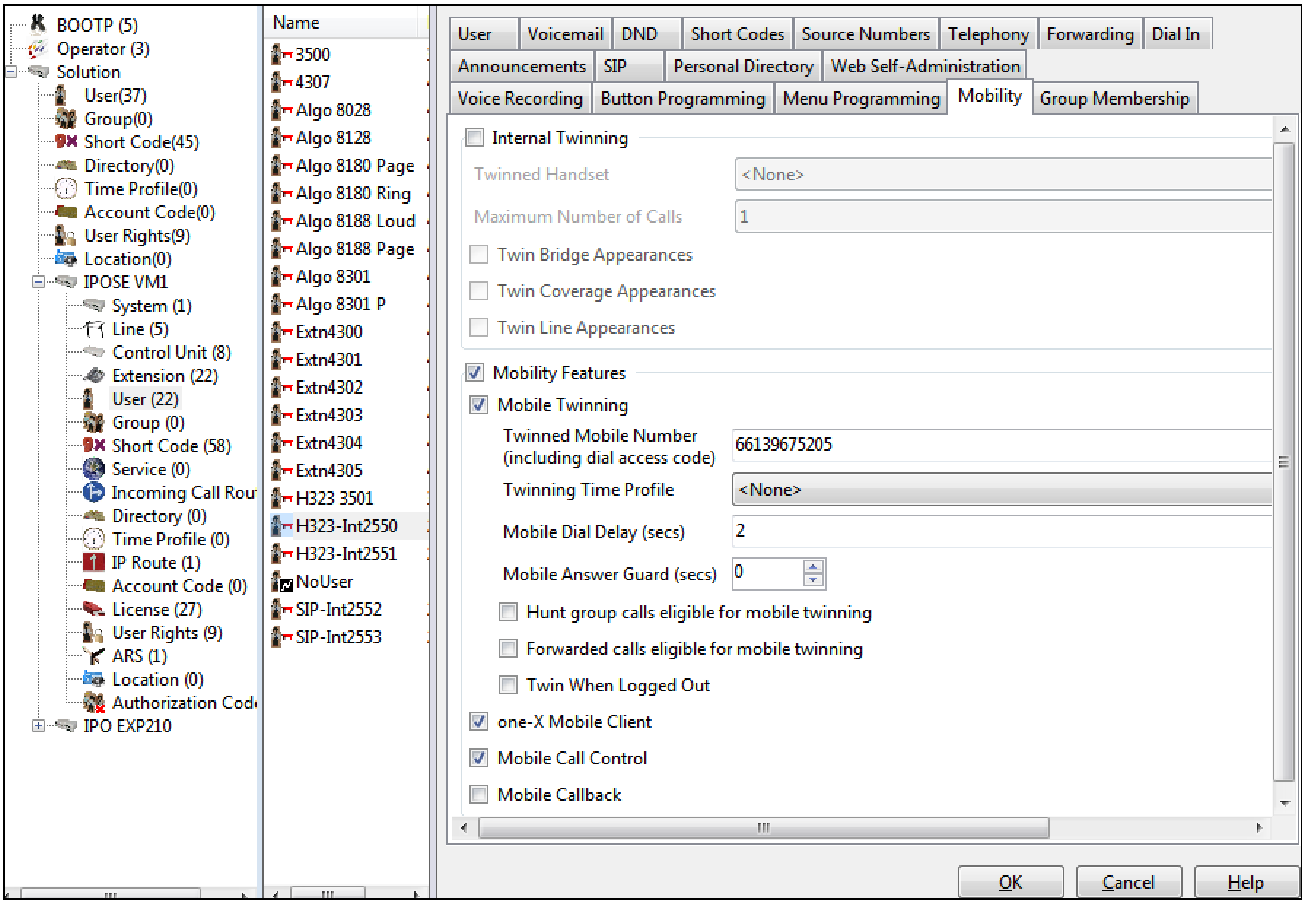
The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.5**). The example below shows the settings for user H323- Int2550.

- The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from service provider.
- The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.
- If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network.
- Click **OK**.



One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for User H323- Int2550.

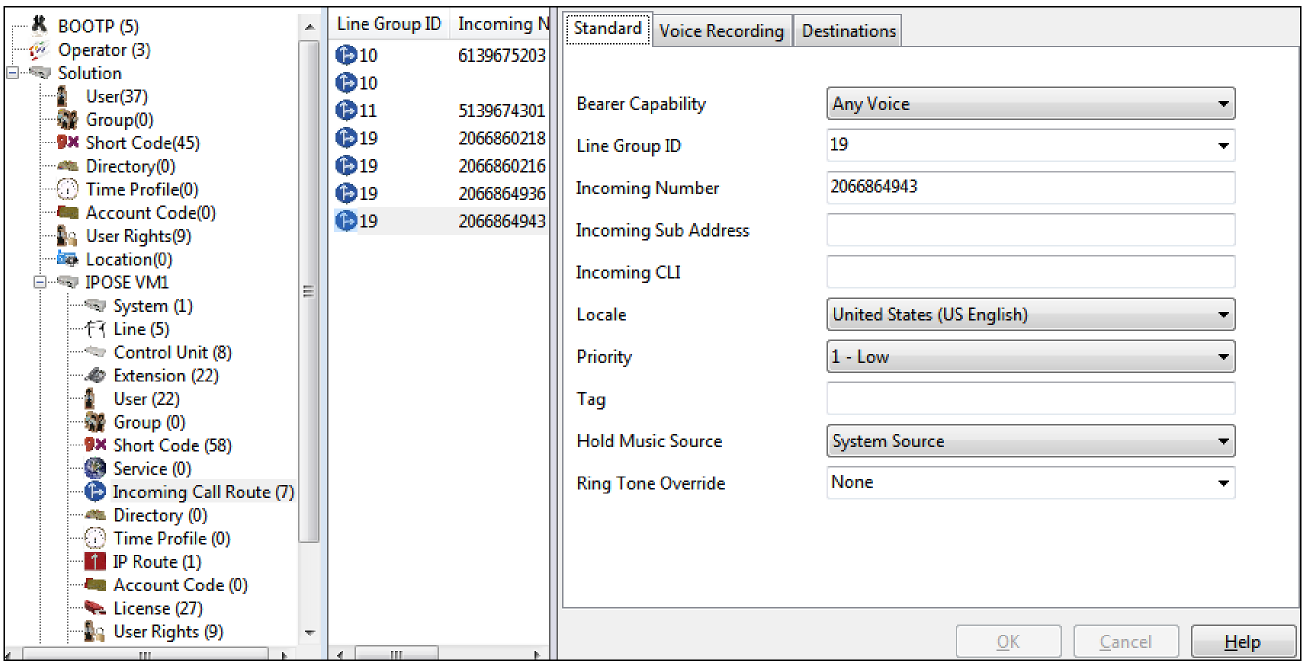
- The **Mobility Features** and **Mobile Twinning** boxes are checked.
- The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case ***66139675205***.
- Make sure the **Mobile Call Control** box is checked.
- Other options can be set according to customer requirements.
- Click **OK**.



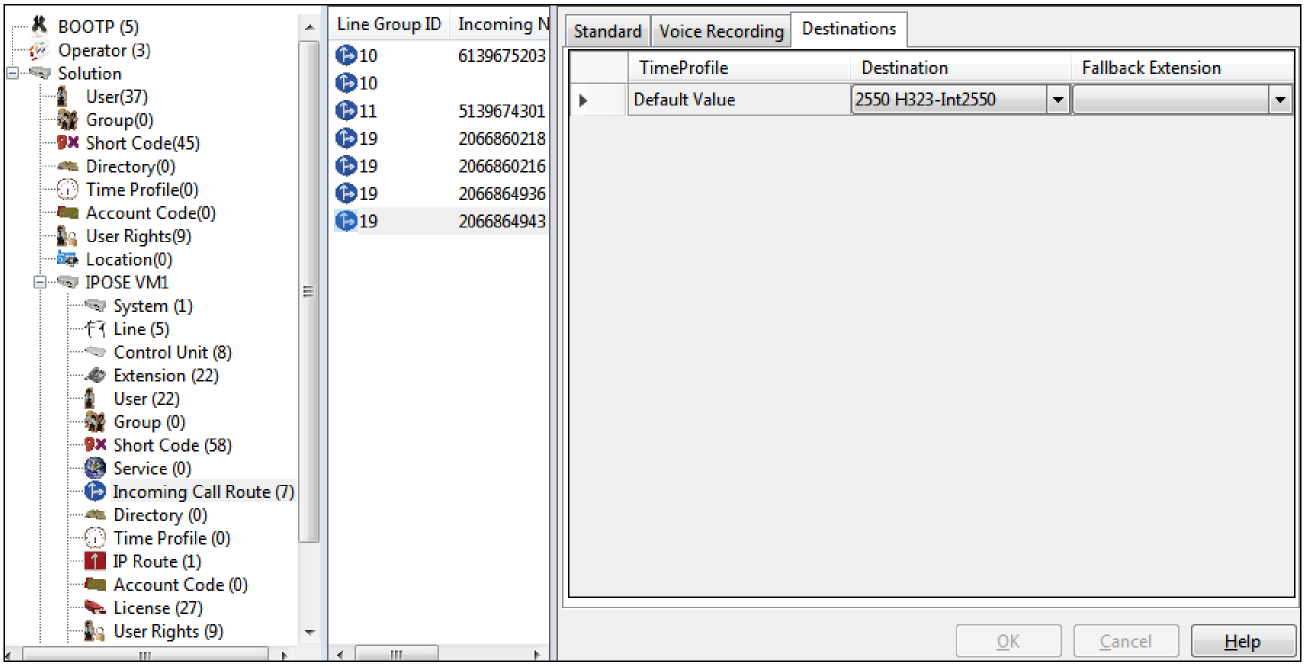
5.8. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capability** to ***Any Voice***.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.5**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Select ***United States (US English)*** for **Locale**.
- Default values can be used for all other fields.
- Click **OK**.



On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **2066864943** on line 19 are routed to extension **4000**. Click **OK**.



5.9. Save Configuration

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

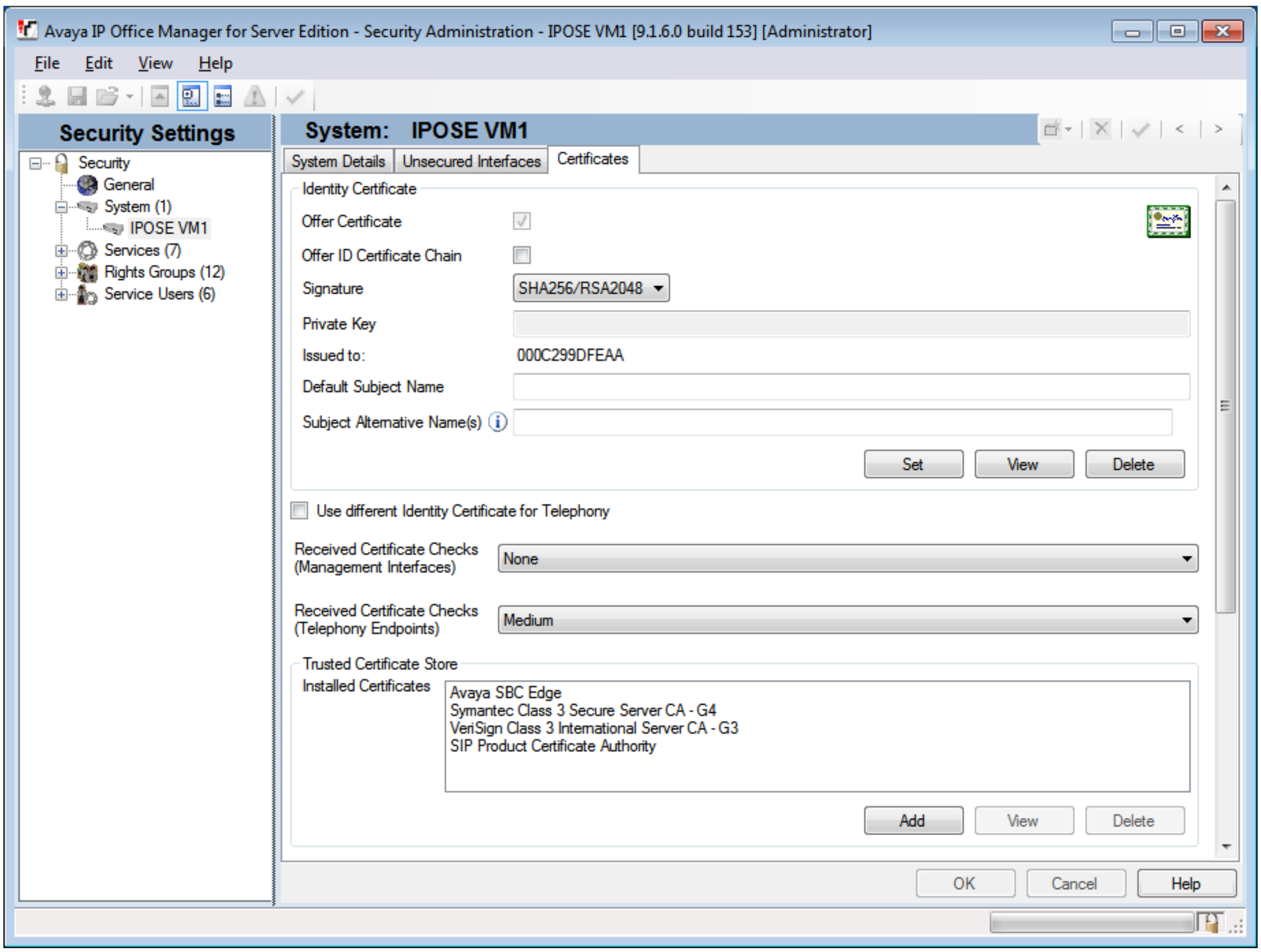
5.10. Security Settings

This setting is used with IP telephony endpoints connecting to the system. This setting is used by IP Office to validate the identity certificate offered by the other end of TLS connection. IP Office does not support mutual authentication for SIP terminals (an identity certificate is not installed in all SIP terminals). Therefore, IP Office does not require a client certificate from a SIP terminal, only SIP and SM trunks. The received certificate is tested as follows:

- **None:** No extra checks are made (The certificate must be in date).
- **Low:** Certificate minimum key size 1024 bits, in date.
- **Medium:** Certificate minimum key size 1024 bits, in date, match to store.
- **High:** Certificate minimum key size 2048 bits, in date, match to store, no self signed, no reflected, chain validation.

