



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 Alestra SIP Trunk Service on the Sonus Platform – 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 the Alestra SIP Trunk Service on the Sonus platform.

The SIP Trunking service offered by Alestra 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 Alestra SIP Trunk Service, operating on the Sonus platform, 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 SIP trunking service provided by Alestra and referenced within these Application Notes is designed for business customers in Mexico. 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.

The Avaya enterprise solution can be configured to authenticate with the SIP service provider using either SIP Trunk Registration or Static IP Authentication. These Application Notes cover the configuration of the Avaya SBCE using Static IP Authentication with service provider Alestra

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 Alestra SIP Trunk 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.

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.711A and G.71MU.
- Fax.
- 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.

Items not supported or not tested included the following:

- Network Call Redirection using the REFER method is not supported by Alestra and it was not tested.
- Operator services such as dialing 0, 0 + 10 digits and Local Directory assistance are not supported by Alestra.
- Inbound toll-free and emergency calls are supported but were not tested as part of the compliance test.

2.2. Test Results

Interoperability testing of the Alestra SIP Trunk Service was completed with successful results for all test cases with the observations and limitations described below:

- **Caller ID on outbound calls:** On calls originating from IP Office extensions to PSTN telephones, the caller ID number shown on the PSTN endpoint was always the main DID number assigned by Alestra to the SIP trunk, not the specific DID assigned to that extension. This includes calls to “twinned” mobile phones, and calls that were forwarded or transferred back on the SIP trunk to the PSTN, where the number displayed on the PSTN endpoint was the main DID number on the trunk, not the originator’s caller ID. This may be a requirement of the Alestra service for outbound calls on the SIP trunk; it is listed here simply as an observation.
- **Caller ID on incoming calls:** On incoming calls made from the test lab in the U.S., the caller IDs displayed on the enterprise extensions was “Unavailable/Restricted”. This seems to be a PSTN restriction for international calls to Mexico and not limited just to Alestra. Inbound calls originating from a test softphone in Monterrey, Mexico displayed the correct caller ID.
- **Outbound Calling Party Number (CPN) Block:** When an IP Office user activated “Withhold Number” on an outbound call for the purpose of privacy, IP Office sent “anonymous” in the “From” header and “Privacy:id”, but on the receiving end at the PSTN the display still showed the main number assigned by Alestra to the SIP trunk. This feature may not be supported on the PSTN in Mexico.
- **Outbound call from an enterprise extension to a busy PSTN number:** Alestra did not send a “486 Busy Here” message on an outbound call to a PSTN number that was busy, as it was expected on this condition. There was no direct impact to the user, who heard busy tone.
- **Codec negotiation:** On long distance outbound calls, national or international, the codec is initially negotiated at G.729A, the codec preferred by Alestra on the SIP trunk, but then is updated and changed to G711A by Alestra. On outbound calls to local PSTN numbers, and on all incoming calls, the codec selected remains at G.729A.
- **Fax support:** Inbound and outbound fax calls using the T.38 protocol failed during the test. Alestra responded with “488 Not Acceptable Here” to the T.38 re-invite sent from the IP Office. Fax using G.711 pass-through was additionally tested. While outbound calls in this mode were successful, inbound calls failed. Fax should not be used with this solution
- **Incoming Call, SIP Trunk Signaling Failure:** On incoming calls when the SIP trunk is on a forced “Out of Service” condition, there was no feedback to the PSTN caller, who just hears silence.
- **Avaya SBCE Remote-Address header** – During the compliance test, a Signaling Rule was used in the Avaya SBCE to remove the “Remote-Address” header, generated 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 on the Alestra SIP Trunk service offer, visit <http://www.alestra.com.mx/>

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Alestra SIP Trunk Service through a public Internet WAN connection.

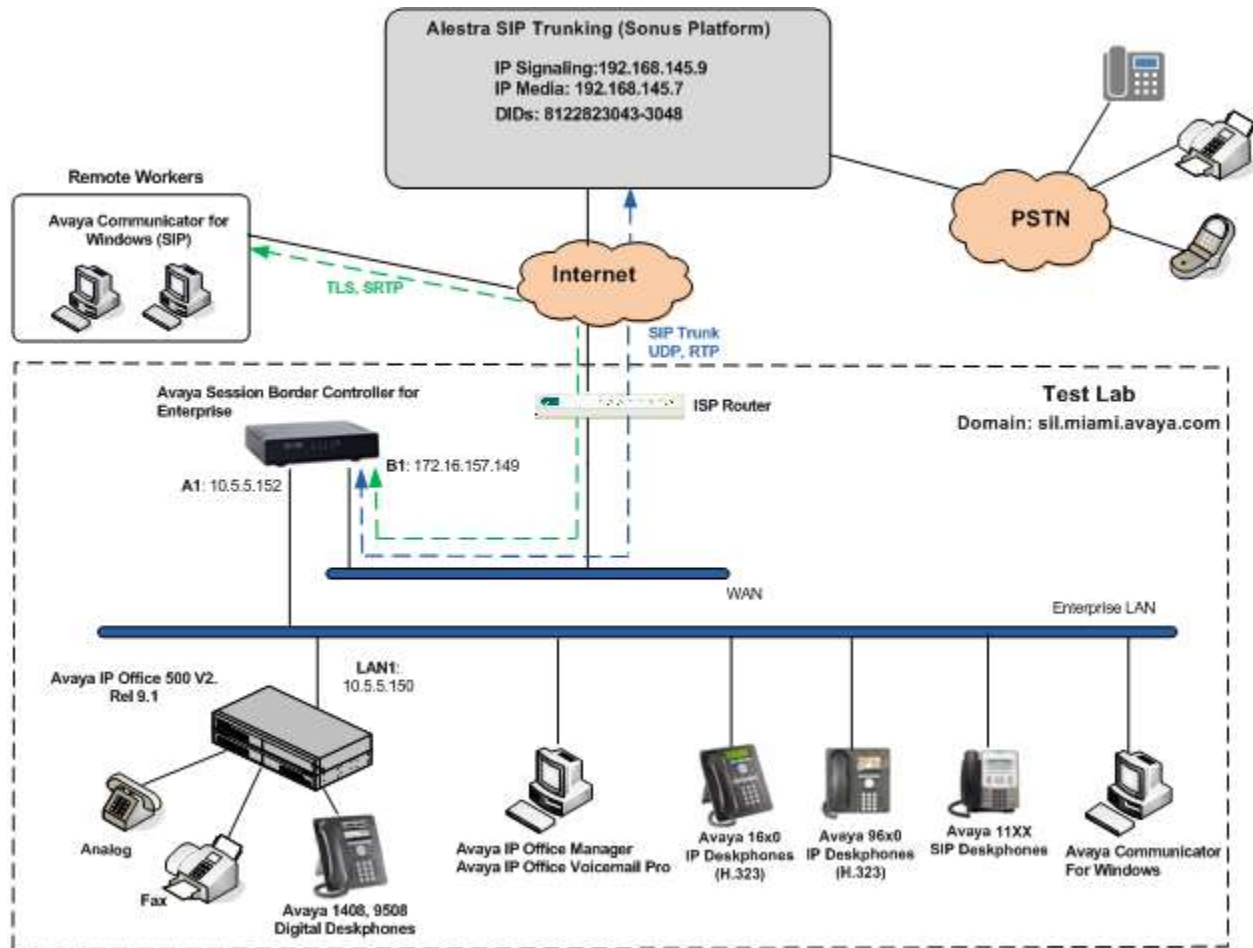


Figure 1: Test Configuration

Note that for security purposes, all public IP addresses of the network elements 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 Deskphones (with H.323 firmware), Avaya 1140E IP Deskphones (with SIP firmware), Avaya 1408 and 9508D Digital Deskphones, 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.

The Avaya SBCE is located at the edge of the enterprise. It 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, the protocols used between the remote workers and the outside interface of the Avaya SBCE were 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

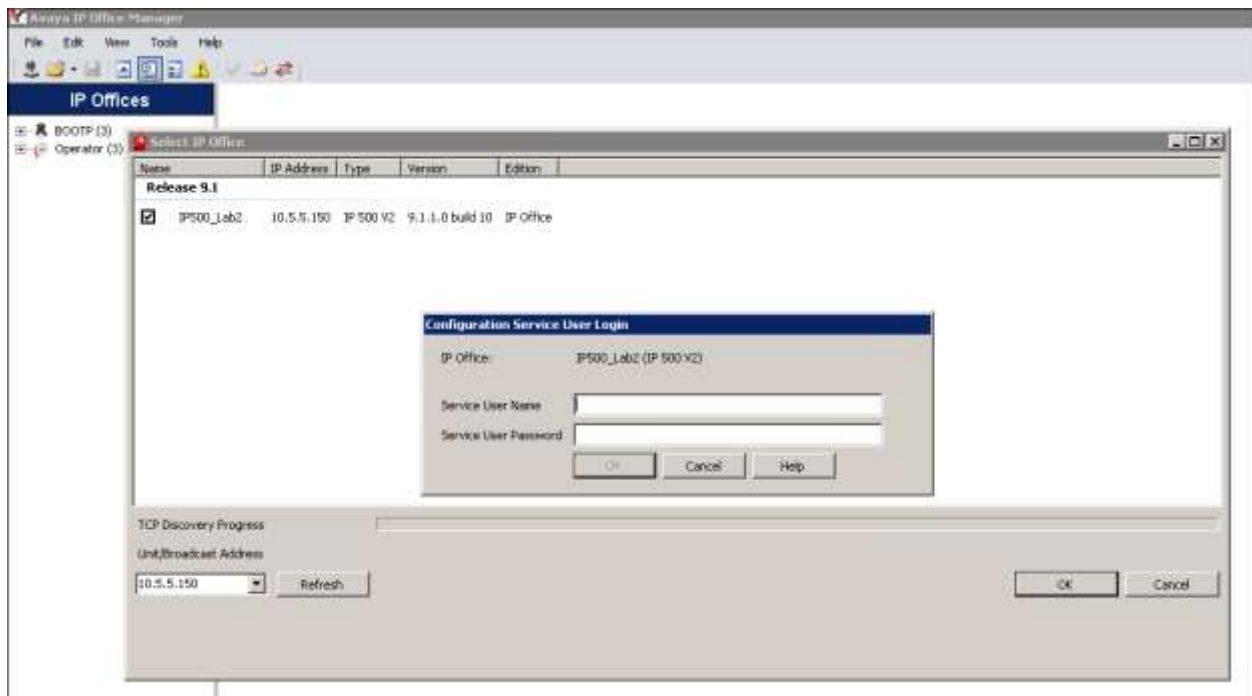
The following equipment and software were used for the sample configuration provided:

Component	Version
Avaya	
Avaya IP Office 500v2	9.1.100.10
Avaya IP Office Digital Expansion Module DCPx16	9.1.100.10
Avaya IP Office Manager	9.1.1.0.Build 10
Avaya IP Office Voicemail Pro	9.1.100.3
Avaya Session Border Controller for Enterprise	6.3. SP2 (6.3.2-08-5478)
Avaya 1608 IP Deskphone (H.323)	1.3.5
Avaya 9640 IP Deskphone (H.323)	Avaya one-X Deskphone Edition S3.230A
Avaya 1140E IP Deskphone (SIP)	04.04.18.00
Avaya Digital Deskphone 1408	40.0
Avaya Digital Deskphone 9508	0.55
Avaya Communicator for Windows	2.0.3.30
Alestra	
Sonus Softswitch	V08.04.09R000
Acme Packet SBC	V6.2
Lucent 5ESS	V16.1

Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition without T.38 Fax Service. (T.38 fax is not supported on IP Office Server Edition). Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

5. Configure IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the Alestra SIP Trunk 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_Lab2** was used as the system name. Under the system name on the Navigation pane, 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 Avaya SBCE and the service provider.

The screenshot shows the IP Office software interface. On the left, the navigation pane lists various system components, with 'License' selected under the 'IP500_Lab2' system. The main pane displays the 'License' configuration page for a 'Remote Server'. It shows the 'License Mode' as 'License Normal', 'Licensed Version' as '9.1', 'Serial Number (A01)' as '33287', 'PLDS Host ID' as '1113287', and 'PLDS File Status' as 'Not Present / Invalid'.

Feature	License Key	Instances	Status	Expires Date	Source
IP500 Voice Networking Channels	ynBHHuBRtV6c0mgs0IM...	255	Valid	Never	ACE Nodal
IP500 Upgrade Standard to Professio...	3ytkoG2MPCK1Un6ARe5Y...	255	Obsolete	Never	ACE Nodal
IP500 Voice Networking Channels	6GqDnSLurpusticrCJuK2...	4	Valid	Never	ACE Nodal
VCM Channel Migration	z4HuoqvV5vnhIzpqgBKhw...	255	Valid	Never	ACE Nodal
SIP Trunk Channels	uanDk0mVAOpf9p7HbUc...	255	Valid	Never	ACE Nodal
WAN IP Extensions	5t4OU2F56k2vzPfhvVw...	255	Obsolete	Never	ACE Nodal
IP500 Universal PRI (Additional chan...	nsjWA2g5GBywARt_VtEoM...	255	Valid	Never	ACE Nodal
RAS LRIQ Support (Rapid Response)	o1C2pPrYADpncGg3K4Dnc5...	255	Valid	Never	ACE Nodal
IP Office Dealer Support - Standard E...	Prc2D7HgwKebFHmgQJm5...	255	Valid	Never	ACE Nodal
IP Office Dealer Support - Profession...	PuMSPmLNV_5na92GvMS...	255	Valid	Never	ACE Nodal
IP Office Distributor Support - Stand...	6t0t0R5v6ePmgntK7WALL...	255	Valid	Never	ACE Nodal
IP Office Distributor Support - Profes...	hG9N6SgQM_Xl0sY3o_r9d...	255	Valid	Never	ACE Nodal
UMS Web Services	3Uu53F6u09NbgMDc16E...	255	Valid	Never	ACE Nodal
Customer Service Agent	FAHEgB4gualDp5qyK_5Y...	255	Obsolete	Never	ACE Nodal
Third Party API	Ymhbtu0cAXgfbuYn6P...	255	Valid	Never	ACE Nodal
Software Upgrade 255	gHCSvd8ds1Z5udk6oedR...	1	Valid	Never	ACE Nodal
onv2 Portal for IP Office	1uwh7ndfncK4M8M_L0a6k...	255	Valid	Never	ACE Nodal

5.2. LAN Settings

In the sample configuration, the LAN1 port was used to connect the IP Office to the enterprise network. The LAN2 (WAN) port was not used. 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.

