Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Fusion Connect SIP Trunking Service with Avaya IP Office R9.1 and Avaya Session Border Controller for Enterprise R6.3 - Issue 0.1

Abstract

These Application Notes describe the procedures for configuring Avaya IP Office Release 9.1 and Avaya Session Border Controller for Enterprise Release 6.3 to inter-operate with the Fusion Connect SIP Trunking Service. Fusion Connect is a member of the Avaya DevConnect Service Provider program..

The Fusion Connect SIP Trunking Service provides PSTN access via a SIP trunk between an enterprise site and the Fusion Connect network as an alternative to legacy analog or digital trunks. This approach generally results in easier maintenance and lower cost for the business customer.

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.

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 an enterprise solution using Avaya IP Office Release R9.1 and Avaya Session Border Controller for Enterprise (Avaya SBCE) R6.3 to inter-operate with the Fusion Connect SIP Trunking Service.

In the sample configuration, the enterprise solution consists of an Avaya SBCE, an Avaya IP Office 500 V2 running Release 9.1 software, Avaya Preferred Edition (a.k.a Voicemail Pro) messaging application, Avaya H.323 and SIP deskphones, and the SIP-based Avaya Communicator softphone. Customers using this Avaya IP Office enterprise solution with the Fusion Connect SIP Trunking Service are able to place and receive PSTN calls via a broadband WAN connection using SIP. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to Fusion Connect via Avaya SBCE and the public Internet. The 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

To verify SIP Trunking interoperability, the following features and functionality were covered during the compliance test.

- SIP OPTIONS queries and responses.
- Incoming calls from the PSTN to H.323 and SIP telephones at the enterprise. All inbound calls from the PSTN were routed from the service provider across the SIP trunk to the enterprise.
- Outgoing calls to the PSTN from H.323 and SIP telephones at the enterprise. All outbound calls to the PSTN were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound calls to the Avaya Communicator for Windows (SIP soft client).
- Various call types including: local, long distance, toll-free, international, Local Directory Assistance, and Emergency 911 calls.
- G.711u and G.729a codecs.
- DTMF transmission using RFC 2833.
- Caller ID presentation and Caller ID restriction.
- User features such as hold and resume, internal call forwarding, transfer, and conference.
- Off-net call transfer, conference, forwarding and mobility (mobile twinning).

- Use of the SIP REFER method for call transfers to the PSTN.
- Voicemail navigation for inbound and outbound calls, and voicemail Message Waiting Indicator (MWI).
- T.38 fax and G.711u pass-through fax.
- Inbound and outbound long-duration and long hold time call stability.
- Response to incomplete call attempts and trunk errors.
- Remote Worker which allows Avaya SIP endpoints to connect directly to the public Internet as enterprise extensions.

2.2. Test Results

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

- **Supported Codecs –** Fusion Connect supports both G.711u and G.729a codecs. However, the codecs cannot be changed dynamically from one to the other through Avaya IP Office just on the enterprise side. Fusion Connect must be notified about which one of the two codecs a customer prefers before configuring for and turning up the service. If a customer desires to change to a different codec (e.g., from G.711u to G.729a) afterwards, Fusion Connect must make the configuration change on the service side after receiving notification from the customer.
- **Inbound T.38 Fax –** After initial connect, Fusion Connect would reject the re-INVITE from Avaya IP Office for switching to T.38 with "488 Not Acceptable Here", and issue its own T.38 re-INVITE towards the enterprise. This extra round of T.38 signaling exchange had no negative impact: the T.38 fax job would run to completion successfully.
- **Outbound T.38 Fax –** No re-INVITE was issued by Fusion Connect for switching to T.38 after an outbound fax call was connected. The fax call would go through successfully using the G.711u pass-through mode (treating fax as regular voice call with best effort).
- **Direct Media –** The Direct Media capability on IP Office allows IP endpoints to send RTP media directly to each other rather than having all the media flow through the IP Office, using up VoIP and relay resources. This capability is not supported by Avaya IP Office on the SIP trunk connection which allows T.38 fax in addition to voice calls. Consequently, Direct Media was disabled for the test circuit configured for the compliance test.

Items not supported by the Fusion Connect SIP Trunking Service included the following:

- **Operator Calls** – Fusion Connect does not support Operator (0) and Operator-Assisted (0 + 10-digits) calls.
- **Session Timer –** Session timer based on RFC 4028 is not implemented by Fusion Connect. During compliance testing, the enterprise sent session refresh re-INVITE messages towards Fusion Connect with the configured session timer on Avaya IP Office.
- **UPDATE Message –** Fusion Connect does not support UPDATE (the Allowed header in SIP messages from Fusion Connect does not contain UPDATE). Consequently, Avaya IP Office used re-INVITE messages to refresh active call sessions during the compliance test.

2.3. Support

For support on the Fusion Connect SIP Trunking Service, please contact Fusion Connect via the following:

- Web: https://www.fusionconnect.com/support
- Phone: (888) 301-1721

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com.

3. Reference Configuration

Figure 1 illustrates the test configuration showing an enterprise site connected to the Fusion Connect SIP Trunking Service.

Located at the edge of the enterprise network is the Avaya SBCE. It has a public side that connects to the Fusion Connect network via the public Internet, and a private side that connects to the enterprise LAN network. All SIP and RTP traffic entering or leaving the enterprise flows through the Avaya SBCE. In this way, the Avaya SBCE can protect the enterprise against any SIP-based attacks. The Avaya SBCE provides network address translation at both the IP and SIP layers.

Within the enterprise site is an Avaya IP Office 500 V2 running the Release 9.1 software. Endpoints include various Avaya IP Telephones (with H.323 and SIP firmware) and the SIP-based Avaya Communicator softphone. The site also has a Windows PC running Avaya Preferred Edition (a.k.a. Voicemail Pro) for providing voice messaging service to the Avaya IP Office users, and Avaya IP Office Manager for administering the Avaya IP Office.

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

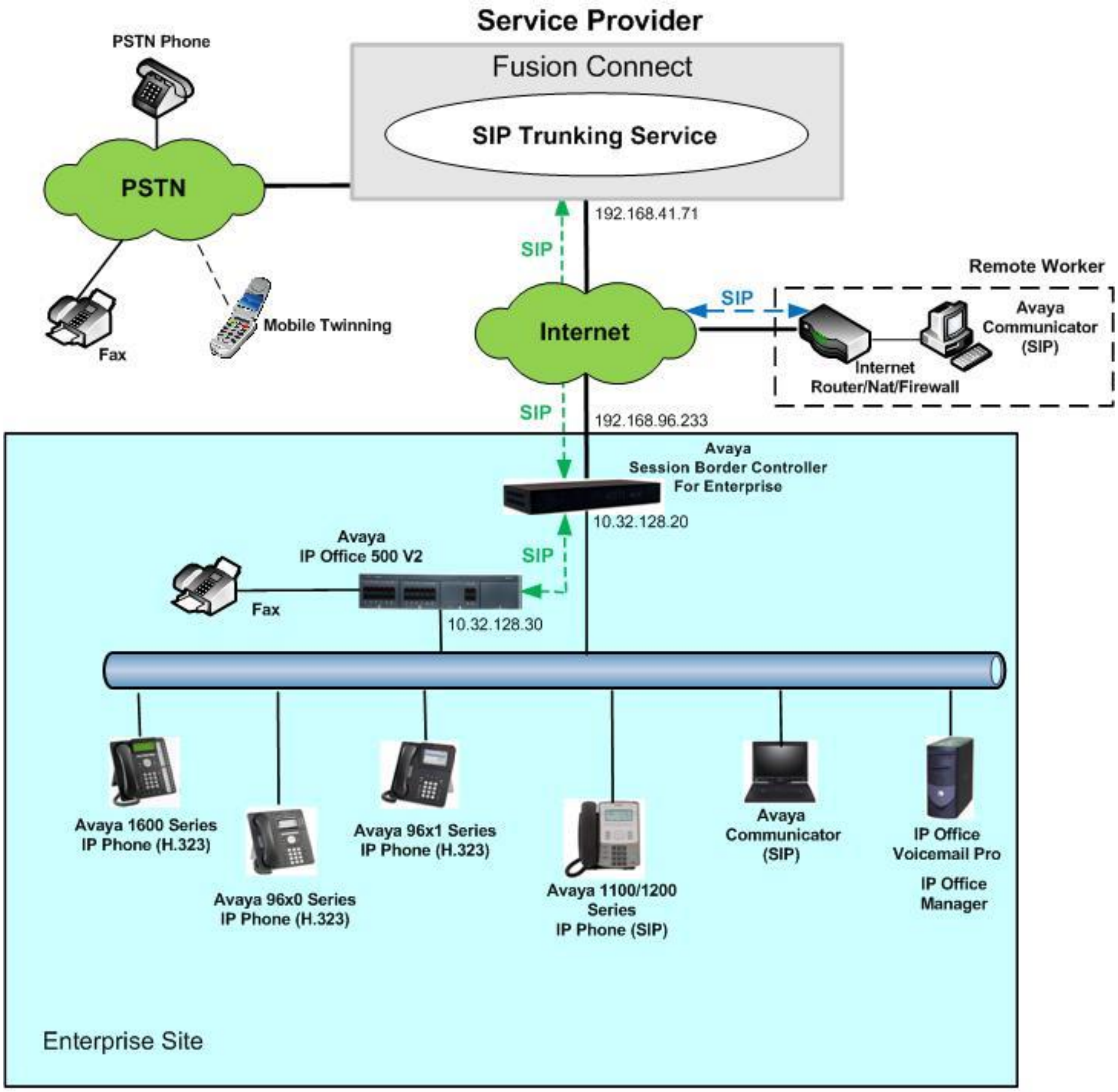


Figure 1: Avaya IP Office with Fusion Connect SIP Trunking Service

For security reasons, any actual public IP addresses used in the configuration have been replaced with private IP addresses in these Application Notes.

During compliance testing, enterprise users dialed a prefix digit 8 or 9 plus N digits to send an outbound call to the number N across the SIP trunk to Fusion Connect. The short code of 8 or 9 was stripped off by Avaya IP Office but the remaining N digits were sent to the service provider network. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits for long distance and local calls. Thus, for these NANP calls, Avaya IP Office sent 11 digits in the

Request URI and the To header of an outbound SIP INVITE message. Fusion Connect sent 10 digits in the Request URI and the To header of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the enterprise network such as a data 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 Avaya IP Office must be allowed to pass through these devices.

The administration of the Avaya Preferred Edition (Voicemail Pro) messaging service and endpoints on Avaya IP Office is standard. Since these configuration tasks are not directly related to the inter-operation with the Fusion Connect SIP Trunking Service, they are not included in these Application Notes.

Remote Worker configuration is also not covered by these Application Notes. For configuration details on Avaya IP Office and Avaya SBCE to support Remote Worker, see [9] in **Section 10**.

4. Equipment and Software Validated

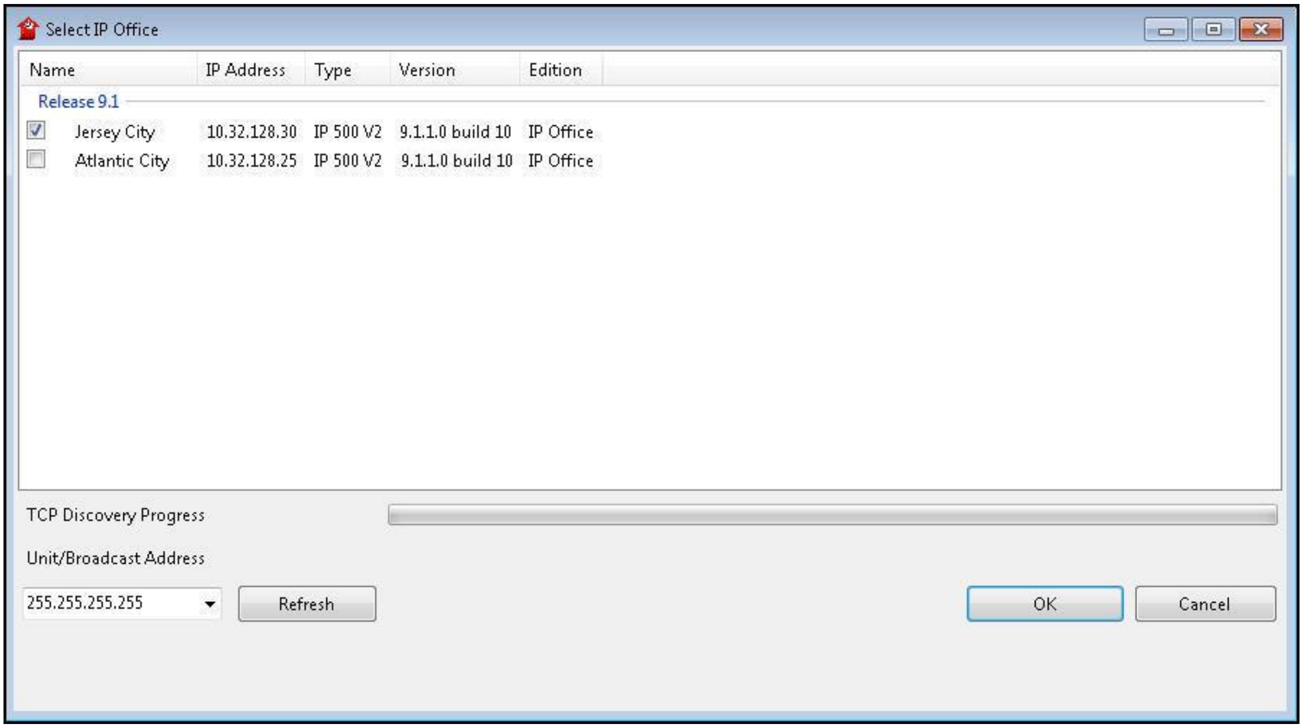
The following equipment and software/firmware were used for the sample configuration provided:

Avaya Telephony Components	
Equipment / Software	**Release / Version**
Avaya IP Office 500 V2	9.1.1.0 build 10
Avaya IP Office COMBO6210/ATM4 Module	9.1.1.0 build 10
Avaya IP Office Manager	9.1.1.0 build 10
Avaya Preferred Edition (a.k.a Voicemail Pro)	9.1.100.3
Avaya 1616 IP Telephones (H.323)	Avaya one-X® Deskphone 1.3 SP5
Avaya 9611G IP Telephones (H.323)	Avaya one-X® Deskphone 6.4.0.14_V452
Avaya 9630G IP Telephones (H.323)	Avaya one-X® Deskphone 3.2.2
Avaya 1120E IP Telephone (SIP)	4.04.18.00
Avaya Communicator for Windows	2.0.3.30
Avaya Session Border Controller for Enterprise running on Portwell CAD-0208 server	6.3.2-08-5478
Fusion Connect Components	
Equipment / Software	**Release / Version**
NBSVoice	3.0
ACME Net-Net 4500 Session Border Controller	SCX6.2

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 (T.38 fax is not supported on IP Office Server Edition). Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog/digital endpoints or trunks.

5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running Avaya IP Office Manager, select **Start → All Programs → IP Office → Manager** to launch the application. A **Select IP Office** pop-up window is displayed as shown below. Select the proper Avaya IP Office system from the pop-up window and click **OK** to log in with the appropriate credentials (not shown). The configuration may alternatively be opened by navigating to **File → Open Configuration** at the top of the Avaya IP Office Manager window.



The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation Pane on the left side, omit the Group Pane in the center, and show the Details Pane on the right side. Since the Group Pane has been omitted, its content is shown as submenus in the Navigation Pane. These panes (Navigation and Details) will be referenced throughout the Avaya IP Office configuration.

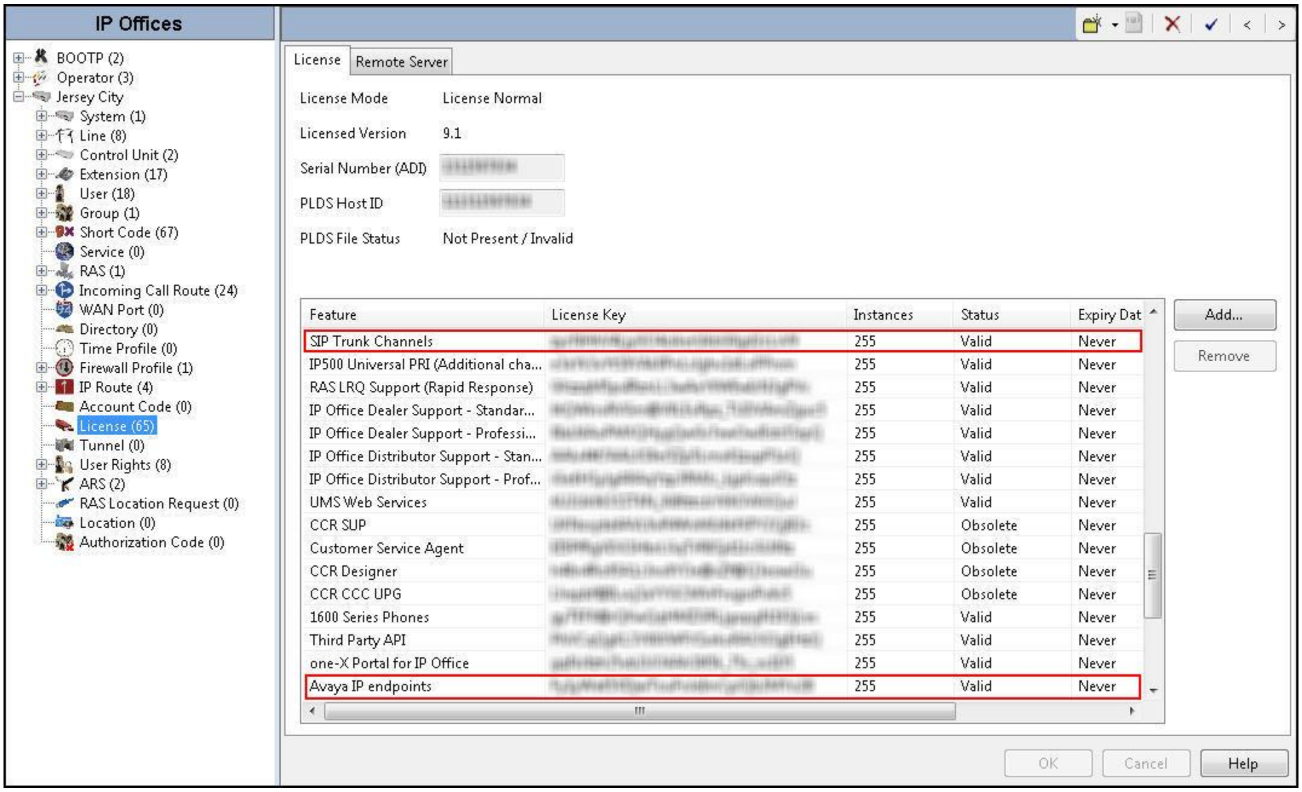
All licensing and feature configuration that is not directly related to the interface with the service provider (such as administering IP endpoints) is assumed to already be in place.

In the sample configuration, **Jersey City** was used as the system name. All navigation described in the following sections (e.g., **Control Unit → IP 500 V2)** appears as submenus underneath the system name **Jersey City** in the Navigation Pane. The configuration screens highlight values/settings configured for the compliance test. Defaults were used for other values and may be customized based upon requirements in the field.

5.1. Licensing and Physical Hardware

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

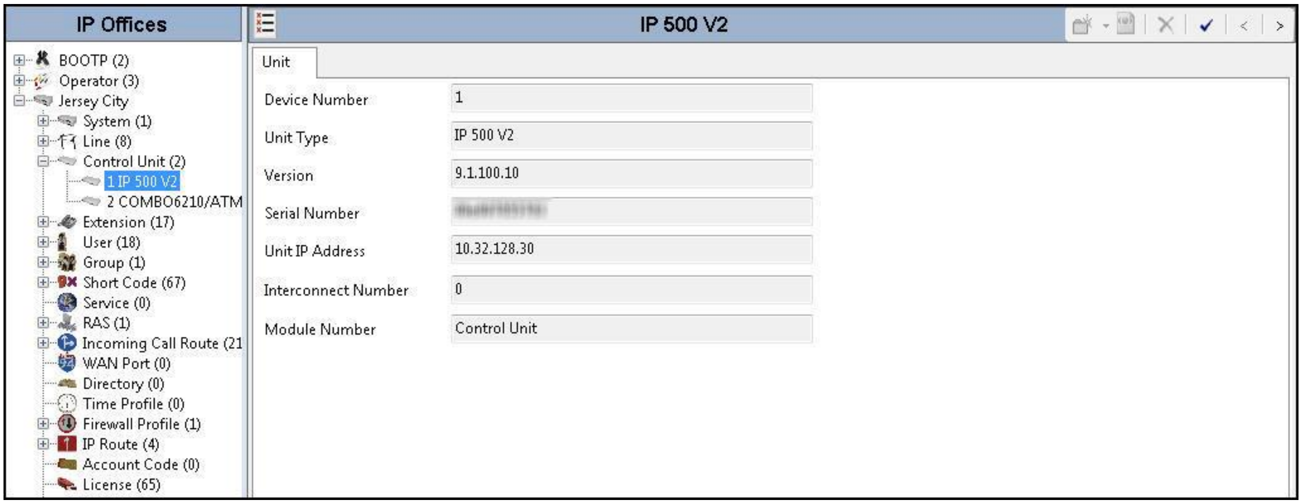
To verify that there is a **SIP Trunk Channels** License with sufficient capacity, click **License** in the Navigation Pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details Pane. The screen below also shows the valid license for **Avaya IP endpoints**.



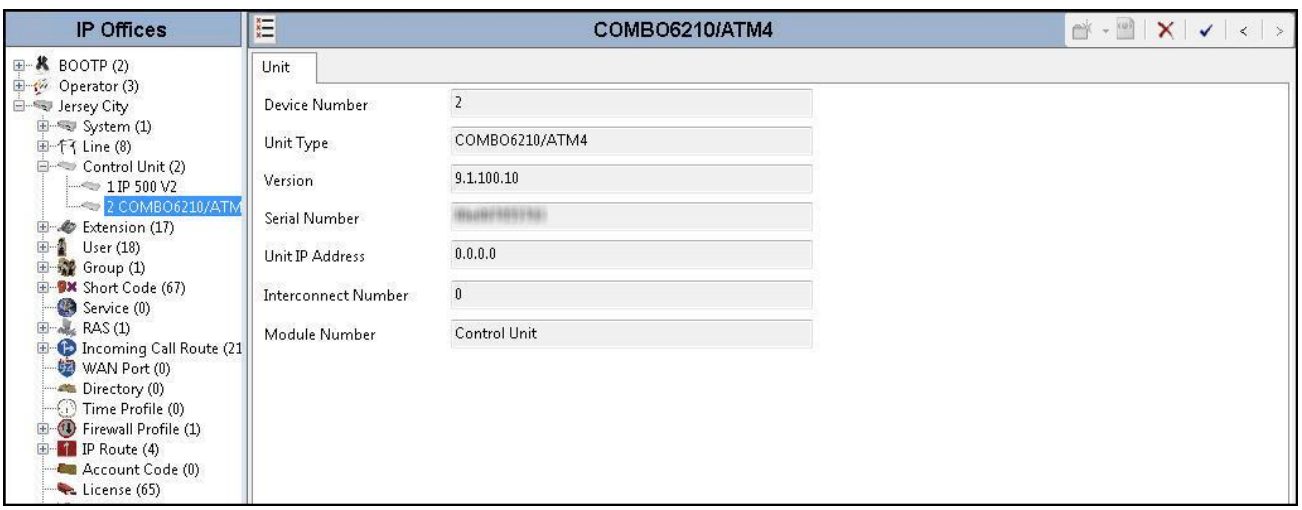
To view the physical hardware comprising the Avaya IP Office system, expand the components under **Control Unit** in the Navigation Pane. In the sample configuration, the second component listed is a Combination Card. This module has 6 digital station ports, two analog extension ports, 4 analog trunk ports and 10 VCM channels. The VCM is a Voice Compression Module supporting VoIP codecs. An Avaya IP Office hardware configuration with a VCM component is necessary to support SIP Trunking.

To view the details of the component, select the component in the Navigation Pane.

The screen below shows the details of the IP 500 V2.



The screen below shows the details of the Combination Card.

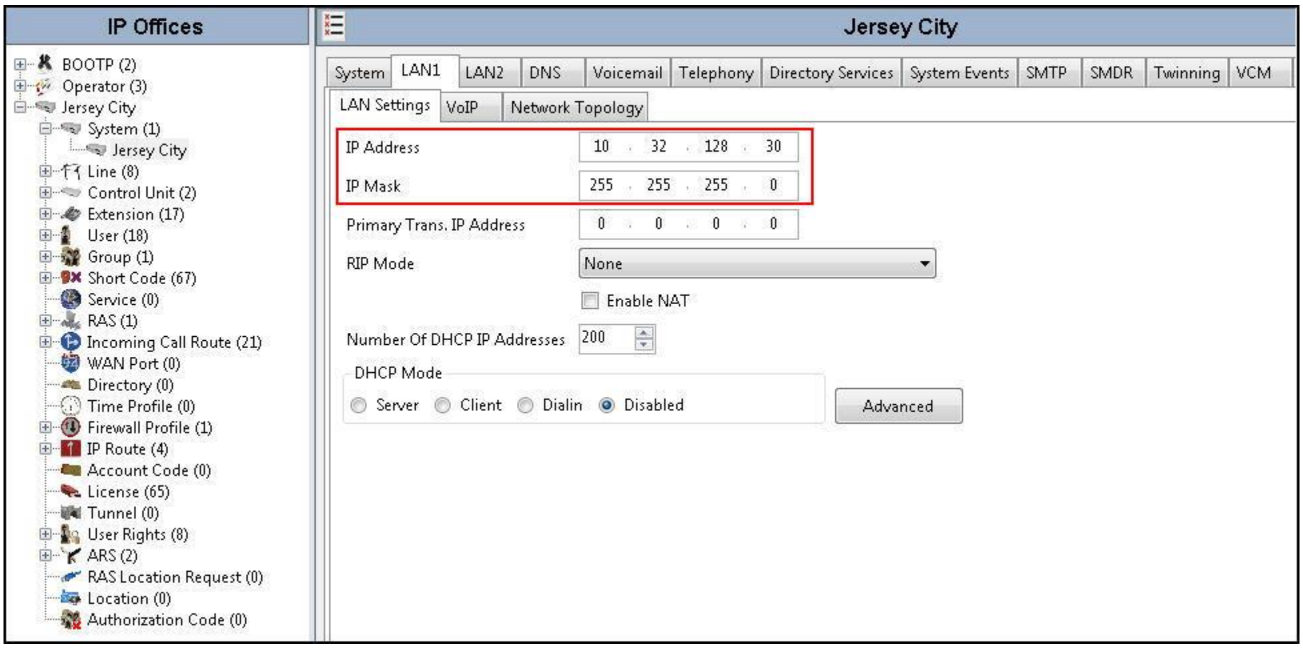


5.2. System

This section configures the necessary system settings.

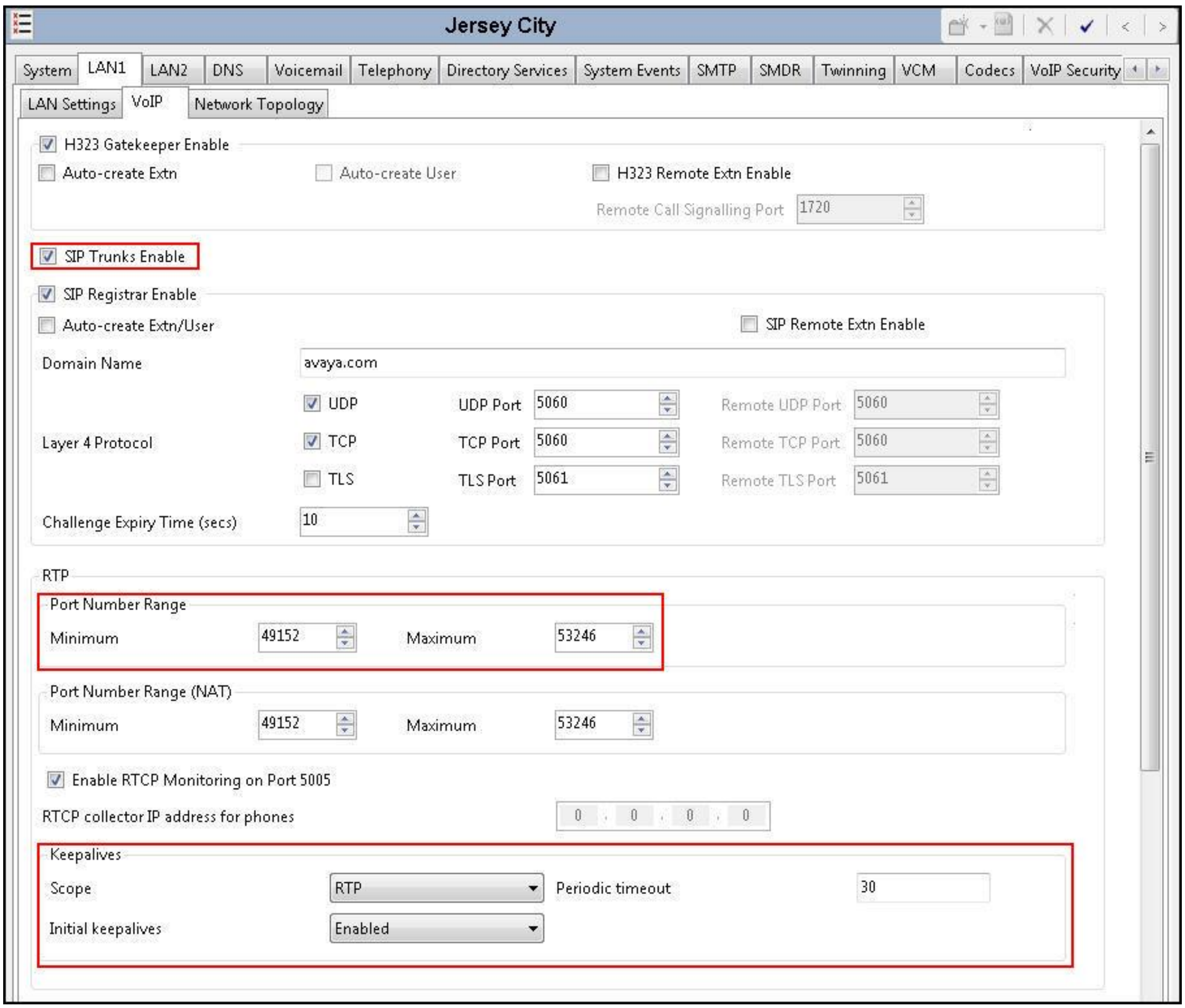
5.2.1. System – LAN1 Tab

In the sample configuration, the Avaya IP Office LAN port was used to connect to the enterprise network. The LAN1 settings correspond to the LAN port on the Avaya IP Office 500 V2. To access the LAN1 settings, first navigate to **System → <*Name*>**, where <*Name*> is the system name assigned to the IP Office. In the case of the compliance test, the system name is **Jersey City**. Next, navigate to the **LAN1 → LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office LAN port. Set the **IP Mask** field to the mask used on the enterprise network.