Navigate to **File →Advanced →Security →System**, choose **Certificates** tab.

- Select ***Medium*** for the **Received Certificate Checks (Telephony Endpoints)**.
- Click **Ok**.
- Click **File → Save Security Settings**.



6. Configure t Avaya Session Border Controller for Enterprise

This section covers the configuration of Avaya SBCE. It is assumed that the software has already been installed. For additional information on these configuration tasks, see **Section 10**.

The compliance testing comprised the configuration for two major components, 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 in the Avaya SBCE web user interface as described in the following sections.

Trunk Server configuration elements for the service provider - Intermedia:

- 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:

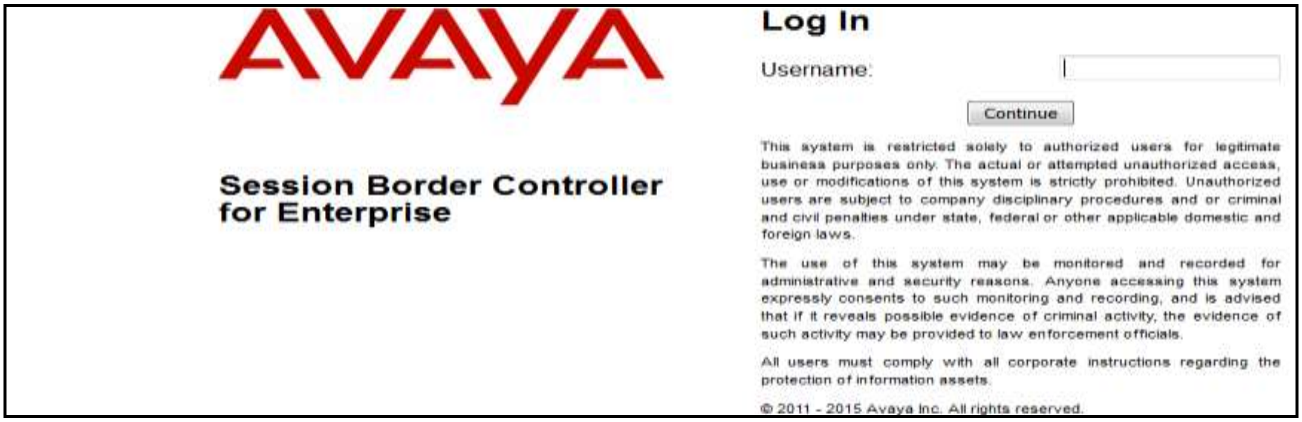
- Global Profiles:
  - URI Groups
  - Routing
  - Topology Hiding
  - Server Interworking
  - Server Configuration
- Domain Policies:
  - Application Rules
  - Media Rules
  - Signaling Rules
  - Endpoint Policy Group
  - Session Policy
- Device Specific Settings:
  - Network Management

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

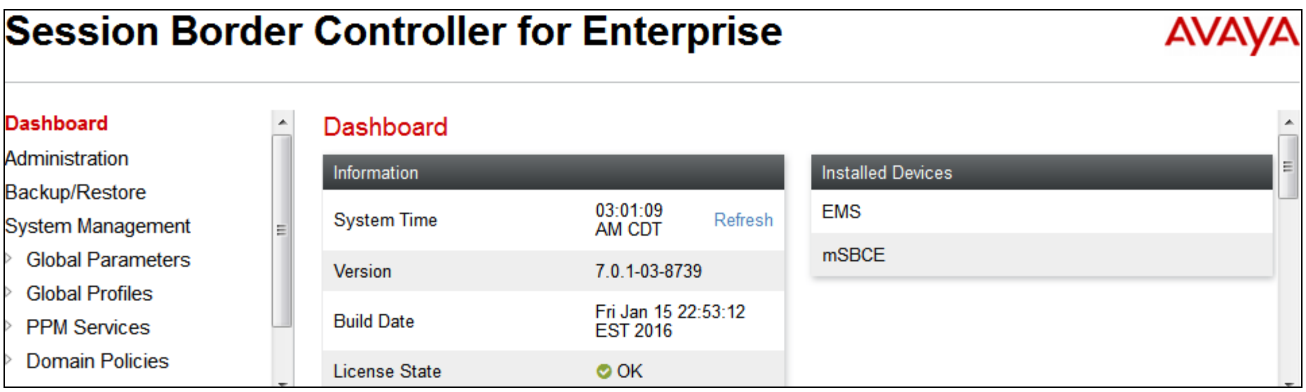
6.1. Log into Avaya Session Border Controller for Enterprise

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.



6.2. Global Profiles

Global Profiles allows for configuration of parameters across all Avaya SBCE appliances.

The naming convention in this entire section is using as follow:

- **SP** is stand for Service Provider, which is Intermedia in this case.
- **EN** is stand for Enterprise Network, which is referred to Avaya IP Office.

6.2.1. Uniform Resource Identifier (URI) Groups

URI Group feature allows user to create any number of logical URI groups that are comprised of individual SIP subscribers located in that particular domain or group. These groups are used by the various domain policies to determine which actions (Allow, Block, or Apply Policy) should be used for a given call flow.

For this configuration testing, "*****" is used for all incoming and outgoing traffic.

6.2.2. Server Interworking Profile

Interworking Profile features are configured differently for Call Server and Trunk Server.

To create a Server Interworking profile, select **Global Profiles → Server Interworking**. Click on the **Add** button.

In the compliance testing, two Server Interworking profiles were created for SP and EN respectively.

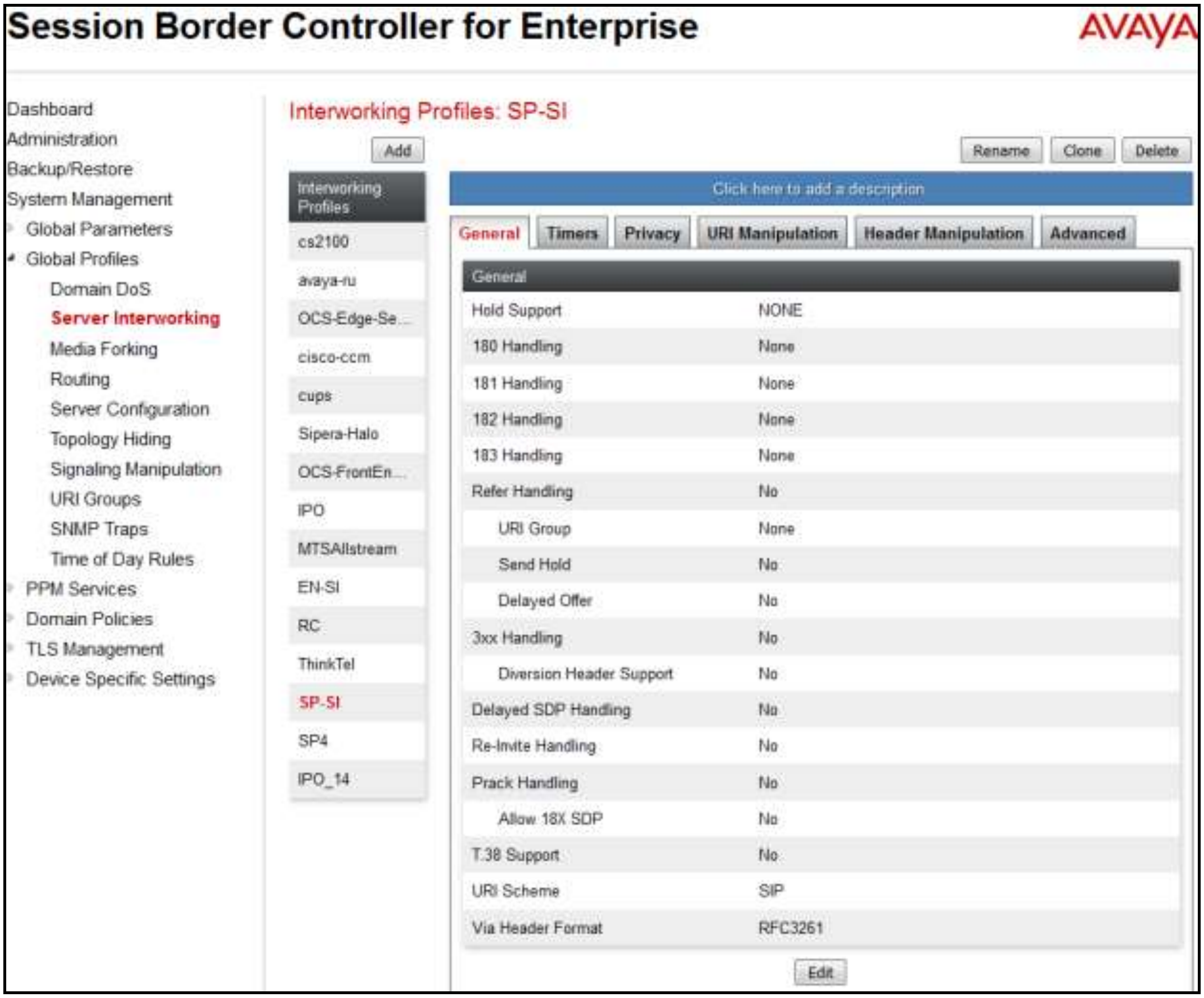
6.2.2.1 Server Interworking profile for SP

Profile **SP-SI** was defined to match the specification of SP. The **General, Header Manipulations** and **Advanced** tabs are configured with the following parameters while the other tabs for **Timers, Privacy** and **URI Manipulation** are kept as default.

