



DevConnect Program

Application Notes for Configuring Avaya IP Office Release 12.0 and Avaya Session Border Controller Release 10.2 to support WorldNet Telecommunications SIP Trunking Service - 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 12.0 and Avaya Session Border Controller Release 10.2 to support WorldNet Telecommunications SIP Trunking Service.

The test was performed to verify SIP trunk features including basic calls, call forward (all calls, busy, no answer), call transfer (blind and consultative), conference, and voice mail. The calls were placed to and from the public switched telephone network (PSTN) with various Avaya endpoints.

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 Avaya DevConnect Program.

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1. Introduction

These Application Notes describe the steps necessary for configuring Session Initiation Protocol (SIP) Trunking service between WorldNet Telecommunications and an Avaya SIP-enabled enterprise solution.

In the configuration used during the testing, the Avaya SIP-enabled enterprise solution consists of an Avaya IP Office Server Edition, two Avaya IP Office 500 V2 as expansion systems, running software release 12.0 (hereafter referred to as IP Office), an Avaya Session Border Controller Release 10.2 (hereafter referred to as Avaya SBC) and various Avaya endpoints, listed in **Section 4**.

The WorldNet Telecommunications SIP Trunking Service referenced within these Application Notes is designed for business customers. Customers using this service with the IP Office solution are able to place and receive PSTN calls via a broadband wide area network (WAN) connection using the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI trunks. This approach generally results in lower cost for the enterprise.

The terms “service provider”, “WorldNet Telecommunications” or “WorldNet” will be used interchangeably throughout these Application Notes.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to WorldNet’s network via the public Internet, as depicted in **Figure 1**, and exercise the features and functionalities 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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and WorldNet did not utilize encryption capabilities, UDP/RTP was used.

2.1. Interoperability Compliance Testing

To verify SIP trunk interoperability the following features and functionalities were exercised during the interoperability compliance test:

- SIP Trunk Registration (Dynamic Authentication).
- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various Avaya endpoints, including SIP, H.323, Digital and Analog telephones at the enterprise. All incoming calls from the PSTN were routed to the enterprise across the SIP trunk from the service provider's network.
- Outgoing PSTN calls from Avaya endpoints, including SIP and H.323, Digital and Analog telephones at the enterprise. All outgoing calls to the PSTN were routed from the enterprise across the SIP trunk to the service provider's network.
- Incoming and outgoing PSTN calls to/from Avaya Workplace Client for Windows (SIP).
- Caller ID presentation.
- Proper disconnect when the caller abandons the call before the call is answered.
- Proper disconnect via normal call termination by the caller or the called parties.
- Proper disconnect by the network for calls that are not answered (with coverage to voicemail off).
- Proper response to busy endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Proper codec negotiation and two-way speech-path. Testing was performed with codecs: G.722 64K and G.711MU, with G.722 64K being the preferred codec.
- Proper response to no matching codecs.
- Proper early media transmissions.
- Voicemail and DTMF tone support using RFC 2833 (leaving and retrieving voice mail messages, etc.).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response systems).
- Call Hold/Resume (long and short duration).
- Call Forward (unconditional, busy, no answer).
- Blind Call Transfers.
- Consultative Call Transfers.
- Station Conference.
- Mobility twinning of incoming calls to mobile phones.

Note: Remote Worker was tested as part of this solution. The configuration necessary to support remote workers is beyond the scope of these Application Notes and is not included in these Application Notes.

Items not supported or not tested included the following:

- Inbound toll-free calls, 911 calls (emergency), “0” calls (Operator), 0+10 digits calls (Operator Assisted) were not tested.
- T.38 fax: WorldNet doesn’t support T.38 fax, G.711 pass-through is the preferred fax method for WorldNet. G.711 pass-through fax was tested successfully.

2.2. Test Results

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

- **SIP endpoints may indicate that a transfer failed even when it is successful** – Occasionally on a transfer operation, Avaya IP Office SIP endpoints (Avaya 1100 Series Deskphones) may indicate on the local call display that the transfer failed even though it was successful. The frequency of this behavior can be reduced by enabling “Emulate Notify for REFER” on the IP Office SIP Line (**Section 5.4.6**).
- **WorldNet does not support REFER for call forward** – WorldNet supports REFER for call transfers to the PSTN but does not support REFER for call forward to the PSTN. The call scenario in which WorldNet does not support REFER is for inbound calls from the PSTN to IP Office which are then forwarded to another PSTN endpoint. In this scenario, if REFER is enabled (**Section 5.4.2**), WorldNet responds with a “403 Forbidden” message in response to the REFER message sent by IP Office. The “403 Forbidden” response does not have any user impact; it’s simply ignored by IP Office (the call does not drop). This issue was solved by enabling “No REFER if using Diversion” on the IP Office SIP Line, which resulted in IP Office sending REFER during call transfers to the PSTN and not sending REFER during call forwards to the PSTN (**Section 5.4.6**).
- **481 Call/Transaction Does Not exist/405 Method Not Allowed** – After a call from the PSTN to the enterprise is successfully transferred to another PSTN party using the SIP REFER method, WorldNet accepted the SIP REFER messages sent by IP Office with “202 Accepted”, which resulted in SIP trunk resources being released with BYE messages, as expected. During the process of releasing the trunk resources, after the acceptance of the SIP REFER message, it was observed that WorldNet sent a “BYE” followed by a “reINVITE”, which resulted in the Avaya SBC responding with “481 Call/Transaction Does Not Exist” and “405 Method Not Allowed”. This behaviour had no negative impact on the transferred call, SIP trunk resources were released successfully after the call transfer. It is being mentioned here simply as an observation.
- **One-Way audio during outbound calls from Avaya Workplace Client for Windows softphone (SIP) to the PSTN** – One-way audio was observed on calls originated from Avaya Workplace Client for Windows softphone (SIP) to the PSTN, there was no audio from Workplace to the PSTN, good audio from the PSTN to Workplace. The issue was observed only when **Media Security** under **System → VoIP → VoIP Security** tab was enabled in IP Office (e.g., set to “Preferred”), when set to “Disabled” audio was good in both directions. This issue is under investigation by Avaya. As a temporary work around, if TLS/SRTP is being used in IP Office, set **Media Security** under **System → VoIP → VoIP Security** tab in IP Office to **Disabled** (refer to **Section 5.2.3.2**).

2.3. Support

For support on WorldNet Telecommunications systems visit the corporate Web page at:
<https://www.worldnetpr.com/en/voice-service/>

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

Figure 1 illustrates the test configuration used for the DevConnect compliance test. The test configuration simulates an enterprise site with an Avaya SIP-enabled enterprise solution connected to the WorldNet Telecommunications SIP Trunking Service through the public Internet.

The Avaya components used to create the simulated enterprise customer site includes:

- IP Office Server Edition running in VMware environment.
 - Avaya IP Office Voicemail Pro.
- Two Avaya IP Office 500 V2 as expansion systems.
- Avaya Session Border Controller.
- Avaya J179 IP Deskphones (H.323).
- Avaya 1100 Series IP Deskphones (SIP).
- Avaya J129 IP Deskphones (SIP).
- Avaya 1400 Series Digital Deskphones.
- Analog Deskphones.
- Avaya Workplace Client for Windows (SIP).

Avaya IP Office provides the voice communications services for the enterprise. In the reference configuration, Avaya IP Office runs on the Avaya IP Office Server Edition platform. Note that this solution is extensible to deployments using the standalone IP500 V2 platform as well.

In the sample configuration, the Primary server runs the Avaya IP Office Server Edition Linux software. Avaya Voicemail Pro runs as a service on the Primary Server. The LAN1 port of the Primary Server is connected to the enterprise LAN. The LAN2 port was not used.

The Expansion Systems (IP500 V2) were used for the support of digital, analog and additional IP stations. The Avaya IP Office 500 V2 is equipped with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module). The LAN1 ports of the Avaya IP Office IP500 V2 systems are connected to the enterprise LAN, the LAN2 ports were not used.

Located at the edge of the enterprise is the Avaya SBC. The Avaya SBC has two physical interfaces, interface **B1** is used to connect to the public network, interface **A1** is used to connect to the private network. All SIP and RTP traffic entering or leaving the enterprise flows through the Avaya SBC. The Avaya SBC provides network address translation at both the IP and SIP layers.

IP endpoints at the enterprise included Avaya 1100 Series IP Deskphones (with SIP firmware), Avaya J100 Series IP Deskphones (with SIP and H.323 firmware), Avaya Workplace Client for Windows (SIP), Avaya Digital and Analog Deskphones. IP endpoints were registered to the Primary Server; non-IP endpoints (analog and digital) were registered to the Expansion Systems. The site also has a Windows PC running Avaya IP Office Manager to configure and administer the system. Mobile Twinning is configured for some of the IP Office users so that calls to these user's extensions will also ring and can be answered at the configured mobile phones.

The transport protocol between the Avaya SBC and WorldNet Telecommunications, across the public Internet, is SIP over UDP. The transport protocol between the Avaya SBC and IP Office, across the enterprise private IP network, is SIP over TLS.

For inbound calls, the calls flowed from WorldNet Telecommunications network to the Avaya SBC, then to IP Office.

Outbound calls to the PSTN were first processed by IP Office. Once IP Office selected the proper SIP trunk, the call was routed to the Avaya SBC for egress to WorldNet Telecommunications network.

For the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to WorldNet Telecommunications network. The short code 9 was stripped off by Avaya IP Office, but the remaining N digits were sent unaltered to WorldNet Telecommunications network.

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

For confidentiality and privacy purposes, public IP addresses and routable DID numbers used during the compliance testing have been masked.