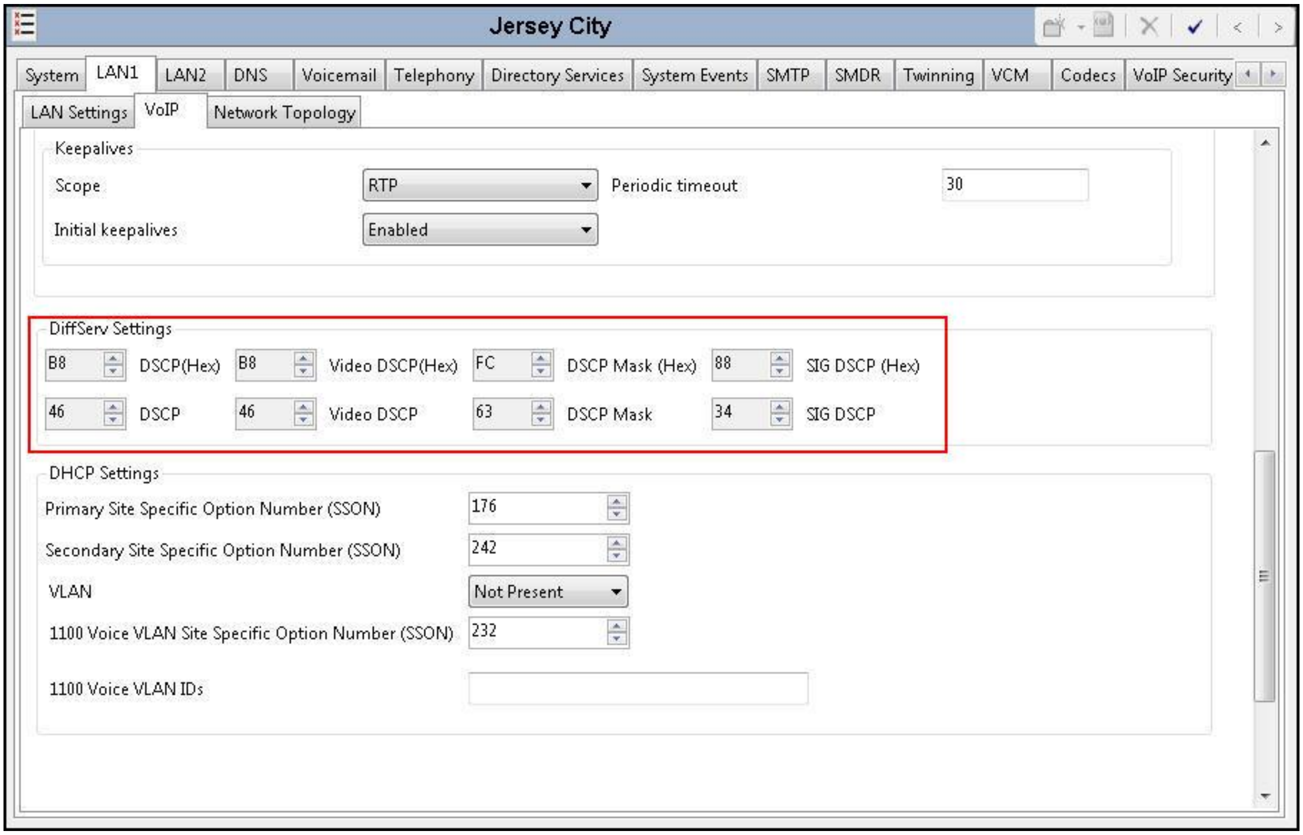
On the **VoIP** tab of LAN1 in the Details Pane, configure the following parameters:

- Check the **SIP Trunks Enable** box to enable the configuration of SIP trunks.
- In the **RTP** section, the **RTP Port Number Range** can be customized to a specific range of receiving ports for the RTP media, as agreed with the service provider. Based on this setting, Avaya IP Office would request RTP media be sent to a port in the configurable range for calls using LAN1.
- In the **Keepalives** section, select ***RTP*** for **Scope**; select ***Enabled*** for **Initial keepalives**; enter ***30*** for **Periodic timeout**. These settings direct IP Office to send a RTP keepalive packet starting at the time of initial connection and every 30 seconds thereafter if no other RTP traffic is present. This facilitates the flow of media in cases where each end of the connection is waiting for media from the other, as well as helping to keep firewall (if used) ports open for the duration of the call.



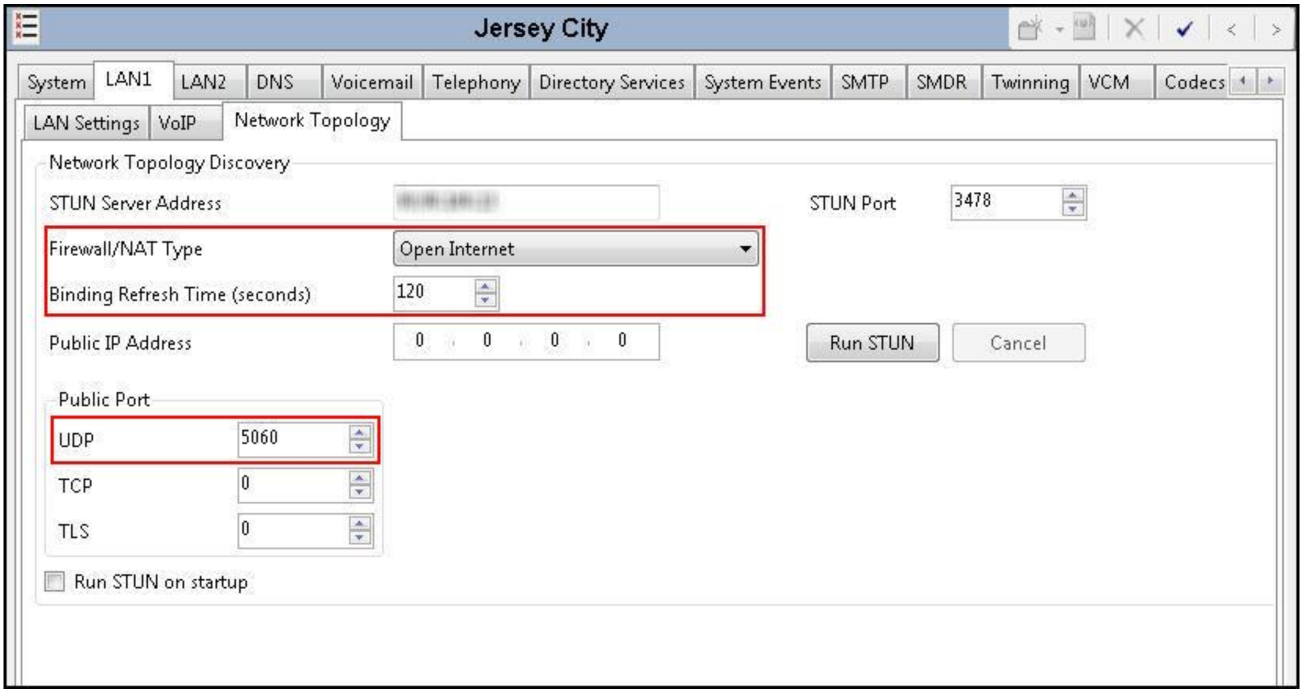
Though not highlighted in the above screen, note the settings for **SIP Registrar Enable**, **Domain Name**, and **Layer 4 Protocol**. These settings are necessary for the IP Office to serve as the SIP Registrar Server for the IP Office SIP endpoints.

Scroll down to the **DiffServ Settings** section. Avaya IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the screen below and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values should be provided by the customer.



On the **Network Topology** tab of LAN1 in the Details Pane, configure the following parameters:

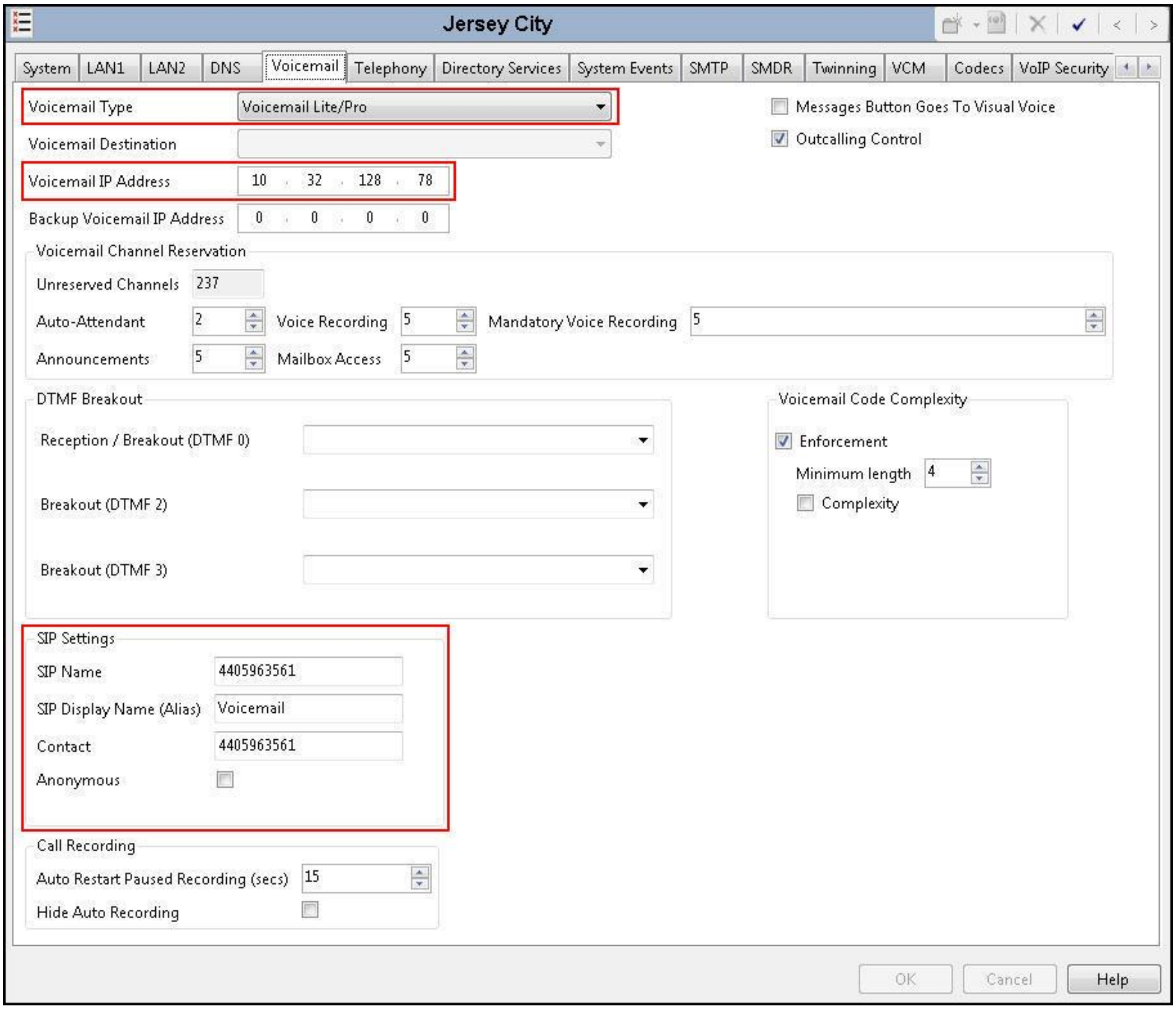
- Select **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***. With the ***Open Internet*** setting, **STUN Server Address** is not used.
- Set **Binding Refresh Time (seconds)** to a desired value. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. See **Section 5.10** for complete details.
- Set **Public Port** to ***5060*** for **UDP**.



5.2.2. System - Voicemail Tab

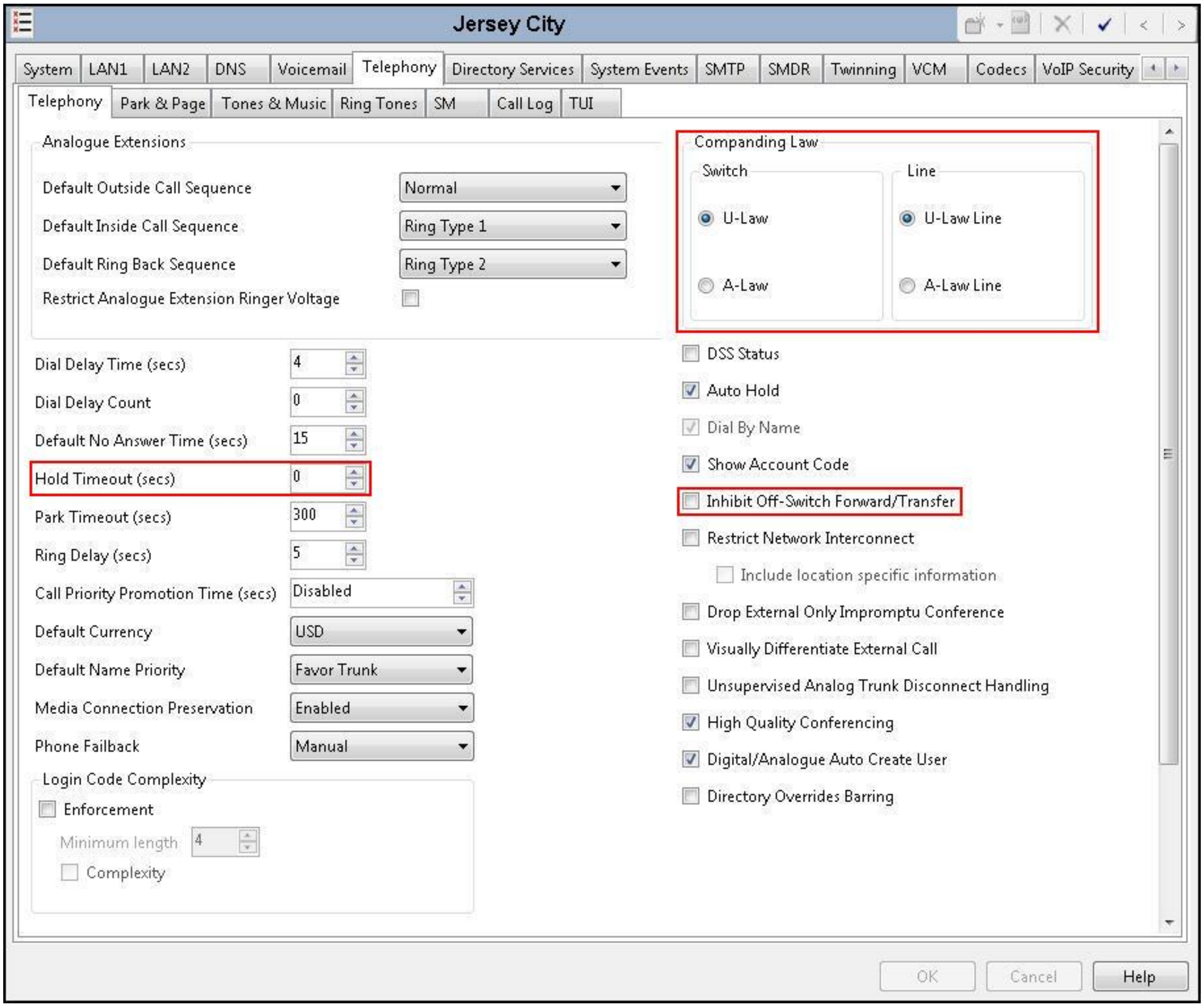
In the **Voicemail** tab of the Details Pane, configure the **SIP Settings** section. The **SIP Name** and **Contact** are set to one of the DID numbers provided by Fusion Connect. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Uncheck the **Anonymous** box to allow the Voicemail Caller ID information to be sent to the network.

Note the selection for **Voicemail Type** and the IP address setting for **Voicemail IP Address**. These are for configuring Voicemail Pro as the voice messaging service for Avaya IP Office users (part of the standard IP Office setup beyond the scope of these Application Notes).



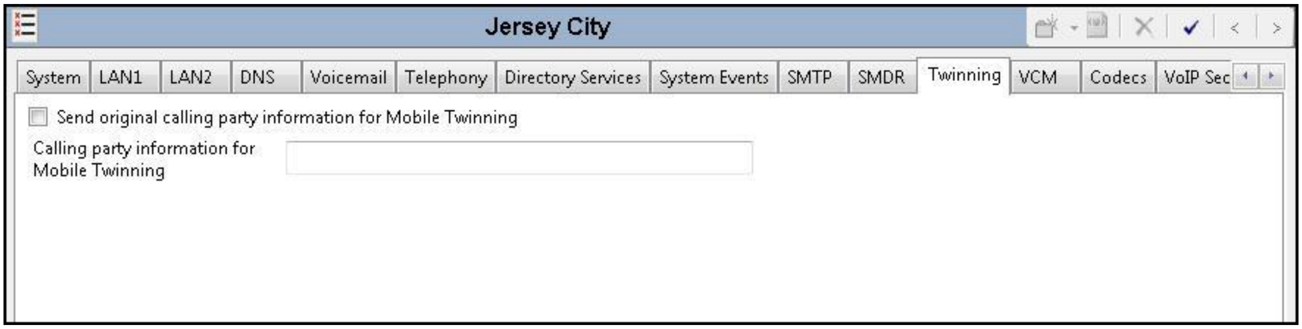
5.2.3. System - Telephony Tab

Navigate to the **Telephony → Telephony** tab in the Details Pane. Enter or select ***0*** for **Hold Timeout (secs)** so that calls on hold will not time out. Choose the **Companding Law** typical for the enterprise site. For the compliance test, ***U-LAW*** was used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider per customer business policies. Note that this configuration might pose a security issue (Toll Fraud). Customers should exercise caution with this configuration.



5.2.4. System - Twinning Tab

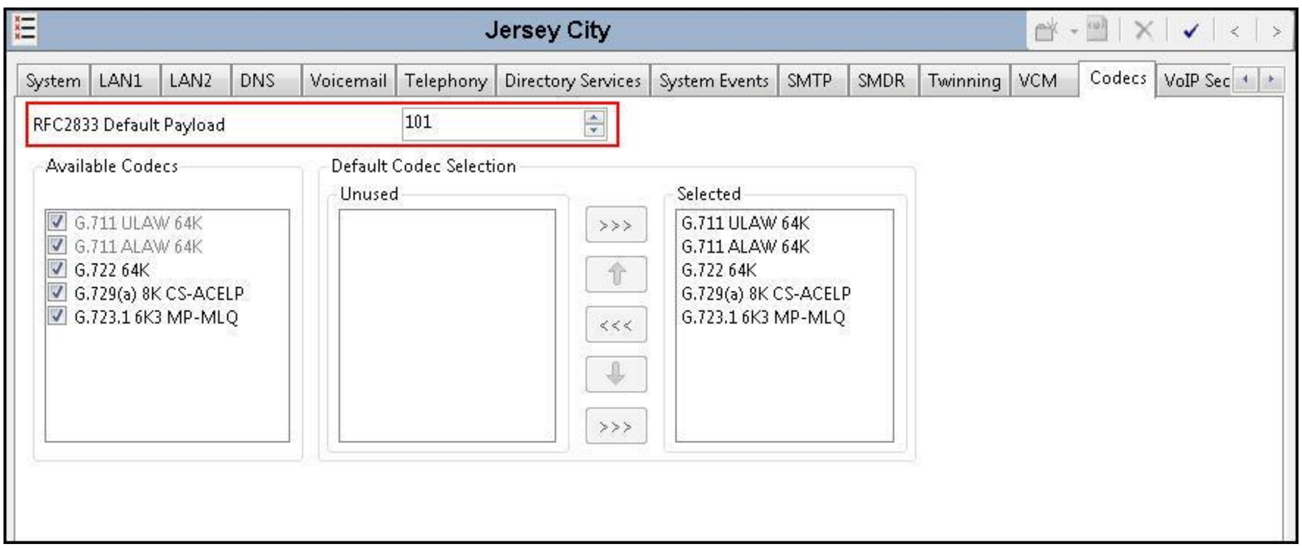
To view or change the System Twinning settings, navigate to the **Twinning** tab in the Details Pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank.



5.2.5. System – Codecs Tab

In the **Codecs** tab of the Details Pane, select or enter ***101*** for **RFC2833 Default Payload**. This setting was preferred by Fusion Connect for use with out-band DTMF tone transmissions.

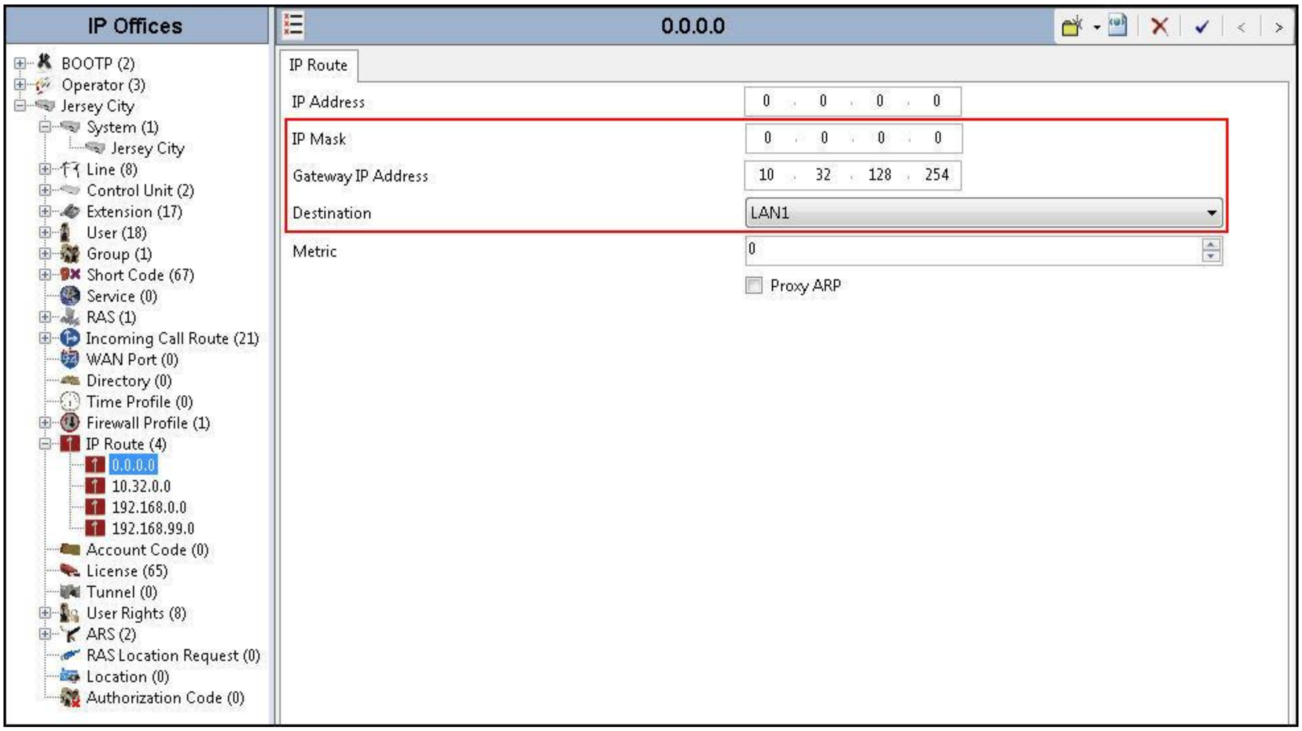
On the left, observe the list of **Available Codecs**. In the screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed. The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis. By default, all IP (SIP and H.323) lines and extensions will assume the system default codec selection, unless configured otherwise for the specific line or extension.



5.3. IP Route

Navigate to **IP Route → 0.0.0.0** in the left Navigation Pane if a default route already exists. Otherwise, to create the default route, right-click on **IP Route** and select **New** (not shown)**.** Create and verify a default route with the following parameters:

- Set **IP Address** and **IP Mask** to ***0.0.0.0***.
- Set **Gateway IP Address** to the IP address of the enterprise LAN gateway for the subnet where the Avaya IP Office is connected.
- Set **Destination** to ***LAN1*** from the drop-down list.



5.4. Administer SIP Line

A SIP Line is needed to establish the SIP connection between Avaya IP Office and the Fusion Connect network. 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 5.4.1** to create the SIP Line from the template.

Note: DevConnect-generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML-format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and IP Office Server Edition systems.

Some items relevant to a specific customer environment are not included in the template associated with these Application Notes, 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 SIP Line **Transport** tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2** through **5.4.8**.

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

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

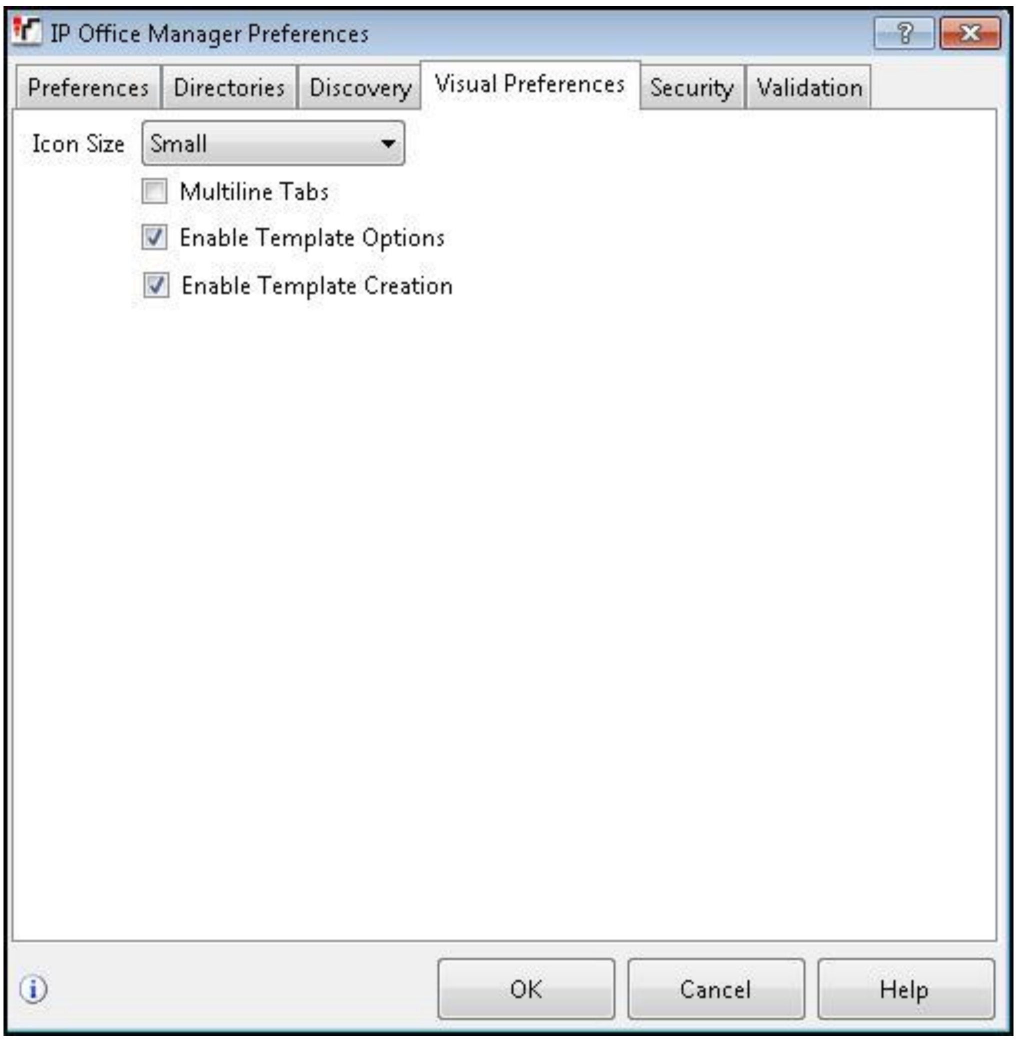
5.4.1. Create SIP Line From Template

1. Copy the template file to a location (e.g., C:\Temp) on the computer where IP Office Manager is installed. Verify that the template file name is

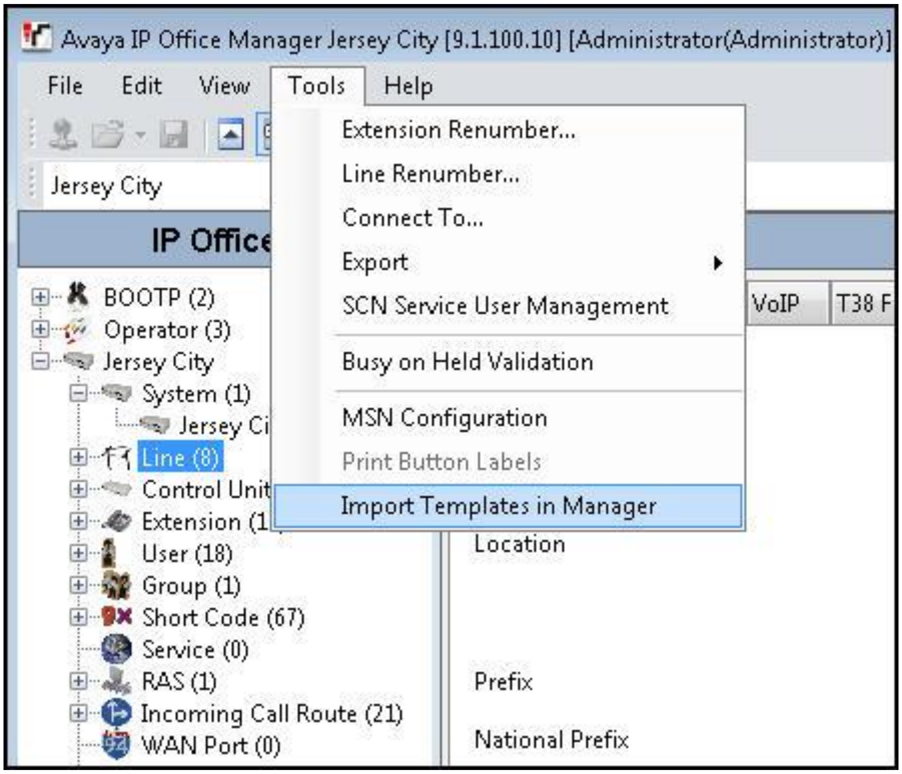
   AF_Fusion Connect_SIPTrunk.xml

   The file name is important in locating the proper template file in **Step 4**.

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 option box is checked next to **Enable Template Options**. Click **OK**.



