Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Voice Carrier IntelliSIP Trunking Service with Avaya IP Office Release 8.1 and Avaya Session Border Controller for Enterprise 6.2 – Issue 1.0

Abstract

These Application Notes describes the steps to configure Session Initiation Protocol (SIP) Trunking between Voice Carrier and an enterprise solution using Avaya IP Office Release 8.1 and Avaya Session Border Controller for Enterprise 6.2.

The Voice Carrier IntelliSIP Trunking Service provides PSTN access via a SIP trunk between the enterprise and the Voice Carrier network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise. Voice Carrier 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 describes the steps to configure Session Initiation Protocol (SIP) Trunking between Voice Carrier and an enterprise solution using Avaya IP Office Release 8.1 and Avaya Session Border Controller for Enterprise 6.2.

The Voice Carrier IntelliSIP Trunking Service referenced within these Application Notes is positioned for customers that have an IP-PBX or IP-based network equipment with SIP functionality, but need a form of IP transport and local services to complete their solution.

The Voice Carrier IntelliSIP Trunking Service will enable delivery of origination and termination of local, long-distance and toll-free traffic across a single broadband connection.

2. General Test Approach and Test Results

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.

The general test approach was to connect a simulated enterprise site to the Voice Carrier IntelliSIP Trunking Service via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of Avaya IP Office, Avaya Session Border Controller for Enterprise (Avaya SBCE) and various Avaya endpoints.

The Voice Carrier IntelliSIP Trunking Service passed compliance testing with the observations or limitations described in **Section 2.2**.

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- Use of UDP or TCP for the transport layer to the service provider
- Sending and receiving SIP OPTIONS queries to the service provider
- Incoming PSTN calls to SIP and H.323 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 SIP and H.323 at the enterprise. All outgoing PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from soft clients. Avaya IP Office Video Softphone and Avaya Flare® Experience for Windows were tested.
- Various call types including: local, long distance, outbound toll-free, international and local directory assistance.
- Codec G.711MU

- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation using DTMF for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and twinning
- G.711 pass-through fax
- REFER message for network call redirection

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

The following items are not supported:

- G.729 codec
- Operator (dial 0) and operator-assisted (dial 0 + 10 digits) services
- T.38 Fax

2.2. Test Results

Interoperability testing of the Voice Carrier IntelliSIP Trunking Service was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **Codec not locked down on outbound calls**: On outbound calls, Voice Carrier responds to the INVITE request with multiple codecs, instead of selecting one from the INVITE SDP list. IP Office uses the first compatible codec in the list. This behavior has no user impact. Calls were successful.
- **Inconsistent SIP headers on inbound Calling Party Number blocked calls:** On inbound Calling Party Number (CPN) blocked calls, Voice Carrier sends a SIP INVITE message containing two headers which contradict each other. The INVITE contains the Privacy = none header which indicates that the calling party number should not be blocked. The INVITE also contains the Remote-Party-ID header with parameter privacy=full which indicates that the calling party number should be blocked. In this scenario, IP Office blocked the calling party number as intended. The call was successful.
- **Blind transfer with Avaya 1100 IP Deskphones calling party number**: An Avaya 1100 Series IP Deskphone (SIP) places an outbound call to the PSTN, then performs a blind transfer of this call to another PSTN endpoint. In this scenario, the wrong calling party number is displayed on the second PSTN endpoint. The display shows the number dialed by the Avaya 1100 Series IP Deskphone for the initial outbound call including the short code (9) + 1 + 10 digits. The behavior is expected to be the same as with IP Office H.323 endpoints which is to display the transferring party number. This is an IP Office issue and is under investigation by the IP Office development team.

2.3. Support

For technical support on the Voice Carrier IntelliSIP Trunking Service, contact Voice Carrier using the Support links at www.voicecarrier.com or by calling 1-888-830-6230.

3. Reference Configuration

Figure 1 illustrates the sample configuration used for the DevConnect compliance testing. The sample configuration shows an enterprise site connected to the Voice Carrier IntelliSIP Trunking Service.

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

The enterprise site contains an Avaya IP Office 500 V2 with various endpoints and a Windows 2003 Server running both Avaya IP Office Manager to configure the Avaya IP Office and Avaya Preferred Edition for voicemail (also known as VoiceMail Pro).

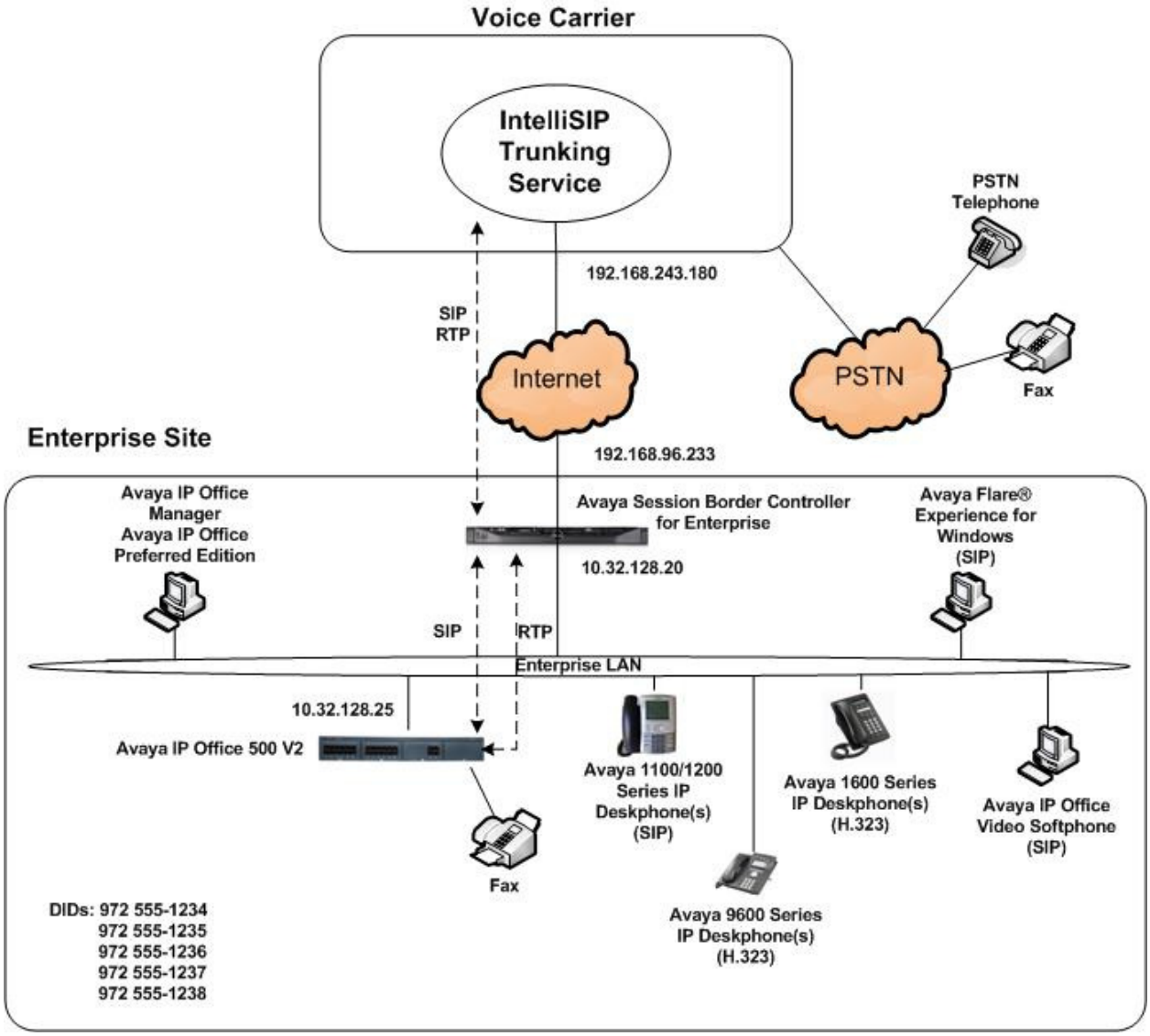


Figure 1: Avaya Interoperability Test Lab Configuration

For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with numbers that cannot be routed over the PSTN.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Voice Carrier. The short code of 9 and any preceding 1 is stripped off by Avaya IP Office and the remaining 10 digits were sent unaltered to Voice Carrier. For outbound calls, Avaya IP Office sent 10 digits in all headers. For inbound calls, the Voice Carrier IntelliSIP Trunking Service 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 were used for the sample configuration provided:

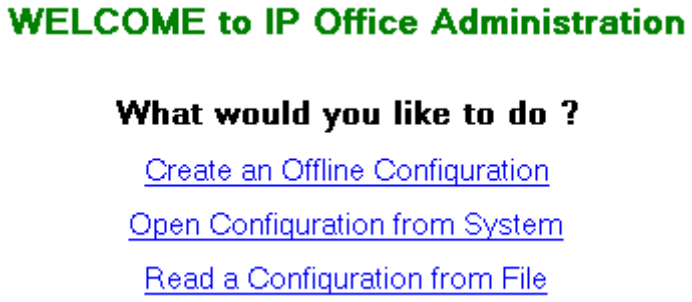
Avaya Telephony Components	
Equipment	**Software**
Avaya IP Office 500 v2	8.1 (69)
Avaya IP Office Manager	10.1 (69)
Avaya IP Office Preferred Edition (Voicemail)	8.1 (9203)
Avaya Session Border Controller for Enterprise running on a Portwell CAD-0208 server	6.2.0.Q36
Avaya 1140E IP Deskphone (SIP)	4.3 SP1 (04.03.12.00)
Avaya 1608 IP Deskphone (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP3 (1.3.3)
Avaya 9641G IP Deskphone (H.323) running Avaya one-X® Deskphone Edition	6.2 SP2 (6.2209)
Avaya IP Office Video Softphone (SIP)	3.2.3.48 (67009)
Avaya Flare® Experience for Windows	1.1.2.11

Voice Carrier Components	
Equipment	**Software**
IntelliSIP	2.1 – av1.6.2.18

Testing was performed with IP Office 500 V2 R8.1, but this testing also applies to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 R8.1 to support analog or digital endpoints or trunks. Also, IP Office 500 V2 does not support SIP Direct Media so Server Edition SIP Direct Media functionality was not compliance tested.

5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the Avaya IP Office Manager PC, select **Start → Programs → IP Office → Manager** to launch the application. A screen that includes the following in the center may be displayed:



Select **Open Configuration from System**. If the above screen does not appear, the configuration may be alternatively opened by navigating to **File → Open Configuration** at the top of the Avaya IP Office Manager window. Select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials.

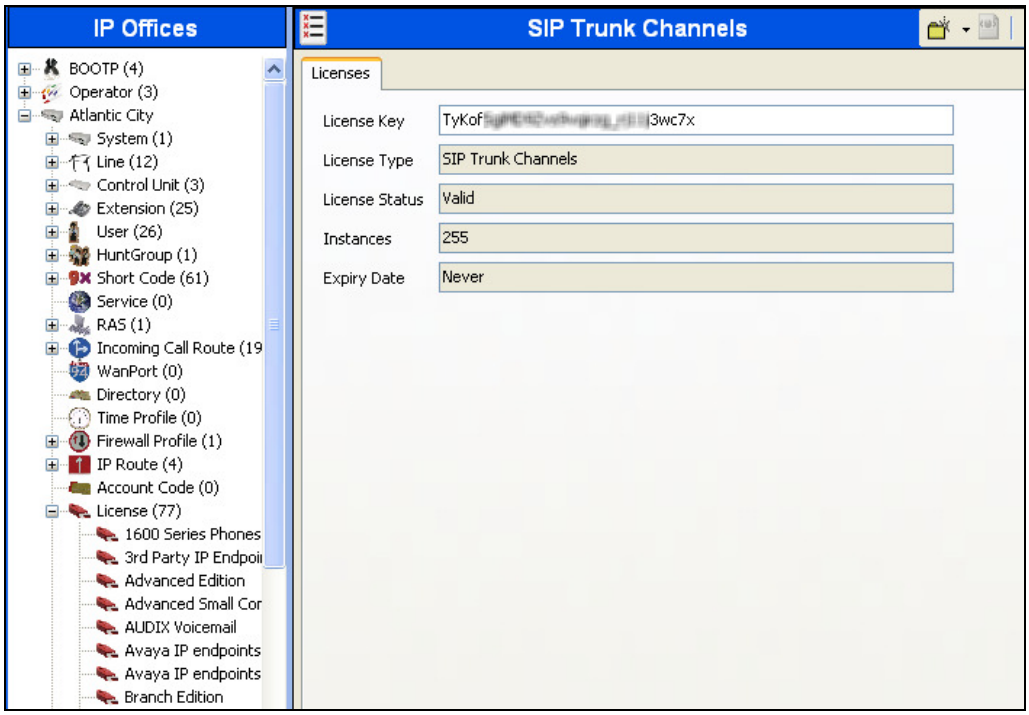
The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and show the Details pane on the right side. Since the Group Pane has been omitted, its content is shown as submenus in the Navigation pane. These panes (Navigation, Group 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 twinning and IP Office Video Softphone support) is assumed to already be in place.

In the sample configuration, **Atlantic City** was used as the system name. All navigation described in the following sections (e.g., **License → SIP Trunk Channels)** appears as submenus underneath the system name **Atlantic City** in the Navigation Pane.

5.1. Licensing and Physical Hardware

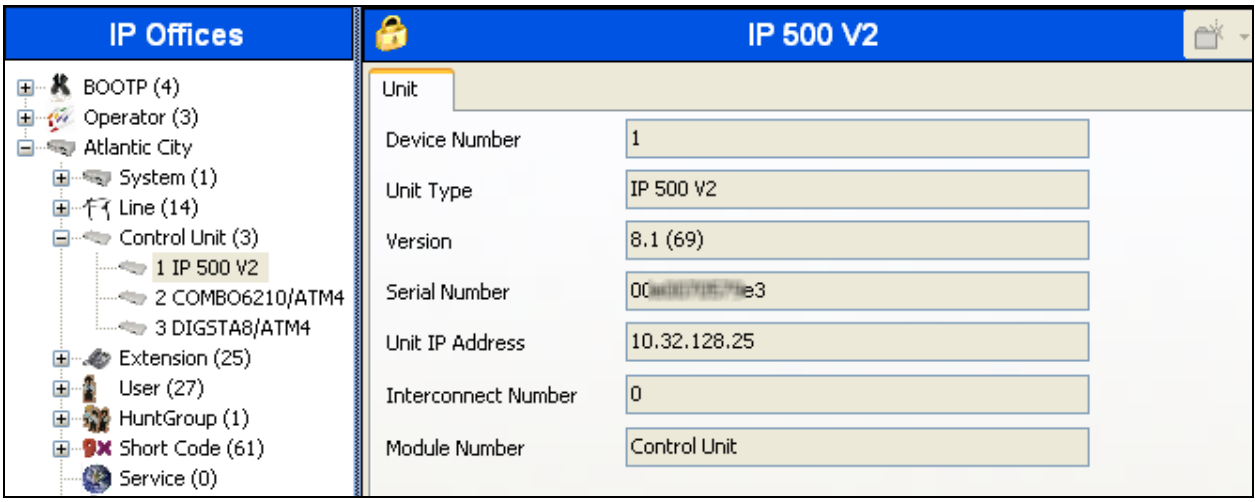
The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License → SIP Trunk Channels** in the Navigation pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details pane.



To view the physical hardware comprising the IP Office system, expand the components under the **Control Unit** in the Navigation pane. In the sample configuration, the second component listed is a Combination Card. This module has 6 digital stations ports, two analog extension ports, 4 analog trunk ports and 10 VCM channels. The VCM is a Voice Compression Module supporting VoIP codecs. An 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 following screen shows the details of the **IP 500 V2**.