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane, showing a tree structure with 'IP500_Lab2' selected, and 'System (1)' expanded. The main pane on the right is titled 'IP500_Lab2' and contains several tabs: 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', 'SMTP', and 'SMDR'. The 'LAN1' tab is active, and within it, the 'LAN Settings' sub-tab is selected. The configuration fields are as follows:

- IP Address:** 10 . 5 . 5 . 150
- IP Mask:** 255 . 255 . 255 . 0
- Primary Trans. IP Address:** 0 . 0 . 0 . 0
- RIP Mode:** None (dropdown menu)
- Enable NAT:** ☐
- Number Of DHCP IP Addresses:** 200 (spinner)
- DHCP Mode:** ☐ Server ☐ Client ☐ Dialin ☒ Disabled
- Advanced:** A button to expand further settings.

On the **VoIP** tab in the Details pane, the **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Deskphones with the H.323 protocol, such as the Avaya 1600 and 9600 Series IP Deskphones 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 Deskphones and the Avaya Communicator 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.

LAN Settings | VoIP | Network Topology

☒ H323 Gatekeeper Enable

☐ Auto-create Extn ☐ Auto-create User ☐ H323 Remote Extn Enable

Remote Call Signalling Port

☒ SIP Trunks Enable

☒ SIP Registrar Enable

☐ Auto-create Extn/User ☐ SIP Remote Extn Enable

Domain Name

Layer 4 Protocol

<input checked="" type="checkbox"/> UDP	UDP Port	<input type="text" value="5060"/>	Remote UDP Port	<input type="text" value="5060"/>
<input checked="" type="checkbox"/> TCP	TCP Port	<input type="text" value="5060"/>	Remote TCP Port	<input type="text" value="5060"/>
<input checked="" type="checkbox"/> TLS	TLS Port	<input type="text" value="5061"/>	Remote TLS Port	<input type="text" value="5061"/>

Challenge Expiry Time (secs)

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.

In the **RTP Keepalives** section, set the **Scope** field to **RTP**. Set the **Periodic timeout** to **30** and **Initial keepalives** to **Enabled**. This will cause the IP Office to send RTP keepalive packets at the beginning of the calls and periodically thereafter, to avoid problems of media deadlock resulting in no audio situations that can occur with certain types of forwarded calls that are routed from the IP Office back to the network, over the same SIP trunk.

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.

The screenshot displays the configuration interface for an Avaya IP Office, specifically the 'VoIP' tab. The 'RTP' section is expanded, showing the following settings:

- Port Number Range:** Minimum is 49152, Maximum is 53246.
- Port Number Range (NAT):** Minimum is 49152, Maximum is 53246.
- Enable RTCP Monitoring on Port 5005:** Checked.
- RTCP collector IP address for phones:** 0.0.0.0.
- Keepalives:**
 - Scope:** RTP
 - Periodic timeout:** 30
 - Initial keepalives:** Enabled

The 'DiffServ Settings' section is also visible, showing the following values:

Field	Value
DSCP (Hex)	B8
Video DSCP (Hex)	B8
DSCP Mask (Hex)	FC
SIG DSCP (Hex)	88
DSCP	46
Video DSCP	46
DSCP Mask	63
SIG DSCP	34

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.

The screenshot shows the 'Network Topology' tab in the 'Details' pane. The 'Network Topology Discovery' section contains the following fields and controls:

- STUN Server Address:** 69.90.168.13
- STUN Port:** 3478
- Firewall/NAT Type:** Open Internet (selected from a dropdown menu)
- Binding Refresh Time (seconds):** 0
- Public IP Address:** 0 . 0 . 0 . 0
- Public Port:**
 - UDP: 5060
 - TCP: 0
 - TLS: 0
- Run STUN on startup:** ☐
- Buttons:** Run STUN, Cancel

5.3. 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 transfers to the PSTN via the SIP trunk to the service provider.

The screenshot displays the 'IP500_Lab2' configuration window, specifically the 'Telephony' tab. The interface is divided into several sections for configuring telephony parameters.

Analogue Extensions:

- Default Outside Call Sequence: Normal
- Default Inside Call Sequence: Ring Type 1
- Default Ring Back Sequence: Ring Type 2
- Restrict Analogue Extension Ringer Voltage: ☐

Companding Law:

- Switch:** ☒ U-Law, ☐ A-Law
- Line:** ☒ U-Law Line, ☐ A-Law Line

Time and Delay Settings:

- Dial Delay Time (secs): 4
- Dial Delay Count: 0
- Default No Answer Time (secs): 15
- Hold Timeout (secs): 0
- Park Timeout (secs): 300
- Ring Delay (secs): 5
- Call Priority Promotion Time (secs): Disabled

Other Settings:

- Default Currency: USD
- Default Name Priority: Favor Trunk
- Media Connection Preservation: Disabled
- Phone Failback: Manual
- Login Code Complexity: ☐ Enforcement

Advanced Features:

- ☐ DSS Status
- ☒ Auto Hold
- ☒ Dial By Name
- ☒ Show Account Code
- ☐ Inhibit Off-Switch Forward/Transfer
- ☐ Restrict Network Interconnect
- ☐ Include location specific information
- ☐ Drop External Only Impromptu Conference
- ☐ Visually Differentiate External Call
- ☐ Unsupervised Analog Trunk Disconnect Handling
- ☒ High Quality Conferencing
- ☒ Digital/Analogue Auto Create User
- ☐ Directory Overrides Barring

5.4. 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.

The screenshot shows the 'IP500_Lab2' window with the 'System' tab selected. The 'RFC2833 Default Payload' field is set to '101'. The 'Available Codecs' list on the left contains five items, all checked: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ. The 'Default Codec Selection' area has an 'Unused' list containing 'G.722 64K' and a 'Selected' list containing 'G.711 ULAW 64K', 'G.711 ALAW 64K', 'G.729(a) 8K CS-ACELP', and 'G.723.1 6K3 MP-MLQ'. Between the lists are five buttons: '>>', an up arrow, '<<', a down arrow, and '>>'.

5.5. 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.

0.0.0.0	
IP Address	0 . 0 . 0 . 0
IP Mask	0 . 0 . 0 . 0
Gateway IP Address	10 . 5 . 5 . 254
Destination	LAN1
Metric	0
<input type="checkbox"/> Proxy ARP	

5.6. 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 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 5.6.1** and **Section 5.6.2** 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 5.6.3 – 5.6.7**.

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 5.6.3 – 5.6.7**.

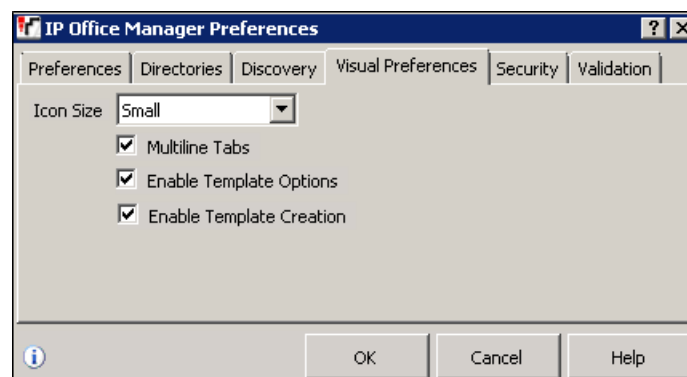
5.6.1. Importing a SIP Line Template

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.

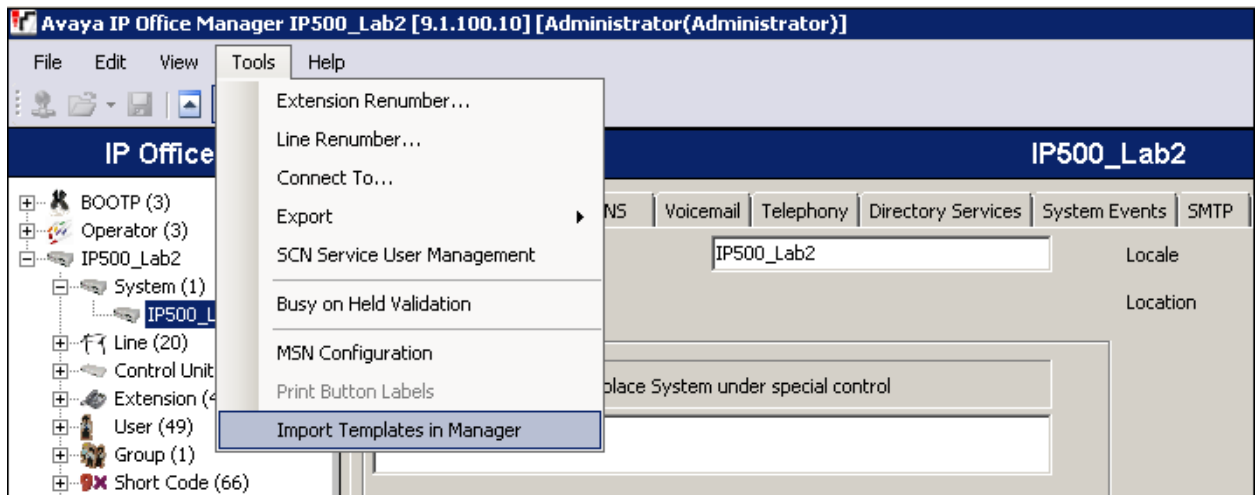
1. Copy a previously created template file to a location (e.g., *\Temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_<user supplied text>_SIPTrunk.xml**, where the *<user supplied text>* portion is entered during template file 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 5.6.2** to create a new SIP Line.

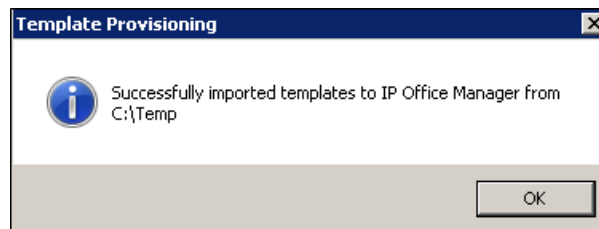
2. Verify that Template Options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Check the box next to **Enable Template Options**. Click **OK**.



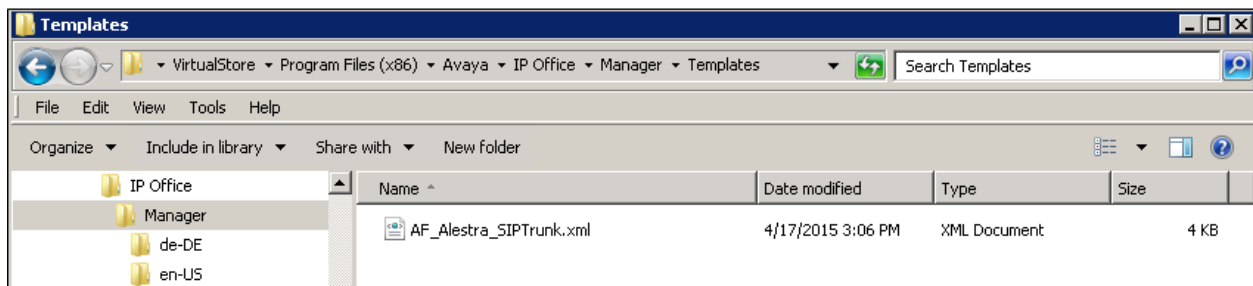
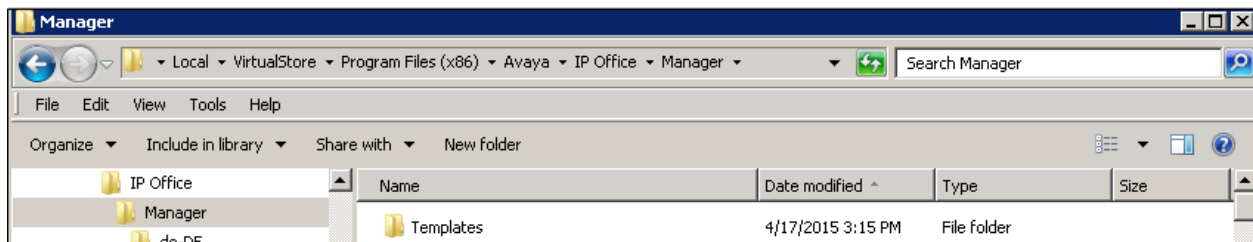
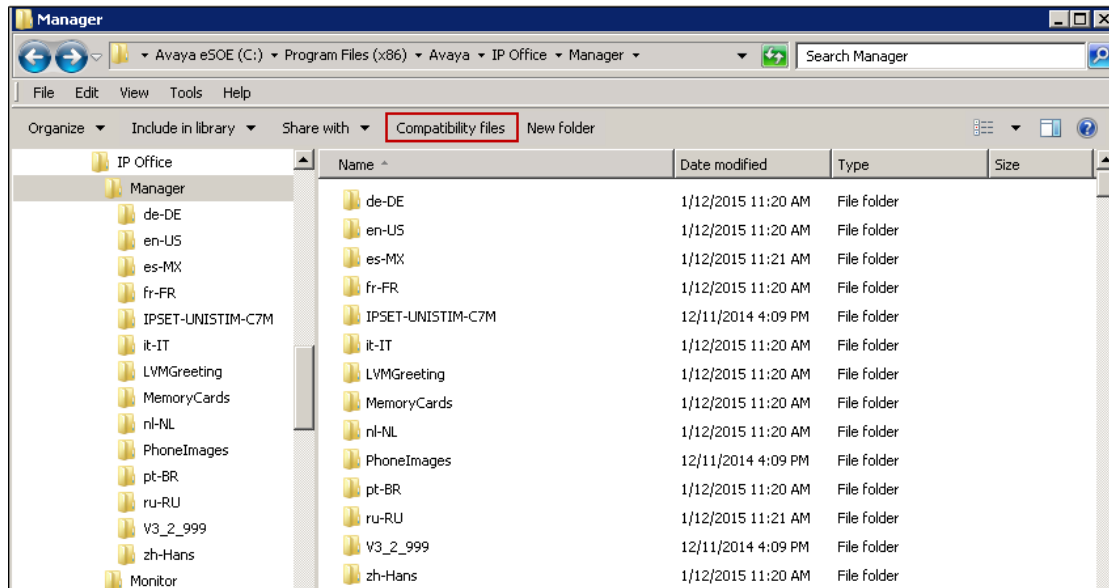
3. Import the template into IP Office Manager. From IP Office Manager, select **Tools** → **Import Templates in Manager**.