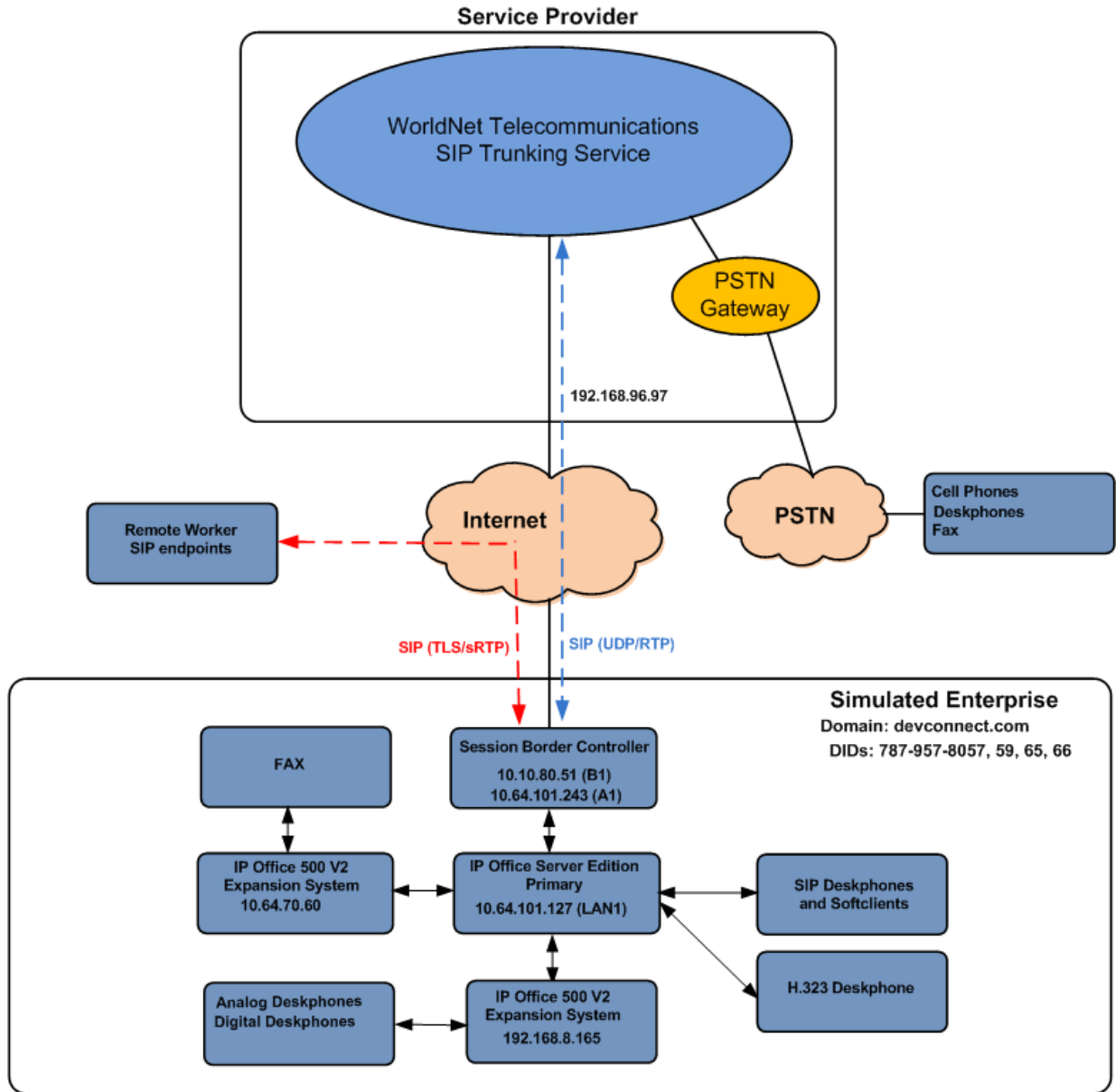


Figure 1: Avaya Interoperability Test Lab Configuration

4. Equipment and Software Validated

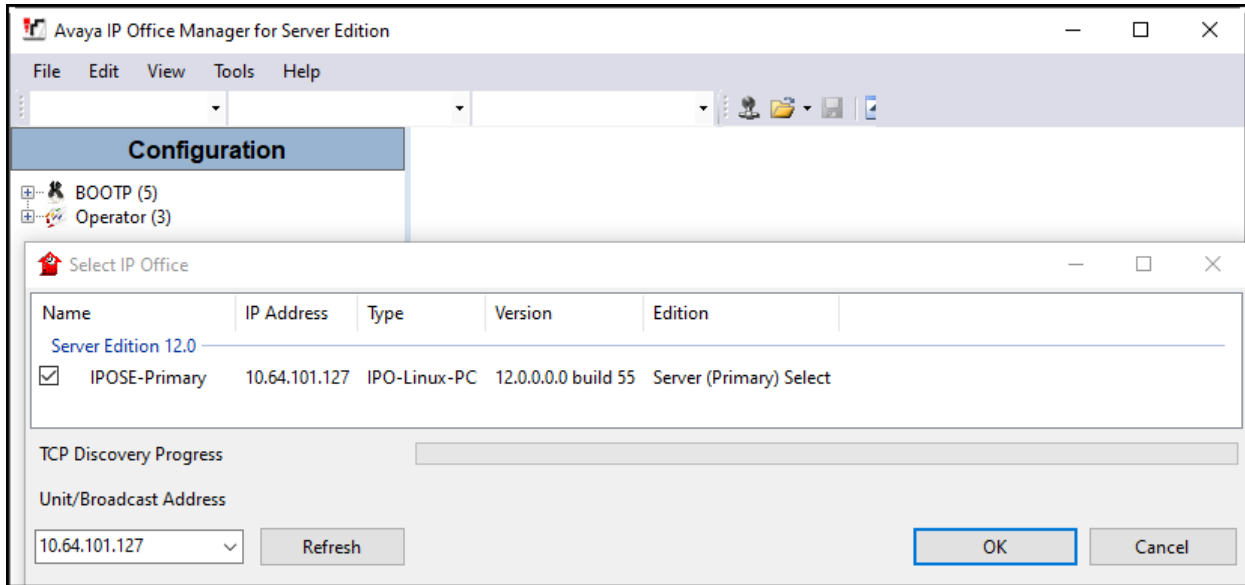
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office Server Edition (Primary Server)	12.0.0.0 Build 55
• Avaya IP Office Voicemail Pro	12.0.0.0 Build 14
Avaya IP Office IP500 V2 (Expansion Systems)	12.0.0.0 Build 55
Avaya IP Office Manager	12.0.0.0 Build 55
Avaya Session Border Controller	ASBC 10.2.0.0-86-24077
Avaya J179 IP Telephone (H.323)	Version 6.8.5.5.1
Avaya 1140E IP Deskphones (SIP)	SIP1140e Ver. 04.04.23.00
Avaya J129 IP Deskphones (SIP)	Version 4.0.10.3.2
Avaya 1408 Digital Telephone	48.02
Avaya Workplace Client for Windows (SIP).	3.36.0.137
Analog Telephone	---
WorldNet Telecommunications	
Metaswitch	CFS: V9.3.20
Oracle SBC	Acme Packet 4600 SCZ8.1.0 GA (Build 33)

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition. IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints.

5. Avaya IP Office Primary Server Configuration

Avaya IP Office is configured through the Avaya IP Office Manager application. From the PC running the IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the Manager application. Log in using the appropriate credentials.



On Server Edition systems, the Solution View screen will appear, similar to the one shown below. All the Avaya IP Office configurable components are shown in the left pane, known as the Navigation Pane. Clicking the “plus” sign next to the Primary server system name, e.g., **IPOSE-Primary**, on the navigation pane will expand the menu on this server.

Configuration | **Server Edition**

Summary

Server Edition Primary

Hardware Installed

- Control Unit: IPO-Linux-PC
- Secondary Server: NONE
- Expansion Systems: 10.64.70.60; 192.168.8.165
- System Identification: bb2fda16200635373b6925fd6df8d37e291e5030

System Settings

- IP Address: 10.64.101.127
- Sub-Net Mask: 255.255.255.0
- System Locale: United States (US English)
- System Location: 3: Thornton, CO
- Device ID: NONE
- Number of Extensions on System: 6

Description	Name	Address	Primary Link	Secondary Link	Users Configured	Extensions Configured
Solution					32	54
Primary Server	IPOSE-Primary	10.64.101.127			6	6
Expansion System	IP500V2-One	192.168.8.165	Bothway		25	24
Expansion System	IP500V2-Two	10.64.70.60	Bothway		1	24

Ready

In the screens presented in the following sections, 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 rest of this document.

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, **IPOSE-Primary** was used as the system name of the Primary Server, **IP500V2-One** and **IP500V2-Two** were used as the system name for the two Expansion Systems. All navigation described in the following sections (e.g., **License**) appears as submenus underneath the system name in the Navigation Pane.

Navigate to **License** in the Navigation Pane. In the Details Pane verify that the **License Status** for **SIP Trunk Channels** is Valid and that the number of **Instances** is sufficient to support the number of channels provisioned for the SIP trunk.

The screenshot displays the Configuration Manager interface. On the left is the Navigation Pane showing a tree view of the system configuration, with 'IPOSE-Primary' selected. On the right is the Details Pane, which is currently showing the 'License' configuration for a 'Remote Server'. The 'License Mode' is set to 'License Normal' and the 'Licensed Version' is '12.0'. Below this, there is a table listing various features and their licensing details. The 'SIP Trunk Channels' feature is highlighted with a red border, showing 256 instances, a 'Valid' status, and an expiration date of 'Never'. Other features listed include Customer Service Agent, Customer Service Supervisor, Avaya IP endpoints, IP500 Universal PRI, CTI Link Pro, Wave User, 3rd Party IP Endpoints, Server Edition, UMS Web Services, and Avaya Mac Softphone.

Feature	Instances	Status	Expiration Date	Source
Customer Service Agent	20	Dormant	Never	PLDS Nodal
Customer Service Supervisor	20	Dormant	Never	PLDS Nodal
Avaya IP endpoints	1000	Valid	Never	PLDS Nodal
SIP Trunk Channels	256	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal
Wave User	16	Obsolete	Never	PLDS Nodal
3rd Party IP Endpoints	1000	Valid	Never	PLDS Nodal
Server Edition	150	Valid	Never	PLDS Nodal
UMS Web Services	1000	Valid	Never	PLDS Nodal
Avaya Mac Softphone	1000	Valid	Never	PLDS Nodal

5.2. System Settings

Configure the necessary system settings. In an Avaya IP Office, the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side). For the compliance test, the **LAN1** interface was used to connect IP Office to the enterprise private network (LAN), **LAN2** was not used.

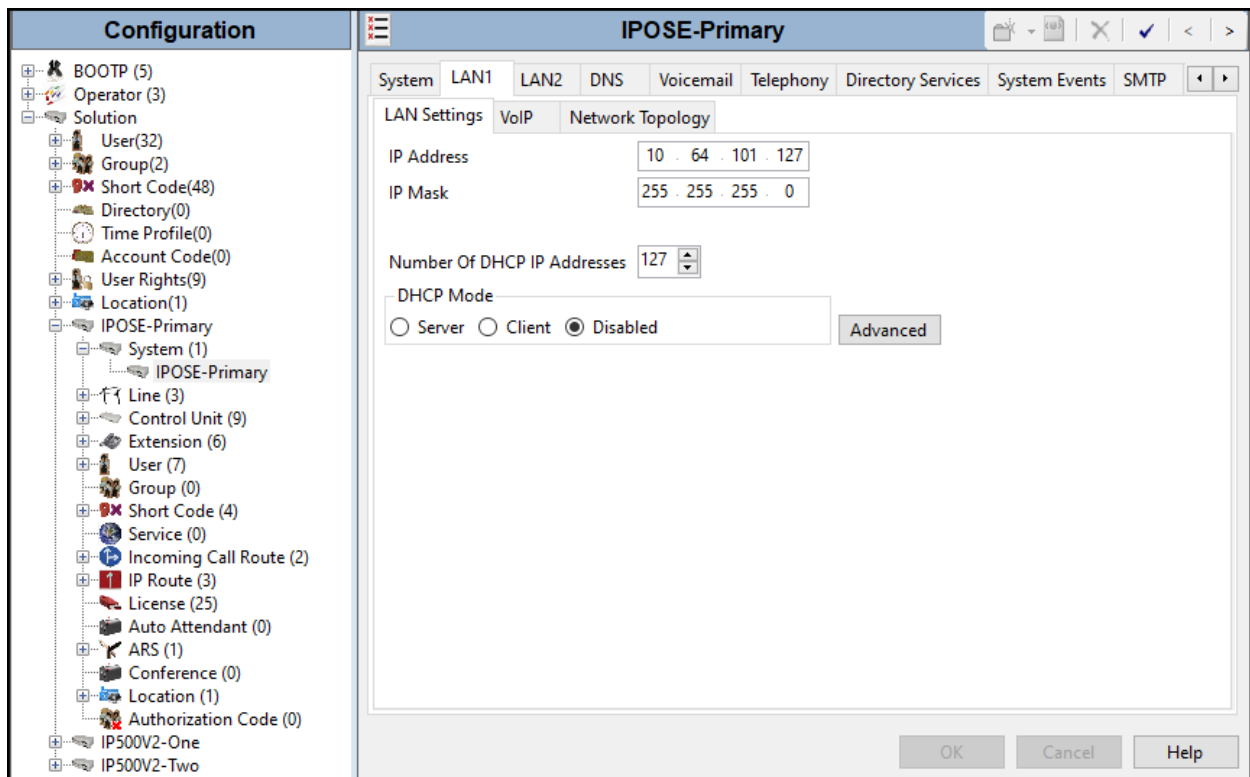
5.2.1. System - LAN1 Tab

In the sample configuration, **IPOSE-Primary** was used as the system name, the **LAN1** port connects to the inside interface (enterprise private network side) of the Avaya SBC across the enterprise LAN (private) network. The outside interface of the Avaya SBC connects to WorldNet Telecommunications network via the public internet. To access the **LAN1** settings, navigate to **System (1) → IPOSE-Primary** in the Navigation Pane.

