Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 9.1 and Avaya Session Border Controller for Enterprise 6.3 to support Group of Gold Line SIP Trunking – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking on an enterprise solution consisting of Avaya IP Office 9.1 and Avaya Session Border Controller for Enterprise 6.3, to interoperate with Group of Gold Line SIP Trunking.

The SIP Trunking service offered by Group of Gold Line provides customers with PSTN access via a SIP trunk between the enterprise and the service provider's network, as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

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

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

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Group of Gold Line SIP Trunking service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya IP Office Release 9.1, Avaya Session Border Controller for Enterprise (Avaya SBCE) Release 6.3 and various Avaya endpoints.

The Group of Gold Line SIP Trunking service referenced within these Application Notes is designed for business customers. Customers using this service with this Avaya enterprise solution are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

2. General Test Approach and Test Results

A simulated enterprise site containing all the Avaya equipment for the SIP-enabled solution was installed at the Avaya Solution and Interoperability Lab. The enterprise site was configured to connect to the Group of Gold Line SIP Trunking service via a broadband connection.

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.

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.

2.1. Interoperability Compliance Testing

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

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various phone types. Phone types included SIP, H.323, digital and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included SIP, H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya Communicator for Windows softphones.
- Inbound and outbound PSTN calls to/from SIP remote workers using Avaya Communicator for Windows softphones.
- Various call types including: local, long distance national, long distance international, outbound toll free and local directory assistant.
- Codecs G.729A, G.711MU and G.711A.
- Fax support.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call transfer, call forwarding and twinning.

The following functionality was not supported or it was not tested in the test configuration:

- Network Call Redirection using the REFER method is not currently supported by Group of Gold Line.
- Operator (0) and operator assisted calls (0+10) are not supported.
- Inbound toll-free and emergency (911) calls are supported but were not tested as part of the compliance test

2.2. Test Results

Interoperability testing of the Group of Gold Line SIP Trunking service was completed with successful results for all test cases with the observations and limitations described below:

- **Call Transfer to the PSTN** – Network Call Redirection using the REFER method is not currently supported by Group of Gold Line. REFER needs to be disabled on the SIP Line tab of the IP Office configuration. Inbound/outbound calls that are transferred back to the PSTN are allowed to complete, but IP Office is not released after the call is transferred, and two trunks remain busy for the complete duration of the call.
- **Fax Support** – Inbound T.38 fax calls to the enterprise failed during testing. There seems to be an interoperability issue related to the timing of the T.38 re-invites sent from each end on incoming calls. The V.29 negotiation during the call setup fails to complete and the calls disconnect. Hence, T.38 fax should not be used in this solution.
  Fax was successfully tested using G.711 pass through mode.
- **SIP Header Manipulation** – During the compliance test, a Sigma Script was used in the Avaya SBCE to remove the "Remote-Address" header used by the Avaya SBCE from outbound messages to the service provider. This header has local significance only and should not be propagated on the SIP trunk to the service provider.

2.3. Support

For technical support and information on the Group of Gold Line solutions, please visit http://www.groupofgoldline.com

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Group of Gold Line SIP Trunking service through a public Internet WAN connection.

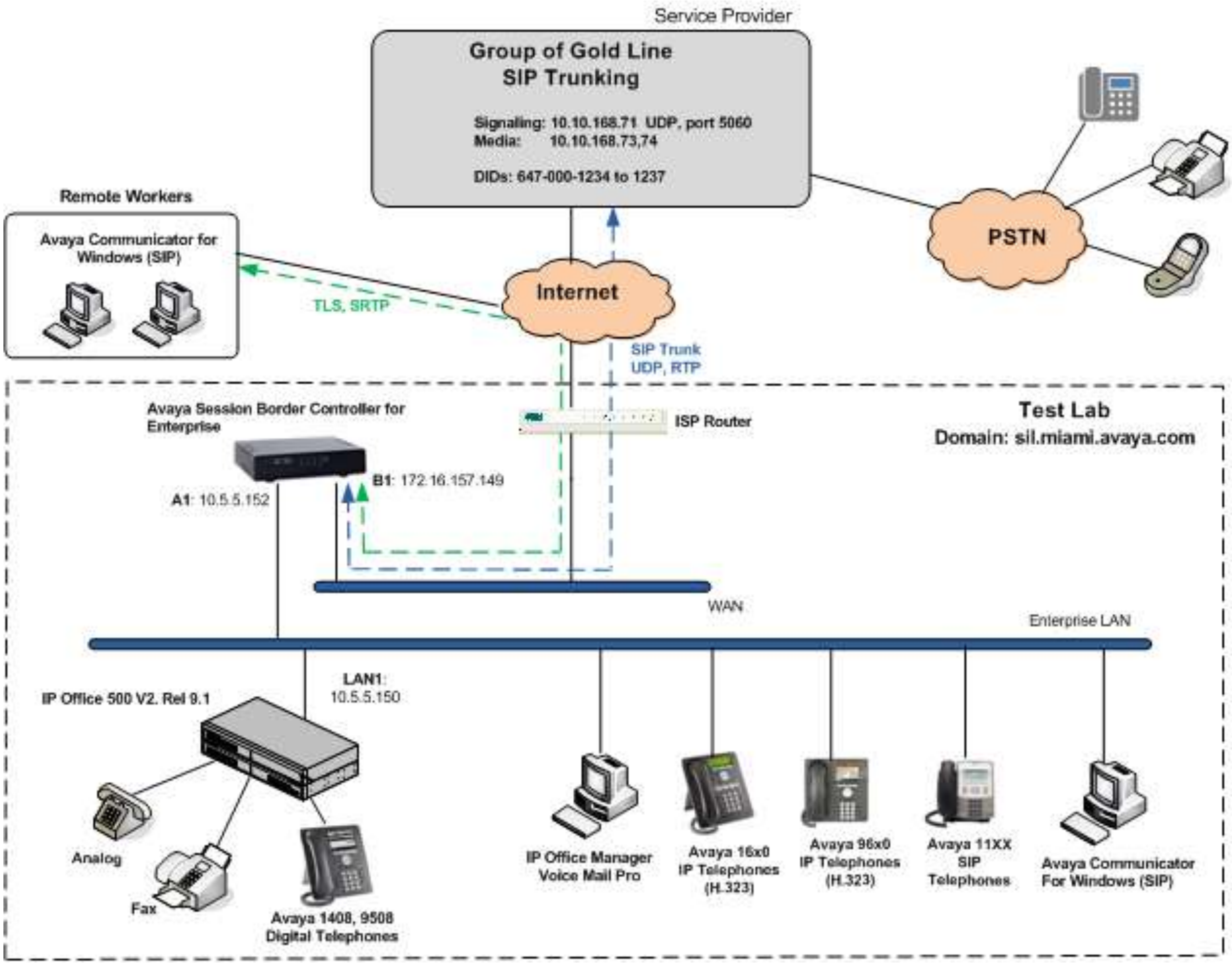


Figure 1: Test Configuration

Note that for security purposes, all public IP addresses of the network elements and public PSTN numbers shown throughout these Application Notes have been edited so the actual values are not revealed.

The enterprise site contains the Avaya IP Office 500v2 with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. The LAN1 port of Avaya IP Office is connected to the enterprise LAN. Endpoints include Avaya 1600 and 9600 Series IP Telephones (with H.323 firmware), Avaya 1140E IP Telephones (with SIP firmware), Avaya 1408 and 9508D Digital Telephones, analog telephones and PCs running Avaya Communicator for Windows.

The site also has a Windows PC running Avaya IP Office Manager to configure and administer the Avaya IP Office system, and Avaya Voicemail Pro providing voice messaging service to the Avaya IP Office users. Mobile Twinning is configured for some of the Avaya IP Office users so that calls to these users' extensions will also ring and can be answered at the configured mobile telephones.

Located at the edge of the enterprise, the Avaya SBCE has two physical interfaces. Interface B1 was used to connect to the public network, while interface A1 was used to connect to the private enterprise infrastructure. All signaling and media traffic entering or leaving the enterprise flows through the Avaya SBCE, in this way protecting the enterprise against any SIP-based attacks. The Avaya SBCE also performs network address translation at both the IP and SIP layers.

Additionally, the reference configuration included the support for IP Office soft-clients in a remote worker environment. A remote worker is a SIP endpoint that resides in the untrusted network, registered to the IP Office at the enterprise via the Avaya SBCE. Remote workers feature the same functionality as any other endpoint at the enterprise. The Avaya Communicator for Windows soft-client was used for this purpose. For security over the public network, remote workers used Transport Layer Security (TLS) as the signaling protocol and Secure Real Time Protocol (SRTP) for the media.

The configuration tasks required to support remote workers are beyond the scope of these Application Notes; hence they are not discussed in this document. Consult section *Configuring the Avaya Session Border Controller for IP Office Remote Workers* in [**2**] in the **Additional References**, for more information on this topic.

In an actual customer configuration, the enterprise site may include additional network components between the service provider and the Avaya IP Office system, such as routers or data firewalls. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that all SIP and RTP traffic between the service provider and the Avaya IP Office system must be allowed to pass through these devices.

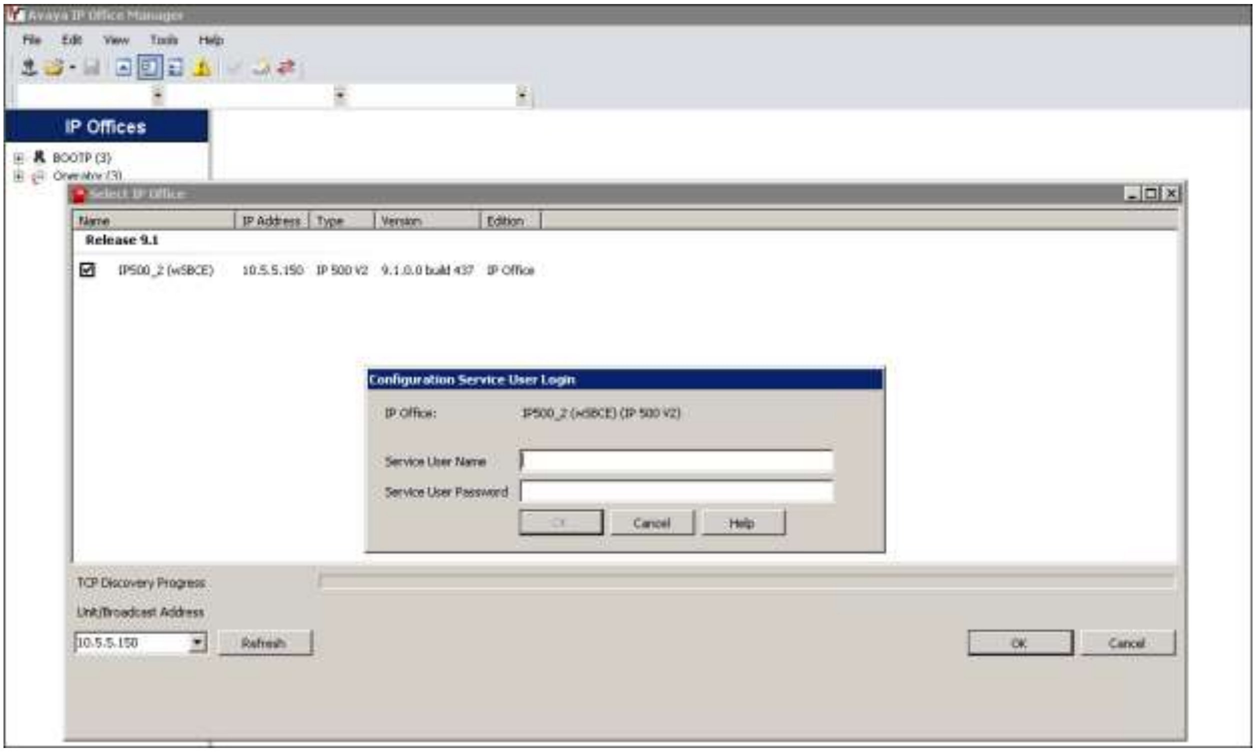
4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya IP Office 500v2	9.1.0.437
Avaya IP Office Digital Expansion Module DCPx16	9.1.0.437
Avaya IP Office Manager	9.1.0.0.Build 437
Avaya IP Office Voicemail Pro	9.1.0.166
Avaya Session Border Controller for Enterprise	6.3.000-19-4338
Avaya 1608 IP Telephone (H.323)	1.3.5
Avaya 9640 IP Telephone (H.323)	Avaya one-X Deskphone Edition S3.230A
Avaya 1140E IP Telephone (SIP)	04.04.18.00
Avaya Digital Telephone 1408	40.0
Avaya Digital Phone 9508	0.55
Avaya Communicator for Windows	2.0.3.30
Group of Gold Line	
Sonus GSX9000HD (Network Border Switch)	V09.00.04 R000

5. Configure IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the Group of Gold Line SIP Trunking service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running IP Office Manager, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials.



A management window will appear similar to the one shown in the next section.

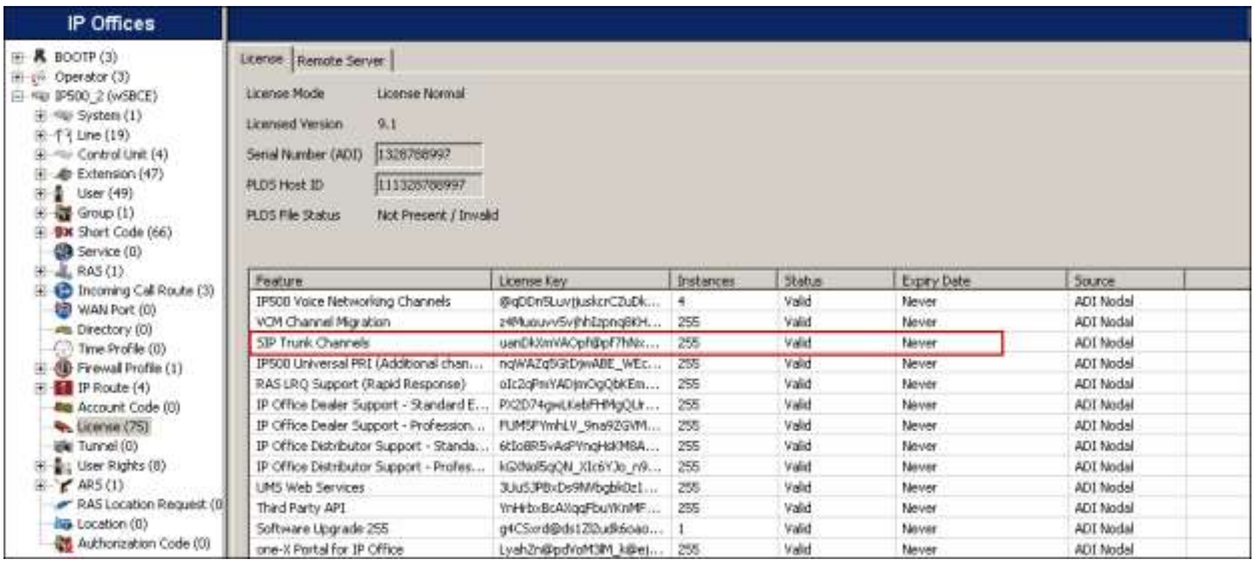
The appearance of the IP Office Manager can be customized using the View menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration.

Standard feature configurations that are not directly related to the interfacing with the service provider are assumed to be already in place, and they are not part of these Application Notes.

5.1. Licensing

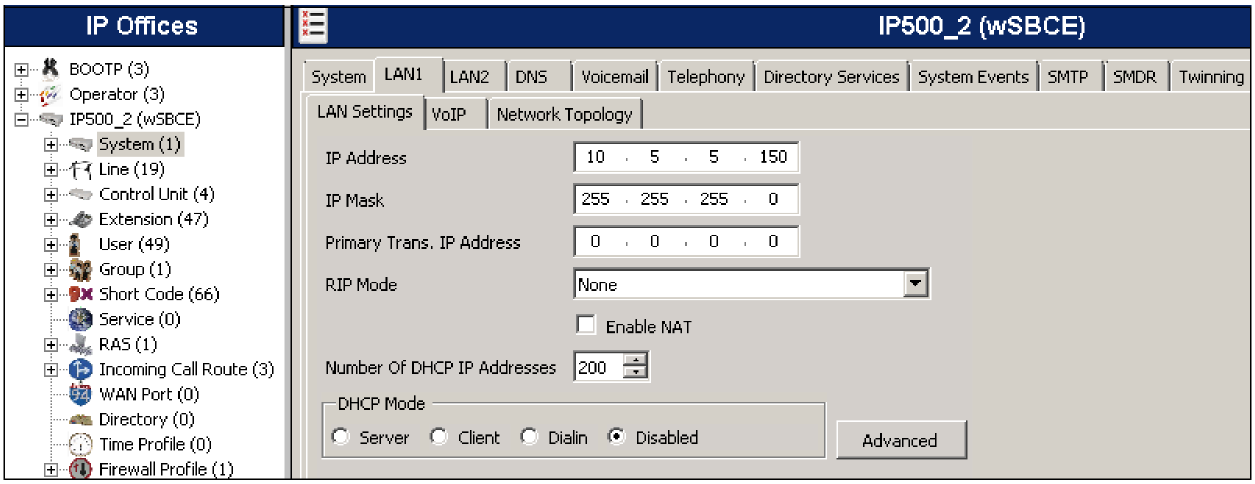
The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

In the reference configuration, ***IP500_2 (wSBCE)*** was used as the system name. Navigate to **IP500_2 (wSBCE)** in the Navigation pane and select **License**. Confirm that there is a valid **SIP Trunk Channels** license with sufficient "Instances" in the Details pane, enough to support the number of channels to be deployed on the SIP trunk to the service provider.

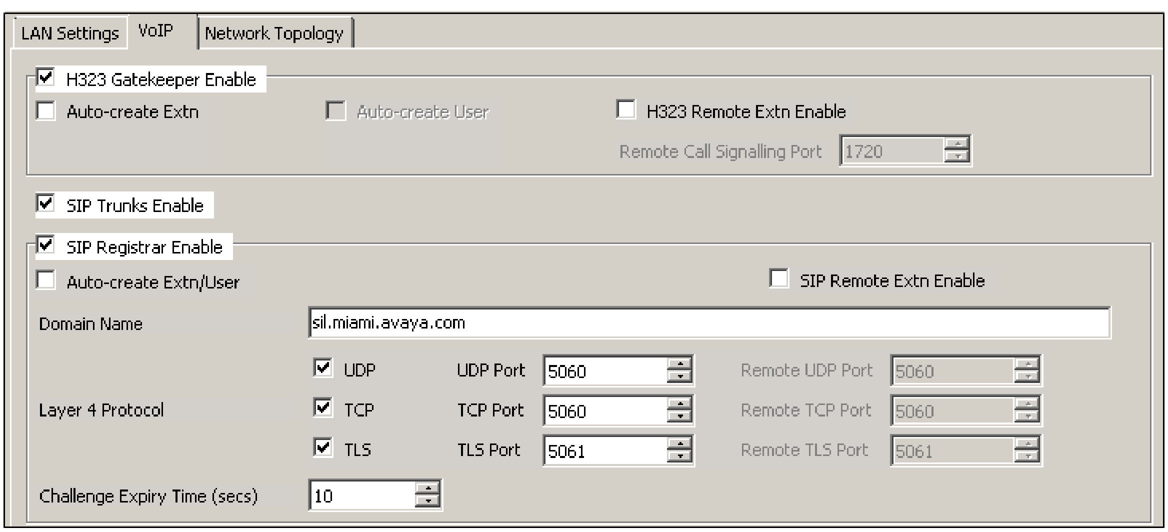


5.2. LAN Settings

In the sample configuration, the LAN1 port was used to connect the IP Office to the enterprise network. To access the LAN1 settings, first navigate to **System (1)** under the system name in the Navigation pane and select the **LAN1 → LAN Settings** tab in the Details pane. Set the **IP Address** and **IP Mask** fields to the IP address and subnet mask assigned to the Avaya IP Office LAN1 port. All other parameters should be set according to customer requirements.