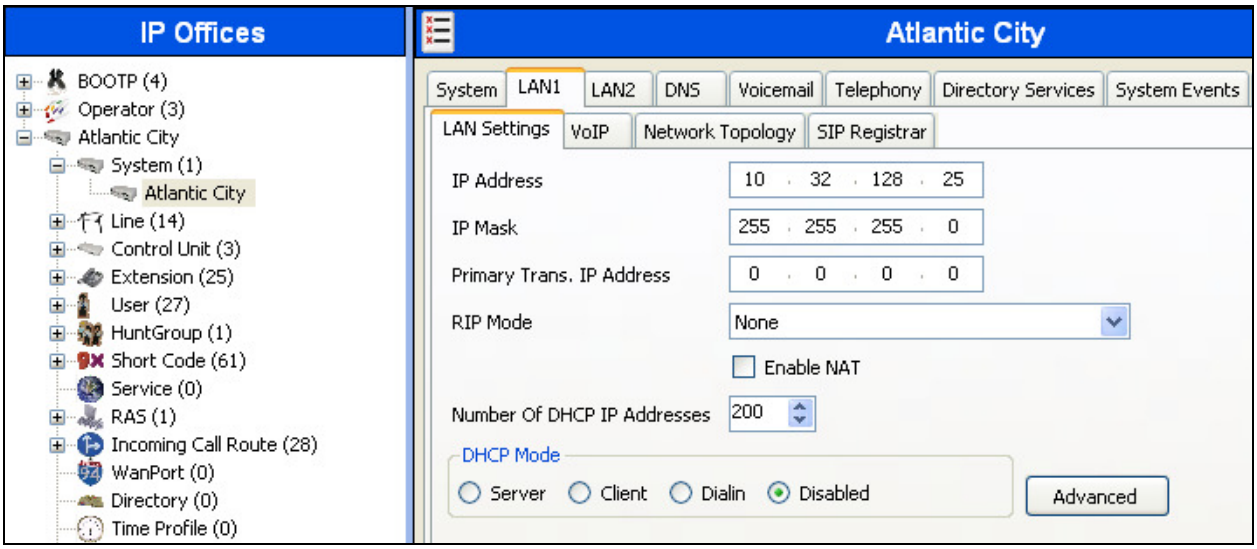


5.2. System

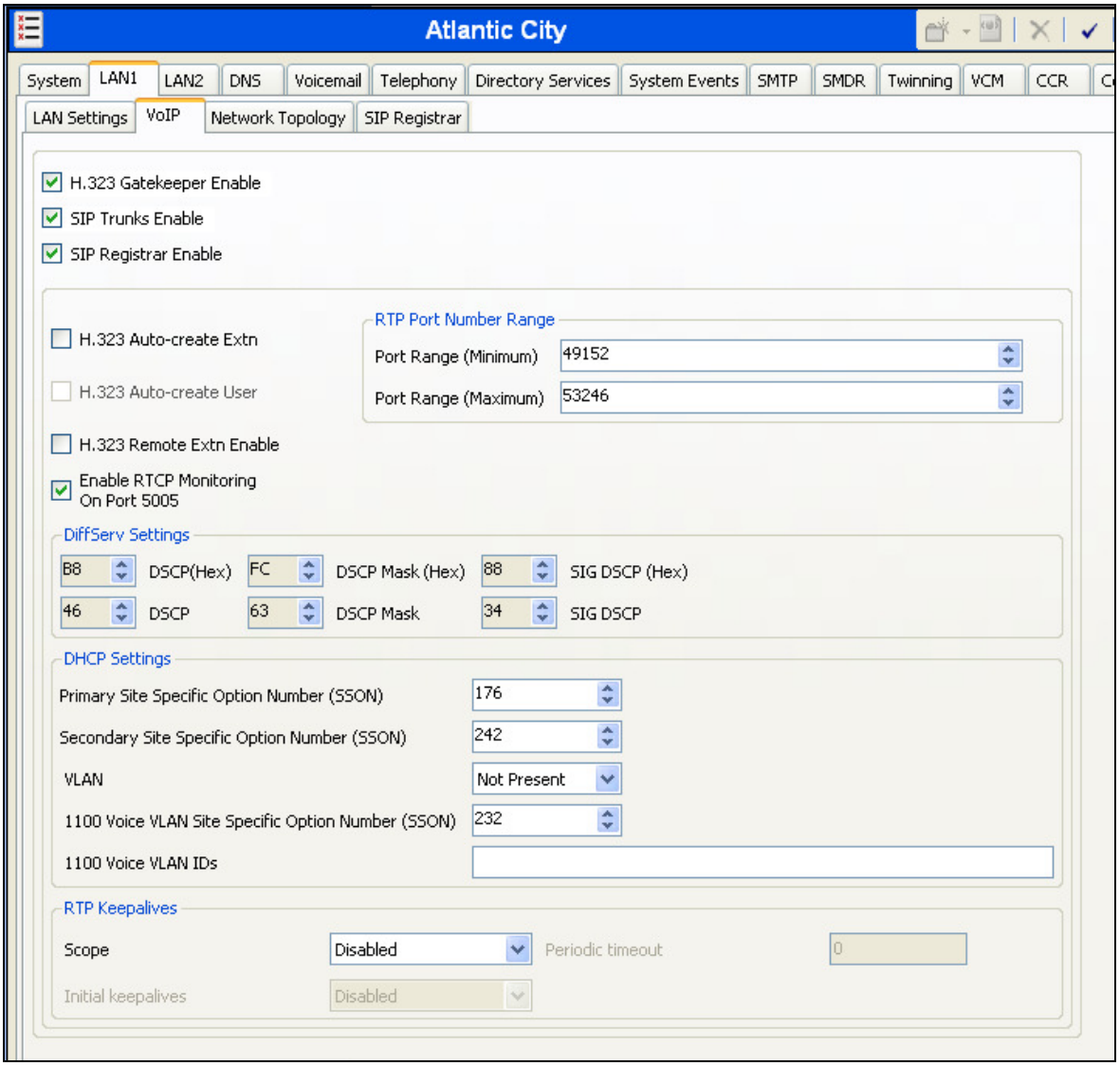
Configure the necessary system settings.

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 **Atlantic 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. All other parameters should be set according to customer requirements.

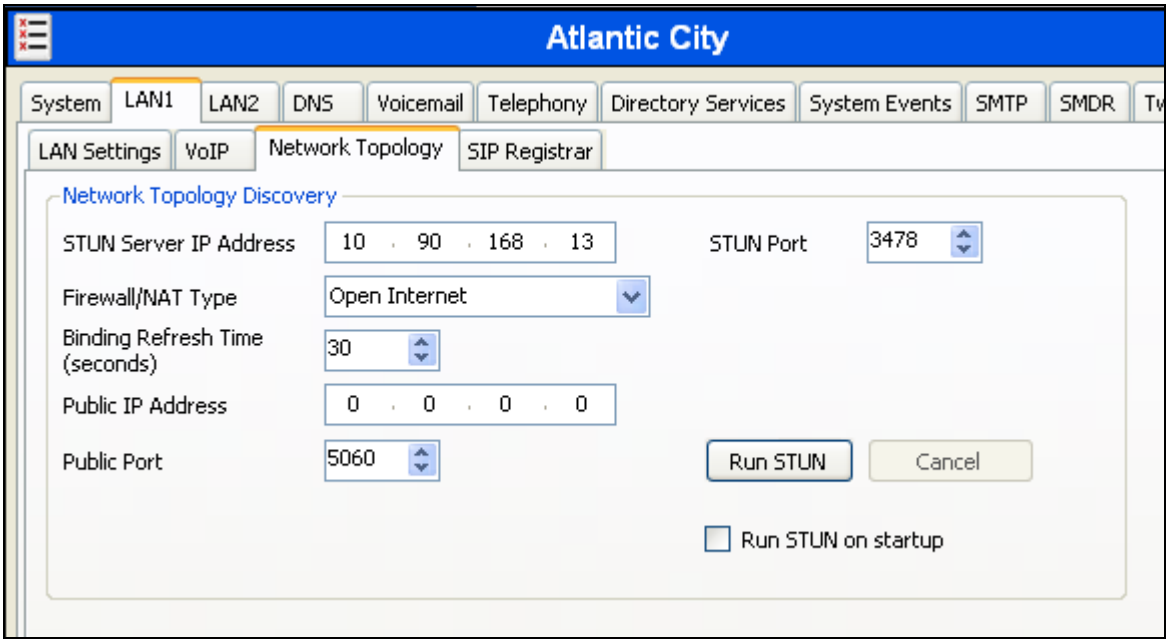


On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN1. Avaya IP Office can also 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 example below and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values will be provided by Voice Carrier. All other parameters should be set according to customer requirements.



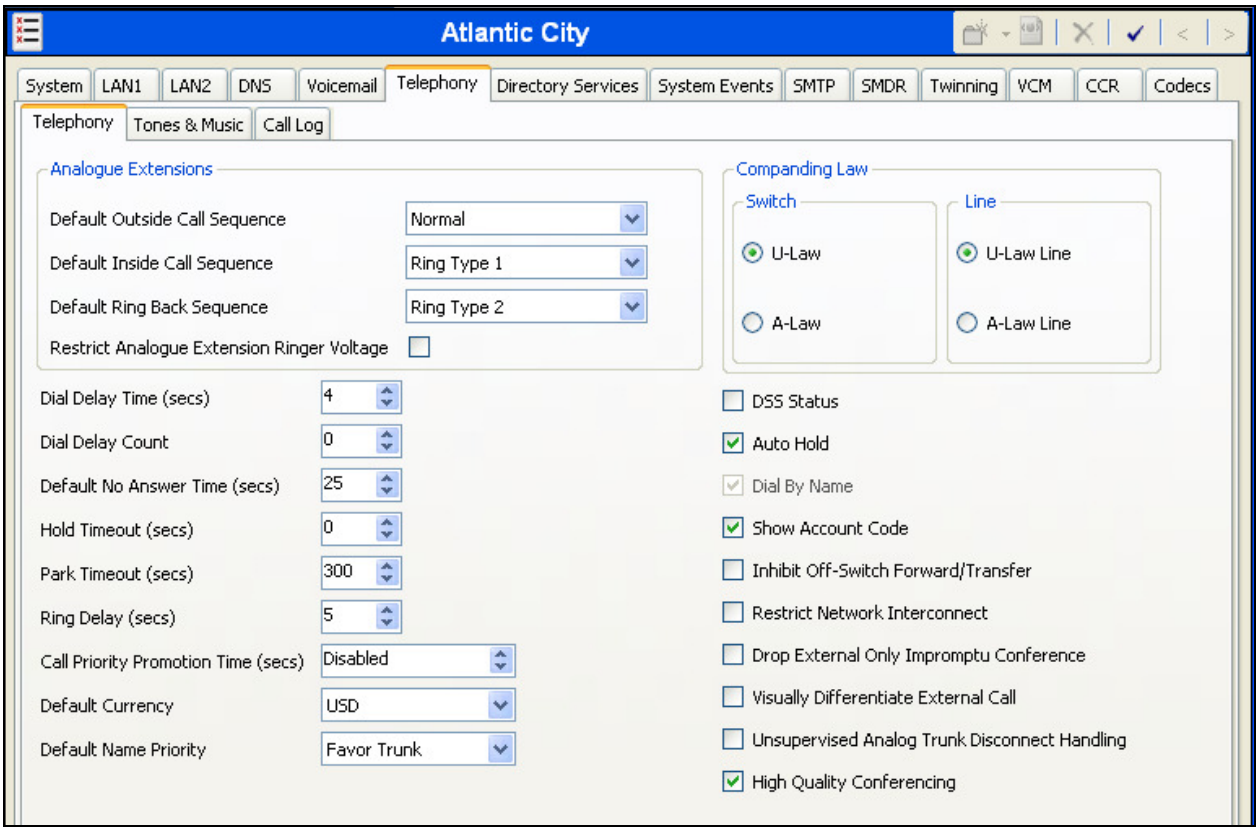
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. The Avaya SBCE will perform network address translation of SIP traffic but it is not necessary for IP Office to have any knowledge of this translation. Thus, the parameter was set to **Open Internet**.
- Set **Binding Refresh Time (seconds)** to **30**. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set the **Public Port** to the port Avaya IP Office will listen on.
- All other parameters should be set according to customer requirements.



5.2.2. System - Telephony Tab

To access the System Telephony settings, navigate to the **Telephony → Telephony** tab in the Details Pane. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN. If for security reasons incoming calls should not be allowed to transfer back to the PSTN then leave this setting checked.



5.2.3. System - Twinning Tab

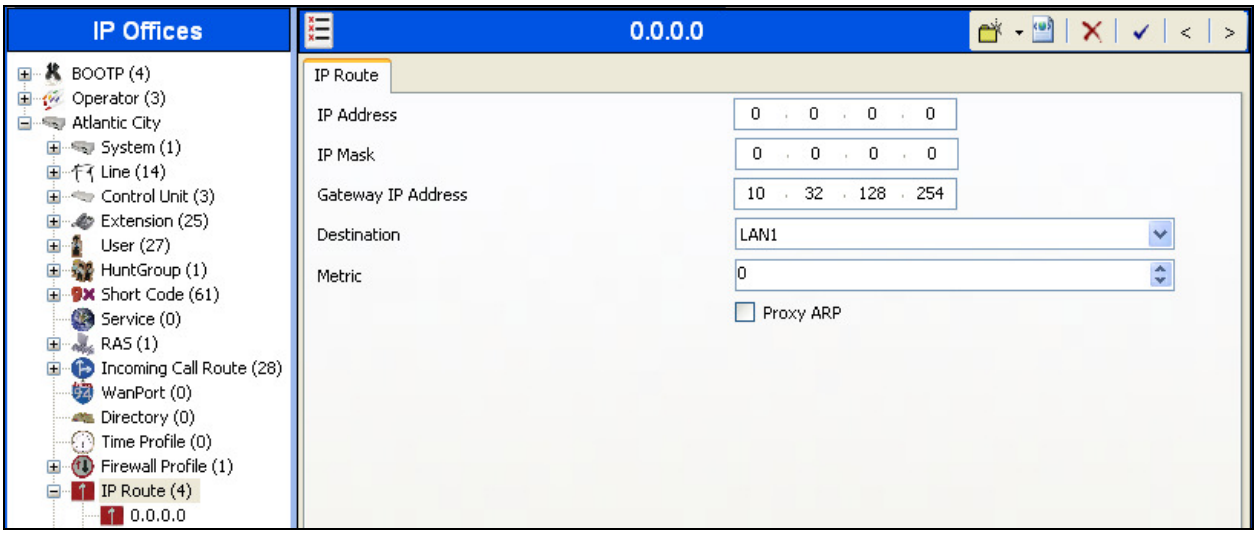
To view or change the System Twinning settings, navigate to the **Twinning** tab in the Details Pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning**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

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.** Create/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 default router for the enterprise network.
- Set **Destination** to **LAN1** from the drop-down list.

Click the **OK** button at the bottom of the page (not shown).



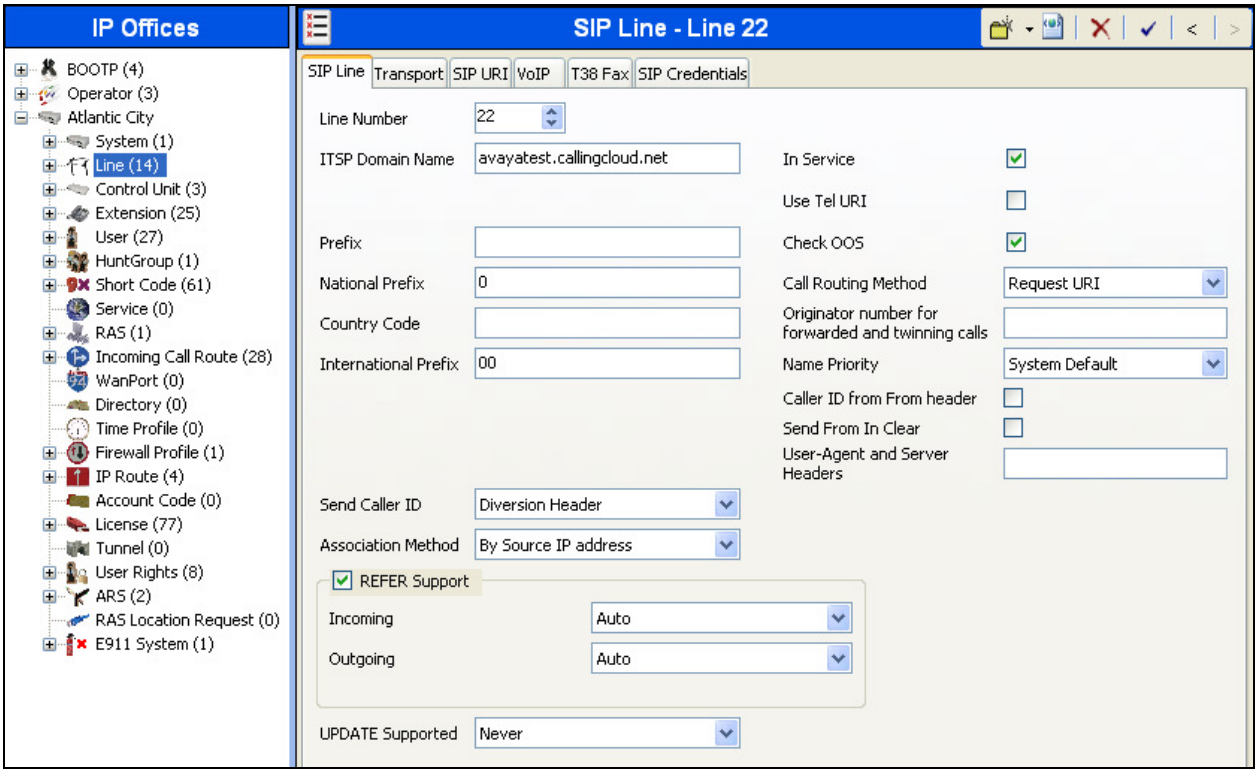
5.4. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Voice Carrier IntelliSIP Trunking Service. To create a SIP line, right-click **Line** in the Navigation Pane and select **New → SIP Line**.

5.4.1. SIP Line – SIP Line Tab

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

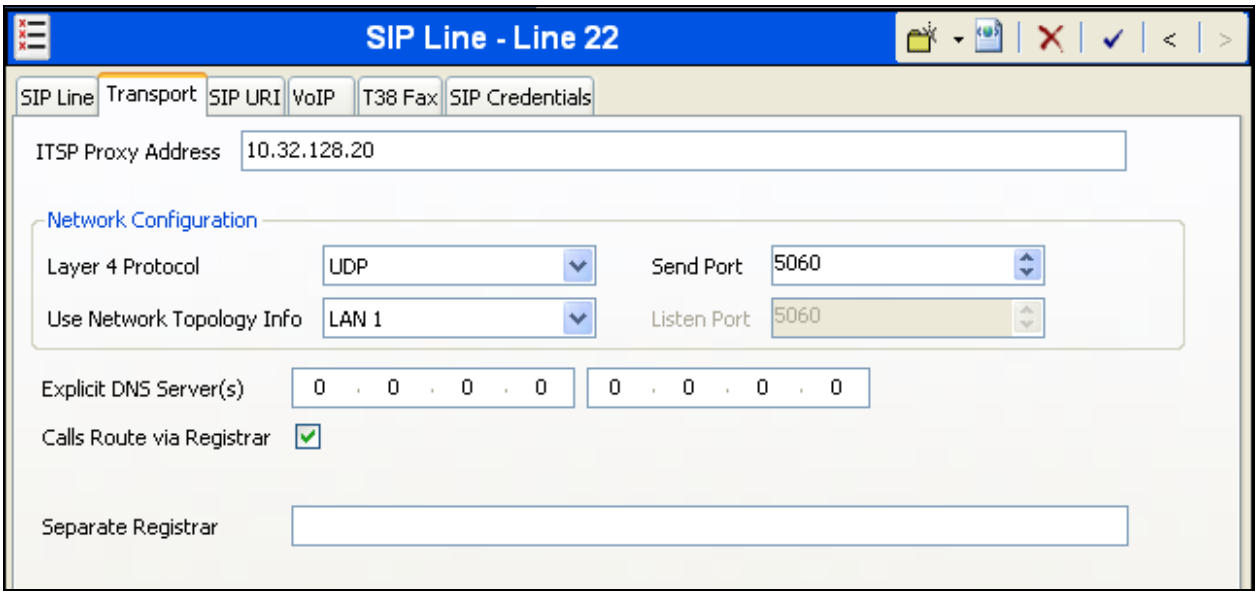
- Set **ITSP Domain Name** to the domain provided by Voice Carrier.
- Set **Send Caller ID** to **Diversion Header**. With this setting and the related configuration in **Section 5.2.3**, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Voice Carrier. It will also include the Diversion Header for calls that are call forwarded out the SIP Line.
- Check **REFER Support**.
- Check the **In Service** box. This makes the trunk available to incoming and outgoing calls.
- Check the **Check OOS** box. IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN1, as shown in **Section 5.2.1**.
- Default values may be used for all other parameters.