5.2.1.1 LAN1 LAN Settings tab

The **LAN Settings** tab as shown in the screenshot below was configured with following settings:

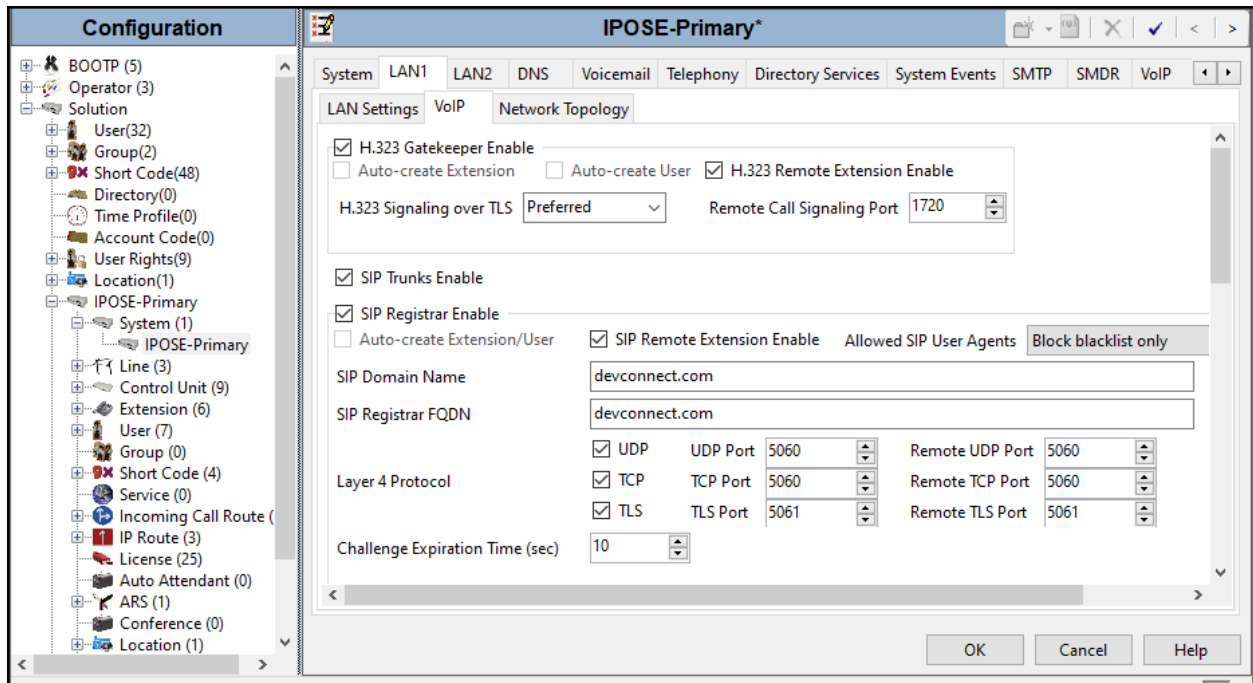
- Set the **IP Address** field to the LAN IP address, e.g., **10.64.101.127**.
- Set the **IP Mask** field to the subnet mask of the enterprise private network, e.g., **255.255.255.0**.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit.



5.2.1.2 LAN1 VoIP Tab

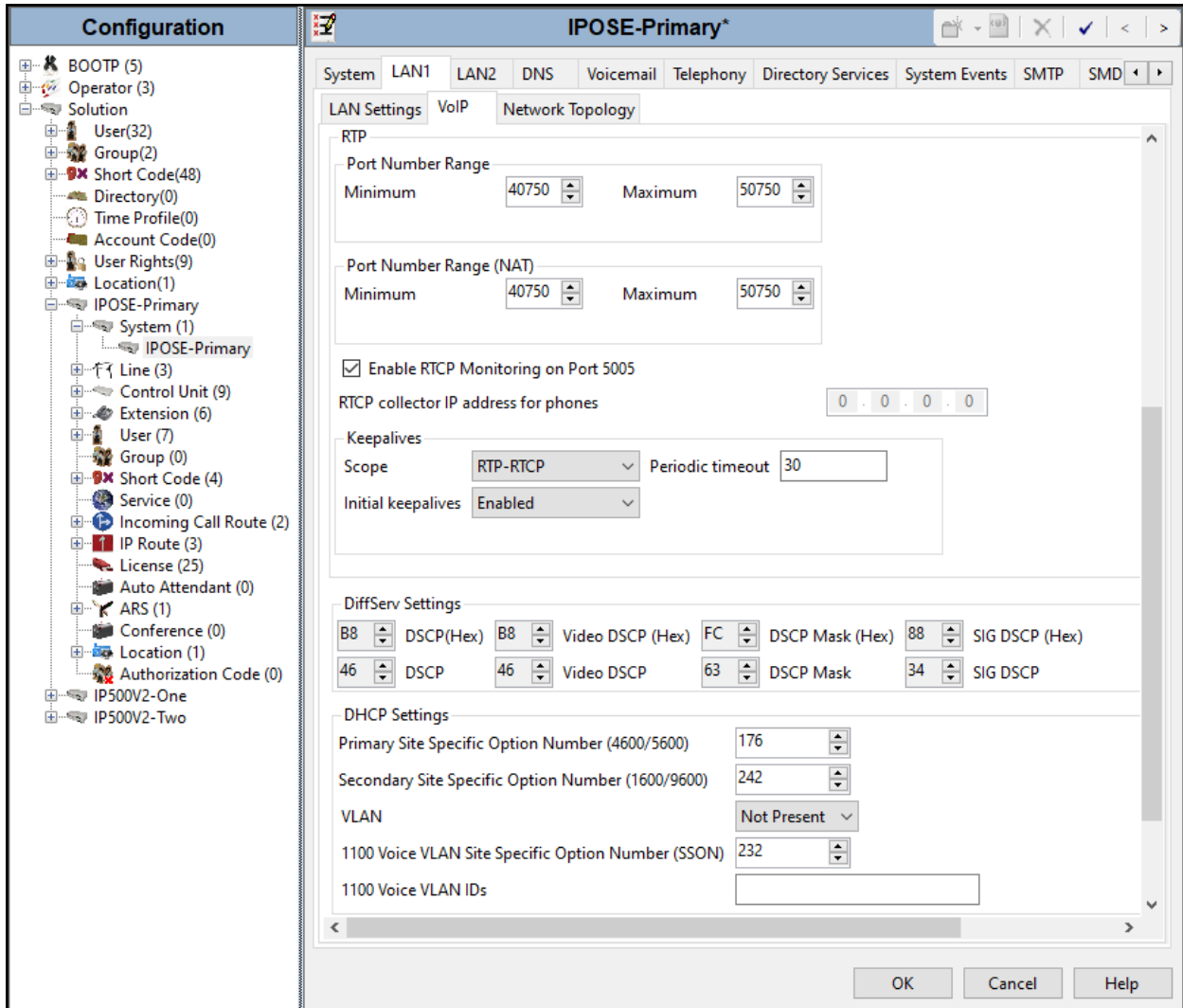
The **VoIP** tab as shown in the screenshot below was configured with following settings:

- Check the **H323 Gatekeeper Enable** to allow Avaya IP Telephones/Softphone using the H.323 protocol to register.
- Select **Preferred** under **H.323 Signaling over TLS**. When enabled, TLS is used to secure the registration and call signaling communication between IP Office and endpoints that support TLS. The H.323 phones that support TLS are 9608, 9611, 9621, 9641 running firmware version 6.6 or higher and the Avaya J100 Series IP Deskphones.
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to WorldNet Telecommunications.
- Check the **SIP Registrar Enable** to allow Avaya IP Telephones/Softphone to register using the SIP protocol.
- Enter the Domain Name of the enterprise under **SIP Domain Name**.
- Enter the SIP Registrar FQDN of the enterprise under **SIP Registrar FQDN**.
- Check TLS and verify the **TLS Port** numbers under **Layer 4 Protocol** are set to **5061**.



Scroll down the page:

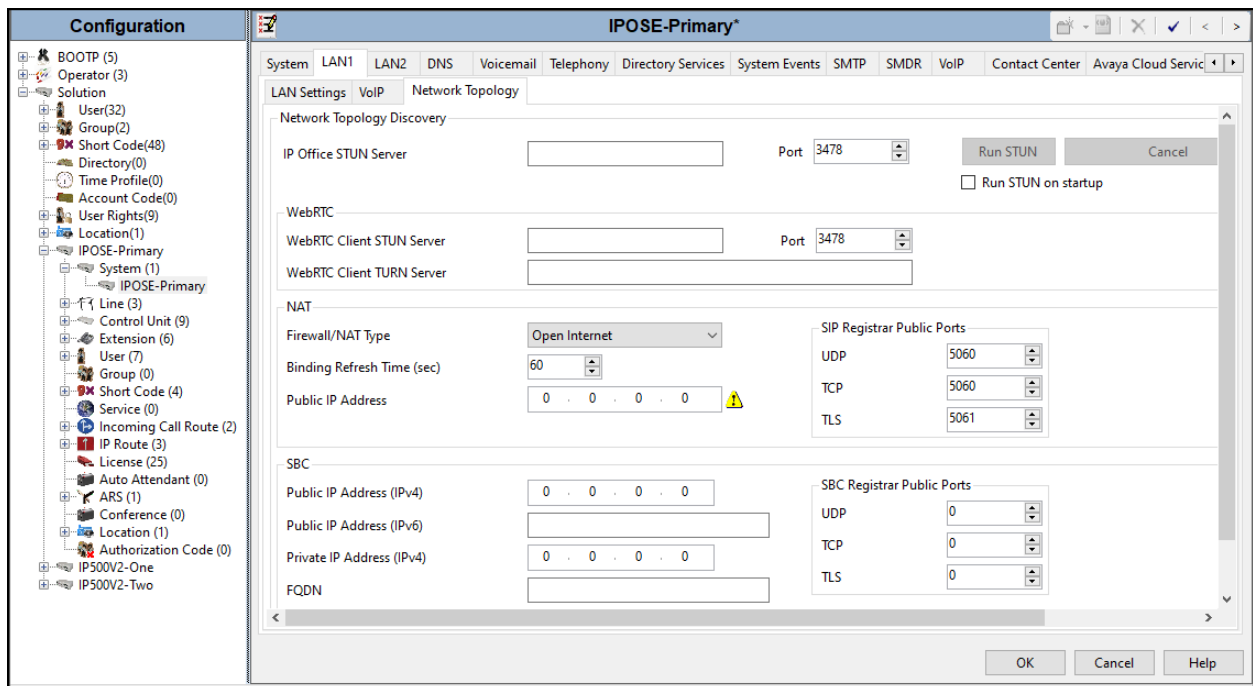
- Verify the **RTP Port Number Range** settings for a specific range for the RTP traffic. The **Port Range (Minimum)** and **Port Range (Maximum)** values were kept as default.
- In the **Keepalives** section at the bottom of the page, set the **Scope** field to **RTP-RTCP**, **Periodic Timeout** to **30**, and **Initial keepalives** to **Enabled**. This will cause the IP Office to send RTP and RTCP keepalive packets at the beginning of the calls and every 30 seconds thereafter if no other RTP/RTCP traffic is present.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit.