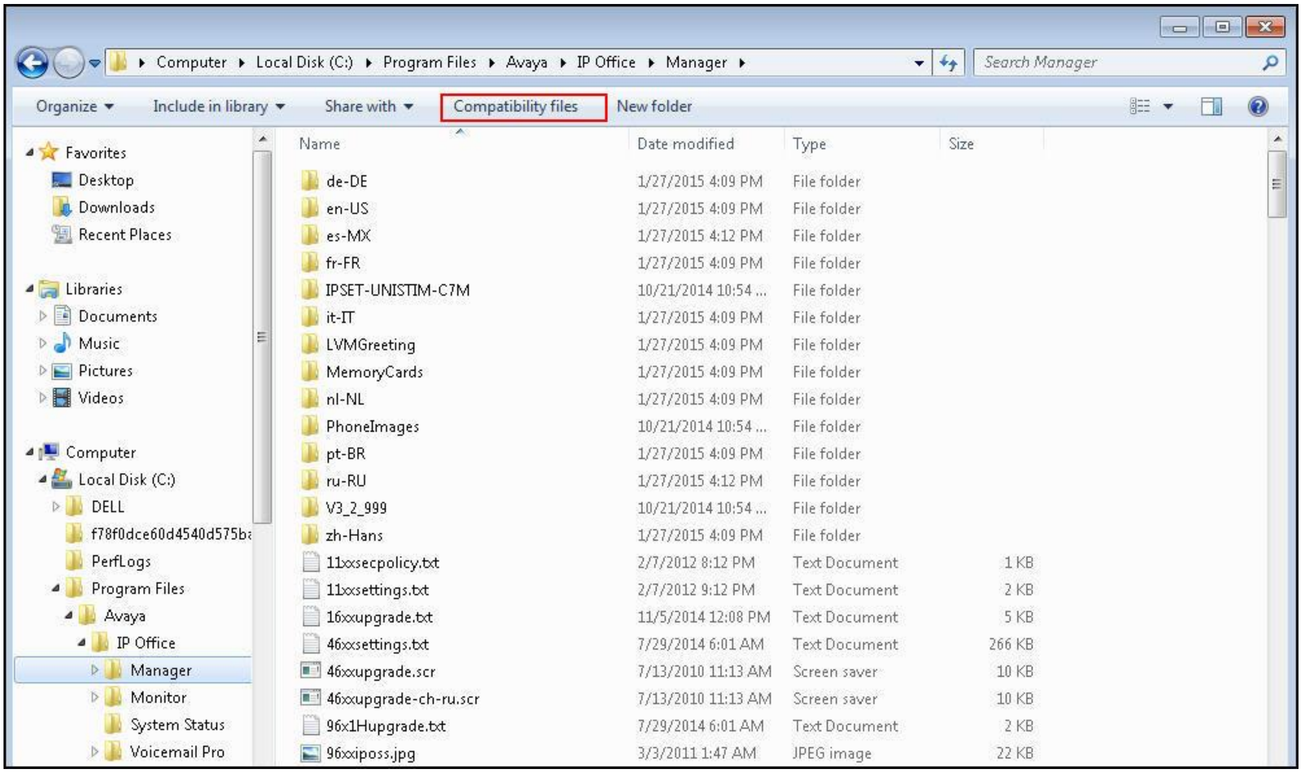
3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 4**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



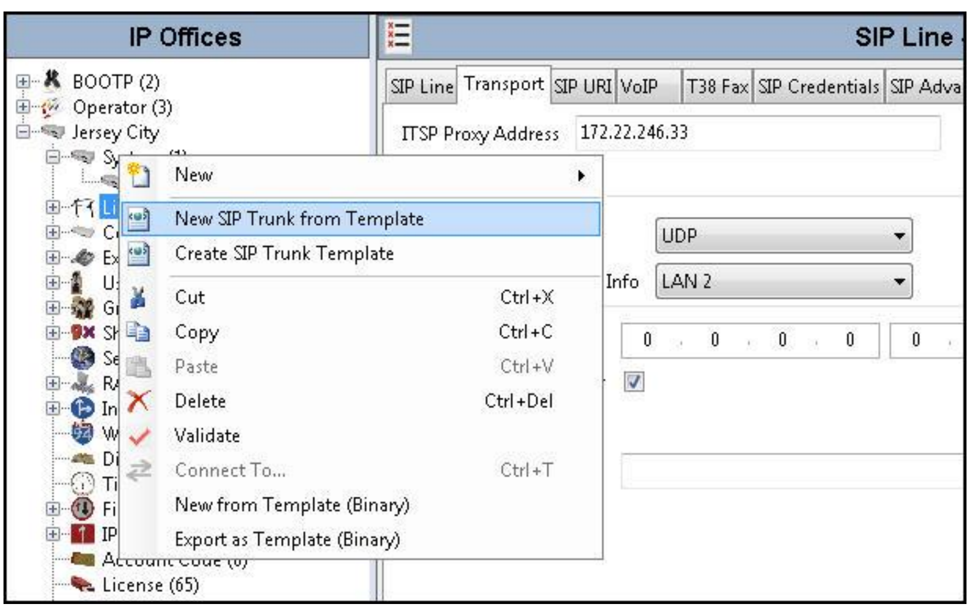
In the pop-up window that appears (not shown), select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue.

If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

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



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



In the subsequent **Template Type Selection** pop-up window, select ***Fusion Connect*** from the **Service Provider** drop-down list as shown below. This selection corresponds to parts of the template file name as specified in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.

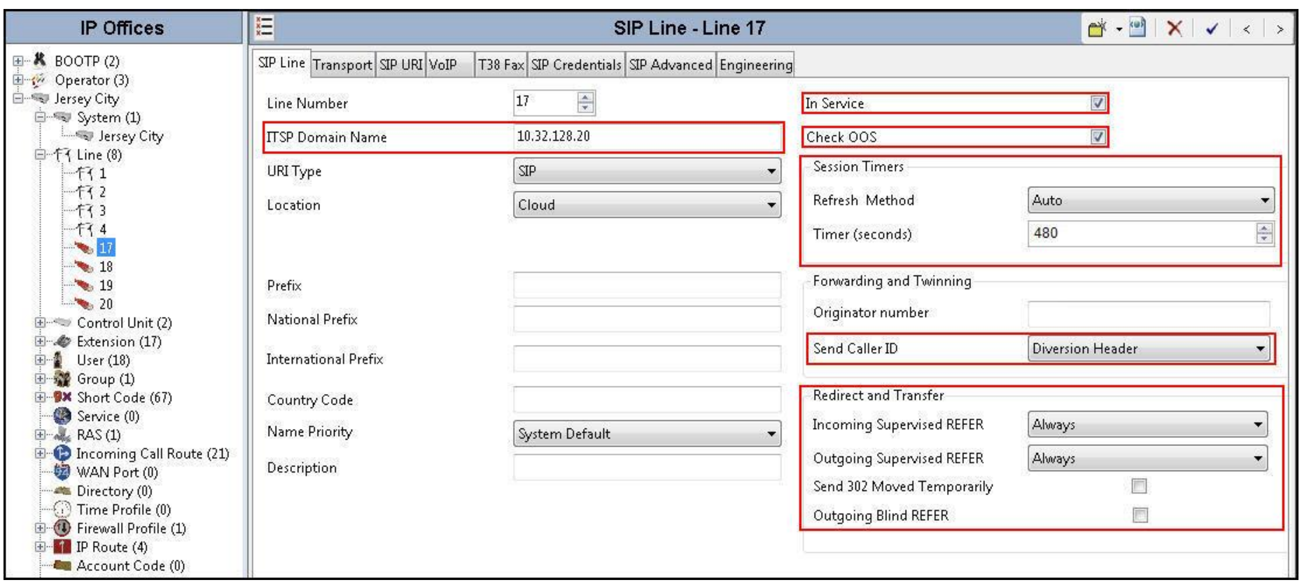
Note that the newly created SIP Line may not immediately appear in the Navigation pane until the configuration was saved, closed and reopened in IP Office Manager.

5. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2** through **5.4.8**.

5.4.2. SIP Line – SIP Line Tab

In the **SIP Line** tab of the Details Pane, configure the parameters as shown below:

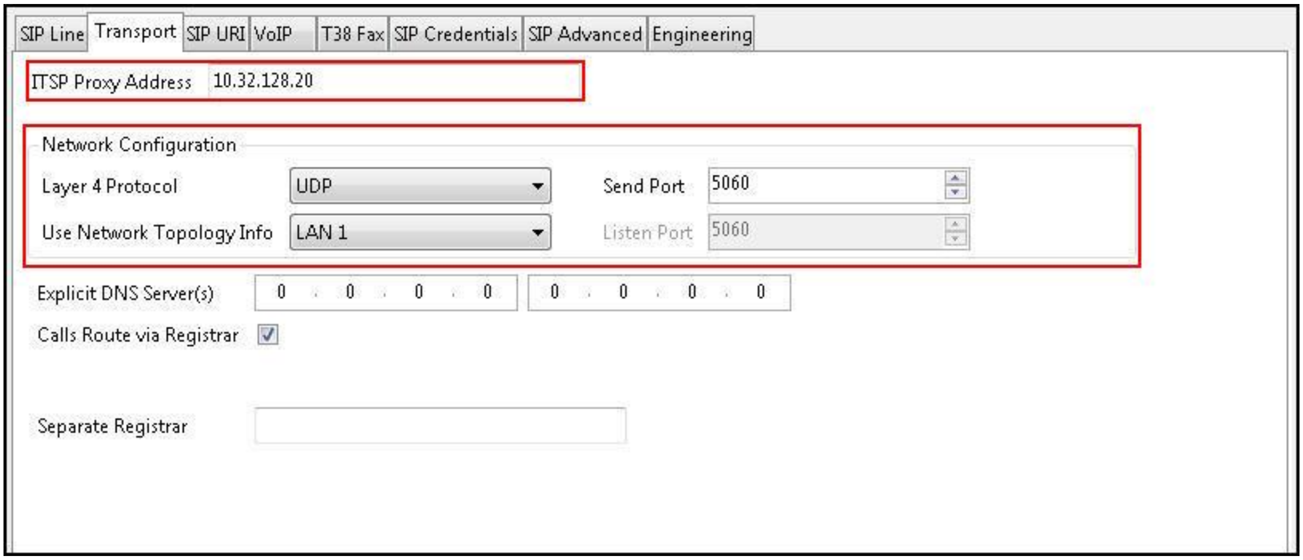
- Set **ITSP Domain Name** to the IP address of the internal signaling interface of the Avaya SBCE.
- Check the **In Service** box.
- Check **OOS** box. Avaya IP Office will check the SIP OPTIONS response from the far end to determine whether to take the SIP Line out of service.
- In the **Session Timers** section, set **Method for Session Refresh** to ***Auto***. With this setting Avaya IP Office will send UPDATE messages for session refresh if the other party supports UPDATE. If UPDATE is not supported, re-INVITE messages are sent. Set **Timer (seconds)** to a desired value. Avaya IP Office will send out session refresh UPDATE or re-INVITE at the specified intervals (half of the specified value).
- Set **Send Caller ID** under **Forwarding and Twinning** to ***Diversion Header***. With this setting and the related configuration in **Section 5.2.4**, Avaya IP Office will include the Diversion Header for calls that are redirected via Mobile Twinning out the SIP Line to the PSTN. It will also include the Diversion Header for calls that are forwarded out the SIP Line.
- Under Redirect and Transfer, select ***Always*** for **Incoming Supervised REFER** and **Outgoing Supervised REFER**. Fusion Connect supports use of the REFER method for off-net call transfer.



5.4.3. SIP Line – Transport Tab

Navigate to the **Transport** tab and set the following:

- Set the **ITSP Proxy Address** to the IP address of the internal signaling interface of the Avaya SBCE.
- Set the **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to the network port used by the SIP line to access the far-end as configured in **Section 5.2.1**.
- Set the **Send Port** to ***5060***.



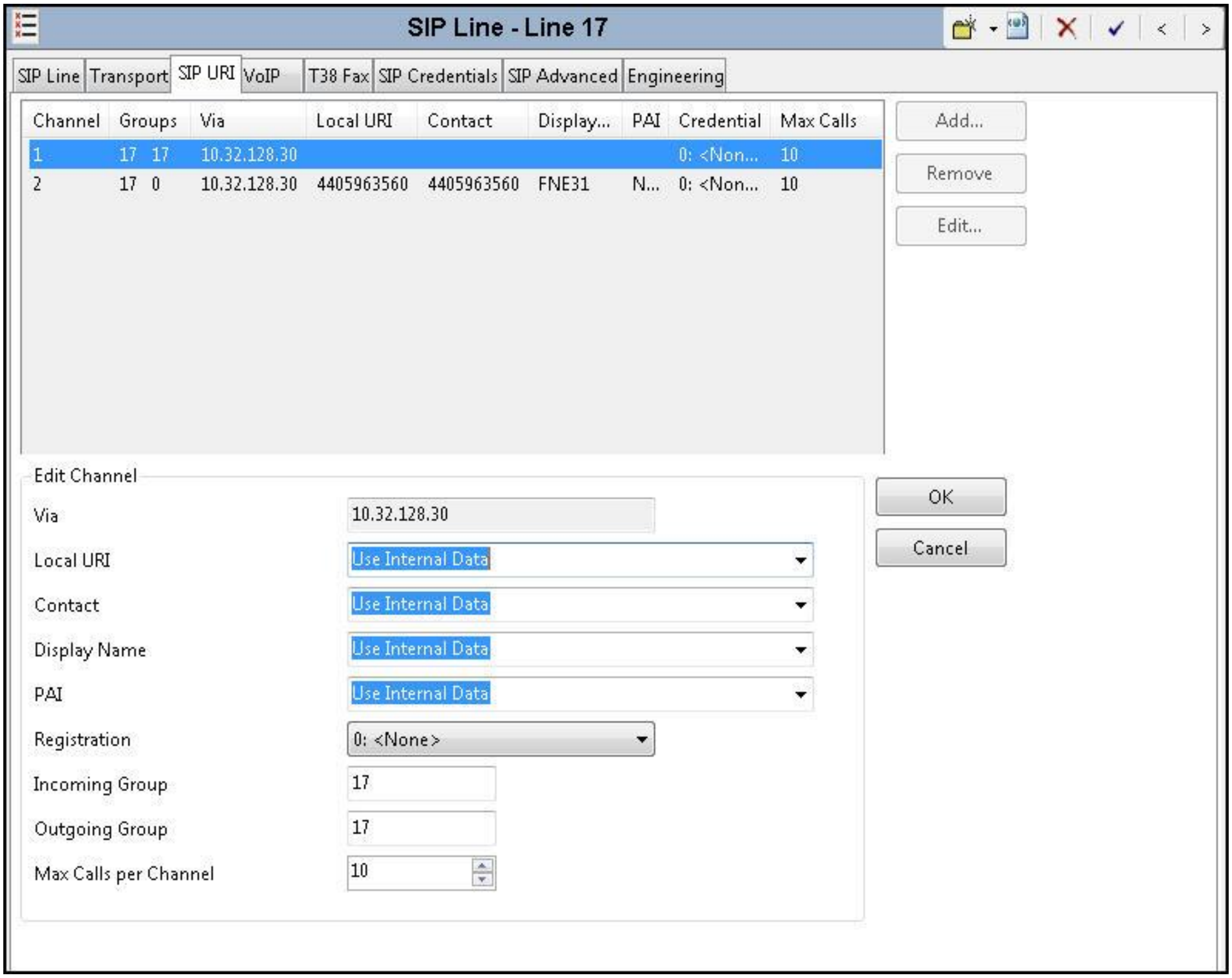
5.4.4. SIP Line – SIP URI Tab

Select the **SIP URI** tab to create or edit a SIP URI entry. A SIP URI entry matches each incoming number that Avaya IP Office will accept on this line. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. For the compliance test, a single SIP URI entry was created to match any DID number assigned to Avaya IP Office users. The following screen shows the edit window on this URI entry for the compliance test.

- Set **Local URI** to ***Use Internal Data***. This setting allows calls on this line whose SIP URI matches the **SIP Name** set on the **SIP** tab of any **User** as shown in **Section 5.6**, or the **SIP Name** set in the **SIP Settings** area of the System **Voicemail** tab as shown in **Section 5.2.2**.
- Set **Contact** and **Display Name** to ***Use Internal Data***. This setting will cause the Contact and Display Name data for outbound messages to be set from the corresponding fields on the **SIP** tab of the individual **User** as shown in **Section 5.66**.
- Set **PAI** to ***Use Internal Data***. This setting directs Avaya IP Office to send the PAI (P-Asserted-Identity) header when appropriate. The PAI header will be populated from the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.66**.
- Select ***0: <None>*** for **Registration**.
- Associate this line with an incoming line group by entering line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes

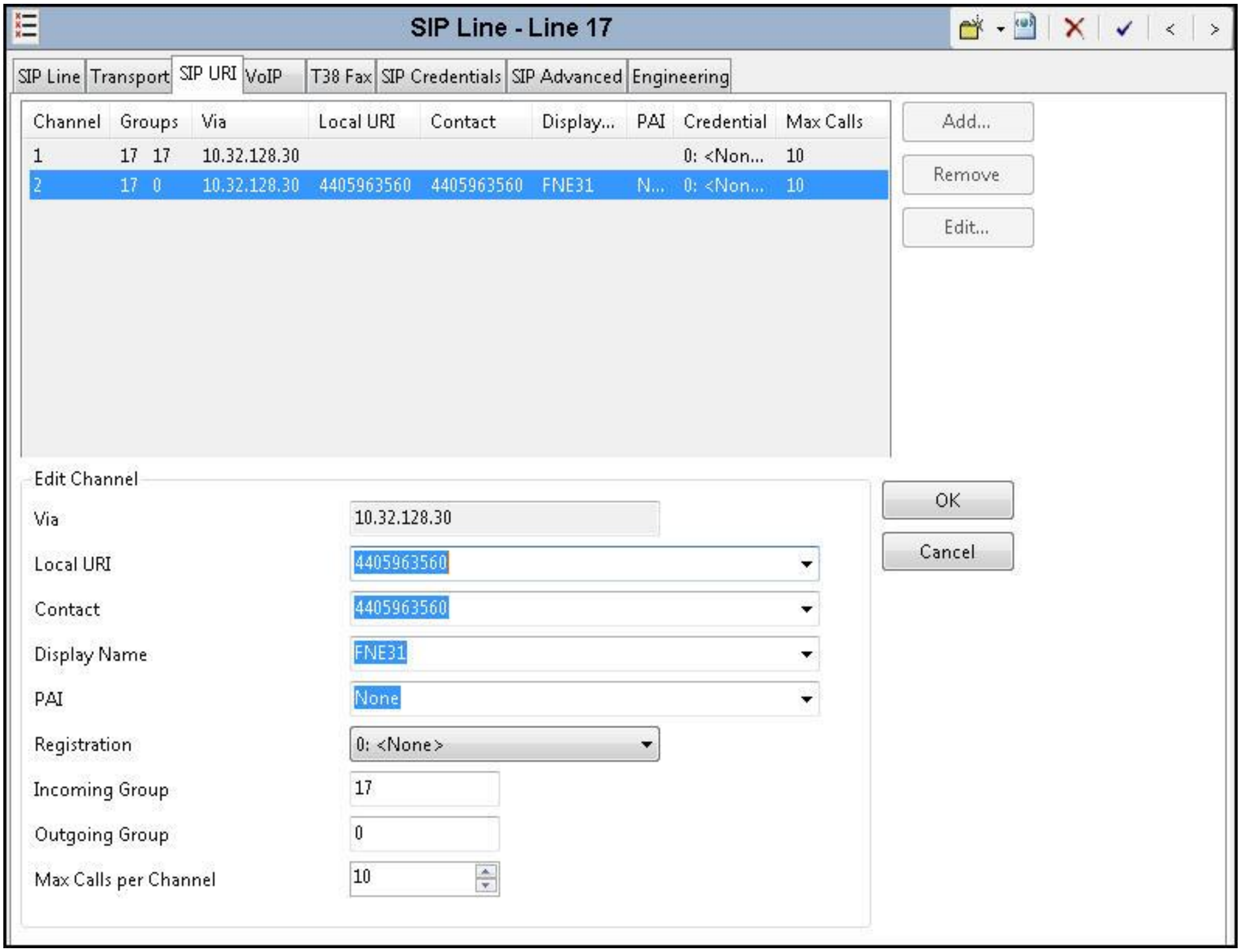
for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. For the compliance test, the incoming and outgoing group ***17*** was specified. Note that this group number can be different than the SIP Line number.

- Set **Max Calls per Channel** to the number of simultaneous SIP calls allowed using this SIP URI pattern.



The screen below shows the edit window for the pre-configured SIP URI entry for matching inbound calls to the Mobile Call Control application (see **Section 5.9**). This entry was necessary since the DID number assigned to the Mobile Call Control application was not configured elsewhere for matching the incoming call Request URI. Without this SIP URI entry, the Avaya IP Office would have responded to an incoming call to the DID meant for the Mobile Call Control application with a "404 Not Found" status message and the call would have failed.

The DID ***4405963560*** will be configured in the Incoming Call Route in **Section** Error! Reference source not found. to deliver the call to the Mobile Call Control application.

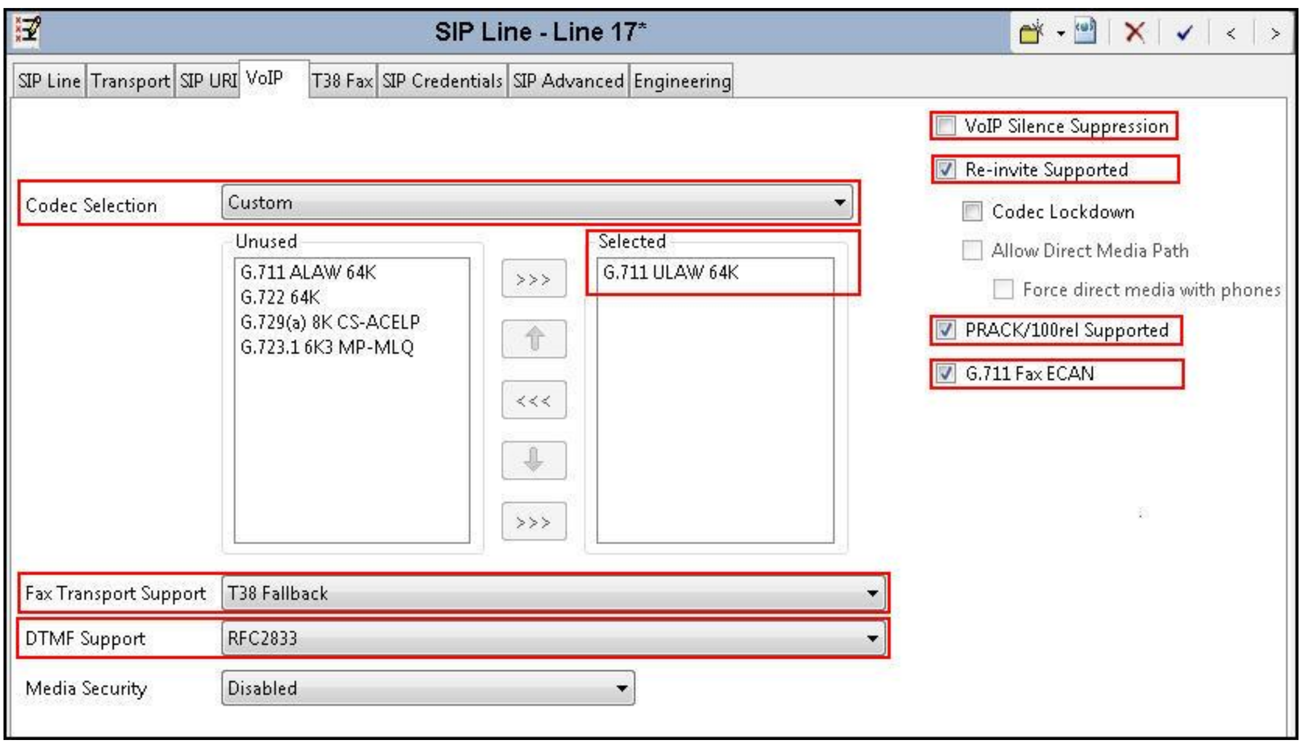


Note that a ***0*** setting means no line group number was configured for **Outgoing Group** for this SIP URI entry. This is because this SIP URI entry is used only for inbound calls to the Mobile Call Control application.

5.4.5. SIP Line – VoIP Tab

Select the **VoIP** tab. Set the parameters as shown below.

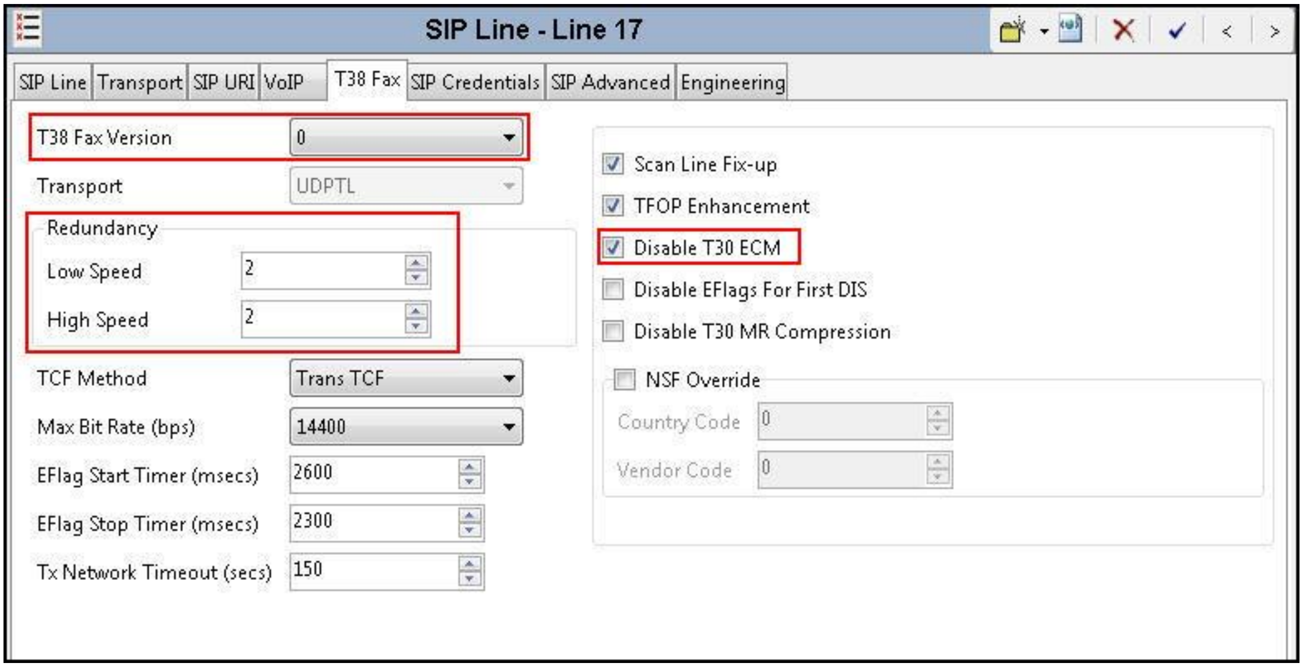
- Select ***Custom*** for **Codec Selection**.
- Choose ***G.711 ULAW 64K*** or ***G.729(a) 8K CS-ACELP*** from the **Unused** box and move the selection to the **Selected** box. Fusion Connect supports both G.711u and G.729a codecs, but customer must inform Fusion Connect about the one preferred codec to use so that the codec configuration on the service side can match the enterprise. See the item **Supported Codecs** in the observation/limitation list in **Section 2.2** for more details. The screen below shows configuration of G.711u as the preferred codec.
- Select ***T38 Fallback*** for **Fax Transport Support** to direct Avaya IP Office to use T.38 for fax calls and use G.711u pass-through for fax if the remote end does not support T.38.
- Select ***RFC2833*** for **DTMF Support**. This directs Avaya IP Office to send DTMF tones as out-band RTP events as per RFC2833.
- Uncheck the **VoIP Silence Suppression** option box.
- Check the **Re-invite Supported** option box. When enabled, re-INVITE can be used during a call session to change the characteristics of the session including codec re-negotiation.
- Check the **PRACK/100rel Supported** option box. This setting enables support by Avaya IP Office for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.
- Check **G.711 Fax ECAN**.



5.4.6. SIP Line – T38 Fax

The settings on this tab configures T.38 fax parameters and are only accessible if **Re-invite Supported** was checked and either ***T38*** or ***T38 Fallback*** was selected for **Fax Transport Support** in the **VoIP** tab in **Section 5.4.5**.

The screen below shows the settings used for the compliance test. The **T38 Fax Version** is set to ***0***. In the **Redundancy** area, **Low Speed** and **High Speed** are set to ***2***. The **Disable T30 ECM** must be checked or fax errors may be experienced when using T.38 Fax. When selected, it disables the T.30 Error Correction Mode used for fax transmission. All other values are left at default.



5.4.7. SIP Line – SIP Credentials Tab

SIP Credentials are used to register the SIP Trunk with a service provider that requires SIP Registration. SIP Credentials are also used to provide the required information for Digest Authentication of outbound calls. SIP Credentials are unique per customer and therefore customers must contact the service provider to obtain the proper registration and/or Digest Authentication credentials for their deployment.

For this compliance test, Fusion Connect configured the test circuit as a static trunk that did not require trunk registration or Digest Authentication for outbound calls. Therefore, this tab did not need to be visited.

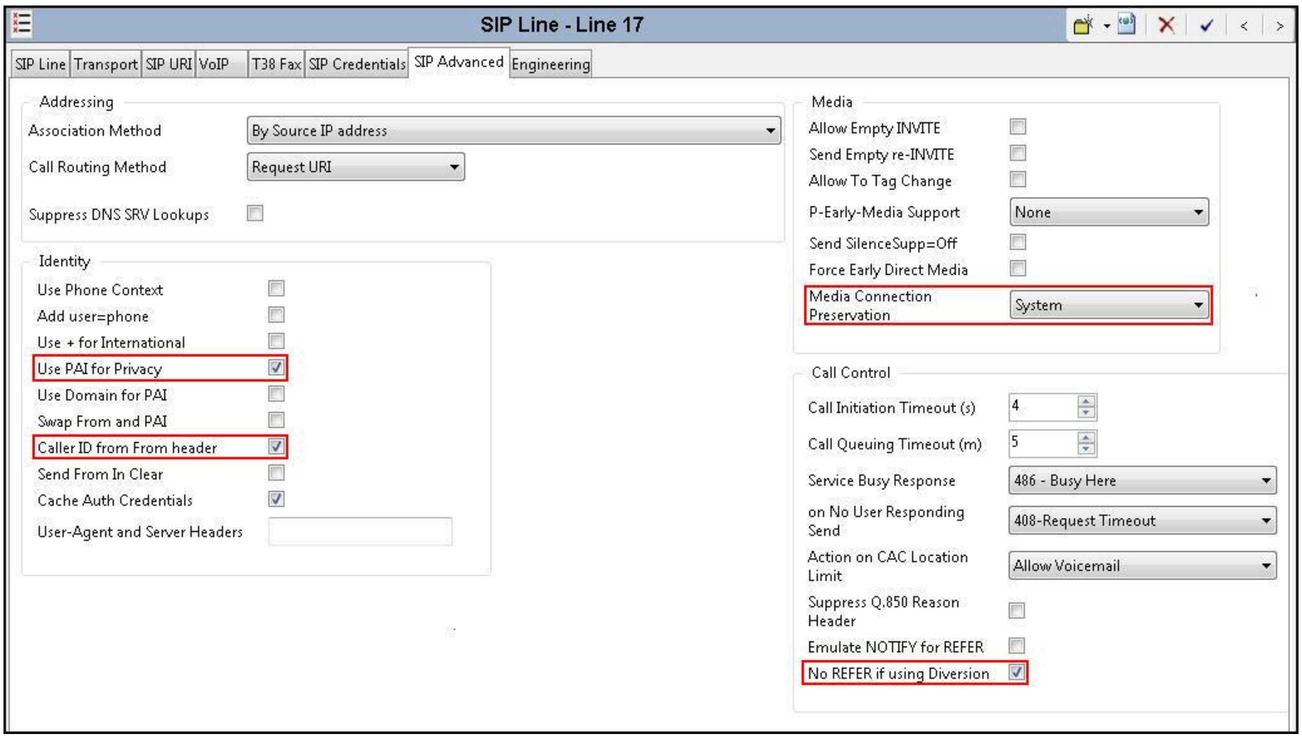
5.4.8. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab to configure advanced SIP Line parameters.