5.4.2. SIP Line - Transport Tab

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

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

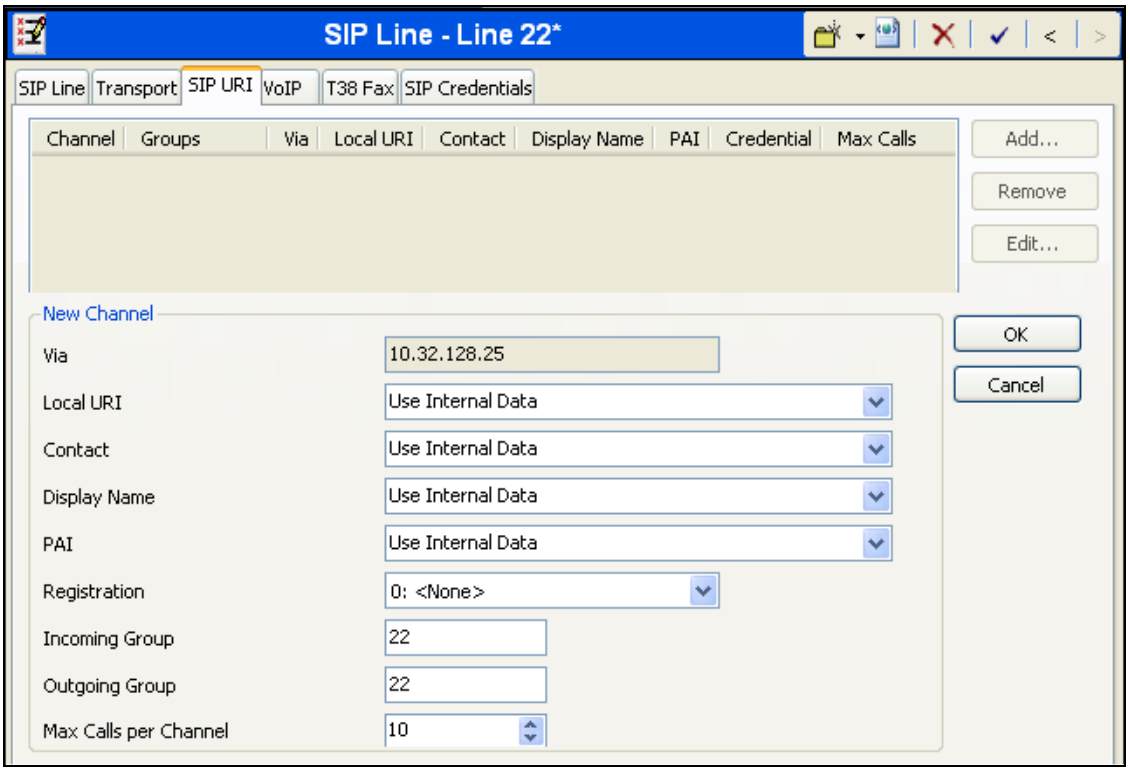


5.4.3. SIP Line - SIP URI Tab

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit** button. In the example screen below, a new entry is created. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact**, **Display Name** and **PAI** to **Use Internal Data**. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any User as shown in **Section 5.7**.
- For **Registration**, select **0: <None>** from the pull-down menu.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line in **Section 5.8**. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining ARS entries for routing outbound traffic to this line in **Section 5.6**. For the compliance test, a new incoming and outgoing group **22** was defined that only contained this line (line 22).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

Click **OK**.



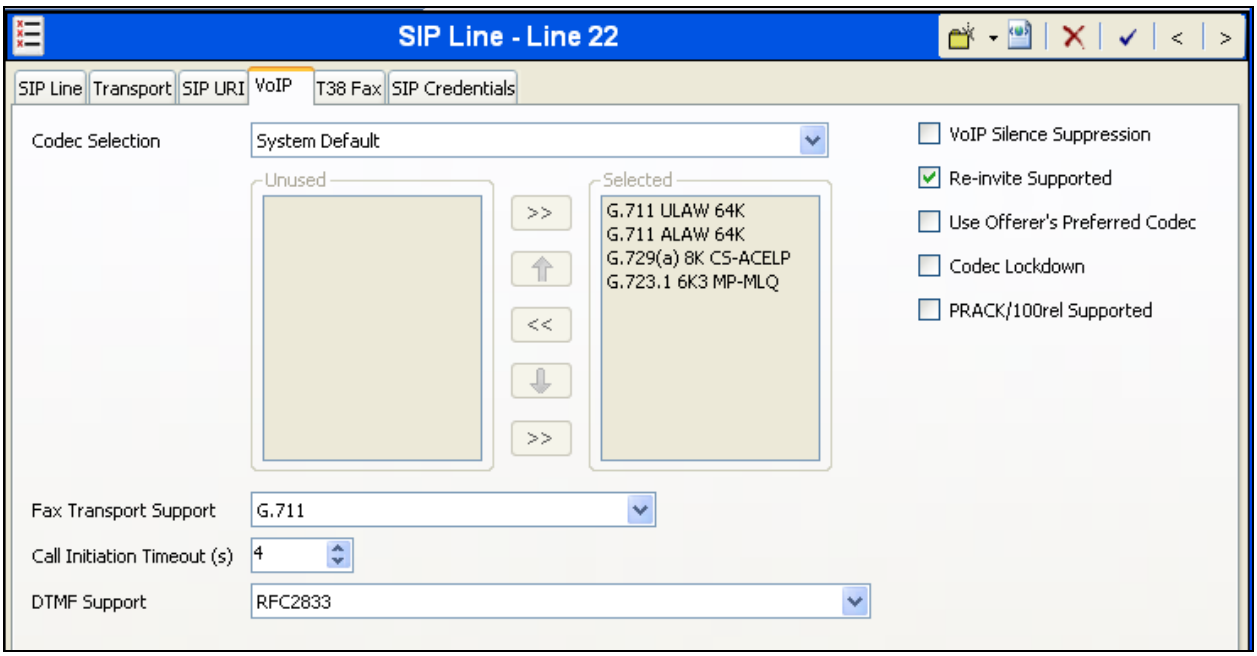
Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user such as a short code. These URIs are created in the same manner as shown above with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, **Display Name** and **PAI** fields.

5.4.4. SIP Line - VoIP Tab

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

- For **Codec Selection**, select **System Default** from the pull-down menu. A list of the codecs in their current order of preference is shown on the right in the **Selected** column. The compliance test used the default codec list. To use a custom list of codecs, select **Custom** for **Codec Selection**. Next, move unwanted codecs from the **Selected** column to the **Unused** column. Lastly, move the codecs up or down the list in the **Selected** column to achieve the desired order of preference.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Set the **Fax Transport Support** to **G.711**.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters.

Click the **OK** button at the bottom of the page (not shown).

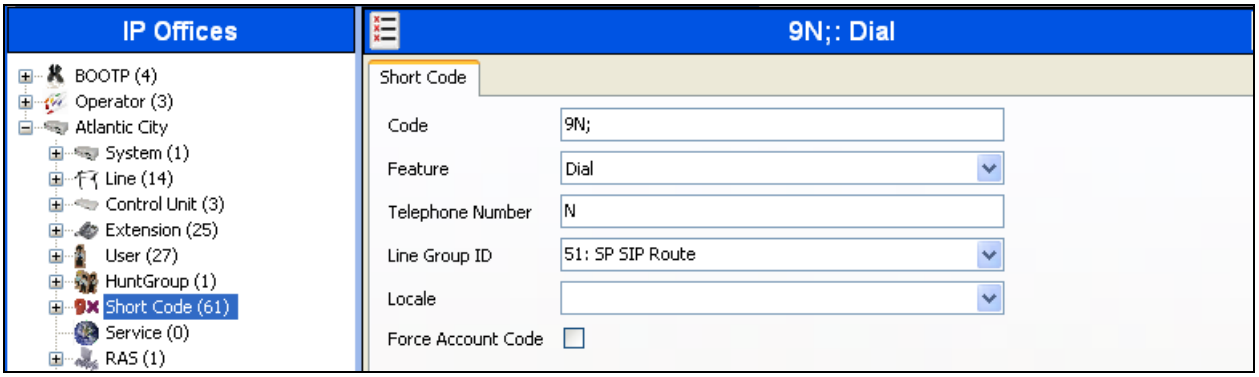


5.5. Short Codes

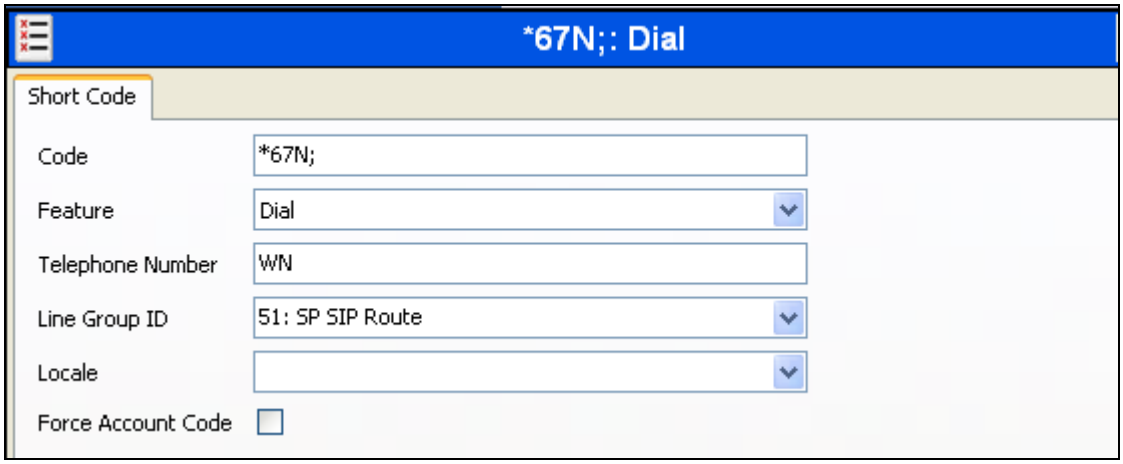
ARS is used to route outbound traffic to the SIP line. A short code is used to route outbound traffic to ARS. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;**. This short code 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**. The value **N** represents the number dialed by the user after removing the **9** prefix. This value is passed to ARS.
- Set the **Line Group Id** to the ARS route to be used which is defined in **Section 5.6**.

Click the **OK** button (not shown).



Optionally, add or edit a short code that can be 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 SIP Line to Voice Carrier documented in these Application Notes, when a user dials *67 plus the number, IP Office will include the user's telephone number in the P-Asserted-Identity (PAI) header and will include the Privacy: Id header. Voice Carrier will allow the call due to the presence of a valid DID in the PAI header, but will prevent presentation of the caller id to the called PSTN destination.



5.6. ARS

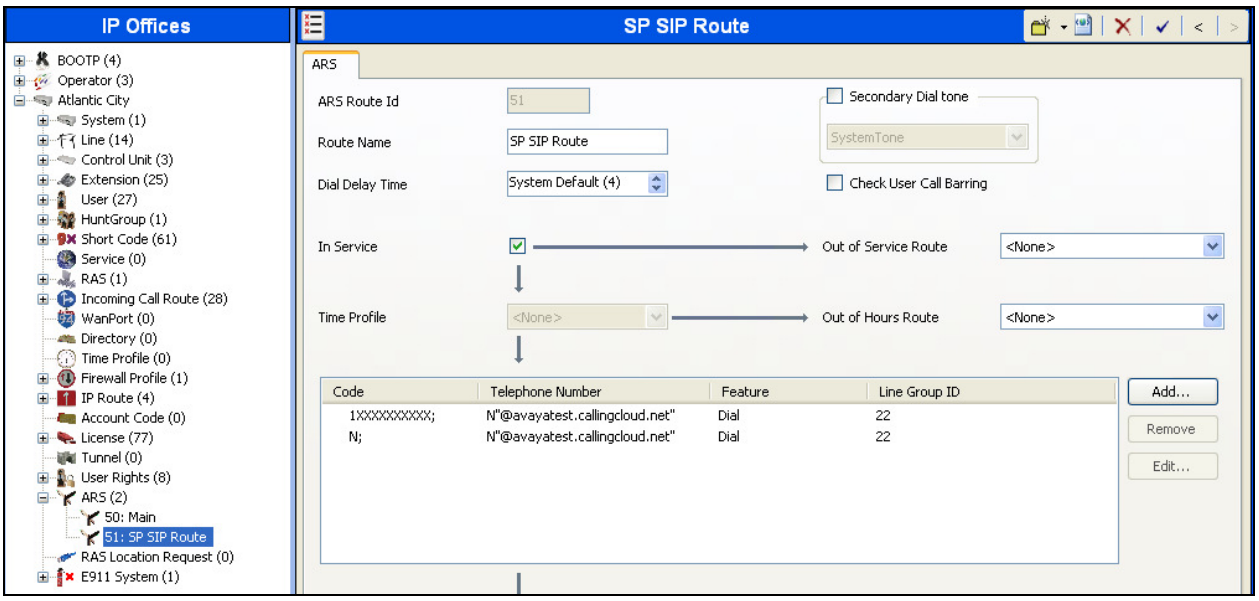
ARS is used to route outbound traffic to the SIP line. To define a new ARS route, right-click **ARS** in the Navigation pane and select **New**. In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

For the compliance test, two entries were created. The first entry matches on any 1 + 10 digit number (**1XXXXXXXXXX**) and then sends only 10 digits (**N**) in the SIP INVITE message on the line group defined in **Section 5.4.3** (e.g., line group 22). The second entry matches on any other number (**N**) and passes it unaltered to the line group.

To create an entry, click the **Add** button and enter the following in the pop-up window.

- In the **Code** field, enter the pattern to match the number passed to ARS from the short code in **Section 5.5** followed by a semi-colon.
- Set **Feature** to **Dial.** This is the action that the short code will perform.
- Set **Telephone Number** to **N"@avayatest.callingcloud.net"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. In the first entry, **N** represents the remaining 10 digits after removing the preceding 1. In the second entry, the value **N** represents the complete number passed to ARS. The domain **avayatest.callingcloud.net** is the service provider domain provided by Voice Carrier.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4.3**. This short code will use this line group when placing the outbound call.

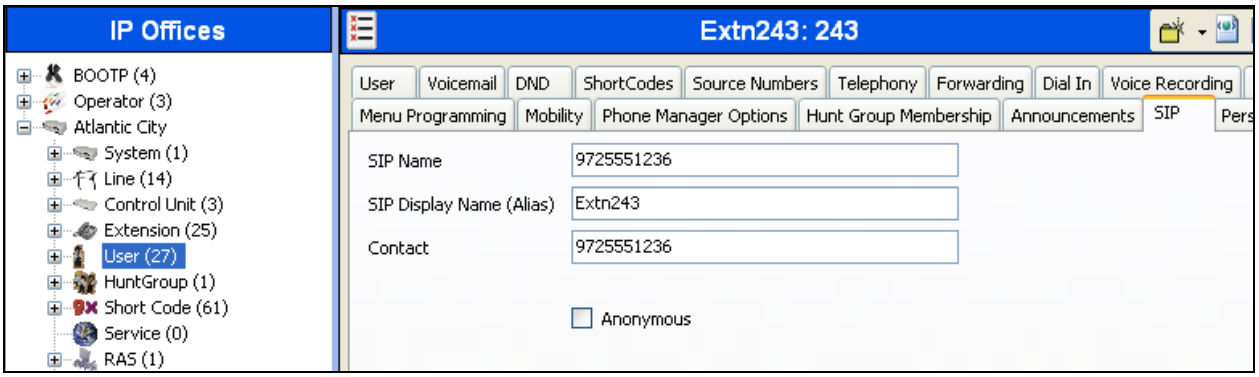
Click the **OK** button (not shown).



5.7. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. 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 **Extn243**. 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 and Contact headers for outgoing SIP trunk calls and 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 5.4.3**). The example below shows the settings for User **Extn243**. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Voice Carrier. 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 the **OK** button (not shown).