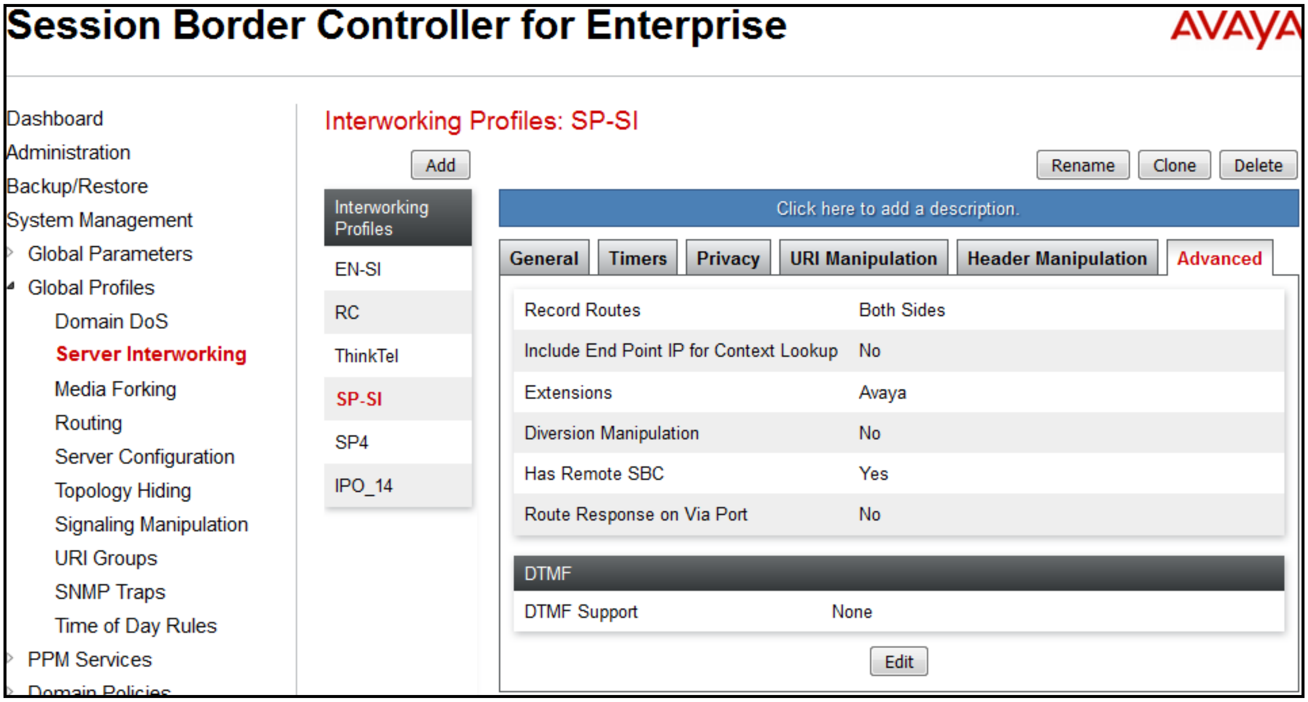
General tab:

- **Hold Support** = ***NONE***. Avaya SBCE will not modify the hold/ resume signaling from EN to SP.
- **18X Handling** = ***None***. Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from EN to SP.
- **Refer Handling** = ***No.*** Avaya SBCE will not handle REFER. It will keep the REFER message unchanged from EN to SP.
- **T.38 Support** = ***No***. SP does not support T.38 fax in the compliance testing.
- Others are left as default values.

The screenshots below illustrate the Server Interworking profile **SP-SI, General.**



The screenshots below illustrate the Server Interworking profile **SP-SI**, **Advanced**.



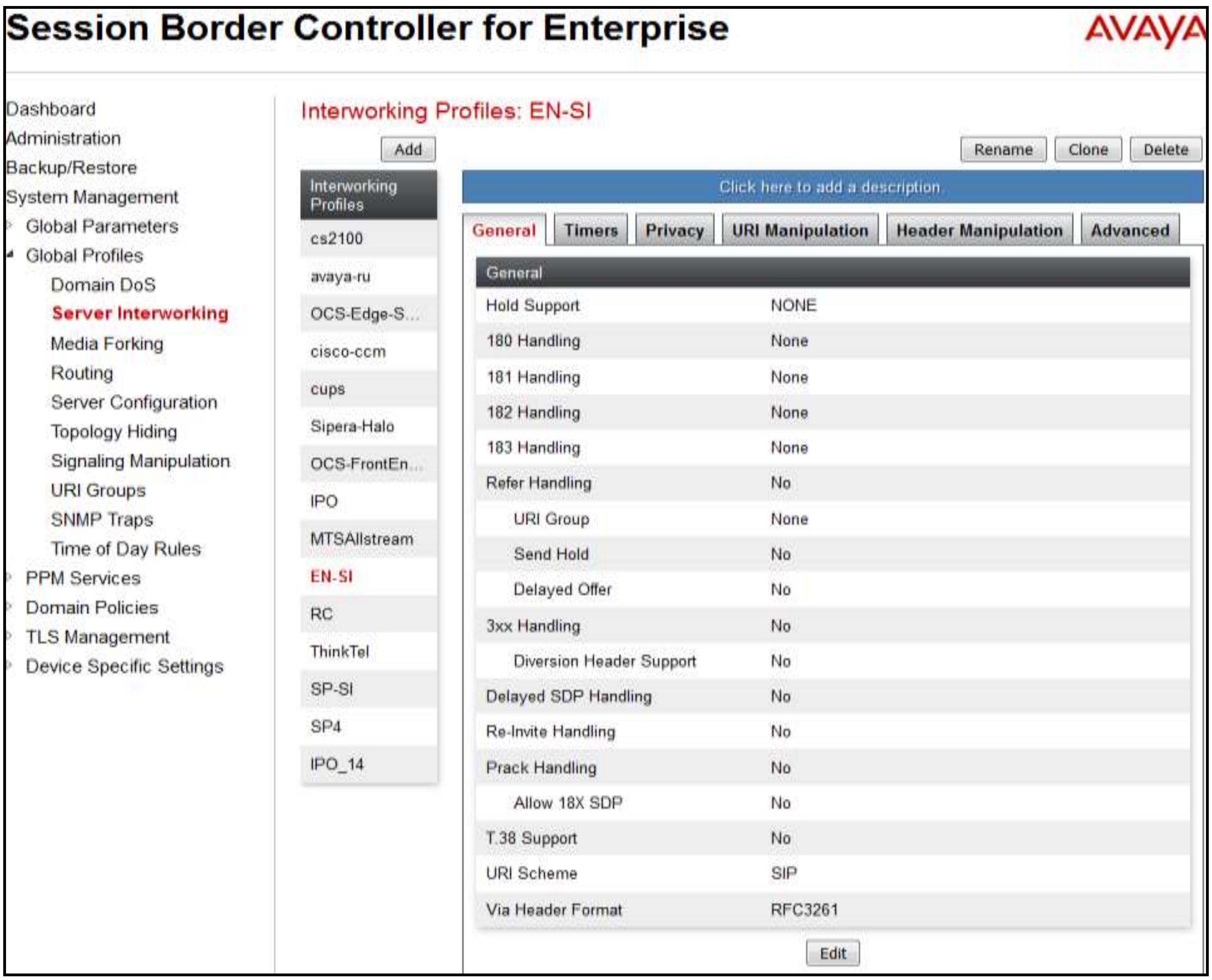
6.2.2.2 Server Interworking profile for EN

Profile **EN-SI** was defined to match the specification of EN. The **General** and **Advanced** tabs are configured with the following parameters while the other settings for **Timers**, **Privacy, URI Manipulation** and **Header Manipulation** are kept as default.

General tab:

- **Hold Support** = ***NONE***.
- **18X Handling** = ***None***. Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from SP to EN.
- **Refer Handling** = ***No***. Avaya SBCE will not handle REFER, it will keep the REFER messages unchanged from SP to EN.
- **T.38 Support** = ***No***. EN does support T.38 fax, but SP doesn't in the compliance testing.
- Others are left as default values.

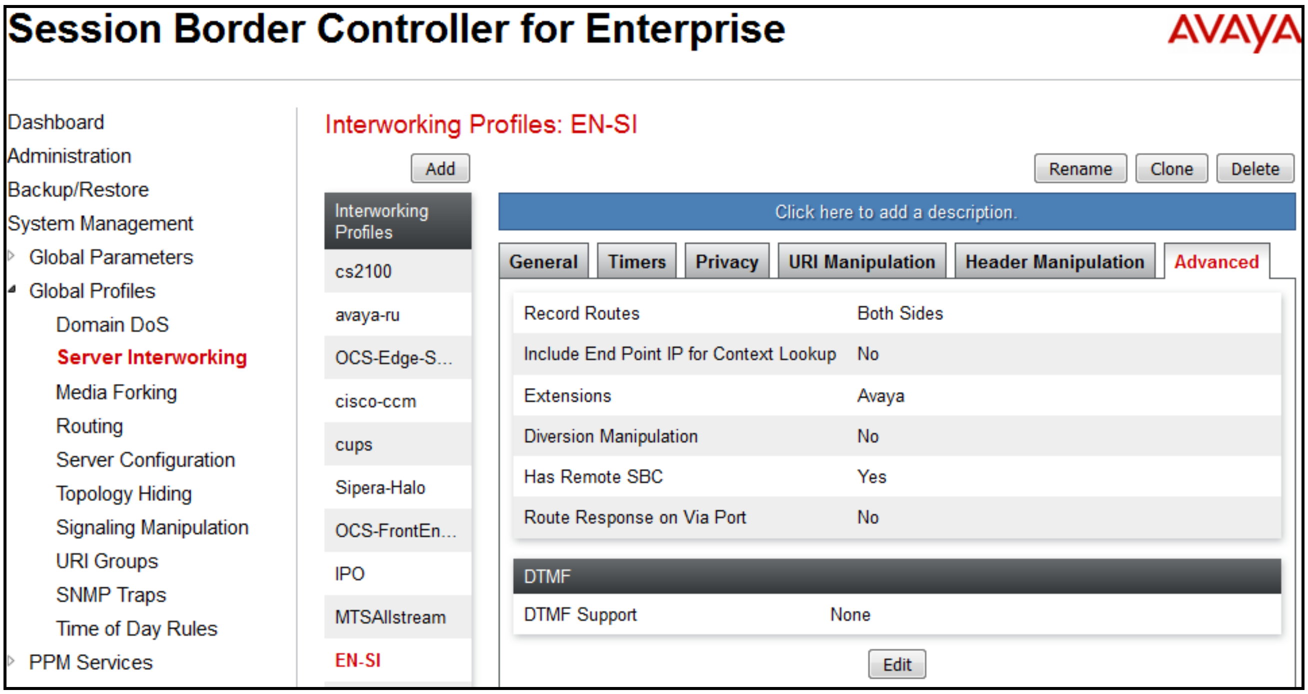
The screenshots below illustrate the Server Interworking profile **EN-SI**, **General**.



Advanced tab:

- **Record Routes** = ***Both Sides***. Avaya SBCE will send Record-Route header to both call and trunk servers.
- **Include End Point IP for Context Lookup** = ***No***.
- **Extensions** = ***Avaya***.
- **Has Remote SBC** = ***Yes***. This setting allows Avaya SBCE to always use the SDP received from EN for the media.
- **DTMF Support** = ***None***. Avaya SBCE will send original DTMF method from SP to EN.
- Others are left as default values.

The screenshots below illustrate the Server Interworking profile **EN-SI**, **Advanced.**



6.2.3. Server Configuration

Server Configuration screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. These tabs are used to configure and manage various SIP Call Server specific parameters such as TCP, UDP or TLS port assignments, heartbeat signaling parameters, DoS security statistics and trusted domains.

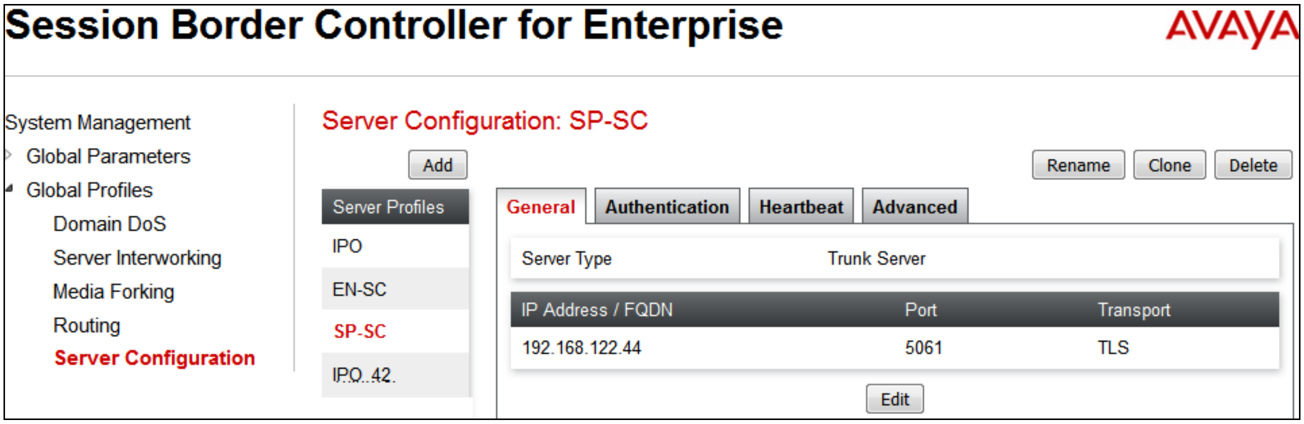
To create a Server Configuration entry, select **Global Profiles →Server Configuration**. Click on the **Add** button.
In the compliance testing, two separate Server Configurations were created, server entry **SP-SC** for SP and server entry **EN-SC** for EN.

6.2.3.1 Server Configuration for SP

Server Configuration named **SP-SC** was created for SP. It will be discussed in detail below. **General** and **Advanced** tabs are provisioned for SP on the SIP trunk for every outbound call from enterprise to PSTN. The **Authentication** is disabled to allow IP Office to send out authentication to SP. **Heartbeat** tab is kept as ***disabled*** as default to allow Avaya SBCE to forward the OPTIONS heartbeat from EN to SP to query the status of the SIP trunk.