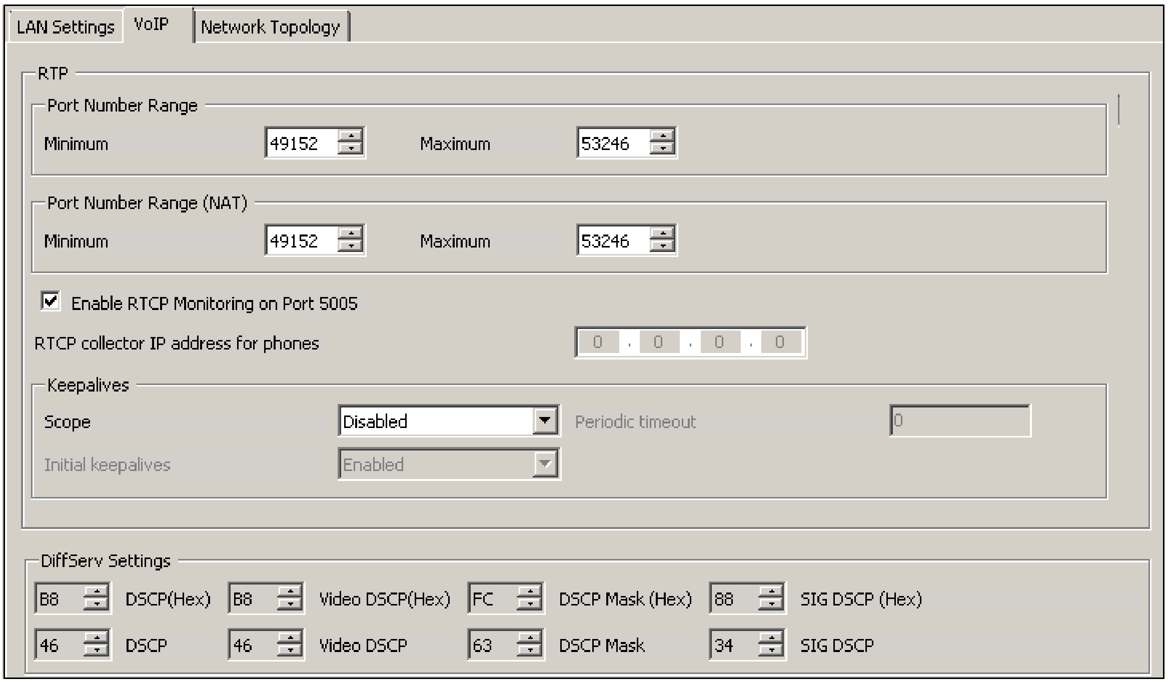
On the **VoIP** tab in the Details pane, the **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones with the H.323 protocol, such as the Avaya 1600 and 9600 Series IP Telephones present in the sample configuration. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks on this interface. The **SIP Registrar Enable** box is checked to allow the registration of Avaya 1140E Telephones and the Avaya Communicator and Avaya IP Office Softphones using the SIP protocol. On the **Domain Name** field, the local SIP registrar domain name ***sil.miami.avaya.com*** was used. This domain name will need to be configured on the SIP endpoints in order to register with the system. On the **Layer 4 Protocol** section, the default **UDP**, **TCP** and **TLS** protocols and ports were used.



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.

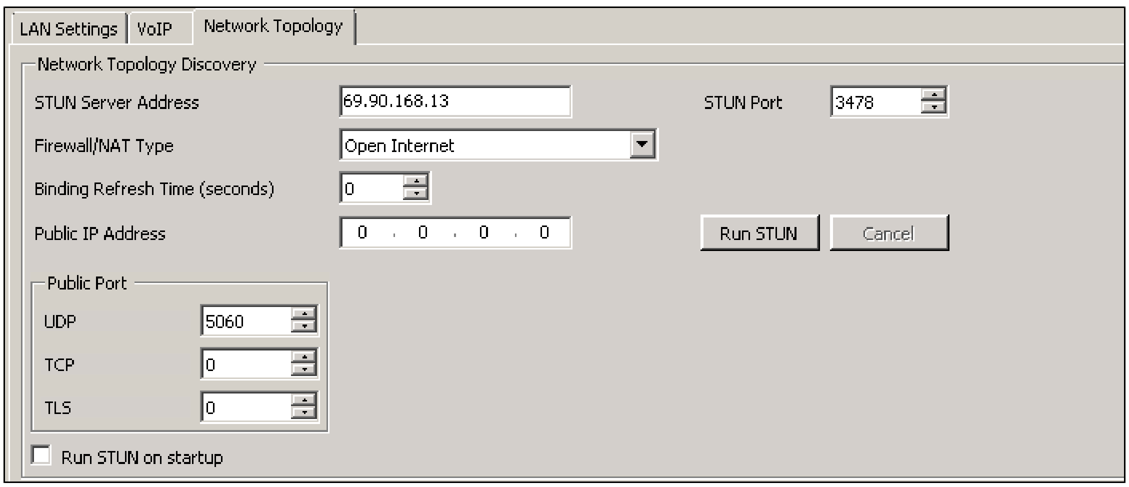
Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below.

All other parameters should be set according to customer requirements.



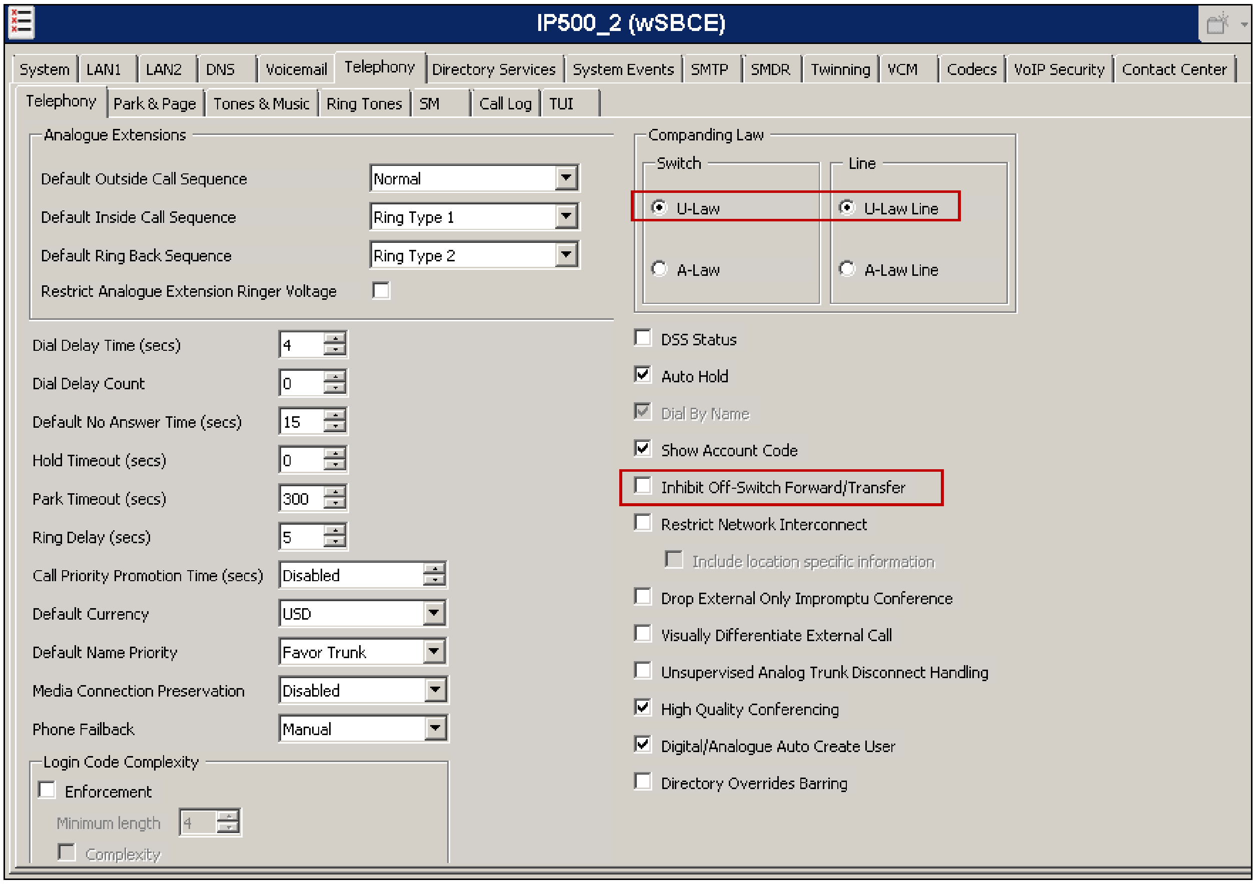
On the **Network Topology** tab in the Details pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu to the option that matches the network configuration. Since no network address translation (NAT) was used in the compliance test, the parameter was set to ***Open Internet***. With this configuration, settings obtained by STUN lookups are ignored. The IP address used is the one assigned to the interface.
- **Binding Refresh Time (seconds)** is used to determine the frequency at which Avaya IP Office will send SIP OPTION messages to the SIP trunk using this interface. In the reference configuration the Avaya SBCE was used to send OPTIONS to the service provider. This parameter was left at the default value ***0***.
- Set **Public Port** to ***5060*** for **UDP**.
- Defaults were used for all other fields.



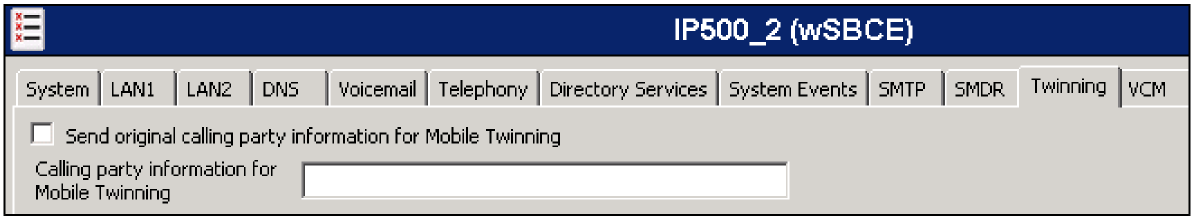
5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** Tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. ***U-Law*** was used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the SIP trunk to the service provider.



5.4. Twinning Calling Party Settings

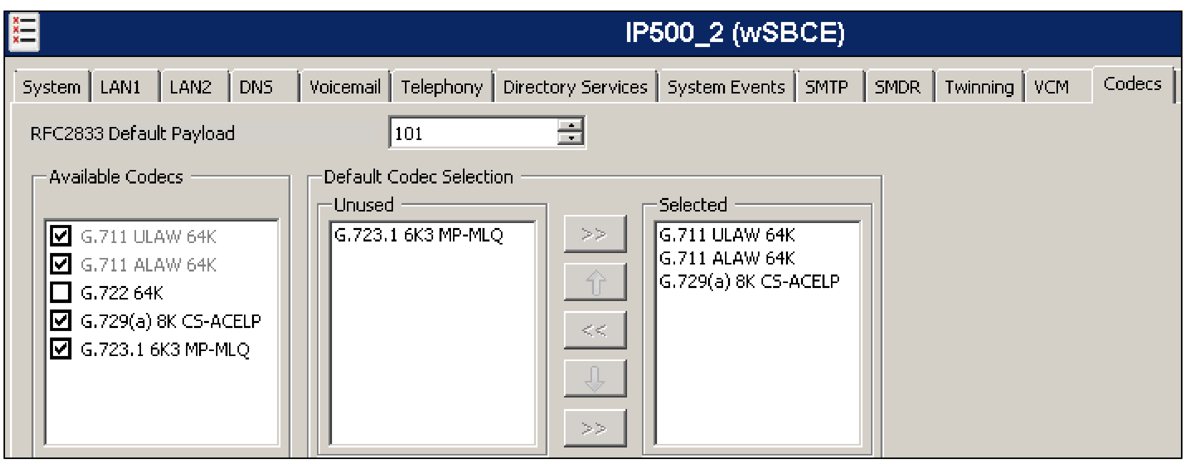
Navigate to the **Twinning** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be controlled by the setting on the SIP Line (**Section 5.7**). This setting also impacts the Caller ID for call forwarding.



5.5. System Codecs Settings

Navigate to the **Codecs** tab in the Details Pane. The **RFC2833 Default Payload** field allows the manual configuration of the payload type used on SIP calls that are initiated by the IP Office. The default value ***101*** was used. The list of **Available Codecs** shows all the codecs supported by the system, and those selected as usable. The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis. The buttons between the two lists can be used to move codecs between the **Unused** and **Selected** lists, and to change the order of the codecs in the **Selected** codecs list. By default, all IP (SIP and H.323) lines and extensions will use this system default codec selection, unless configured otherwise for a specific line or extension.

Click **OK** (not shown) to save any changes made to any of the various **System** tabs.

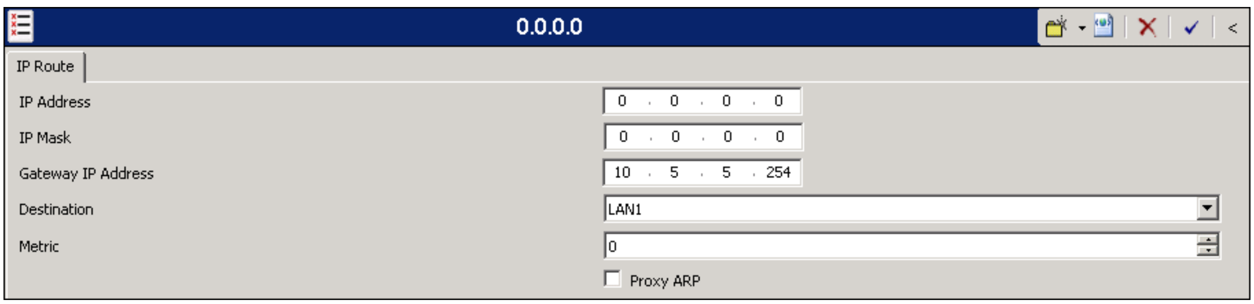


5.6. IP Route

In the reference configuration, the IP Office LAN1 interface and the private interface of the Avaya SBCE resided on the same subnet, so an IP route was not necessary. In an actual customer configuration, these two interfaces may be in different subnets, and in that case an IP route would need to be created to specify the IP address of the local gateway or router where the IP Office needs to send the packets, in order to reach the subnet where the Avaya SBCE is located.

To create an IP route, on the left navigation pane, right-click on **IP Route**. Select **New** (not shown).

- Set the **IP Address** and **IP Mask** of the subnet of the private side of the Avaya SBCE, or enter **0.0.0.0** to make this the default route.
- Set **Gateway IP Address** to the IP Address of the default router in the IP Office subnet.
- Set **Destination** to ***LAN1*** from the pull-down menu.
- Click **OK** (not shown) to save any changes.



5.7. Administer SIP Line

A SIP line is created to establish the SIP connection between the Avaya IP Office and the private interface of the Avaya SBCE. This line will carry outbound and inbound traffic between to and from the service provider.

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 5.7.1** and **Section 5.7.2** to create the SIP Line from the template.

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

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

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.7.3 – 5.7.7**.

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.7.3 – 5.7.7**.

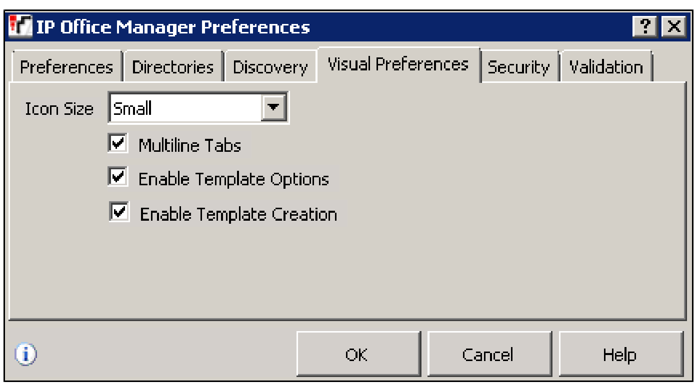
5.7.1. Importing a SIP Line Template

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 (500v2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

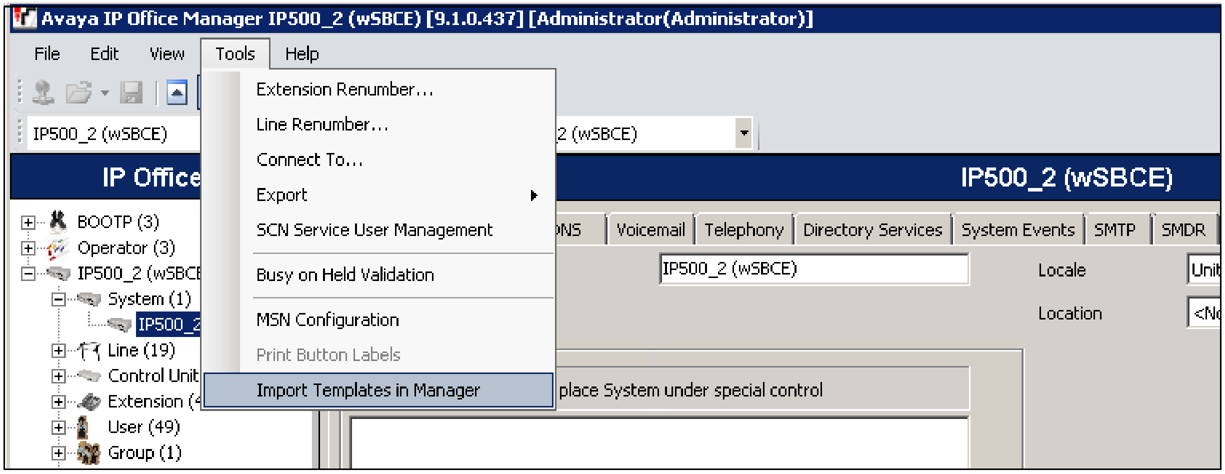
1. Copy a previously created template file to a location (e.g., *\Temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_<*user supplied text*>_SIPTrunk.xml**, where the **<*user supplied text*>** portion is entered during template file creation.

   Note – If necessary, the **<*user supplied text*>** portion of the template file name may be modified, however the **AF_<*user supplied text*>_SIPTrunk.xml** format of the file name must be maintained. For example, an original template file **AF_*TEST* _SIPTrunk.xml** could be changed to **AF_*Test1*_SIPTrunk.xml**. The template file name is selected in **Section 5.7.2** to create a new SIP Line.

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. Check the box next to **Enable Template Options**. Click **OK**.

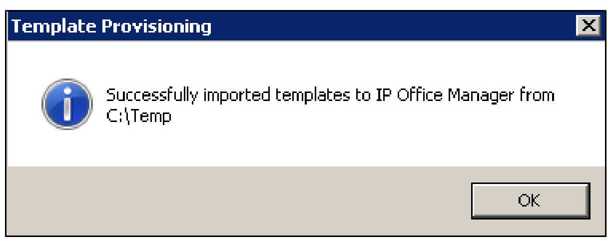


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.

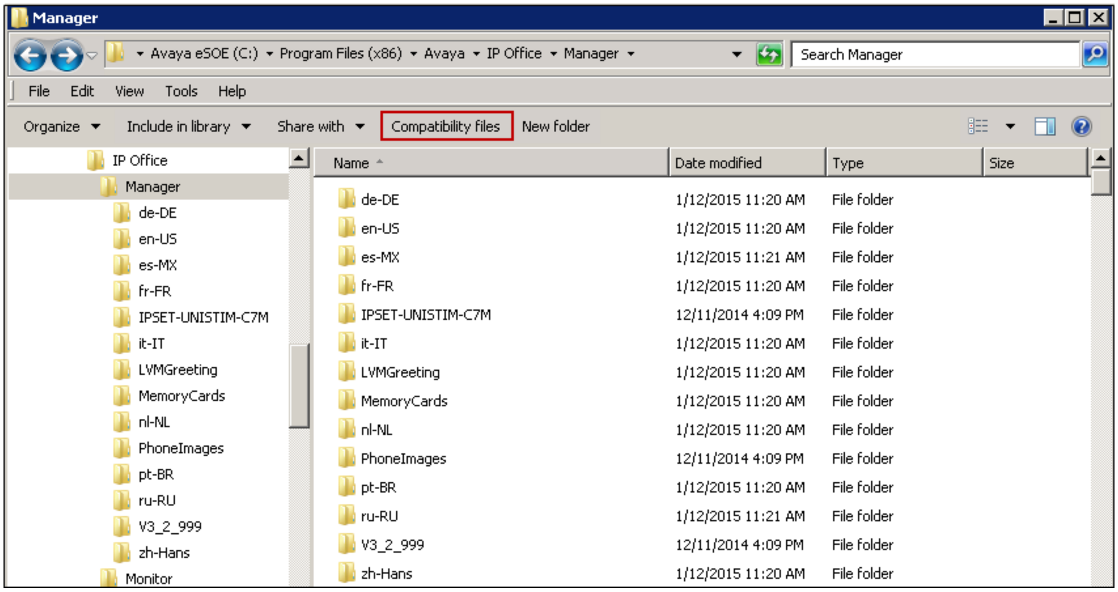


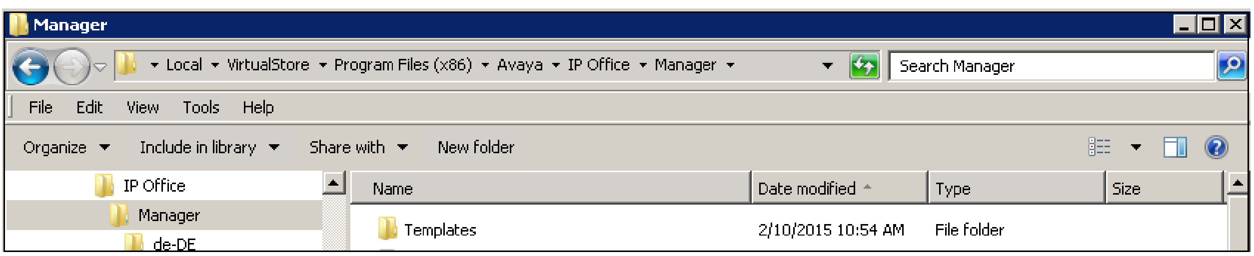
4. A folder browser will open (not shown). Select the directory used in **step 1** to store the template (e.g., *\Temp*). In the reference configuration, template file **AF*_Group of Gold Line_* SIPTrunk.xml** was imported. The template file is automatically copied into the default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

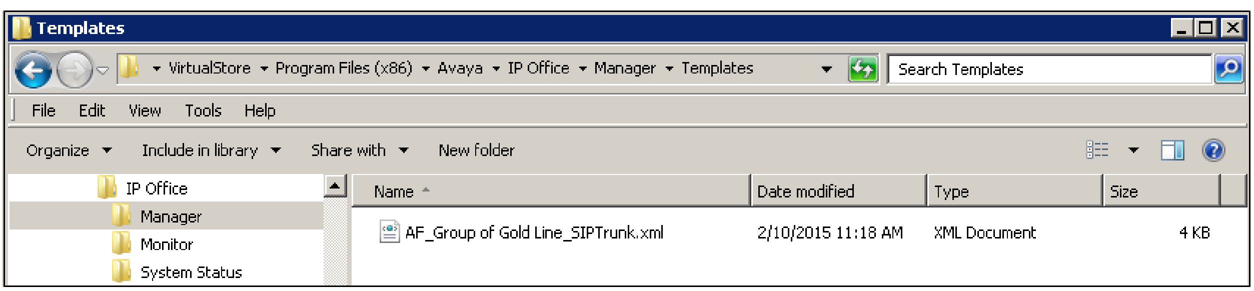
5. After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.