In the **Identity** area, the **Use PAI for Privacy** box is checked for Avaya IP Office to use the P-Asserted-Identity (PAI) SIP header for privacy-requested outbound calls. With this configuration, Avaya IP Office will populate the From and Contact headers of the anonymous outbound call INVITE with "anonymous" as the URI user part, but include the normal calling user information in the PAI header. The **Caller ID from From header** box is checked for Avaya IP Office to use the Caller ID information in the From SIP header rather than the PAI or Contact SIP header for inbound calls.

In the **Media** area, select ***System*** for **Media Connection Preservation** to allow established calls to continue despite brief network failures.

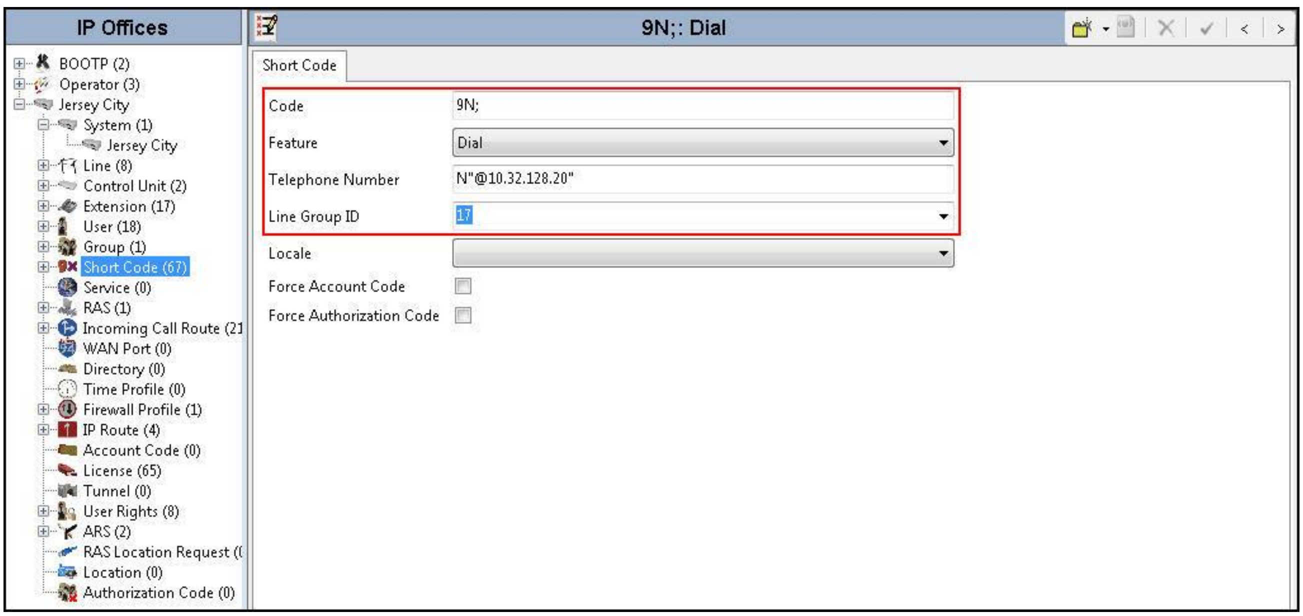
In the **Call Controll** area, **No REFER if using Diversion** is checked to prevent Avaya IP Office from using the SIP REFER method on call scenarios that use the Diversion SIP header (e.g., off-net call forward or outbound call to mobile twinning number).



5.5. Short Code

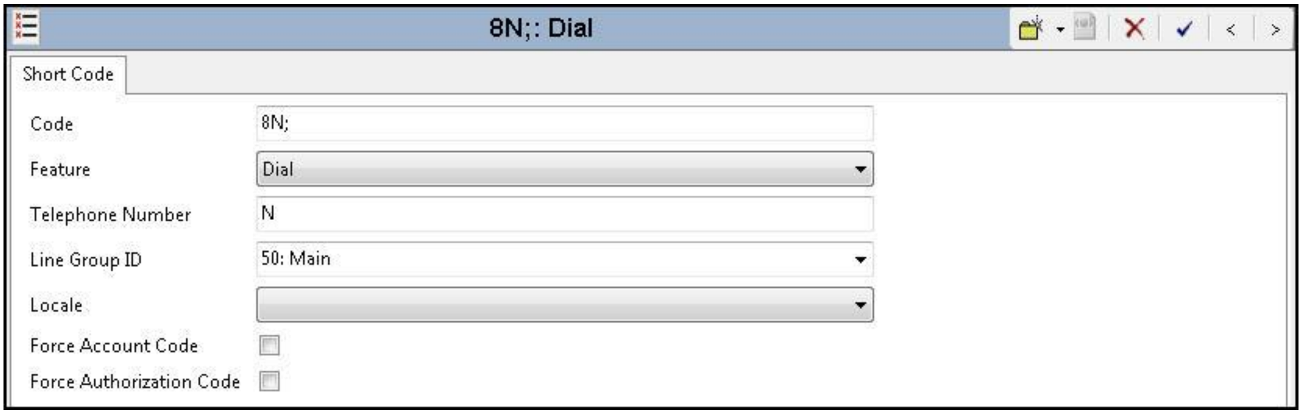
Define a short code to route outbound calls to the SIP Line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters as shown below:

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. The ***9N;*** short code, used for the compliance test, will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to ***N"@10.32.128.20"***. This field is used to construct the Request URI and the To header in the outgoing SIP INVITE message. The value ***N*** represents the number dialed by the user. The IP address following the **@** sign is the IP address of the private interface of the Avaya SBCE.
- Set the **Line Group Id** to the **Outgoing Group** number defined on the **SIP URI** tab of the SIP Line in **Section 5.4.4**. This short code will use this line group when placing outbound calls.

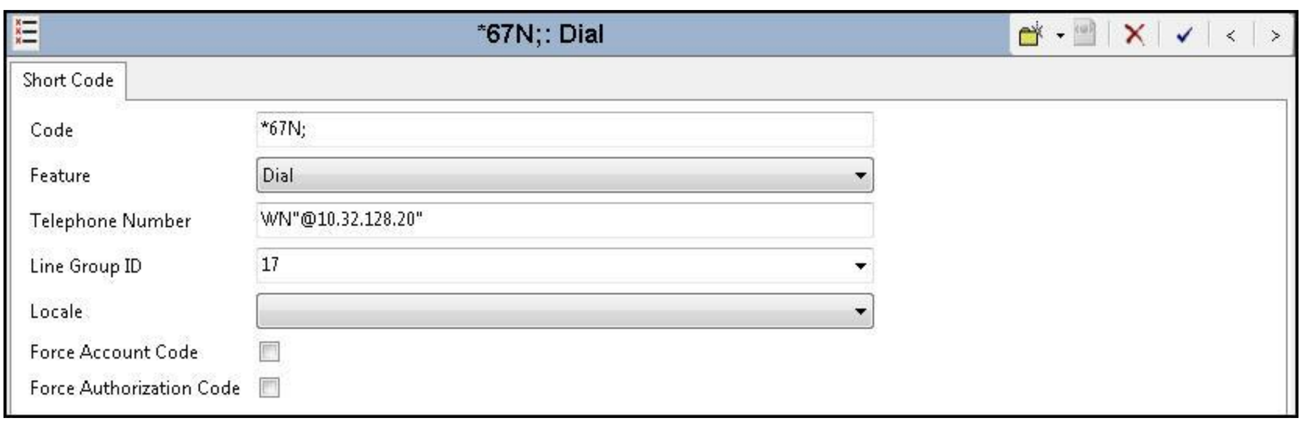


The simple **9N;** short code illustrated above does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the dialed digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used.

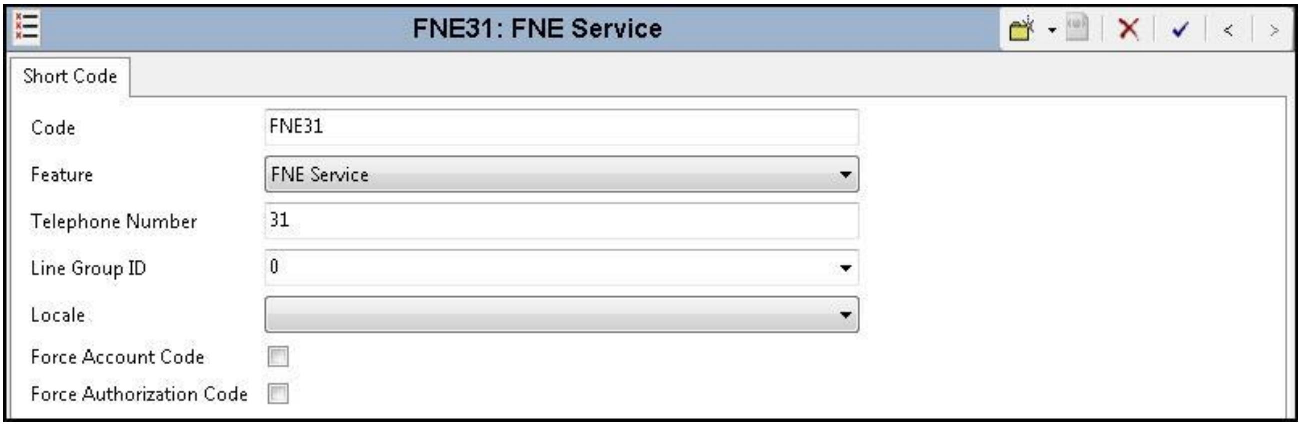
In the screen below, the short code ***8N;*** is illustrated for access to ARS. When the Avaya IP Office user dials 8 plus any number ***N***, rather than being directed to a specific **Line Group ID**, the call is directed to ***50: Main***, configurable via ARS. See **Section 5.8** for example ARS route configuration.



Optionally, add or edit a short code used to access the SIP Line anonymously. In the screen shown below, the short code ****67N;*** is illustrated. This short code is similar to the **9N;** short code except that the **Telephone Number** field begins with the letter ***W***, which means "withhold the outgoing calling line identification". In the case of the compliance test, when a user dialed *67 plus the destination number, Avaya IP Office would include the user's telephone number (DID number assigned to the user) in the **P-Asserted-Identity** (PAI) header, populate the URI user part with "anonymous" in the From and Contact headers, and include the **Privacy: id** header in the outbound INVITE message. Consequently Fusion Connect would prevent presentation of the caller id to the called PSTN destination.



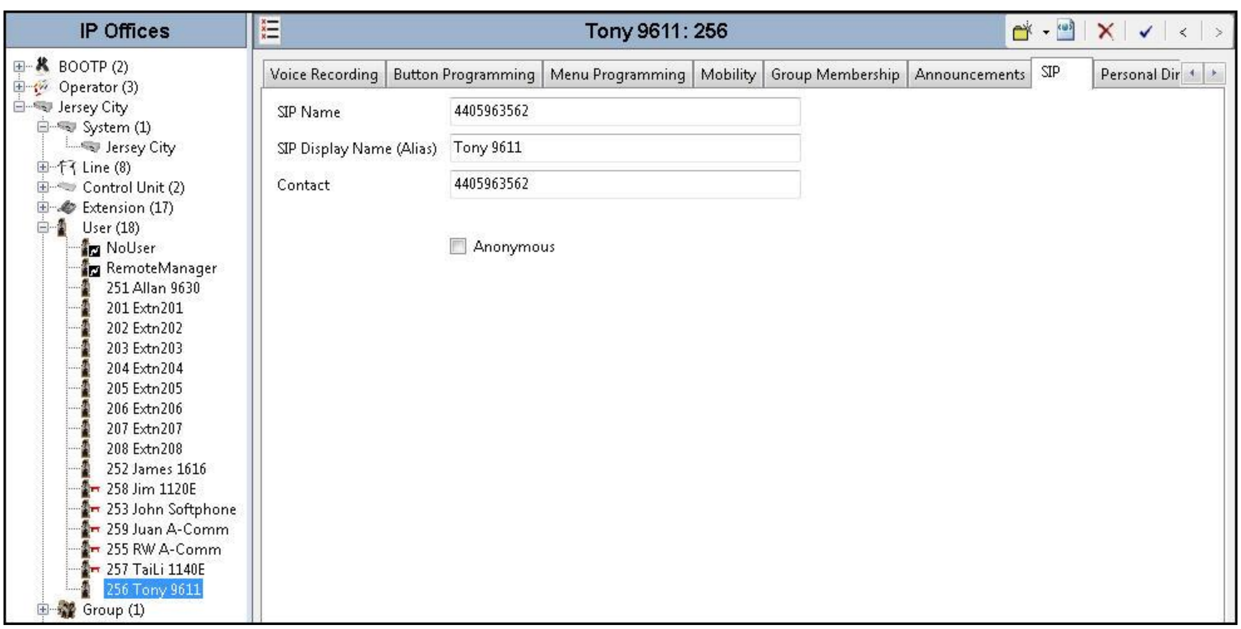
For completeness, the short code ***FNE31*** for the Mobile Call Control application is shown below. See **Section 5.7** for routing incoming call to this application to receive internal IP Office dial tones. See **Section 5.9** for configuration to enable this mobility feature.



5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line. To configure these settings, first navigate to **User→Name** in the Navigation Pane, where **Name** is the name of the user to be modified. In the example below, the name of the user is “Tony 9611” at extension 256. Select the **SIP** tab in the Details Pane. The **SIP Name** and **Contact** are set to one of the DID numbers provided by Fusion Connect. The **SIP Display Name (Alias)** can optionally be configured with a descriptive text string. The value entered for the **Contact** field will be used in the Contact header for outgoing SIP INVITE to the service provider. The value entered for the **SIP Name** is used as the user part of the SIP URI in the From header for outgoing SIP INVITE.

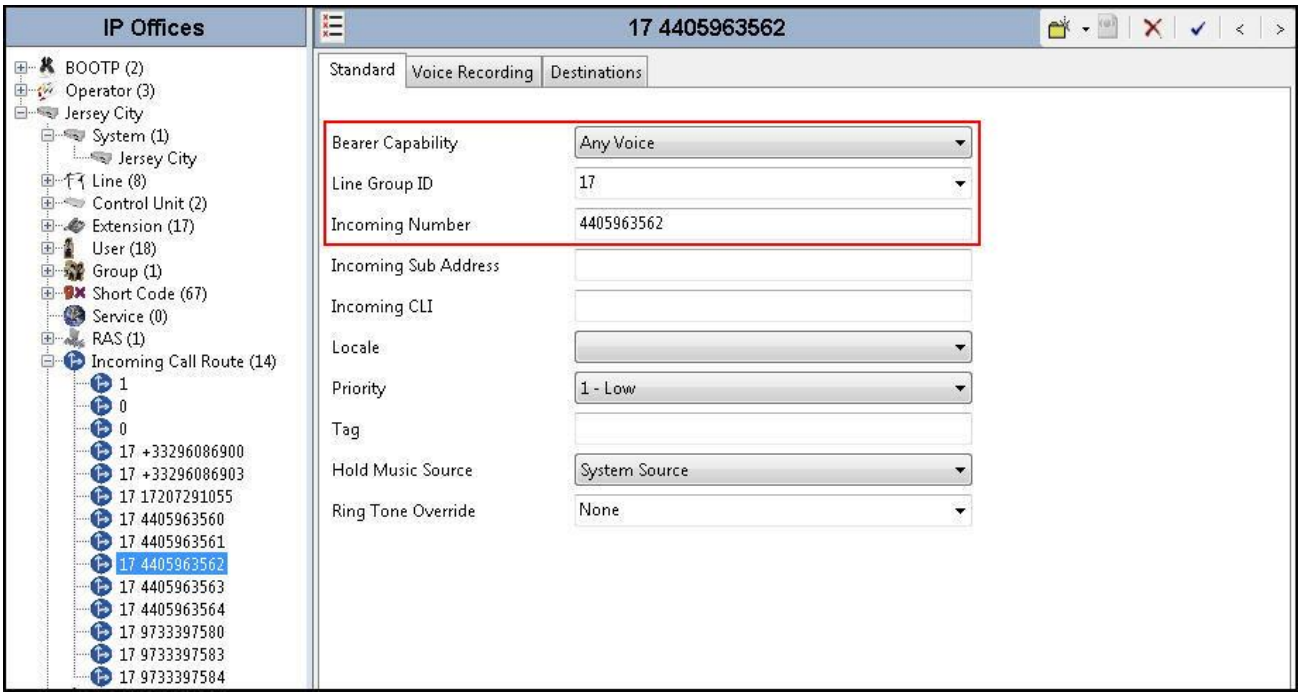
If outbound calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user information from the network (or alternatively use the ***67N;** short code as defined in **Section 5.5**).



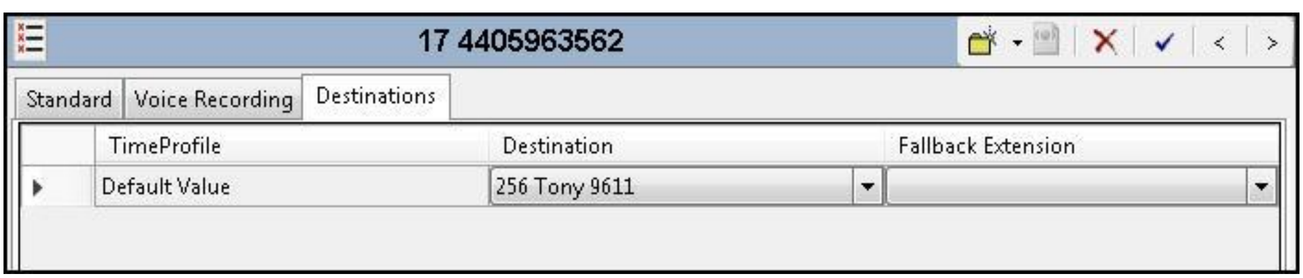
5.7. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal destination. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, right-click **Incoming Call Route** in the Navigation Pane and select **New** (not shown). On the **Standard** tab in the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to ***Any Voice***.
- Set the **Line Group Id** to the **Incoming Group** of the SIP Line defined in **Section 5.4.4**.
- Set the **Incoming Number** to the incoming number on which this route should match.



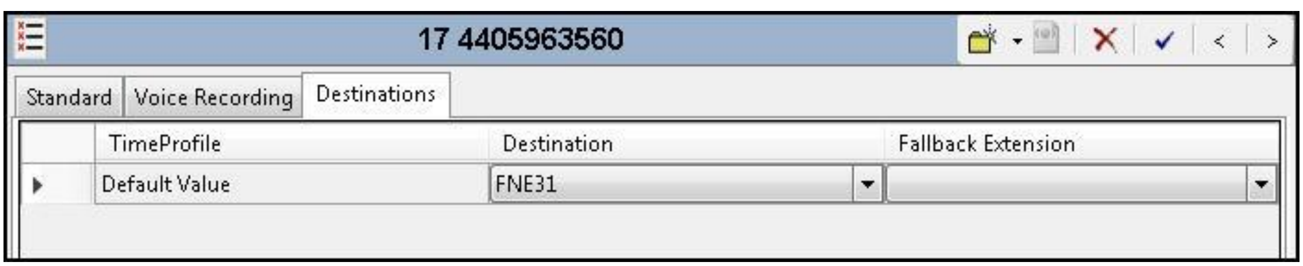
On the **Destinations** tab, select the destination from the pull-down list of the **Destination** field. In this example, incoming calls to 4405963562 on Incoming Group 17 are to be routed to the user "Tony 9611" at extension 256.



The screen below shows calls routed to the IP Office fax endpoint which is an analog extension (Extn 208).

The screen below shows calls routed to IP Office Voicemail Pro for message retrieval. Note that the DID 4405963561 was assigned to Voicemail in **Section 5.2.2**.

The following **Destinations** tab for an incoming call route contains the **Destination** "FNE31" entered manually. The name "FNE31" is the short code for accessing the Mobile Call Control application. An incoming call to 4405963560 from an IP Office user's twinned mobile phone will be delivered directly to an internal dial tone from the Avaya IP Office, allowing the caller to dial call destinations, both internal and external. See **Section 5.9** on configuration to enable the Mobile Call Control application.



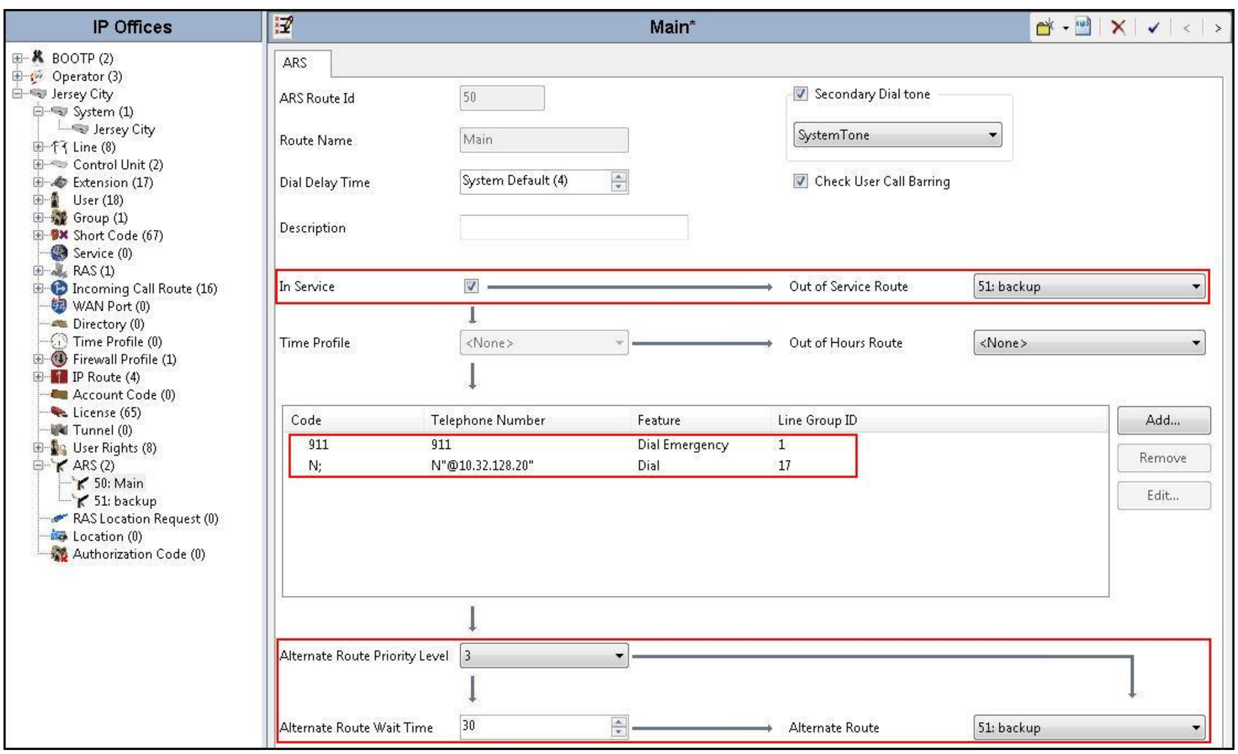
5.8. ARS and Alternate Routing

While detailed coverage of Automatic Route Selection (ARS) is beyond the scope of these Application Notes, this section includes basic ARS screen illustration and considerations. ARS is shown here mainly to illustrate alternate routing should the SIP Line be out of service or temporarily not responding.

Optionally, ARS can be used to supplement or replace the simple **9N;** short code approach documented in **Section 5.5**. With ARS, secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, 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. ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all local and long distance calls should use the SIP Line, but service numbers should prefer a different outgoing line group, ARS can be used to distinguish between the two call patterns.

To add a new ARS route, right-click **ARS** in the Navigation Pane and select **New** (not shown). To view or edit an existing ARS route, expand ARS in the Navigation Pane and select a route name.

The following screen shows an example ARS configuration for the route named ***50: Main***. The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter. IP Office short codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service Route and the means to manually activate the Out of Service Route can be helpful for scheduled maintenance or other known service-affecting events for the primary route.

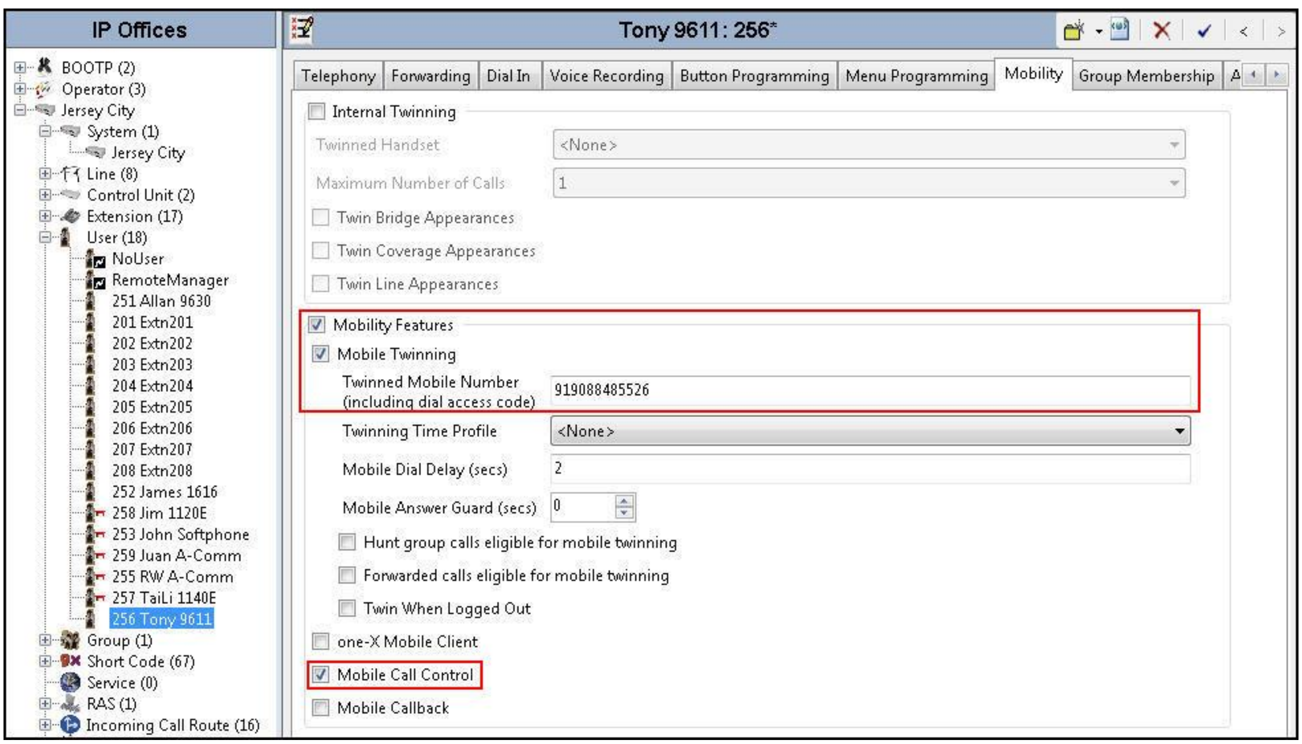


Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., **8N;** in **Section 5.5**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 8 plus any number, the processing for the short code **8N;** would direct the call via ARS to Line Group 17. A short code **911** can be configured to send the emergency call out using Line Group 1 when the user dials "911". If the primary route cannot be used, the call can automatically route to the route name specified in the **Alternate Route** field in the lower right of the screen (***51: Backup***). Since alternate routing is considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority, configured in the **User** tab of individual users, to the value in the **Alternate Route Priority Level** field.

5.9. Mobility

With Mobility configured for an Avaya IP Office user, an inbound call routed to this user automatically triggers an outbound all to the configured Mobile Twinning number for this user.

The following screen shows the **Mobility** tab for User "Tony 9611" at extension 256. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number for the twinned mobile telephone including the dial access code (short code), in this case ***919088485526*** (short code 9 plus the ensuing twinned mobile number). The **Mobile Call Control** option box is also checked so that an inbound call from the twinned mobile number (9088485526 in this example) to the Mobile Call Control application (see Incoming Call Route to "FNE31" in **Section 5.7**) will be delivered directly to an internal dial tone from the Avaya IP Office, allowing the caller to perform further dialing actions including making calls and activating Short Codes. Other options can be set according to customer requirements.



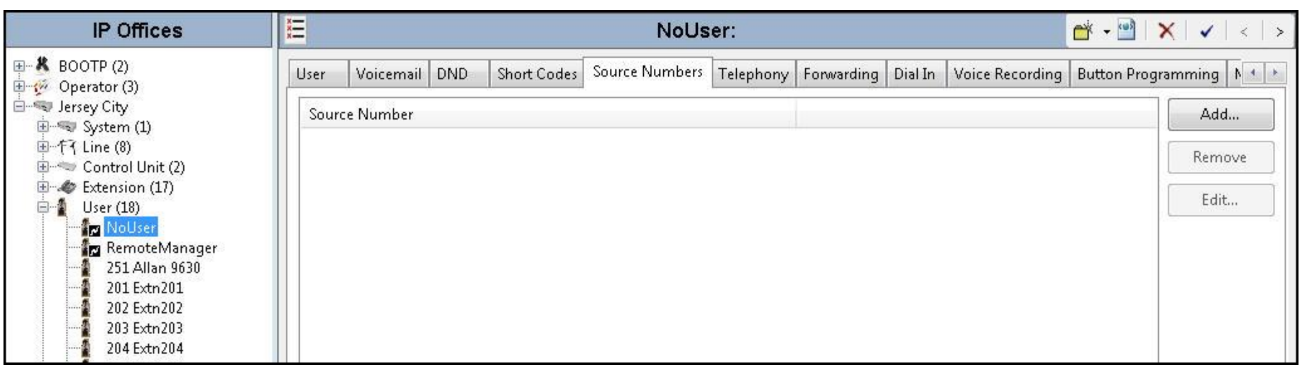
Note that when an inbound call is from the twinned mobile number to the Mobile Call Control application, the caller ID contained in the From header of the incoming INVITE must match the twinned mobile number (without the leading short code digit and the PSTN access code 1 for the North American Numbering Plan), otherwise the Avaya IP Office responds with a “486 Busy Here” message and the caller will hear busy tones.