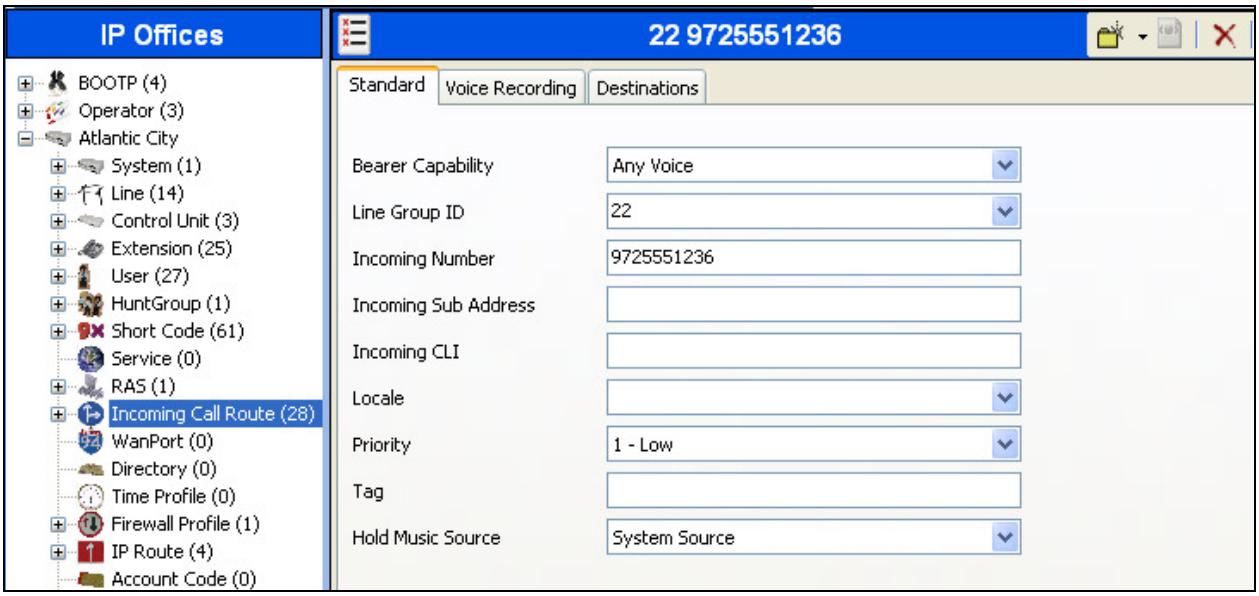
5.8. Incoming Call Route

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, right-click **Incoming Call Routes** in the Navigation Pane and select **New**.

5.8.1. Incoming Call Route – Standard Tab

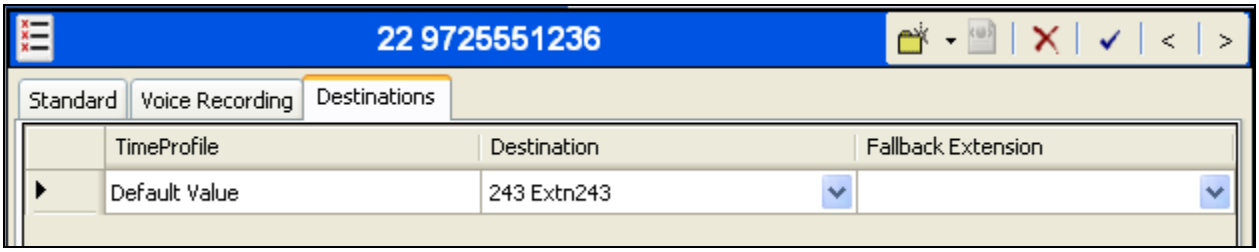
On the **Standard** tab of the Details Pane, enter the parameters as shown below.

- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4.3**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Default values can be used for all other fields.



5.8.2. Incoming Call Route – Destinations Tab

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 9725551236 on line 22 are routed to extension 243.

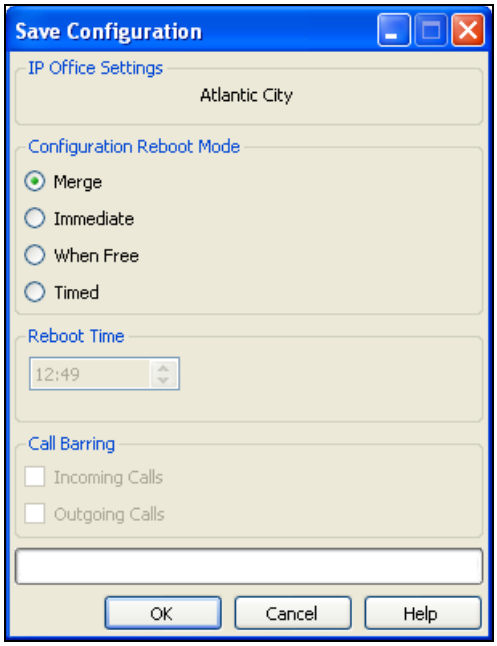


Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Figure 1** are omitted here, but can be configured in the same fashion.

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.

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



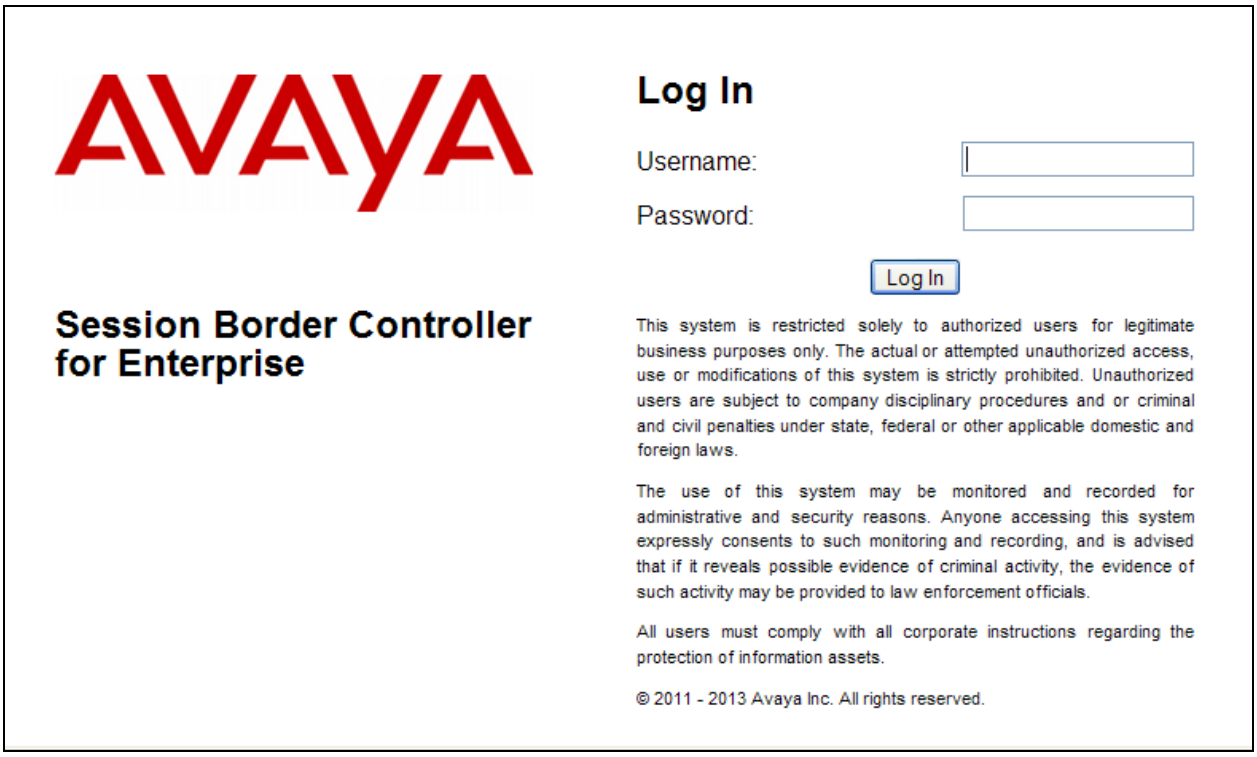
6. Configure Avaya Session Border Controller for Enterprise

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 (e.g., 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.

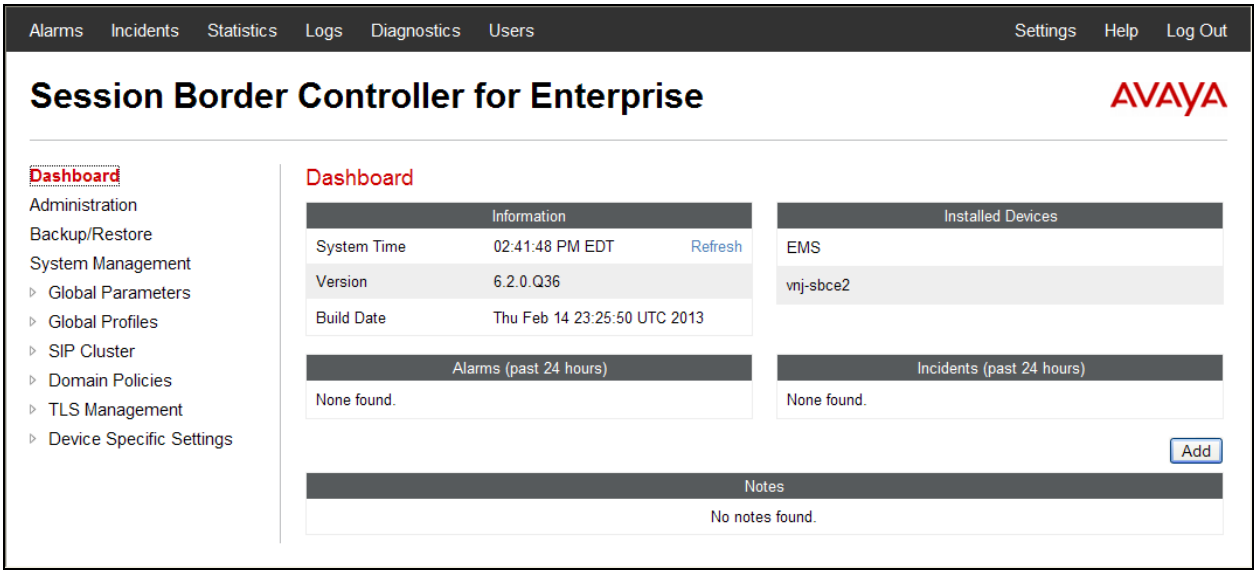
On all screens described in this section, it is to be assumed that parameters are left at their default values unless specified otherwise.

6.1. Access the 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 appropriate credentials.

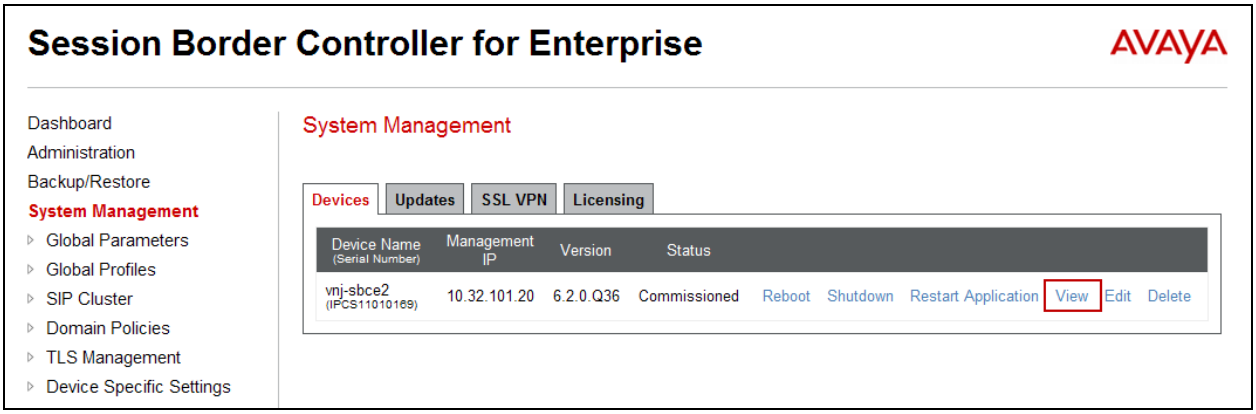


After logging in, the Dashboard screen will appear as shown below. All configuration screens of the Avaya SBCE are accessed by navigating the menu tree in the left pane.

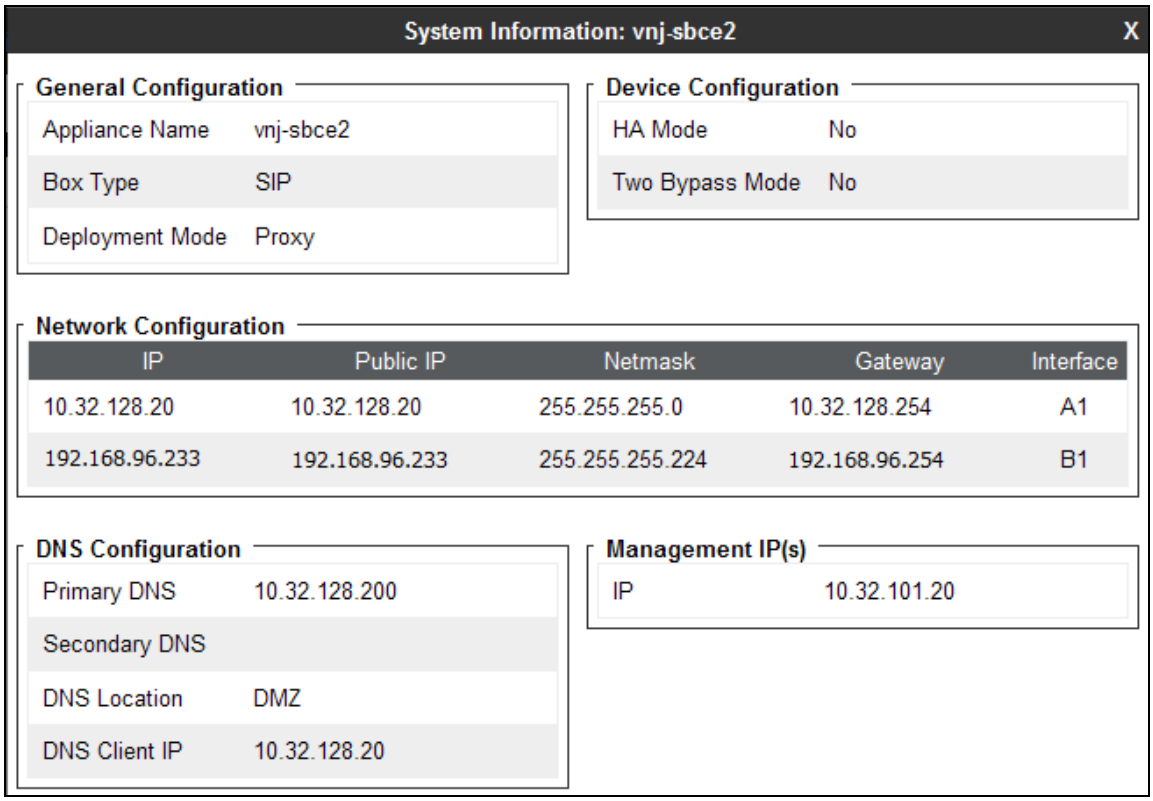


6.2. Verify Network Configuration and Enable Interfaces

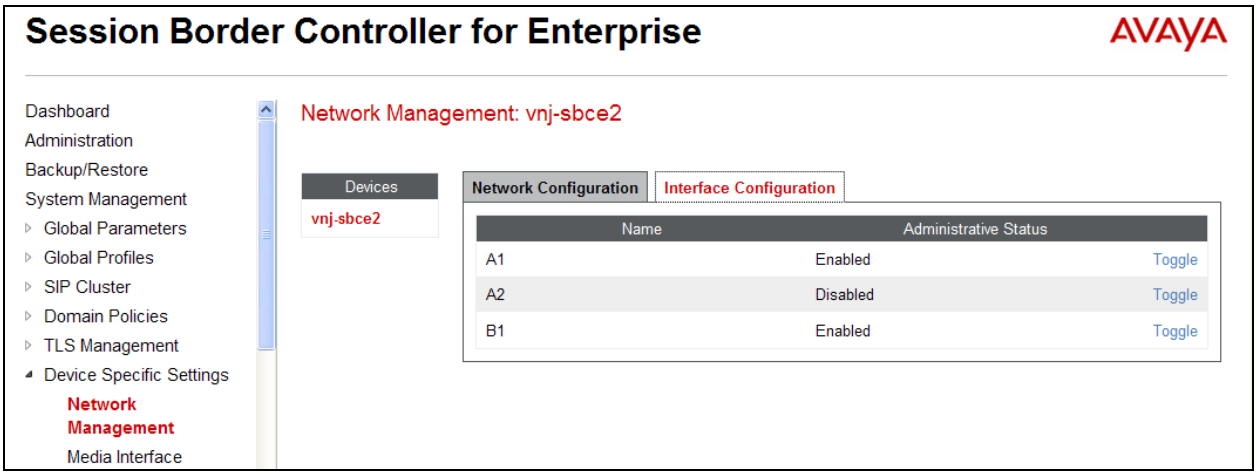
To view the network information provided during installation, navigate to **System Management**. In the right pane, click **View** highlighted below.



A System Information page will appear showing the information provided during installation. In 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 and public interfaces of the Avaya SBCE. Each of these interfaces must be enabled after installation.



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, click on the **Interface Configuration** tab. Verify the **Administrative Status** is **Enabled** for both the **A1** and **B1** interfaces. If not, click **Toggle** to enable the interface.



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.