Note –Windows 7 (and later) locks the Avaya IP Office 9.1 **\Templates** directory, and it cannot be viewed. To enable browsing of the **\Templates** directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager** (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.

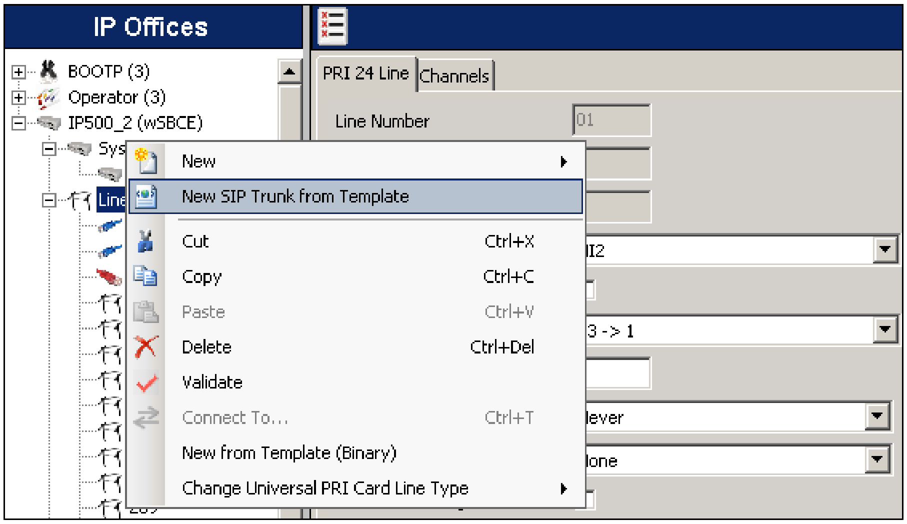






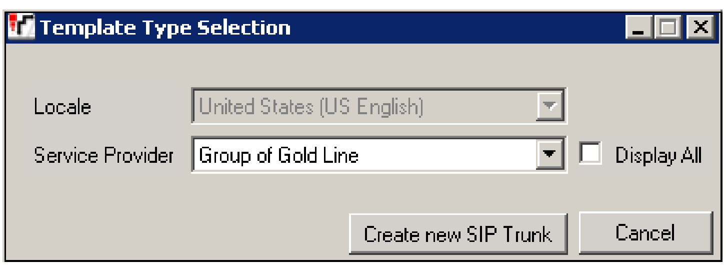
5.7.2. Creating a SIP Trunk from an XML Template

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



2. In the subsequent **Template Type Selection** pop-up window, from the **Service Provider** pull-down menu, select the XML template name from **Section 5.7.1**.

Note – The drop down menu will display the **<*user supplied text*>** part of the template file name (see **Section 5.7.1**). If you check the **Display All** box, then the full template file name is displayed.



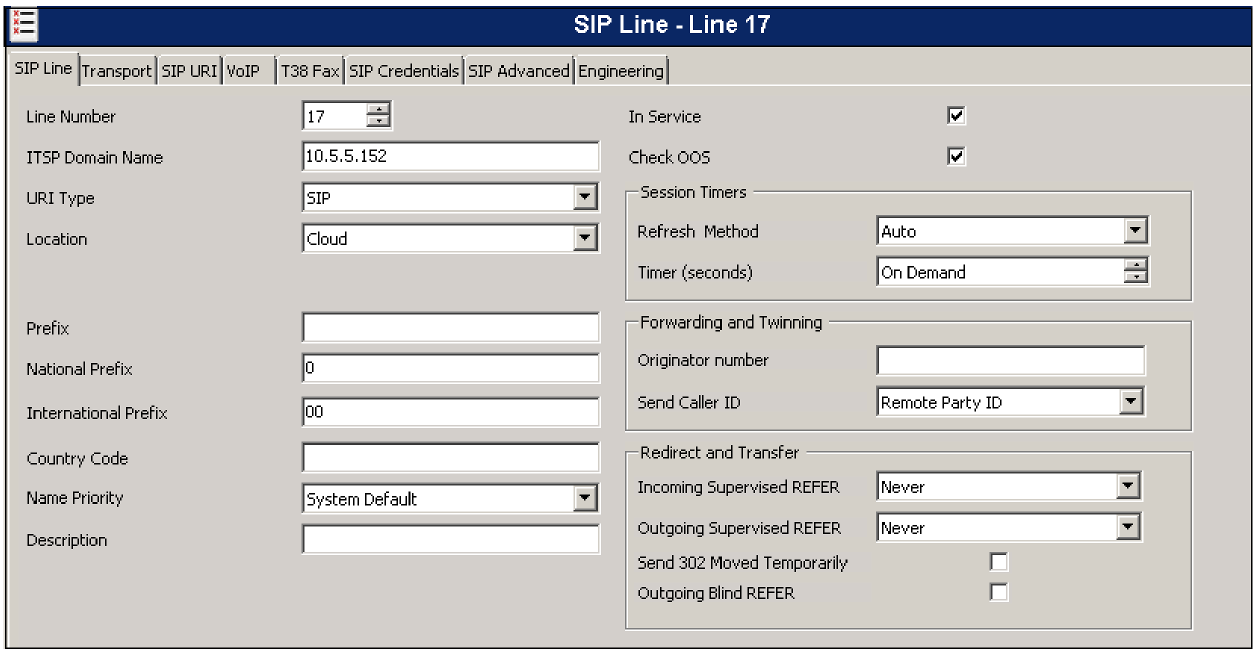
Click **Create new SIP Trunk** to finish creating the trunk.

3. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.7.3 – 5.7.7.**

5.7.3. SIP Line Tab

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

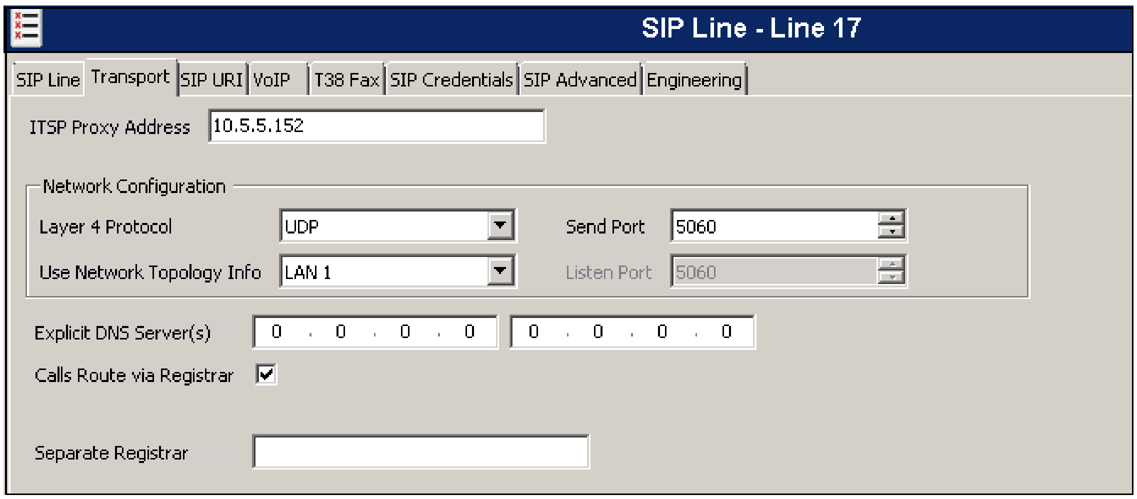
- Set the **ITSP Domain Name** to the IP address of the private interface of the Avaya SBCE.
- Check the **In Service** box.
- Check the **Check OOS** box.
- On the **Forwarding and Twinning** section, set **Send Caller ID** to ***Remote Party ID***. With this setting, Avaya IP Office will include the Remote-Party-ID header, with the originator's calling party information, on calls that are redirected via Call Forward or Mobile Twinning out the SIP Line to the service provider.
- On the **Redirect and Transfer section**, since REFER is not supported by the service provider, set **Incoming Supervised REFER** and **Outbound Supervised REFER** to ***Never***.
- Default values may be used for all other parameters.



5.7.4. Transport Tab

Select the **Transport** tab and set the following:

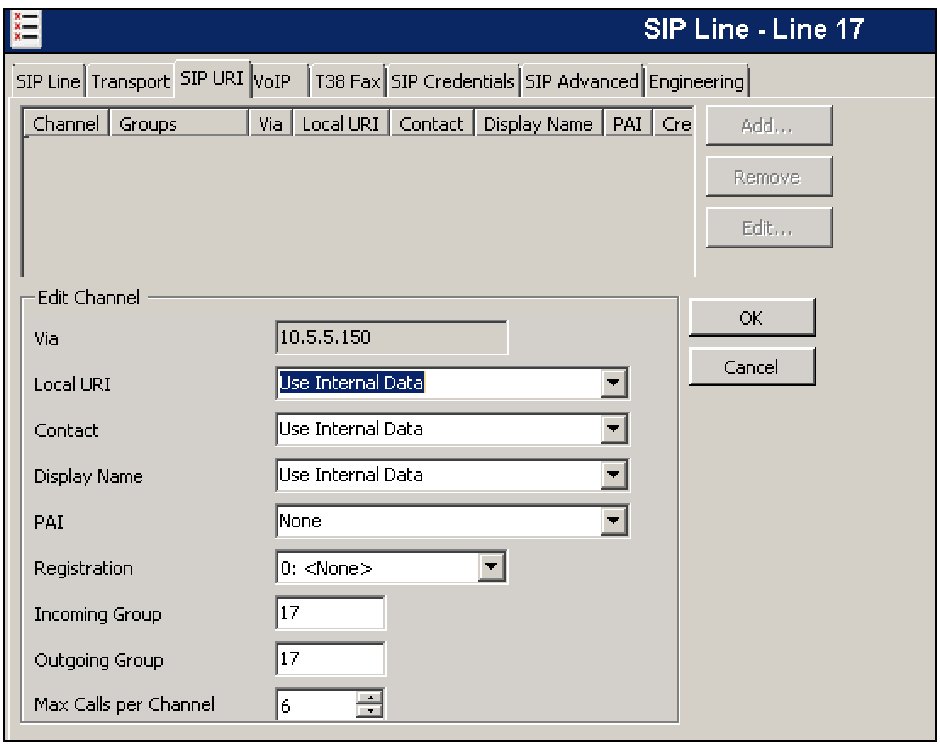
- Set the **ITSP Proxy Address** to the IP address of the private interface of the Avaya SBCE.
- Set the **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to ***LAN1*** as configured in **Section 5.2**.
- Set the **Send Port** to ***5060***.
- Default values may be used for all other parameters.



5.7.5. SIP URI Tab

A SIP URI entry needs to be created to match each number that Avaya IP Office and the service provider will accept on this line. Select the **SIP URI** tab, click the **Add** button and the **New Channel** area will appear at the bottom of the pane. In the example screen below, a previously configured entry was edited to use the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to ***Use Internal Data***. This setting allows calls on this line that have a SIP URI that matches the number set in the **SIP** tab of any user as shown later in **Section 5.8**.
- Set **PAI** to ***None***.
- Under **Registration**, select ***0: <None>*** from the pull-down menu. Group of Gold Line did not require SIP trunk registration.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group 17 was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous calls to be allowed on the SIP trunk using this SIP URI pattern.
- Click **OK**.

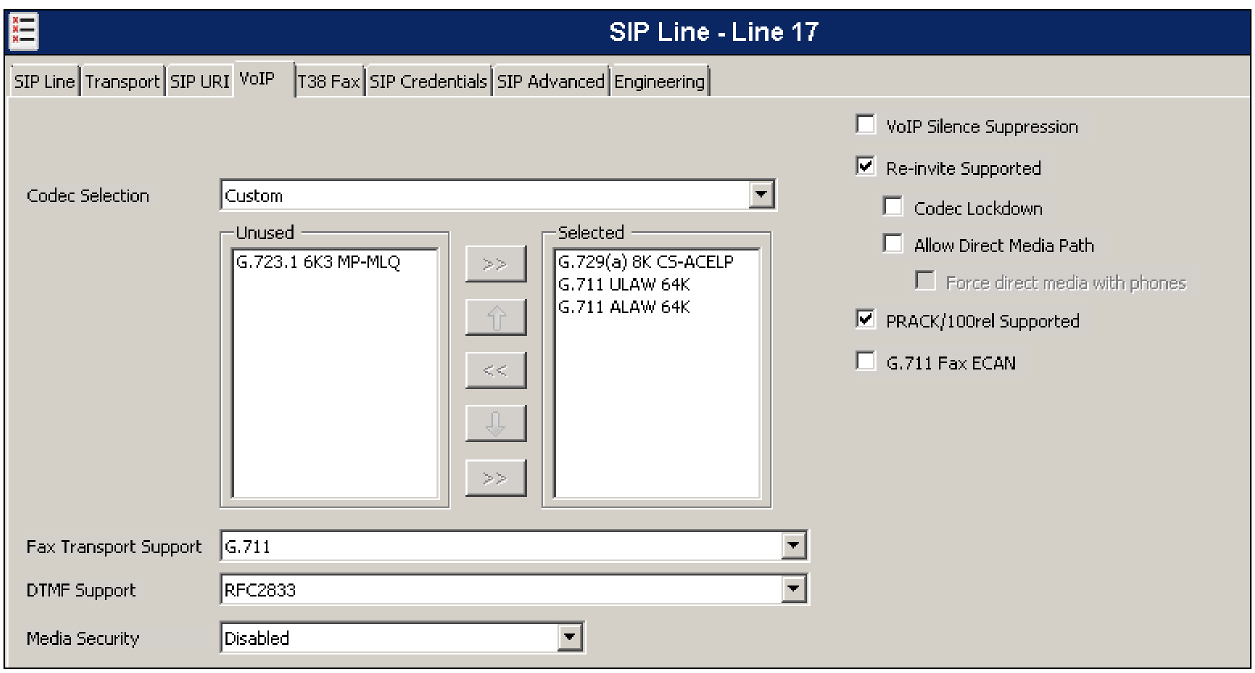


Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user, such as a short code. These URIs are created in the same manner as shown previously, with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, and **Display Name**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

5.7.6. VoIP Tab

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

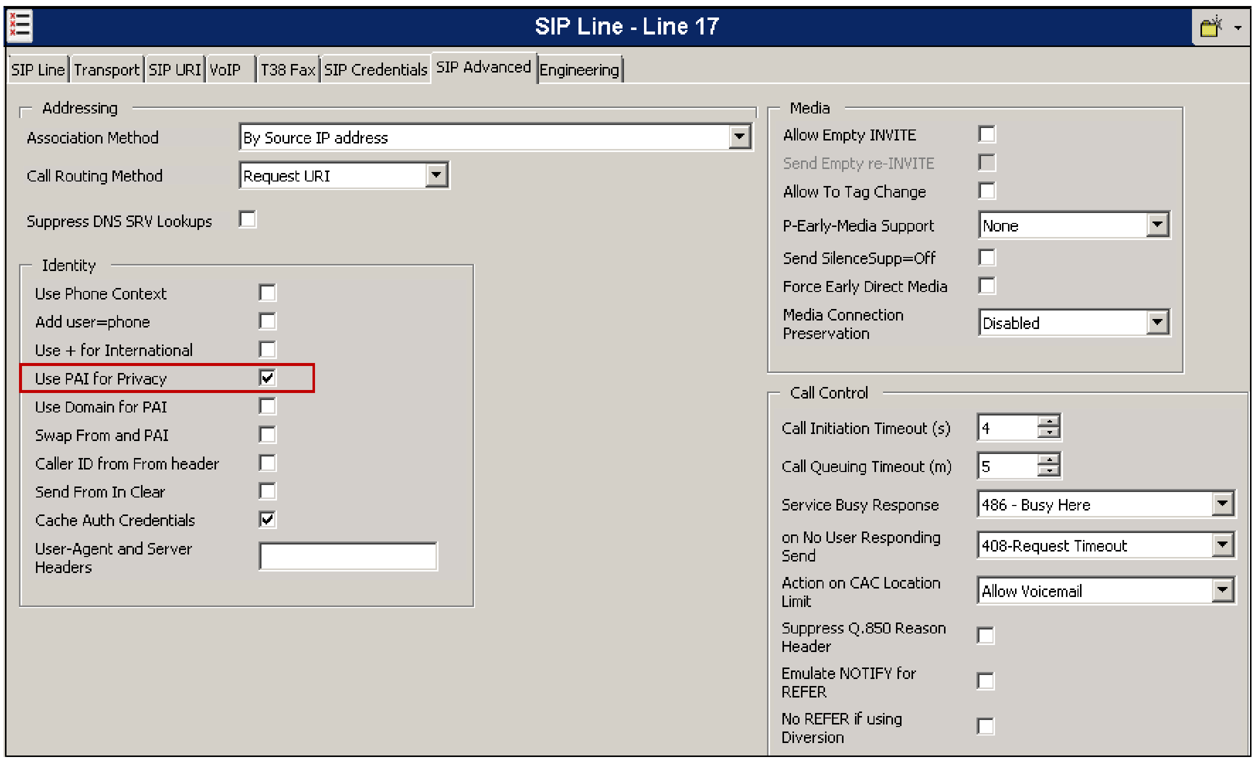
- In the sample configuration, the **Codec Selection** was configured using the ***Custom*** option, allowing an explicit ordered list of codecs to be specified. The buttons allow setting the specific order of preference for the codecs to be used on the line, as shown. During the compliance test, **G729A**, **G711U and G711A**, in this order of preference, were the codecs supported by Group of Gold Line.
- Set **Fax Transport Support** to ***G711***. See **Section 2.2**.
- Set the **DTMF Support** field to ***RFC2833.*** This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check the **PRACK/100rel Supported** box, to advertise the support for provisional responses and Early Media to the service provider.
- Default values may be used for all other parameters.



5.7.7. SIP Advanced Tab

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

To configure Avaya IP Office to use the PAI header for privacy calls, on the **SIP Advanced** tab, check **Use PAI for Privacy**. All other fields retained their default values.



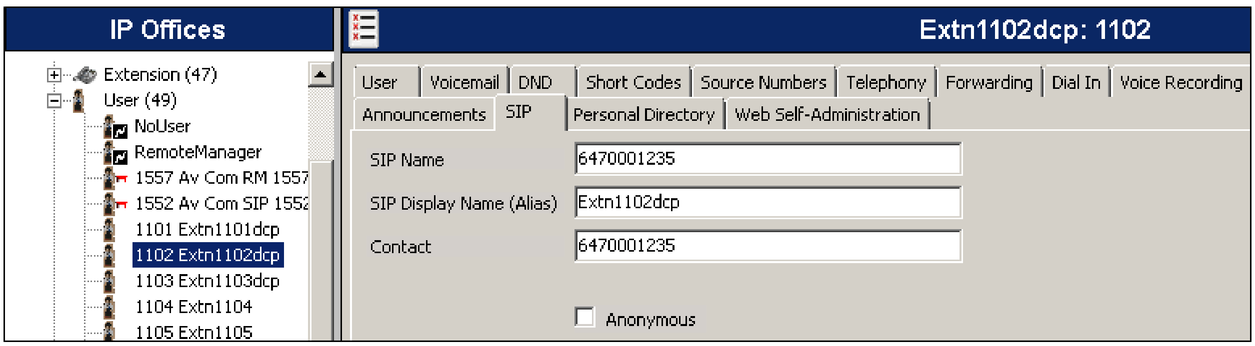
Click **OK** (not shown) to save any changes made to any of the various “SIP Line” tabs.

No changes were made to the **T38 Fax**, **SIP Credentials** and **Engineering** tabs, so they will not be visited**.**

5.8. Users

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.7**. To configure these settings, navigate to **User** in the left Navigation Pane and select the name of the user to be modified. In the example below, the name of the user is ***Extn 1102dcp***. 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. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.7.5**). The example below shows the settings for user "Extn1102dcp". The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by Group of Gold Line. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Click **OK** (not shown) to save any changes.