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_Alestra_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**.

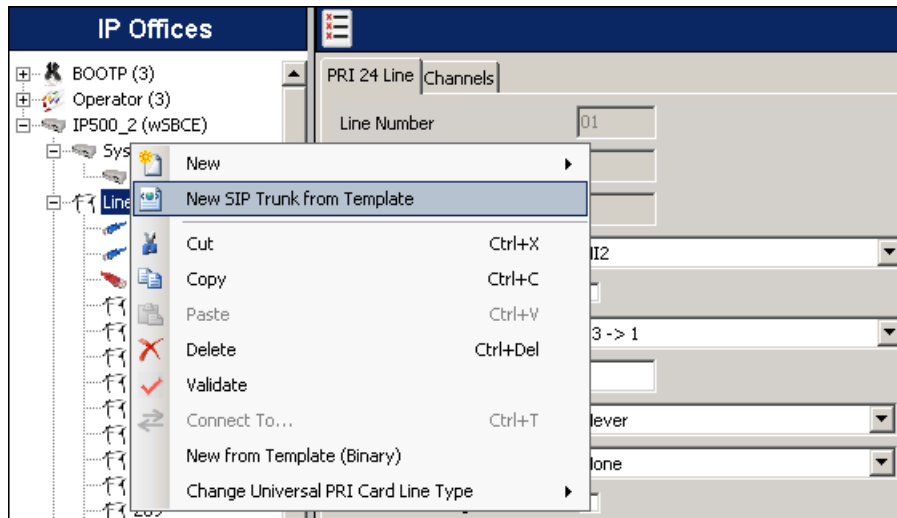


Note –Windows 7 (and later) locks the Avaya IP Office 9.1 \Templates directory, and it cannot be viewed. To enable browsing of the \Templates directory, open Windows Explorer, navigate to C:\Program Files\Avaya\IP Office\Manager (or C:\Program Files (x86)\Avaya\IP Office\Manager), and then click on the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



5.6.2. 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 5.6.1**.

Note – The drop down menu will display the *<user supplied text>* part of the template file name (see **Section 5.6.1**). If the **Display All** box is checked, then the full template file name is displayed.



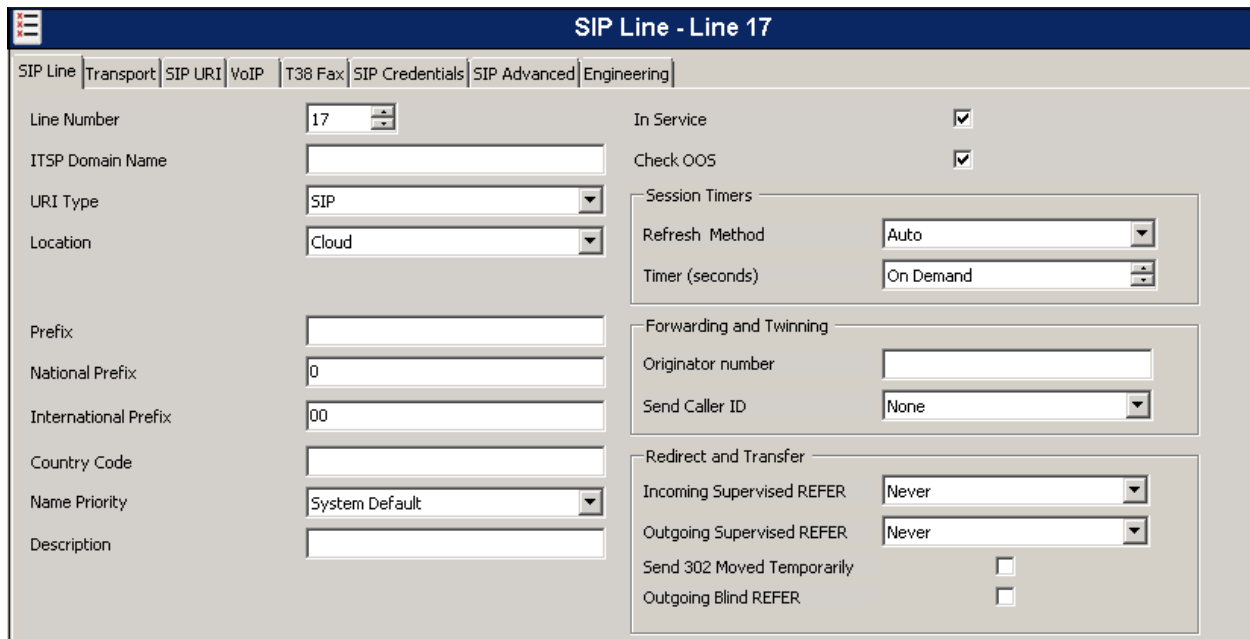
Click **Create new SIP Trunk** to finish creating the trunk.

3. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.6.3 – 5.6.7**.

5.6.3. SIP Line Tab

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

- Leave the **ITSP Domain Name** field blank. IP Office will use the IP address entered in the **Transport / ITSP Proxy Address** field in **Section 5.6.4** as the host portion of the SIP URI of SIP headers in messages sent to the Avaya SBCE.
- Check the **In Service** box.
- Check the **Check OOS** box.
- On the **Forwarding and Twinning** section, set **Send Caller ID** to *None*. This field is not used in this configuration. On outbound calls, the caller ID number shown on the PSTN end was always the main number assigned by Alestra to the enterprise, regardless of the actual number sent in any of the origination headers from the IP Office.
- On the **Redirect and Transfer** section, set **Incoming Supervised REFER** and **Outbound Supervised REFER** to *Never*. REFER is not supported by Alestra.
- Default values may be used for all other parameters.



SIP Line - Line 17	
SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials SIP Advanced Engineering	
Line Number	17
ITSP Domain Name	
URI Type	SIP
Location	Cloud
Prefix	
National Prefix	0
International Prefix	00
Country Code	
Name Priority	System Default
Description	
In Service	<input checked="" type="checkbox"/>
Check OOS	<input checked="" type="checkbox"/>
Session Timers	
Refresh Method	Auto
Timer (seconds)	On Demand
Forwarding and Twinning	
Originator number	
Send Caller ID	None
Redirect and Transfer	
Incoming Supervised REFER	Never
Outgoing Supervised REFER	Never
Send 302 Moved Temporarily	<input type="checkbox"/>
Outgoing Blind REFER	<input type="checkbox"/>

5.6.4. 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.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '10.5.5.152'. The 'Network Configuration' section contains the following settings: 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is set to '5060', 'Use Network Topology Info' is set to 'LAN 1', and 'Listen Port' is set to '5060'. The 'Explicit DNS Server(s)' field is empty, showing two sets of IP address input boxes. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

Field	Value
ITSP Proxy Address	10.5.5.152
Layer 4 Protocol	UDP
Send Port	5060
Use Network Topology Info	LAN 1
Listen Port	5060
Explicit DNS Server(s)	0 . 0 . 0 . 0
Calls Route via Registrar	<input checked="" type="checkbox"/>
Separate Registrar	

5.6.5. SIP URI Tab

SIP URI entries need to be created to match each number that Avaya IP Office and Alestra 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. To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button. In the example screen below, a previously configured entry is edited.

For the compliance test, a single SIP URI was created for calls in the outbound direction. This SIP URI will populate the user part of the origination headers on outbound calls with the 10 digit DID number set in the **SIP** tab of that User, as shown in **Section 5.7**. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to *Use Internal Data*. Set **PAI** to *None*.
- The **Registration** parameter was set to *0: <None>*. The Alestra SIP Trunk service does not require SIP trunk registration.
- Leave the **Incoming Group** field at the default value *0*.
- 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 outgoing group *17* was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Click **OK**.

SIP Line - Line 17

Tabs: SIP Line | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials | SIP Advanced | Engineering

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Cre
1	0 17	1..				N...	0: <
2	17 0	1..	3044	3044	3044	N...	0: <
3	17 0	1..	3045	3045	3045	N...	0: <
4	17 0	1..	3046	3046	3046	N...	0: <
5	17 0	1..	3047	3047	3047	N...	0: <
6	17 0	1..	3048	3048	3048	N...	0: <

Buttons: Add..., Remove, Edit...

Edit Channel

Via: 10.5.5.150

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: None

Registration: 0: <None>

Incoming Group: 0

Outgoing Group: 17

Max Calls per Channel: 10

Buttons: OK, Cancel

On inbound calls to the IP Office, Alestra sent in the Request-URI header the last 4 digits of the dialed DID numbers. In the reference configuration, separate SIP URI entries were created to match the last 4 digits of each DID number assigned by Alestra to the SIP trunk.

To configure the entries for the SIP URIs for inbound calls, select the **SIP URI** tab and click the **Add** button. To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button. The example screen below shows a previously configured entry for the DID number with last 4 digits ending in 3044. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to **3044**. Set **PAI** to **None**.
- Under **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. For the compliance test, a new incoming group **17** was defined that only contains this line (line 17).
- Leave the **Outgoing Group** field with the default value **0**.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern. In the example shown, 2 simultaneous incoming calls are allowed to this DID number.
- Click **OK**.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Cre
1	0 17	1..				N...	0: <
2	17 0	1..	3044	3044	3044	N...	0: <
3	17 0	1..	3045	3045	3045	N...	0: <
4	17 0	1..	3046	3046	3046	N...	0: <
5	17 0	1..	3047	3047	3047	N...	0: <
6	17 0	1..	3048	3048	3048	N...	0: <

Edit Channel

Via: 10.5.5.150

Local URI: 3044

Contact: 3044

Display Name: 3044

PAI: None

Registration: 0: <None>

Incoming Group: 17

Outgoing Group: 0

Max Calls per Channel: 2

OK

Cancel

Repeat the steps above for each DID number assigned by Alestra, to be accepted by the IP Office on this SIP line.

5.6.6. 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, G711A and G711U**, in this order of preference, were the codecs supported by Alestra.
- Set **Fax Transport Support** to **None**. 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.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'VoIP' tab selected. The window has a dark blue header and a light gray body. At the top, there are tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', 'SIP Credentials', 'SIP Advanced', and 'Engineering'. The 'VoIP' tab is active. The main configuration area is divided into several sections. On the left, there is a 'Codec Selection' section with a dropdown menu set to 'Custom'. Below this, there are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.723.1 6K3 MP-MLQ'. The 'Selected' list contains 'G.729(a) 8K C5-ACELP', 'G.711 ALAW 64K', and 'G.711 ULAW 64K'. Between these lists are buttons for moving items: '>>', '<<', '<', '>', and '<<<'. Below the codec lists, there are three dropdown menus: 'Fax Transport Support' set to 'None', 'DTMF Support' set to 'RFC2833', and 'Media Security' set to 'Disabled'. On the right side of the window, there are several checkboxes: 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'Allow Direct Media Path' (unchecked), 'Force direct media with phones' (unchecked), 'PRACK/100rel Supported' (checked), and 'G.711 Fax ECAN' (unchecked).

5.6.7. SIP Advanced Tab

On the **SIP Advanced** tab, verify that the **Call Routing Method** is set to the default selection, **Request URI**. On incoming calls, the “To” header arriving from Alestra contained the complete 10 digit DID number dialed, preceded by an 8 digit prefix, while the “Request-URI” header contained the last 4 digits of the DID number dialed. By making this selection, IP Office will use the “Request URI” header on the incoming INVITEs to route the call to the intended destination, while the “To” header is ignored in this decision.

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”. 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, IP Office will use the PPI header for privacy. To configure IP Office to use the PAI header for privacy calls, check the box for **Use PAI for Privacy**.

All other fields retained their default values.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'SIP Advanced' tab selected. The window has a tabbed interface with tabs for SIP Line, Transport, SIP URI, VoIP, T38 Fax, SIP Credentials, SIP Advanced, and Engineering. The SIP Advanced tab contains several sections: Addressing, Identity, Media, and Call Control. In the Addressing section, 'Association Method' is set to 'By Source IP address' and 'Call Routing Method' is set to 'Request URI'. In the Identity section, 'Use PAI for Privacy' is checked. In the Media section, 'P-Early-Media Support' is set to 'None' and 'Media Connection Preservation' is set to 'Disabled'. In the Call Control section, 'Call Initiation Timeout (s)' is 4, 'Call Queuing Timeout (m)' is 5, 'Service Busy Response' is '486 - Busy Here', 'on No User Responding Send' is '408-Request Timeout', and 'Action on CAC Location Limit' is 'Allow Voicemail'. Other options like 'Suppress DNS SRV Lookups', 'Use Phone Context', 'Add user=phone', 'Use + for International', 'Use Domain for PAI', 'Swap From and PAI', 'Caller ID from From header', 'Send From In Clear', 'Cache Auth Credentials', 'User-Agent and Server Headers', 'Allow Empty INVITE', 'Send Empty re-INVITE', 'Allow To Tag Change', 'Send SilenceSupp=Off', 'Force Early Direct Media', 'Suppress Q.850 Reason Header', 'Emulate NOTIFY for REFER', and 'No REFER if using Diversion' are all unchecked.