5.2.1.3 LAN1 Network Topology tab

The **Network Topology** tab as shown in the screenshot below was configured with following settings:

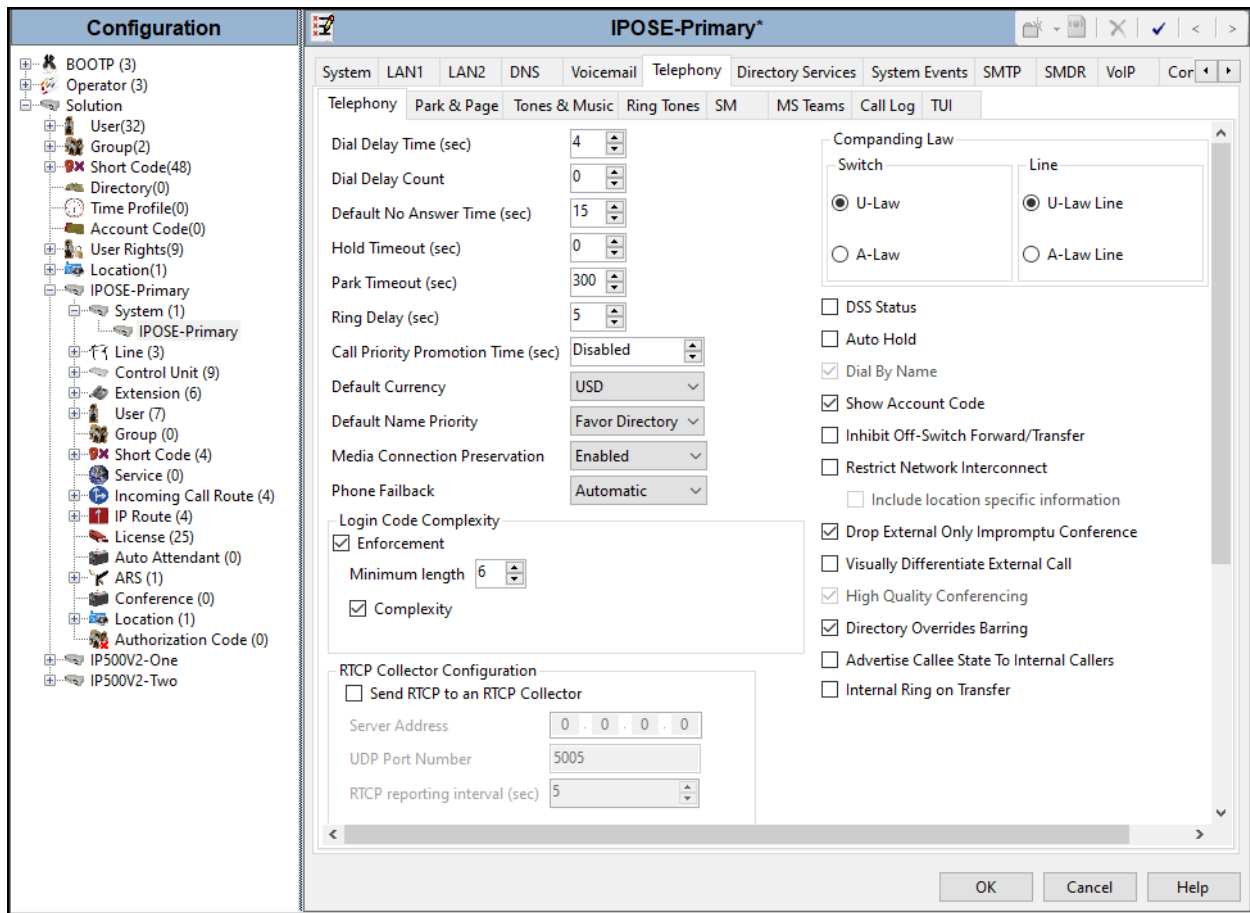
- The **Firewall/NAT Type** was set to **Open Internet** in the reference configuration.
- The **Binding Refresh Time (sec)** was set to **60** seconds. This is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages, to periodically check the status of the SIP lines configured on this interface.
- The **Public IP Address** and **Public Port** sections are not used.
- Click **OK** to commit.



5.2.2. System - Telephony Tab

To access the System Telephony settings, navigate to the **Telephony** → **Telephony** tab in the **Details** pane, configure the following parameters:

- Choose the **Companding Law** typical for the enterprise location; **U-Law** was used for the compliance test.
- Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN. If for security reasons incoming calls should not be allowed to transfer back to the PSTN then leave this setting checked.
- All other parameters should be set to default or according to customer requirements.
- Click **OK** to commit.



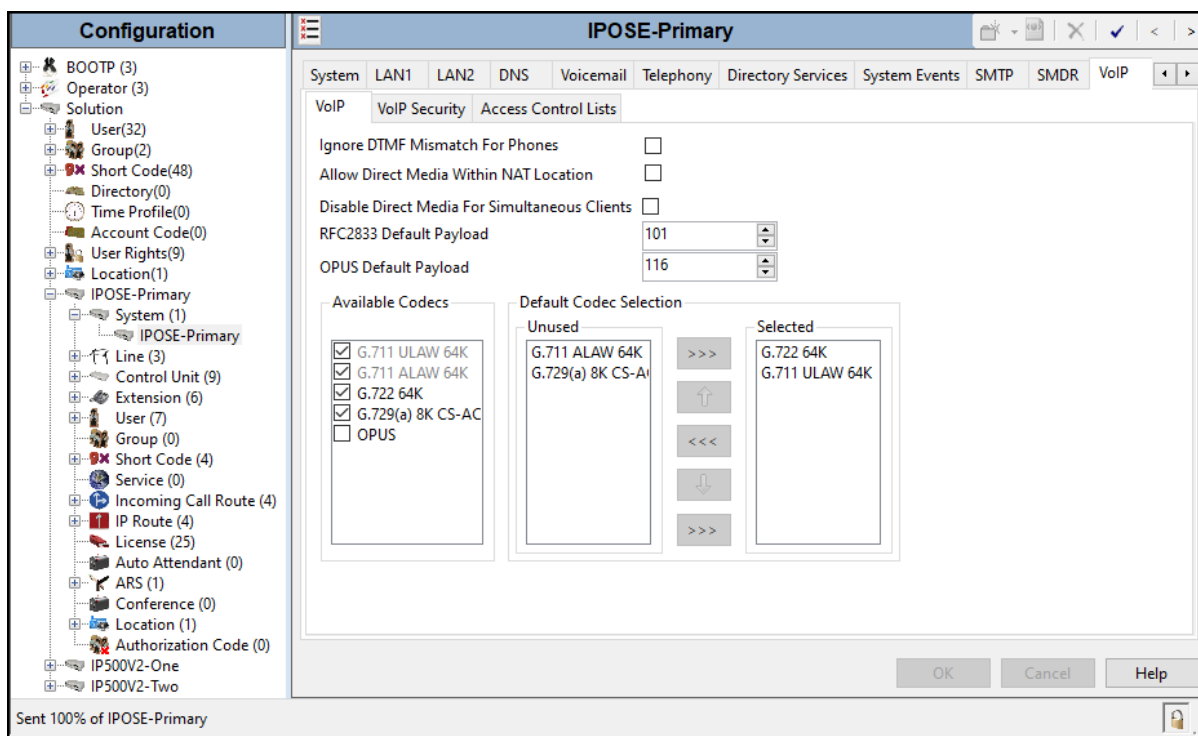
5.2.3. System - VoIP Tab

Navigate to the **VoIP** tab in the Details pane to view or change the system codecs and VoIP security settings.

5.2.3.1 VoIP - VoIP Tab

Select the **VoIP → VoIP** tab, configure the following parameters:

- The **RFC2833 Default Payload** field allows for the manual configuration of the payload type used on SIP calls that are initiated by the IP Office. The default value **101** was used.
- For codec selection, select the codecs and codec order of preference on the right, under the **Selected** column. 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 lines and phones (SIP and H.323) will use the system codec selection shown here, unless configured otherwise for a specific line or extension. The example below shows the codecs used for IP phones (SIP and H.323). Testing was performed with codecs G.722 and G.711MU as requested by WorldNet, with G.722 being the preferred codec.
- Click **OK** to commit.



Note: The codec selections defined under this section (VoIP – VoIP tab) are the codecs selected for the IP phones/extensions. The codec selections defined under **Section 5.4.5** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

5.2.3.2 VoIP – VoIP Security Tab

Secure Real-Time Transport Protocol (SRTP) refers to the application of additional encryption and or authentication to VoIP calls (SIP and H.323). SRTP can be applied between telephones, between ends of an IP trunk or in various other combinations.

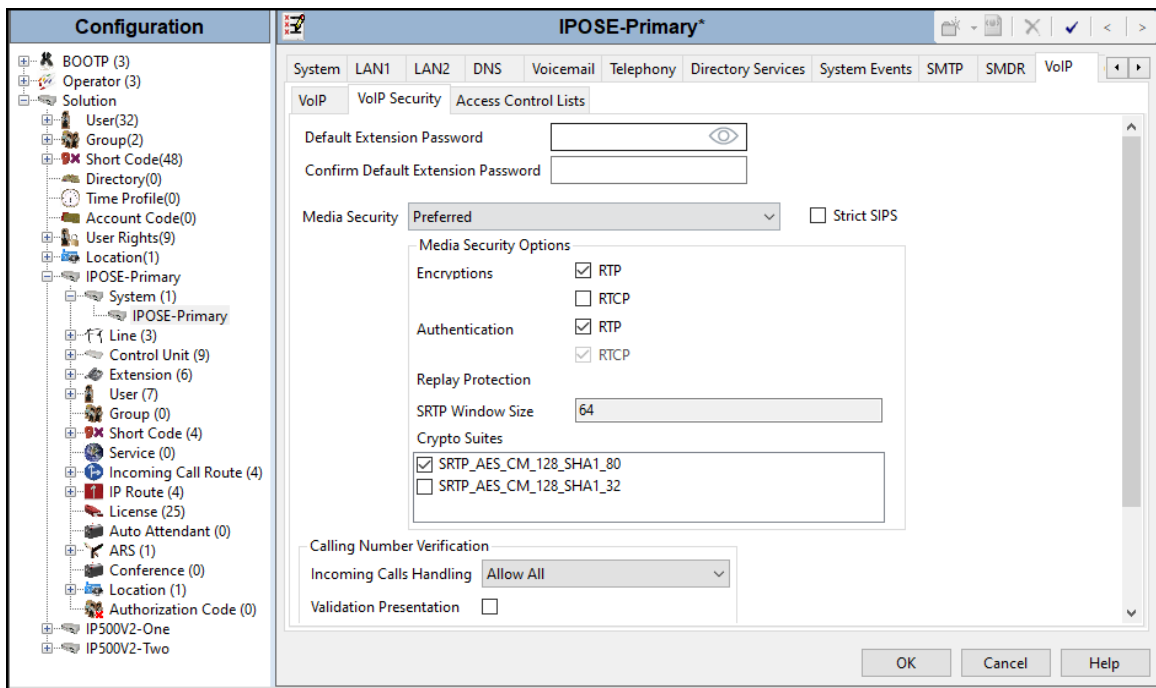
Configuring the use of SRTP at the system level is done on the **VoIP Security** tab using the Media Security setting. The options are:

- Disabled (default).
- Preferred.
- Enforced.

When enabling SRTP on the system, the recommended setting is **Preferred**. In this scenario, IP Office uses SRTP if supported by the far-end, otherwise uses RTP. If the **Enforced** setting is used, and SRTP is not supported by the far-end, the call is not established (refer to **Section 2.2** for one-way audio issue involving Avaya Workplace client for Windows softphones).