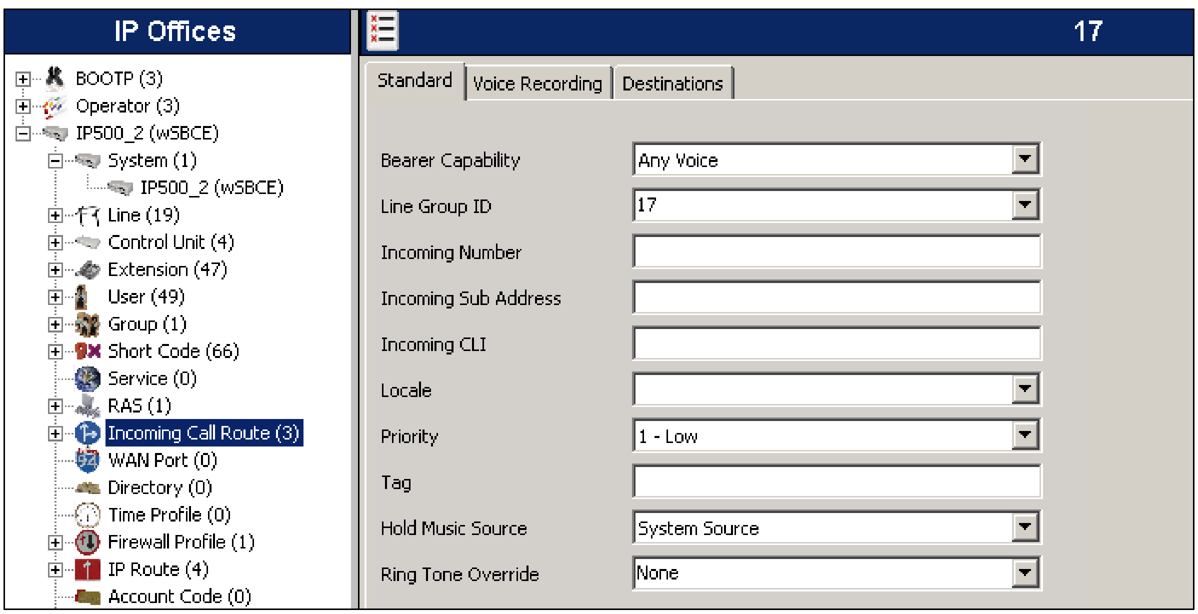
5.9. Incoming Call Route

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

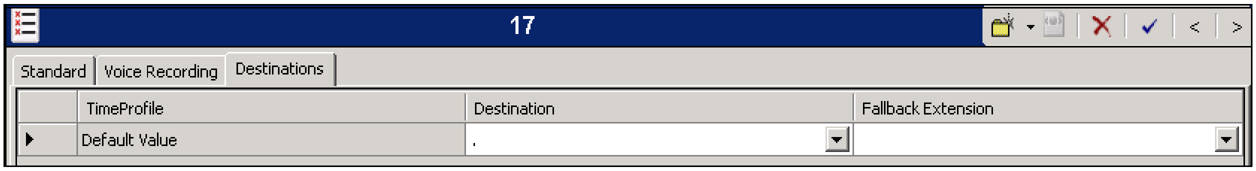
In a scenario like the one used for the compliance test, only one incoming route was needed, which allowed any incoming number arriving on the SIP trunk to reach any predefined extension in the IP Office. The routing decision for the call is based on the parameters previously configured for the **SIP URI (Section 5.7.5)** and the users **SIP Name** and **Contact**, already populated with the assigned DID numbers **(Section 5.8)**

To add a new incoming call route, from the left Navigation Pane, right-click on **Incoming Call Route** and select **New** (not shown). On the Details Pane, under the **Standard** tab, set the parameters as show below:

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



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



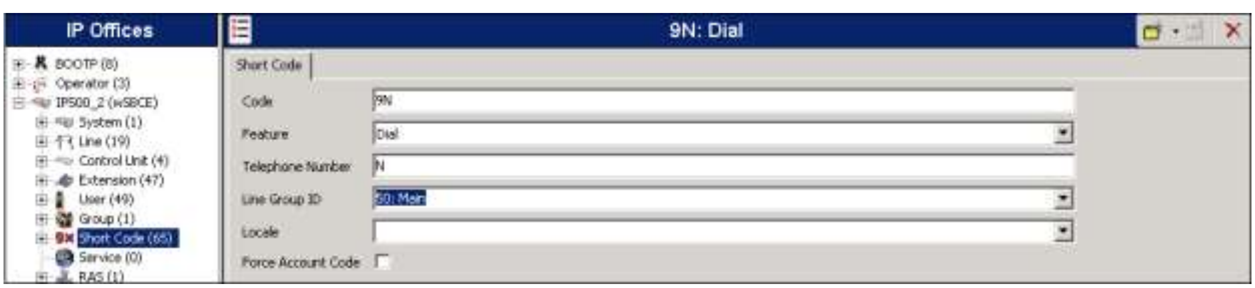
Additional incoming call routes may be required to allow inbound calls to numbers not associated with a user, such as a short code. These routes are created in the same manner as shown, with the exception that the incoming DID number is entered directly in the **Incoming Number** field on the **Standard** tap, and the specific destination (short code, etc.) needs to be entered on the **Default Value** field of the **Destinations** tab. Click **OK** (not shown) to save any changes.

5.10. Short Code

In the reference configuration, Avaya IP Office used Automatic Route Selection (ARS) to route outbound traffic to the SIP line. A short code is needed to send the outbound traffic to the ARS route. To create the short code used for ARS, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). The screen below shows the creation of the short code ***9N*** used in the reference configuration. When the Avaya IP Office users dialed 9 plus any number N, calls were directed to **Line Group *50: Main***, configurable via ARS and defined next in **Section 5.11**

On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

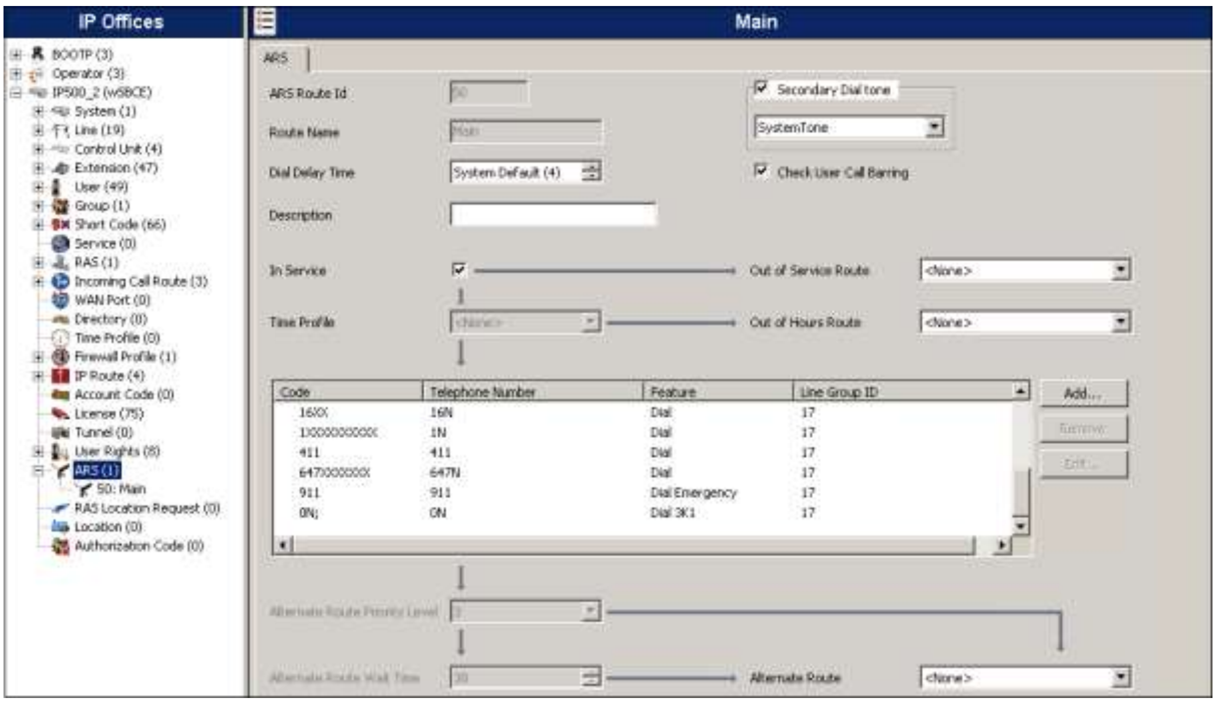
- In the **Code** field, enter the dial string which will trigger this short code, in this case ***9N***. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to **N**. The value **N** represents the number dialed by the user after removing the **9** prefix.
- Set the **Line Group ID** to the ARS route to be used. In the example shown, the call is directed to **Line Group *50: Main***.
- Click **OK** (not shown).



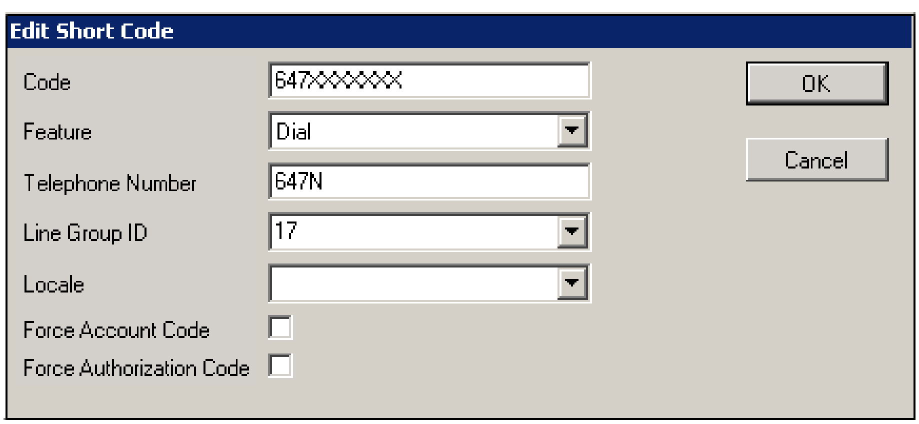
5.11. Automatic Route Selection

While detailed coverage of ARS is beyond the scope of these Application Notes, this section includes some basic screen illustrations of the ARS settings used during the compliance test.

The following screen shows the ARS configuration for the route ***50: Main***. The example shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. Note the sequence of ***X***s used in the **Code** column of some entries, to specify the exact number of digits to be expected following the access code and the first digits on the string. This type of setting results in a much quicker response in the delivery of the calls by the IP Office.



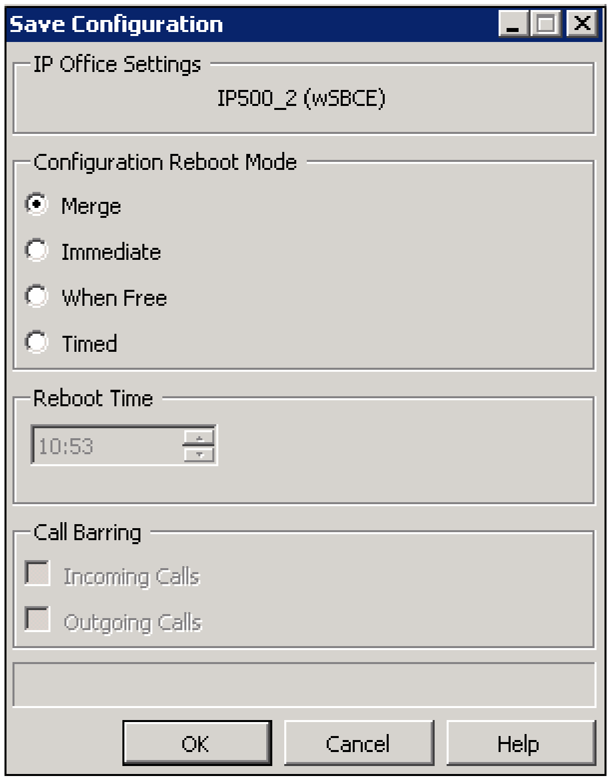
For example, during the compliance test, to dial local PSTN calls the user dialed 9 plus the 10 digit local number, starting with the area code 647 and then the remaining 7 digits.



5.12. Save Configuration

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

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.

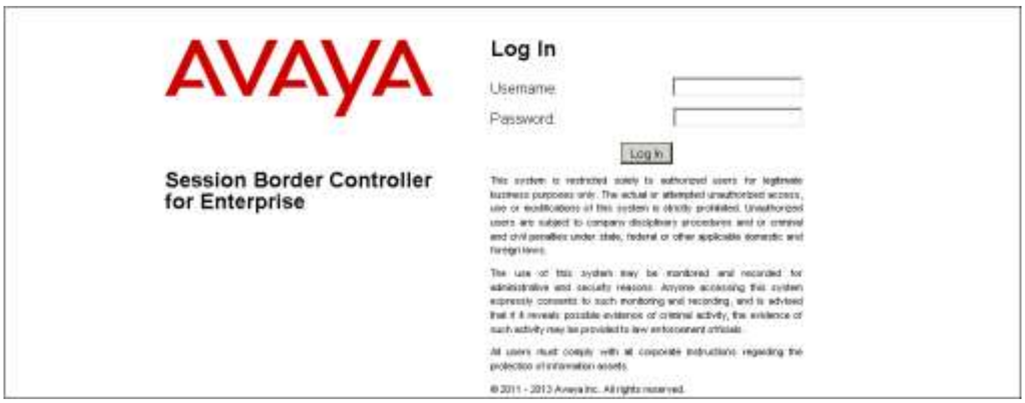


6. Configure Avaya Session Border Controller for Enterprise

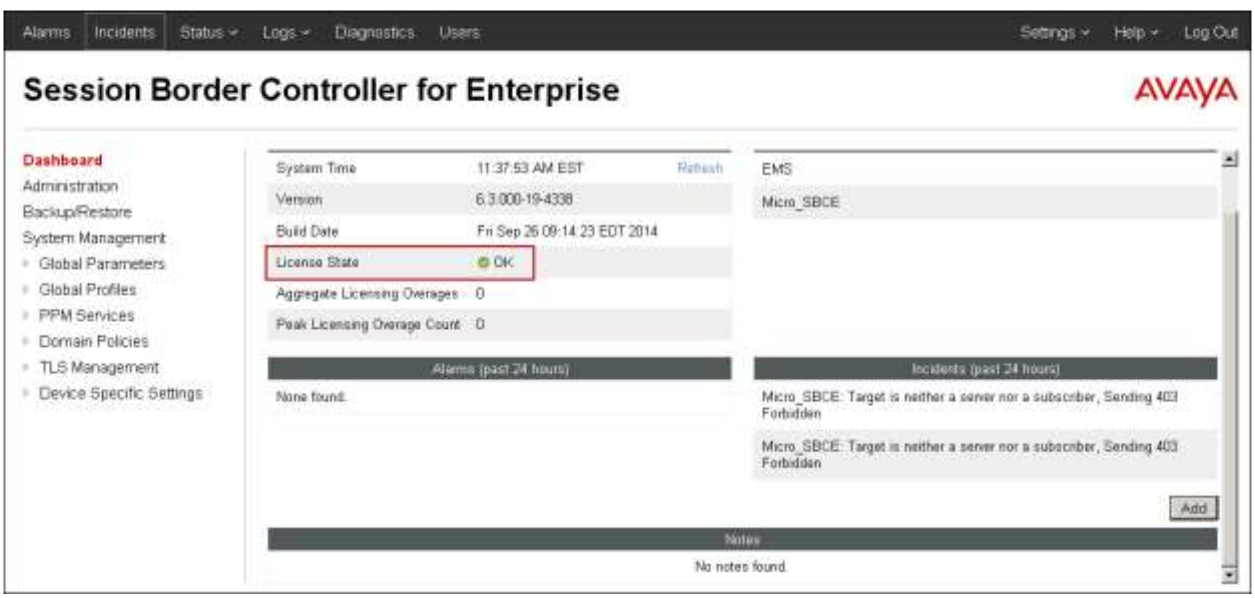
This section describes the configuration of the Avaya SBCE. It is assumed that the initial installation of the Avaya SBCE, the assignment of the management interface IP Address and license installation have already been completed; hence these tasks are not covered in these Application Notes. For more information on the installation and initial provisioning of the Avaya SBCE consult the Avaya SBCE documentation in the **Additional References** section.

6.1. System Access

Access the Session Border Controller web management interface by using a web browser and entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation. Log in using the appropriate credentials.



Once logged in, the Dashboard screen is presented. The left navigation pane contains the different available menu items used for the configuration of the Avaya SBCE. New in Release 6.3 of the Avaya SBCE is the **License State** field. In the example below, the status **OK** indicates that a valid license is present.

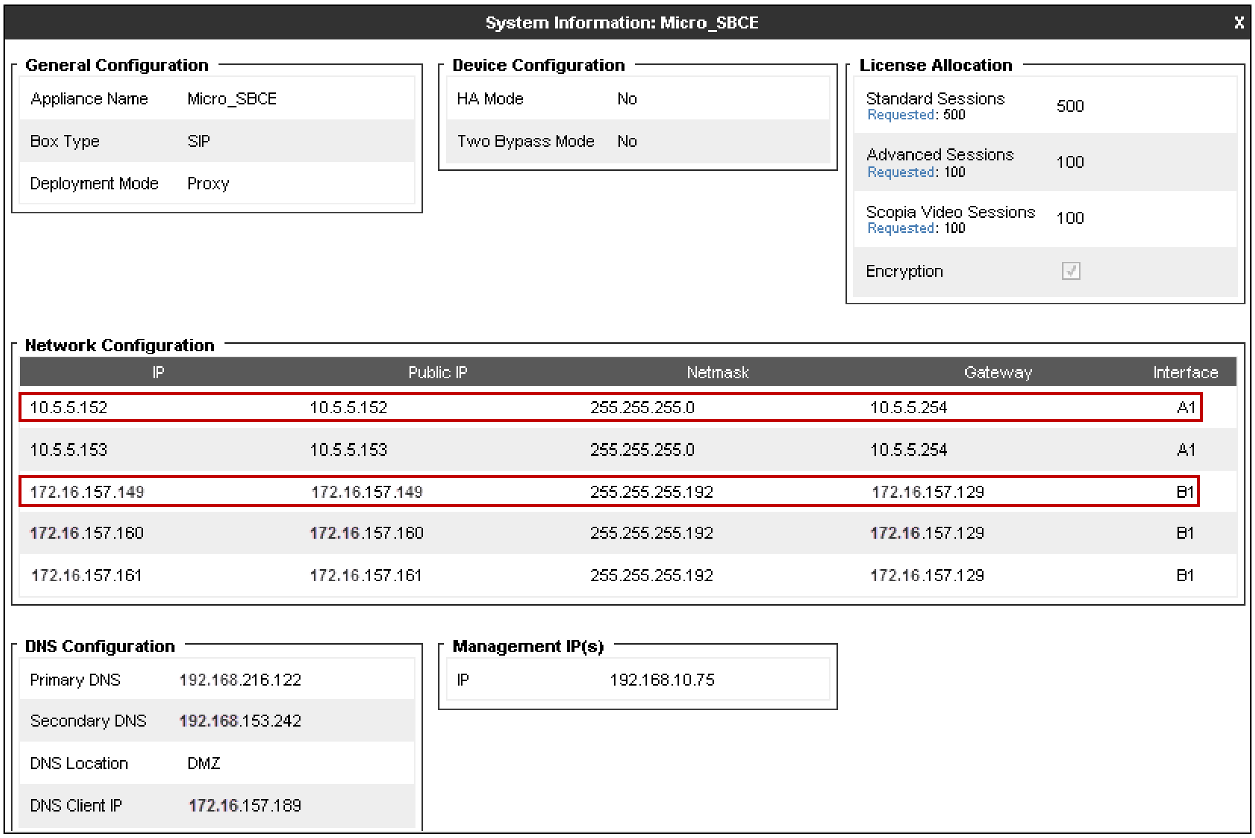


6.2. System Management

To view current system information, select **System Management** on the left navigation pane. A list of installed devices is shown in the **Devices** tab on the right pane. In the reference configuration, a single device named ***Micro_SBCE*** is shown. The management IP address that was configured during installation is shown here. Note that the management IP address needs to be on a subnet separate from the ones used in all other interfaces of the Avaya SBCE, segmented from all VoIP traffic. Verify that the **Status** is ***Commissioned***, indicating that the initial installation process of the device has been previously completed, as shown on the screen below.

To view the network configuration assigned to the Avaya SBCE, click **View** on the screen above. The **System Information** window is displayed, as shown on the screen on the next page, containing the current device configuration and network settings.