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

No changes were made to the **T.38 Fax**, **SIP Credentials** and the **Engineering** tabs, so they will not be visited.

5.7. Users

Configure the SIP parameters for each user that will be placing calls via the SIP line defined in **Section 5.6**. 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 *Ext 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. 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 Alestra. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.

Click **OK** (not shown) to save any changes.

The screenshot shows the Avaya SIP configuration interface. On the left, the 'IP Offices' pane lists a hierarchy: 'Extension (47)' > 'User (49)' > 'NoUser' > 'RemoteManager' > '1557 Av Com RM 1557' > '1552 Av Com SIP 1552' > '1101 Extn1101dcp' > '1102 Extn1102dcp' (selected) > '1103 Extn1103dcp' > '1104 Extn1104' > '1105 Extn1105'. The main pane is titled 'Extn1102dcp: 1102' and contains several tabs: 'User', 'Voicemail', 'DND', 'Short Codes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', 'Menu Programming', 'Mobility', 'Group Membership', 'Announcements', 'SIP' (active), and 'Personal Directory'. The 'SIP' tab displays three text input fields: 'SIP Name' with value '8122823044', 'SIP Display Name (Alias)' with value 'Extn1102dcp', and 'Contact' with value '8122823044'. Below these fields is an unchecked checkbox labeled 'Anonymous'.

5.8. 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.

To add a new incoming call route, from the left Navigation Pane, right-click on **Incoming Call Route** and select **New** (not shown). The screen below shows the route for one of the DID numbers assigned to the enterprise by Alestra, 8122823044 in this example.

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 5.6**.
- Set the **Incoming Number** to the last 4 digits of the DID number assigned by Alestra. (**3044** in this example). Alestra sent the last 4 digits of the DID numbers in the Request-URI to the IP Office.
- Default values may be used for all other parameters.

The screenshot shows the IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree structure. The 'Incoming Call Route (8)' item is selected. The main pane is titled '17 3044' and has three tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is active, showing the following configuration parameters:

Parameter	Value
Bearer Capacity	Any Voice
Line Group ID	17
Incoming Number	3044
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

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 8122823044 on line 17 are routed to extension 1102.

The screenshot shows the 'Destinations' tab of the 'Incoming Call Route' configuration for DID 17 3044. The tab is titled '17 3044' and has three tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Destinations' tab is active, showing a table with the following configuration parameters:

TimeProfile	Destination	Fallback Extension
Default Value	1102 Extn1102dcp	

5.9. 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 5.10**

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).

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree structure including: BOOTP (3), Operator (3), IP500_Lab2, System (1), IP500_Lab2, Line (20), Control Unit (4), Extension (47), User (49), Group (1), Short Code (56), Service (0), and RAS (3). The 'Short Code (56)' item is selected. The main area on the right is titled '9N: Dial' and contains the 'Short Code' configuration form. The form fields are: Code (9N), Feature (Dial), Telephone Number (N), Line Group ID (50: Main), and Locale. At the bottom, there are checkboxes for 'Force Account Code' and 'Force Authorization Code', both of which are unchecked.

5.10. 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. The highlighted entries show that for example, for calls in the local area code, the user dialed 9 plus the 8 digit local number, starting with a 2, which was the range of local numbers used during the compliance test. For national long distance calls in Mexico, the user dialed 9, then 01, followed by 10 digit numbers.

IP Offices

- BOOTP (3)
- Operator (3)
- IP500_Lab2
 - System (1)
 - IP500_Lab2
 - Line (20)
 - Control Unit (4)
 - Extension (47)
 - User (49)
 - Group (1)
 - Short Code (66)
 - Service (0)
 - RAS (1)
 - Incoming Call Route (4)
 - WAN Port (0)
 - Directory (0)
 - Time Profile (0)
 - Firewall Profile (1)
 - IP Route (5)
 - Account Code (0)
 - License (75)
 - Tunnel (0)
 - User Rights (8)
 - ARS (1)
 - 50: Main**
 - RAS Location Request (0)
 - Location (0)
 - Authorization Code (0)

Main*

ARS

ARS Route Id: 50

Route Name: Main

Dial Delay Time: System Default (4)

Description:

In Service: ☒ Out of Service Route: <None>

Time Profile: <None> Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group ID
01XXXXXXX	0151N	Dial	17
411	411	Dial	17
2XXXXXXX	2N	Dial	17
0800XXXXXX	0800N	Dial	17
031XXXXXXX	031N	Dial	17
01XXXXXXX	01N	Dial	17
001XXXXXXX	001N	Dial	17

Alternate Route Priority Level: 5

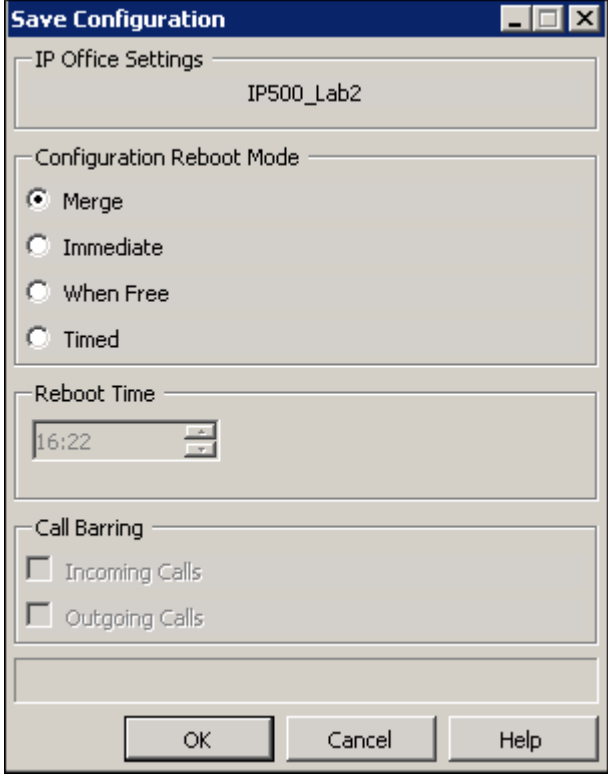
Alternate Route Wait Time: 30

Alternate Route: <None>

5.11. 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.



The image shows a 'Save Configuration' dialog box with a title bar containing a minus, maximize, and close button. The dialog is divided into several sections. The first section, 'IP Office Settings', contains a text field with the value 'IP500_Lab2'. The second section, 'Configuration Reboot Mode', contains four radio buttons: 'Merge' (selected), 'Immediate', 'When Free', and 'Timed'. The third section, 'Reboot Time', contains a time selection control showing '16:22'. The fourth section, 'Call Barring', contains two checkboxes: 'Incoming Calls' and 'Outgoing Calls', both of which are unchecked. At the bottom of the dialog are three buttons: 'OK', 'Cancel', and 'Help'.

Section	Field/Option	Value/State	
IP Office Settings	Text Field	IP500_Lab2	
Configuration Reboot Mode	Merge	Selected	
	Immediate	Unselected	
	When Free	Unselected	
	Timed	Unselected	
Reboot Time	Time Selection	16:22	
Call Barring	Incoming Calls	Unchecked	
	Outgoing Calls	Unchecked	
Buttons			OK, Cancel, Help

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.

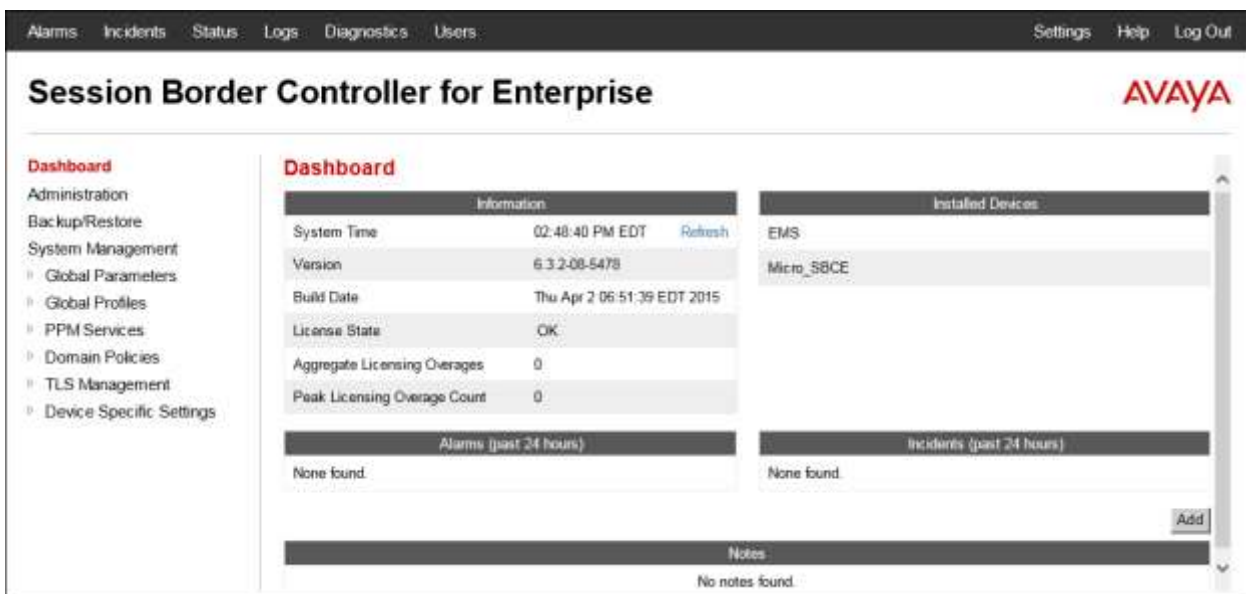
6.1. System Access

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.



The login page features the Avaya logo on the left. To the right, under the heading "Log In", are input fields for "Username:" and "Password:", followed by a "Log In" button. Below the login fields is a block of legal disclaimer text regarding system access and monitoring. At the bottom left, the text "Session Border Controller for Enterprise" is displayed.

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. Verify that the status of the **License State** field is **OK**, indicating that a valid license is present. Contact an authorized Avaya sales representative if a license is needed.



The dashboard interface includes a top navigation bar with links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main content area is titled "Session Border Controller for Enterprise" and features the Avaya logo. A left-hand navigation pane lists menu items: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, PPM Services, Domain Policies, TLS Management, and Device Specific Settings. The main dashboard area contains several sections: "Information" with system details (System Time, Version, Build Date, License State: OK, Aggregate Licensing Overages, Peak Licensing Overage Count), "Installed Devices" (listing EMS and Micro_SBCE), "Alarms (past 24 hours)" (None found), "Incidents (past 24 hours)" (None found), and "Notes" (No notes found). An "Add" button is located at the bottom right of the dashboard area.

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.

Session Border Controller for Enterprise AVAYA

System Management

Dashboard
Administration
Backup/Restore
System Management
 Global Parameters
 Global Profiles
 PPM Services
 Domain Policies
 TLS Management
 Device Specific Settings

Devices | Updates | SSL VPN | Licensing

Device Name	Management IP	Version	Status
Micro_SBCE	192.168.10.75	6.3.2-08-5478	Commissioned

Reboot Shutdown Restart Application **View** Edit Uninstall

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.

System Information: Micro_SBCE

X

General Configuration

Appliance Name	Micro_SBCE
Box Type	SIP
Deployment Mode	Proxy

Device Configuration

HA Mode	No
Two Bypass Mode	No

License Allocation

Standard Sessions Requested: 500	500
Advanced Sessions Requested: 100	100
Scopia Video Sessions Requested: 100	100
Encryption	<input checked="" type="checkbox"/>

Network Configuration

IP	Public IP	Netmask	Gateway	Interface
10.5.5.152	10.5.5.152	255.255.255.0	10.5.5.254	A1
10.5.5.153	10.5.5.153	255.255.255.0	10.5.5.254	A1
172.16.157.149	172.16.157.149	255.255.255.192	172.16.157.129	B1
172.16.157.160	172.16.157.160	255.255.255.192	172.16.157.129	B1
172.16.157.161	172.16.157.161	255.255.255.192	172.16.157.129	B1

DNS Configuration

Primary DNS	192.168.216.122
Secondary DNS	192.168.153.242
DNS Location	DMZ
DNS Client IP	172.16.157.189

Management IP(s)

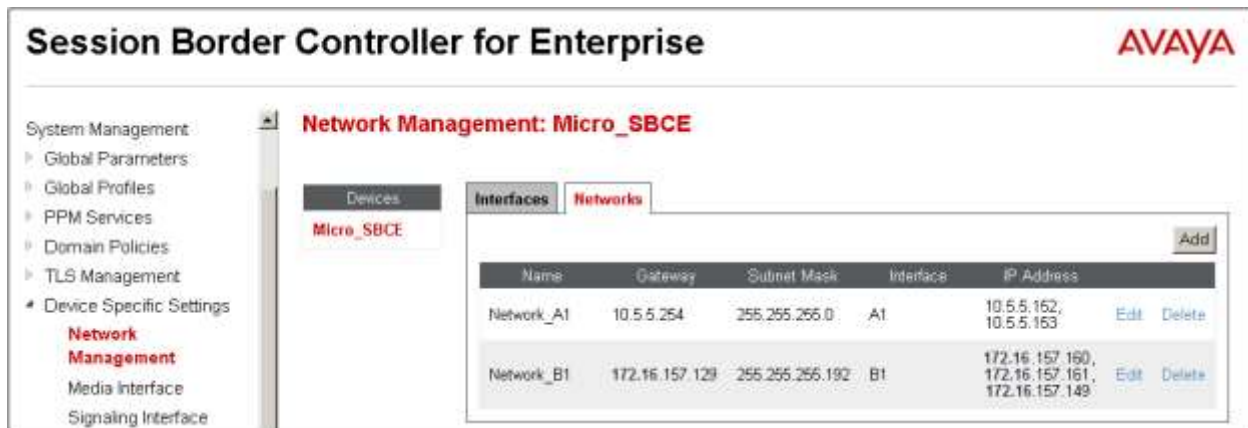
IP	192.168.10.75
----	---------------

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.