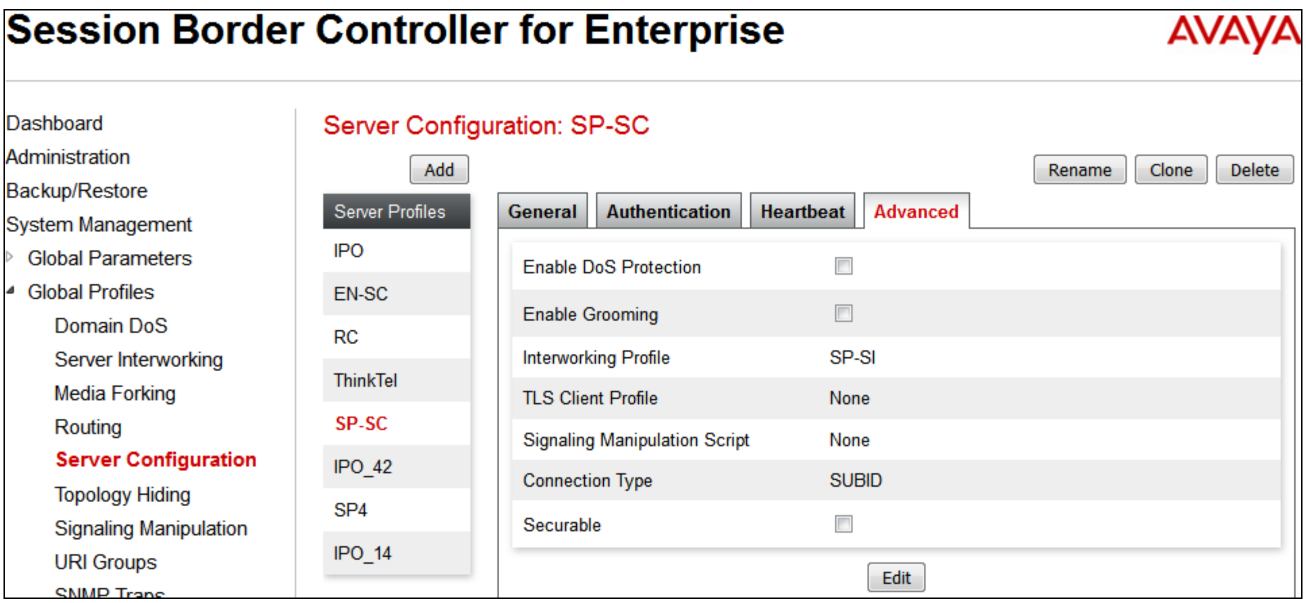
General tab:
Click on the **Edit** button and enter following information.

- Set **Server Type** for SP as ***Trunk Server***.
- In the compliance testing, SP supported ***TLS*** and listened on port ***5061***.



Advanced tab:
Click on the **Edit** button and enter following information.

- **Interworking Profile** drop down list, select ***SP-SI*** as defined in **Section 6.2.2**.
- The other settings are kept as default.

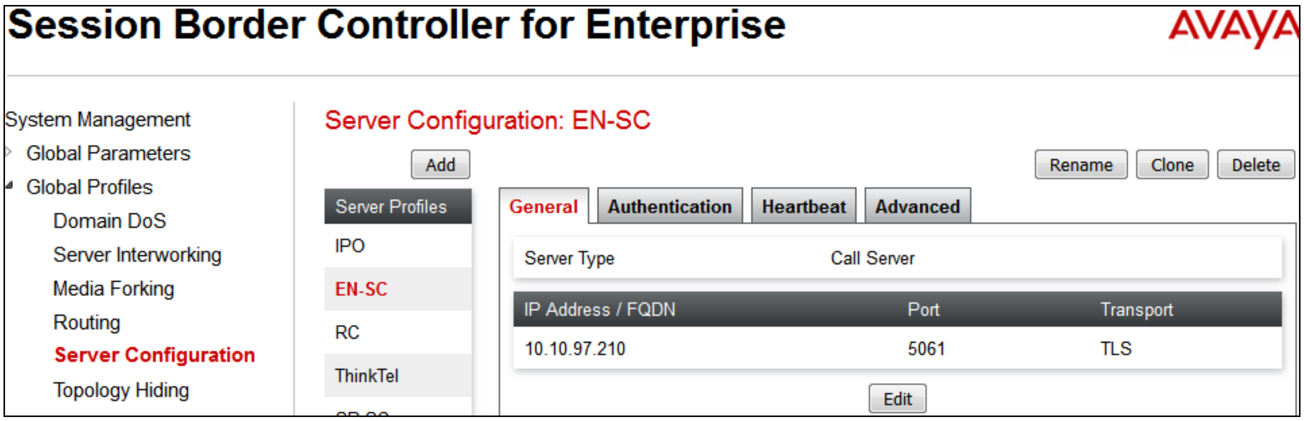


6.2.3.2 Server Configuration for EN

Server Configuration named **EN-SC** created for EN is discussed in detail below. **General** and **Advanced** tabs are provisioned but no configuration is done for **Authentication** tab. The **Heartbeat** tab is kept as ***disabled*** as default to allow Avaya SBCE to forward the OPTIONS heartbeat from SP to EN to query the status of the SIP trunk.

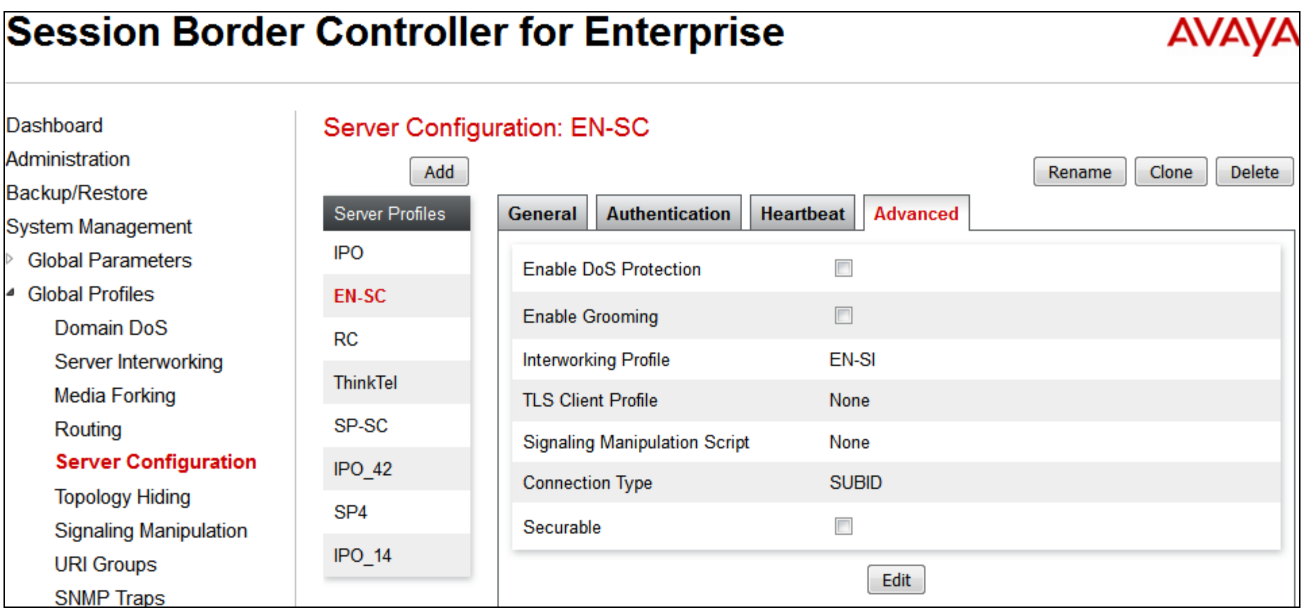
General tab:
Click on the **Edit** button then specify the following.

- **Server Type** for EN as ***Call Server***.
- **IP Address/FQDN** is IP Office IP address.
- **Transport**, the link between Avaya SBCE and EN was ***TLS***. Listening on **Port** ***5061***.



Advanced tab:
Click on the **Edit** button to enter the following information.

- **Interworking Profile** drop down list select ***EN-SI*** as defined in **Section** Error! Reference source not found.**.**
- **Signaling Manipulation Script** drop down list select ***None***.
- The other settings are kept as default.



6.2.4. Routing Profiles

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

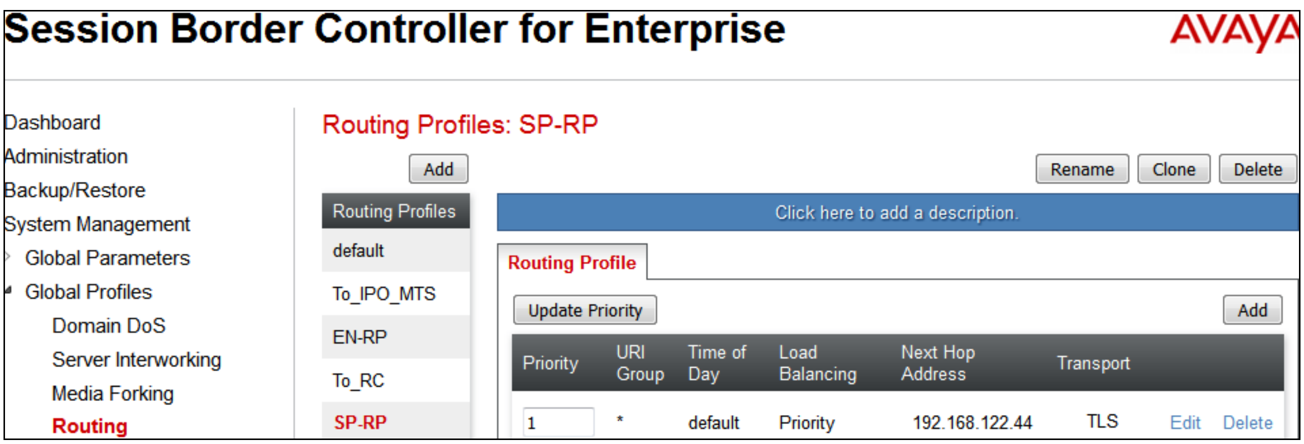
To create a Routing profile, select **Global Profiles → Routing** then click on the **Add** button.

In the compliance testing, Routing profile **SP-RP** was created to be used in conjunction with Server Flow (see **Section 6.4.4**) defined for EN. This entry is to route outgoing calls from the enterprise to SP.

In the opposite direction, Routing profile **EN-RP** was created to be used in conjunction with Server Flow (see **Section 6.4.4**) defined for SP. This entry is to route incoming calls from SP to the EN.

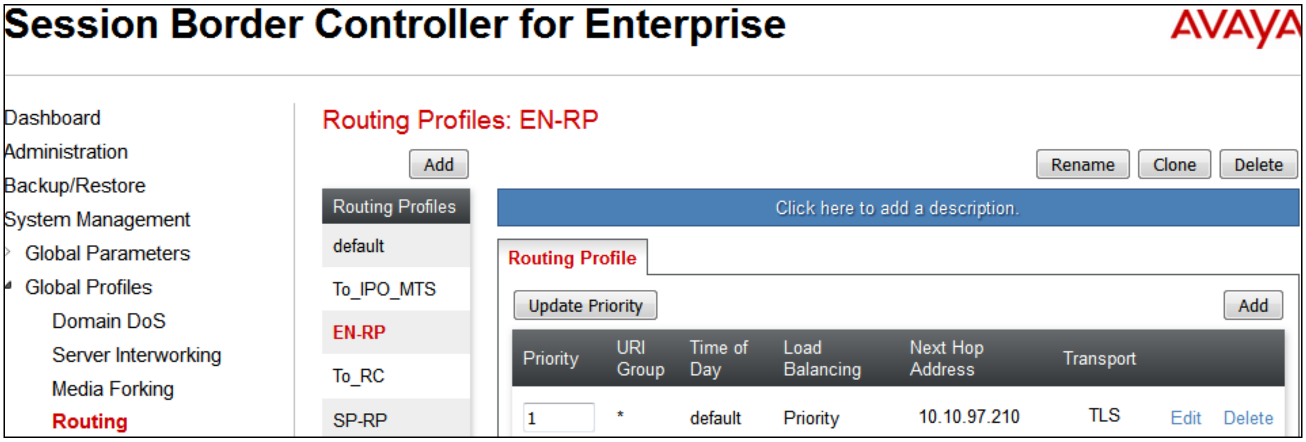
6.2.4.1 Routing Profile for SP

The screenshot below illustrate the routing profile from Avaya SBCE to the SP network, **Global Profiles → Routing**: **SP-RP**. As shown in **Figure 1**, the SP SIP trunk is connected with transportation protocol ***TLS***. If there is a match in the "To" or "Request URI" headers with the URI Group **SP** defined in **Section** Error! Reference source not found., the call will be routed to the **Next Hop Address** which is the IP address of SP SIP trunk.