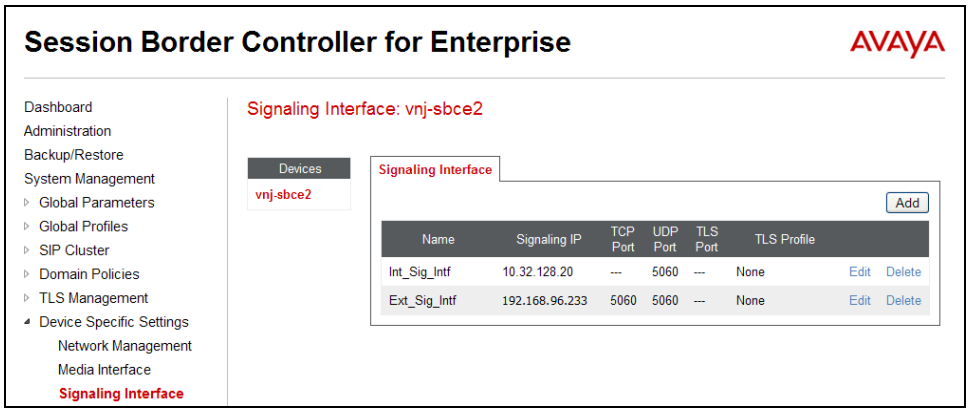
To create a new interface, navigate to **Device Specific Settings → Signaling Interface** in the left pane. In the center pane, select the Avaya SBCE device (**vnj-sbce2**) to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by 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. When configuring the interface, configure the parameters as follows:

- Set **Name** to a descriptive name.
- Set the **Signaling IP** to the IP address associated with the private interface (A1) defined in **Section 6.2**.
- In the **UDP Port**, **TCP Port** and **TLS Port** fields, enter the port the Avaya SBCE will listen on for SIP requests from Avaya IP Office for each transport protocol. For the compliance test, the **UDP Port** was set to **5060**.

Signaling interface **Ext_Sig_Intf** was created for the Avaya SBCE external interface. When configuring the interface, configure the parameters as follows:

- Set **Name** to a descriptive name.
- Set the **Signaling IP** to the IP address associated with the public interface (B1) defined in **Section 6.2**.
- In the **UDP Port**, **TCP Port** and **TLS Port** fields, enter the port the Avaya SBCE will listen on for SIP requests from the service provider for each transport protocol. For the compliance test, both UDP and TCP were tested with Voice Carrier. Thus, the **UDP Port** and **TCP Port** were set to **5060**.



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.

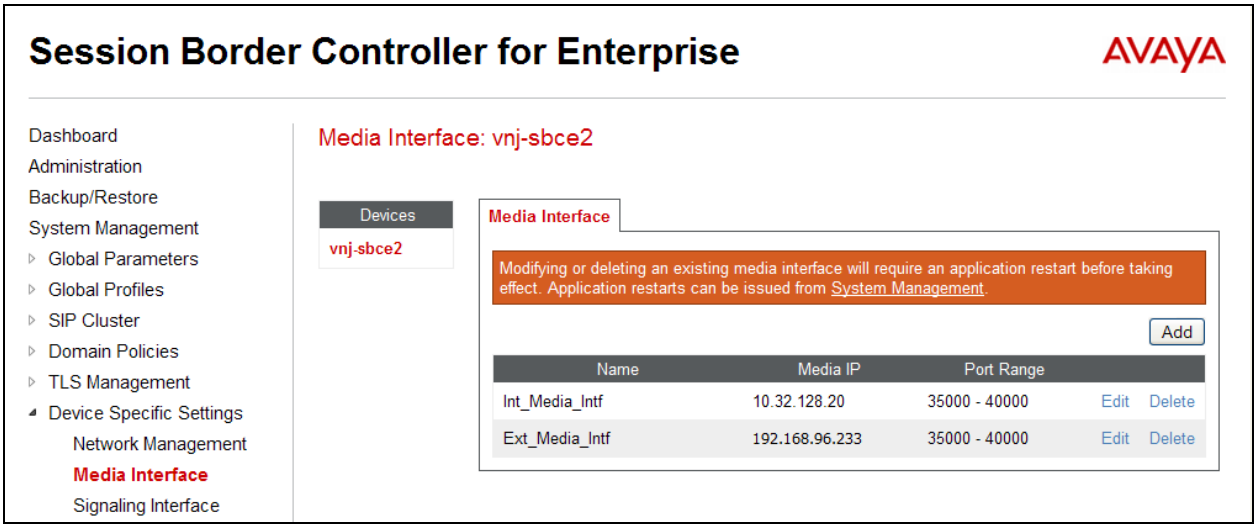
To create a new interface, navigate to **Device Specific Settings → Media Interface** in the left pane. In the center pane, select the Avaya SBCE device (**vnj-sbce2**) to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by 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_Media_Intf** was created for the Avaya SBCE internal interface. When configuring the interface, configure the parameters as follows:

- Set **Name** to a descriptive name.
- Set the **Media IP** to the IP address associated with the private interface (A1) defined in **Section 6.2**.
- Set **Port Range** to a range of ports acceptable to both the Avaya SBCE and Avaya IP Office. For the compliance test, the port range used was selected arbitrarily.

Signaling interface **Ext_Media_Intf** was created for the Avaya SBCE external interface. When configuring the interface, configure the parameters as follows:

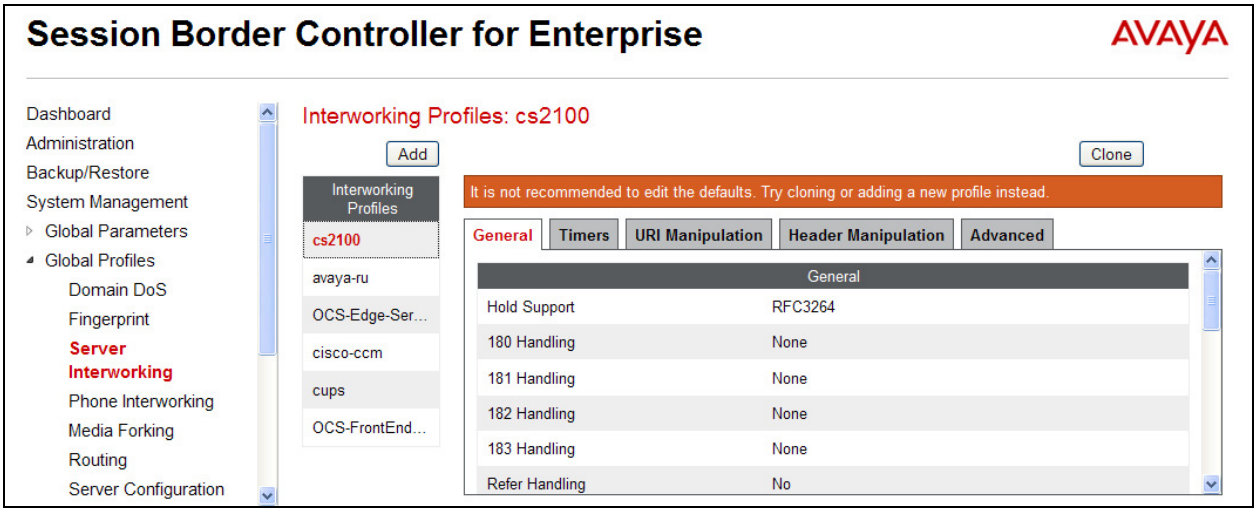
- Set **Name** to a descriptive name.
- Set the **Media IP** to the IP address associated with the public interface (B1) defined in **Section 6.2**.
- Set **Port Range** to a range of ports acceptable to both the Avaya SBCE and the service provider. For the compliance test, the port range used was selected arbitrarily.



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 a server interworking profile for Avaya IP Office and the service provider SIP server. These profiles will be applied to the appropriate server in **Section 6.7.1** and **6.7.2**.

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 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.



6.5.1. Server Interworking – Avaya IP Office

For the compliance test, server interworking profile **IPOffice** was created for Avaya IP Office by creating a new profile and accepting the default values for all settings. The **General** tab parameters are shown below.

General | Timers | URI Manipulation | Header Manipulation | Advanced

General	
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
T.38 Support	No
URI Scheme	SIP
Via Header Format	RFC3261

Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

DTMF	
DTMF Support	None

Edit

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

The **Advanced** tab parameters are shown below.

General | Timers | URI Manipulation | Header Manipulation | Advanced

Record Routes	Both
Topology Hiding: Change Call-ID	Yes
Call-Info NAT	No
Change Max Forwards	Yes
Include End Point IP for Context Lookup	No
OCS Extensions	No
AVAYA Extensions	No
NORTEL Extensions	No
Diversion Manipulation	No
Metaswitch Extensions	No
Reset on Talk Spurt	No
Reset SRTP Context on Session Refresh	No
Has Remote SBC	Yes
Route Response on Via Port	No
Cisco Extensions	No

Edit

6.5.2. Server Interworking – Voice Carrier

For the compliance test, server interworking profile **SP-General** was created for the Voice Carrier SIP server. When creating the profile, the default values were used for all parameters. Thus, the **SP-General** profile is identical to the **IPOffice** profile created in **Section 6.5.1**.

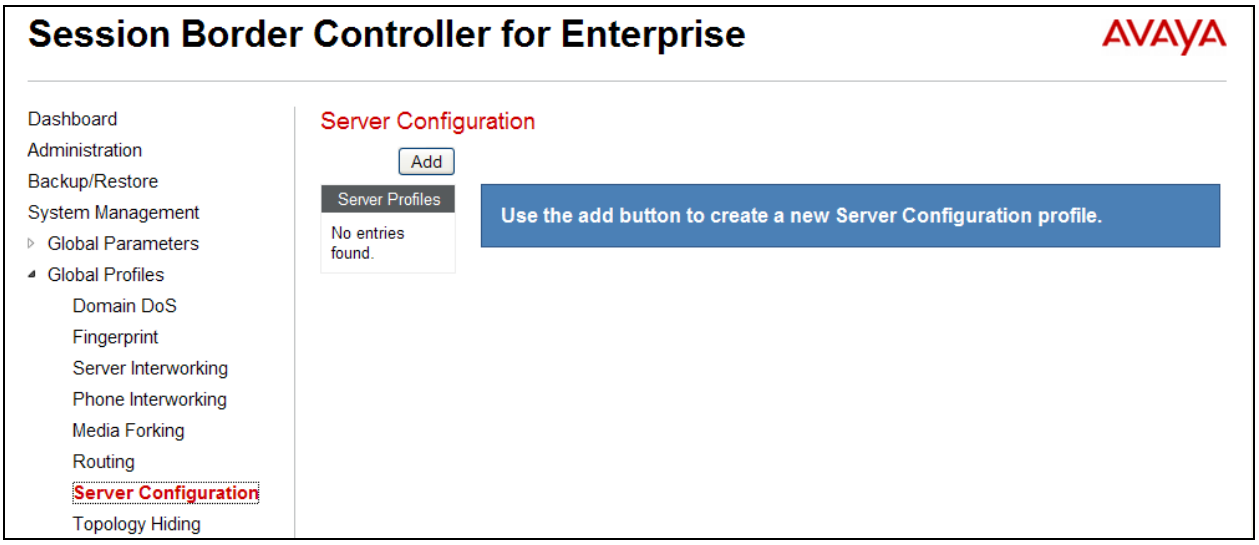
6.6. Signaling Manipulation

Signaling manipulation scripts provides for the manipulation of SIP messages which cannot be done by other configuration within the Avaya SBCE. It was not necessary to create any signaling manipulation scripts for interoperability with Voice Carrier.

6.7. Server Configuration

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

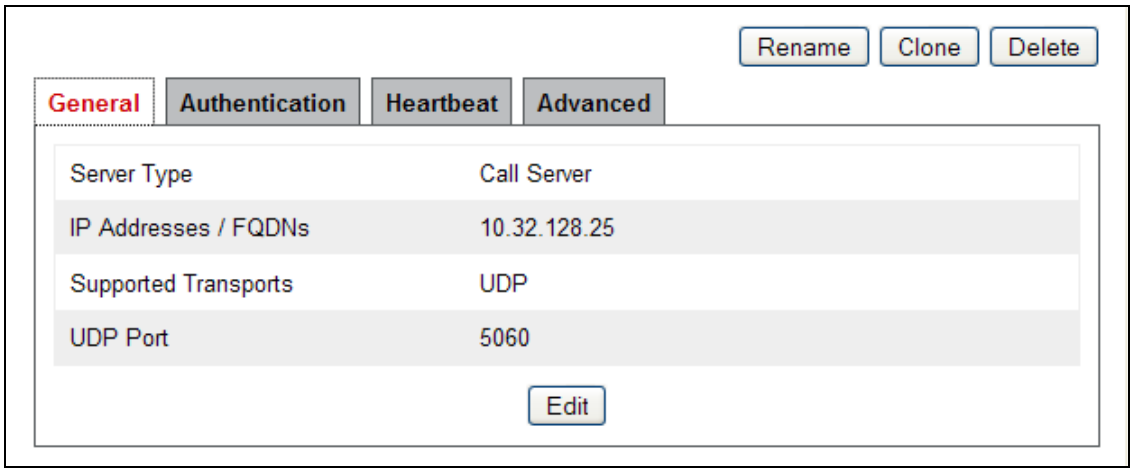
To create a new profile, navigate to **Global Profiles → Server Configuration** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by series of pop-up windows in which the profile parameters can be configured.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



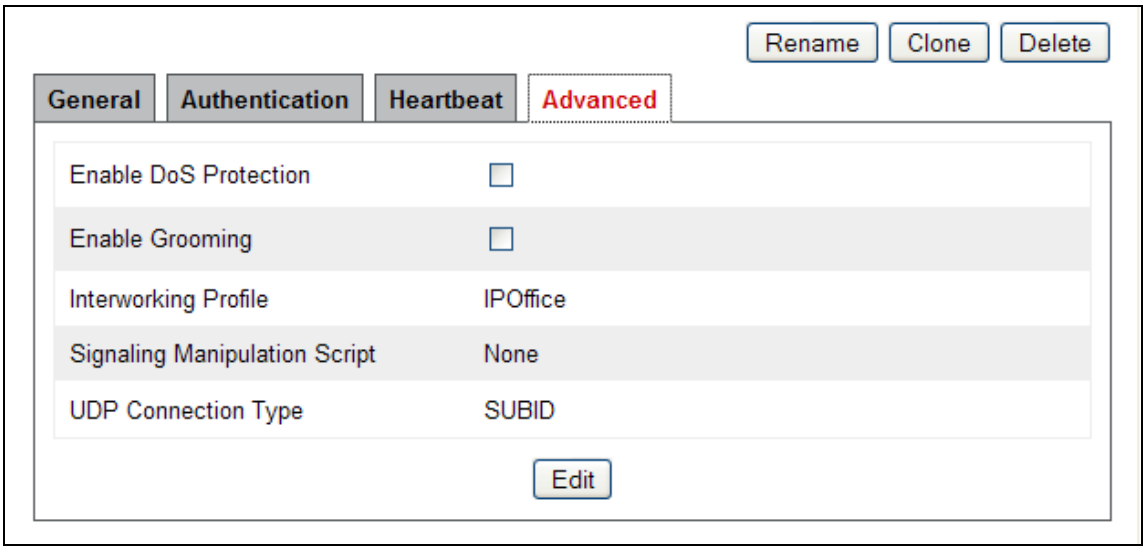
6.7.1. Server Configuration – Avaya IP Office

For the compliance test, server configuration profile **IPO-ACity** 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 signaling interface.
- Set **Supported Transports** to the transport protocol used for SIP signaling between Avaya IP Office and the Avaya SBCE.
- Set the **UDP Port** to the port Avaya IP Office will listen on for SIP requests from the Avaya SBCE.



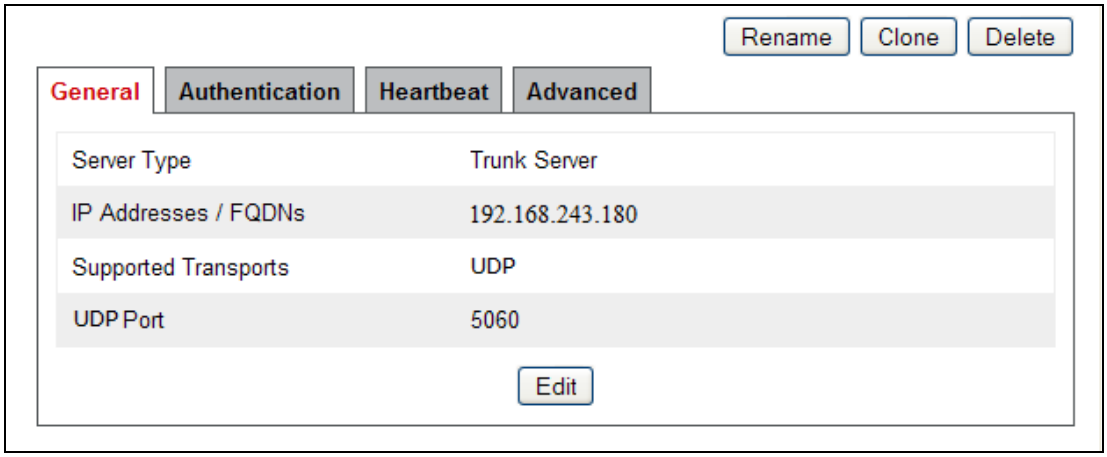
On the **Advanced** tab, set the **Interworking Profile** field to the interworking profile for Avaya IP Office defined in **Section 6.5.1**.