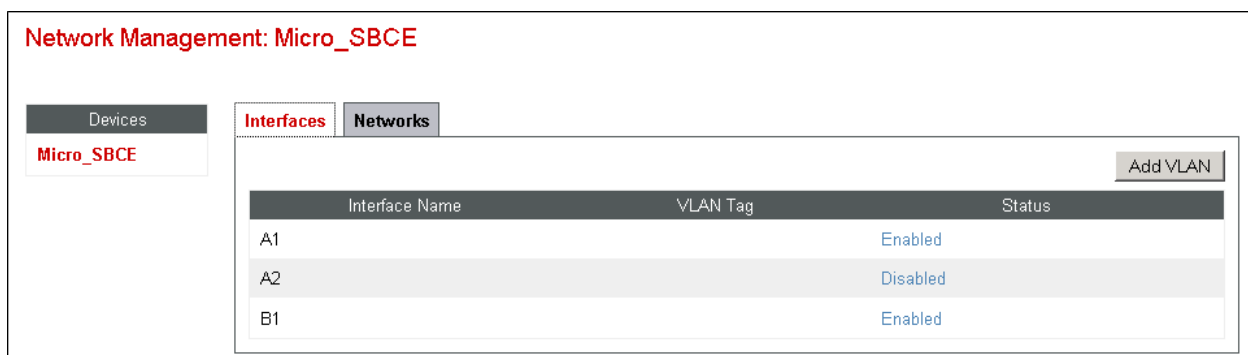
Select **Network Management** under **Device Specific Settings** on the left-side menu.

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**.



Name	Gateway	Subnet Mask	Interface	IP Address	Edit	Delete
Network_A1	10.5.5.254	255.255.255.0	A1	10.5.5.152, 10.5.5.153	Edit	Delete
Network_B1	172.16.157.129	255.255.255.192	B1	172.16.157.160, 172.16.157.161, 172.16.157.149	Edit	Delete

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.

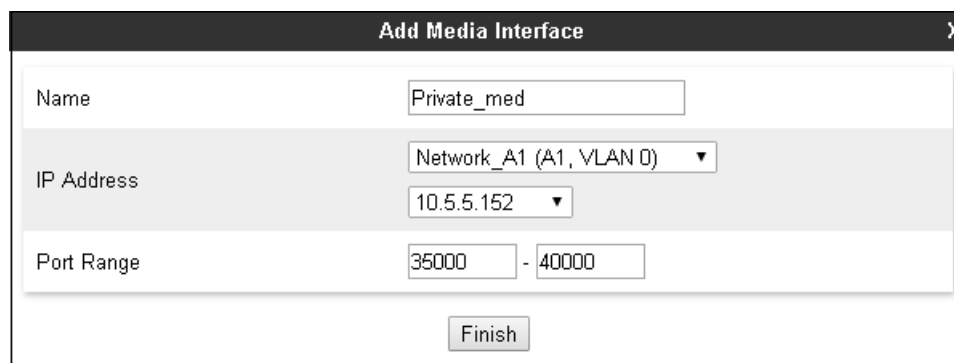


Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled

6.4. Media Interfaces

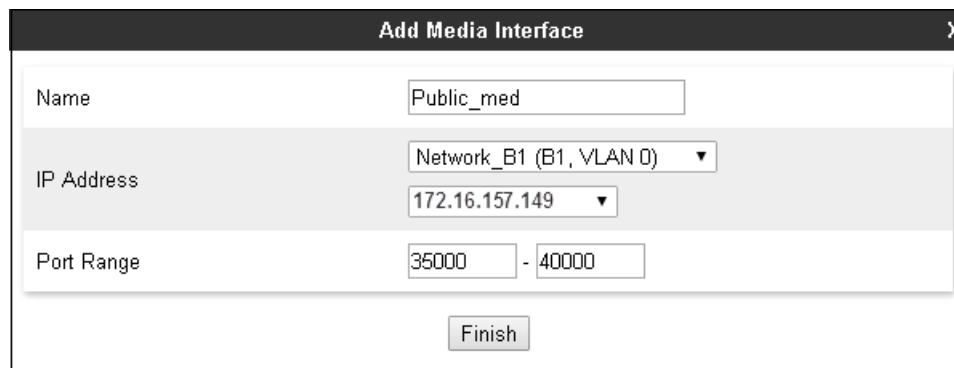
Media Interfaces were created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address, and one of the ports in this range as the listening IP address and port in which 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. Under **IP Address**, select the network associated with the private interface of the SBCE (A1) and the private IP Address used for SIP trunking, from the drop-down menus. The **Port Range** was left at the default values of **35000-40000**. Click **Finish**.



The screenshot shows a dialog box titled "Add Media Interface" with a close button (X) in the top right corner. The dialog contains three main sections: "Name" with a text input field containing "Private_med"; "IP Address" with a dropdown menu showing "Network_A1 (A1, VLAN 0)" and a sub-dropdown showing the IP address "10.5.5.152"; and "Port Range" with two input fields containing "35000" and "40000" separated by a hyphen. A "Finish" button is located at the bottom center of the dialog.

A Media Interface facing the public network side was similarly created with the name **Public_med**, as shown below. On the **IP Address** drop-down menus, the network associated with the public interface of the SBCE (B1) and the public IP Address used for SIP trunking are selected. The **Port Range** was left at the default values. Click **Finish**.



The screenshot shows a dialog box titled "Add Media Interface" with a close button (X) in the top right corner. The dialog contains three main sections: "Name" with a text input field containing "Public_med"; "IP Address" with a dropdown menu showing "Network_B1 (B1, VLAN 0)" and a sub-dropdown showing the IP address "172.16.157.149"; and "Port Range" with two input fields containing "35000" and "40000" separated by a hyphen. A "Finish" button is located at the bottom center of the dialog.

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

Media Interface: Micro_SBCE

Devices
Micro_SBCE

Media Interface

Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from [System Management](#).

Add

Name	Media IP Network	Port Range	Edit	Delete
Private_med	10.5.5.152 Network_A1 (A1, VLAN 0)	35000 - 40000	Edit	Delete
Public_media	172.16.157.149 Network_B1 (B1, VLAN 0)	35000 - 40000	Edit	Delete

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. Under **IP Address**, select the network associated with the private interface of the SBCE (A1) and the private IP Address used for SIP trunking, from the drop-down menus. Enter **5060** for **UDP Port**, since UDP port 5060 is used for signaling traffic from IP Office in the sample configuration, **Section 5.6.4**. Click **Finish**.

Add Signaling Interface X

Name: Private_sig

IP Address: Network_A1 (A1, VLAN 0) 10.5.5.152

TCP Port: Leave blank to disable

UDP Port: 5060

TLS Port: Leave blank to disable

TLS Profile: None

Enable Shared Control: ☐

Shared Control Port:

Finish

A second Signaling Interface with the name **Public_sig** was similarly created in the service provider's direction. On the **IP Address** drop-down menus, the network associated with the public interface of the SBCE (B1) and the public IP Address used for SIP trunking are selected. Enter **5060** for **UDP Port**. Click **Finish**.

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

Devices

Micro_SBCE

Signaling Interface

Modifying or deleting an existing signaling interface will require an application restart before taking effect. Application restarts can be issued from [System Management](#).

Add

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile	
Private_sig	10.5.5.152 Network_A1 (A1, VLAN 0)	---	5060	---	None	Edit Delete
Public_sig	172.16.157.149 Network_B1 (B1, VLAN 0)	---	5060	---	None	Edit Delete

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**.

Clone Profile X

Profile Name	avaya-ru
Clone Name	<input type="text" value="IP Office"/>

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			
Re-Invite Handling	No			

Scroll down to the bottom of the tab to see the rest of the settings. Click **Edit** if changes to any of the parameters are needed.

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

General	Timers	URI Manipulation	Header Manipulation	Advanced
Record Routes				Both Sides
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
Lync Extensions				No

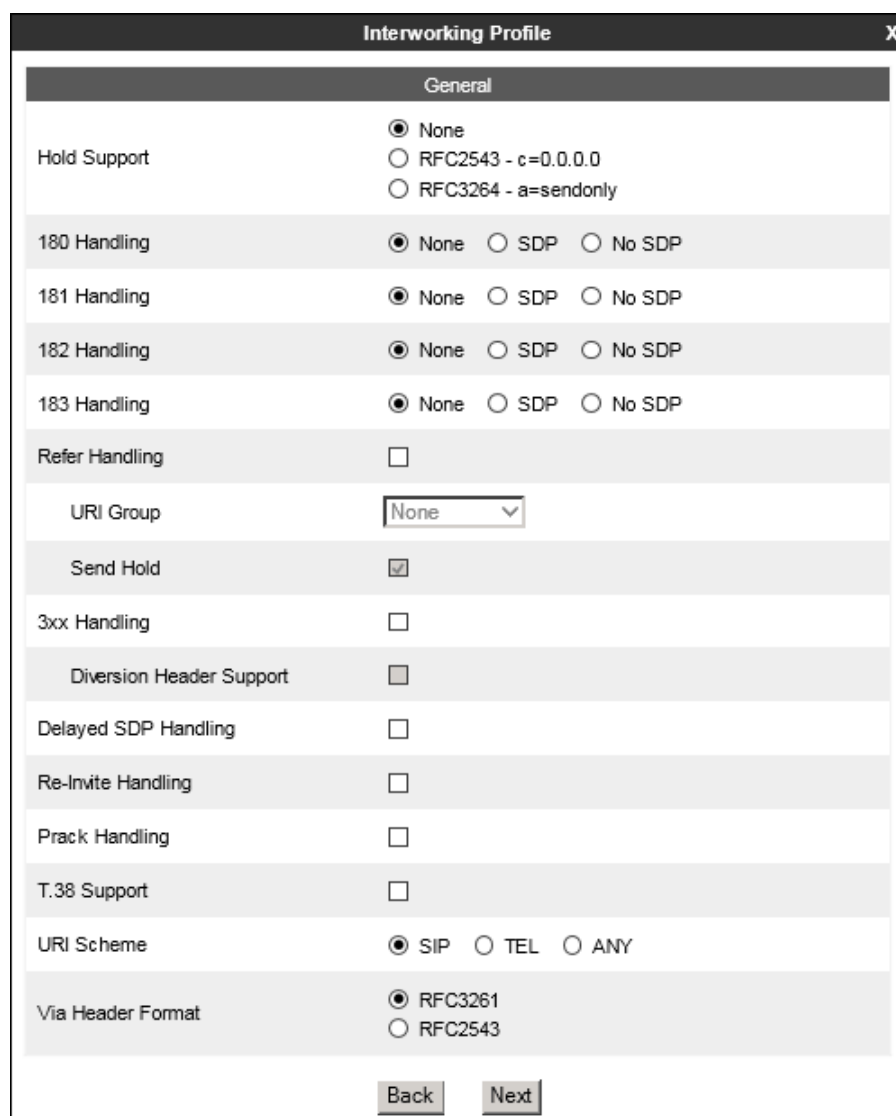
6.6.2. Server Interworking Profile – Service Provider

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**.



The screenshot shows a dialog box titled "Interworking Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" which contains the text "Service Provider". Below this field, there is a "Next" button.

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



The screenshot shows the "Interworking Profile" dialog box with the "General" tab selected. The dialog contains the following configuration options:

- Hold Support:** Radio buttons for ☒ None, ☐ RFC2543 - c=0.0.0.0, and ☐ RFC3264 - a=sendonly.
- 180 Handling:** Radio buttons for ☒ None, ☐ SDP, and ☐ No SDP.
- 181 Handling:** Radio buttons for ☒ None, ☐ SDP, and ☐ No SDP.
- 182 Handling:** Radio buttons for ☒ None, ☐ SDP, and ☐ No SDP.
- 183 Handling:** Radio buttons for ☒ None, ☐ SDP, and ☐ No SDP.
- Refer Handling:** A checkbox that is unchecked.
- URI Group:** A dropdown menu showing "None".
- Send Hold:** A checkbox that is checked.
- 3xx Handling:** A checkbox that is unchecked.
- Diversion Header Support:** A checkbox that is unchecked.
- Delayed SDP Handling:** A checkbox that is unchecked.
- Re-Invite Handling:** A checkbox that is unchecked.
- Prack Handling:** A checkbox that is unchecked.
- T.38 Support:** A checkbox that is unchecked.
- URI Scheme:** Radio buttons for ☒ SIP, ☐ TEL, and ☐ ANY.
- Via Header Format:** Radio buttons for ☒ RFC3261 and ☐ RFC2543.

At the bottom of the dialog, there are "Back" and "Next" buttons.