6.2.4.2 Routing Profile for EN

The Routing Profile for SP to EN, **SP-to-EN**, was defined to route call where the “To” header matches the URI Group **SP** defined in **Section** Error! Reference source not found. to **Next Hop Address** which is the IP address of IP Office as a destination. As shown in **Figure 1**, the SIP trunk between EN and Avaya SBCE is connected with transportation protocol ***TLS***.



6.2.5. Topology Hiding

Topology Hiding is a security feature of Avaya SBCE which allows changing certain key SIP message parameters to ‘hide’ or ‘mask’ how the enterprise network may appear to an unauthorized or malicious user.

To create a Topology Hiding profile, select **Global Profiles → Topology Hiding** then click on the **Add Profile** (not shown).

In the compliance testing, two Topology Hiding profiles were created: **SP-TH** and **EN-TH**.

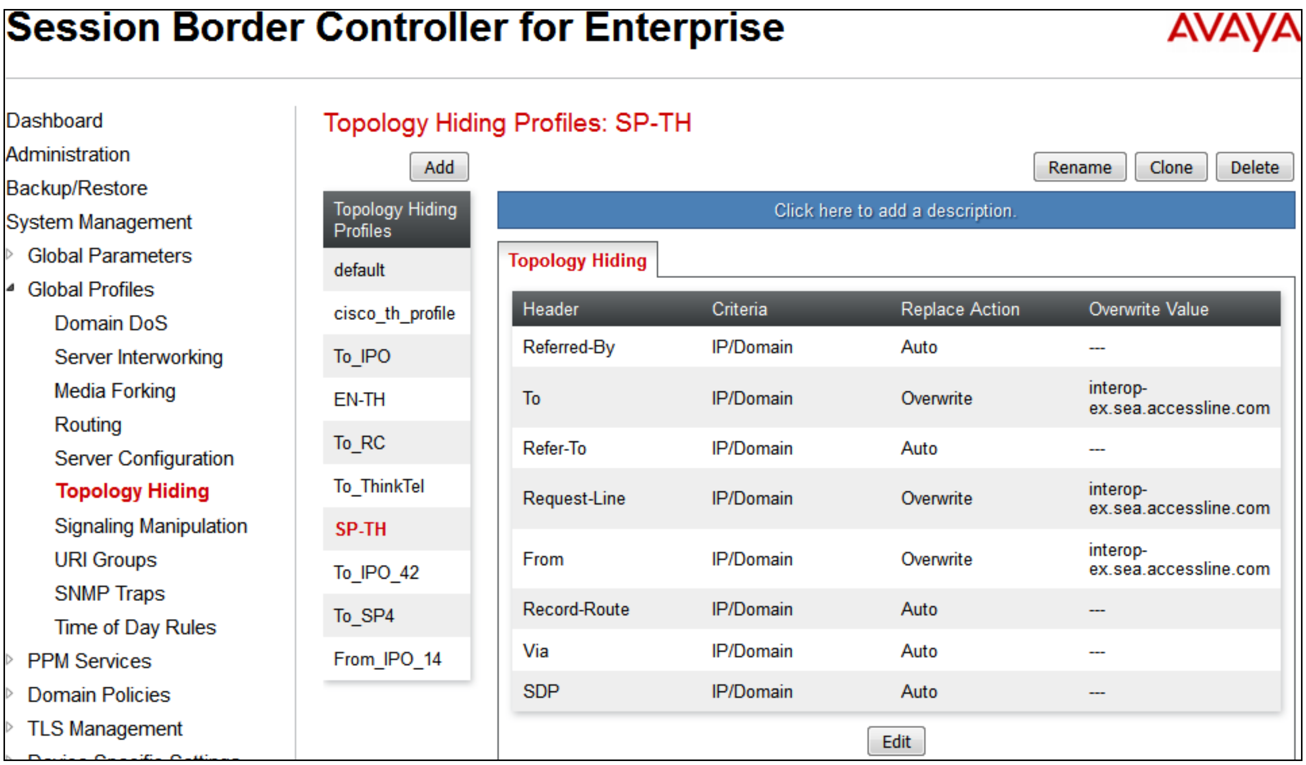
6.2.5.1 Topology Hiding Profile for SP

Topology Hiding profile **SP-TH** was defined for outgoing calls to SP to:

- Mask URI-Host of the “Request-Line” and “To” headers with service provider SIP domain to meet the requirements of SP.
- Mask URI-Host of the “From” header to SP SIP domain as shown in capture below.

This implementation is to secure the enterprise network topology and also to meet the SIP requirements from the service provider.

The screenshots below illustrate the Topology Hiding profile **SP-TH**.

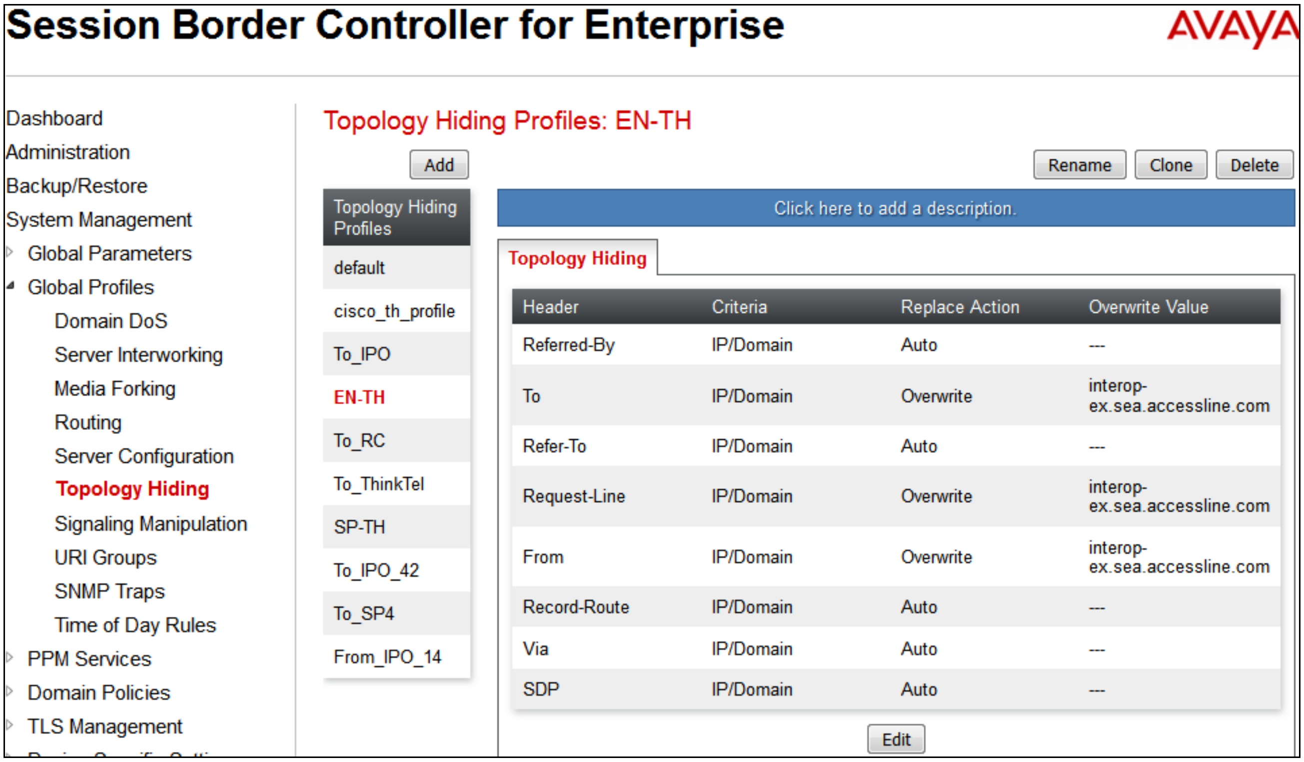


6.2.5.2 Topology Hiding Profile for EN

Topology Hiding profile **EN-TH** was defined for incoming calls to IP Office to:

- Mask URI-Host of the “Request-Line”, “To”, and “From” headers with the enterprise SIP domain.
- Leave the “Record-Route”, “Via” headers and SDP to default **Auto**.

The screenshots below illustrate the Topology Hiding profile **EN-TH**.



6.3. Domain Policies

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

6.3.1. Application Rules

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

Select Domain Policies Application Rules from the left side menu as shown below. In the sample configuration, a single default application rule "default" was used. For field deployment create an application rule with the concurrent sessions purchased.

6.3.1.1 Application Rule for SP

Clone the Application Rule default with a descriptive name e.g. SP-AR for service provider and click the Edit button to change value of **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** to ***500*** respectively as shown and then click the Finish button (not shown). Others are left as default.

6.3.1.2 Application Rule for EN

Similarly, clone the Application Rule default with a descriptive name e.g. EN-AR for IP Office and click the Edit button to change value of **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** to ***500*** respectively as shown and then click the Finish button (not shown). Others are left as default.

6.3.2. Media Rules

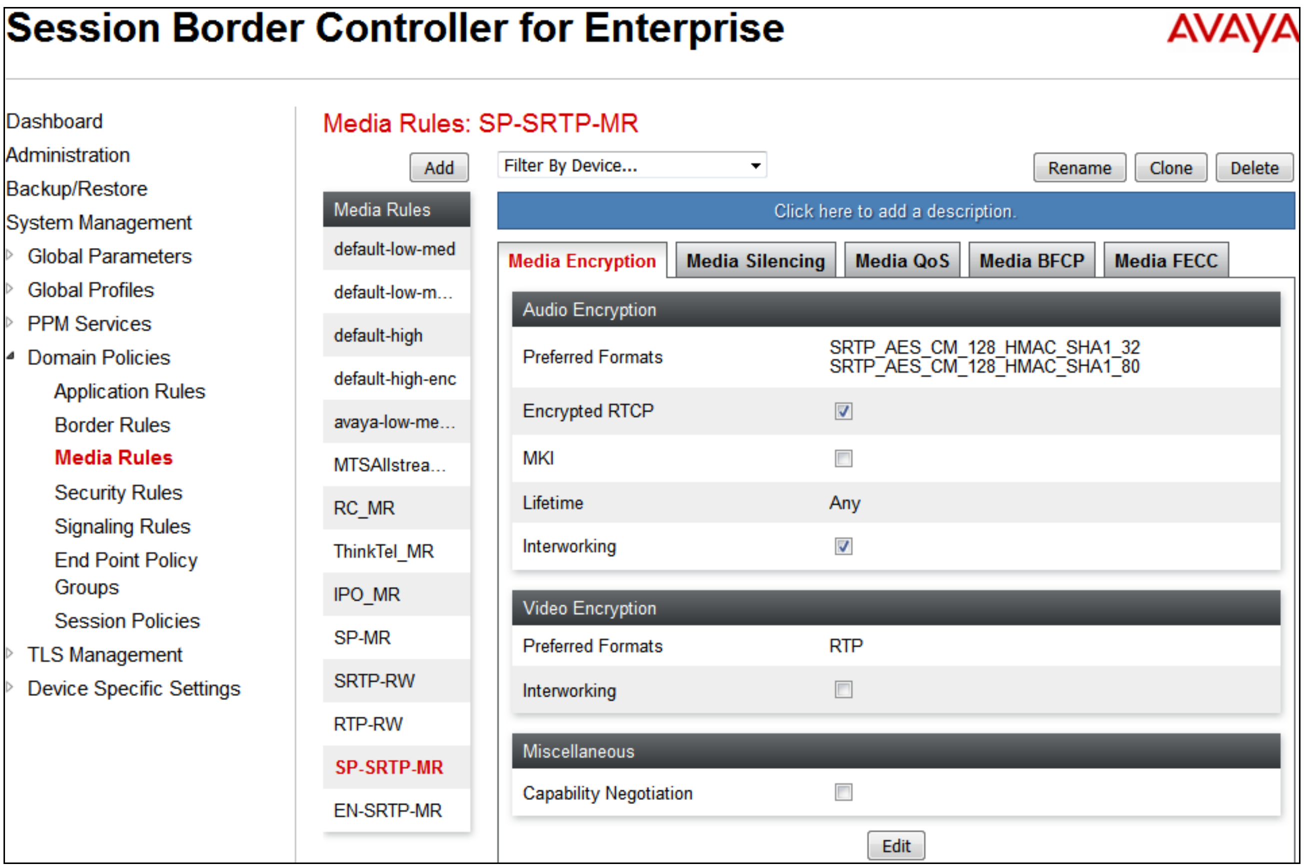
Media Rules define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media related parameters define a strict profile that is associated with other SIP specific policies to determine how media packets matching these criteria will be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

To clone a signaling rule, navigate to **Domain Policies → Media Rules,** select the **default** rule then click on the **Clone Rule** button (not shown).

In the compliance testing, two **Media Rules** were created for SP and EN.