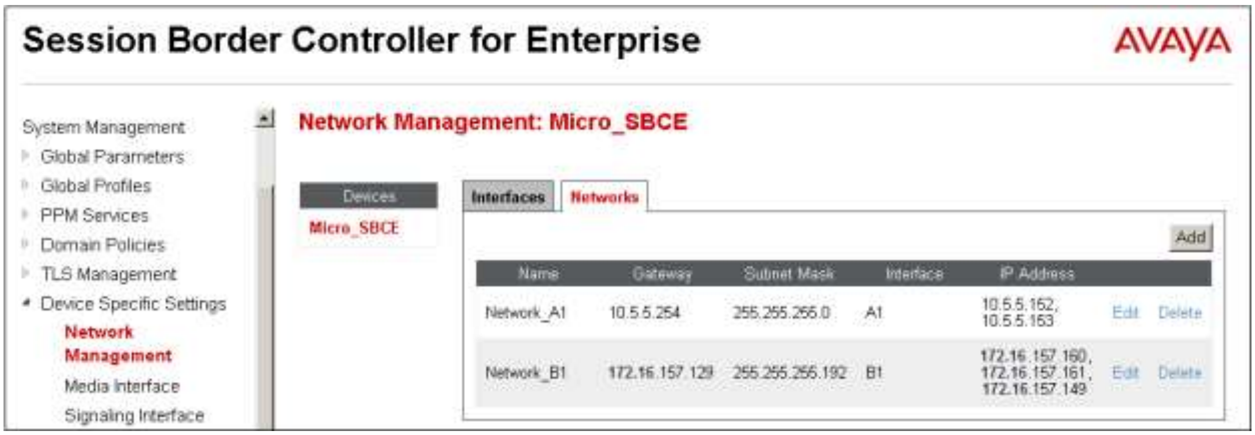
Note that the **A1** and **B1** interfaces correspond to the private and public interfaces for the Avaya SBCE. The highlighted **A1** and **B1** IP addresses are the ones relevant to these Application Notes. Other IP addresses assigned to these interfaces on the screen below are used to support remote workers and they are not discussed in this document. On the **License Allocation** area of the **System Information**, verify that the number of **Standard Sessions** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise. The number of sessions and encryption features are primarily controlled by the license file installed.



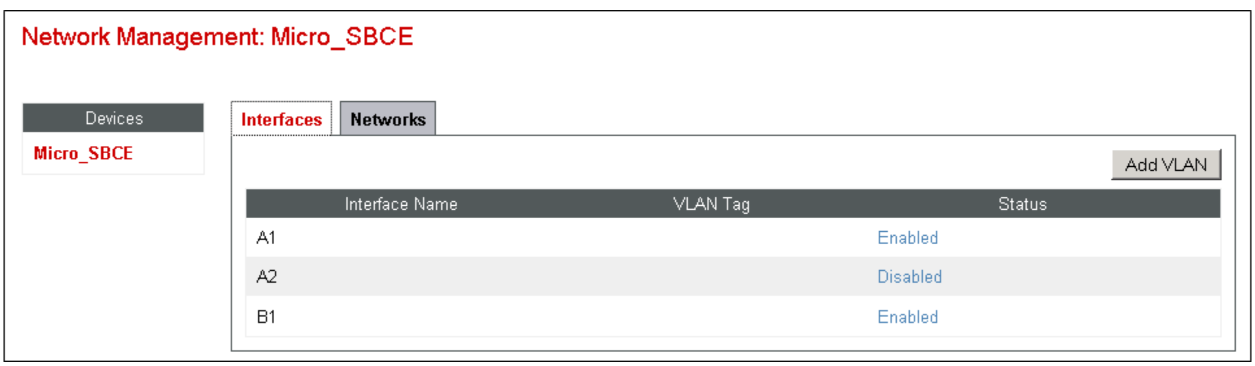
6.3. Network Management

The network configuration parameters should have been previously specified during installation of the Avaya SBCE. In the event that changes need to be made to the network configuration, they can be entered here.

Select **Network Management** under **Device Specific Settings** on the left-side menu. Under **Devices** in the center pane, select the device being managed, **Micro_SBCE** in the sample configuration. On the **Networks** tab, verify or enter the network information as needed. Note that the **A1** and **B1** interfaces correspond to the private and public interfaces for the Avaya SBCE. In the configuration used during the compliance test, IP address ***10.5.5.152*** was assigned to interface **A1**, and IP address ***172.16.157.149*** was assigned to interface **B1**. Other IP addresses assigned to these interfaces on the screen below are used to support remote workers and they are not discussed in this document. See **Figure 1** in **Section 3**.



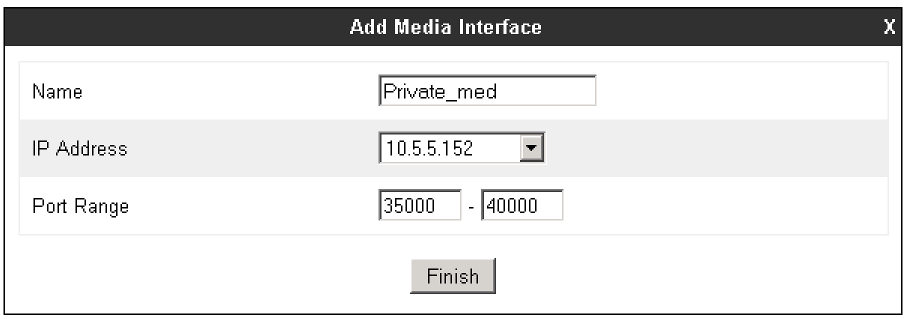
On the **Interfaces** tab, verify the **Status** is ***Enabled*** for both the **A1** and **B1** interfaces. Click the buttons if necessary to enable the interfaces.



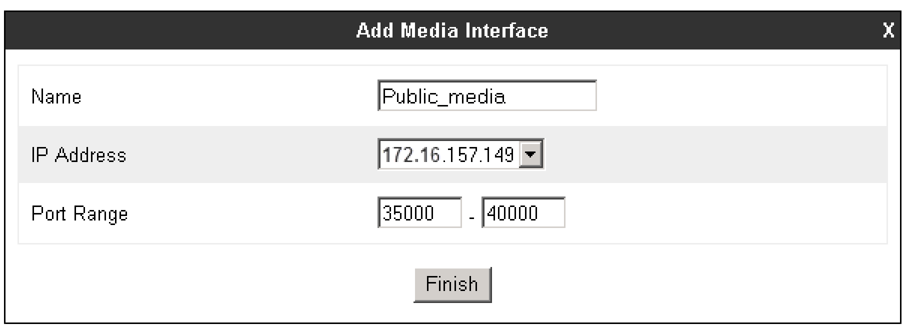
6.4. Media Interfaces

Media Interfaces were created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address, and one of the ports in this range as the listening IP address and port in which it will accept media from the Call or the Trunk Server.

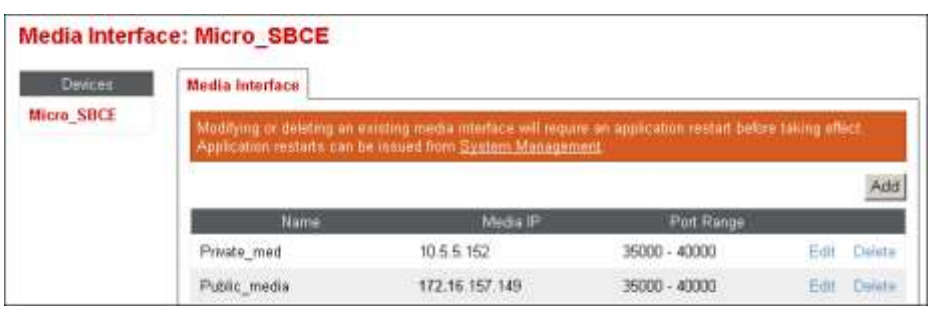
To add the Media Interface in the enterprise direction, select **Media Interface** from the **Device Specific Settings** menu on the left-hand side, select the **Micro_SBCE** device and click the **Add** button (not shown). On the **Add Media Interface** screen, enter an appropriate **Name** for the Media Interface. Select the private IP Address for the Avaya SBCE facing the enterprise from the **IP Address** drop-down menu. The **Port Range** was left at the default values of ***35000-40000***. Click **Finish**.



A Media Interface facing the public network side was similarly created with the name ***Public_med***, as shown below. The outside IP Address of the Avaya SBCE was selected from the drop-down menu. The **Port Range** was left at the default values. Click **Finish**.



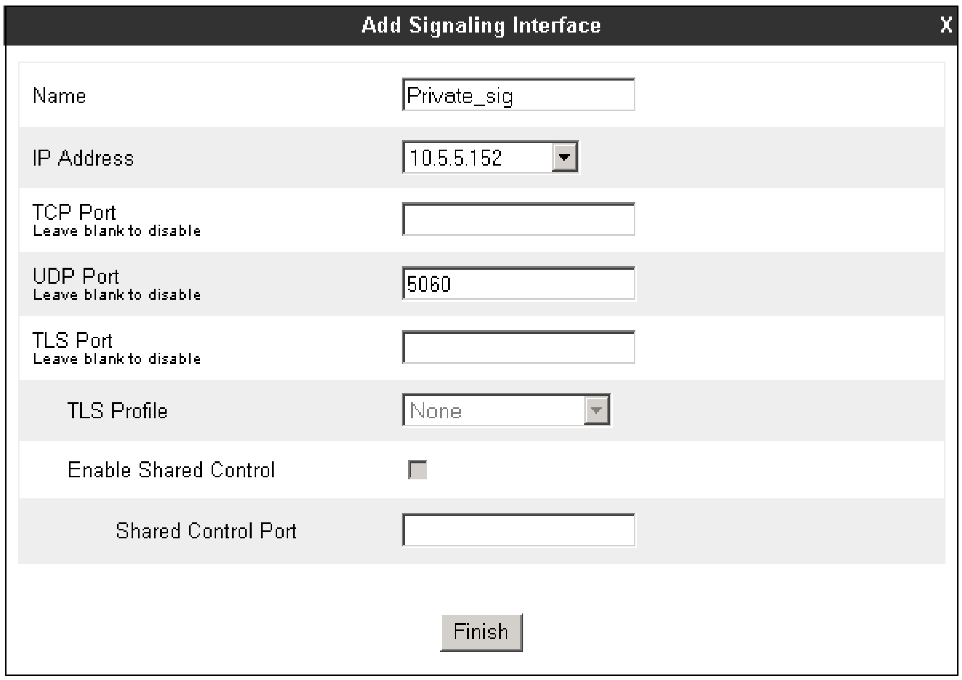
Once the configuration is completed, the **Media Interface** screen will appear as follows.



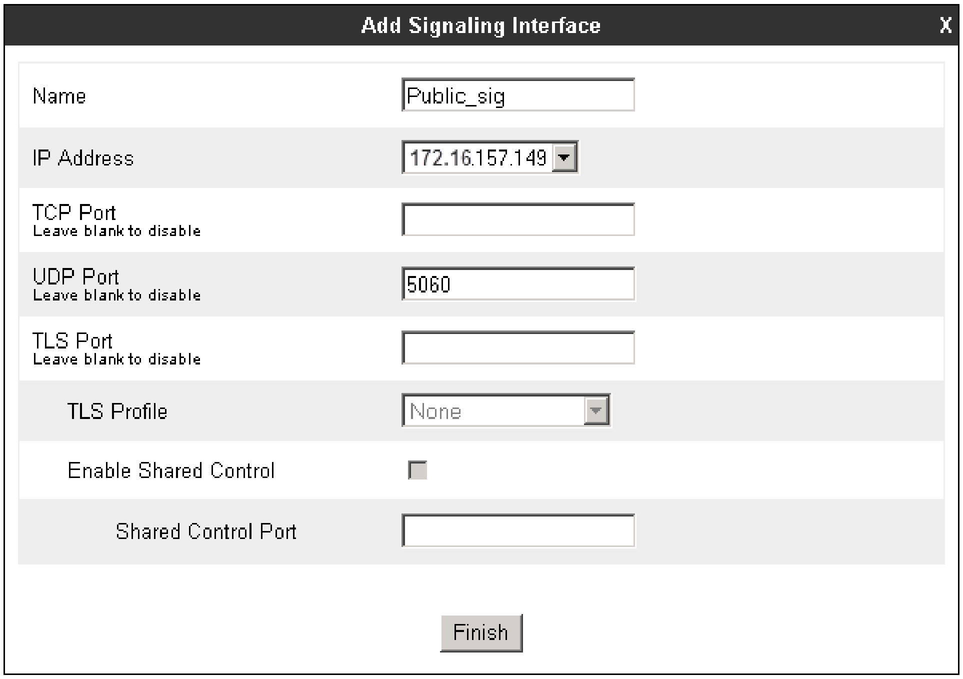
6.5. Signaling Interfaces

Signaling Interfaces are created to specify the IP addresses and ports in which the Avaya SBCE will expect the signaling traffic in the connected networks.

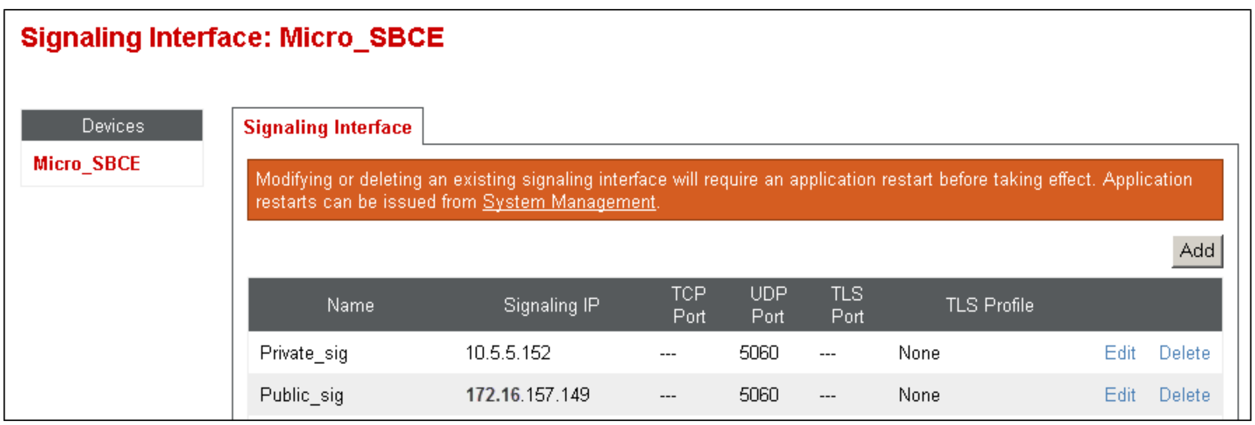
To add the Signaling Interface in the enterprise direction, select **Signaling Interface** from the **Device Specific Settings** menu on the left-hand side, select the **Micro_SBCE** device and click the **Add** button (not shown). On the **Add Signaling Interface** screen, enter an appropriate **Name** for the interface. Select the private IP Address of the Avaya SBCE from the **IP Address** drop-down menu. Enter ***5060*** for **UDP Port**, since UDP port 5060 is used for signaling traffic from IP Office in the sample configuration, **Section 5.7.4**. Click **Finish**.



A second Signaling Interface with the name **Public_sig** was similarly created in the service provider's direction. The public IP Address of the Avaya SBCE was selected from the **IP Address** drop-down menu. Enter ***5060*** for **UDP Port**. Click **Finish**.



Once the configuration is completed, the **Signaling Interface** screen will appear as follows:



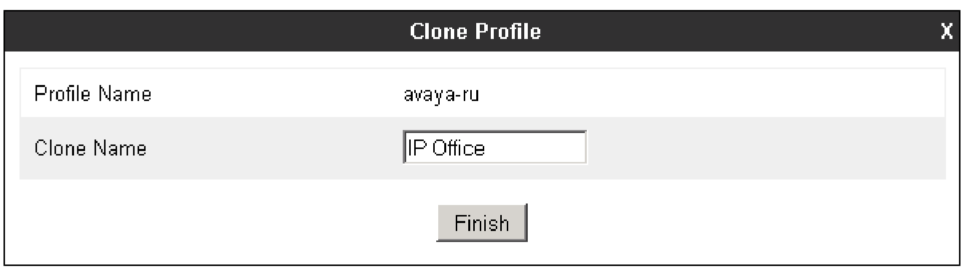
6.6. Server Interworking

Interworking Profile features are configured to facilitate the interoperability between the enterprise SIP-enabled solution (Call Server) and the SIP trunk service provider (Trunk Server).

6.6.1. Server Interworking Profile – Avaya IP Office

Interworking profiles can be created by cloning one of the pre-defined default profiles, or by adding a new profile. To configure the interworking profile in the enterprise direction, select **Global Profiles → Server Interworking** on the left navigation pane. Under **Interworking Profiles**, select ***avaya-ru*** from the list of pre-defined profiles. Click **Clone**.

Enter a descriptive name for the cloned profile. Click **Finish**.



On the newly cloned ***IP Office*** interworking profile, verify the settings on the **General** tab:

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
URI Group	None
Send Hold	No
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No

Scroll down to the bottom of the tab to see the rest of the settings. Click **Edit** (not shown) if changes to any of the parameters are needed.

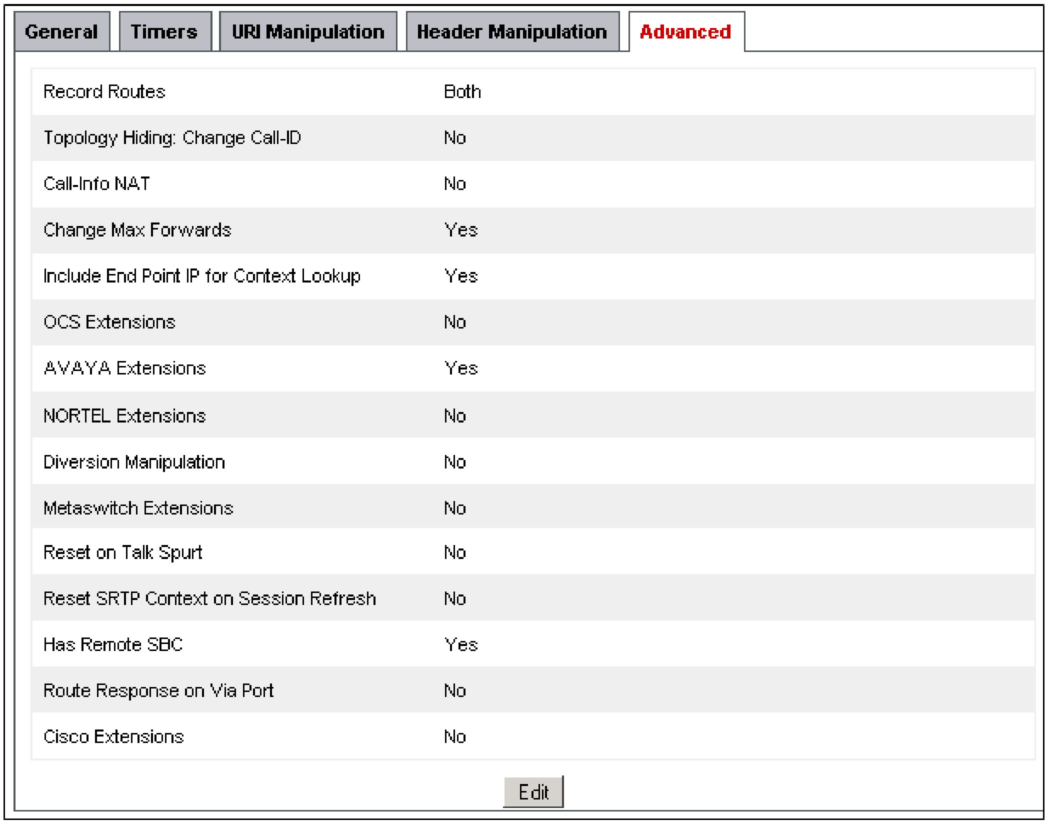
General | Timers | URI Manipulation | Header Manipulation | Advanced

Re-Invite 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

The **Timers**, **URI Manipulation** and **Header Manipulation** tabs contain no entries. The **Advaced** tab settings are shown on the screen below:

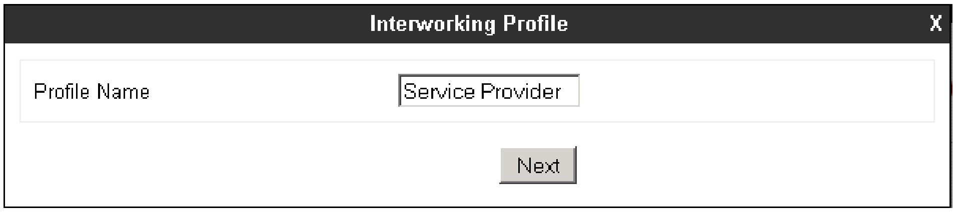

General | Timers | URI Manipulation | Header Manipulation | Advanced

Record Routes	Both
Topology Hiding: Change Call-ID	No
Call-Info NAT	No
Change Max Forwards	Yes
Include End Point IP for Context Lookup	Yes
OCS Extensions	No
AVAYA Extensions	Yes
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