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

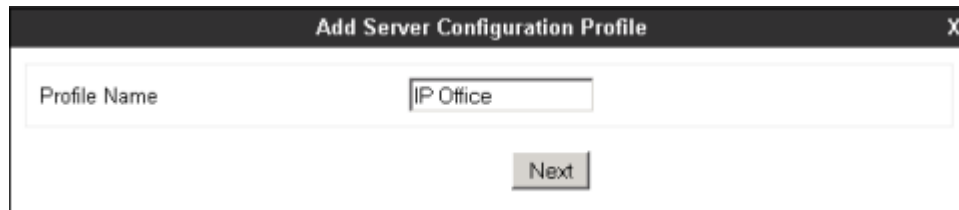
Interworking Profile	
Record Routes	<input type="radio"/> None <input type="radio"/> Single Side <input type="radio"/> Dialog-Initiate Only (Single Side) <input type="radio"/> Both Sides
Topology Hiding: Change Call-ID	<input checked="" type="checkbox"/>
Call-Info NAT	<input type="checkbox"/>
Change Max Forwards	<input checked="" type="checkbox"/>
Include End Point IP for Context Lookup	<input type="checkbox"/>
OCS Extensions	<input type="checkbox"/>
AVAYA Extensions	<input type="checkbox"/>
NORTEL Extensions	<input type="checkbox"/>
Diversion Manipulation	<input type="checkbox"/>
Diversion Condition	<input type="text" value="None"/>
Diversion Header URI	<input type="text"/>
Metaswitch Extensions	<input type="checkbox"/>
Reset on Talk Spurt	<input type="checkbox"/>
Reset SRTP Context on Session Refresh	<input type="checkbox"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Cisco Extensions	<input type="checkbox"/>
Lync Extensions	<input type="checkbox"/>
SBC FQDN	<input type="text"/>
<input type="button" value="Back"/> <input type="button" value="Finish"/>	

6.7. 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).

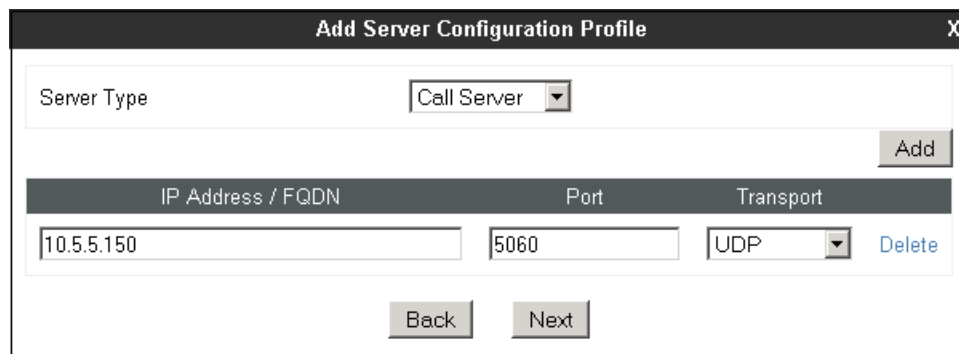
6.7.1. Server Configuration Profile – Avaya IP Office

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**.



The screenshot shows a dialog box titled "Add Server Configuration Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" containing the text "IP Office". Below this field is a "Next" button.

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 5.6**. Click **Next**.



The screenshot shows the "Add Server Configuration Profile" dialog box with the following details:

- Server Type:** A dropdown menu set to "Call Server".
- IP Address / FQDN:** A text field containing "10.5.5.150".
- Port:** A text field containing "5060".
- Transport:** A dropdown menu set to "UDP".
- Buttons:** An "Add" button is located to the right of the Transport dropdown. At the bottom of the dialog are "Back" and "Next" buttons.

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**.

The screenshot shows a dialog box titled "Add Server Configuration Profile - Advanced". It contains the following fields and controls:

- Enable DoS Protection**: A checkbox that is currently unchecked.
- Enable Grooming**: A checkbox that is currently unchecked.
- Interworking Profile**: A dropdown menu with "IP Office" selected.
- Signaling Manipulation Script**: A dropdown menu with "None" selected.
- Connection Type**: A dropdown menu with "SUBID" selected.
- At the bottom, there are two buttons: "Back" and "Finish".

6.7.2. 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**.

The screenshot shows a dialog box titled "Add Server Configuration Profile". It contains the following fields and controls:

- Profile Name**: A text input field containing "Service Provider".
- At the bottom, there is a "Next" button.

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 **192.168.145.9**, the IP address of the service provider SIP proxy server. Enter **5060** under **Port**, and select **UDP** for **Transport**. Click **Next**.

The screenshot shows a dialog box titled "Add Server Configuration Profile". It contains the following fields and controls:

- Server Type**: A dropdown menu with "Trunk Server" selected.
- An **Add** button is located to the right of the Server Type dropdown.
- Below the Server Type field is a table with three columns: "IP Address / FQDN", "Port", and "Transport".
- The table contains one row with the following values: "192.168.145.9" in the IP Address / FQDN column, "5060" in the Port column, and "UDP" in the Transport column.
- A "Delete" link is located to the right of the table row.
- At the bottom, there are two buttons: "Back" and "Next".

Click **Next** on the **Authentication** tab. On the **Heartbeat** tab:

- 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 Alestra proxy server in order to refresh the registration binding of the SIP trunk. **300** seconds was the value used during the compliance test.
- The **From URI** and **To URI** entries for the OPTIONS messages are built using the IP addresses of the Avaya SBCE and the service's provider SIP proxy, like shown on the example below.
- Click **Next**.

The screenshot shows a dialog box titled "Add Server Configuration Profile - Heartbeat". It contains the following fields and controls:

- Enable Heartbeat**: A checkbox that is checked.
- Method**: A dropdown menu with "OPTIONS" selected.
- Frequency**: A text input field containing "300", followed by the label "seconds".
- From URI**: A text input field containing "sip@172.16.157.149".
- To URI**: A text input field containing "sip@192.168.145.9".
- At the bottom, there are two buttons: "Back" and "Next".

On the **Advanced** tab, select *Service Provider* from the **Interworking Profile** drop down menu. Click **Finish**

The screenshot shows a dialog box titled "Add Server Configuration Profile - Advanced". It contains the following fields and controls:

- Enable DoS Protection**: A checkbox that is unchecked.
- Enable Grooming**: A checkbox that is unchecked.
- Interworking Profile**: A dropdown menu with "Service Provider" selected.
- Signaling Manipulation Script**: A dropdown menu with "None" selected.
- Connection Type**: A dropdown menu with "SUBID" selected.
- At the bottom, there are two buttons: "Back" and "Finish".

6.8. 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 Alestra SIP trunk.

6.8.1. Routing Profile – Avaya IP Office

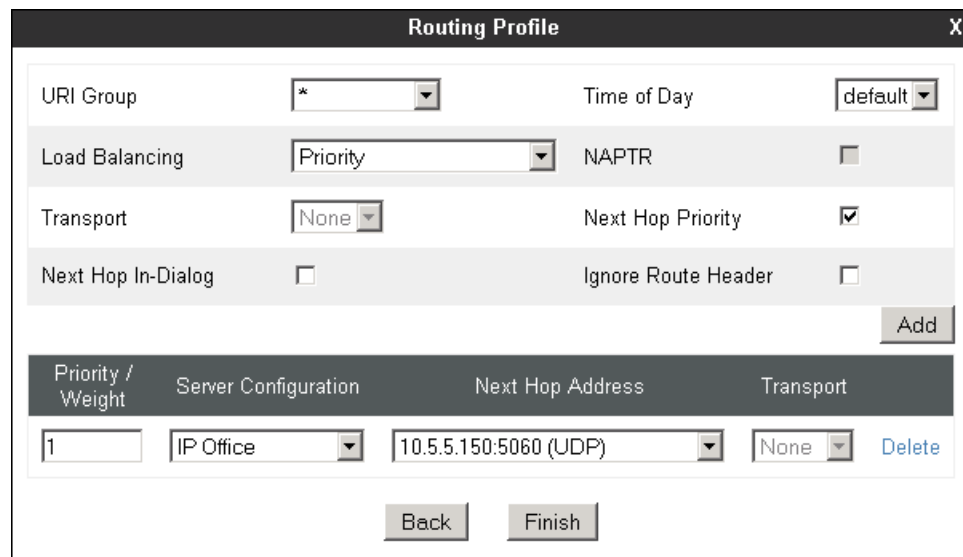
To create the inbound route, select the **Routing** tab from the **Global Profiles** menu on the left-hand side and select **Add** (not shown). Enter an appropriate **Profile Name** similar to the example below. Click **Next**.



The image shows a 'Routing Profile' dialog box. It has a title bar with 'Routing Profile' and a close button 'X'. Inside, there is a text field labeled 'Profile Name' containing the text 'Route to IP Office'. Below the text field is a button labeled '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 6.7.1**. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the IP Office Server Profile in **Section 6.7.1**. Defaults were used for all other parameters. Click **Finish**.



The image shows a 'Routing Profile' dialog box with various configuration options and a table of next-hop addresses.

Configuration options:

- URI Group: *
- Time of Day: default
- Load Balancing: Priority
- NAPTR: ☐
- Transport: None
- Next Hop Priority: ☒
- Next Hop In-Dialog: ☐
- Ignore Route Header: ☐

Buttons: Add, Back, Finish

Priority / Weight	Server Configuration	Next Hop Address	Transport	
1	IP Office	10.5.5.150:5060 (UDP)	None	Delete

6.8.2. Routing Profile – Service Provider

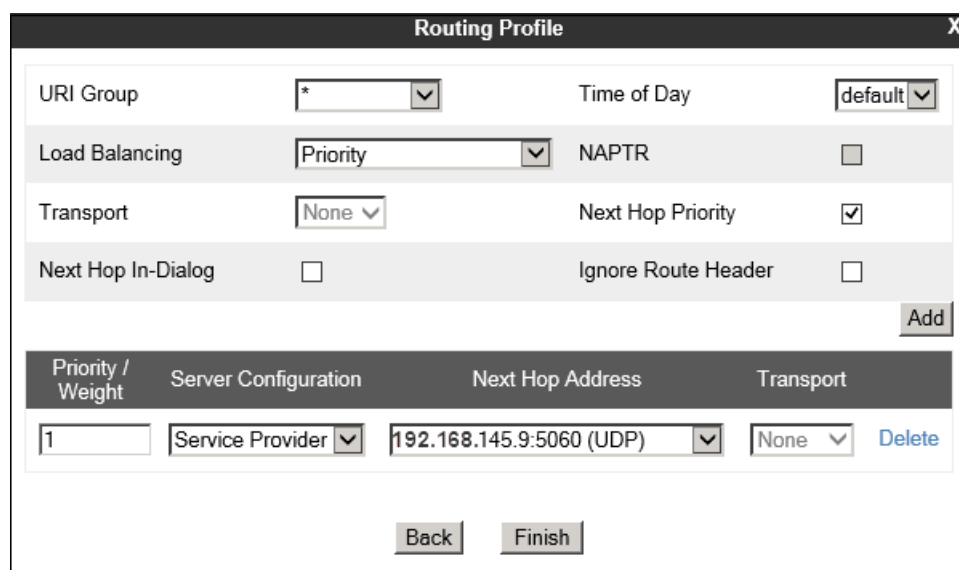
Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route. Enter an appropriate **Profile Name** similar to the example below. Click **Next**.



The image shows a dialog box titled "Routing Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" containing the text "Route to SP". Below the input field is a button labeled "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 6.7.2**. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the Server Profile corresponding to the Alestra SIP proxy server in **Section 6.7.2**. Defaults were used for all other parameters. Click **Finish**.



The image shows a "Routing Profile" dialog box with various configuration options and a table of next-hop addresses.

Configuration options:

- URI Group: *
- Time of Day: default
- Load Balancing: Priority
- NAPTR: ☐
- Transport: None
- Next Hop Priority: ☒
- Next Hop In-Dialog: ☐
- Ignore Route Header: ☐

Buttons: Add, Back, Finish

Priority / Weight	Server Configuration	Next Hop Address	Transport	
1	Service Provider	192.168.145.9:5060 (UDP)	None	Delete

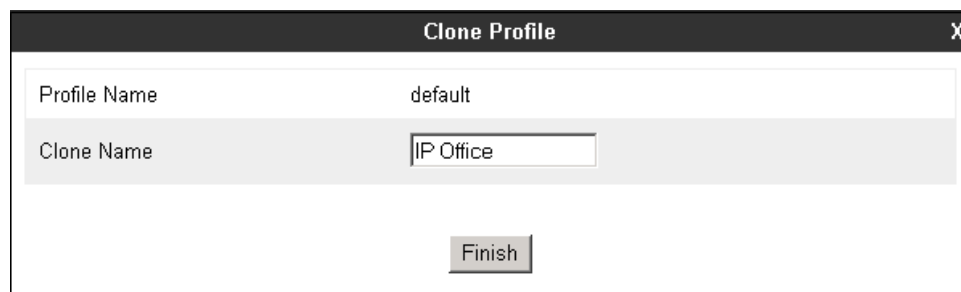
6.9. Topology Hiding

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.

6.9.1. 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**.



Clone Profile	
Profile Name	default
Clone Name	IP Office
<div>Finish</div>	

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 the host part of the SIP URIs in all SIP headers 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**.

Header	Criteria	Replace Action	Overwrite Value	
Request-Line	IP/Domain	Auto		Delete
Refer-To	IP/Domain	Auto		Delete
SDP	IP/Domain	Auto		Delete
To	IP/Domain	Auto		Delete
Record-Route	IP/Domain	Auto		Delete
Referred-By	IP/Domain	Auto		Delete
From	IP/Domain	Auto		Delete
Via	IP/Domain	Auto		Delete

Finish

6.9.2. Topology Hiding Profile – Service Provider

A Topology Hiding profile named **Service Provider** (not shown) was similarly configured in the direction of the SIP trunk to the service provider. During the compliance test, IP addresses were used in the host part of the SIP URIs in all SIP headers between the Avaya SBCE and the Alestra SIP proxy. As previously mentioned, the default action of **Auto** implies the insertion of IP addresses in the host portion of these headers. Hence it was not necessary to modify any of the headers sent to the service provider, and default values were used for all fields.