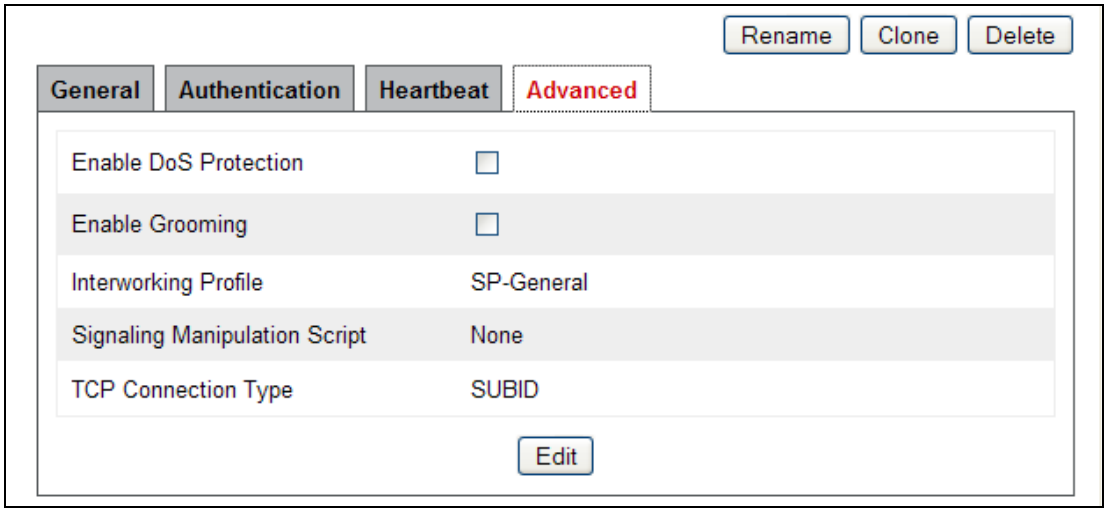
6.7.2. Server Configuration – Voice Carrier

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

- Set **Server Type** to **Trunk Server.**
- Set **IP Addresses / FQDNs** to the IP address of the Voice Carrier SIP server.
- Set **Supported Transports** to the transport protocol used for SIP signaling between Voice Carrier and the Avaya SBCE. In the compliance test, both UDP and TCP were tested.
- Set the **UDP Port** or **TCP Port**, whichever applies, to the standard SIP port of 5060. This is the port Voice Carrier will listen on for SIP requests from the Avaya SBCE. The example below shows the use of UDP.



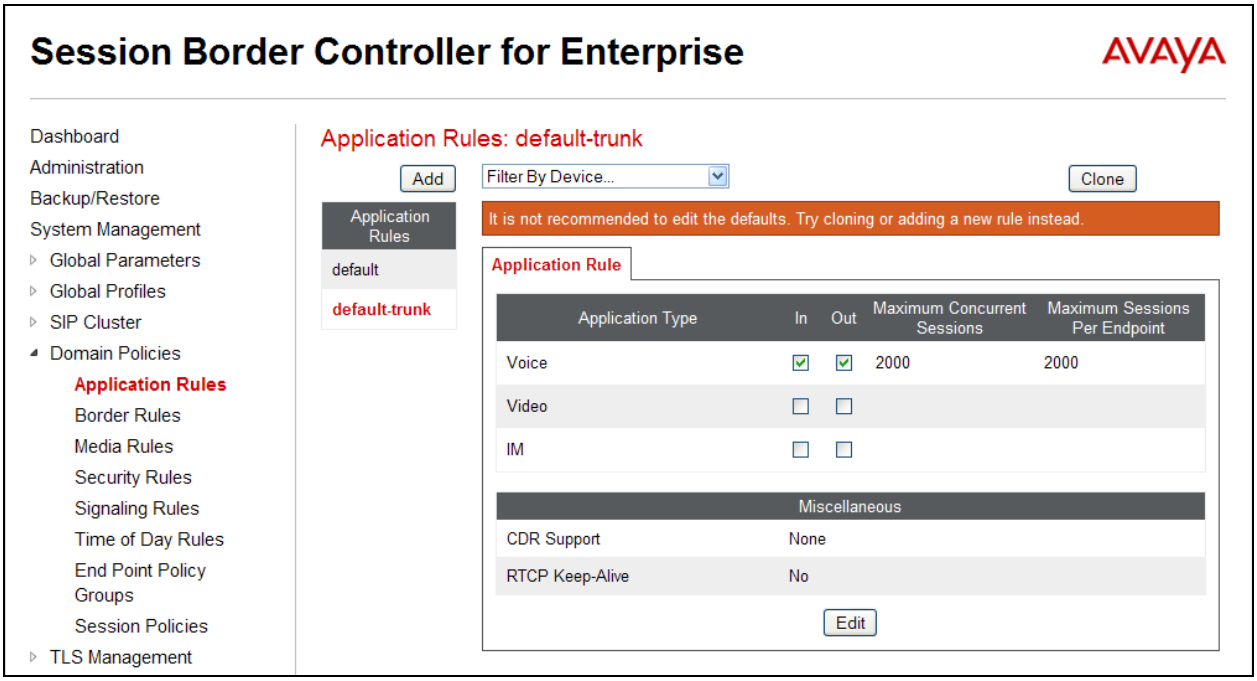
On the **Advanced** tab, set the **Interworking Profile** field to the interworking profile for Voice Carrier defined in **Section 6.5.2**.



6.8. 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 6.11**. For the compliance test, the predefined **default-trunk** application rule (shown below) was used for both Avaya IP Office and the Voice Carrier 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.



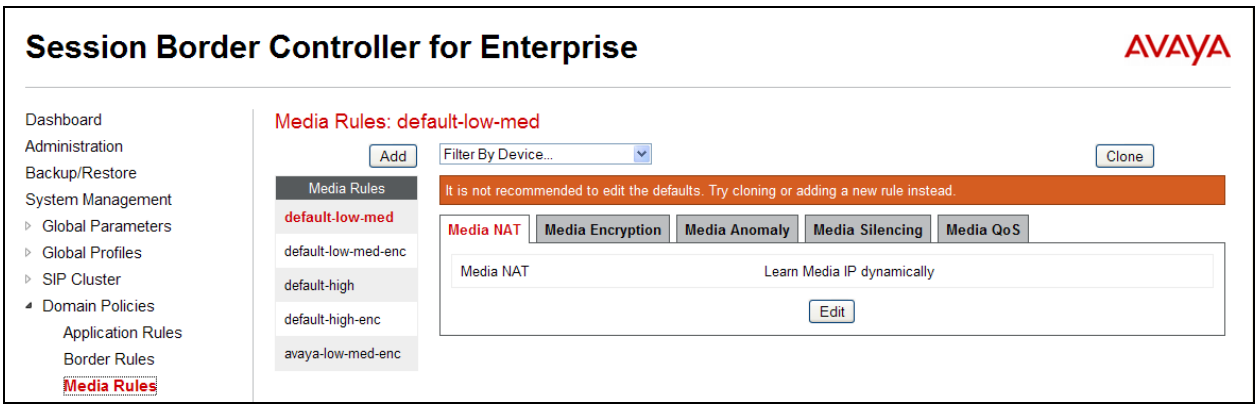
6.9. Media Rules

A media rule defines the processing to be applied to the selected media. A media rule is one component of the larger endpoint policy group defined in **Section 6.11**. For the compliance test, the predefined **default-low-med** media rule (shown below) was used for both Avaya IP Office and the Voice Carrier SIP server.

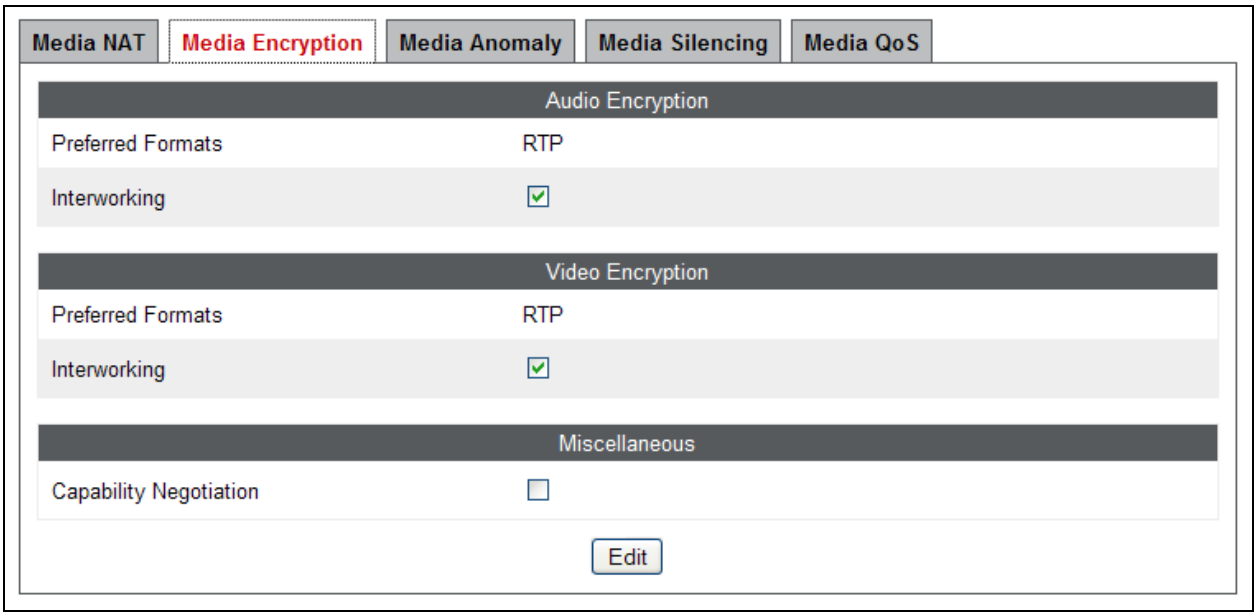
To view an existing rule, navigate to **Domain Policies → Media Rules** in the left pane. In the center pane, select the rule (e.g., **default-low-med**) to be viewed.

Each of the tabs of the **default-low-med** media rule containing data is shown below.

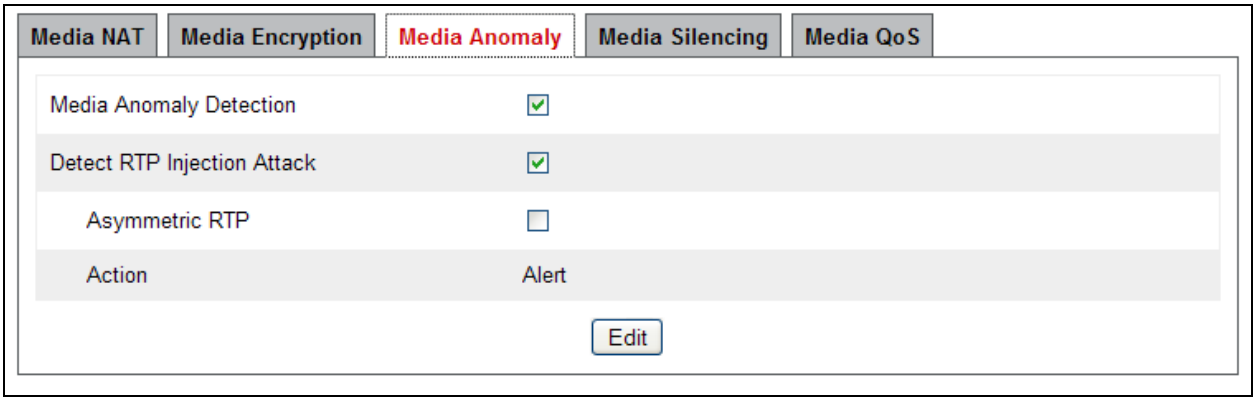
The **Media NAT** tab has no entries.



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

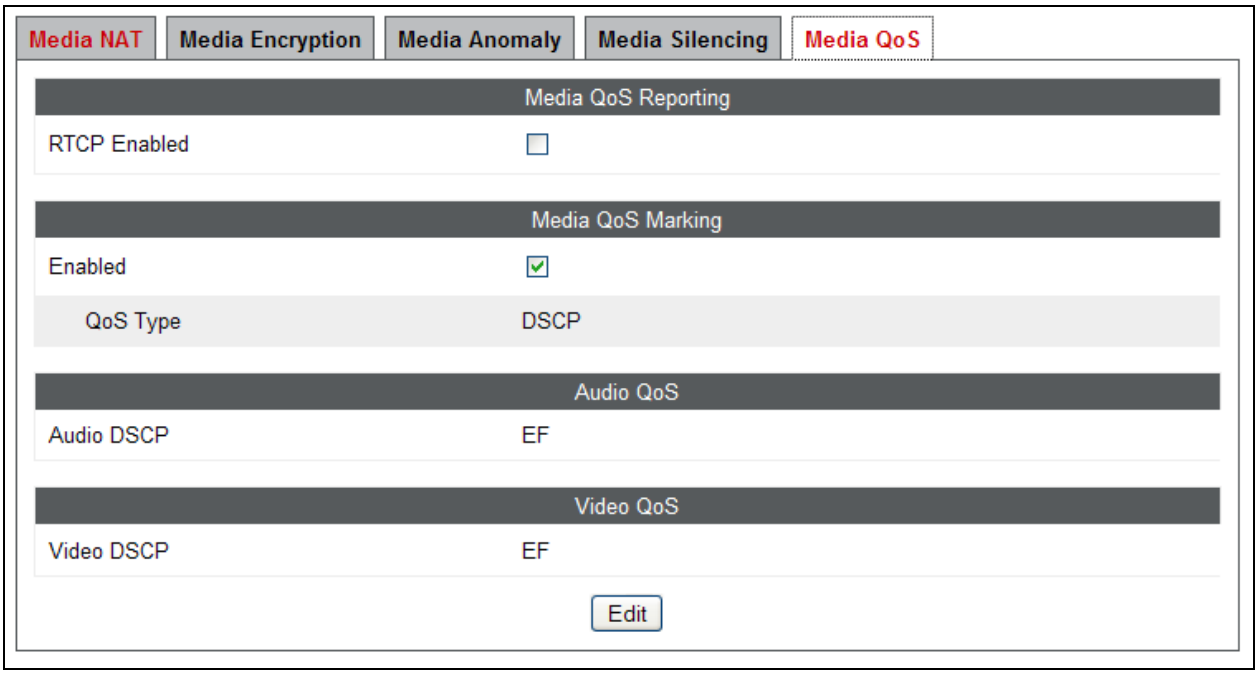


The **Media Anomaly** tab shows **Media Anomaly Detection** was enabled.



The **Media Silencing** tab has no entries.

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

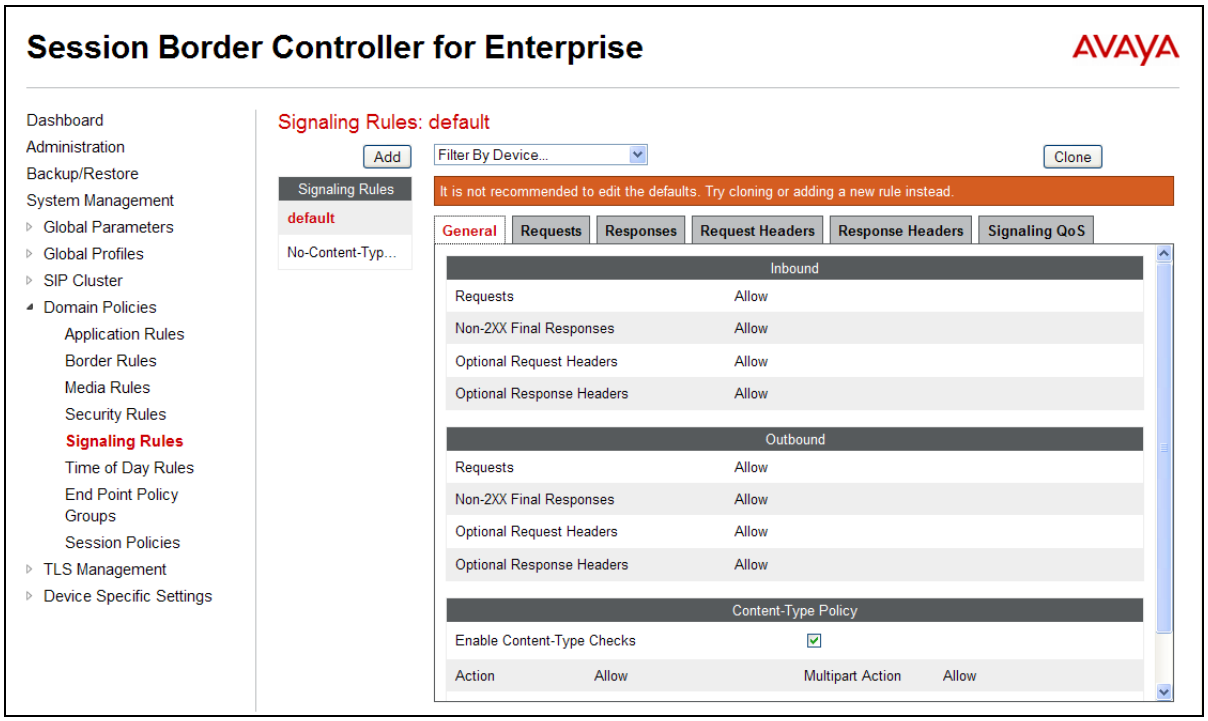


6.10. Signaling Rules

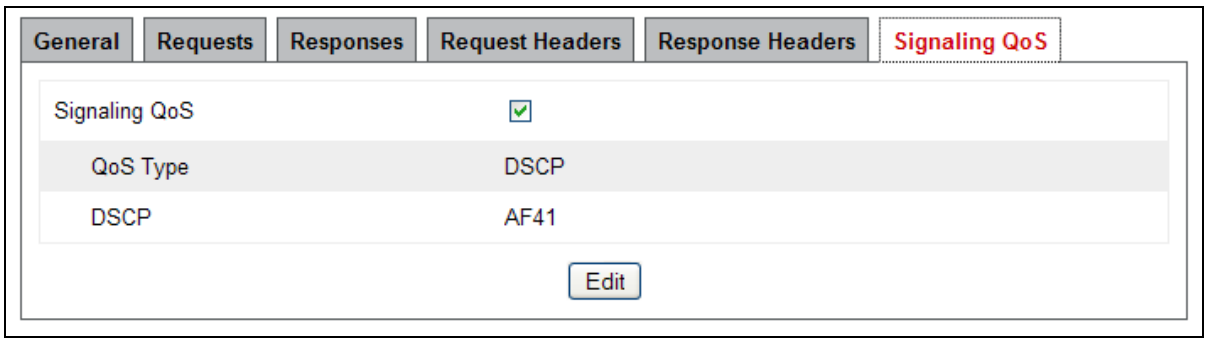
A signaling rule defines the processing to be applied to the selected signaling traffic. A signaling rule is one component of the larger endpoint policy group defined in **Section 6.11**. For the compliance test, the predefined **default** signaling rule (shown below) was used for both Avaya IP Office and the Voice Carrier 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 are shown below.