To configure the use of SRTP, select the **VoIP → VoIP Security** tab on the Details pane.

- Set the **Media Security** drop-down menu to **Preferred** to have IP Office attempt use encrypted RTP for devices that support it and fall back to RTP for devices that do not support encryption.
- Verify **Strict SIPS** is not checked.
- Under **Media Security Options**, select **RTP** for the **Encryptions** and **Authentication** fields.
- Under **Crypto Suites**, select **SRTP_AES_CM_128_SHA1_80**.
- Click **OK** to commit.

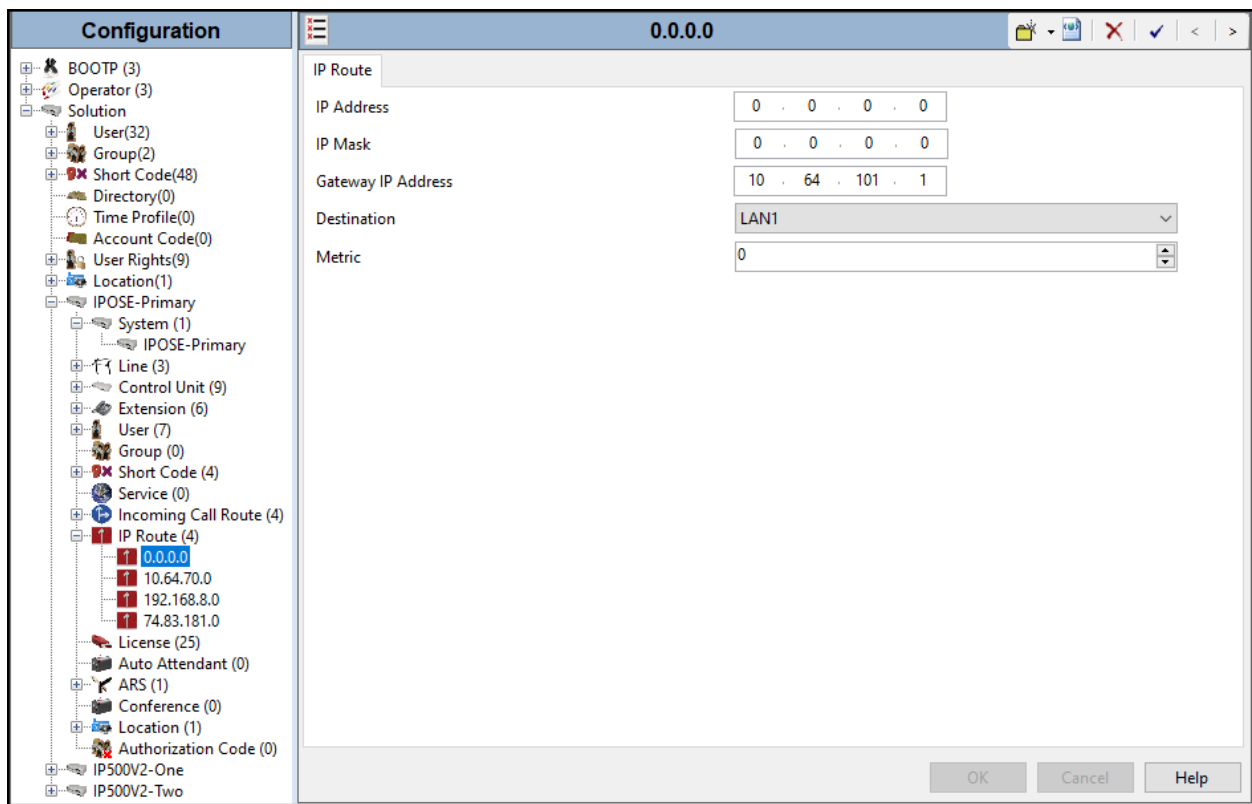


5.3. IP Route

Create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets in order to route calls to WorldNet Telecommunications network.

Navigate to **IP Route**, right-click on **IP Route** and select **New**. The values used during the compliance test are shown below:

- Set the **IP Address** and **IP Mask** to **0.0.0.0** to make this the default route.
- Set **Gateway IP Address** to the IP address of the gateway/router used to route calls to the public network, e.g., **10.64.101.1**.
- Set **Destination** to **LAN1** from the pull-down menu.
- Click **OK** to commit.



5.4. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and WorldNet Telecommunications. 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 **Sections 5.4.1** to create the SIP Line from the template.

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

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries

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 **Section 5.4.2** to **5.4.6**.

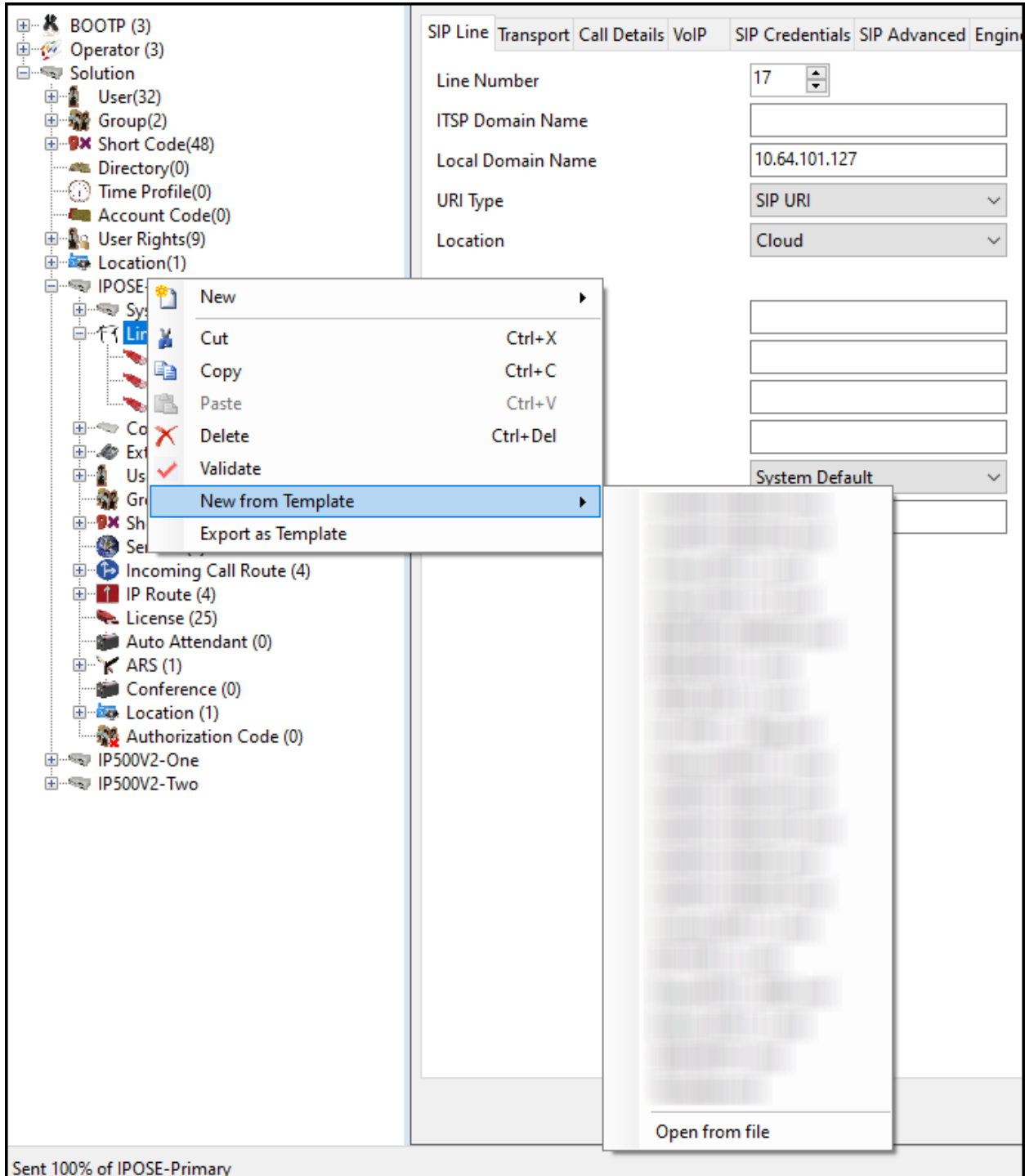
Alternatively, a SIP Line can be created manually. To do so, right-click on **Line** in the **Navigation** pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.2** to **5.4.6**.

5.4.1. Creating a SIP Trunk from an XML Template

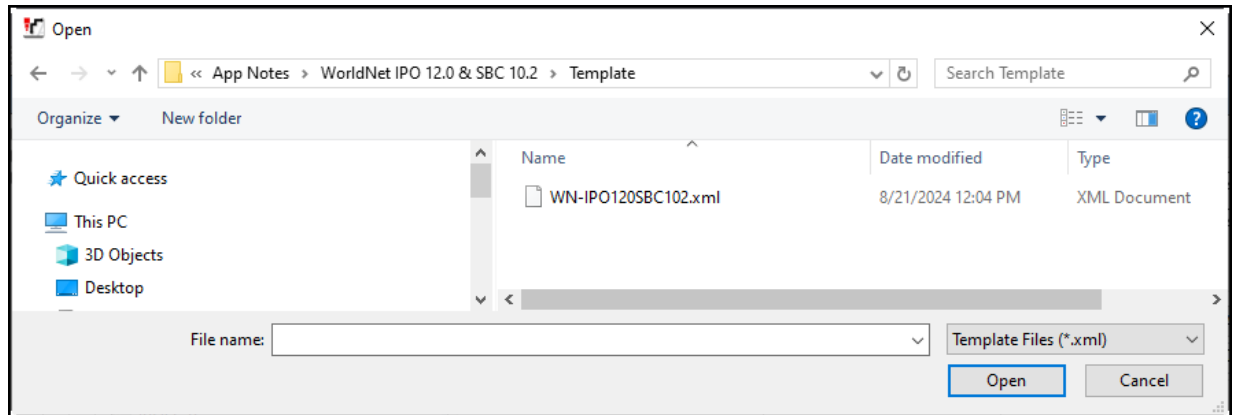
DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and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

Copy a previously created template file to a location (e.g., \Temp) on the same computer where IP Office Manager is installed.

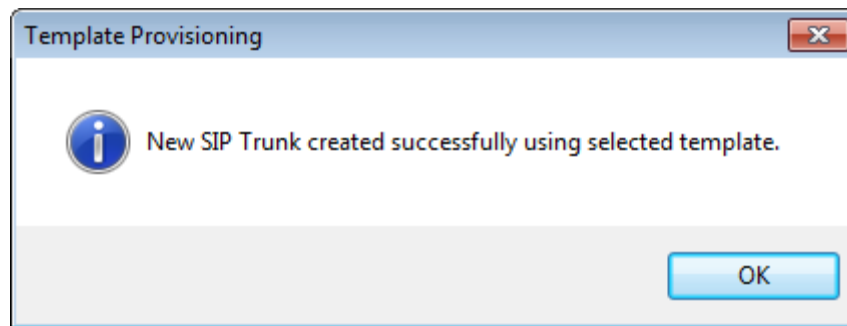
To create the SIP Trunk from the template, from the **Primary** server, right-click on **Line** in the Navigation Pane, then navigate to **New → New from Template → Open from file**.