6.10. 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**.

Application Rules: default-trunk

It is not recommended to edit the defaults. Try cloning or adding a new rule instead.

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Audio	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	2000	2000
Video	<input type="checkbox"/>	<input type="checkbox"/>		
IM	<input type="checkbox"/>	<input type="checkbox"/>		

Miscellaneous

CDR Support	None
RTCP Keep-Alive	No

Clone

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

Clone Rule

Rule Name: default-trunk

Clone Name: Sessions=500

Finish

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**.

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Voice	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	500	500
Video	<input type="checkbox"/>	<input type="checkbox"/>		
IM	<input type="checkbox"/>	<input type="checkbox"/>		

Miscellaneous	
CDR Support	<input checked="" type="radio"/> None <input type="radio"/> CDR w/ RTP <input type="radio"/> CDR w/o RTP
RTCP Keep-Alive	<input type="checkbox"/>

Finish

6.11. Signaling Rules

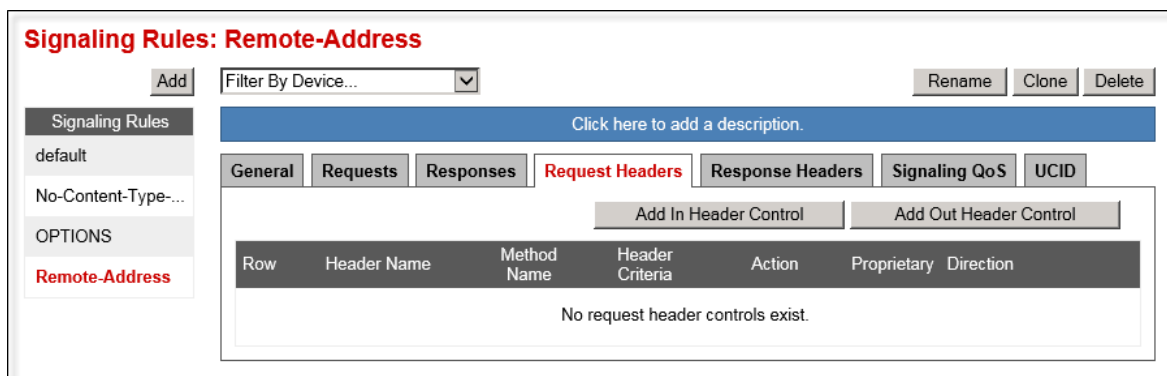
As mentioned in **Section 2.2**, a Signaling Rule was used in the Avaya SBCE to remove the “Remote-Address” header, generated by the Avaya SBCE, from outbound messages to the service provider. This header has local significance only and should not be propagated on the SIP trunk to the service provider.

In the **Domain Policies** menu on the left-hand side, select **Signaling Rules**, then **Add Rule** (not shown). Enter an appropriate name like in the example below. Click **Next**.



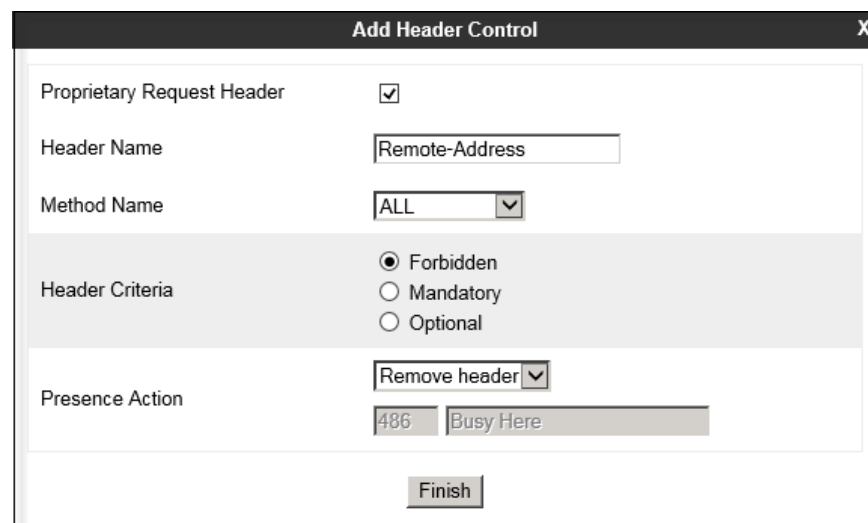
A dialog box titled "Signaling Rule" with a close button (X) in the top right corner. It contains a text field labeled "Rule Name" with the value "Remote-Address" entered. Below the text field is a "Next" button.

On the newly created **Remote-Address** Signaling Rule, select the **Request Headers** tab to create the manipulations performed on request messages. Select **Add Out Header Control**.



A screenshot of the "Signaling Rules: Remote-Address" configuration screen. The left sidebar shows a list of signaling rules: "default", "No-Content-Type...", and "Remote-Address" (highlighted in red). The main area has tabs for "General", "Requests", "Responses", "Request Headers" (selected), "Response Headers", "Signaling QoS", and "UCID". Below the tabs are buttons for "Add In Header Control" and "Add Out Header Control". A table with columns "Row", "Header Name", "Method Name", "Header Criteria", "Action", "Proprietary", and "Direction" is shown, with the message "No request header controls exist." below it. At the top right of the main area are buttons for "Add", "Filter By Device...", "Rename", "Clone", and "Delete".

Enter the settings on the **Add Header Control** screen as show below. Click **Finish**.



A dialog box titled "Add Header Control" with a close button (X) in the top right corner. It contains the following fields and controls:

- Proprietary Request Header:** A checkbox that is checked.
- Header Name:** A text field with the value "Remote-Address".
- Method Name:** A dropdown menu with the value "ALL".
- Header Criteria:** Three radio buttons: "Forbidden" (selected), "Mandatory", and "Optional".
- Presence Action:** A dropdown menu with the value "Remove header".
- 486:** A text field with the value "Busy Here".
- Finish:** A button at the bottom.

Select the **Response Headers** tab to create the manipulations performed on response messages. Select **Add Out Header Control**.

Signaling Rules: Remote-Address

Add Filter By Device... Rename Clone Delete

Click here to add a description.

General Requests Responses Request Headers **Response Headers** Signaling QoS UCID

Add In Header Control Add Out Header Control

Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction
No response header controls exist.							

Enter the settings on the **Add Header Control** screen as show below. Click **Finish**.

Add Header Control X

Proprietary Response Header ☒

Header Name Remote-Address

Response Code 2XX

Method Name ALL

Header Criteria ☒ Forbidden ☐ Mandatory ☐ Optional

Presence Action Remove header

486 Busy Here

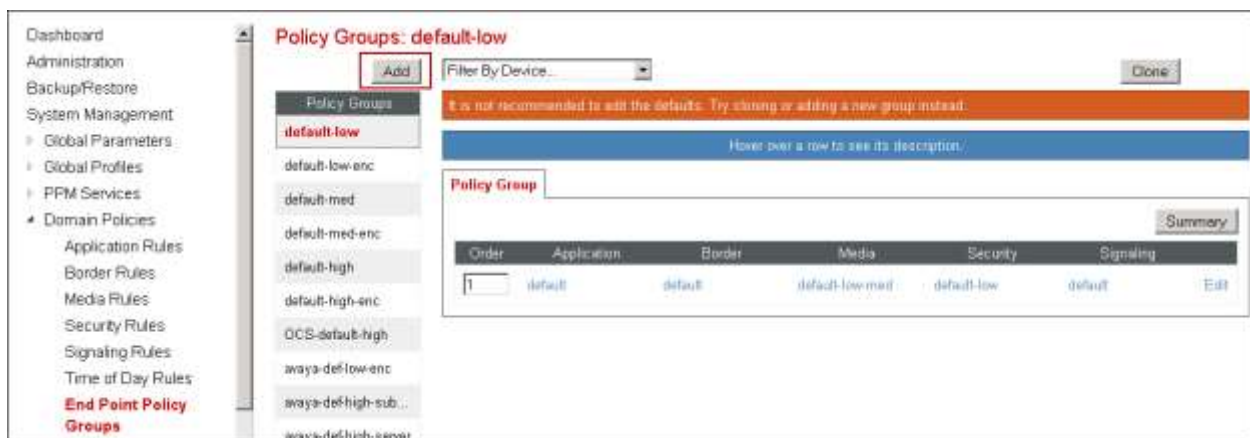
Finish

6.12. End Point Policy Groups

End Point Policy Groups associate the different sets of rules under Domain Policies (Application, Media, Signaling, 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 and Signaling Rules defined in **Sections 6.10 and 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 of 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**.

Policy Group

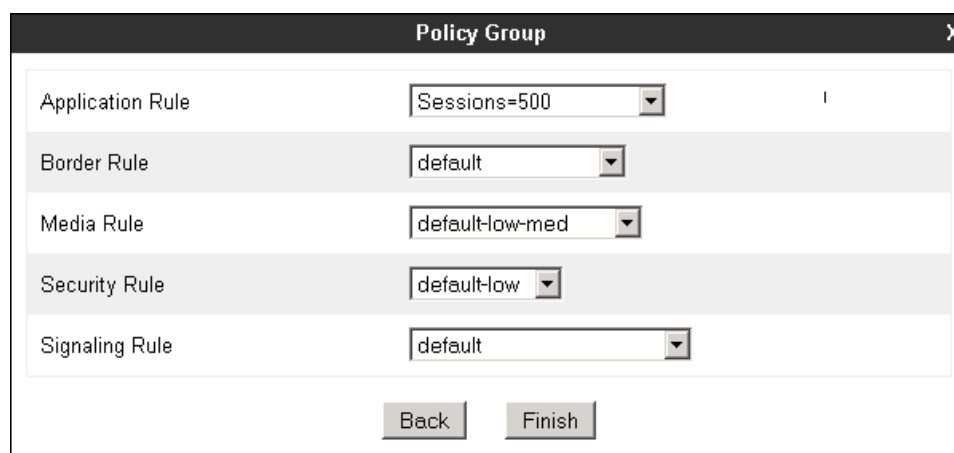
X

Group Name

IP Office

Next

In the Policy Group tab, defaults were used for all fields, with the exception of the **Application Rule**, where the *Sessions=500* rule created in **Section 6.10** was selected. Click **Finish**.



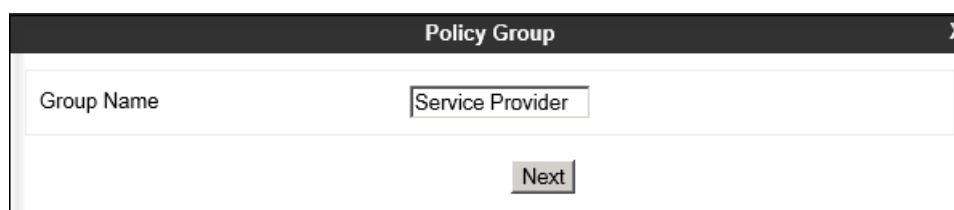
The screenshot shows a 'Policy Group' window with a close button (X) in the top right corner. It contains five rows of configuration options, each with a label and a dropdown menu:

Label	Selected Value
Application Rule	Sessions=500
Border Rule	default
Media Rule	default-low-med
Security Rule	default-low
Signaling Rule	default

At the bottom of the window are two buttons: 'Back' and 'Finish'.

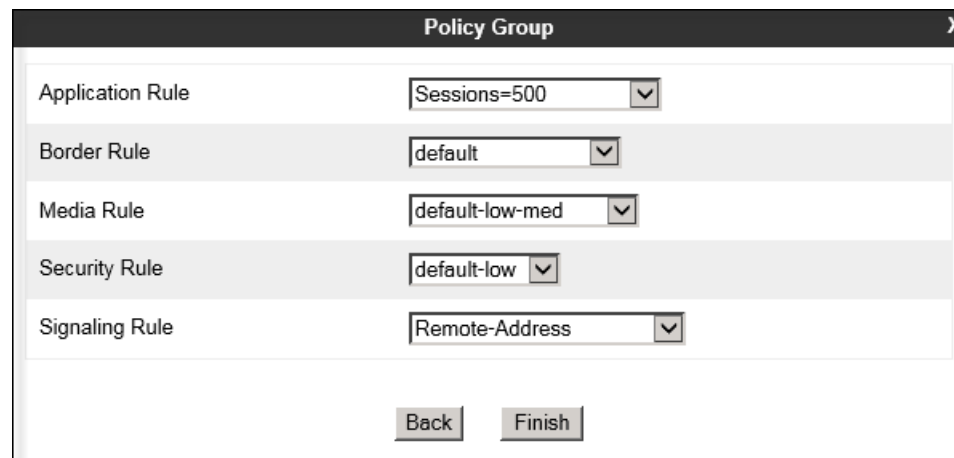
6.12.2. End Point Policy Group – Service Provider

Back at the **End Point Policy Groups** screen, select **Add** to create the End Point Policy Group for the service provider (not shown). Enter an appropriate name in the **Group Name** field. Click **Next**.



The screenshot shows a 'Policy Group' window with a close button (X) in the top right corner. It contains a single row with a label 'Group Name' and a text input field containing the text 'Service Provider'. Below the input field is a 'Next' button.

In the Policy Group tab, under **Application Rule**, select the *Sessions=500* rule created in **Section 6.10**. Under **Signaling Rule**, select the *Remote-Address* rule created in **Section 6.11**. Defaults were used for all other fields. Click **Finish**.



The screenshot shows a 'Policy Group' window with a close button (X) in the top right corner. It contains five rows of configuration options, each with a label and a dropdown menu:

Label	Selected Value
Application Rule	Sessions=500
Border Rule	default
Media Rule	default-low-med
Security Rule	default-low
Signaling Rule	Remote-Address

At the bottom of the window are two buttons: 'Back' and 'Finish'.