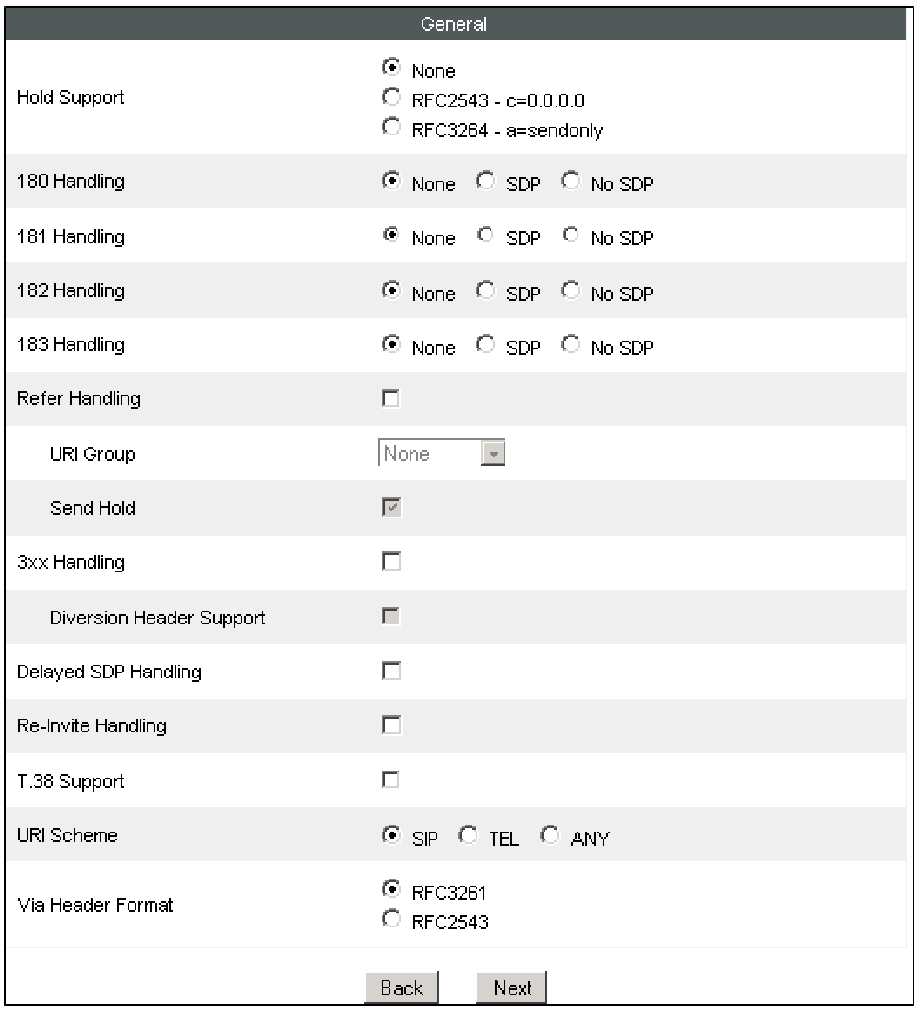
Edit

6.6.2. Server Interworking Profile – Service Provider

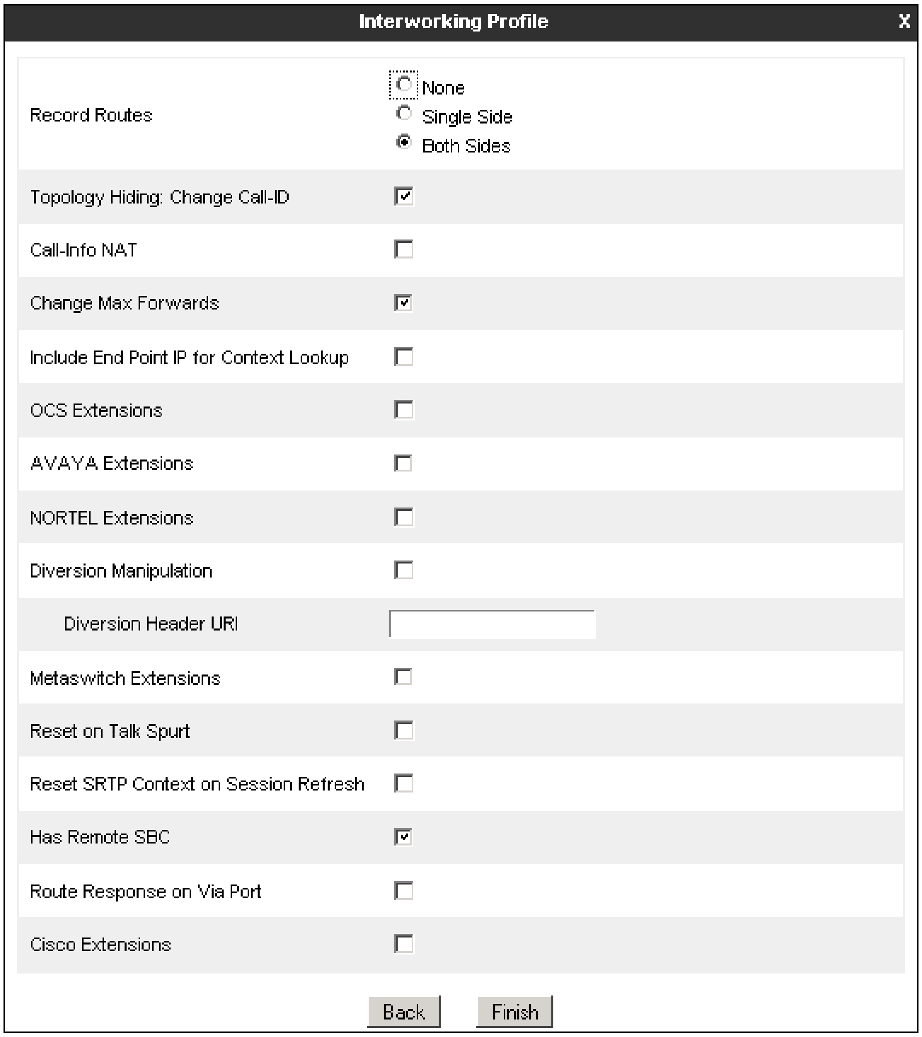
A second interworking profile in the direction of the SIP trunk to the service provider was created, by adding a new profile in this case. Select **Global Profiles → Server Interworking** on the left navigation pane and click **Add** (not shown). Enter a descriptive name for the new profile. Click **Next**.



On the **General** screen, all parameters retain their default values. Click **Next**.



Click **Next** on the **Privacy/DTMF** and **SIP Timers/Transport Timers** tabs (not shown). Accept all defaults in the **Advanced Settings** tab. Click **Finish**.



6.7. Signaling Manipulation

The screen below shows the finished Signaling Manipulation script named ***Remote-Address*** created during the compliance test. This script was used to remove the "Remote-Address" header from outbound INVITE and 200 OK messages. This header is generated by the Avaya SBCE and should not be propagated to the service provider,

To add a Signaling Manipulation script, from the **Global Profiles** menu on the left panel, select **Signaling Manipulation**. Click **Add** to open the SigMa Editor screen, where the text of the script can be entered.

This script will be applied to the Sever Configuration profile corresponding to the service provider, later in **Section 6.8.2**.

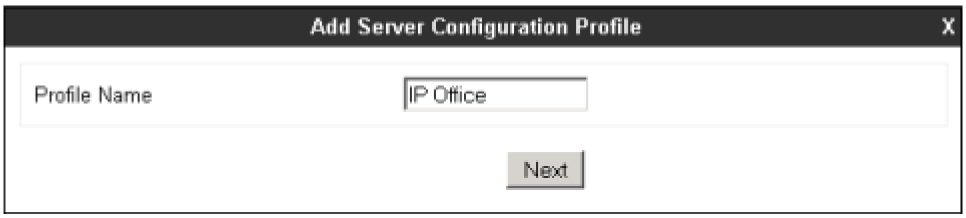
The details of the script used can be found in **Appendix A** of this document.

6.8. Server Configuration

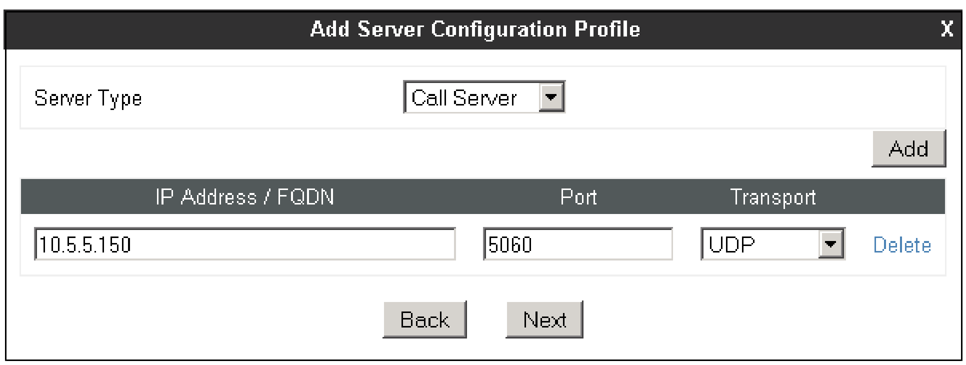
Server Profiles are created to define the parameters for the Avaya SBCE two peers, i.e., Avaya IP Office (Call Server) and the SIP Proxy at the service provider's network (Trunk Server).

6.8.1. Server Configuration Profile – Avaya IP Office

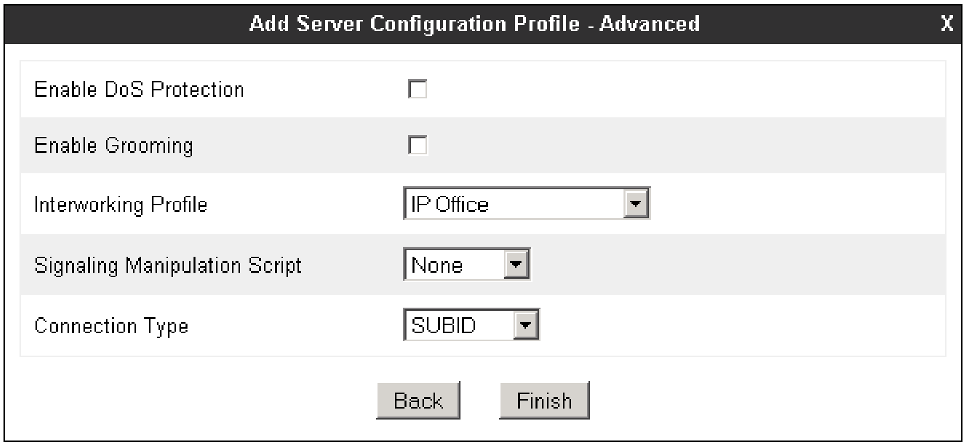
From the **Global Profiles** menu on the left-hand navigation pane, select **Server Configuration** and click the **Add** button (not shown) to add a new profile for the Call Server. Enter an appropriate **Profile Name** similar to the screen below. Click **Next**.



On the **Add Server Configuration Profile** Tab select ***Call Server*** from the drop down menu for the **Server Type**. On the **IP Addresses / FQDN** field**,** enter the IP address of the IP Office LAN1, as defined in **Section 5.2**. Enter ***5060*** under **Port** and select ***UDP*** for **Transport**. The transport protocol and port selected here must match the values used on the IP Office SIP line on **Section 5.7**. Click **Next**.

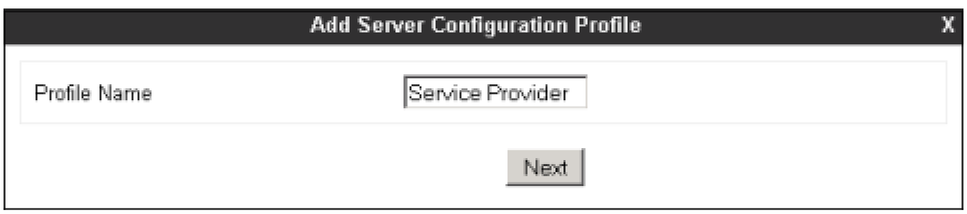


Click **Next** on the **Authentication** and **Heartbeat** tabs (not shown). On the **Advanced** tab, select ***IP Office*** from the **Interworking Profile** drop down menu. Click **Finish**.

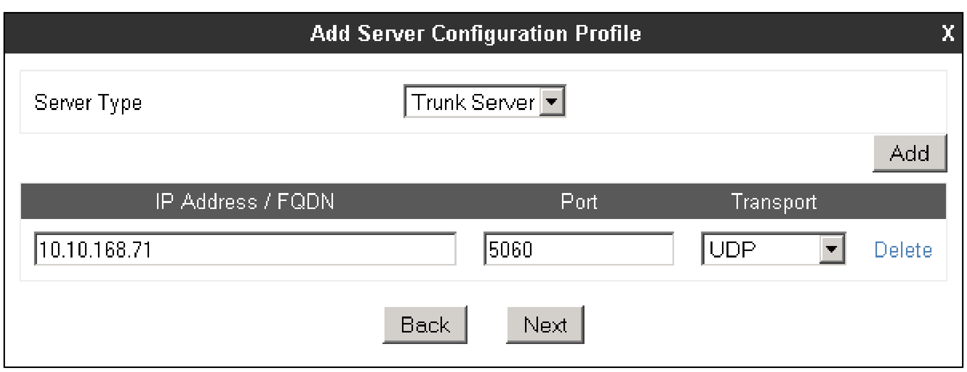


6.8.2. Server Configuration Profile – Service Provider

Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown). Enter an appropriate **Profile Name** similar to the screen below. Click **Next**.



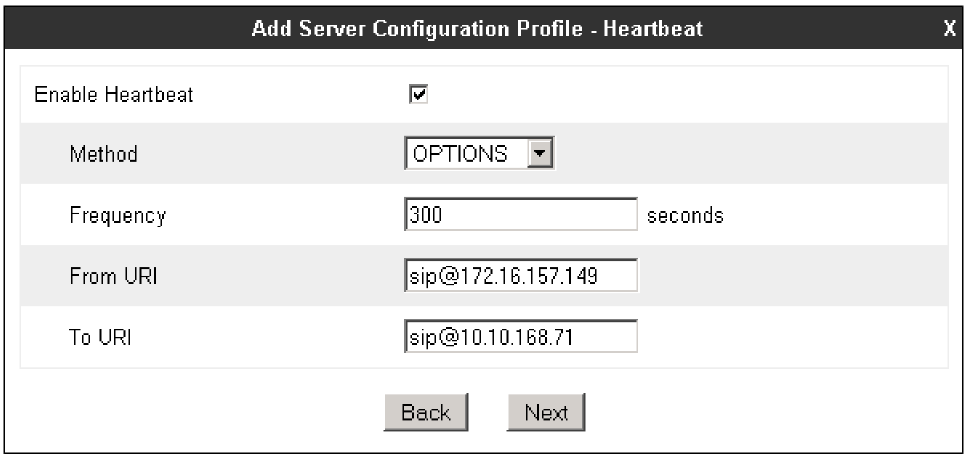
On the **Add Server Configuration Profile** Tab select ***Trunk Server*** from the drop down menu for the **Server Type**. On the **IP Addresses / FQDN** field, enter ***10.10.168.71***, the IP Address of the service provider SIP proxy server. Enter ***5060*** under **Port**, and select **UDP** for **Transport**, as required by Group of Gold Line.



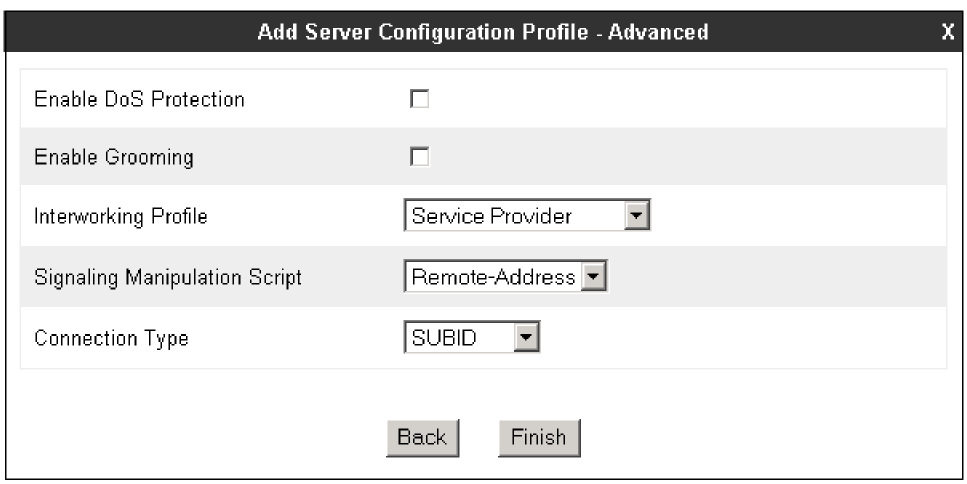
Click **Next** on the **Authentication** tab (not shown).

On the **Heartbeat** tab, OPTIONS can be configured to periodically check the integrity of the SIP trunk to the service provider. To do this, set the following:

- Check the **Enable Heartbeat** box.
- Under **Method**, select ***OPTIONS*** from the drop down menu.
- **Frequency:** Enter the amount of time (in seconds) between OPTIONS messages that will be sent from the enterprise to the service provider proxy server. ***300*** seconds was the value used during the compliance test.
- The **From URI** and **To URI** entries for the OPTIONS messages were built using the IP addresses of the public interface of the Avaya SBCE and the service provider proxy server respectively.
- Click **Next**.



On the **Advanced** tab, select ***Service Provider*** from the **Interworking Profile** drop down menu. Under **Signaling Manipulation Script**, select the script created in **Section 6.7**. Click **Finish**

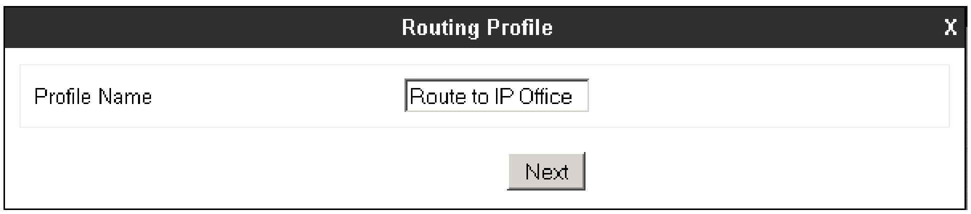


6.9. Routing

Routing profiles define a specific set of routing criteria that is used, in addition to other types of domain policies, to determine the path that the SIP traffic will follow as it flows through the Avaya SBCE interfaces. Two Routing Profiles were created in the test configuration, one for inbound calls, with the IP Office as the destination, and the second one for outbound calls, which are routed to the Group of Gold Line SIP trunk.

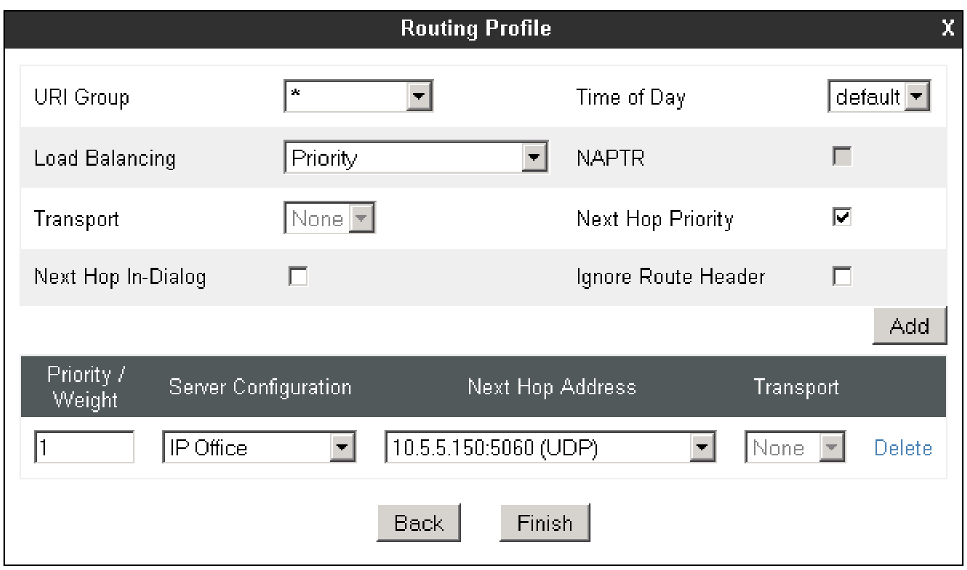
6.9.1. Routing Profile – Avaya IP Office

To create the inbound route, select the **Routing** tab from the **Global Profiles** menu on the left-hand side and select **Add** (not shown). Enter an appropriate **Profile Name** similar to the example below. Click **Next.**



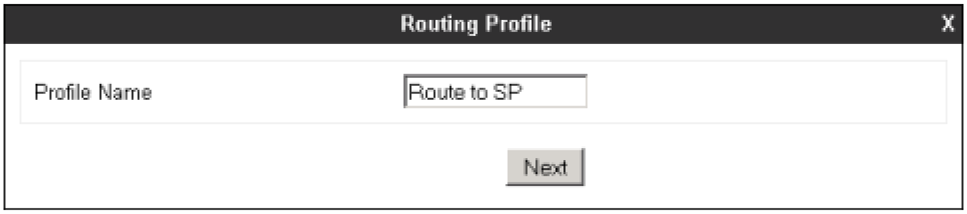
On the **Routing Profile** tab, click the **Add** button to enter the next-hop address.

Since only one next-hop is defined, enter ***1*** under **Priority/Weight**. Under **Server Configuration**, select the ***IP Office*** profile created in **Section 6.8.1**. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the IP Office Server Profile in **Section 6.8.1**. Defaults were used for all other parameters. Click **Finish**.