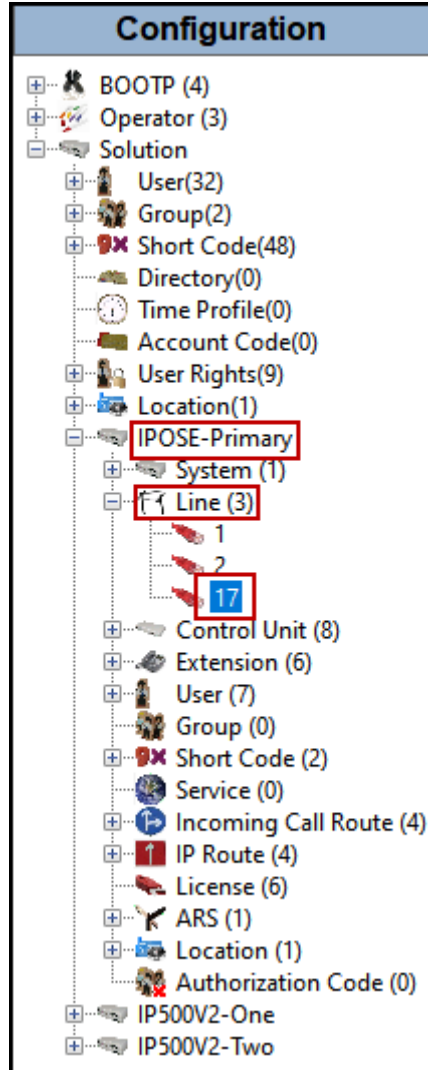
Navigate to the directory on the local machine where the template was copied and select the template.



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



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line 17).



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 5.4.2 to 5.4.6**.

5.4.2. SIP Line – 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** blank. Note that if this field is left blank, then IP Office inserts the ITSP Proxy Address from the Transport tab as the ITSP Domain in the SIP messaging.
- **Local Domain Name** is set to the IP address of the Avaya IP Office LAN1 interface (e.g., **10.64.101.127**).
- Verify that **In Service** box is checked, the default value. This makes the trunk available to incoming and outgoing calls.
- Verify that **Check OOS** box is checked, the default value. IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Verify that **Refresh Method** is set to **Auto**.
- Verify that **Timer (sec)** is set to **On Demand**.
- Under **Redirect and Transfer**, set **Incoming Supervised REFER** and **Outgoing Supervised REFER** to **Always**.
- Click **OK** to commit.

The screenshot displays the configuration window for 'SIP Line - Line 17'. The left pane shows a hierarchical tree of configuration objects, with 'Line 3' expanded to show 'Line 17'. The right pane contains the configuration fields for the selected line. The 'SIP Line' tab is active, showing the following settings:

Field	Value
Line Number	17
ITSP Domain Name	
Local Domain Name	10.64.101.127
URI Type	SIP URI
Location	Cloud
Prefix	
National Prefix	
International Prefix	
Country Code	
Name Priority	System Default
Description	Service Provider

Additional settings and checkboxes:

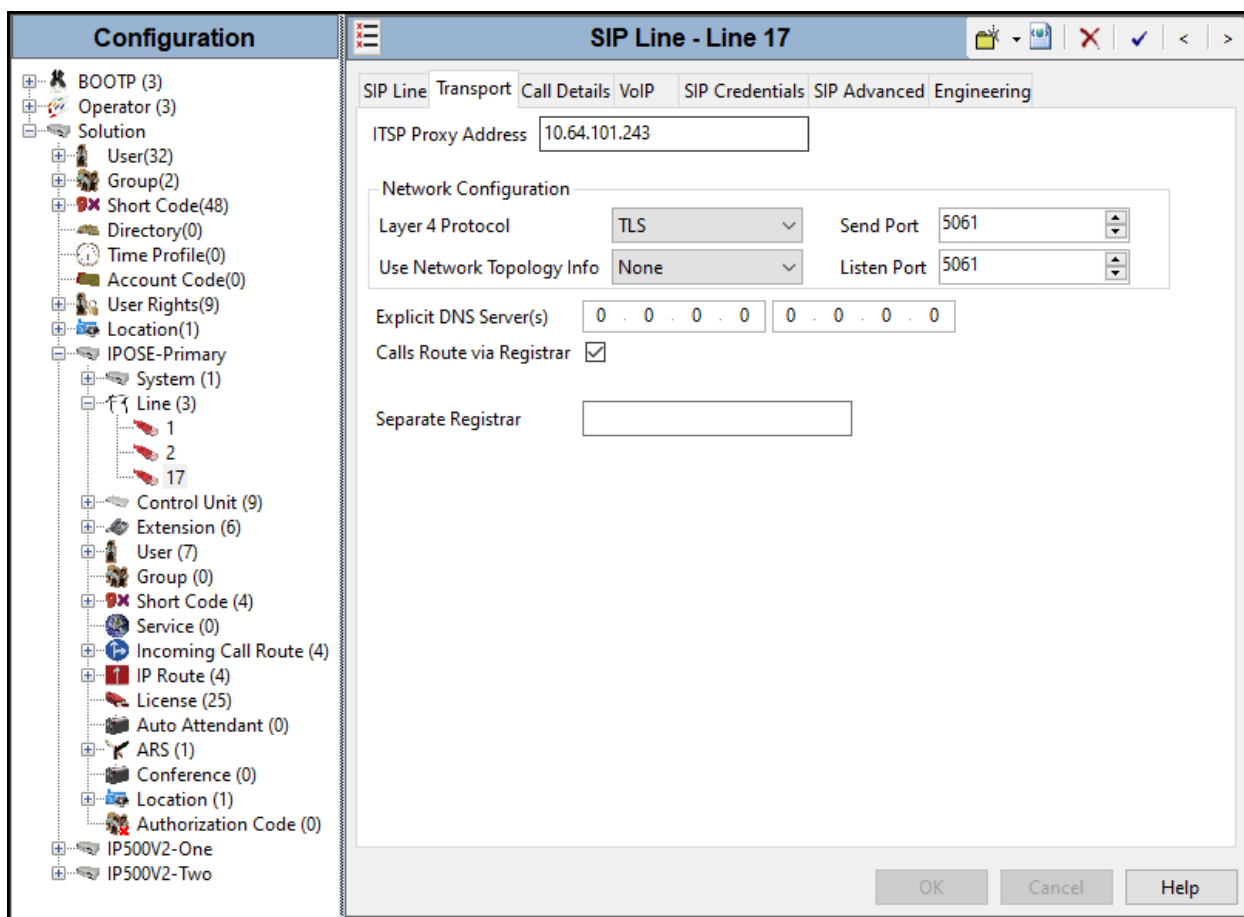
- In Service:**
- Check OOS:**
- Session Timers:**
 - Refresh Method:** Auto
 - Timer (sec):** On Demand
- Redirect and Transfer:**
 - Incoming Supervised REFER:** Always
 - Outgoing Supervised REFER:** Always
 - Send 302 Moved Temporarily:**
 - Outgoing Blind REFER:**

The status bar at the bottom indicates 'Sent 100% of IPOSE-Primary'.

5.4.3. SIP Line - Transport Tab

Select the **Transport** tab. Set or verify the parameters as shown below:

- Set the **ITSP Proxy Address** to the inside IP Address of the Avaya SBC or **10.64.101.243** as shown in **Figure 1**.
- Set **Layer 4 Protocol** to **TLS**.
- Set **Use Network Topology Info** to **None** (see note below).
- Set the **Send Port** to **5061**.
- Default values may be used for all other parameters.
- Click **OK** to commit.



Note – For the compliance testing, the **Use Network Topology Info** field was set to **None**, since no NAT was used in the test configuration. In addition, it was not necessary to configure the **System → LAN1 → Network Topology** tab for the purposes of SIP trunking. If a NAT is used between Avaya IP Office and the other end of the trunk, then the **Use Network Topology Info** field should be set to the LAN interface (LAN1) used by the trunk and the **System → LAN1 → Network Topology** tab needs to be configured with the details of the NAT device.

5.4.4. SIP Line – Call Details Tab

Select the **Call Details** tab, and then click the **Add...** button (not shown) and the screen shown below will appear. To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button. In the example screen below a new entry was created with the parameters shown below:

- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **17** was defined that only contains this line (line 17).
- Under **Credentials**, select **0: <None>** from the pull-down menu.
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Verify **P Asserted ID** and **Diversion Header** are checked.
- Set the **Local URI**, **Contact**, **P Asserted ID** and **Diversion Header** fields to the values shown in the screenshot below.
- Set all remaining fields as shown on the screenshot below.
- Click **OK**.

The screenshot shows a configuration window titled "SIP Line - 17 | Call Details | SIP URI". The window contains the following fields and options:

- New URI** section:
 - Incoming Group: 17
 - Max Sessions: 10
 - Outgoing Group: 17
 - Credentials: 0: <None>
- Display** and **Content** columns:
 - Local URI: Auto
 - Contact: Auto
 - P Asserted ID: Auto
 - P Preferred ID: None
 - Diversion Header: Auto
 - Remote Party ID: None
- Field meaning** table:

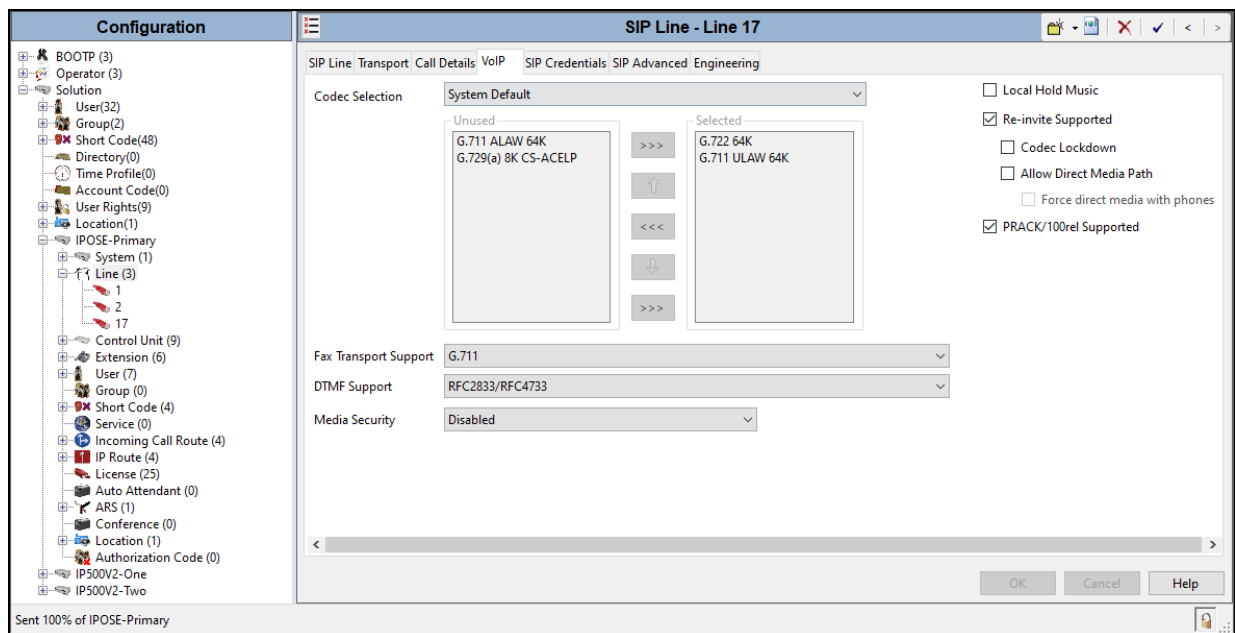
Outgoing Calls		Forwarding/Twinning		Incoming Calls	
Caller	Original Caller	Original Caller	Original Caller	Called	Called
Caller	Original Caller	Original Caller	Original Caller	Called	Called
Caller	Original Caller	Original Caller	Original Caller	Called	Called
None	None	None	None	None	None
None	None	Caller	Caller	None	None
None	None	None	None	None	None

Buttons at the bottom: OK, Cancel, Help.