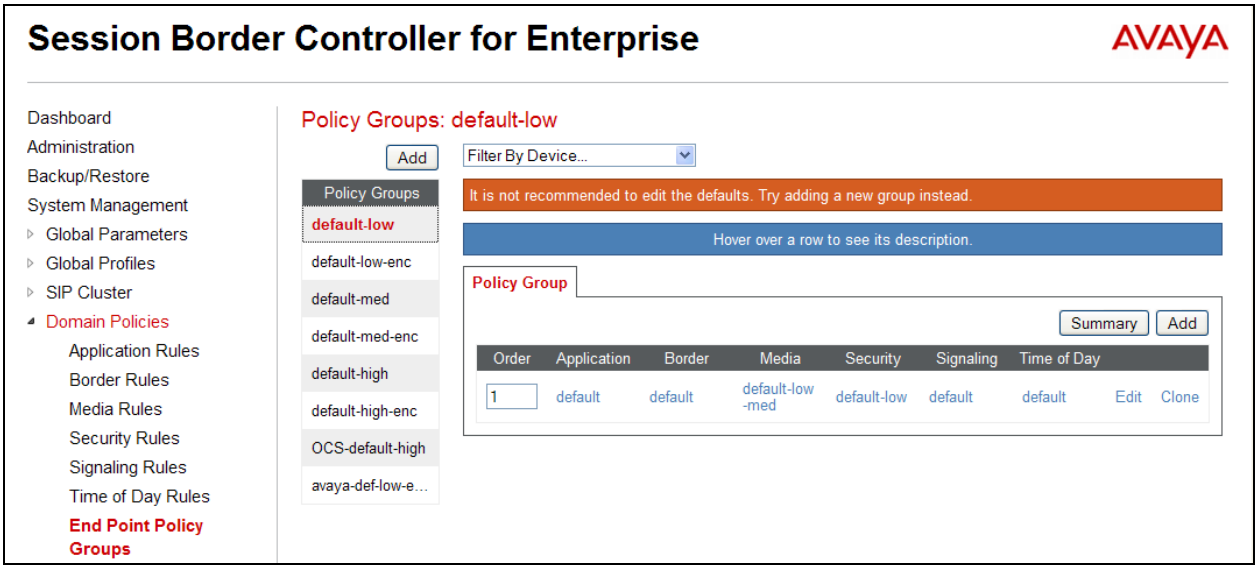
The **Requests**, **Responses**, **Request Headers**, and **Response Headers** tabs have no entries. The **Signaling QoS** tab is shown below.



6.11. Endpoint Policy Groups

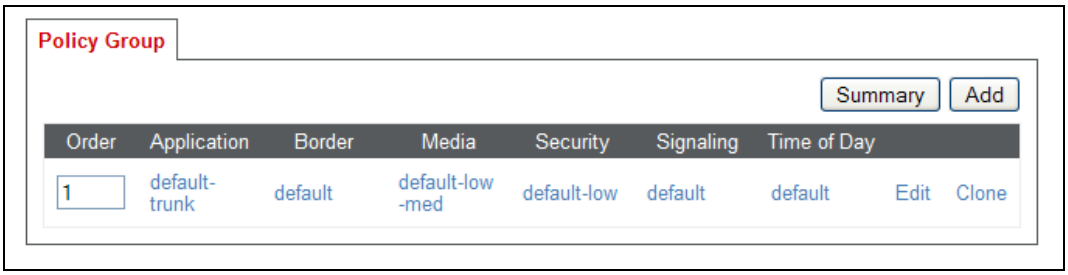
An endpoint policy group is a set of policies that will be applied to traffic between the Avaya SBCE and a signaling endpoint (connected server). Thus, an endpoint policy group must be created for Avaya IP Office and the service provider SIP server. The endpoint policy group is applied to the traffic as part of the endpoint flow defined in **Section 6.14**.

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 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.



6.11.1. Endpoint Policy Group – Avaya IP Office

For the compliance test, endpoint policy group **IPO-EP-Policy** was created for Avaya IP Office. Default values were used for each of the rules which comprise the group with the exception of **Application**. For **Application**, enter the application rule created in **Section 6.8**. The details of the default settings for **Media** and **Signaling** are showed in **Section 6.9** and **Section 6.10** respectively.



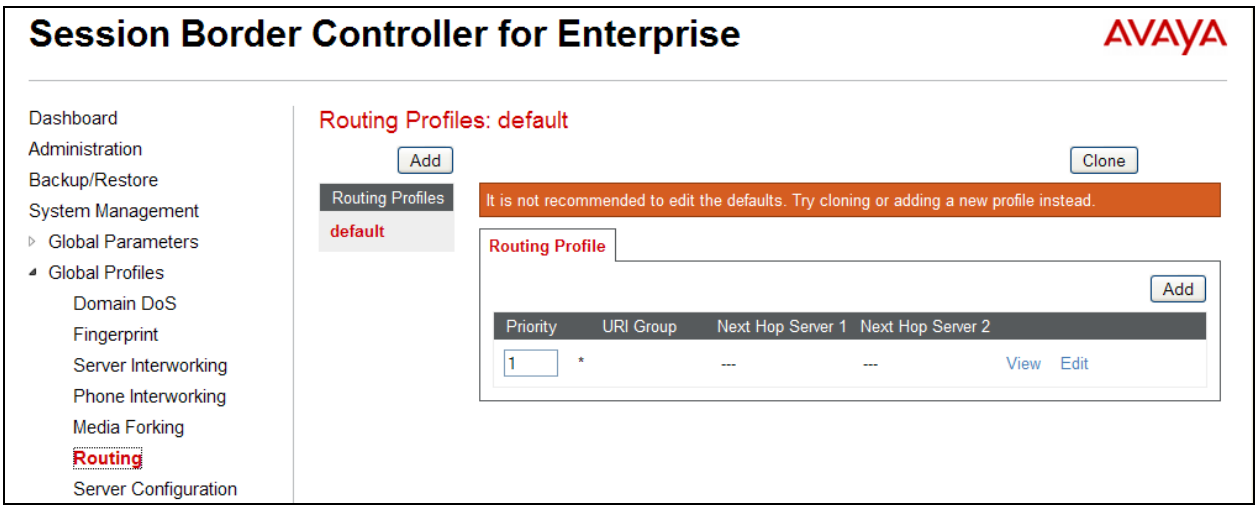
6.11.2. Endpoint Policy Group – Voice Carrier

For the compliance test, endpoint policy group **SP-EP-Policy** was created for the Voice Carrier SIP server. Default values were used for each of the rules which comprise the group with the exception of **Application**. For **Application**, enter the application rule created in **Section 6.8**. Thus, the **SP-EP-Policy** is identical to the **IPO-EP-Policy** created in **Section 6.11.1**.

6.12. 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 6.14**. Create a routing profile for Avaya IP Office and the service provider SIP server.

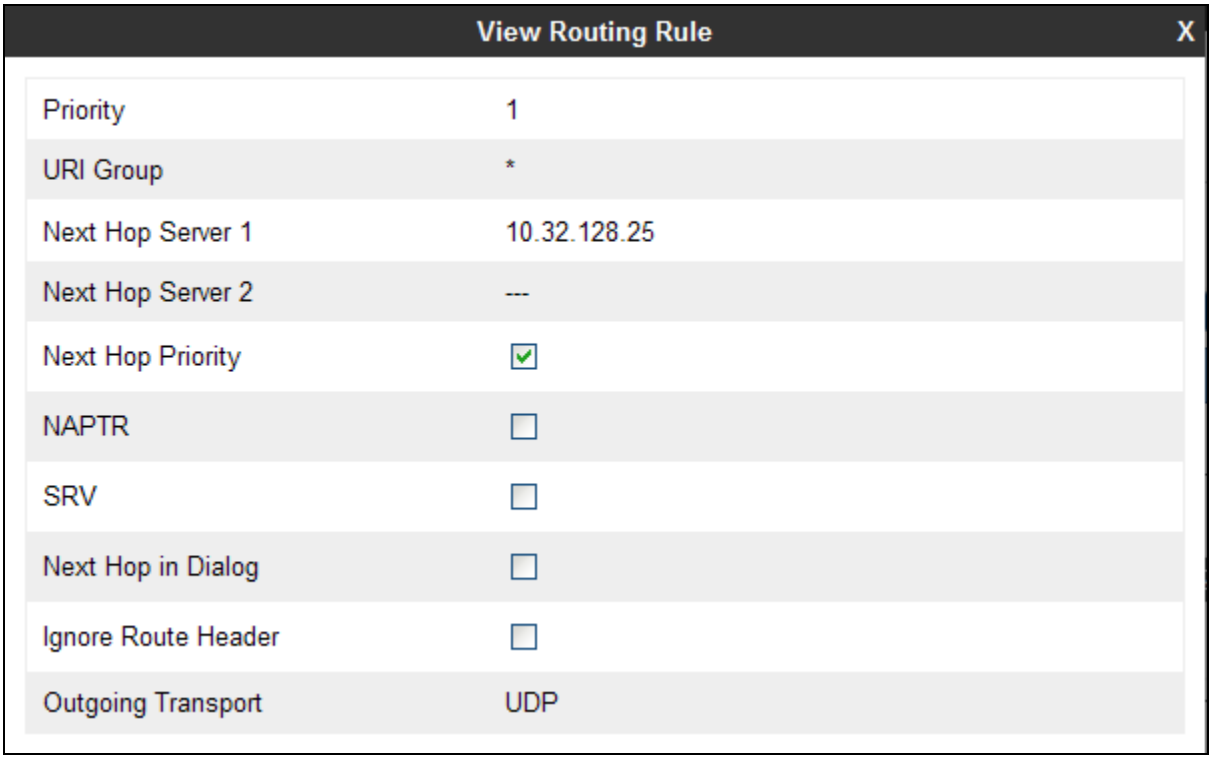
To create a new profile, navigate to **Global Profiles → Routing** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by series of pop-up windows in which the profile parameters can be configured. Once complete,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



6.12.1. Routing – Avaya IP Office

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

- Set the **URI Group** to the wild card * to match on any URI.
- Set the **Next Hop Server 1** field to the IP address of Avaya IP Office signaling interface.
- Enable **Next Hop Priority**.
- Set the **Outgoing Transport** field to **UDP**.



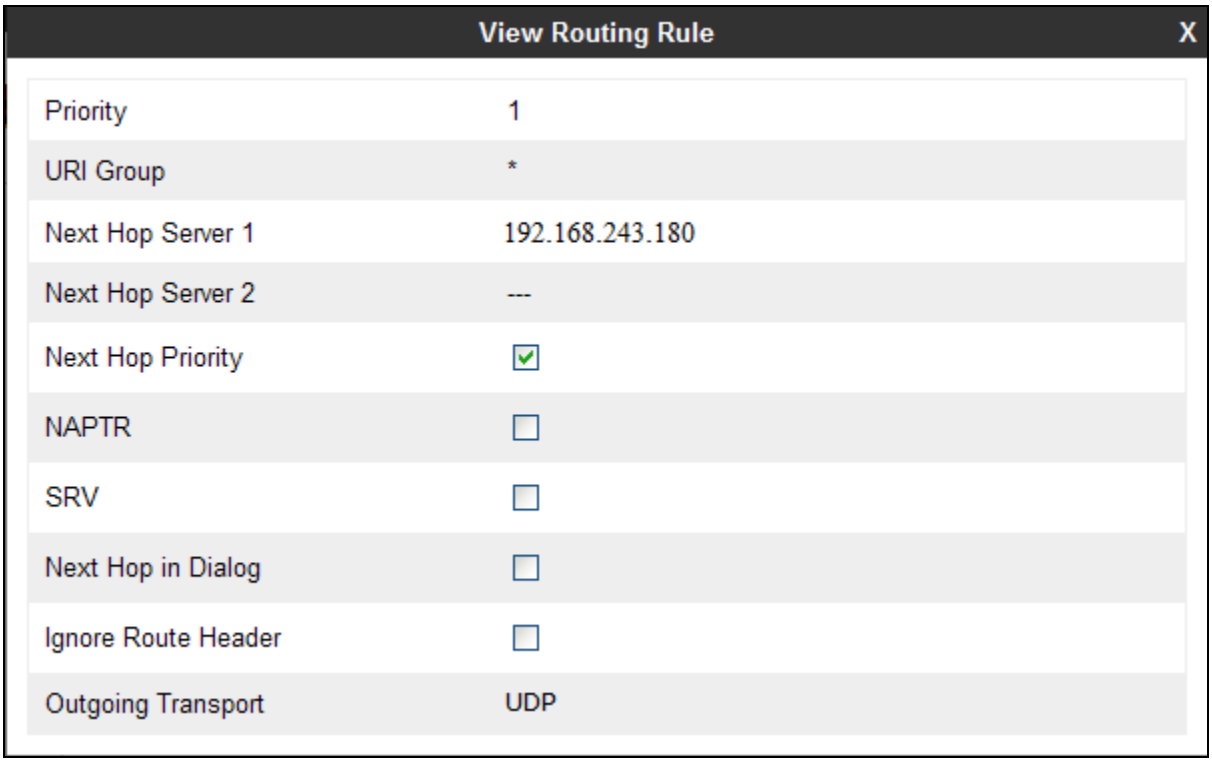
View Routing Rule X

Priority	1
URI Group	*
Next Hop Server 1	10.32.128.25
Next Hop Server 2	---
Next Hop Priority	☑
NAPTR	☐
SRV	☐
Next Hop in Dialog	☐
Ignore Route Header	☐
Outgoing Transport	UDP

6.12.2. Routing – Voice Carrier

For the compliance test, routing profile **To-Trunks** was created for Voice Carrier. When creating the profile, configure the parameters as follows:

- Set the **URI Group** to the wild card * to match on any URI.
- Set the **Next Hop Server 1** field to the IP address of the Voice Carrier SIP server.
- Enable **Next Hop Priority**.
- Set the **Outgoing Transport** field to **UDP** or **TCP** as defined by Voice Carrier.



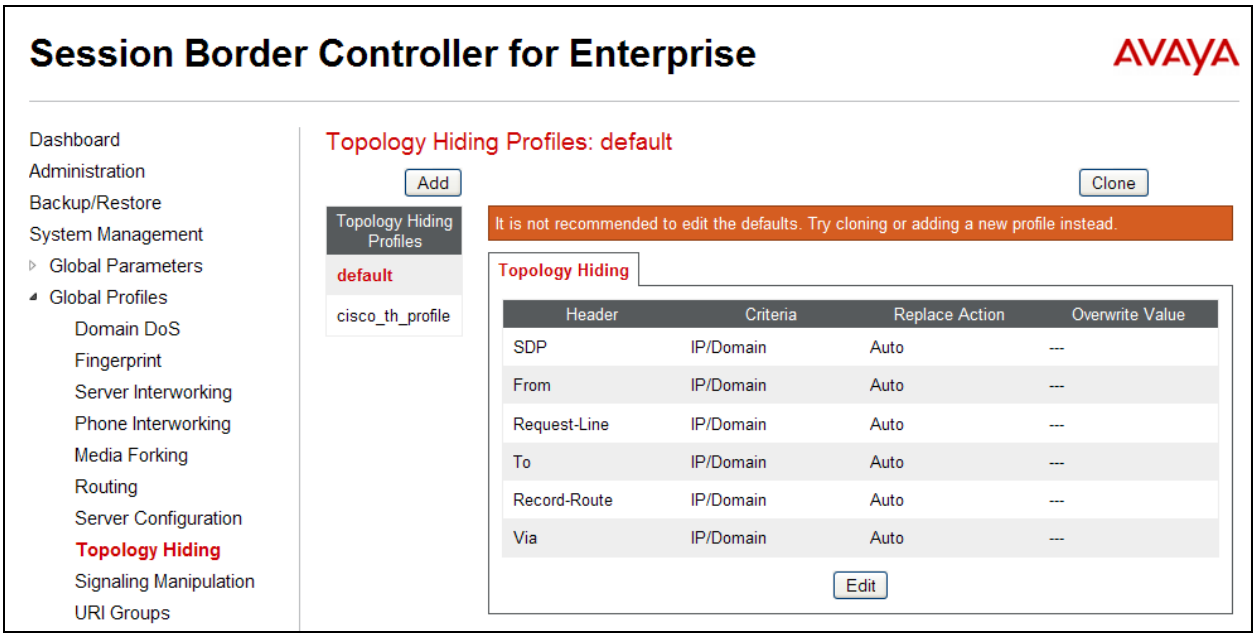
View Routing Rule X

Priority	1
URI Group	*
Next Hop Server 1	192.168.243.180
Next Hop Server 2	---
Next Hop Priority	☑
NAPTR	☐
SRV	☐
Next Hop in Dialog	☐
Ignore Route Header	☐
Outgoing Transport	UDP

6.13. 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 endpoint flow in **Section 6.14**. For the compliance test, the predefined **default** topology hiding profile (shown below) was used for both Avaya IP Office and the Voice Carrier SIP server.