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 6.8.2**, which is the reverse route of the flow. Click **Finish**.

Edit Flow: IP Office Flow	
Flow Name	IP Office Flow
Server Configuration	IP Office
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Public_sig
Signaling Interface	Private_sig
Media Interface	Private_med
End Point Policy Group	IP Office
Routing Profile	Route to SP
Topology Hiding Profile	IP Office
File Transfer Profile	None
Signaling Manipulation Script	None
Remote Branch Office	Any
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 6.8.1**, which is the reverse route of the flow. Click **Finish**.

Edit Flow: SIP Trunk Flow	
Flow Name	SIP Trunk Flow
Server Configuration	Service Provider
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Private_sig
Signaling Interface	Public_sig
Media Interface	Public_media
End Point Policy Group	Service Provider
Routing Profile	Route to IP Office
Topology Hiding Profile	Service Provider
File Transfer Profile	None
Signaling Manipulation Script	None
Remote Branch Office	Any
Finish	

7. Alestra SIP Trunking Configuration

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

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

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

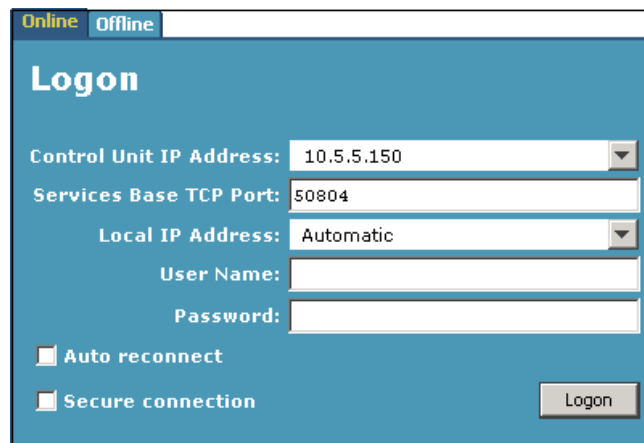
The following sections include steps that may be used to verify the configuration of the Avaya IP Office and the Avaya SBCE with the Alestra SIP Trunk Service.

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



The screenshot shows the 'Logon' window of the Avaya IP Office System Status application. At the top, there are two tabs: 'Online' (selected) and 'Offline'. The window has a blue header with the word 'Logon' in white. Below the header, there are several input fields and checkboxes. The 'Control Unit IP Address' field is a dropdown menu showing '10.5.5.150'. The 'Services Base TCP Port' field is a text box containing '50804'. The 'Local IP Address' field is a dropdown menu showing 'Automatic'. Below these are 'User Name' and 'Password' text boxes. At the bottom left, there are two checkboxes: 'Auto reconnect' and 'Secure connection', both of which are currently unchecked. A 'Logon' button is located at the bottom right of the window.

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).

Avaya IP Office System Status - IP500_Lab2 (10.5.5.150) - IP500 V2 9.1.1.0 build 10

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

System
Alarms (7)
Extensions (28)
Trunks (4)
Line:1
Line:2
Line:17
Line:18
Active Calls
Resources
Voicemail
IP Networking
Locations

Status Utilization Summary Alarms

SIP Trunk Summary

Line Service State: In Service
Peer Domain Name: sip://10.5.5.152
Resolved Address: 10.5.5.152
Line Number: 17
Number of Administered Channels: 20
Number of Channels in Use: 0
Administered Compression: G729 A, G711 A, G711 Mu
Enable Faststart: Off
Silence Suppression: Off
Media Stream: RTP
Layer 4 Protocol: UDP
SIP Trunk Channel Licenses: Unlimited
SIP Trunk Channel Licenses in Use: 0
SIP Device Features:

0%

Channel Number	URI	Call Ref	Current State	Time in State	Remote Media Add...	Codec	Connect...	Caller ID or Diske...	Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Packet ...	Transmit Jitter	Transmit Packet ...
1			Idle	00:03:41											
2			Idle	01:25:55											
3			Idle	01:48:57											
4			Idle	01:48:57											
5			Idle	01:48:57											
6			Idle	01:48:57											
7			Idle	01:48:57											

Tools Trace All Pause Ping Call Details Graceful Shutdown Force Out of Service Print... Save As...

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

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

System
Alarms (7)
Extensions (28)
Trunks (4)
Line:1
Line:2
Line:17
Line:18
Active Calls
Resources

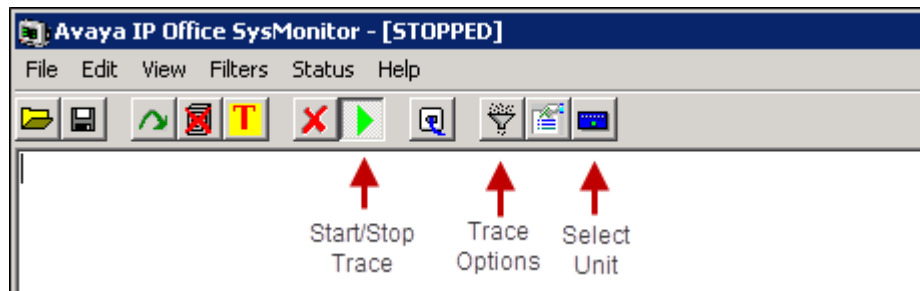
Status Utilization Summary **Alarms**

Alarms for Line: 17 SIP sip://10.5.5.152

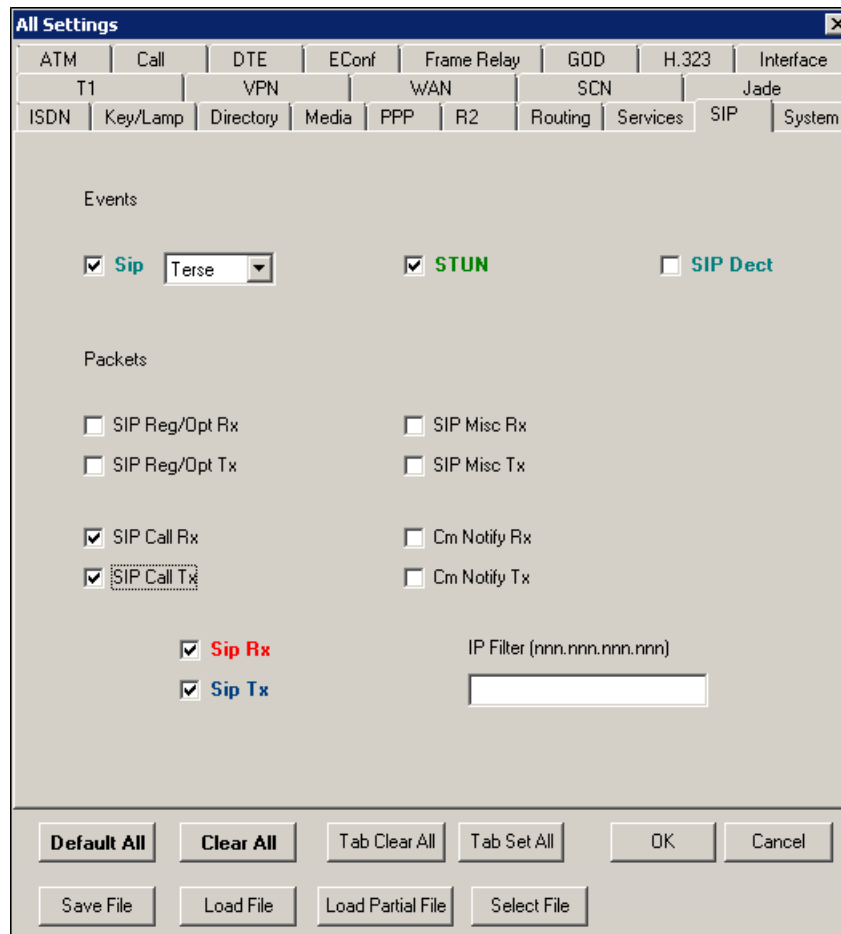
Last Date Of Error	Occurrences	Error Description
--------------------	-------------	-------------------

8.1.2. Monitor

The Avaya IP Office SysMonitor 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.



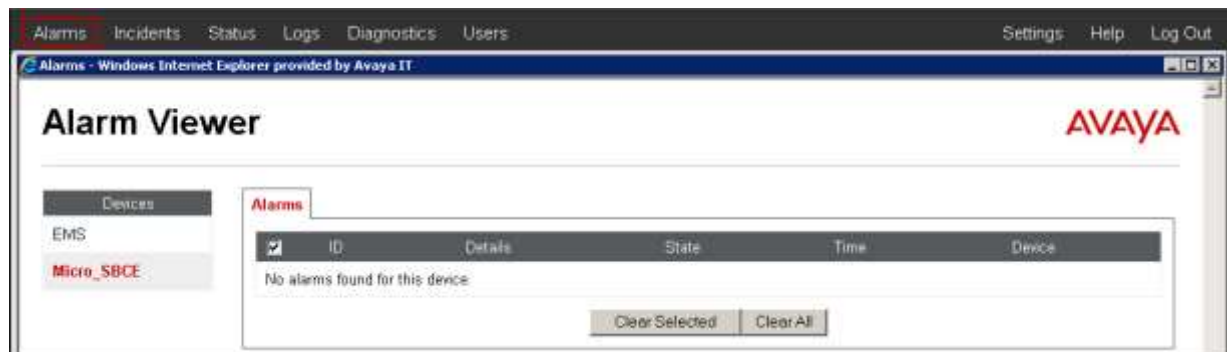
Click the **Trace Options** icon on the taskbar and select the **SIP** tab to modify the threshold used for capturing events, types of packets to be captured, filters, etc. Additionally, the color used to represent the packets in the trace can be customized by right clicking on the type of packet and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



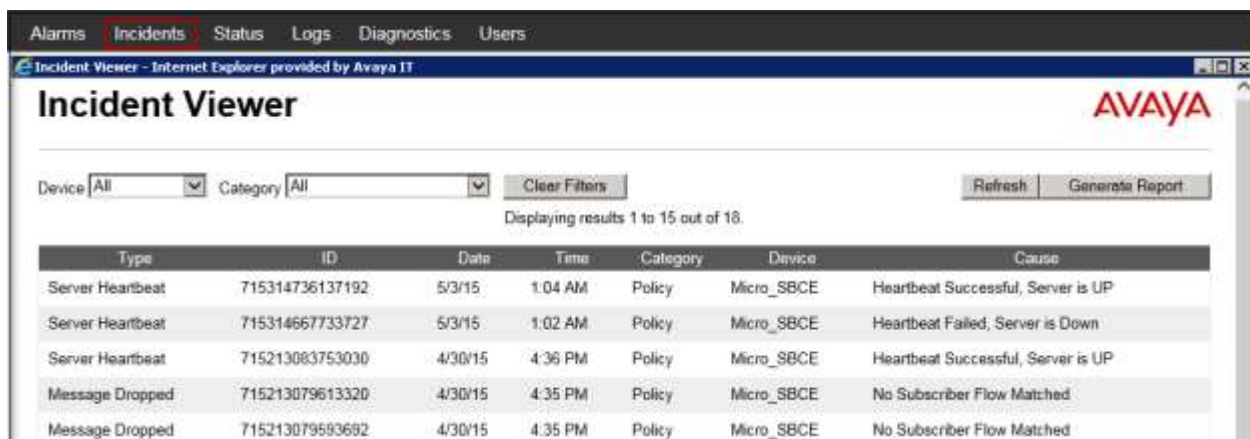
8.2. Avaya Session Border Controller for Enterprise

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.

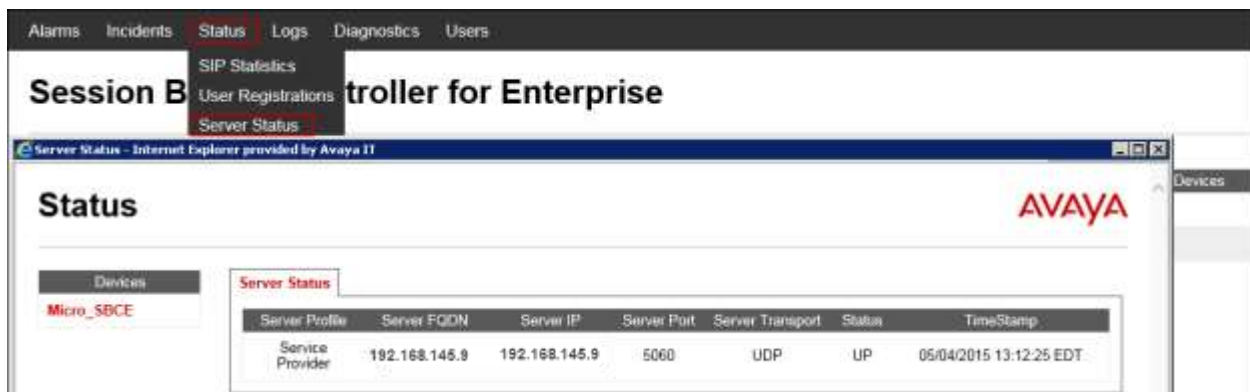
Alarms: Provides information about the health of the SBC.



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 6.8.2**.



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.

Packet Capture		Captures		
				Refresh
File Name	File Size (bytes)	Last Modified		
test_20150504132157.pcap	274,432	May 4, 2015 1:22:22 PM EDT		Delete

9. Conclusion

These Application Notes describe the procedures required to configure SIP trunk connectivity on the Avaya IP Office Release 9.1 and Avaya Session Border Controller Release 6.3, to support the Alestra SIP Trunk Service on the Sonus platform, 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, April 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, April 2015
<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 Alestra SIP Trunk Service is available from Alestra.

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