5.4.5. SIP Line - VoIP Tab

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

- The **Codec Selection** was configured using the **System Default** option, allowing the same codec order used under **System – VoIP** (refer to **Section 5.2.3.1**). Testing was performed with codecs G.722 64K and G.711MU, with G.722 64K being the preferred codec.
- Select **G.711** for **Fax Transport Support** (refer to **Section 2.1**).
- Set the **DTMF Support** field to **RFC2833/RFC4733**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Set the **Media Security** field to **Disabled**.
- Check the **Re-invite Supported** box.
- Check the **PRACK/100rel Supported** box.
- Default values may be used for all other parameters.
- Click the **OK** to commit.



Note: The codec selections defined under this section are the codecs selected for the SIP Line (Trunk). The codec selections defined under **Section 5.2.3** are the codecs selected for the IP phones/extension (H.323 and SIP).

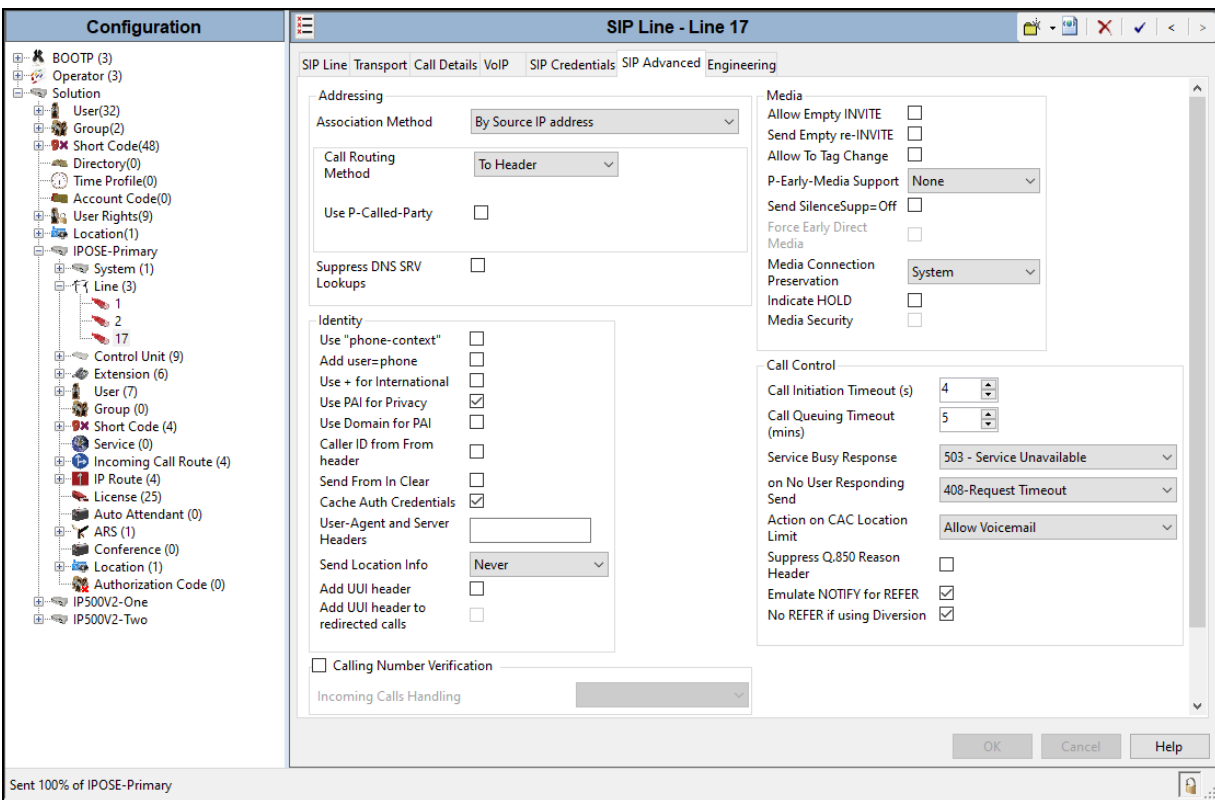
5.4.6. SIP Line – SIP Advanced Tab

In the **Addressing** area:

- Select **To Header** for **Call Routing Method**.

In the **Identity** area:

- Check the box for **Use PAI for Privacy**.
- Under Call Control, check **Emulate NOTIFY for REFER** and **No REFER if using Diversion** (refer to **Section 2.2**).
- Default values may be used for all other parameters.
- Click **OK** to commit.



5.5. IP Office Line – Primary Server

In IP Office Server Edition systems, IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate line to be configured in the Group pane. The screen below shows the IP Office Line to the IP500V2-One Expansion System.

The screenshot displays the configuration interface for an IP Office Line. On the left is a navigation tree under the 'Configuration' header, showing a hierarchy from Solution down to IP500V2-One and IP500V2-Two. The main area is titled 'IP Office Line - Line 1' and contains several tabs: 'Line', 'Short Codes', and 'VoIP Settings'. The 'Line' tab is active, showing the following fields:

- Line Number: 1
- Transport Type: WebSocket Server
- Networking Level: SCN
- Security: Unsecured
- Telephone Number: (empty)
- Prefix: (empty)
- Outgoing Group ID: 99999
- Number of Channels: 250
- Outgoing Channels: 250

Below these fields is a 'Gateway' section with the following fields:

- Address: 192 . 168 . 8 . 165
- Location: 3: Thornton, CO
- Password: (masked with dots)
- Confirm Password: (masked with dots)

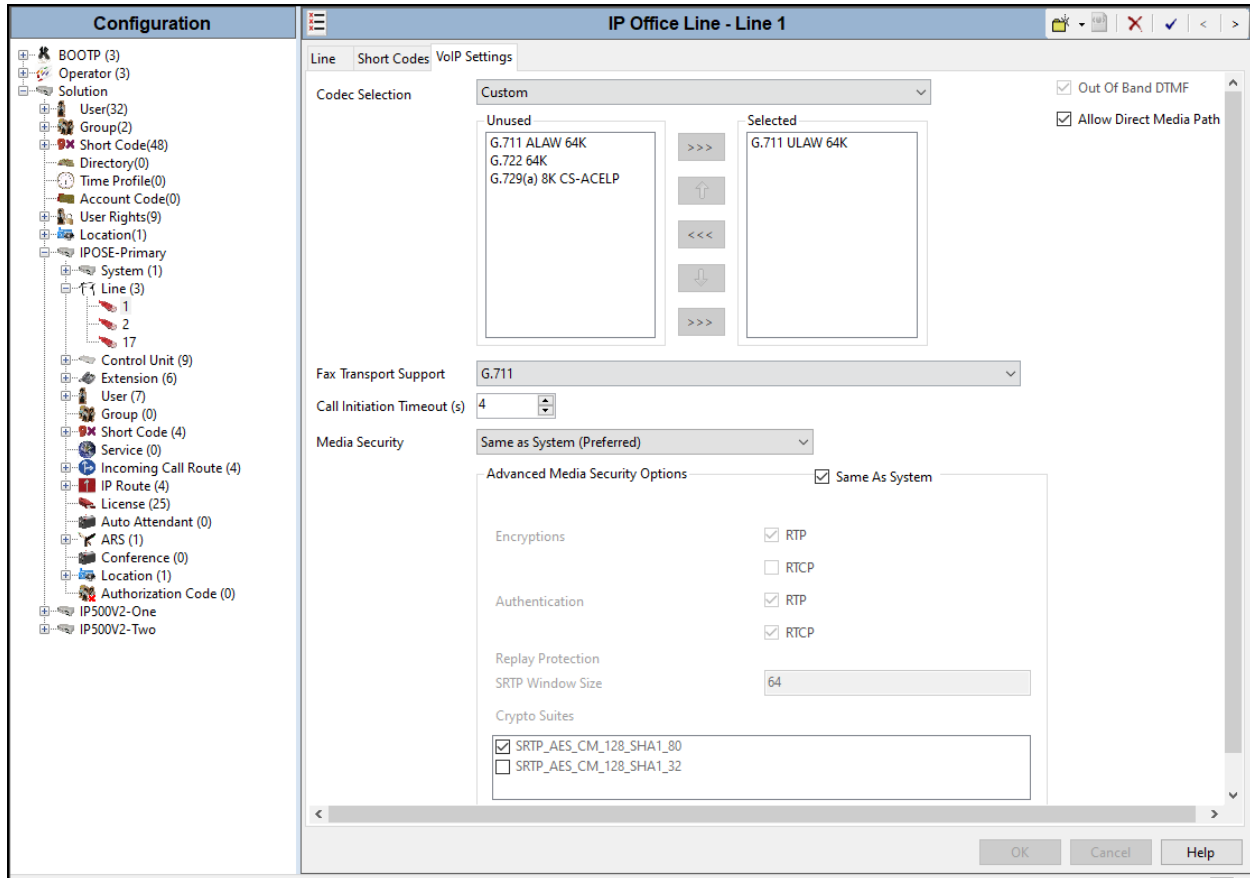
To the right of the Gateway fields is the 'SCN Resiliency Options' section, which includes a checkbox for 'Supports Resiliency' and three sub-options, all of which are currently unchecked:

- Supports Resiliency
 - Backs up my IP phones
 - Backs up my hunt groups
 - Backs up my voicemail
 - Backs up my IP DECT phones

At the bottom of the window is a 'Description' field (empty) and three buttons: 'OK', 'Cancel', and 'Help'.

The screen below shows the IP Office Line, **VoIP Settings** tab:

- The **Codec Selection** was configured using the **Custom** option, allowing an explicit order of codecs to be specified for the SIP Line. The buttons allow setting the specific order of preference for the codecs to be used on the SIP Line, as shown. Only codec **G.711ULAW 64K** was selected for the IP500V2 expansion systems (G.722 64K was not included), this was required in order for G.711 pass-through fax to work properly.
- Select **G.711** for **Fax Transport Support** (refer to **Section 2.1**).
- Under **Media Security** verify **Same as System (Preferred)** is selected (default value).
- On the **Advanced Media Security Options** check **Same As System**.



Repeat this process as needed to add additional Secondary server or Expansion Systems to the solution.