5.10. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. By default, Avaya IP Office Release 9.1 sends out OPTIONS every 300 seconds. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2.1** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **noUser** user. The OPTIONS period is determined in the following manner:

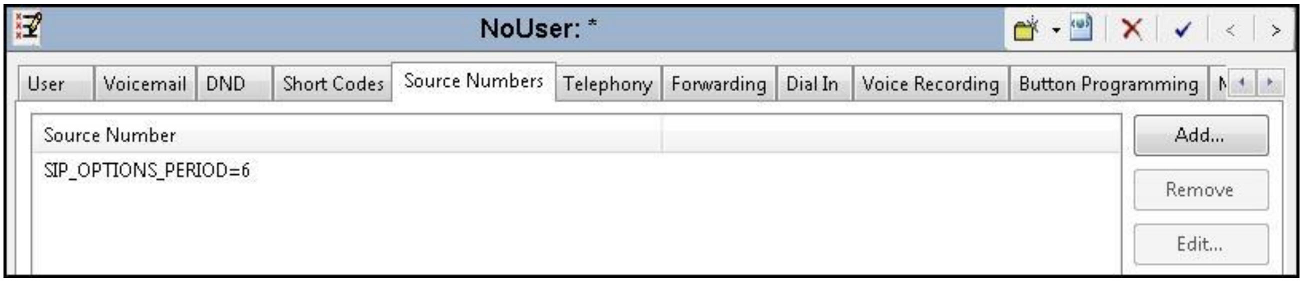
- To use the default value, set **Binding Refresh Time** to 300. OPTIONS will be sent at the 300 second frequency.
- To establish a period of less than 300 seconds, do not define the **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message period will be equal to the **Binding Refresh Time** setting.
- To establish a period greater than 300 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 seconds. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD** settings.

To configure the **SIP_OPTIONS_PERIOD** parameter, 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_OPTIONS_PERIOD=X***, where ***X*** is the desired value in minutes. Click **OK**.

The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. Click **OK** at the bottom of the screen (not shown).

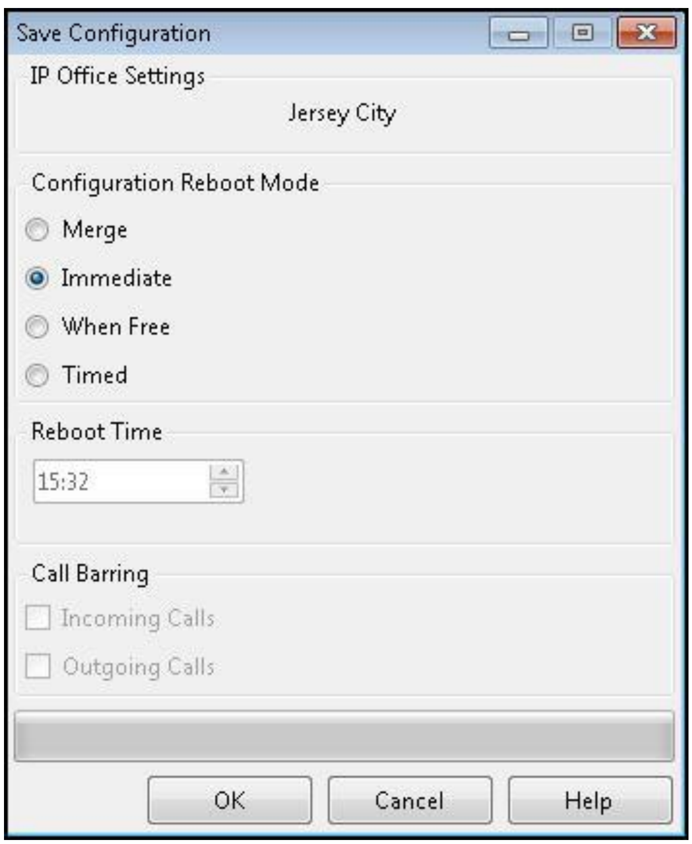


For the compliance test, an OPTIONS period of 2 minutes was desired. The **Binding Refresh Time** was set to ***120*** seconds in **Section 5.2.1**. Thus, there was no need to define **SIP_OPTIONS_PERIOD**.

5.11. 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.

The following **Save Configuration** screen will appear, with either **Merge** or **Immediate** automatically selected, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a system reboot or a service disruption. Click **OK** to proceed.



6. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Avaya SBCE. It is assumed that the initial installation of the Avaya SBCE has been completed, including the assignment of a management IP address. The management interface **must** be provisioned on a different subnet than either the Avaya SBCE private or public network interfaces (i.e., A1 and B1). If the management interface has not been configured on a separate subnet, then contact your Avaya representative for guidance in correcting the configuration.

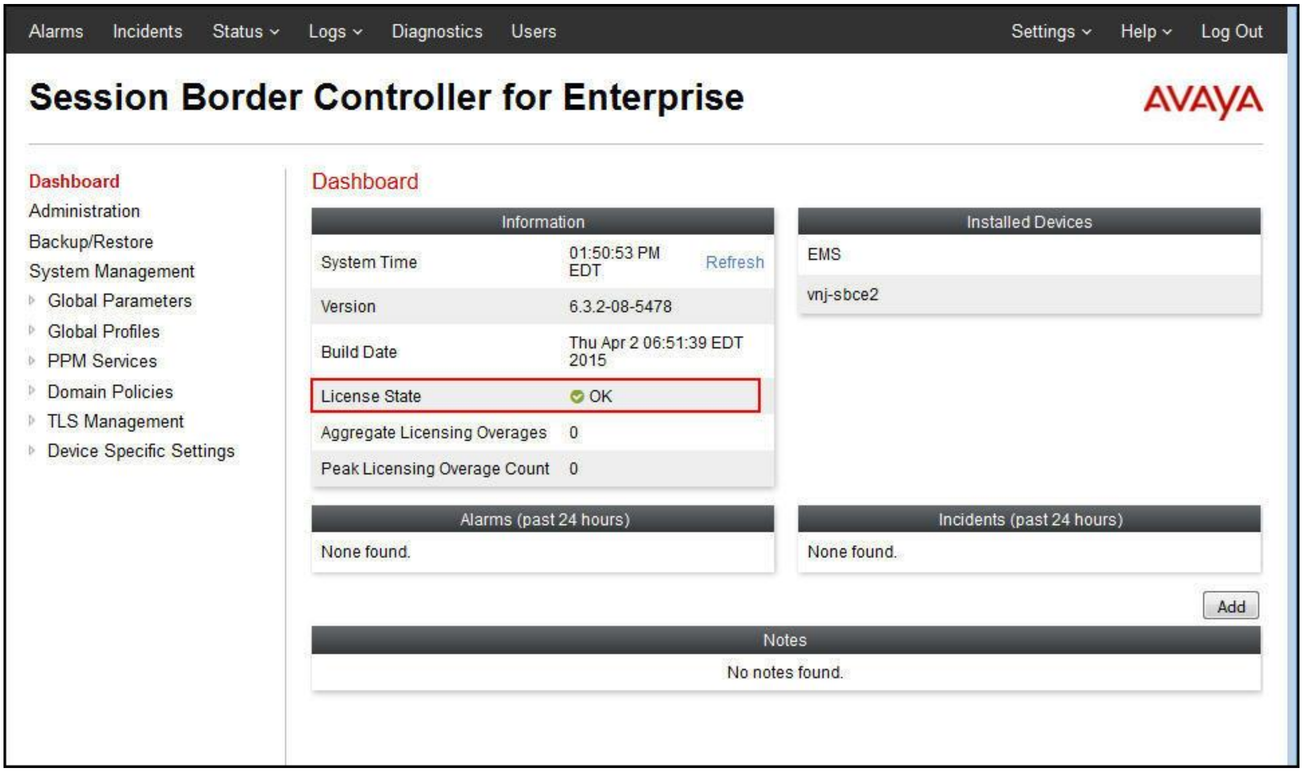
On all screens described in this section, it is to be assumed that parameters are left at their default values unless specified otherwise.

6.1. Access Management Interface

Use a web browser to access the web interface by entering the URL **https://<ip-addr>**, where **<ip-addr>** is the management IP address assigned during installation. The Avaya SBCE login page will appear as shown below. Log in with the appropriate credentials.

After logging in, the Dashboard screen will appear as shown below. Verify that **License State** is **OK** as highlighted. The Avaya SBCE will only operate for a short time without a valid license. Contact your Avaya representative to obtain a license if necessary.

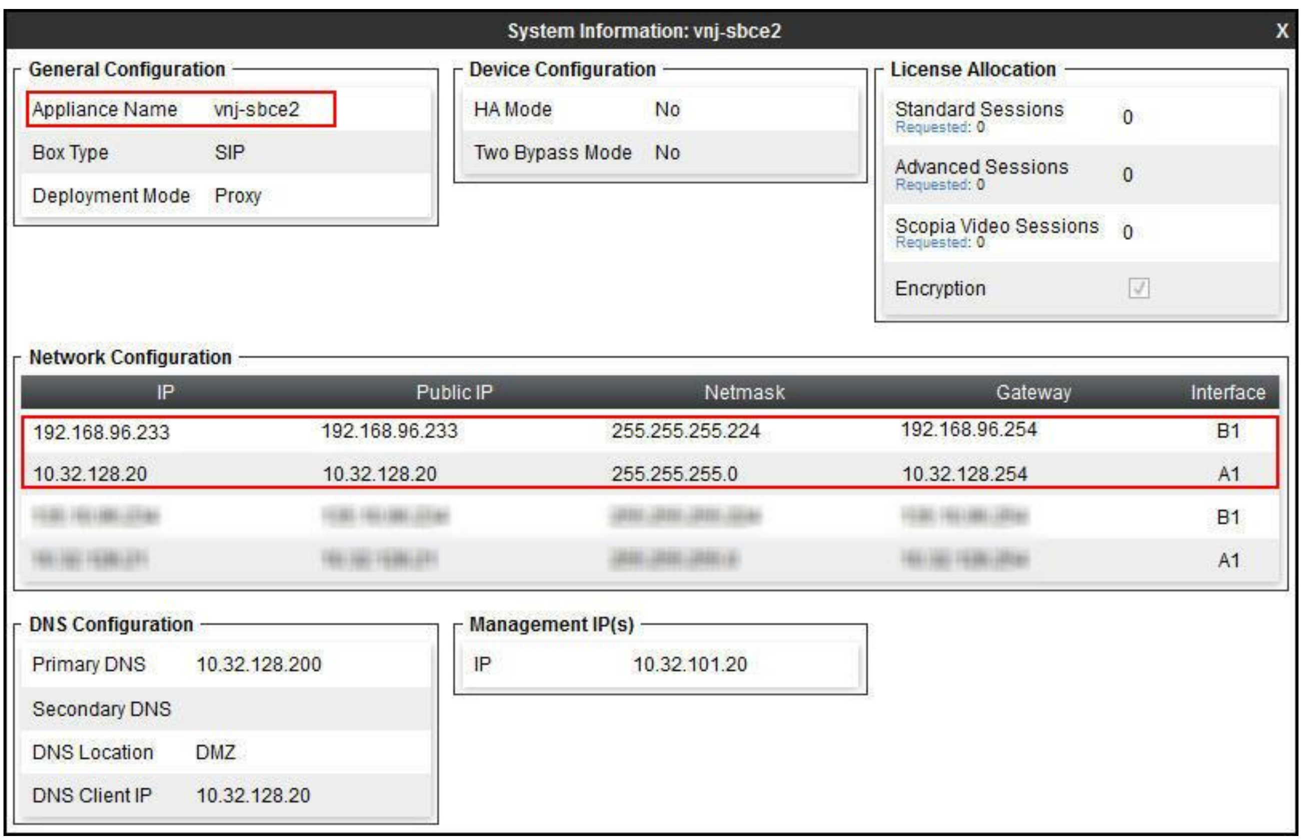
All configuration screens of the Avaya SBCE are accessed by navigating the menu tree in the left pane.



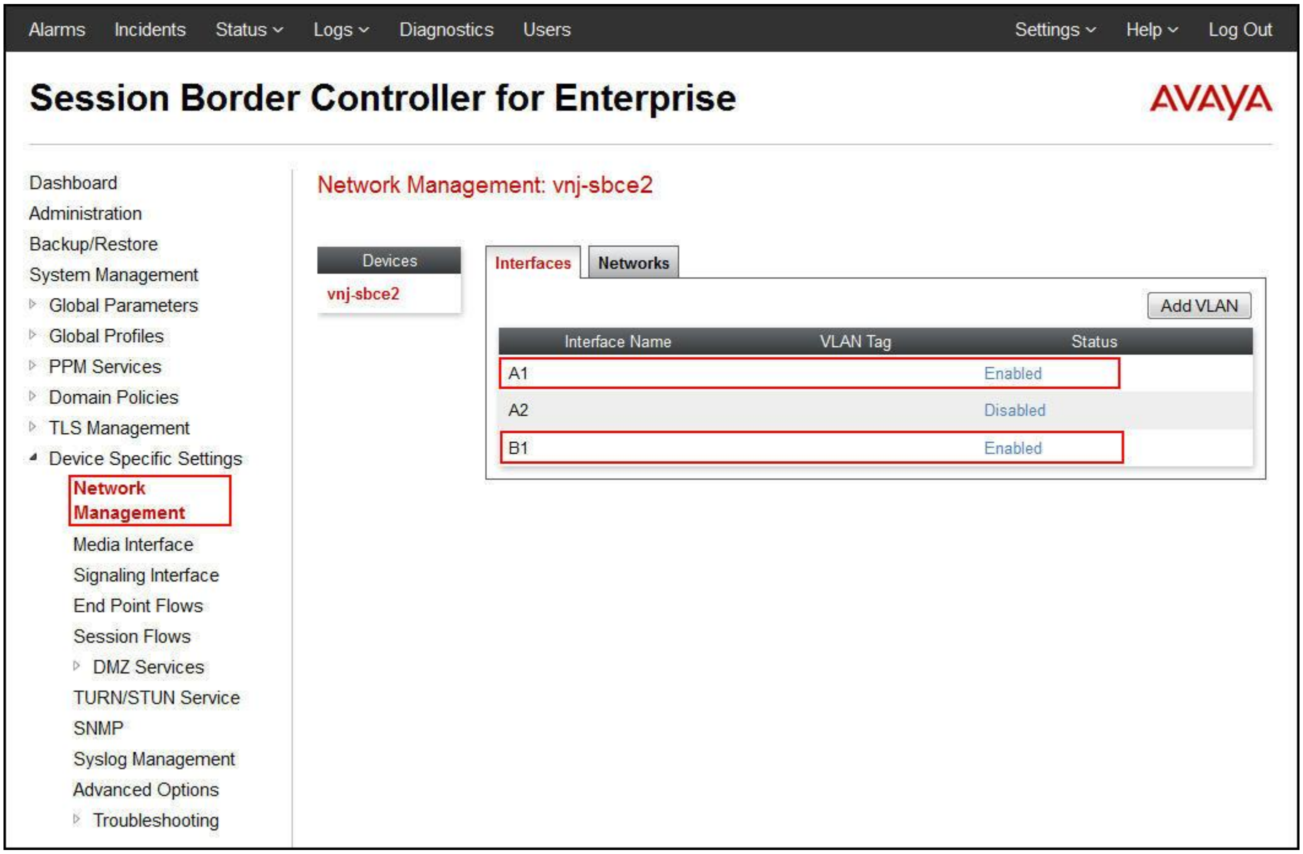
6.2. Verify Network Configuration and Enable Interfaces

To view the network information provided during installation, navigate to **System Management**. In the right pane, click **View** highlighted below.

A System Information page will appear showing the information provided during installation. The **Appliance Name** field is the name of the device (***vnj-sbce2***). This name will be referenced in other configuration screens. Interfaces **A1** and **B1** represent the private (or internal) and public (or external) interfaces of the Avaya SBCE. Each of these interfaces must be enabled after installation. Note that the **Management IP** is on a different subnet than either the A1 or B1 interfaces.



To enable the interfaces, first navigate to **Device Specific Settings → Network Management** in the left pane and select the device being managed in the center pane. In the right pane, in the **Interfaces** tab verify that **Status** is **Enabled** for both the **A1** and **B1** interfaces. If not, click on **Disabled** and confirm in the pop-up confirmation window to toggle to **Enabled**.



6.3. Signaling Interface

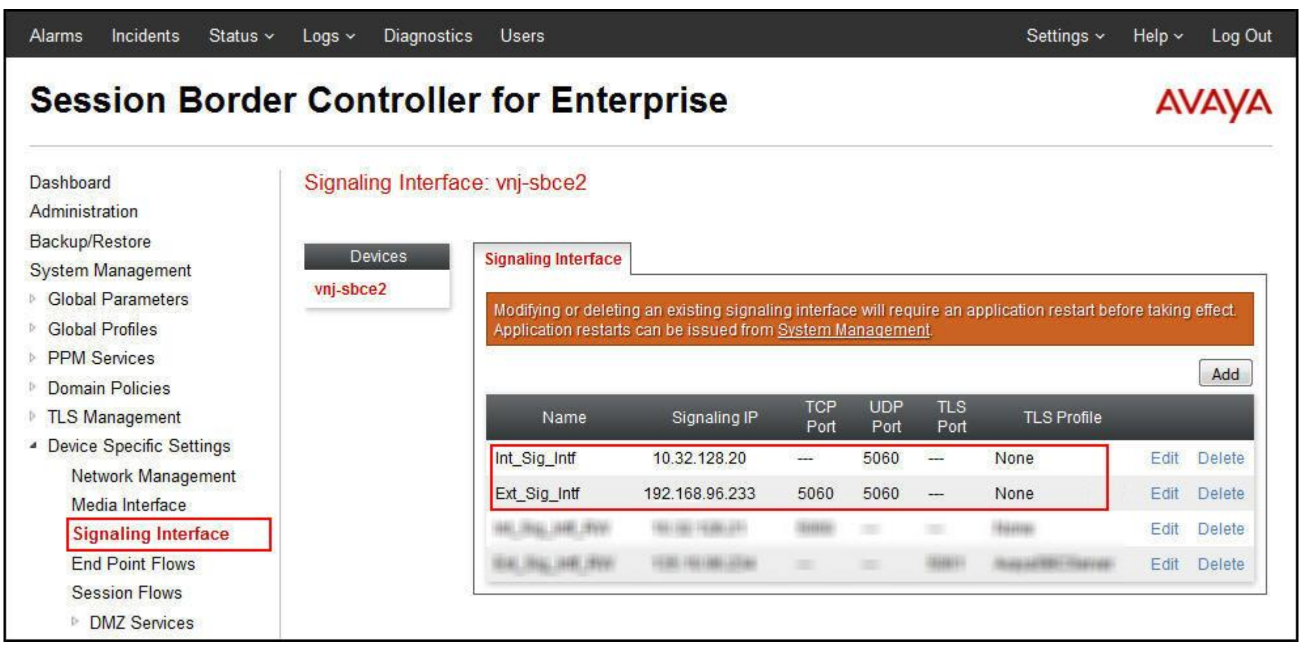
A signaling interface defines an IP address, protocols and listen ports that the Avaya SBCE can use for signaling. Create a signaling interface for both the internal and external sides of the Avaya SBCE.

To create a new interface, navigate to **Device Specific Settings → Signaling Interface** in the left pane. In the center pane, select the Avaya SBCE device to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by a series of pop-up windows in which the interface parameters can be configured. Once complete, the settings are shown in the far right pane.

For the compliance test, signaling interface **Int_Sig_Intf** was created for the Avaya SBCE internal interface and signaling interface **Ext_Sig_Intf** was created for the Avaya SBCE external interface. These two signaling interfaces are highlighted below.

When configuring the interfaces, configure the parameters as follows:

- Set **Name** to a descriptive name.
- For the internal interface, set the **Signaling IP** to the IP address associated with the private interface (A1) shown in **Section 6.2**. For the external interface, set the **Signaling IP** to the IP address associated with the public interface (B1) shown in **Section 6.2**.
- In the **UDP Port**, **TCP Port** and **TLS Port** fields, enter the port the Avaya SBCE will listen on for each transport protocol. For the internal interface, the Avaya SBCE was configured to listen for UDP on port 5060. For the external interface, the Avaya SBCE was configured to listen for UDP or TCP on port 5060. Since the Fusion Connect SIP Trunking Services uses UDP, it would have been sufficient to simply configure the Avaya SBCE for UDP.



6.4. Media Interface

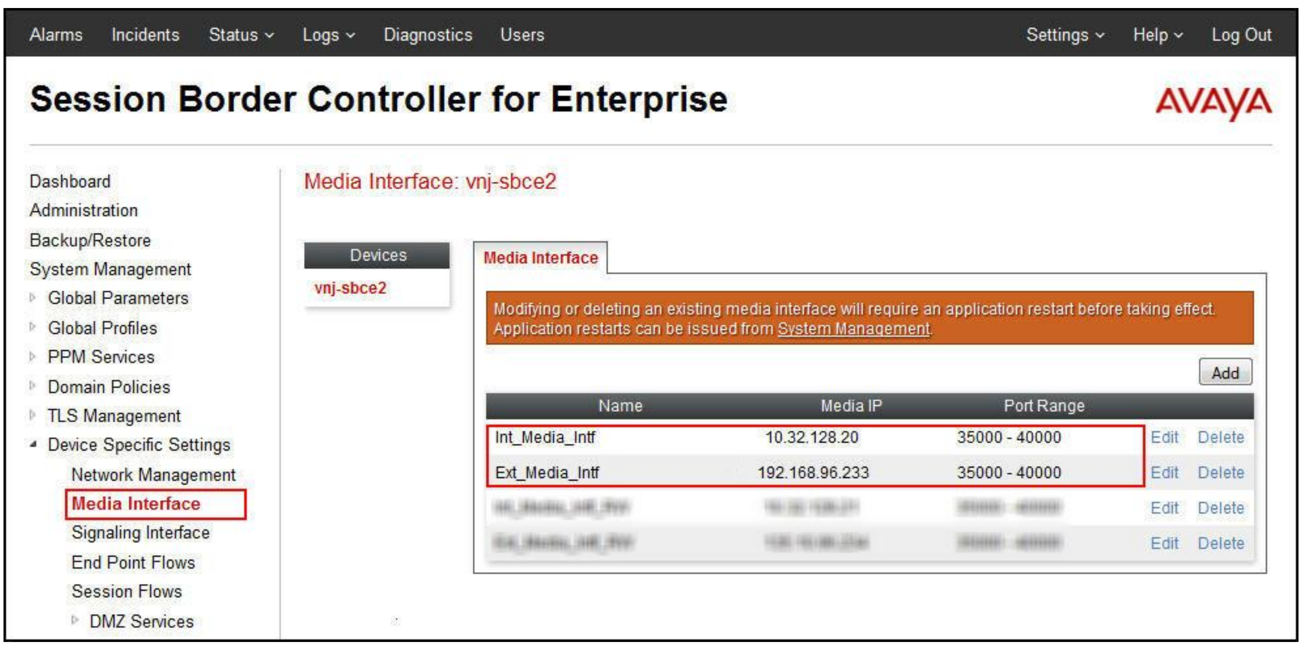
A media interface defines an IP address and port range for transmitting media. Create a media interface for both the internal and external sides of the Avaya SBCE.

To create a new interface, navigate to **Device Specific Settings → Media Interface** in the left pane. In the center pane, select the Avaya SBCE device to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by a series of pop-up windows in which the interface parameters can be configured. Once complete, the settings are shown in the far right pane.

For the compliance test, media interface **Int_Media_Intf** was created for the Avaya SBCE internal interface and media interface **Ext_Media_Intf** was created for the Avaya SBCE external interface. Both are highlighted below.

When configuring the interfaces, configure the parameters as follows:

- Set **Name** to a descriptive name.
- For the internal interface, set the **Media IP** to the IP address associated with the private interface (A1) shown in **Section 6.2**. For the external interface, set the **Media IP** to the IP address associated with the public interface (B1) shown in **Section 6.2**.
- Set **Port Range** to a range of ports acceptable to both the enterprise and the far end. For the compliance test, the default port range was used for both interfaces.

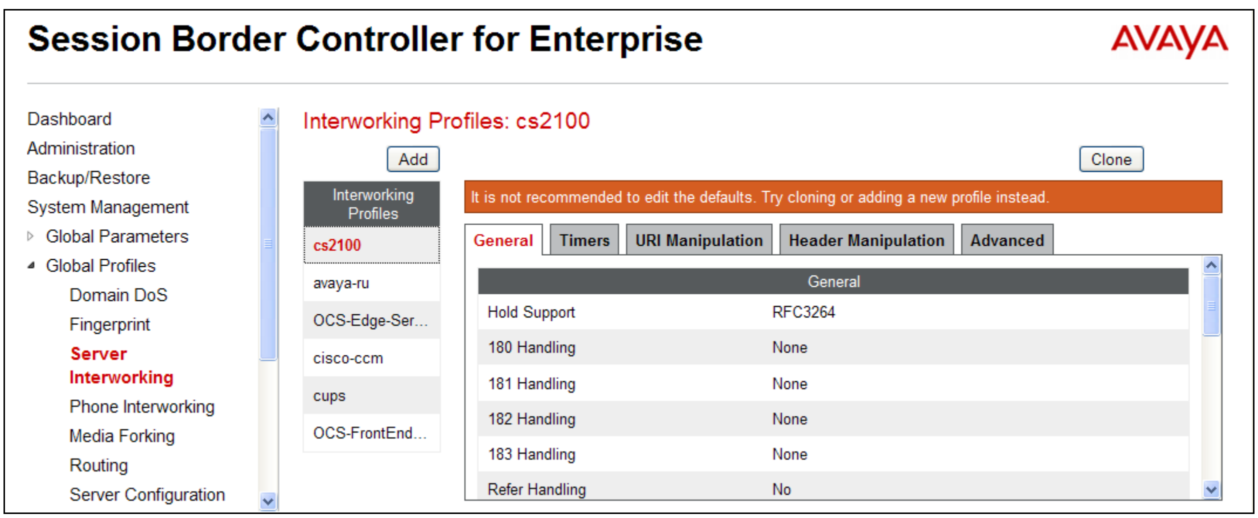


6.5. Server Interworking

A server interworking profile defines a set of parameters that aid in interworking between the Avaya SBCE and a connected server. Create one server interworking profile for Avaya IP Office and another for the service provider SIP server. These profiles will be applied to the appropriate servers in **Section 6.6.1** and **6.6.2**.

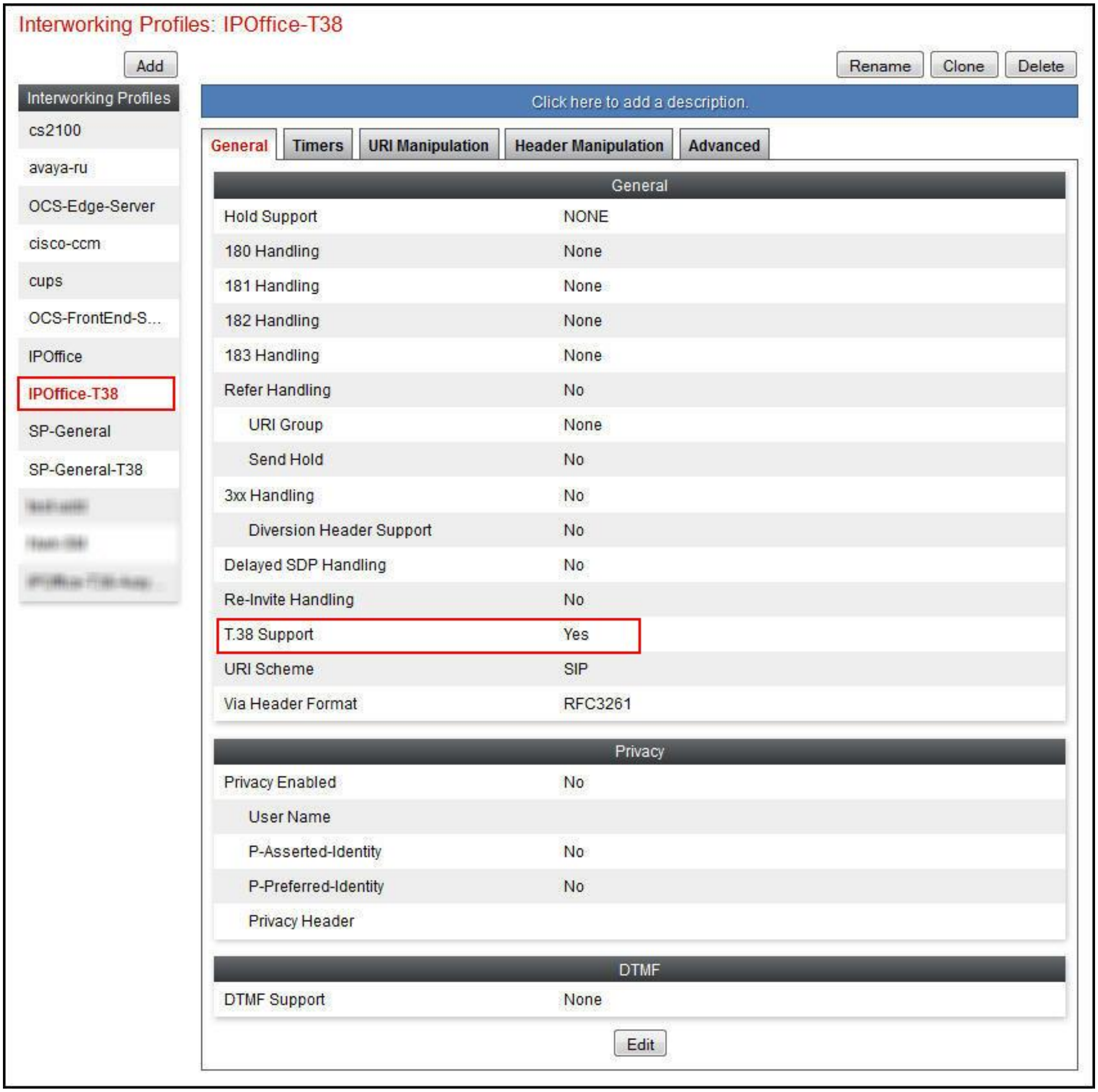
To create a new profile, navigate to **Global Profiles → Server Interworking** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by a series of pop-up windows in which the profile parameters can be configured. Once complete, the settings are shown in the far right pane. Alternatively, a new profile may be created by selecting an existing profile in the center pane and clicking the **Clone** button in the right pane. This will create a copy of the selected profile which can then be edited as needed. To view the settings of an existing profile, select the profile from the center pane. The settings will appear in the right pane.

The screen below shows the user interface as described above, before creating the specific server interworking profiles used for the compliance test.