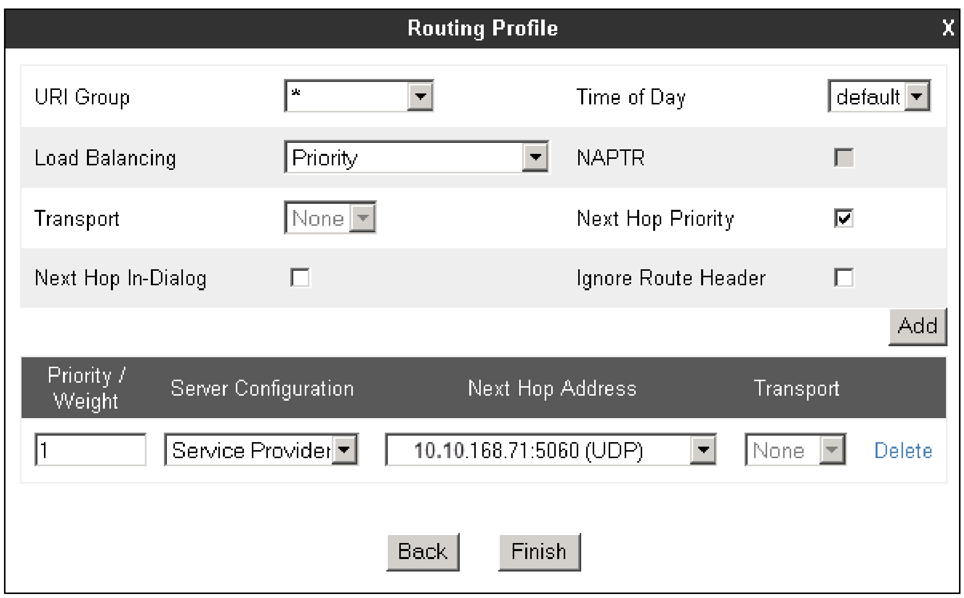
6.9.2. Routing Profile – Service Provider

Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route. Enter an appropriate **Profile Name** similar to the example below. Click **Next.**



On the **Routing Profile** tab, click the **Add** button to enter the next-hop address.

Since only one next-hop is defined to the service provider, enter ***1*** under **Priority/Weight**. Under **Server Configuration**, select the ***Service Provider*** profile created in **Section 6.8.2**. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the Server Profile corresponding to the Group of Gold Line SIP proxy server in **Section 6.8.2**. Defaults were used for all other parameters. Click **Finish**.



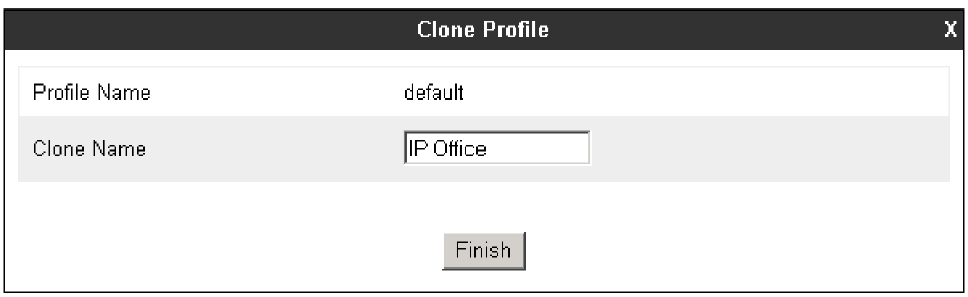
6.10. Topology Hiding

Topology Hiding is a security feature that allows the modification of several SIP headers, preventing private enterprise network information from being propagated to the untrusted public network.

Topology Hiding can also be used as an interoperability tool to adapt the host portion in the SIP headers to the IP addresses or domains expected on the service provider and the enterprise networks. For the compliance test, the Topology Hiding Profiles were created by cloning the default profile. Only the minimum configuration required to achieve interoperability on the SIP trunk was performed. Additional steps can be taken in this section to further mask the information that is sent from the enterprise to the public network.

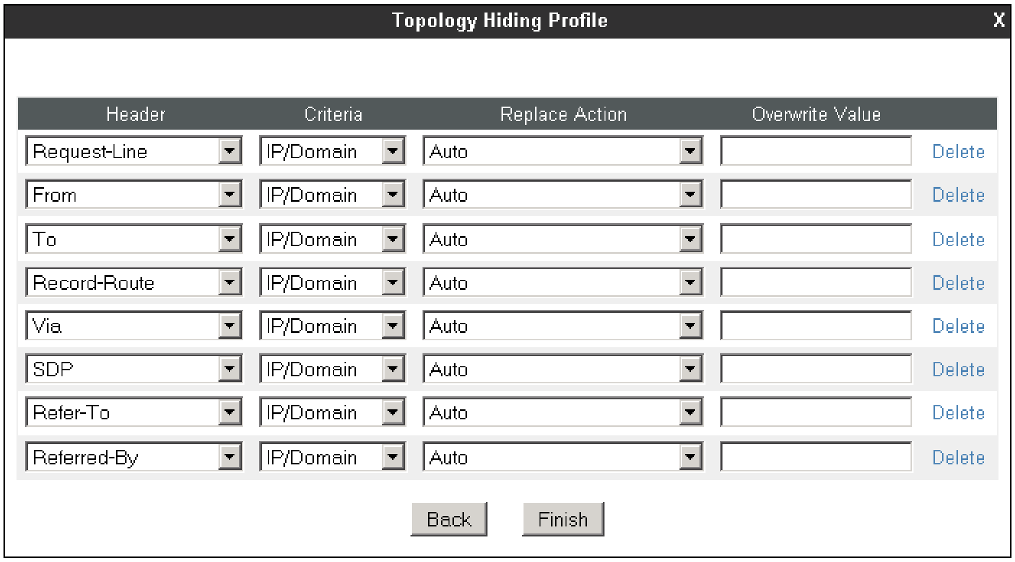
6.10.1. Topology Hiding Profile – Avaya IP Office

To add the Topology Hiding Profile in the enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side, select ***default*** from the list of pre-defined profiles and click the **Clone** button (not shown). Enter a **Clone Name** such as the one shown below. Click **Finish**.



On the newly cloned **IP Office** profile screen, click the **Edit** button (not shown).

During the compliance test, IP addresses instead of domains were used in all SIP messages between the IP Office and the Avaya SBCE. Note that since the default action of ***Auto*** implies the insertion of IP addresses in the host portion of these headers, it was not necessary to modify any of the headers sent to the enterprise. Default values were used for all fields. Click **Finish**.



6.10.2. Topology Hiding Profile – Service Provider

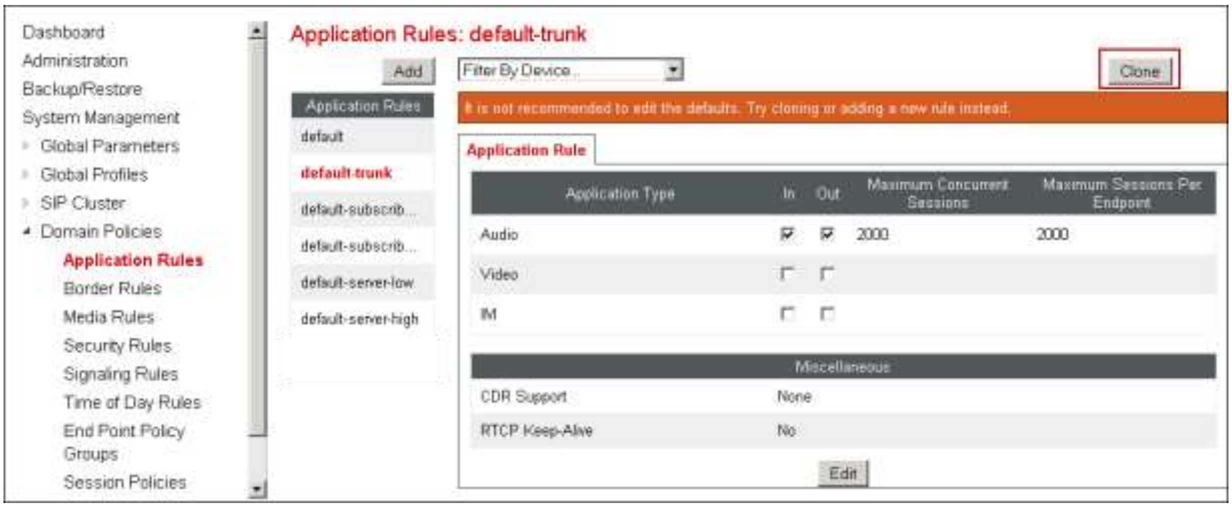
A Topology Hiding profile named ***Service Provider*** was similarly configured in the direction of the SIP trunk to the service provider. Since IP addresses instead of domains were used in all SIP messages between the Group of Gold Line SIP proxy server and the Avaya SBCE, the default action of ***Auto*** was also used in this profile. Note that even though both profiles used the same default settings, they were separately defined with the purpose of allowing possible future changes to be made to the profile in one of the directions, without affecting the settings in the other direction.

The screen below shows the **Service Provider** profile once the configuration was completed.

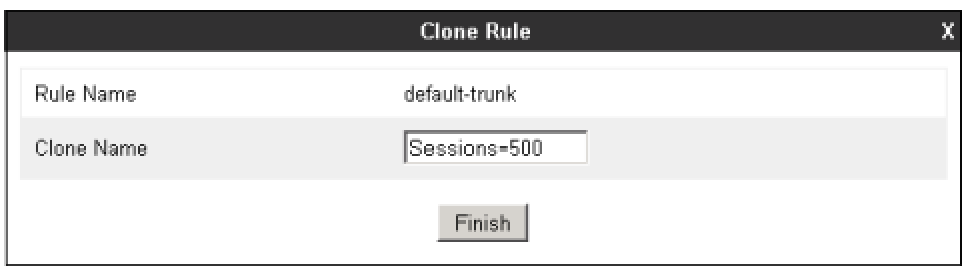
6.11. Application Rules

Application Rules define the types of SIP-based Unified Communications (UC) applications to be protected by the Avaya SBCE, as well as the maximum number of concurrent sessions allowed to be processed by the device. A single new Application Rule was created, by cloning the pre-defined **default-trunk** rule.

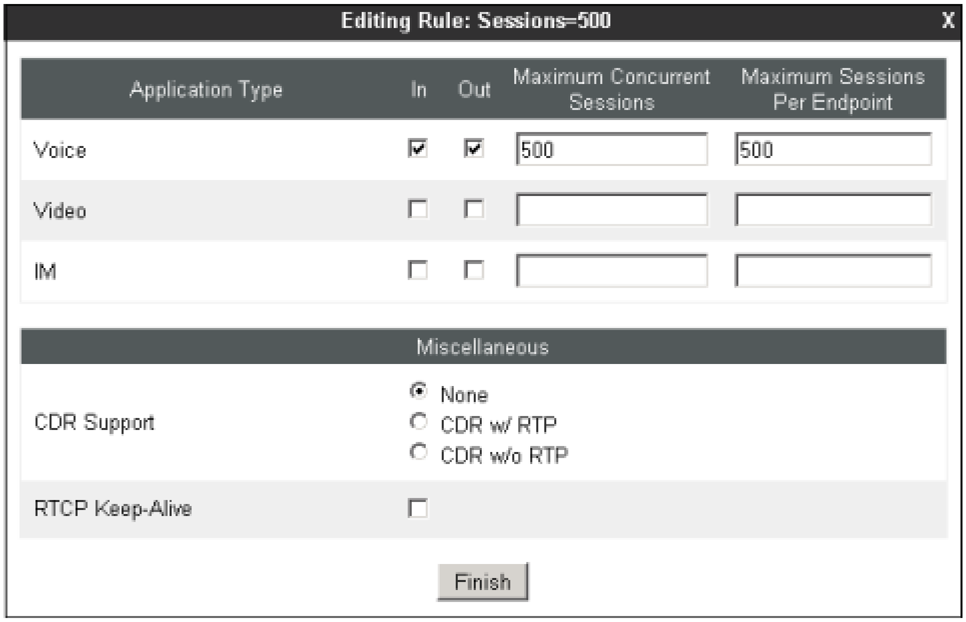
Select **Application Rules** under the **Domain Policies** menu on the left hand side, select the **default-trunk** Application Rule and click **Clone**.



Under **Clone Name** enter the new rule name. Click **Finish** to save.



On the Application Rules screen, select the newly created rule and click **Edit** (not shown). For SIP trunking, **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** should have the same value. In the example below, they were set to ***500***, which is the number of maximum simultaneous sessions supported on the Avaya SBCE Portwell CAD-0208 platform. Click **Finish**.



6.12. End Point Policy Groups

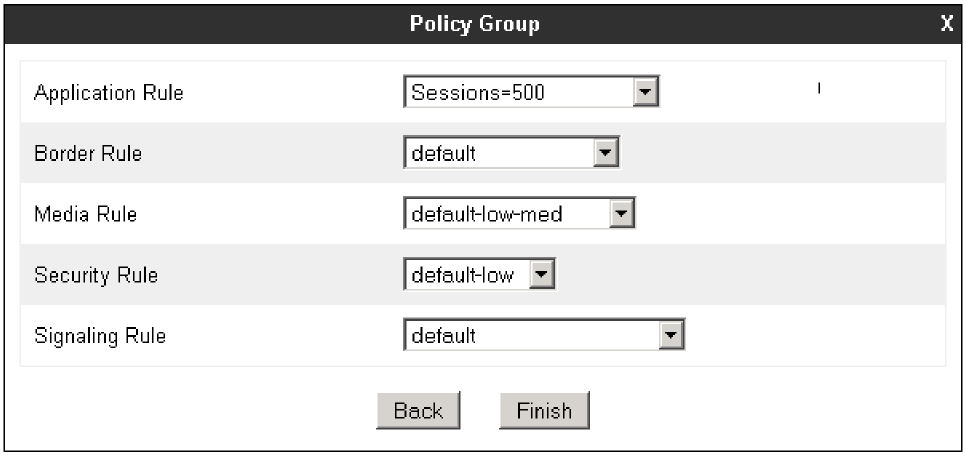
End Point Policy Groups associate the different sets of rules under Domain Policies (Media, Signaling, Security, etc) to be applied to specific SIP messages traversing through the Avaya SBCE. In the reference configuration, the End Point Policy Groups used default sets of rules already pre-defined in the configuration, with the exception of the new Application Rule defined in **Section 6.11**. Please note that changes should not be made to any of the defaults. If changes are needed, it is recommended to create a new rule by cloning one the defaults and then make the necessary changes to the new rule.

6.12.1. End Point Policy Group – Avaya IP Office

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

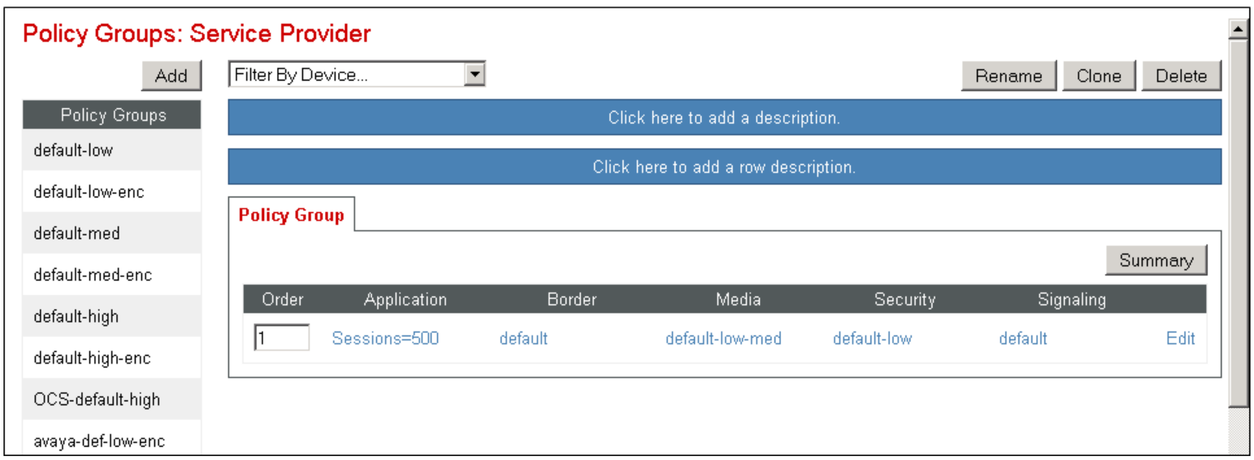
Enter an appropriate name in the **Group Name** field. Click **Next**.

In the Policy Group tab, defaults were used for all fields, with the exception of the **Application Rule**, where the ***Sessions=500*** rule was selected. Click **Finish**.



6.12.2. End Point Policy Group – Service Provider

A second End Point Policy Group was created for the service provider, repeating the steps described above. This is done with the purpose of allowing changes to be made to one of the groups in the future if needed, without affecting the settings in the other group. The screen below shows the ***Service Provider*** End Point Policy Group after the configuration was completed.

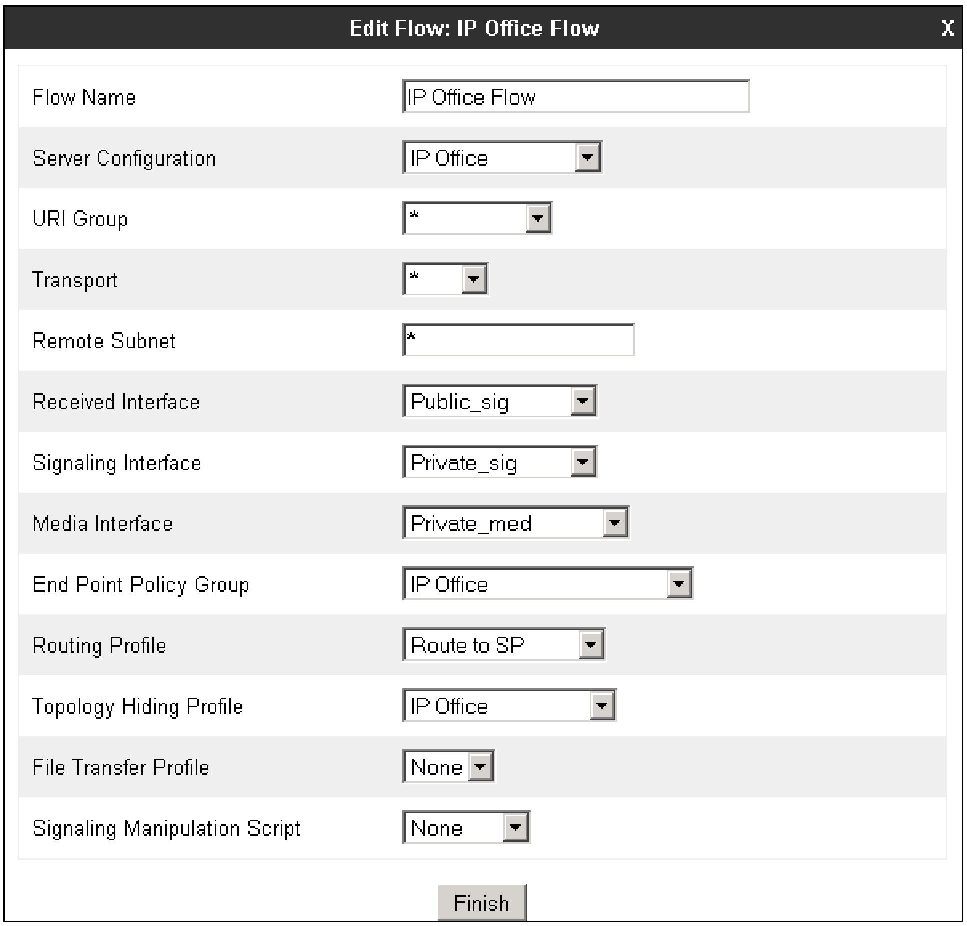


6.13. End Point Flows

End Point Flows determine the path to be followed by the packets traversing through the Avaya SBCE. They also combine the different sets of rules and profiles previously configured, to be applied to the SIP traffic traveling in each direction.

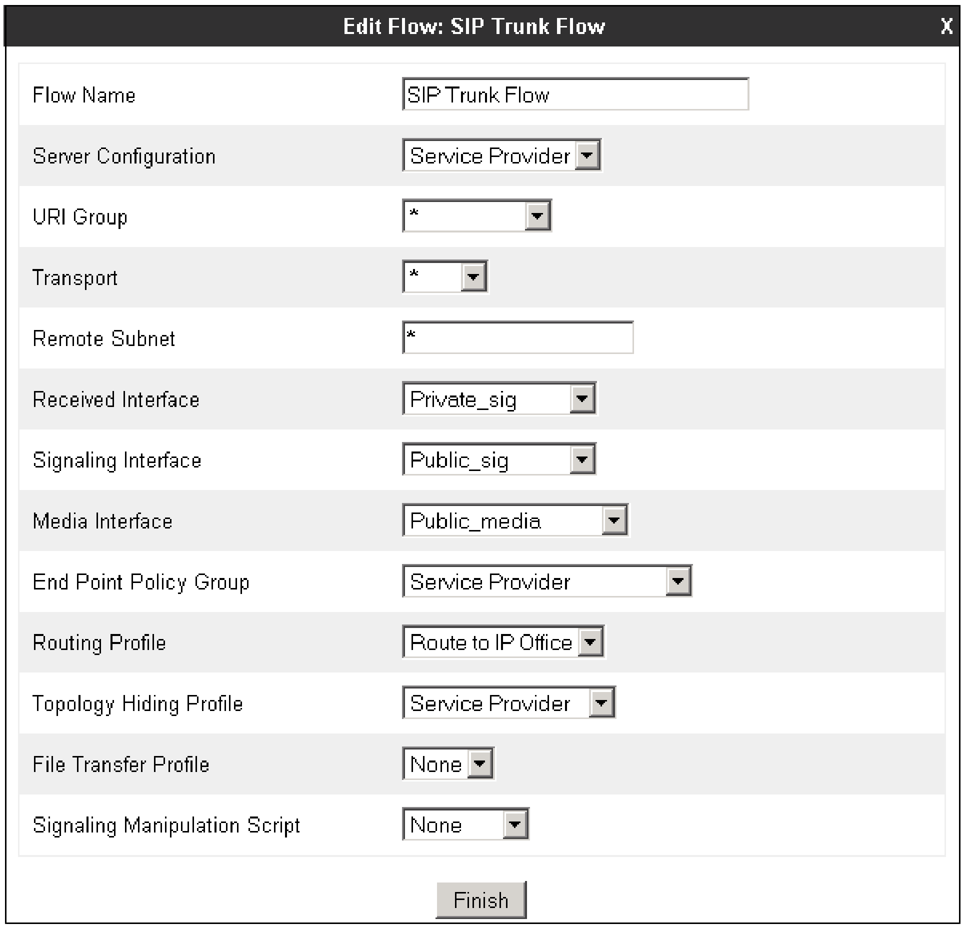
6.13.1. End Point Flow – Avaya IP Office

To create the call flow toward the enterprise, from the **Device Specific** menu, select **End Point Flows**, then select the **Server Flows** tab. Click **Add** (not shown). The screen below shows the flow named **IP Office Flow** created in the sample configuration. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for the Service Provider in **Section 6.9.2**, which is the reverse route of the flow. Click **Finish**.