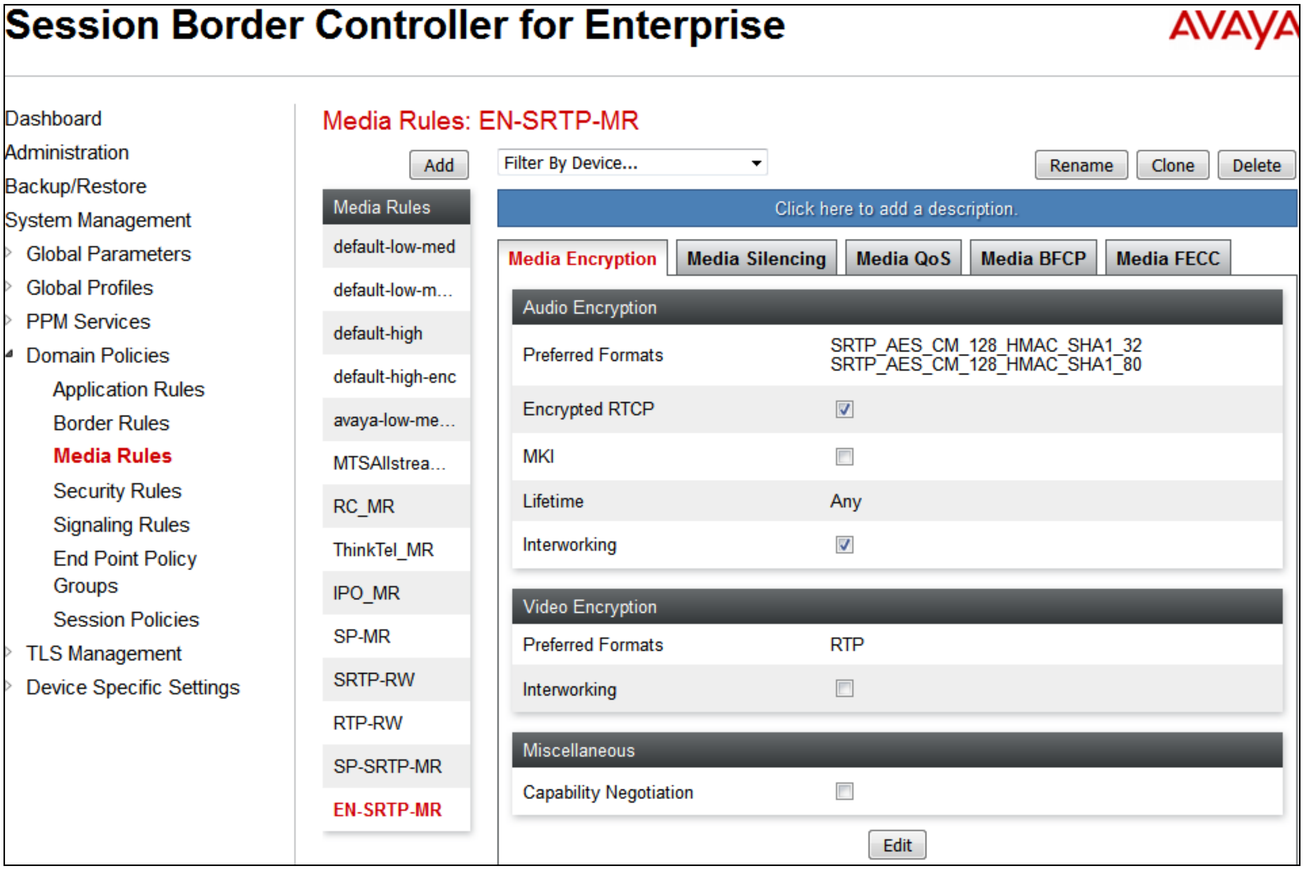
6.3.2.1 Media Rule for SP

Clone the Signaling Rule **default-low-med** with a descriptive name e.g. **SP-SRTP-MR** and click on the **Finish** button (not shown). Select this newly created rule and click on **Edit** to modify the parameters as shown below. Then click on **Finish** (not shown) again to save the changes.



6.3.2.2 Media rule for EN

Clone the Media Rule **default-low-med** with a descriptive name e.g. **EN-SRTP-MR** and click on the **Finish** button (not shown). Select this newly created rule and click on **Edit** to modify the parameters as shown below. Then click on **Finish** again to save the changes.



6.3.3. Signaling Rules

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

To clone a signaling rule, navigate to **Domain Policies → Signaling Rules**, select the **default** rule then click on the **Clone Rule** button (not shown).

In the compliance testing, two **Signaling Rules** were created for SP and EN.

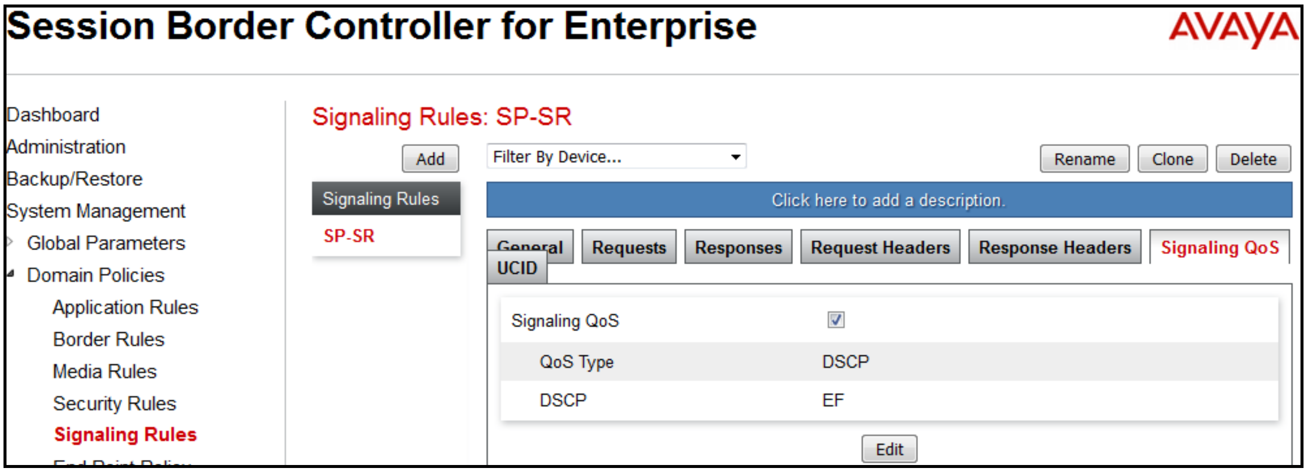
6.3.3.1 Signaling Rule for SP

Clone the Signaling Rule **default** with a descriptive name e.g. **SP-SR** and click on the **Finish** button (not shown). Verify that **General** settings of **SP-SR** with **Inbound** and **Outbound Request** were set to **Allow**, and **Enable Content-Type Checks** was enabled with **Action** and **Multipart-Action** were set to **Allow** (not shown).

On the **Signaling QoS** tab and enter the following information.

- Select the proper Quality of Service (QoS).
- Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP packet header with specific values to support Quality of Services policies for signaling.

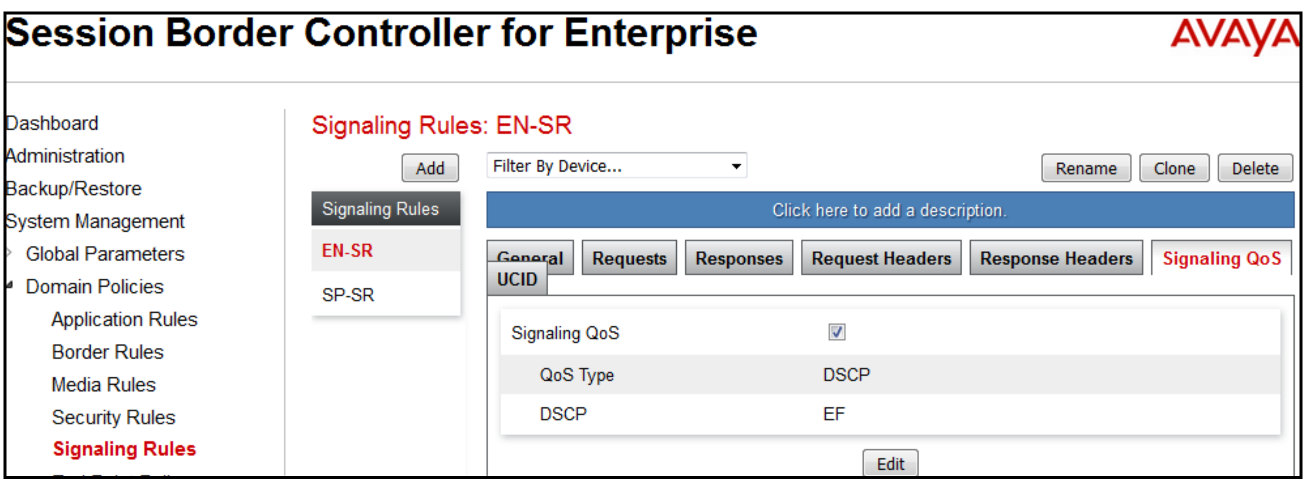
The following screen shows the QoS value used for the compliance testing.



6.3.3.2 Signaling Rule for EN

Clone the Signaling Rule **default** with a descriptive name e.g. **EN-SR** for EN and click on the **Finish** button (not shown). Verify that **General** settings of **EN-SR** with **Inbound** and **Outbound Request** were set to **Allow**, and **Enable Content-Type Checks** was enabled with **Action** and **Multipart-Action** were set to **Allow** (not shown). Similarly the Signaling QoS rules are set as shown in Figure below.

Similarly the Signaling QoS rules are set as shown in Figure below.



6.3.4. Endpoint Policy Groups

The rules created within the Domain Policy section are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to Server Flow defined in **Section 6.4.4**.

Endpoint Policy Groups were separately created for SP and EN.

To create a policy group, navigate to **Domain Policies → Endpoint Policy Groups** and click on the **Add Group** button (not shown).

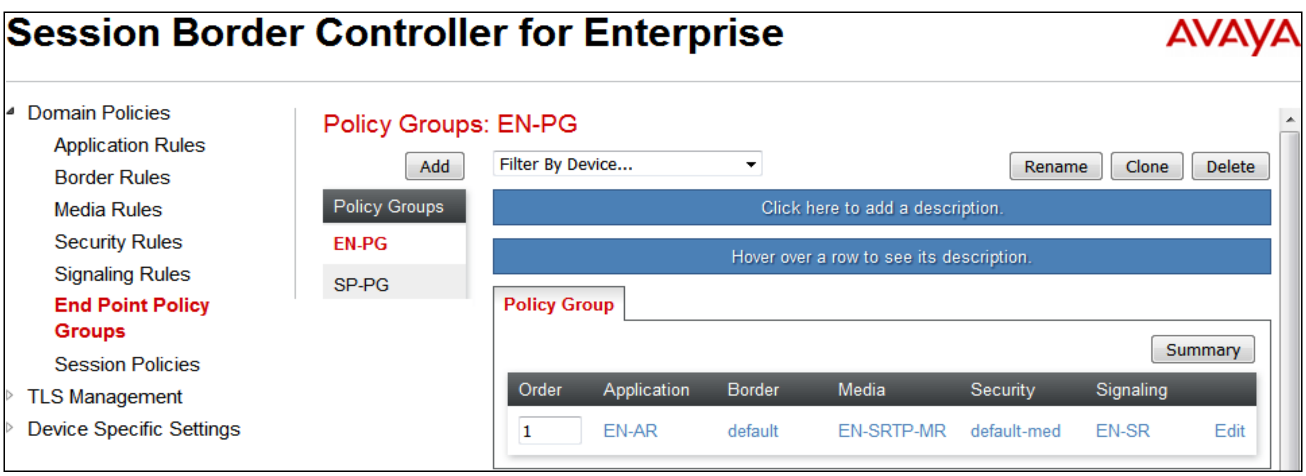
6.3.4.1 Endpoint Policy Group for SP

The following screen shows **SP-PG** created for SP.

- Set Application Rule to ***SP-AR*** which was created in **Section 6.3.1.1**.
- Set Media Rule to ***SP-SRTP-MR*** which was created in **Section 6.3.2.1**.
- Set Signaling Rule to ***SP-SR*** which was created in **Section 6.3.3.1**.
- Set Border Rule to ***default***.
- Set Security Rule to ***default-med***.

6.3.4.2 Endpoint Policy Group for EN

Similarly, the following screen shows policy group **EN-PG** created for EN.