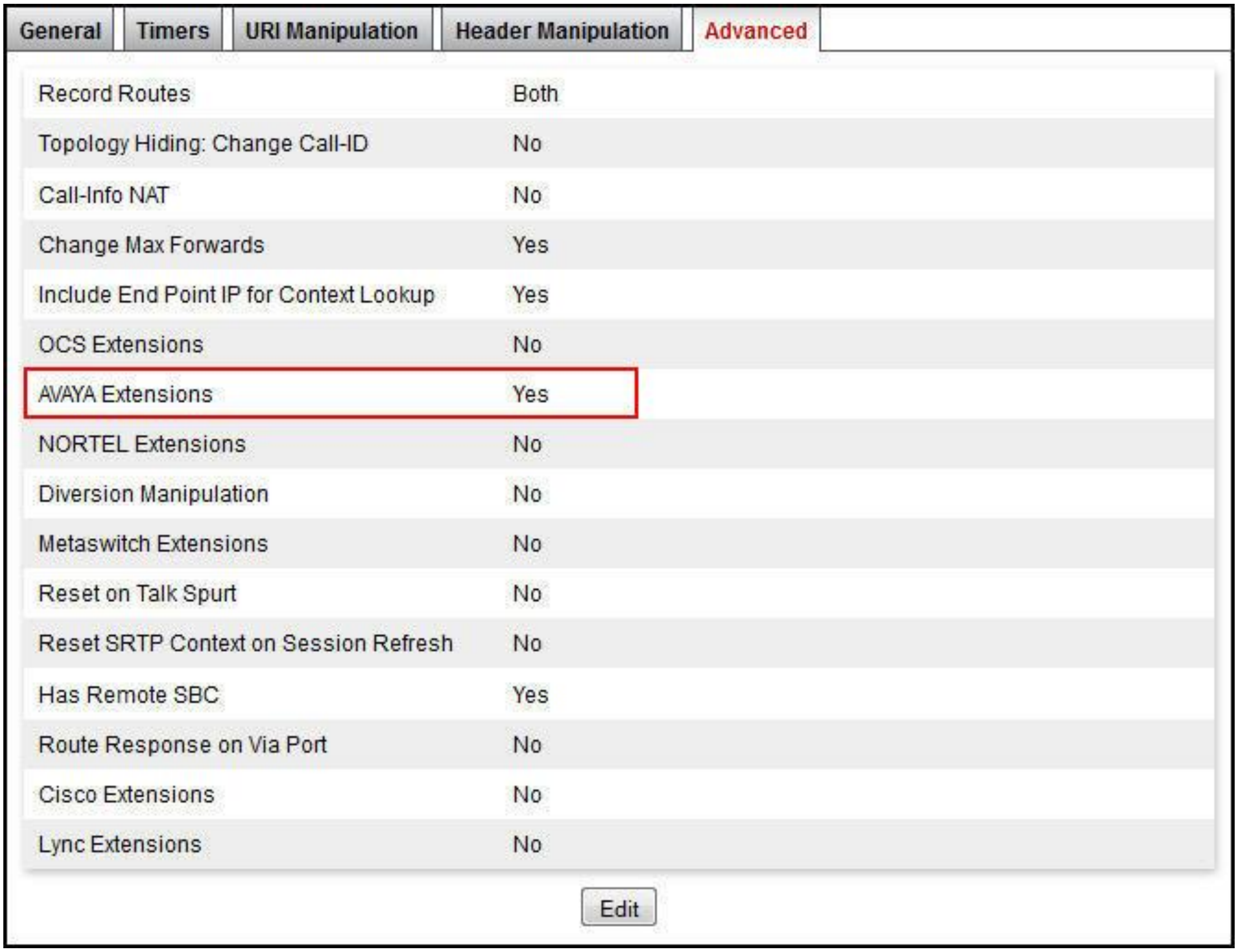
6.5.1. Server Interworking – Avaya IP Office

For the compliance test, the server interworking profile ***IPOffice-T.38*** was created for Avaya IP Office. The **General** tab parameters are shown below. Note the setting for **T.38 Support**.



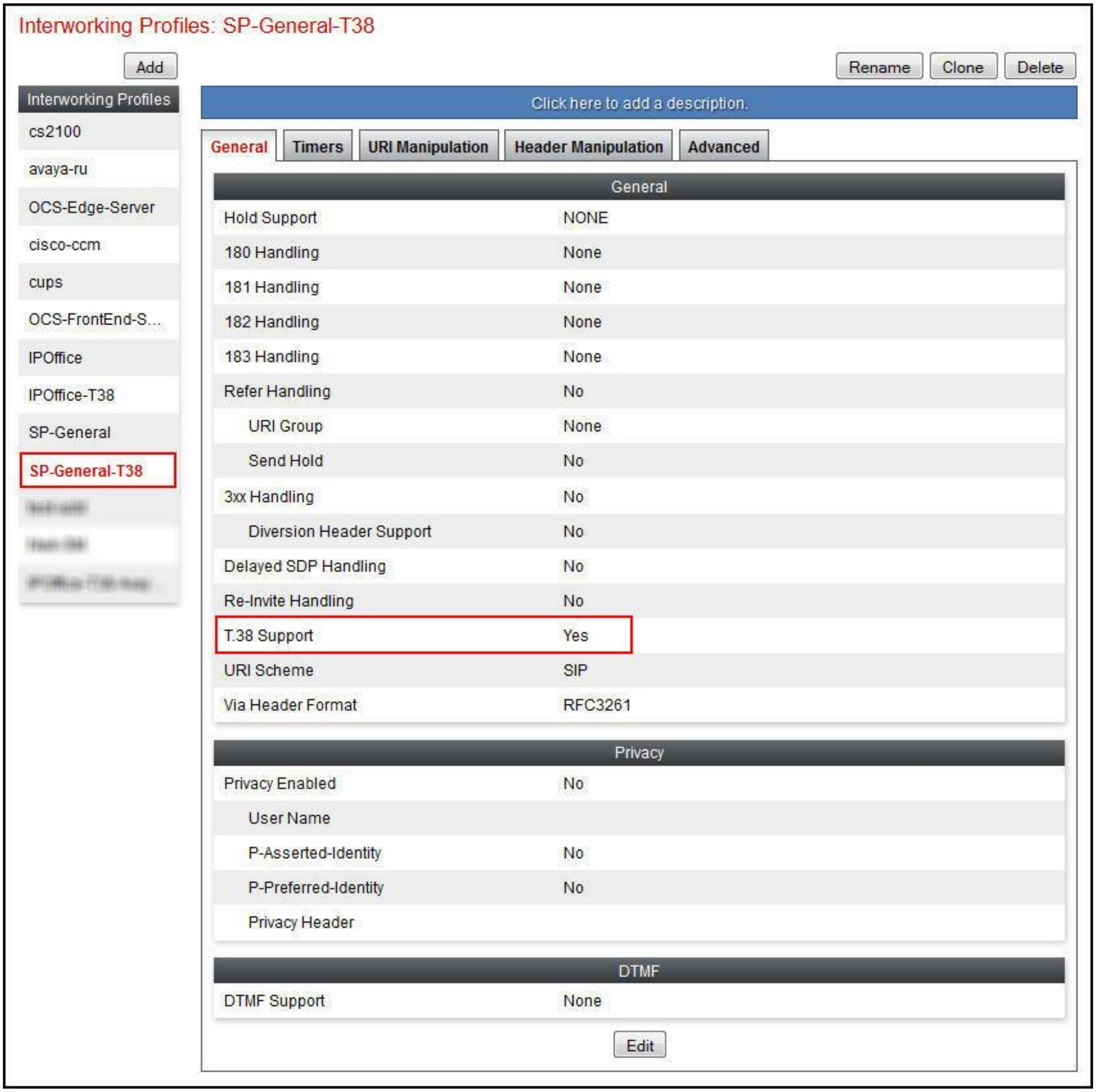
The **Timers**, **URI Manipulation** and **Header Manipulation** tabs have no configured entries.

The **Advanced** tab parameters are shown below. Note that **AVAYA Extensions** is set to ***Yes***.



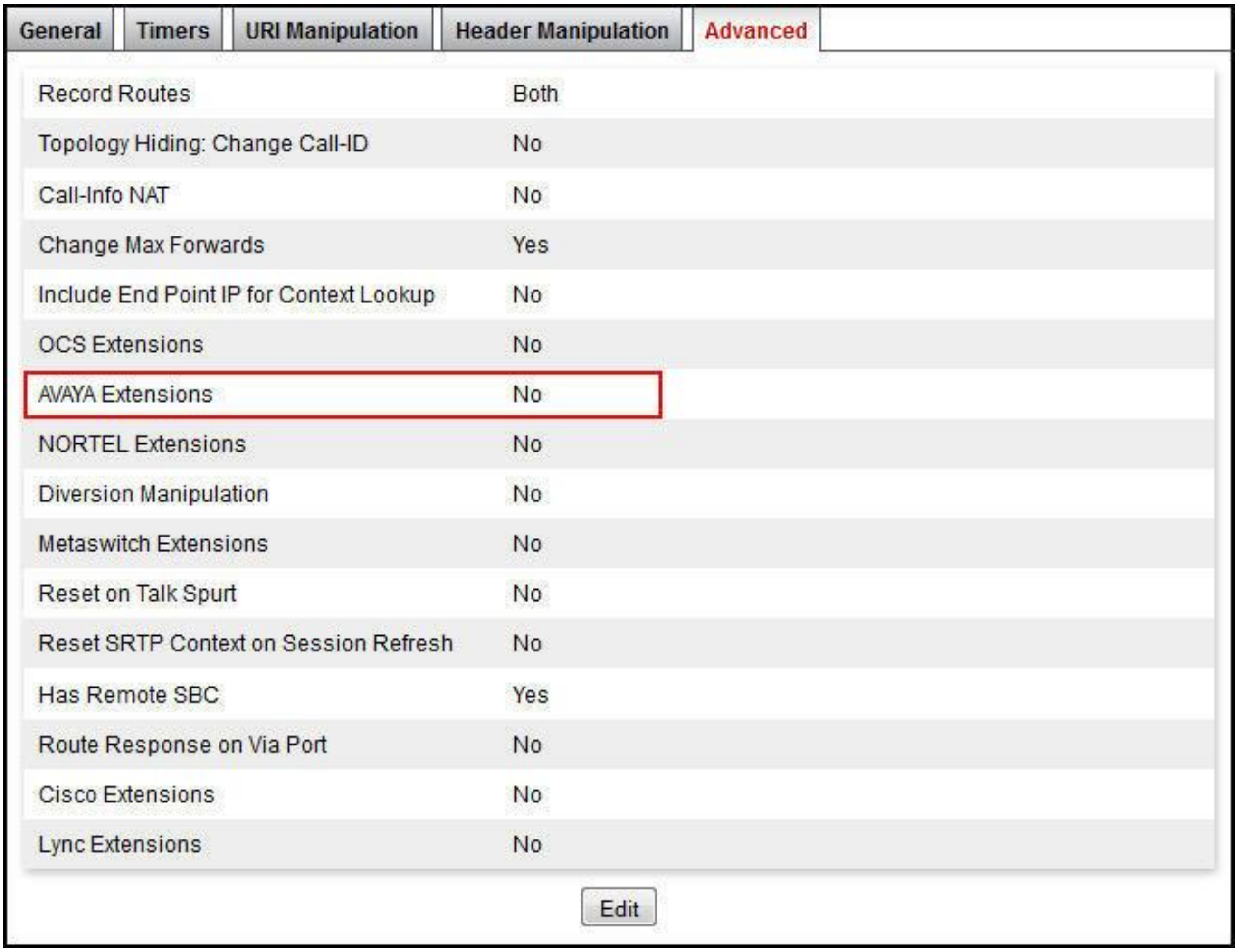
6.5.2. Server Interworking – Fusion Connect

For the compliance test, server interworking profile ***SP-General-T38*** was created for the Fusion Connect SIP server. The **General** tab parameters are shown below. Note the setting for **T.38 Support**.



The **Timers**, **URI Manipulation**, **Header Manipulation** tabs have no entries.

The **Advanced** tab parameters are shown below. Note that **AVAYA Extensions** is set to ***No***.

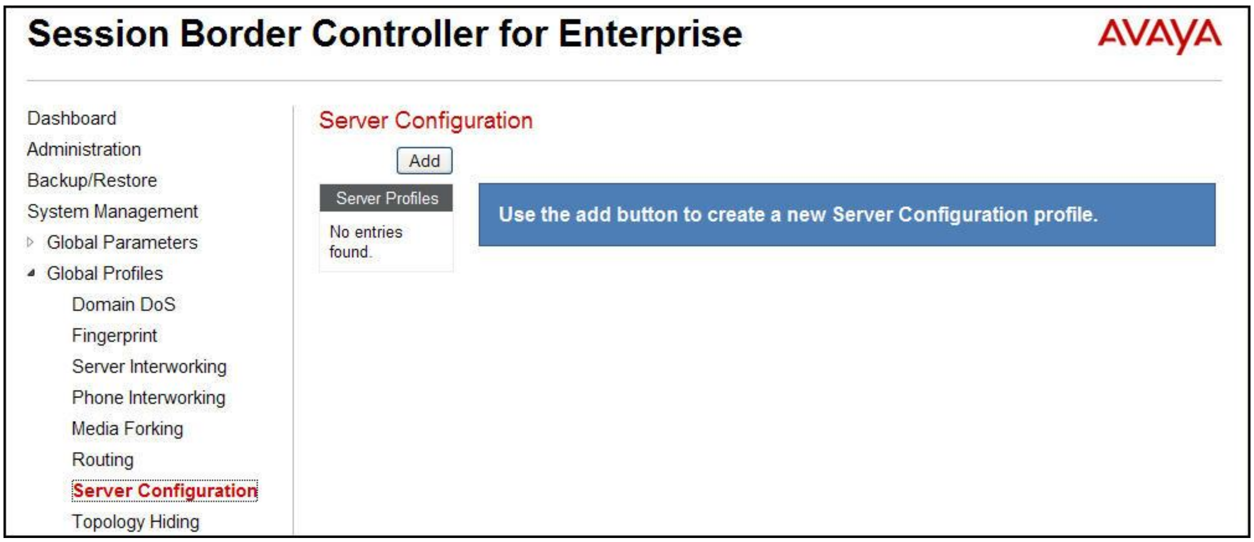


6.6. Server Configuration

A server configuration profile defines the attributes of the physical server. Create separate server configuration profiles for Avaya IP Office and the service provider SIP server.

To create a new profile, navigate to **Global Profiles → Server Configuration** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by a series of pop-up windows in which the profile parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing profile, select the profile from the center pane. The settings will appear in the right pane.

The screen below shows the GUI elements described above before the servers profiles were added for the compliance test.

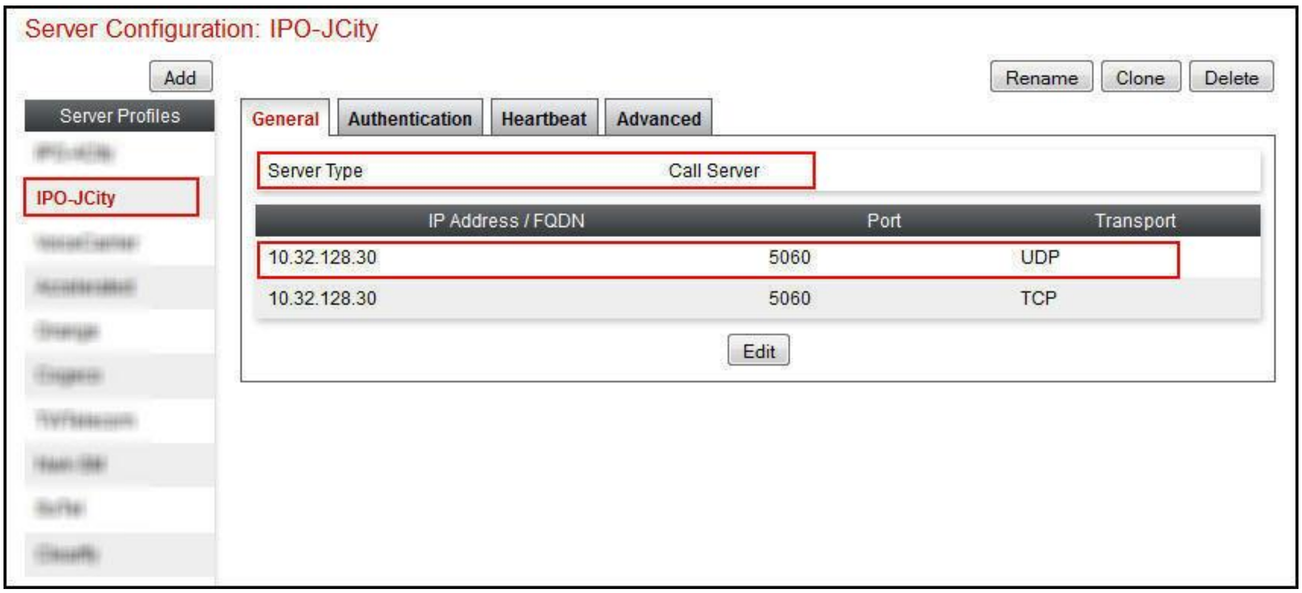


6.6.1. Server Configuration – Avaya IP Office

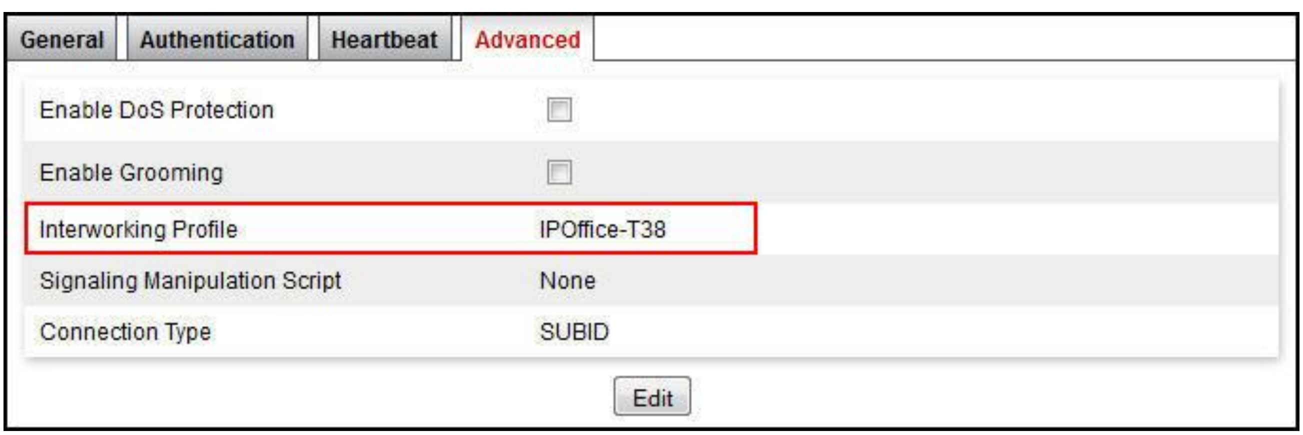
For the compliance test, the server configuration profile ***IPO-JCity*** was created for Avaya IP Office. When creating the profile, configure the **General** tab parameters as follows:

- Set **Server Type** to ***Call Server***.
- Set **IP Addresses / FQDNs** to the IP address of the Avaya IP Office LAN1 port.
- Set **Transport** to ***UDP***, the transport protocol used for SIP signaling between Avaya IP Office and the Avaya SBCE.
- Set **Port** to the port Avaya IP Office will listen on for SIP requests from the Avaya SBCE.

Note that TCP was also set in the screen below, though UDP connectivity would have been sufficient.



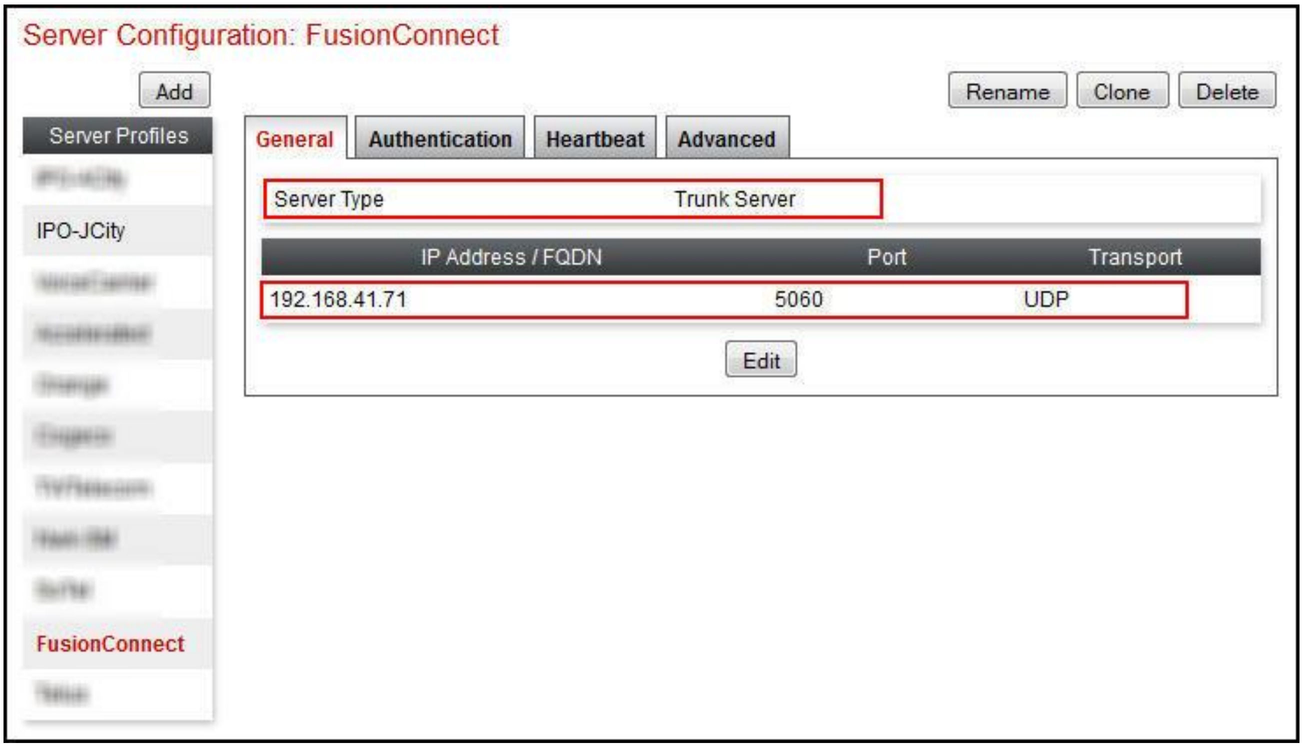
On the **Advanced** tab, set the **Interworking Profile** field to the interworking profile for Avaya IP Office defined in **Section 6.5.1**.



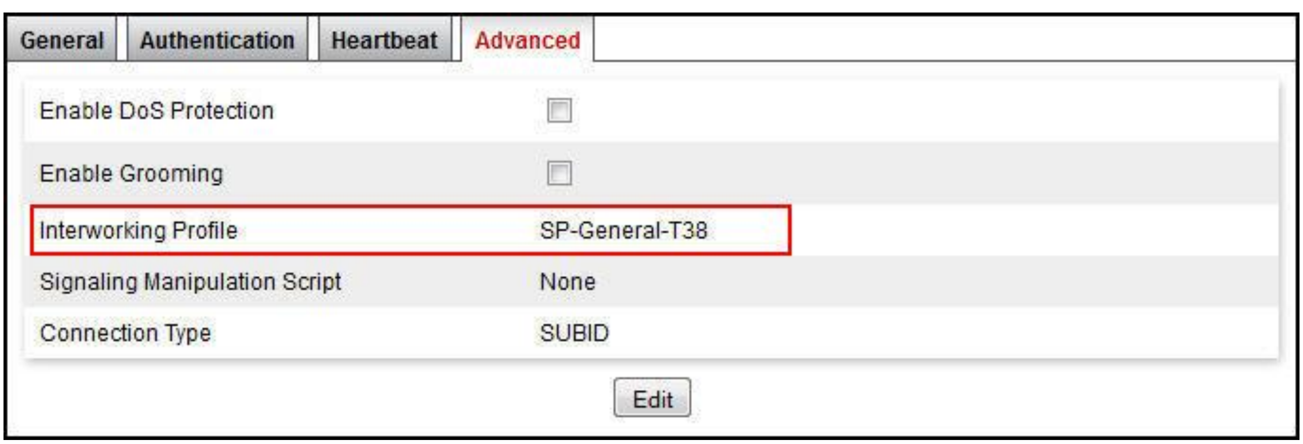
6.6.2. Server Configuration – Fusion Connect

For the compliance test, server configuration profile Fusion Connect was created for Fusion Connect. When creating the profile, configure the **General** tab parameters as follows:

- Set **Server Type** to ***Trunk Server***.
- Set **IP Addresses / FQDN** to the IP address of the Fusion Connect network access interface.
- Select the appropriate **Transport** protocol used for SIP signaling between Fusion Connect and the Avaya SBCE. In the compliance test, **UDP** was tested.
- Set **Port** to the standard SIP port of **5060**. This is the port the Fusion Connect SIP server will listen on for SIP messages from the Avaya SBCE.



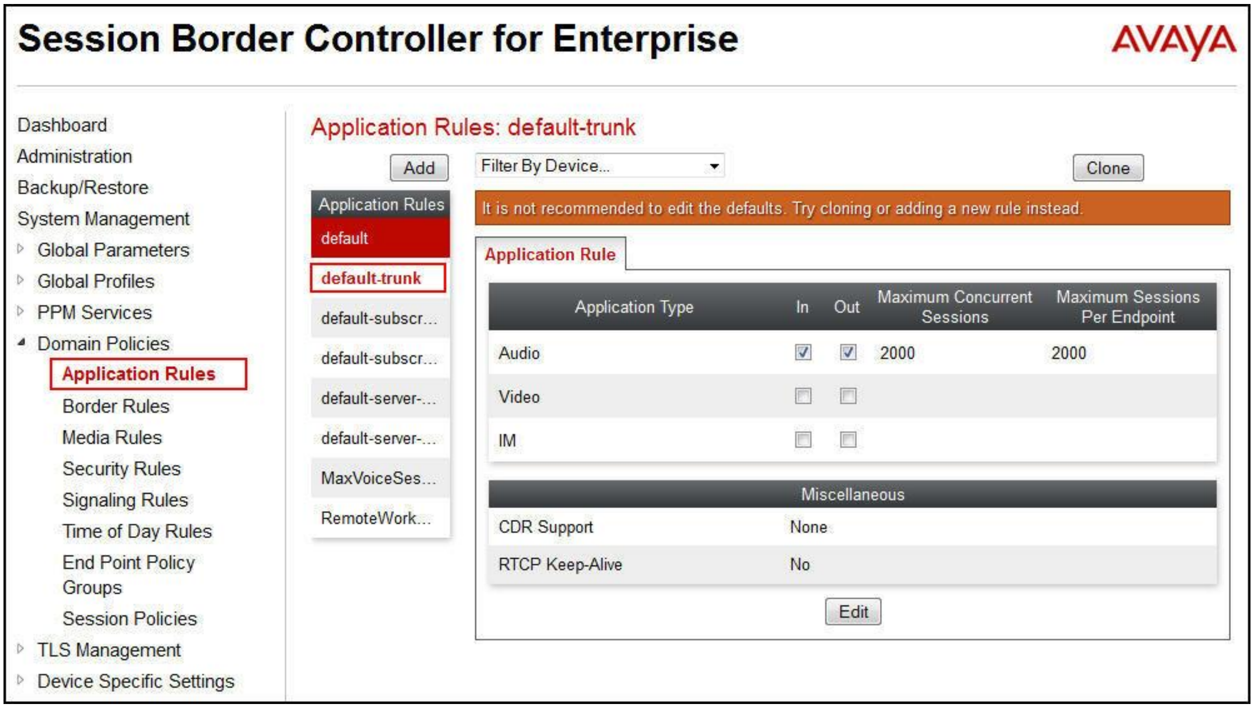
On the **Advanced** tab, select the **Interworking Profile** for Fusion Connect defined in **Section 6.5.2**.



6.7. Application Rules

An application rule defines the allowable SIP applications and associated parameters. An application rule is one component of the larger endpoint policy group defined in **Section 6.10**. For the compliance test, the predefined **default-trunk** application rule (shown below) was used for both Avaya IP Office and the Fusion Connect SIP server.

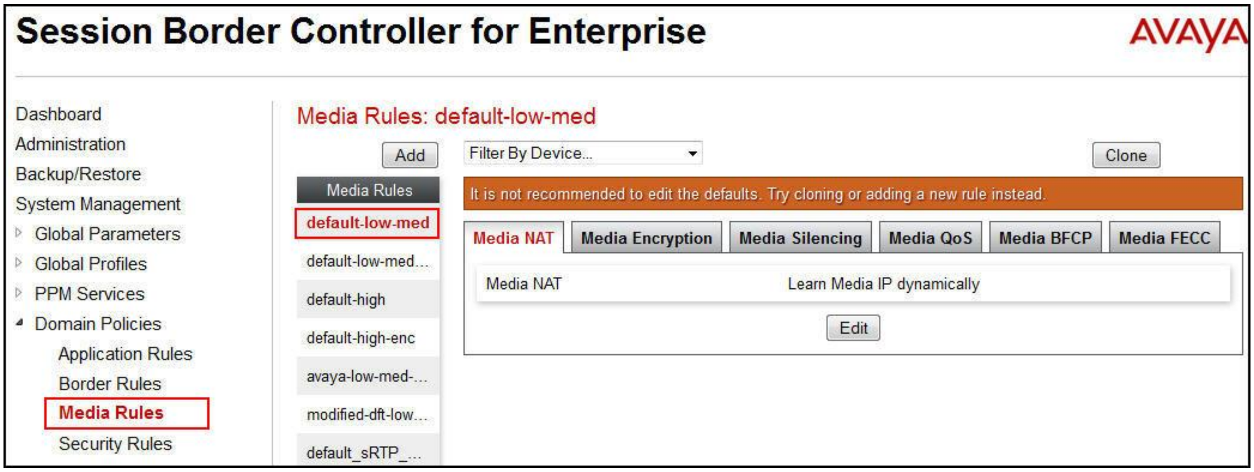
To view an existing rule, navigate to **Domain Policies → Application Rules** in the left pane. In the center pane, select the rule (e.g., **default-trunk**) to be viewed.



6.8. Media Rules

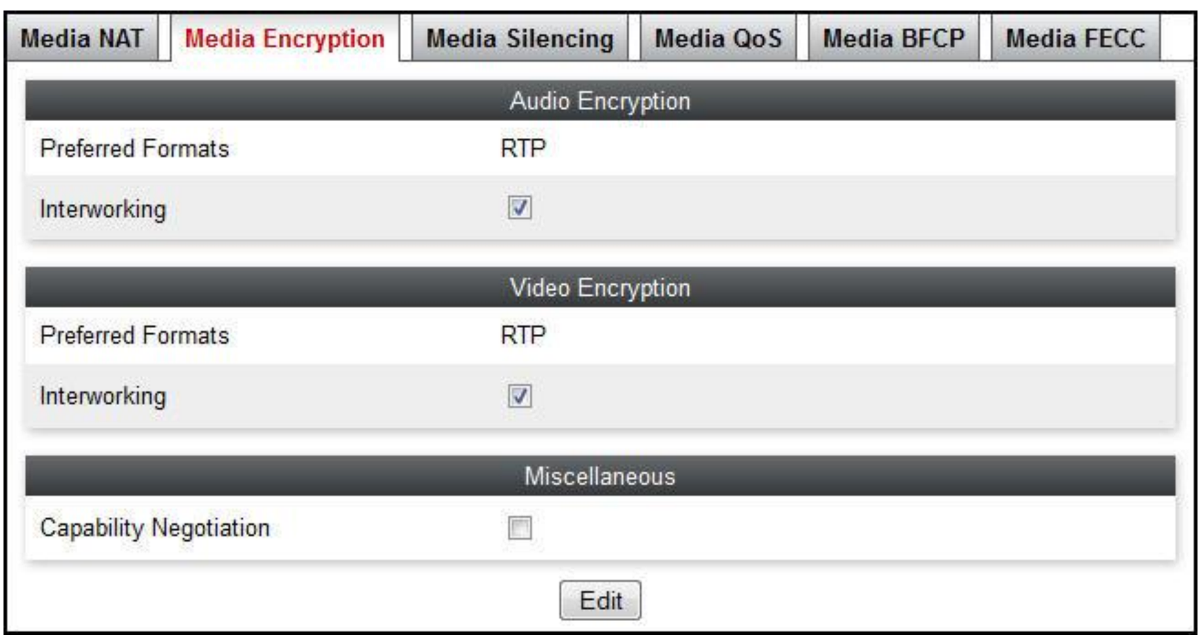
A media rule defines the processing to be applied to the selected media. A media rule is one component of the larger end point policy group defined in **Section 6.10**. For the compliance test, the predefined **default-low-med** media rule (shown below) was used for both Avaya IP Office and the Fusion Connect SIP server.