6.13.2. End Point Flow – Service Provider

A second Server Flow with the name **SIP Trunk Flow** was similarly created in the network direction. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for the IP Office in **Section 6.9.1**, which is the reverse route of the flow. Also note that there is no selection under the **Signaling Manipulation Script** field. Since the script created in **Section 6.7** was already applied to the service provider's Server Configuration Profile in **Section 6.8.2**, it is not necessary to make a selection here. Click **Finish**.



7. Group of Gold Line SIP Trunking Configuration

Group of Gold Line is responsible for the configuration of the SIP Trunking service in its network. The customer will need to provide the IP address and port used to reach the Avaya SBCE at the enterprise. Group of Gold Line will provide the customer the necessary information to configure the SIP trunk connection from the enterprise site to the network, including:

- IP address and port of the Group of Gold Line SIP Proxy server.
- Supported codecs and order of preference.
- DID numbers.
- All other IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

This information is used to complete the configuration of the Avaya IP Office and the Avaya SBCE discussed in the previous sections.

8. Verification Steps

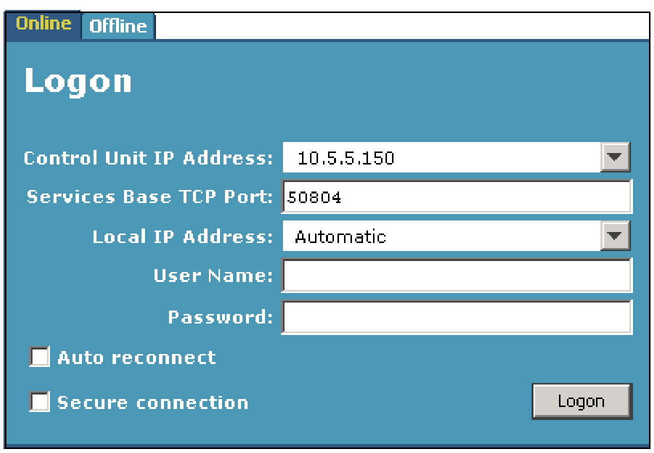
The following sections include steps that may be used to verify the configuration of the Avaya IP Office and the Avaya SBCE with the Group of Gold Line SIP Trunking service.

8.1. Avaya IP Office

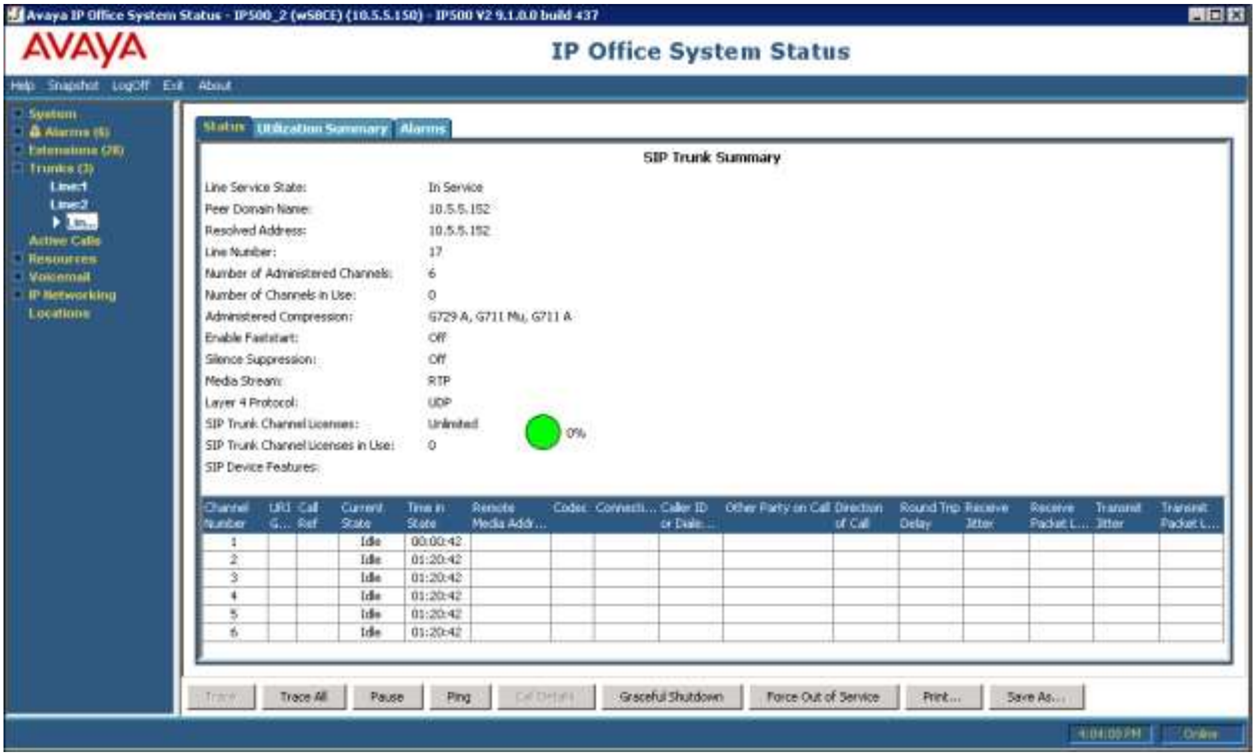
The Avaya IP Office System Status and Monitor applications are useful tools used for the verification and troubleshooting of the SIP connection to the service provider via the Avaya SBCE.

8.1.1. System Status

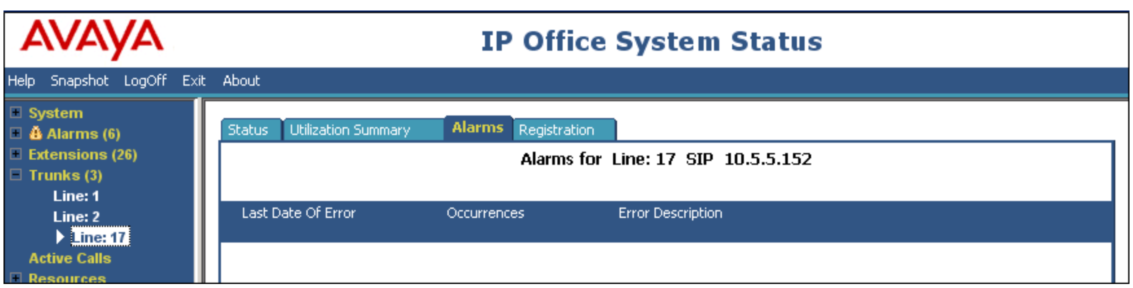
The Avaya IP Office System Status application can be used to verify the service state of the SIP line. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Under **Control Unit IP Address** select the IP address of the IP Office system under verification. Log in using the appropriate credentials



Select the SIP line of interest from the left pane (**Line 17** in the reference configuration). On the **Status** tab in the right pane, verify that the **Current State** is ***Idle*** for each channel (assuming no active calls at present time).

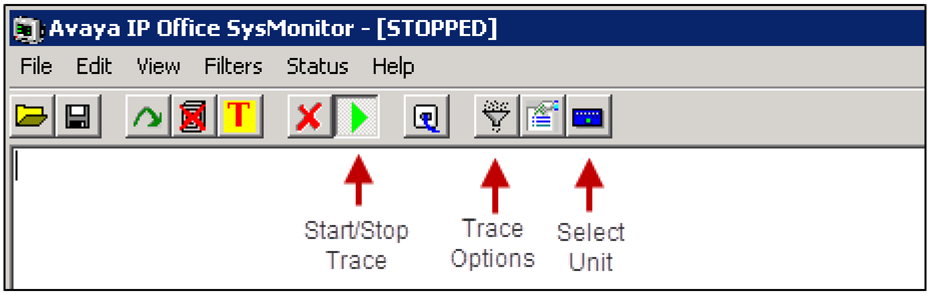


Select the **Alarms** tab and verify that no alarms are active on the SIP line.

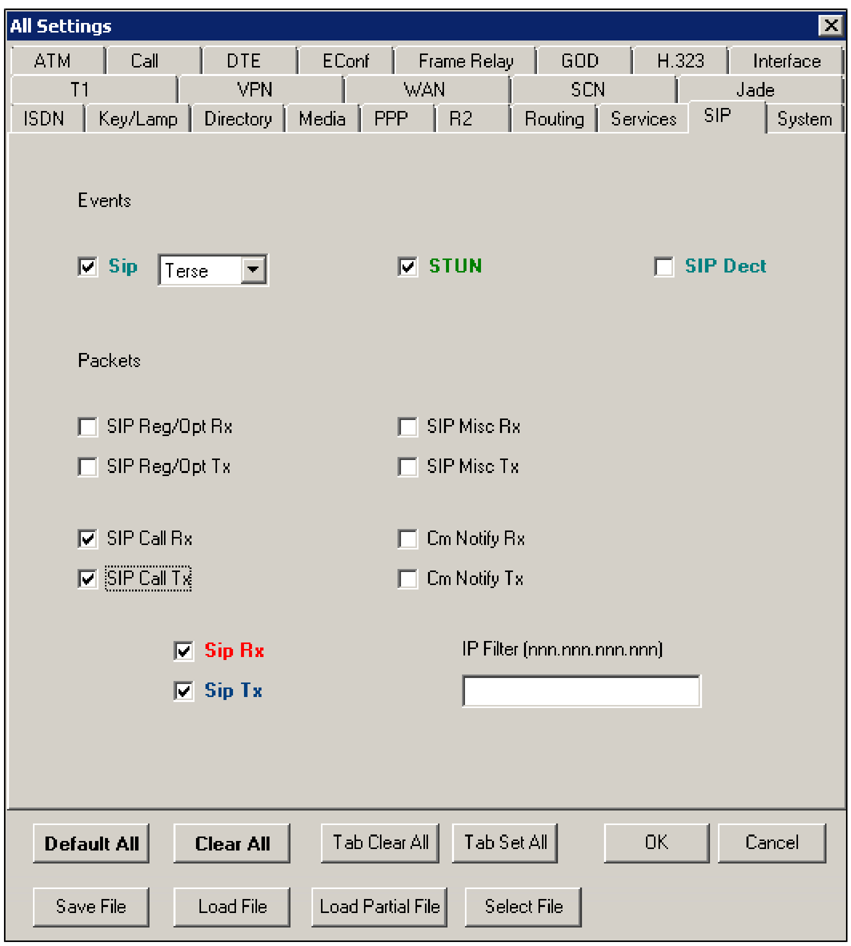


8.1.2. Monitor

The Avaya IP Office Monitor application can be used to monitor and troubleshoot signaling messaging on the SIP trunk. Launch the application from **Start → Programs → IP Office → Monitor** on the PC where Avaya IP Office Manager was installed. Click the **Select Unit** icon on the taskbar and Select the IP address of the IP Office system under verification.



Click the **Trace Options** icon on the taskbar and select the **SIP** tab to modify the threshold used for capturing events, types of packets to be captured, filters, etc. Additionally, the color used to represent the packets in the trace can be customized by right clicking on the type of packet and selecting the desired color.

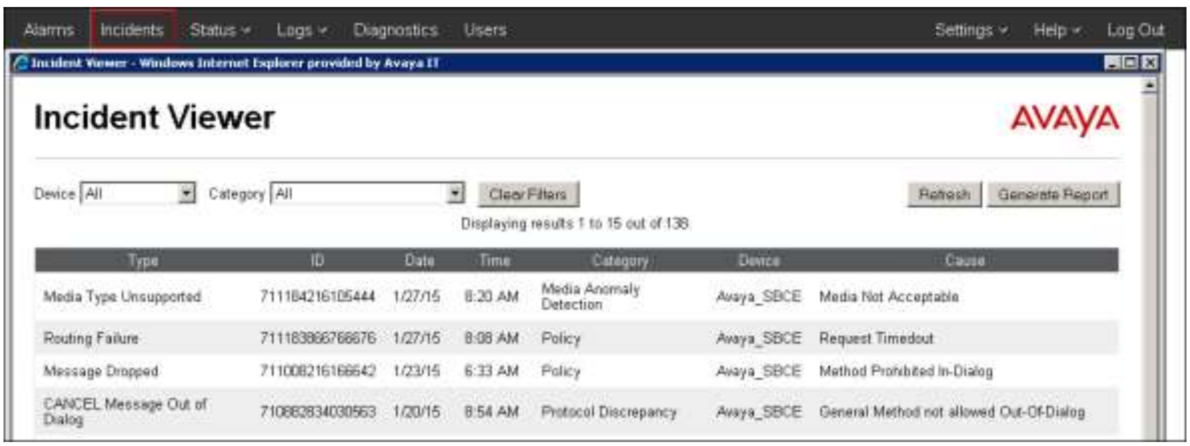


8.2. Avaya Session Border Controller for Enterprise

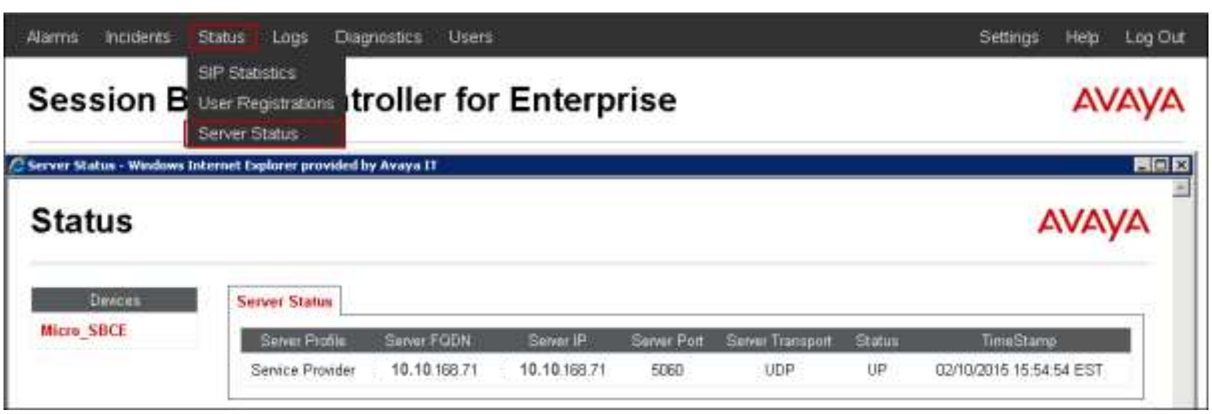
There are several links and menus located on the taskbar at the top of the screen of the web interface that can provide useful diagnostic or troubleshooting information.

Alarms: Provides information about the health of the SBC.

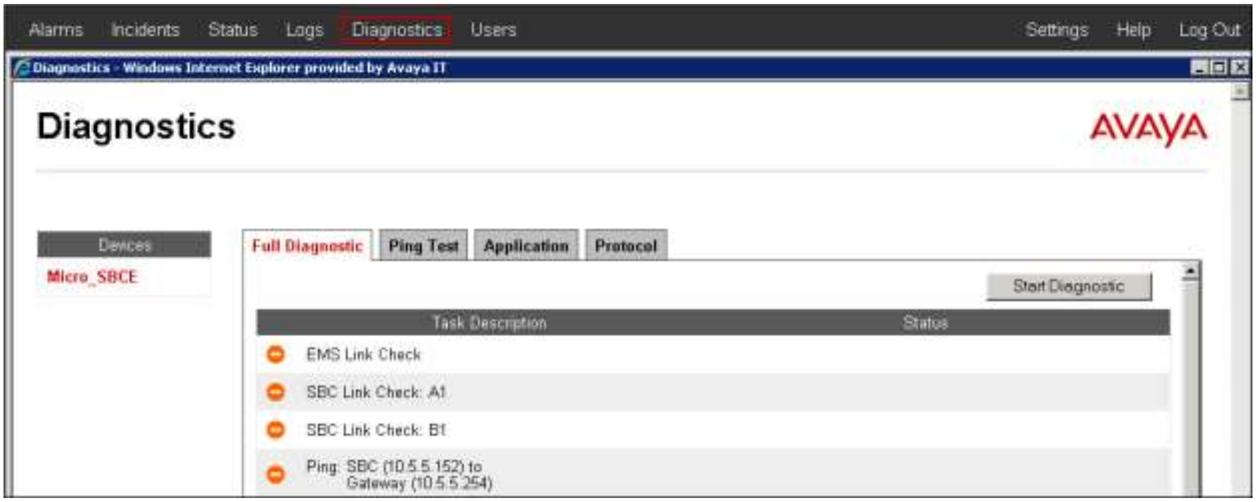
Incidents : Provides detailed reports of anomalies, errors, policies violations, etc.



Status: Statistical and current status information. The **Server Status** screen below provides information about the condition of the connection to the Service Provider. This requires Heartbeat to be enabled on the Server Configuration profile, as configured in **Section 7.8.2**.

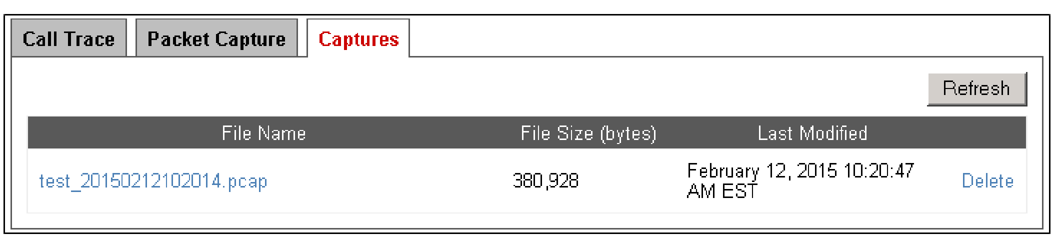


Diagnostics: This screen provides a variety of tools to test and troubleshoot the Avaya SBCE network connectivity.



Additionally, the Avaya SBCE contains an internal packet capture tool that allows the capture of packets on any of its interfaces, saving them as *pcap* files. Navigate to **Device Specific Settings → Troubleshooting → Trace**. Select the **Packet Capture** tab, set the desired configuration for the trace and click **Start Capture**.

Once the capture is stopped, click the Captures tab and select the proper pcap file. Note that the date and time is appended to the filename specified previously. The file can now be saved to the local PC, where it can be opened with an application such as Wireshark.



9. Conclusion

These Application Notes describe the procedures required to configure SIP trunk connectivity between Avaya IP Office Release 9.1 and Avaya Session Border Controller Release 6.3 with the Group of Gold Line SIP Trunking, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with the exception of the observations/limitations described in **Section 2.2**.

10. Additional References

[1] *IP Office Platform 9.1, Deploying Avaya IP Office Platform IP500V2*, Document 15-601042, January 2015
https://downloads.avaya.com/css/P8/documents/101005082

[2] *Administering Avaya IP Office Platform with Manager*, *Release 9.1.0*, January 2015
https://downloads.avaya.com/css/P8/documents/101005673

[3] *Administering Avaya Communicator on IP Office, Release 9.1*, December 2014
https://downloads.avaya.com/css/P8/documents/101005862

[4] *IP Office Platform 9.1, Using Avaya IP Office Platform System Status*, Document 15-601758, October 2014
https://downloads.avaya.com/css/P8/documents/101005061

[5] *Avaya IP Office Knowledgebase*
http://marketingtools.avaya.com/knowledgebase

[6] *Deploying Avaya Session Border Controller for Enterprise*, *Release 6.3*, October 2014
https://downloads.avaya.com/css/P8/documents/101001303

[7] *Administering Avaya Session Border Controller for Enterprise*, *Release 6.3*, October 2014
https://downloads.avaya.com/css/P8/documents/101001325

Product documentation for Avaya products may be found at http://support.avaya.com.
Product documentation for the Group of Gold Line SIP Trunking service is available from Group of Gold Line.

11. Appendix A: SigMa Script

The following is the Signaling Manipulation script used in **Section 6.7** of the Avaya SBCE configuration:

```
//Remove Remote-Address header in outbound INVITE and 200 OK

within session "ALL"

{

   act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"

   {
       remove(%HEADERS["Remote-Address"][1]);
   }

}
```

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.