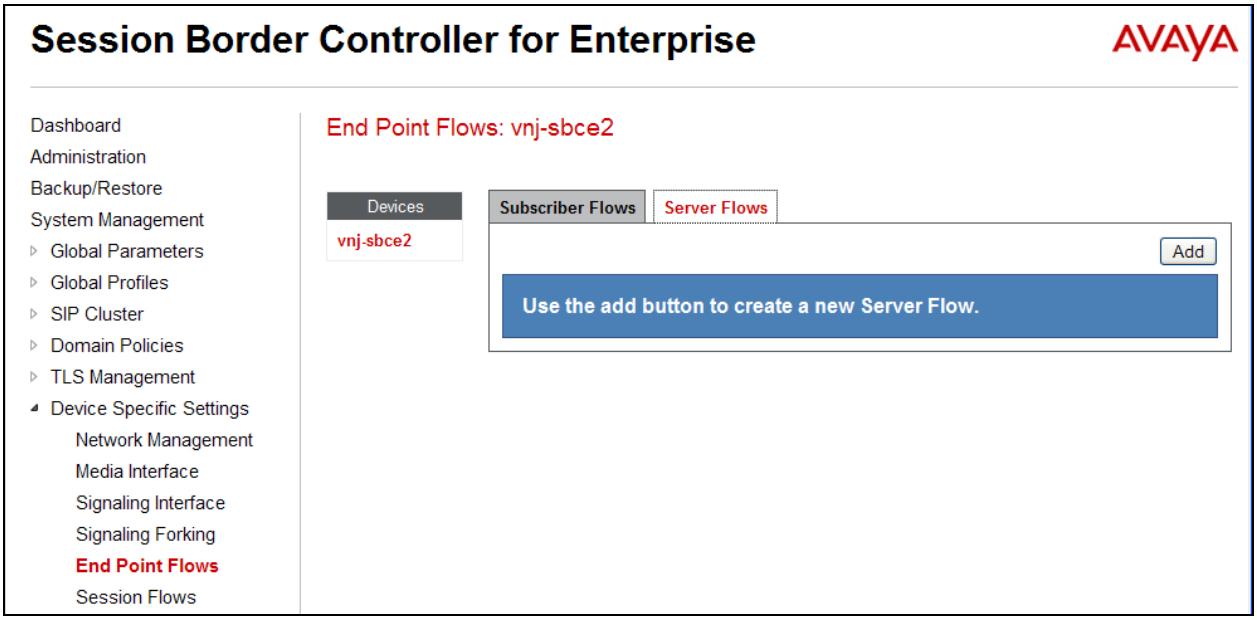
To add a new profile or view an existing profile, navigate to **Global Profiles → Topology Hiding** in the left pane. In the center pane, select **Add** to add a new profile. In the center pane, select an existing profile (e.g., **default**) to be viewed.



6.14. End Point Flows

Endpoint 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 endpoint flow and the destination endpoint flow. In the case of the compliance test, the signaling endpoints are Avaya IP Office and the service provider 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 (**vnj-sbce2**) 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 settings are shown in the far right pane.



6.14.1. End Point Flow – Avaya IP Office

For the compliance test, endpoint flow **IPO-ACity** 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-Trunks** to determine the destination server and corresponding destination flow. The **End Point Policy** and **Topology Hiding Profile** will be applied as appropriate. When creating the flow, configure the parameters as follows:

- For the **Flow Name**, enter a descriptive name.
- For **Server Configuration**, select the Avaya IP Office server created in **Section 6.7.1**.
- To match all traffic, set the **URI Group**, **Transport**, and **Remote Subnet** to *.
- Set the **Received Interface** to the external signaling interface.
- Set the **Signaling Interface** to the internal signaling interface.
- Set the **Media Interface** to the internal media interface.
- Set the **End Point Policy Group** to the endpoint policy group defined for Avaya IP Office in **Section 6.11.1**.
- Set the **Routing Profile** to the routing profile defined in **Section 6.12.2** used to direct traffic to the Voice Carrier SIP server.
- Set the **Topology Hiding Profile** to the topology hiding profile defined for Avaya IP Office in **Section 6.13**.

View Flow: IPO-ACity X

Criteria		Profile	
Flow Name	IPO-ACity	Signaling Interface	Int_Sig_Intf
Server Configuration	IPO-ACity	Media Interface	Int_Media_Intf
URI Group	*	End Point Policy Group	IPO-EP-Policy
Transport	*	Routing Profile	To-Trunks
Remote Subnet	*	Topology Hiding Profile	default
Received Interface	Ext_Sig_Intf	File Transfer Profile	None

6.14.2. End Point Flow – Voice Carrier

For the compliance test, endpoint flow **VoiceCarrier** was created for the Voice Carrier SIP server. All traffic from Voice Carrier will match this flow as the source flow and use the specified **Routing Profile To-IPO-ACity** to determine the destination server and corresponding destination flow. The **End Point Policy** and **Topology Hiding Profile** will be applied as appropriate. When creating the flow, configure the parameters as follows:

- For the **Flow Name**, enter a descriptive name.
- For **Server Configuration**, select the Voice Carrier SIP server created in **Section 6.7.2**.
- To match all traffic, set the **URI Group**, **Transport**, and **Remote Subnet** to *.
- Set the **Received Interface** to the internal signaling interface.
- Set the **Signaling Interface** to the external signaling interface.
- Set the **Media Interface** to the external media interface.
- Set the **End Point Policy Group** to the endpoint policy group defined for Voice Carrier in **Section 6.11.2**.
- Set the **Routing Profile** to the routing profile defined in **Section 6.12.1** used to direct traffic to Avaya IP Office.
- Set the **Topology Hiding Profile** to the topology hiding profile defined for Voice Carrier in **Section 6.13**.

View Flow: VoiceCarrier X

Criteria

Flow Name	VoiceCarrier
Server Configuration	VoiceCarrier
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Int_Sig_Intf

Profile

Signaling Interface	Ext_Sig_Intf
Media Interface	Ext_Media_Intf
End Point Policy Group	SP-EP-Policy
Routing Profile	To-IPO-ACity
Topology Hiding Profile	default
File Transfer Profile	None

7. Voice Carrier IntelliSIP Trunking Configuration

Voice Carrier is responsible for the configuration of the Voice Carrier IntelliSIP Trunking Service. The customer will need to provide the IP address used to reach the enterprise. In the case of the compliance test, this is the Avaya SBCE public address. Voice Carrier will provide the customer the necessary information to configure Avaya IP Office and Avaya SBCE at the enterprise including:

- Voice Carrier SIP domain
- IP address of the Voice Carrier SIP proxy
- Supported codecs
- DID numbers

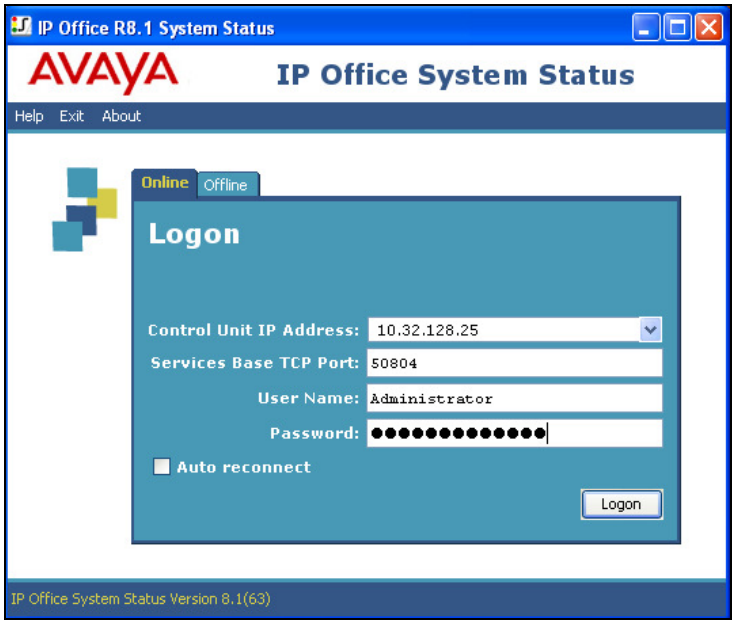
8. Verification Steps

This section provides verification steps that may be performed in the field to verify that the solution is configured properly.

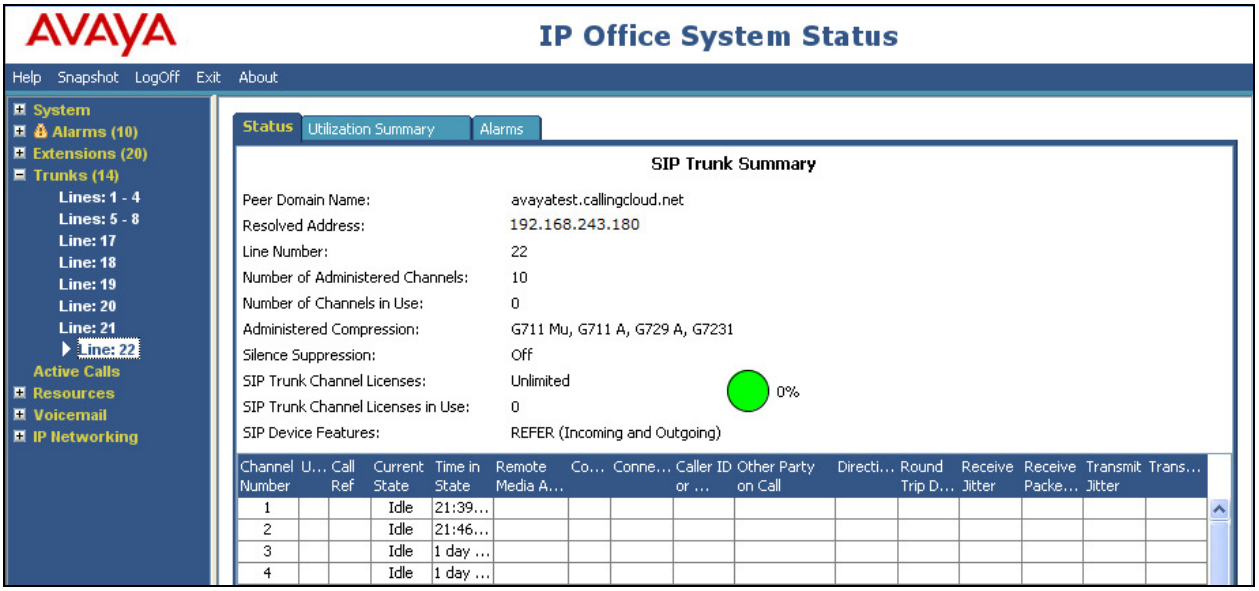
8.1. System Status

The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → Programs → IP Office → System Status**.

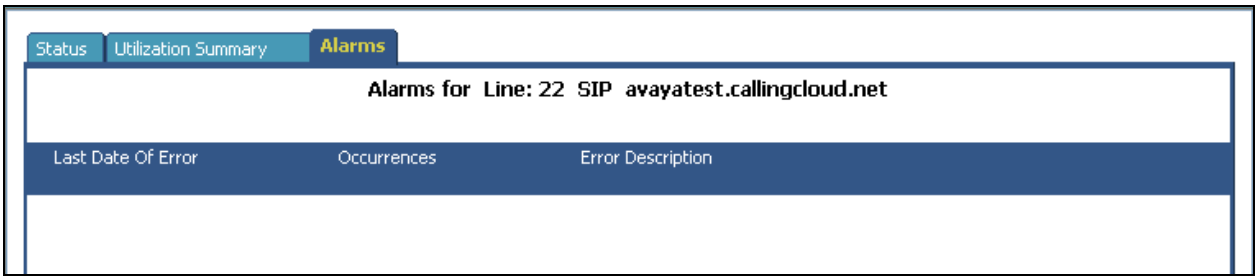
The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current Stat**e for each channel. If no active call is currently in session (as shown below), the state should be **Idle**. If some channels are taken by active calls, then the **Current Sate** will be shown as **Connected**.



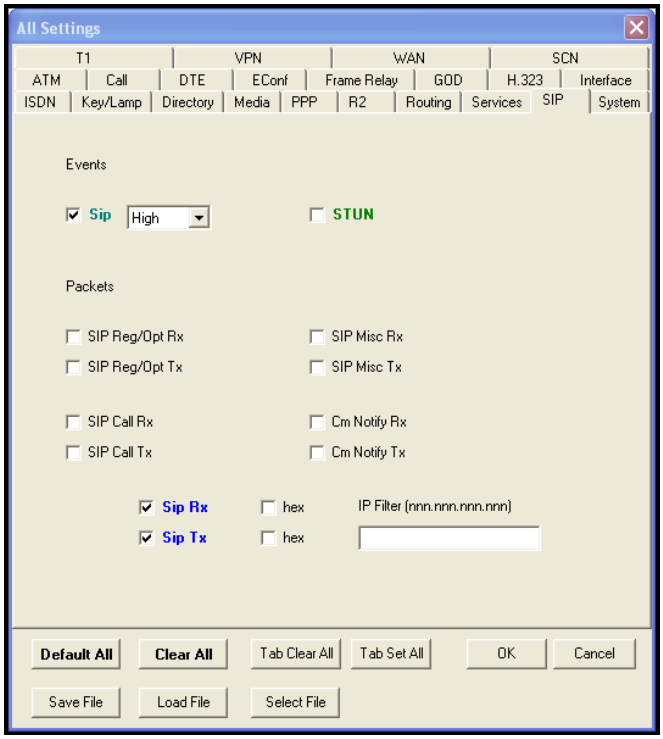
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



8.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select **Filters → Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked.



9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office 8.1 to the Voice Carrier IntelliSIP Trunking Service. The Voice Carrier IntelliSIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks. The Voice Carrier IntelliSIP Trunking Service passed compliance testing. Please refer to **Section 2.2** for any exceptions.

10. Additional References

This section references documentation relevant to these Application Notes. In general, Avaya product documentation is available at http://support.avaya.com.

[1] *IP Office 8.1 IP500/IP500 V2 Installation*, Document Number 15-601042, Issue 27m, July 2, 2013.
[2] *IP Office Release 8.1Manager 10.1*, Document Number 15-601011, Issue 29u, April 5, 2013.
[3] *IP Office System Status Application*, Document Number 15-601758, Issue 07a, November 26, 2012.
[4] *IP Office Release 8.1 Administering Voicemail Pro*, Document Number 15-601063, Issue 8b, December 11, 2012.
[5] *IP Office System Monitor*, Document Number 15-601019, Issue 03c, March 1, 2013.

Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/

11. Appendix A: SIP Line Template

Avaya IP Office supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Note that not all of the configuration information, particularly items relevant to a specific installation environment, is included in the SIP Line Template. Therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported and additional configuration be supplemented using **Section 5.4** in these Application Notes as a reference.

The SIP Line Template created from the configuration as documented in these Application Notes is as follows:

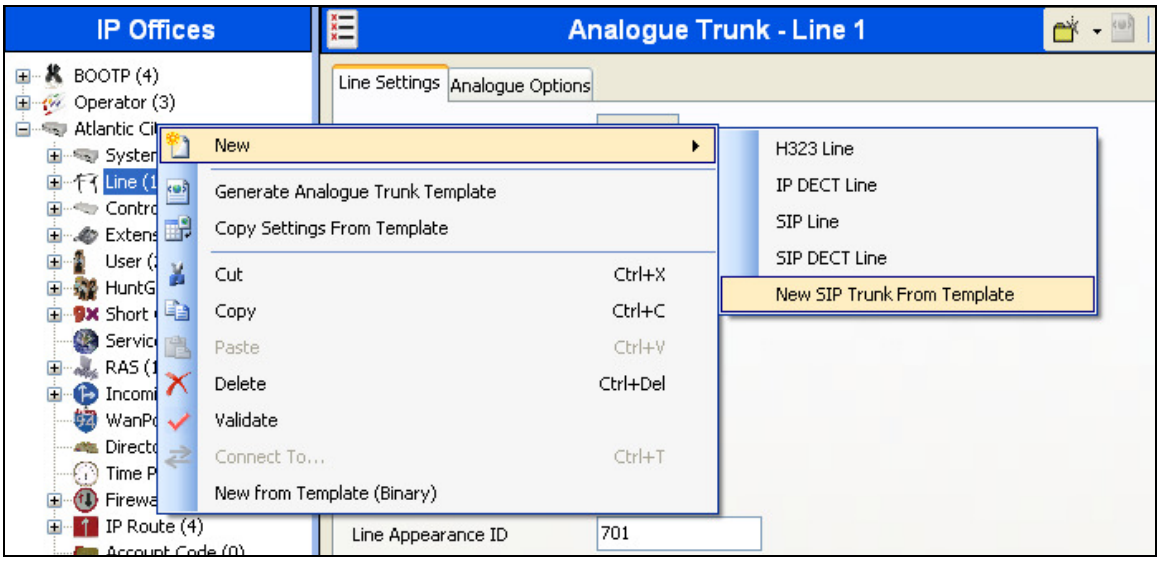

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130816</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Voice Carrier</DescriptiveName>
  <ITSPDomainName>avayatest.callingcloud.net</ITSPDomainName>
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>2</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateNever</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>10.32.128.20</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>false</UseAdvVoiceCodecPrefs>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_G711</FaxTransportSupport>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
```

```
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>false</Rel100Supported>
  <T38FaxVersion>3</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
  <HighSpeed>0</HighSpeed>
  <TCFMethod>Trans_TCF</TCFMethod>
  <MaxBitRate>FaxRate_14400</MaxBitRate>
  <EflagStartTimer>2600</EflagStartTimer>
  <EflagStopTimer>2300</EflagStopTimer>
  <UseDefaultValues>true</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
  <TFOPEnhancement>true</TFOPEnhancement>
  <DisableT30ECM>false</DisableT30ECM>
  <DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
  <DisableT30MRCompression>false</DisableT30MRCompression>
  <NSFOverride>false</NSFOverride>
</Template>
```


To import the above template into a new installation:

1. On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US_VoiceCarrier_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates). It may be necessary to create this directory.

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



3. Verify that **United States** is automatically populated for **Country** and **VoiceCarrier** is automatically populated for **Service Provider** in the resulting Template Type Selection screen as shown below. Click **Create new SIP Trunk** to finish the importing process.

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.