To view an existing rule, navigate to **Domain Policies → Media Rules** in the left pane. In the center pane, select the rule (e.g., **default-low-med**) to be viewed.



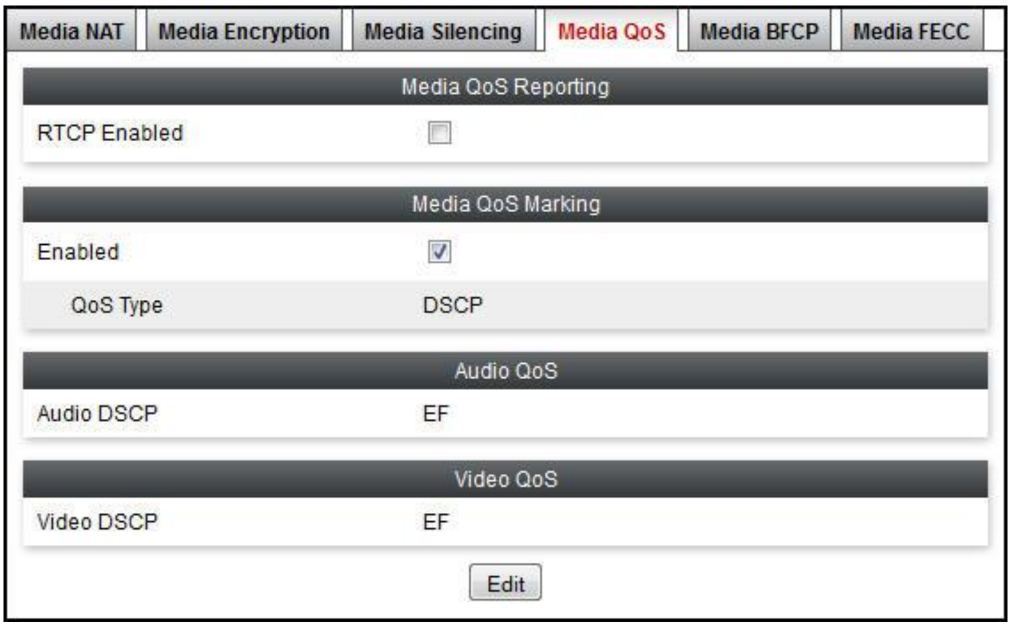
Each of the tabs of the **default-low-med** media rule is shown below (the **Media NAT** tab is shown above).

The **Media Encryption** tab indicates that no encryption was used.



The **Media Silencing** tab shows **Media Silencing** was disabled.

The **Media QoS** settings are shown below.



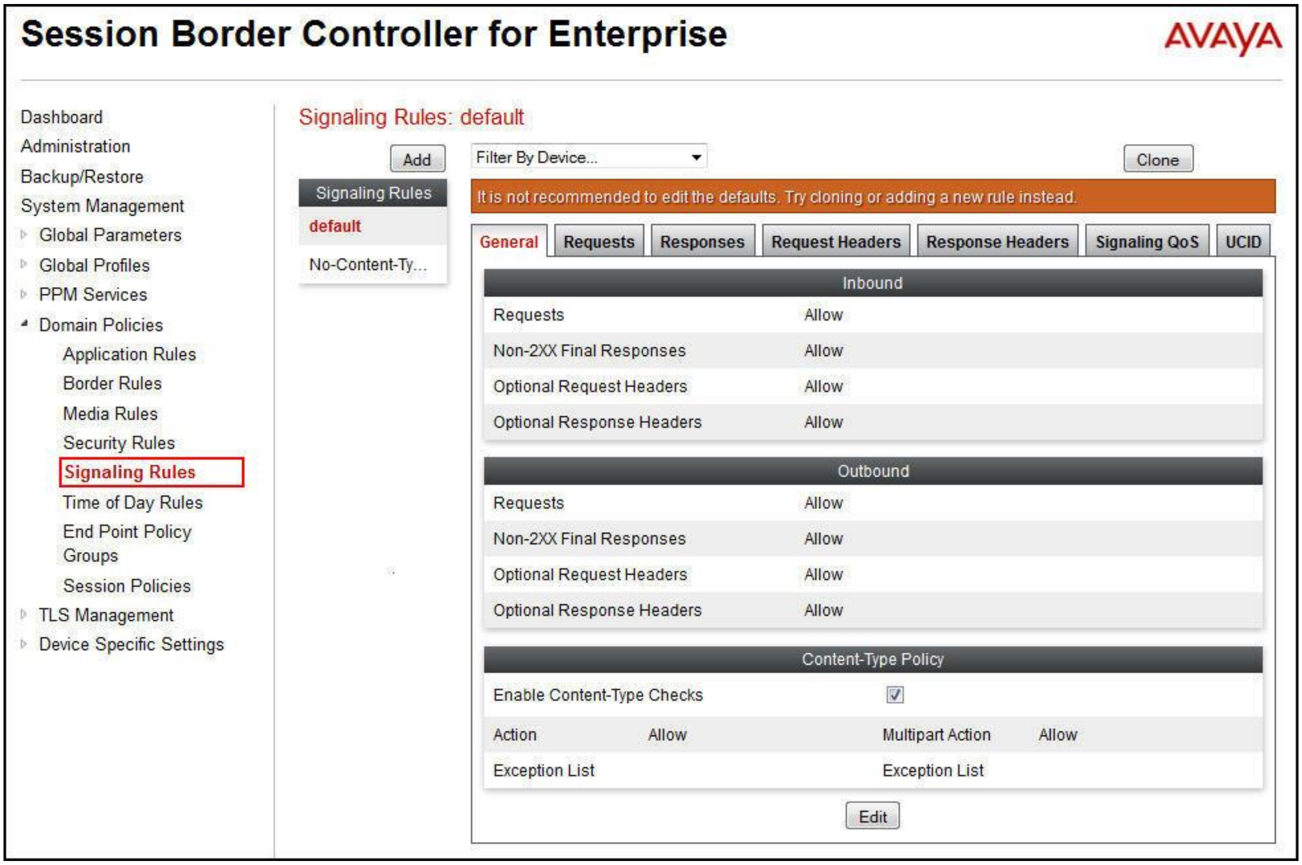
The **Media BFCP** tab is shown below.

The **Media FECC** tab is shown below.

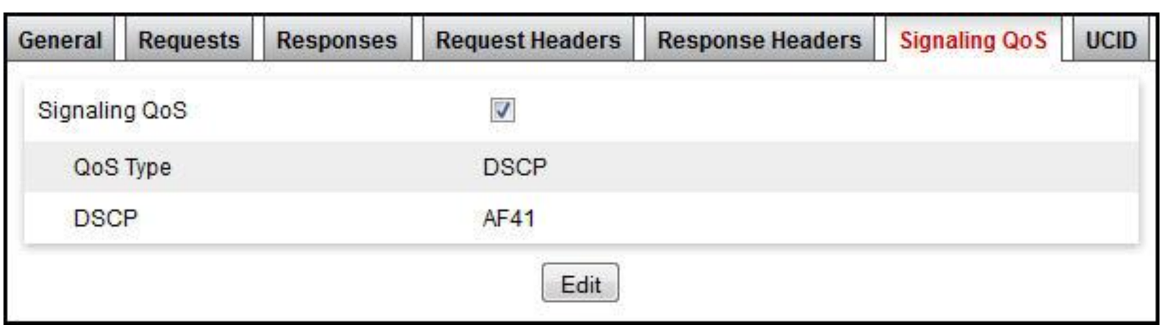
6.9. Signaling Rules

A signaling rule defines the processing to be applied to the selected signaling traffic. A signaling rule is one component of the larger end point policy group defined in **Section 6.10**. For the compliance test, the predefined **default** signaling rule (shown below) was used for both Avaya IP Office and the Fusion Connect SIP server.

To view an existing rule, navigate to **Domain Policies → Signaling Rules** in the left pane. In the center pane, select the rule (e.g., **default**) to be viewed. The **General** tab settings of the default signaling rule are shown below.



The **Requests**, **Responses**, **Request Headers, Response Headers** and **UCID** tabs have no entries. The **Signaling QoS** tab is shown below.

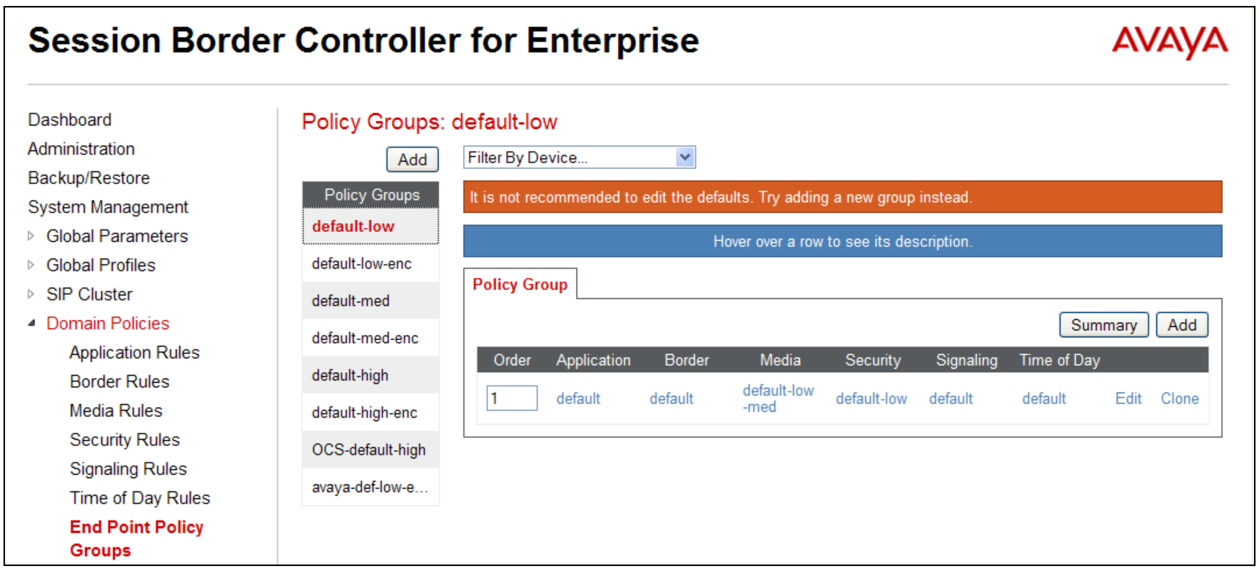


6.10. End Point Policy Groups

An end point policy group is a set of policies that will be applied to traffic between the Avaya SBCE and a signaling endpoint (connected server). Thus, one end point policy group must be created for Avaya IP Office and another for the service provider SIP server. The end point policy group is applied to the traffic as part of the end point flow defined in **Section 6.13**.

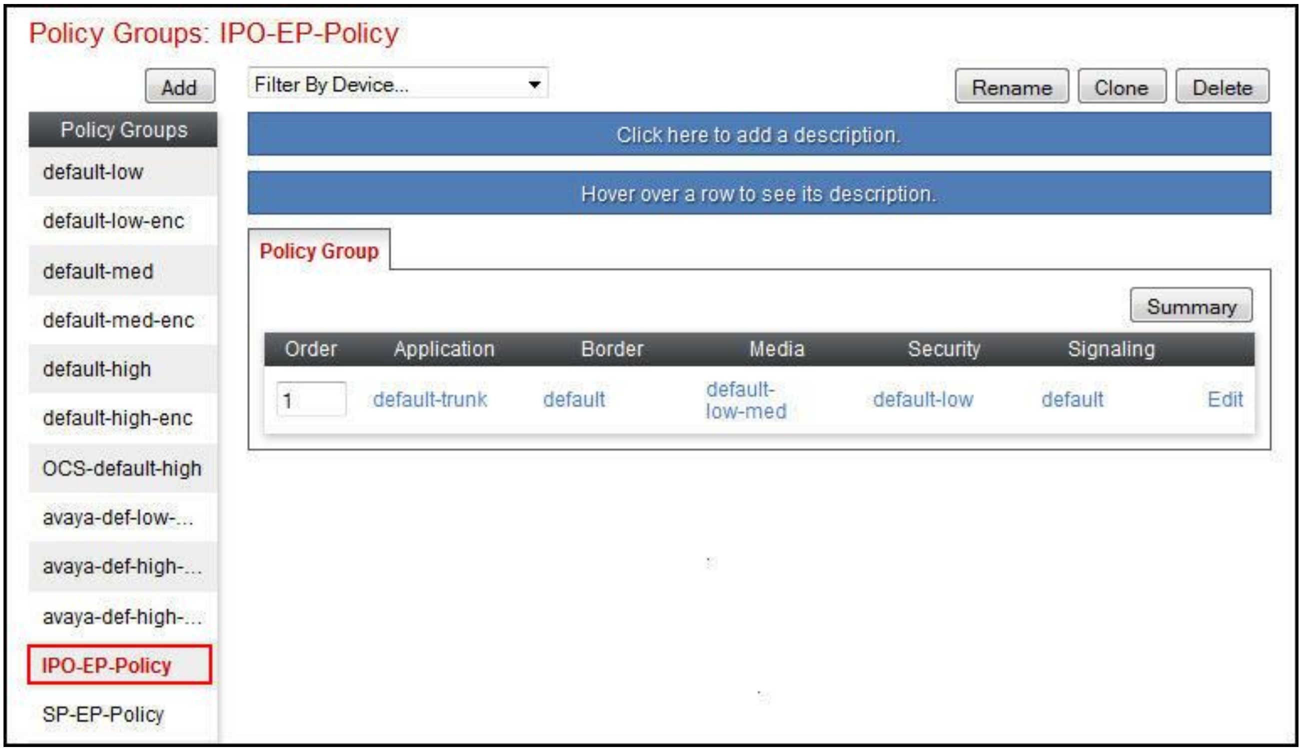
To create a new group, navigate to **Domain Policies → End Point Policy Groups** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new group, followed by a series of pop-up windows in which the group parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing group, select the group from the center pane. The settings will appear in the right pane.

The screen below shows the GUI elements described above before specific endpoint policy groups were added for the compliance test.



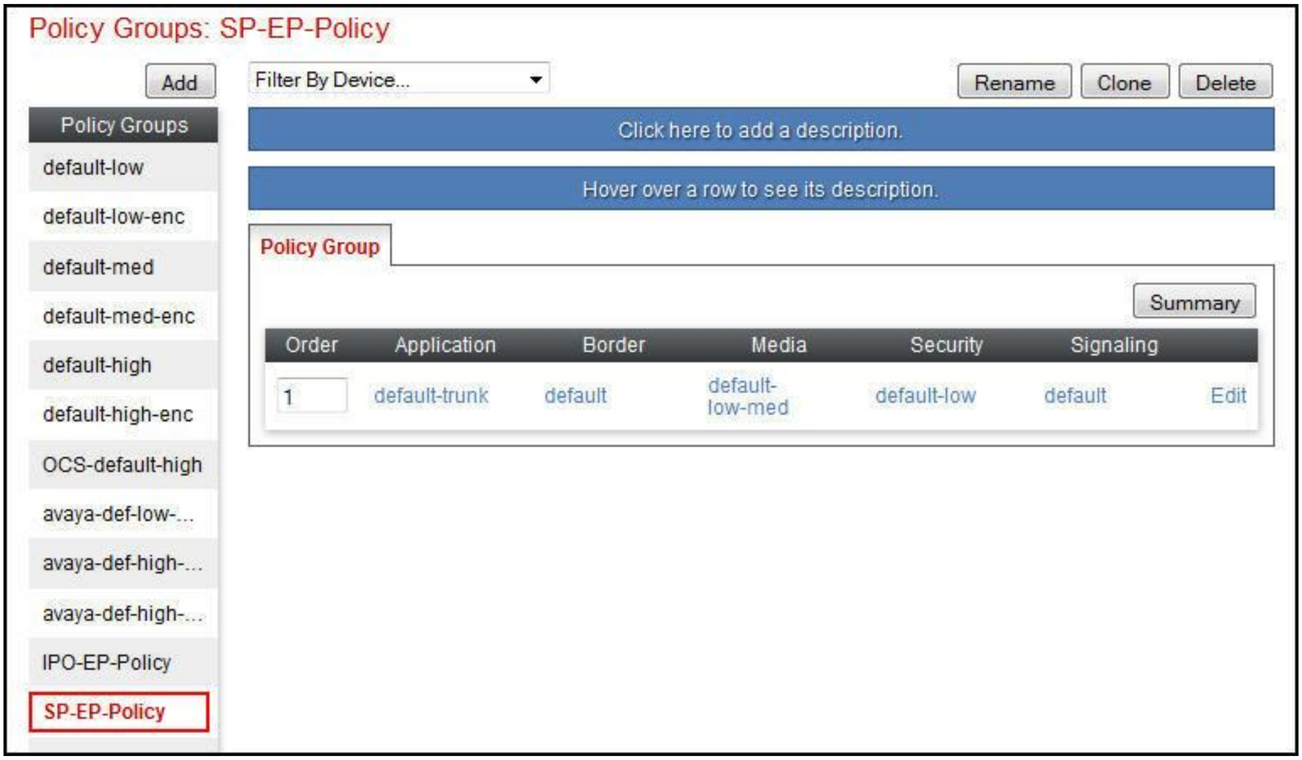
6.10.1. End Point Policy Group – Avaya IP Office

For the compliance test, the end point policy group ***IPO-EP-Policy*** was created for Avaya IP Office. Default values were used for each of the rules which comprise the group. The details of the default settings for **Application**, **Media** and **Signaling** are shown in **Section 6.7**, **Section 6.8** and **Section 6.9** respectively.



6.10.2. End Point Policy Group – Fusion Connect

For the compliance test, the end point policy group ***SP-EP-Policy*** was created for the Fusion Connect SIP server. Same default values were used for each of the rules which comprise the group. Thus, the **SP-EP-Policy** is identical to the **IPO-EP-Policy** created in **Section 6.10.1**.

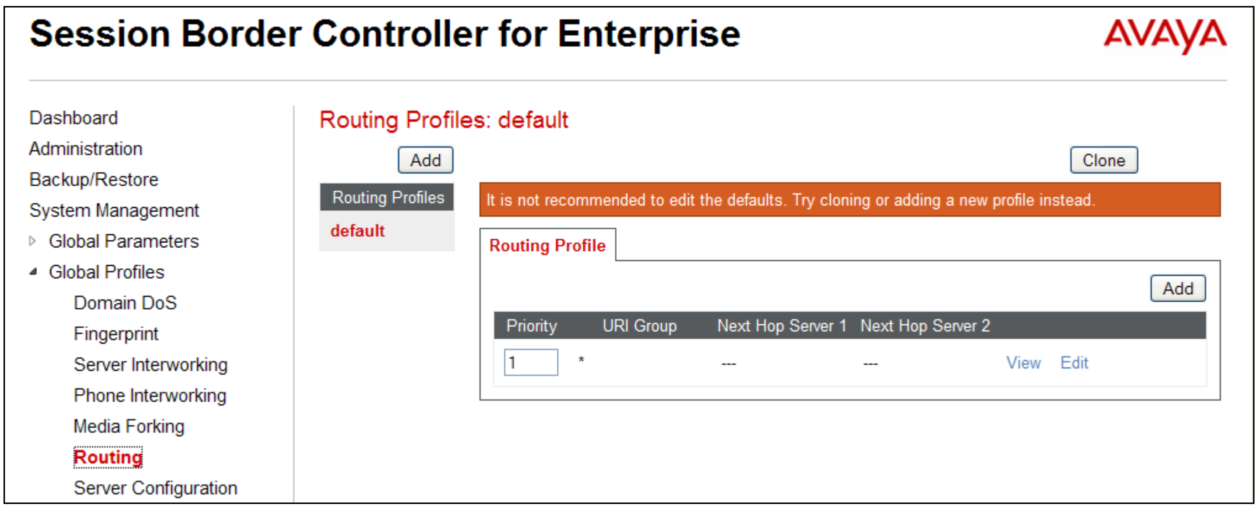


6.11. Routing

A routing profile defines where traffic will be directed based on the contents of the URI. A routing profile is applied only after the traffic has matched an endpoint server flow defined in **Section 6.13**. Create one routing profile for Avaya IP Office and another for the service provider SIP server.

To create a new profile, navigate to **Global Profiles → Routing** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by a series of pop-up windows in which the profile parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing profile, select the profile from the center pane. The settings will appear in the right pane.

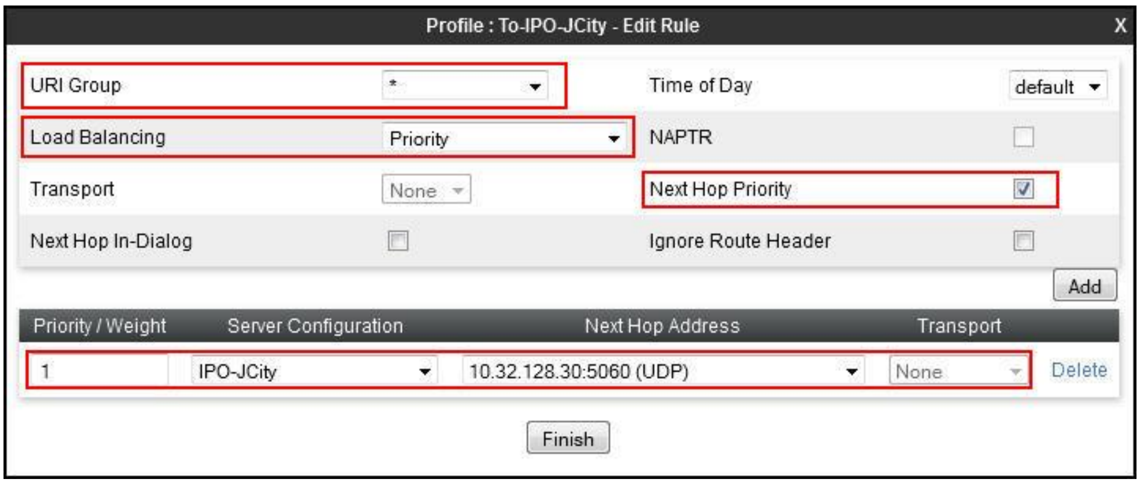
The screen below shows the GUI elements described above before specific routing profiles were added for the compliance test.



6.11.1. Routing – Avaya IP Office

For the compliance test, the routing profile ***To-IPO-JCity*** was created for Avaya IP Office. When creating the profile, configure the parameters as follows:

- Set **URI Group** to the wild card ***** to match on any URI.
- Select ***Priority*** for **Load Balancing**.
- Enable **Next Hop Priority**.
- When adding an entry for routing destination (Next Hop Address)
  - Enter ***1*** for **Priority/Weight**.
  - For **Server Configuration**, select the Server for Avaya IP Office as configured in **Section 6.6.1**.
  - Set **Next Hop Address** to the IP address of Avaya IP Office LAN1 port.
  - Select ***UDP*** for **Transport** (the transport will be displayed in the Next Hop Address field once the entry is added).

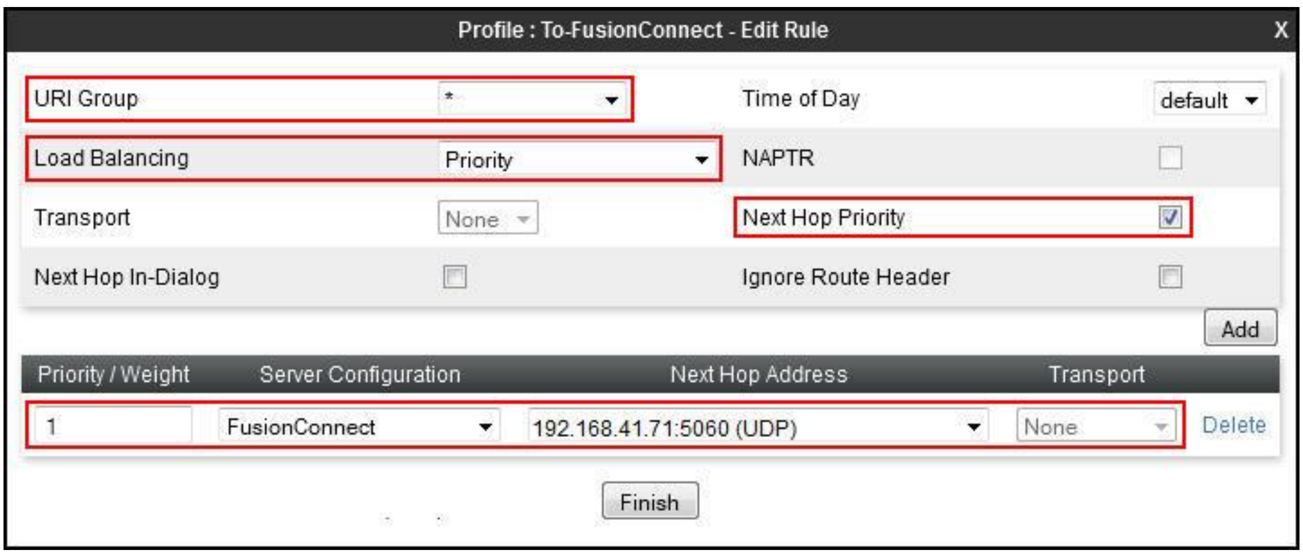


The following screen shows the routing profile for Avaya IP Office when configured.

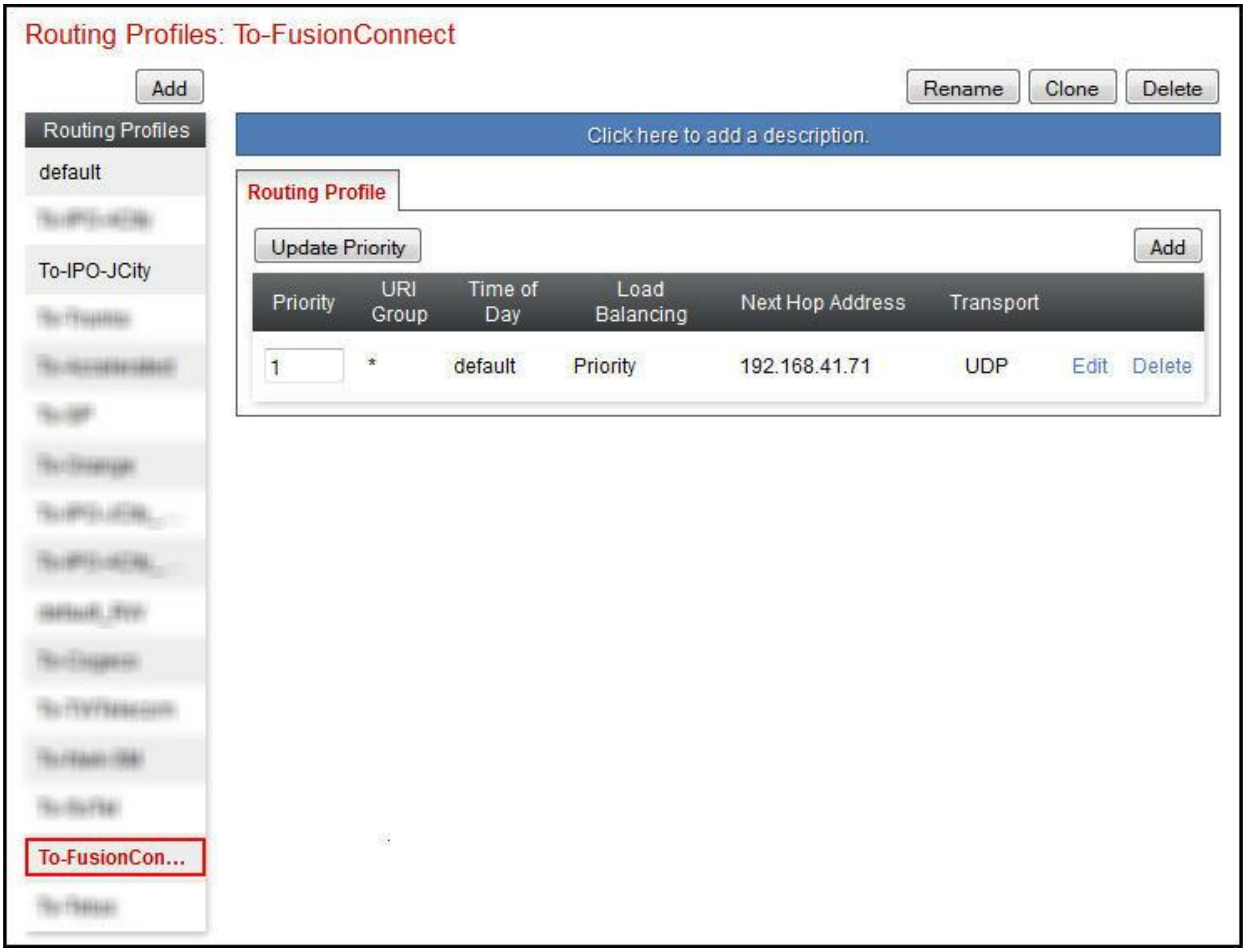
6.11.2. Routing – Fusion Connect

For the compliance test, routing profile ***To-FusionConnect*** was created for routing calls to Fusion Connect. When creating the profile, configure the parameters as follows:

- Set **URI Group** to the wild card ***** to match on any URI.
- Select ***Priority*** for **Load Balancing**.
- Enable **Next Hop Priority**.
- When adding an entry for routing destination (Next Hop Address)
  - Enter a sequential number starting with ***1*** for **Priority/Weight**.
  - For **Server Configuration**, select the Server for Fusion Connect as configured in **Section 6.6.2**.
  - Set **Next Hop Address** to the IP address of the Fusion Connect SIP server as configured in **Section 6.6.2**.
  - Select ***UDP*** for **Transport** (the transport will be displayed in the Next Hop Address field once the entry is added).