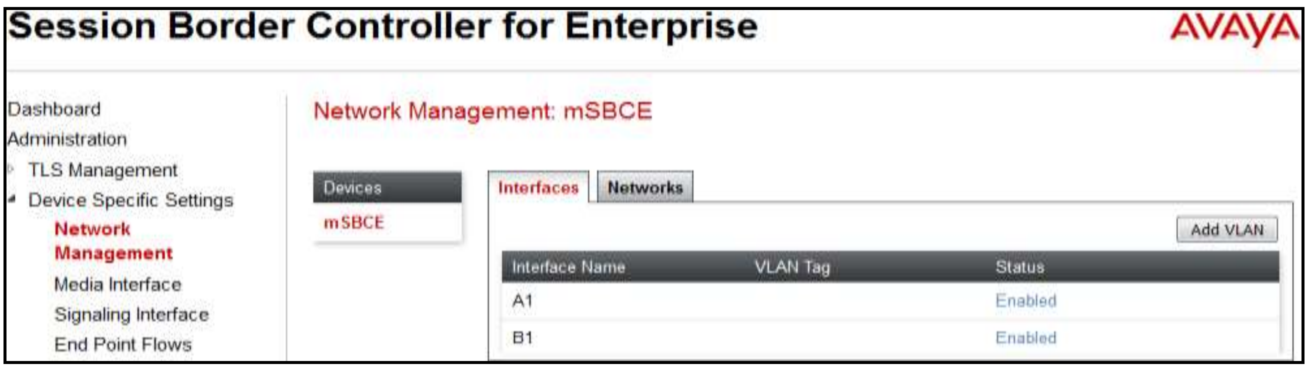
6.4. Device Specific Settings

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

6.4.1. Network Management

The Network Management page is where the network interface settings are configured and enabled. During the installation process of Avaya SBCE, certain network-specific information is defined such as device IP address, public IP address, subnet mask, gateway, etc. to interface the device to the networks. This information populates the various Network Management tabs which can be edited as needed to optimize device performance and network efficiency.

Navigate to **Device Specific Settings → Network Management**, under **Interfaces** tab, enable the interfaces connecting to the inside enterprise and outside service provider networks. To enable an interface, click on "Disable" Status. The following screen shows interface **A1** and **B1** were **Enabled**.



On the **Networks** tab and verify the IP addresses assigned to the interfaces and that the interfaces were enabled. The following screen shows the private interface was assigned to **A1** and the public interface was assigned to **B1** appropriate to the parameters shown in the **Figure 1**.

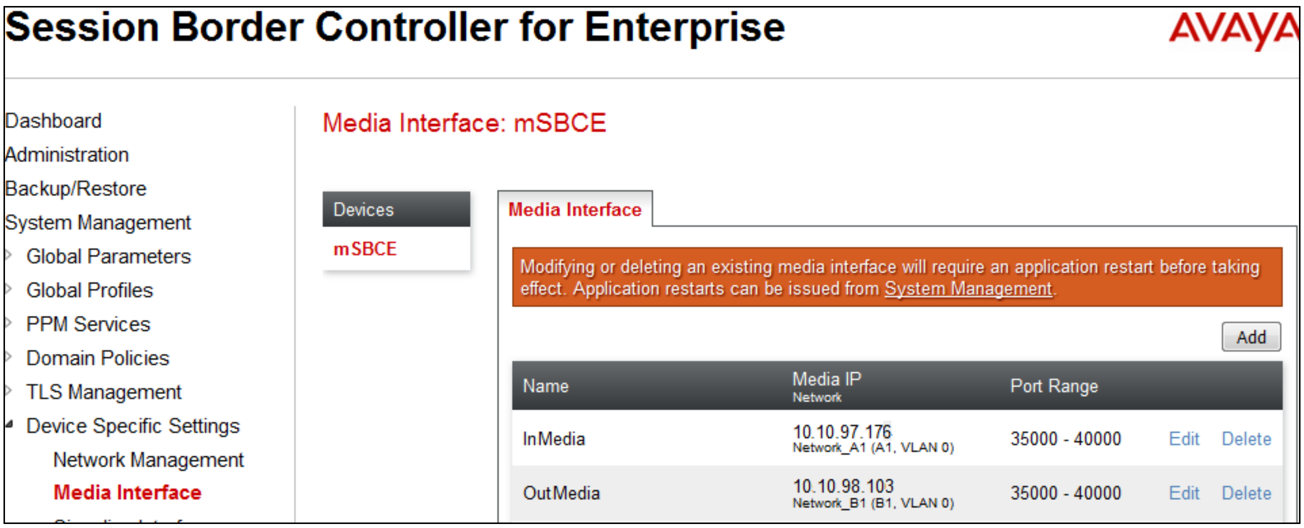
6.4.2. Media Interface

The Media Interface screen is where the media ports are defined. Avaya SBCE will open connection for RTP traffic on the defined ports.

To create a new **Media Interface**, navigate to **Device Specific Settings → Media Interface** and click on the **Add Media Interface** button (not shown).

Two separate Media Interfaces are needed for both the inside and outside interfaces. The following screen shows the Media Interfaces **InsideMedia** and **OutsideMedia** were created for the compliance testing.

Note: After the media interfaces are created, an application restart is necessary before the changes will take effect.

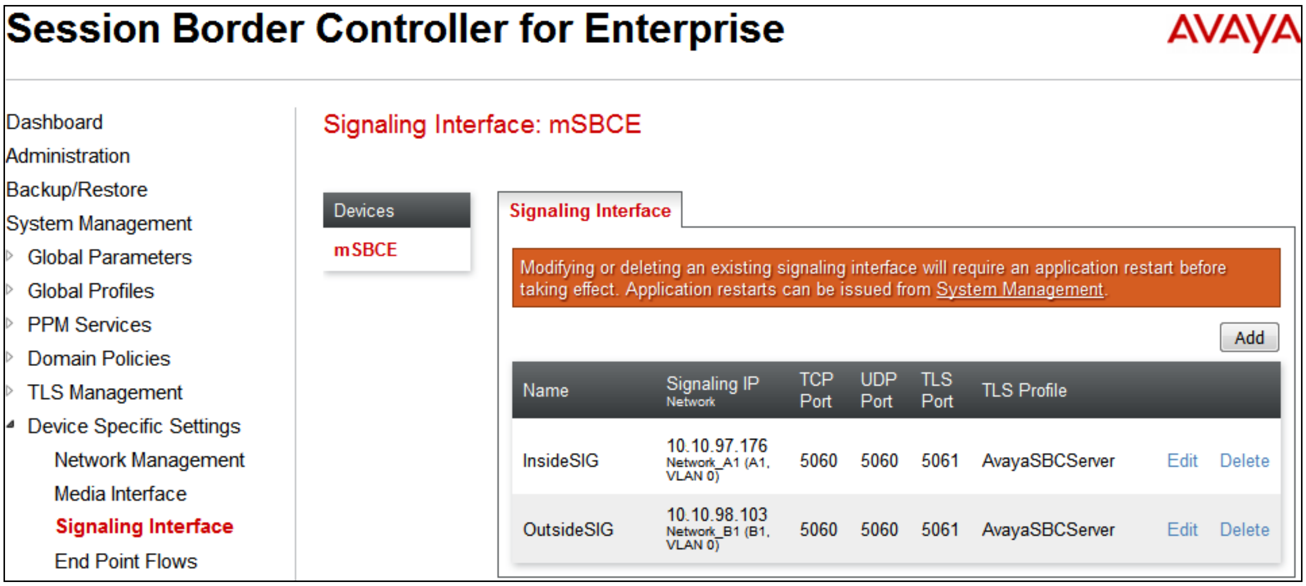


6.4.3. Signaling Interface

The Signaling Interface screen is where the SIP signaling port is defined. Avaya SBCE will listen for SIP requests on the defined port.

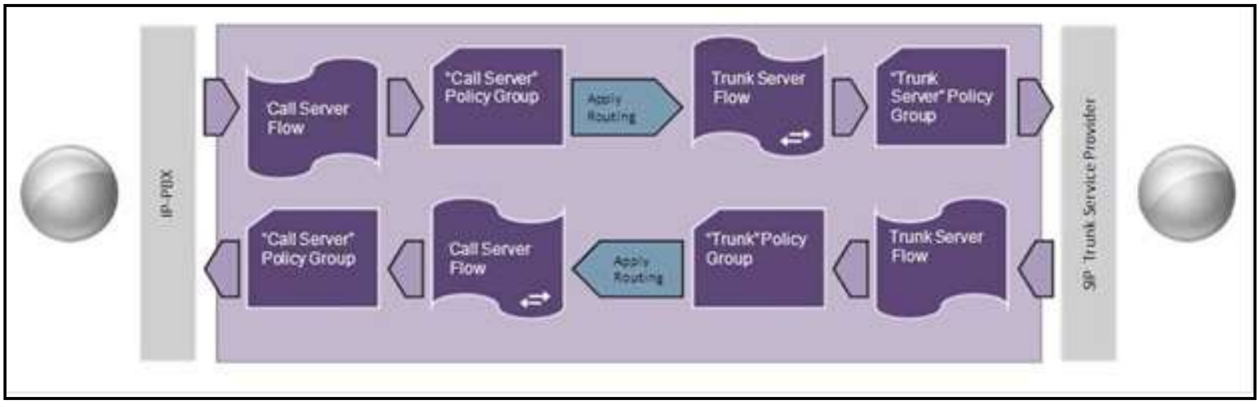
To create a new **Signaling Interface**, navigate to **Device Specific Settings → Signaling Interface** and click on the **Add Signaling Interface** button (not shown).

Two separate Signaling Interfaces are needed for both inside and outside interfaces. The following screen shows the Signaling Interfaces **InsideSIP** and **OutsideSIP** were created in the compliance testing with **TLS/5061** and **TLS/5061** respectively configured for inside and outside interfaces.



6.4.4. End Point Flows - Server Flow

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

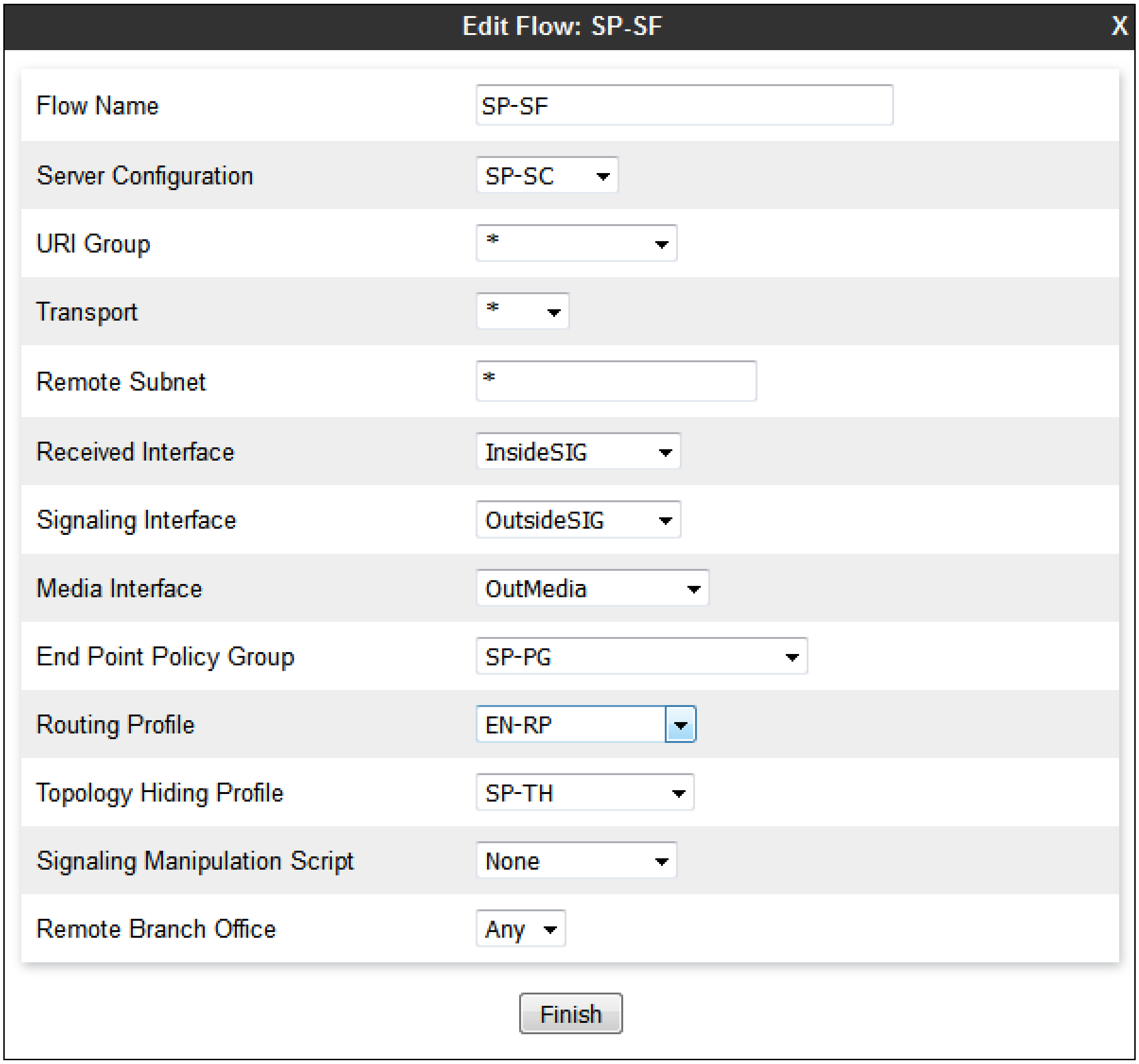


In the compliance testing, two separate Server Flows were created for SP and EN.