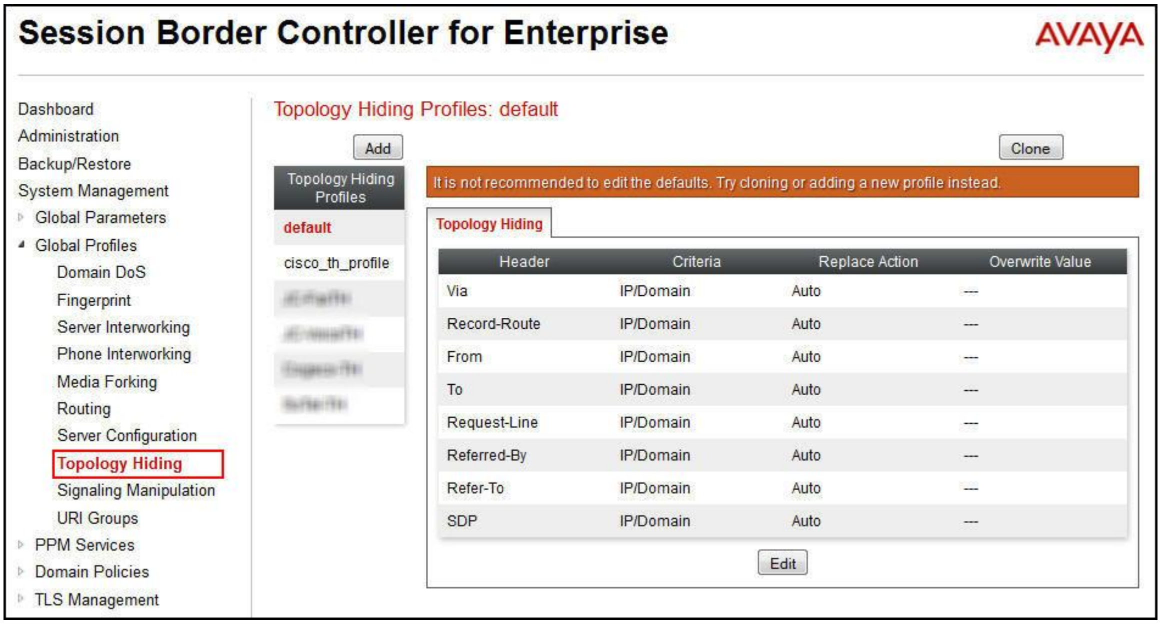
The following screen shows the routing profile for Fusion Connect when configured.



6.12. Topology Hiding

Topology hiding allows the host part of some SIP message headers to be modified in order to prevent private network information from being propagated to the untrusted public network. It can also be used as an interoperability tool to adapt the host portion of these same headers to meet the requirements of the connected servers. The topology hiding profile is applied as part of the end point flow in **Section 6.13**. For the compliance test, the predefined **default** topology hiding profile (shown below) was used for both Avaya IP Office and the Fusion Connect SIP servers.

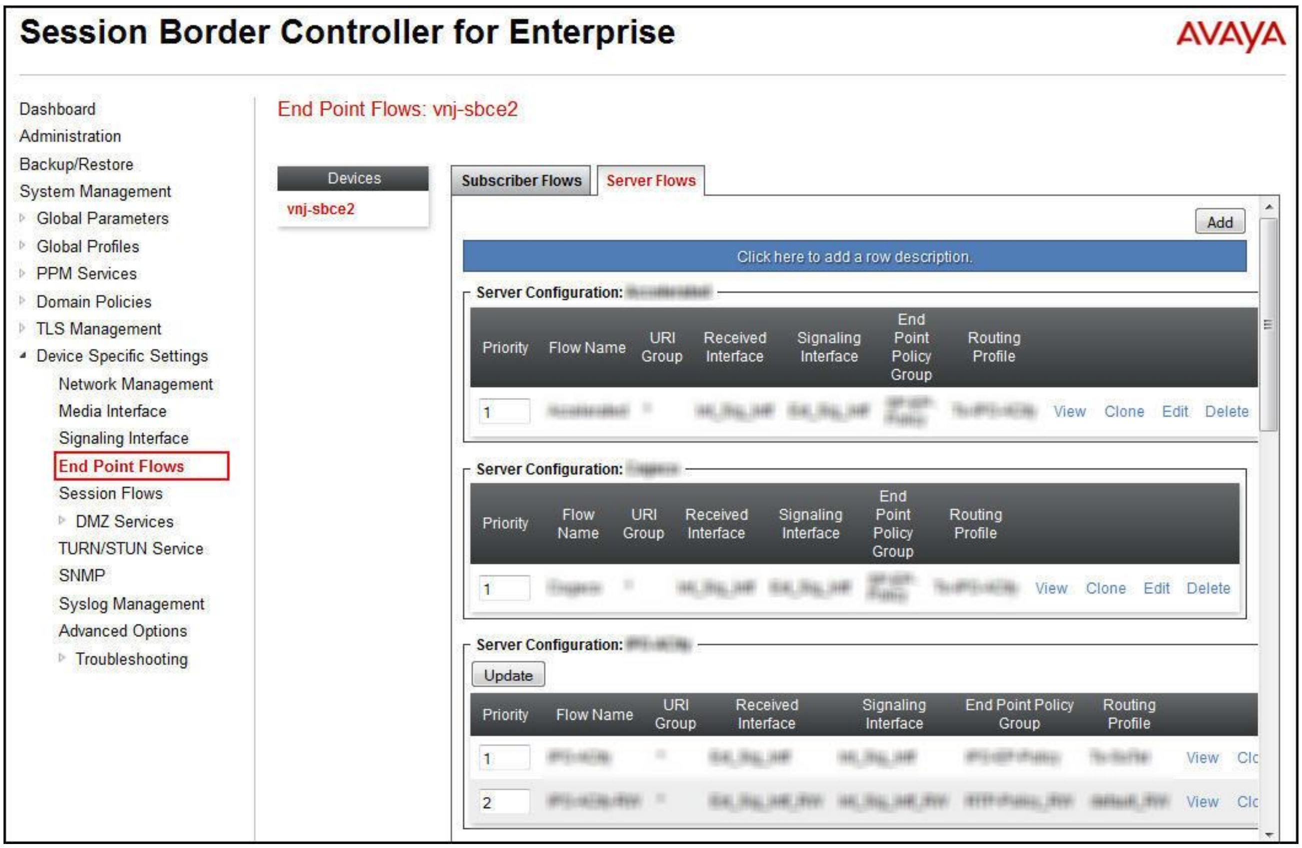
To add a new or view an existing profile, navigate to **Global Profiles → Topology Hiding** in the left pane. In the center pane, select **Add** to add a new profile, or select an existing profile (e.g., **default**) to be viewed.



6.13. End Point Flows

End point flows are used to determine the signaling endpoints involved in a call in order to apply the appropriate policies. When a packet arrives at the 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 policies and profiles which control processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for the destination endpoint are applied. Thus, two flows are involved in every call: the source end point flow and the destination end point flow. In the case of the compliance test, the signaling endpoints are Avaya IP Office and the Fusion Connect SIP server.

To create a new flow for a server endpoint, navigate to **Device Specific Settings → End Point Flows** in the left pane. In the center pane, select the Avaya SBCE device to be managed. In the right pane, select the **Server Flows** tab and click the **Add** button. A pop-up window (not shown) will appear requesting the name of the new flow and the flow parameters. Once complete, the configured flow is shown in the far right pane under the server name listed beside the **Server Configuration** heading.

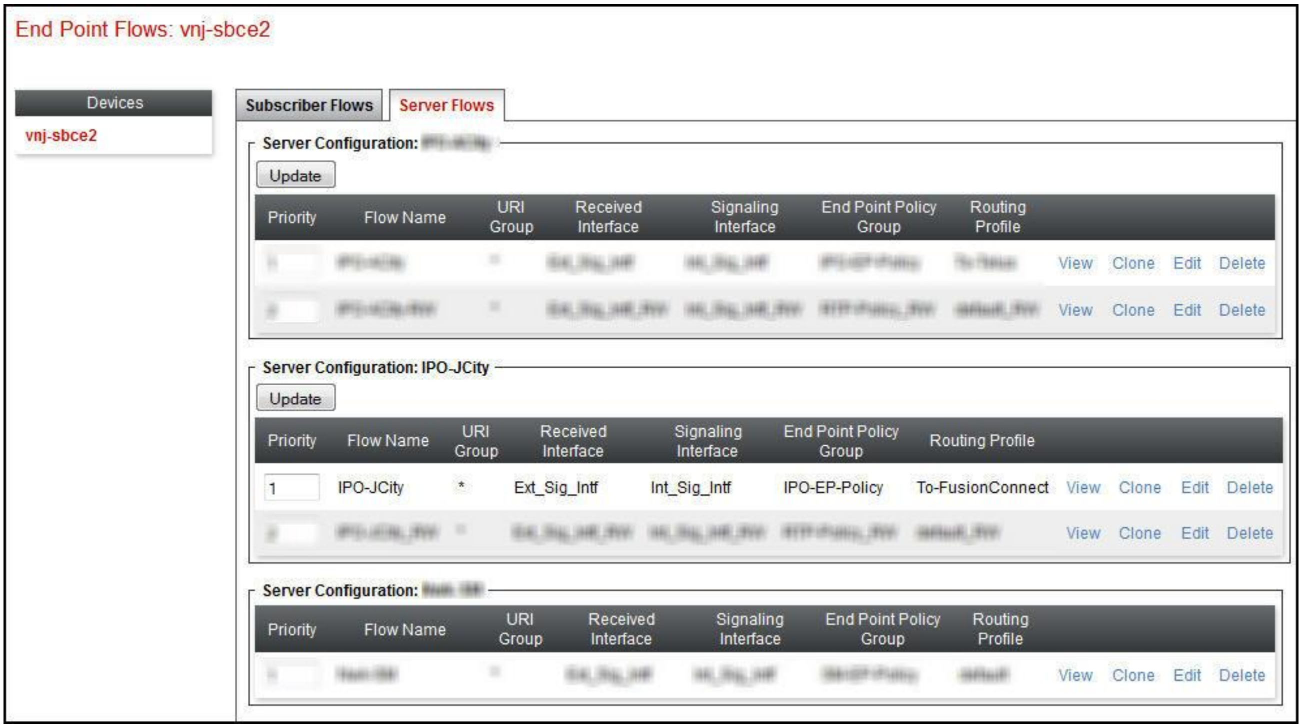


6.13.1. End Point Flow – Avaya IP Office

For the compliance test, the end point flow ***IPO-JCity*** was created for Avaya IP Office. All traffic from Avaya IP Office will match this flow as the source flow and use the specified routing profile ***To-FusionConnect*** to determine the destination server and corresponding destination flow. The **End Point Policy Group** and **Topology Hiding Profile** will be applied as appropriate. When creating the flow, configure the parameters as follows:

- For **Flow Name**, enter a descriptive name.
- For **Server Configuration**, select the Avaya IP Office server created in **Section 6.6.1**.
- To match all traffic, set the **URI Group**, **Transport**, and **Remote Subnet** to *****.
- Set **Received Interface** to the external signaling interface.
- Set **Signaling Interface** to the internal signaling interface.
- Set **Media Interface** to the internal media interface.
- Set **End Point Policy Group** to the endpoint policy group defined for Avaya IP Office in **Section 6.10.1**.
- Set **Routing Profile** to the routing profile defined in **Section 6.11.2** used to direct traffic to the Fusion Connect SIP server.
- Set **Topology Hiding Profile** to the topology hiding profile specified for Avaya IP Office in **Section 6.12**.

The screen below shows the saved **IPO-JCity** configuration as a Server Flow. Note the server name by the **Server Configuration** heading.

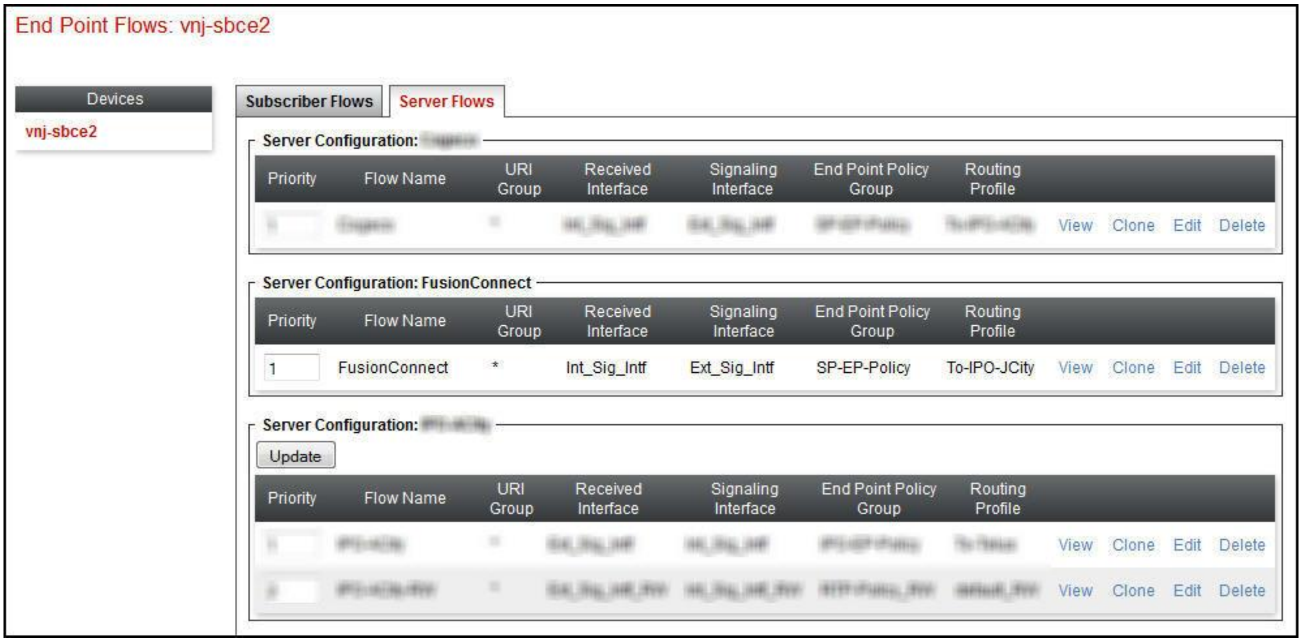


6.13.2. End Point Flow – Fusion Connect

For the compliance test, the end point flow ***FusionConnect*** was created for the Fusion Connect SIP server. All traffic from Fusion Connect will match this flow as the source flow and use the specified routing profile ***To-IPO-JCity*** to determine the destination server and corresponding destination flow. The **End Point Policy Group** and **Topology Hiding Profile** will be applied as appropriate. When creating the flow, configure the parameters as follows:

- For **Flow Name**, enter a descriptive name.
- For **Server Configuration**, select the Fusion Connect SIP server created in **Section 6.6.2**.
- To match all traffic, set the **URI Group**, **Transport**, and **Remote Subnet** to *****.
- Set **Received Interface** to the internal signaling interface.
- Set **Signaling Interface** to the external signaling interface.
- Set **Media Interface** to the external media interface.
- Set **End Point Policy Group** to the endpoint policy group defined for Fusion Connect in **Section 6.10.2**.
- Set **Routing Profile** to the routing profile defined in **Section 6.11.1** used to direct traffic to Avaya IP Office.
- Set **Topology Hiding Profile** to the topology hiding profile specified for Fusion Connect in **Section 6.12**.

The screen below shows the saved Fusion Connect configuration as a Server Flow. Note the server name by the **Server Configuration** heading.



7. Fusion Connect SIP Trunking Configuration

Clearfly is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise site (i.e., the IP address of the public interface on the Avaya SBCE) and the codec preferred (G.711u or G.729a). Fusion Connect will provide the customer the necessary information to configure the Avaya IP Office and Avaya SBCE including:

- Access interface IP address of the Fusion Connect SIP Trunking Service.
- Transport and port for the Fusion Connect SIP connection to the Avaya SBCE at the enterprise.
- DID numbers to assign to users at the enterprise.

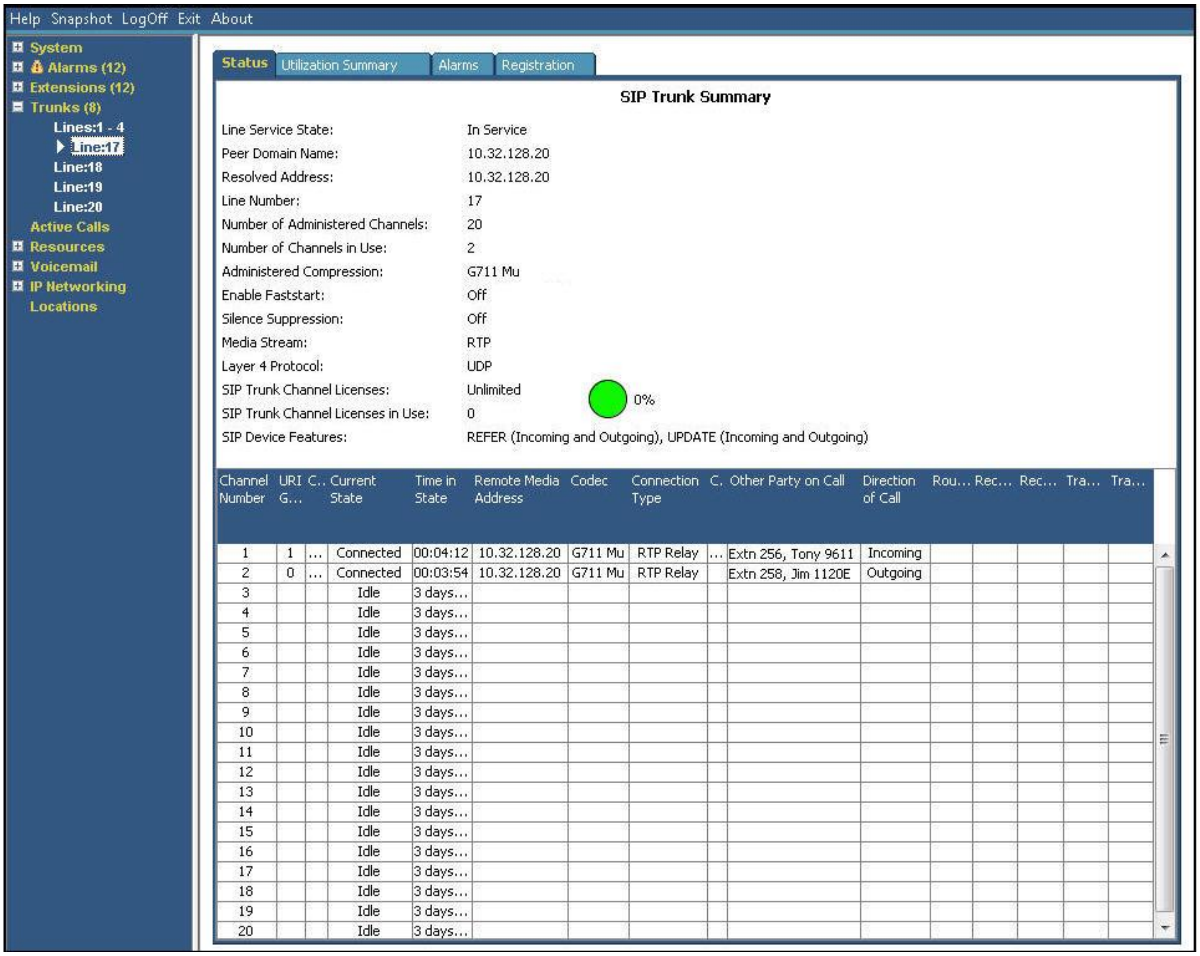
8. Verification Steps

This section provides verification steps that may be performed to verify the solution configuration.

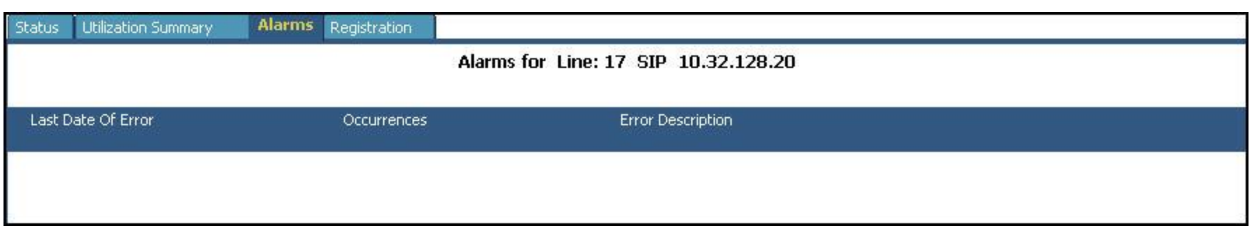
8.1. Avaya IP Office System Status

Use the Avaya IP Office System Status application to check the SIP Line channels state and alarms:

- Launch the application from **Start → Programs → IP Office → System Status** on the Avaya IP Office Manager PC. Select the SIP Line under **Trunks** from the left pane. In the **Status** tab in the right pane, verify the **Current State** is ***Idle*** for channels not taken by active calls; the state should be ***Connected*** for the channels engaged in active calls with the PSTN.

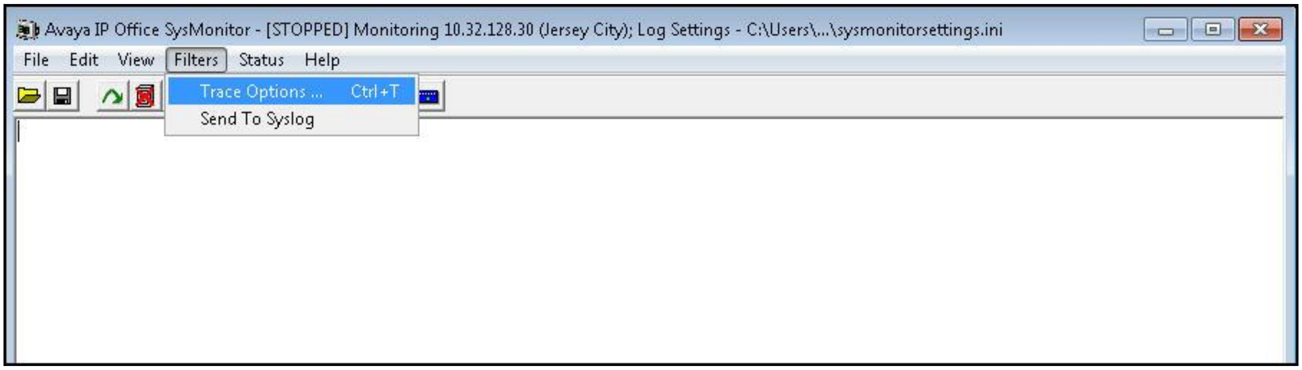


- Select the **Alarms** tab and verify that no alarms are active on the SIP Line.

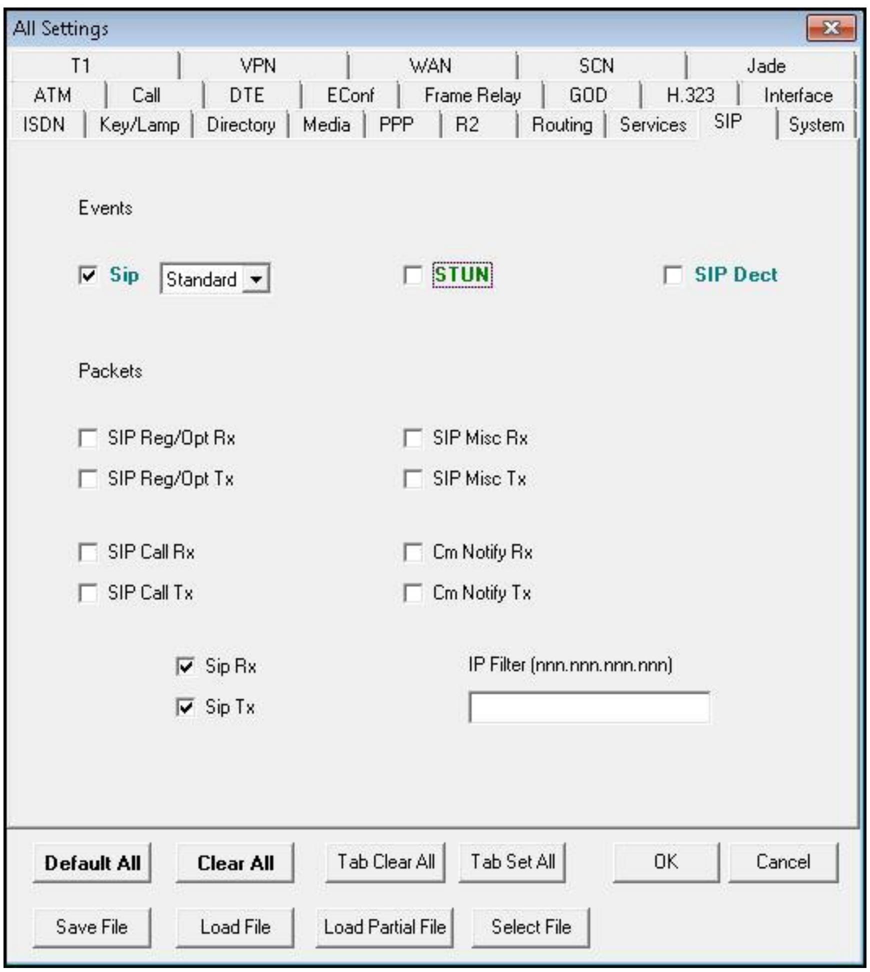


8.2. Avaya IP Office Monitor

The Monitor application can be used to monitor and troubleshoot Avaya IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor** on the Avaya IP Office Manager PC. The application allows monitored information to be customized. To customize, select **Filters → Trace Options…** as shown below:

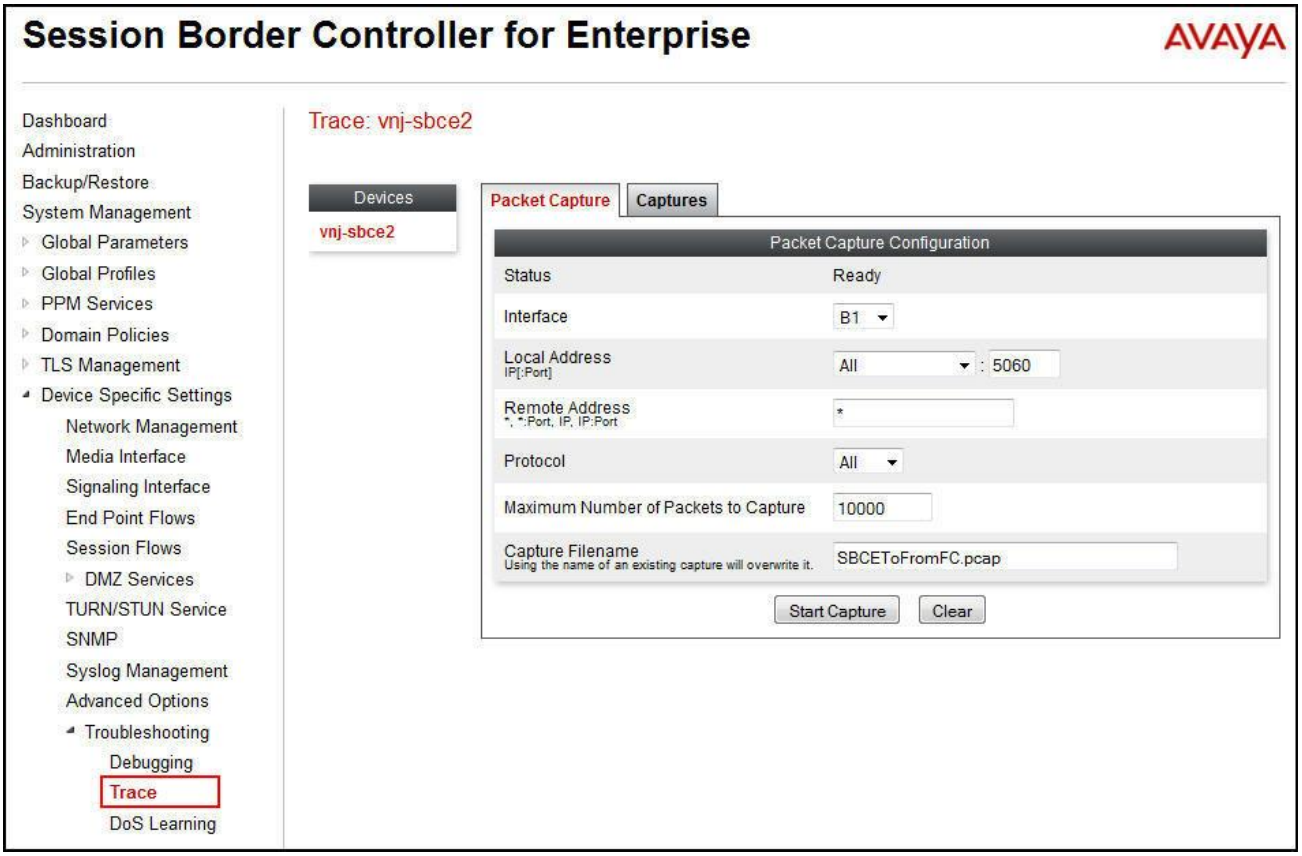


The following screen shows the **SIP** tab of trace options. In this example, **Standard Sip Events** and the **Sip Rx** and **Sip Tx** boxes are checked.



8.3. Avaya SBCE Traces

The Avaya SBCE can take traces on specified interfaces. SIP signaling crossing both interfaces A1 and B1 can be captured for troubleshooting. In the Avaya SBCE web interface, navigate to **Device Specific Settings → Troubleshooting → Trace** to invoke this facility. In the **Packet Capture** tab, select or supply the relevant information (e.g., A1 or B1 or any interfaces, IP/port, protocol, number of packets to capture, capture file name, etc.), then press the **Start Capture** button to start the trace. After the trace capture has been stopped, the captured trace file can then be downloaded from the **Captures** tab for examination using a protocol sniffer application such as Wireshark.



9. Conclusion

The Fusion Connect SIP Trunking Service passed compliance testing with Avaya IP Office R9.1 and Avaya Session Border Controller for Enterprise R6.3. These Application Notes describe the configuration necessary to connect Avaya IP Office R9.1 and Avaya SBCE R6.3 to Fusion Connect as shown in **Figure 1**. Test results and observations are noted in **Section 2.2**.

10. Additional References

[1] *IP Office™ Platform 9.1, Deploying Avaya IP Office™ Platform IP500 V2,* Document Number 15-601042, Issue 30g, January 2015.
[2] *Administering Avaya IP Office™ Platform with Manager*, Release 9.1, Issue 10.04, February 2015.
[3] *IP Office ™ Platform 9.1, Administering Avaya IP Office™ Platform Voicemail Pro*, Document Number 15-601063, Issue 10c, December 2014.
[4] *IP Office™ Platform 9.1, Using IP Office System Monitor*, Document Number 15-601019, Issue 06b, November 2014.
[5] *IP Office™ Platform 9.1, Using Avaya IP Office™ Platform System Status*, Document Number 15-601758, October 2014.
[6] *Using Avaya Communicator on IP Office*, Release 9.1, December 2014.
[7] *Deploying Avaya Session Border Controller for Enterprise*, Release 6.3, Issue 4, October 2014.
[8] *Administering Avaya Session Border Controller for Enterprise*, Release 6.3, Issue 4, October 2014.
[9] *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, February 2015.
[10] RFC 3261 *SIP: Session Initiation Protocol*, http://www.ietf.org/.
[11] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, http://www.ietf.org/.

Product documentation for Avaya products may be found at http://support.avaya.com or http://marketingtools.avaya.com/knowledgebase/ipoffice/general/rss2html.php?XMLFILE=manuals.xml&TEMPLATE=pdf_feed_template.html.

Product documentation for Fusion Connect SIP Trunking Service is available from Fusion Connect.

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.