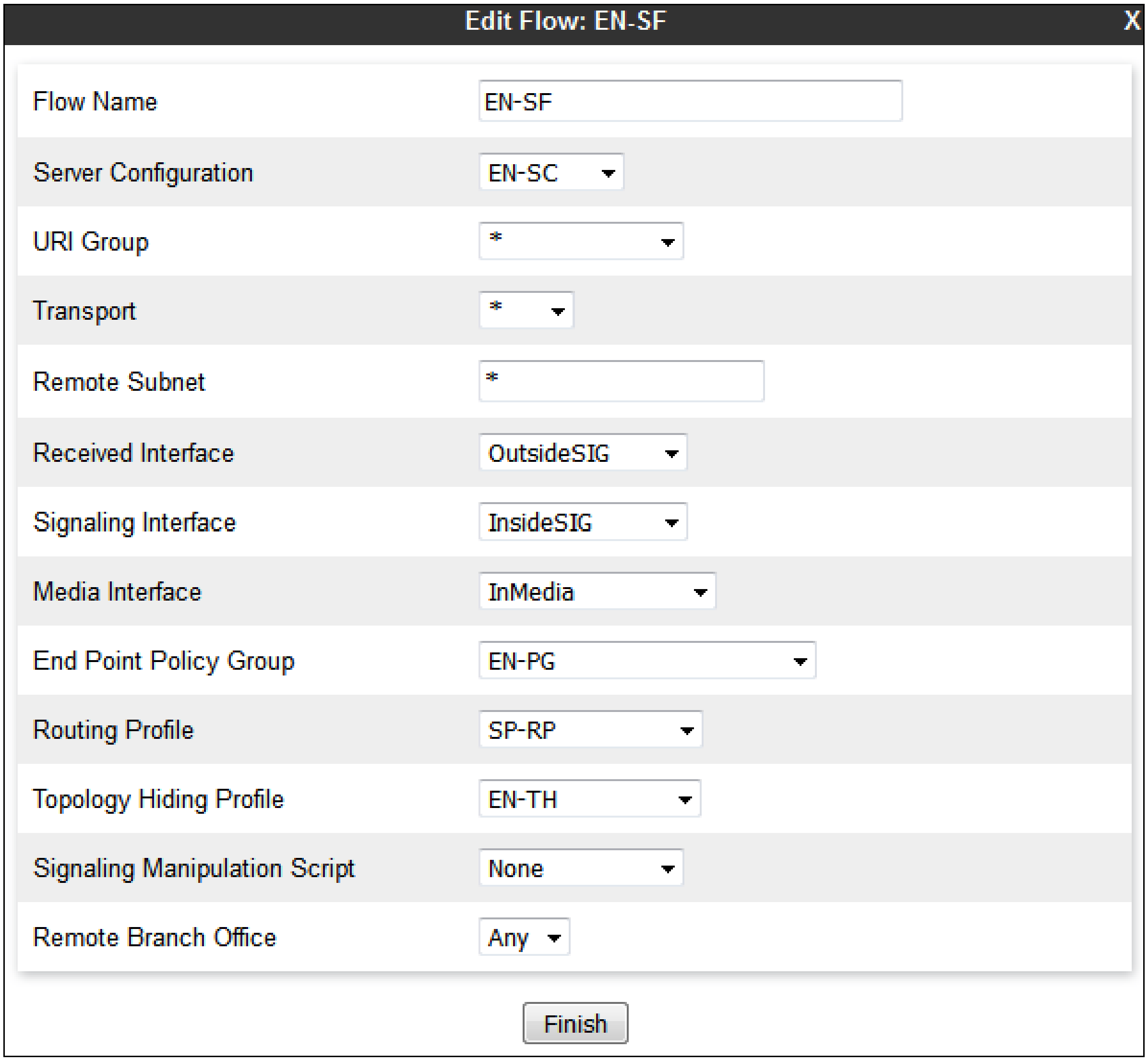
To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**, select the **Server Flows** tab and click on the **Add Flow** button (not shown). In the new window that appears, enter the following values while the other fields were kept as default.

- **Flow Name**: Enter a descriptive name.
- **Server Configuration**: Select Server Configuration created in **Section 6.2.3** which the Server Flow associates to.
- **URI Group**: Select "*****".
- **Received Interface**: Select the Signaling Interface created in **Section 6.4.3** which is the Server Configuration is designed to receive SIP signaling from.
- **Signaling Interface**: Select the Signaling Interface created in **Section 6.4.3** which is the Server Configuration is designed to send the SIP signaling to.
- **Media Interface**: Select the Media Interface created in **Section 6.4.2** which is the Server Configuration is designed to send the RTP to.
- **End Point Policy Group**: Select the End Point Policy Group created in **Section 6.3.4**.
- **Routing Profile**: Select the Routing Profile created in **Section 6.2.4**.
- **Topology Hiding Profile**: Select the Topology Hiding profile created in **Section 6.2.5** to apply toward the Server Configuration.
- Use default values for all remaining fields. Click **Finish** to save and exit.

The following screen shows the Server Flow **SP-SF** for SP.



Similarly, the following screen shows the Server Flow **EN-SF** for EN.



7. Intermedia SIP Trunking Configuration

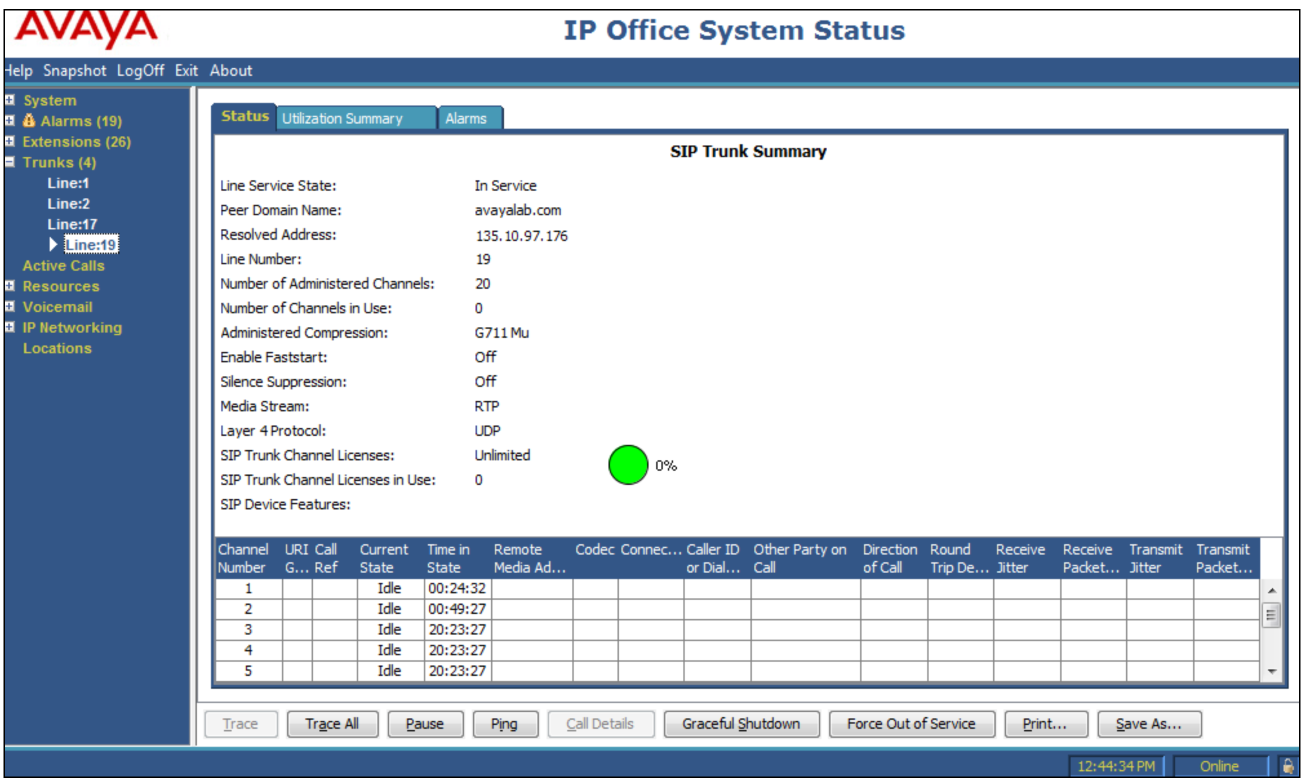
Intermedia is responsible for the configuration of Intermedia SIP Trunking service. The customer will need to provide the IP address used to reach Avaya IP Office at the enterprise, this address will be the outside interface of Avaya SBCE. Intermedia will provide the customer the necessary information to configure Avaya IP Office SIP connection to Intermedia. The provided information from Intermedia includes:

- IP address of the Intermedia SIP proxy.
- Supported codecs.
- DID numbers.
- IP addresses and port numbers used for signaling or media through any security devices.

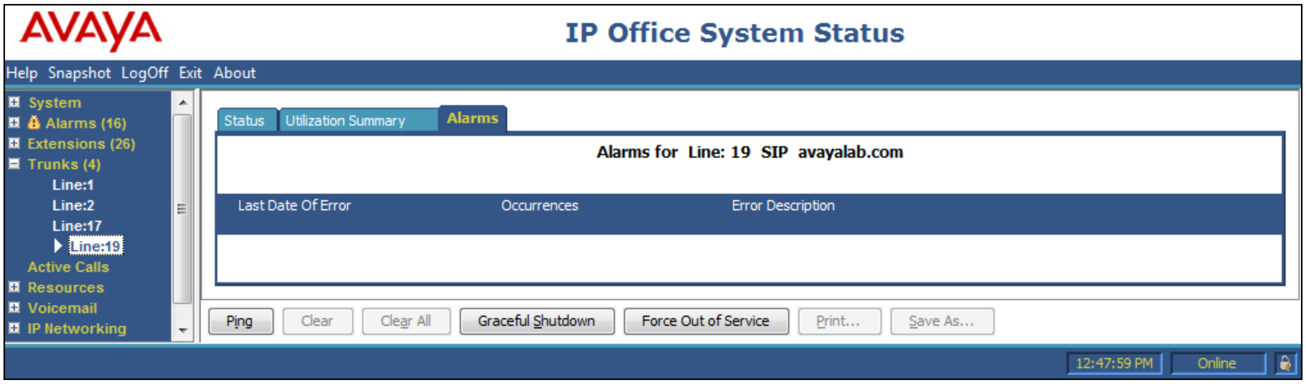
8. Verification Steps

The following steps may be used to verify the configuration:

- Use 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 Avaya IP Office Manager was 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.



- Verify that a phone connected to PSTN can successfully place a call to Avaya IP Office with two-way audio.
- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Using a network sniffing tool e.g. Wireshark to monitor the SIP signaling between the enterprise and Intermedia. The sniffer traces are captured at the public interface of Avaya SBCE.

9. Conclusion

The Intermedia SIP Trunking passed compliance testing. These Application Notes describe the procedures required to configure the SIP connection between Avaya IP Office, Avaya SBCE and the Intermedia SIP Trunking service as shown in **Figure 1**.

10. Additional References

[1] *IP Office 9.1 Administering Avaya IP Office Platform with Manager*, Release 9.1.0, Issue 10.03, February 2015.

[2] *Administering Avaya IP Office™, Platform Voicemail Pro IP Office™ Platform 9.115-601063*, Issue 10c - (09 December 2014).

[3] / *IP Office Embedded Voicemail User Guide (IP Office Mode), Document number 15-604067*, Issue 9.0, 10 September 2013.

[4] *Avaya Session Border Controller for Enterprise Overview and Specification*, Release 7.0, Issue 1, August 2015.

[5] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, Issue 1, August 2015.

[6] *Deploying Avaya Session Border Controller in Virtualized Environment*, Release 7.0, Issue 1, August 2015.

[7] *Administering Avaya Session Border Controller for Enterprise,* Release 7.0, Issue 1, August 2015.

[8] *Application Notes for configuring Avaya IP Office 9.0 and Avaya Session Border Controller for Enterprise 6.3 to support Remote Workers*, Issue 1.0.

[9] *Using Avaya Communicator for Web*, Release 1, Issue 1.0.4, October 2015.

Product documentation for Avaya products may be found at http://support.avaya.com. Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/

Product documentation for Intermedia SIP Trunking is available from Intermedia.

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.