5.6. 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. To add an incoming call route, right click on **Incoming Call Route** in the **Navigation** pane and select **New** (not shown). On the Details Pane, under the **Standard** tab, set the parameters as show below:

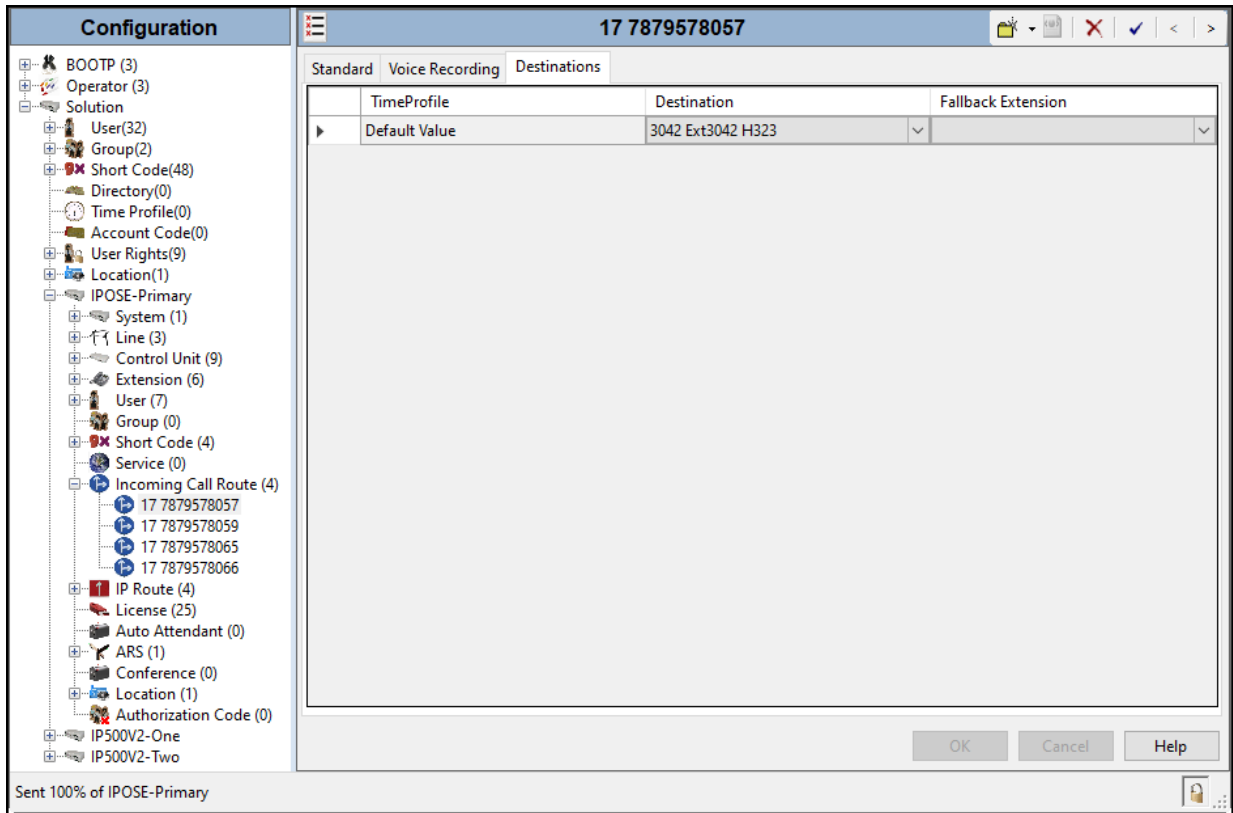
- Set **Bearer Capacity** to **Any Voice**.
- The **Line Group ID** is set to **17**. This matches the **Incoming Group** field configured in the **Call Details** tab for the SIP Line on **Section 5.4.4**.
- On the **Incoming Number**, enter one of the DID numbers provided by WorldNet.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).

The screenshot displays the Avaya IP Office configuration window for an Incoming Call Route. The left pane shows a tree view of the system configuration, with 'Incoming Call Route (4)' expanded to show the specific route '17 7879578057'. The right pane shows the configuration details for this route, with the 'Standard' tab selected. The configuration parameters are as follows:

Parameter	Value
Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	7879578057
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

At the bottom of the configuration pane, there are buttons for 'OK', 'Cancel', and 'Help'. The status bar at the bottom left indicates 'Sent 100% of IPOSE-Primary'.

Select the **Destinations** tab. From the **Destination** drop-down menu, select the IP Office extension associated with this DID number. In the reference configuration, the DID number 7879578057 provided by WorldNet was associated with the Avaya IP Office extension **3042**.



Repeat this process as needed to assign incoming call routes to additional IP Office users, as well as for other Avaya IP Office destinations (Hunt Group, Voicemail, Short Codes, etc.).

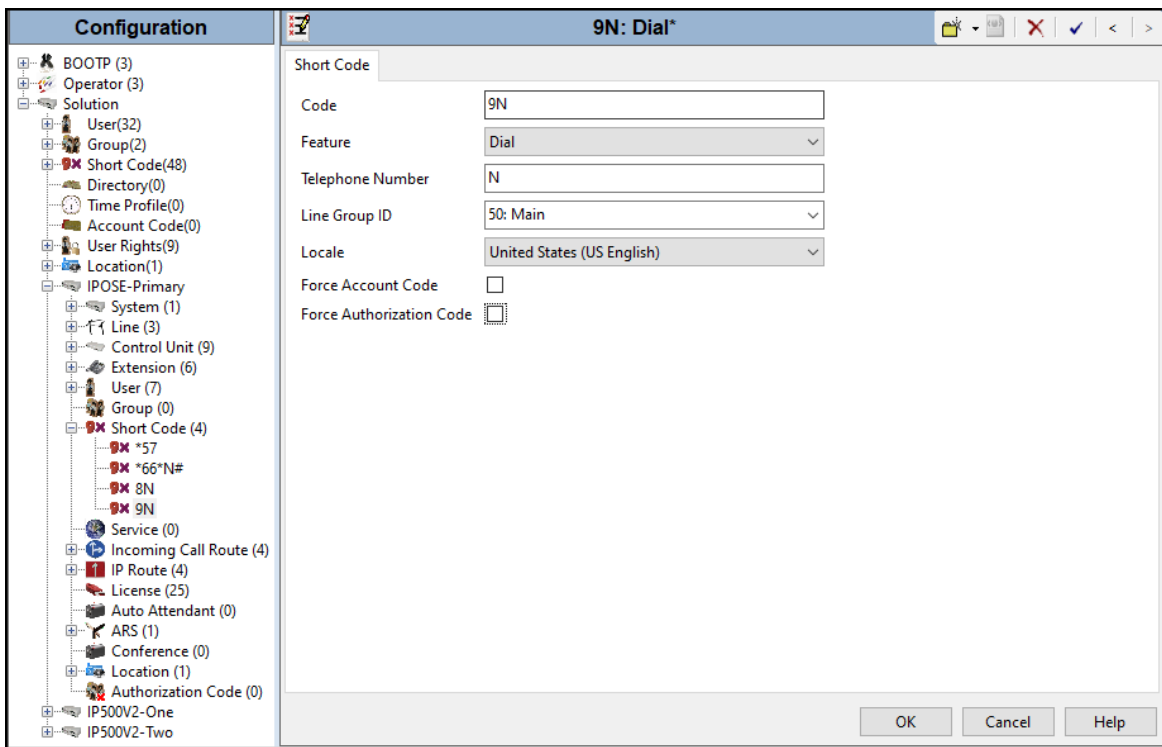
5.7. Outbound Call Routing

For outbound call routing, a combination of system short codes and Automatic Route Selection (ARS) entries are used. With ARS, features like time-based routing criteria and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. While detailed coverage of ARS is beyond the scope of these Application Notes, and alternate routing was not used in the reference configuration, this section includes some basic screen illustrations of the ARS settings used during the compliance testing.

5.7.1. Short Codes and Automatic Route Selection

To create a short code to be used for ARS, right-click on **Short Code**, the **Navigation** pane and select **New**. The screen below shows the short code **9N** created (note that the semi-colon is not used here). In this case, when the IP Office user dials 9 plus any number **N**, instead of being directed to a specific Line Group ID, the call is directed to **Line Group 50: Main**, which is configurable via ARS.

- In the **Code** field, enter the dial string which will trigger this short code. In this case, **9N** was used (note that the semi-colon is not used here).
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N**. The value **N** represents the number dialed by the user after removing the **9** prefix. This value is passed to ARS.
- Set the **Line Group ID** to **50: Main** to be directed to **Line Group 50: Main**, this is configurable via ARS.
- For **Locale**, **United States (US English)** was used.
- Click the **OK** to commit.

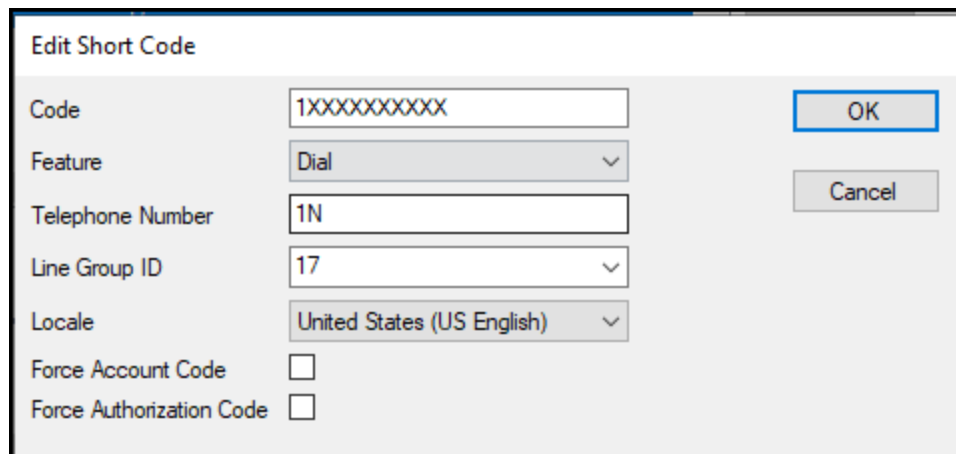


The following screen shows the example ARS configuration for the route **Main**. Note the sequence of **Xs** used in the **Code** column of the entries to specify the exact number of digits to be expected, following the access code and the first set of digits on the string. This type of setting results in a much quicker response in the delivery of the call by IP Office.

To create a short code to be used for ARS, select **ARS → 50: Main** on the Navigation Pane and click **Add** (not shown). Configure the following parameters:

- In the **Code** field, enter the dial string which will trigger this short code. In this case, **1** followed by **10 Xs** to represent the exact number of digits.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **1N**. The value **N** represents the additional number of digits dialed by the user after dialing **1** (The **9** will be stripped off).
- Set the **Line Group ID** to the Line Group number being used for the SIP Line, in this case **Line Group ID 17** was used.
- For **Locale**, **United States (US English)** was used.
- Click **OK** to commit.

The following example shows the dial pattern for calls within Puerto Rico and calls to the United States.



Edit Short Code	
Code	1XXXXXXXXXX
Feature	Dial
Telephone Number	1N
Line Group ID	17
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

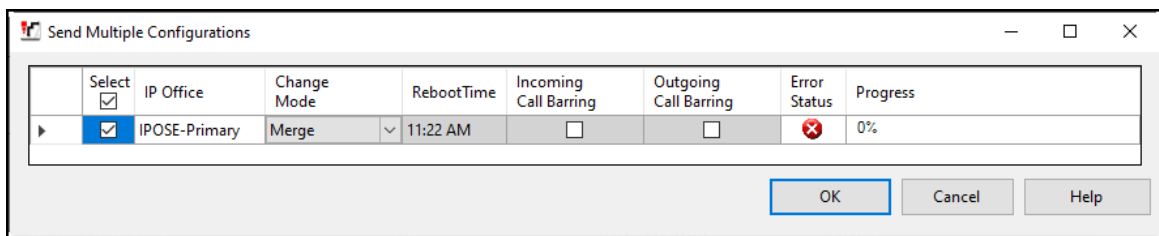
Repeat the above procedure for additional dial patterns to be used by the enterprise to dial out from IP Office.

5.8. Save IP Office Primary Server Configuration

The provisioning changes made in Avaya IP Office Manager must be applied to the Avaya IP Office server in order for the changes to take effect. At the top of the Avaya IP Office Manager page, click **File** → **Save Configuration** (if that option is grayed out, no changes are pending).

A screen similar to the one below will appear, with either **Merge** or **Reboot** automatically selected, based on the nature of the configuration changes. The **Merge** option will save the configuration change with no impact to the current system operation. The **Reboot** option will save the configuration and cause the Avaya IP Office server to reboot.

Click **OK** to execute the save.



6. Avaya IP Office Expansion System Configuration

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. Clicking the “plus” sign next to **IP500V2-One** on the left navigation pane will expand the menu on this server.

The screenshot displays the Avaya IP Office configuration interface, divided into two main panes: **Configuration** and **System Inventory**.

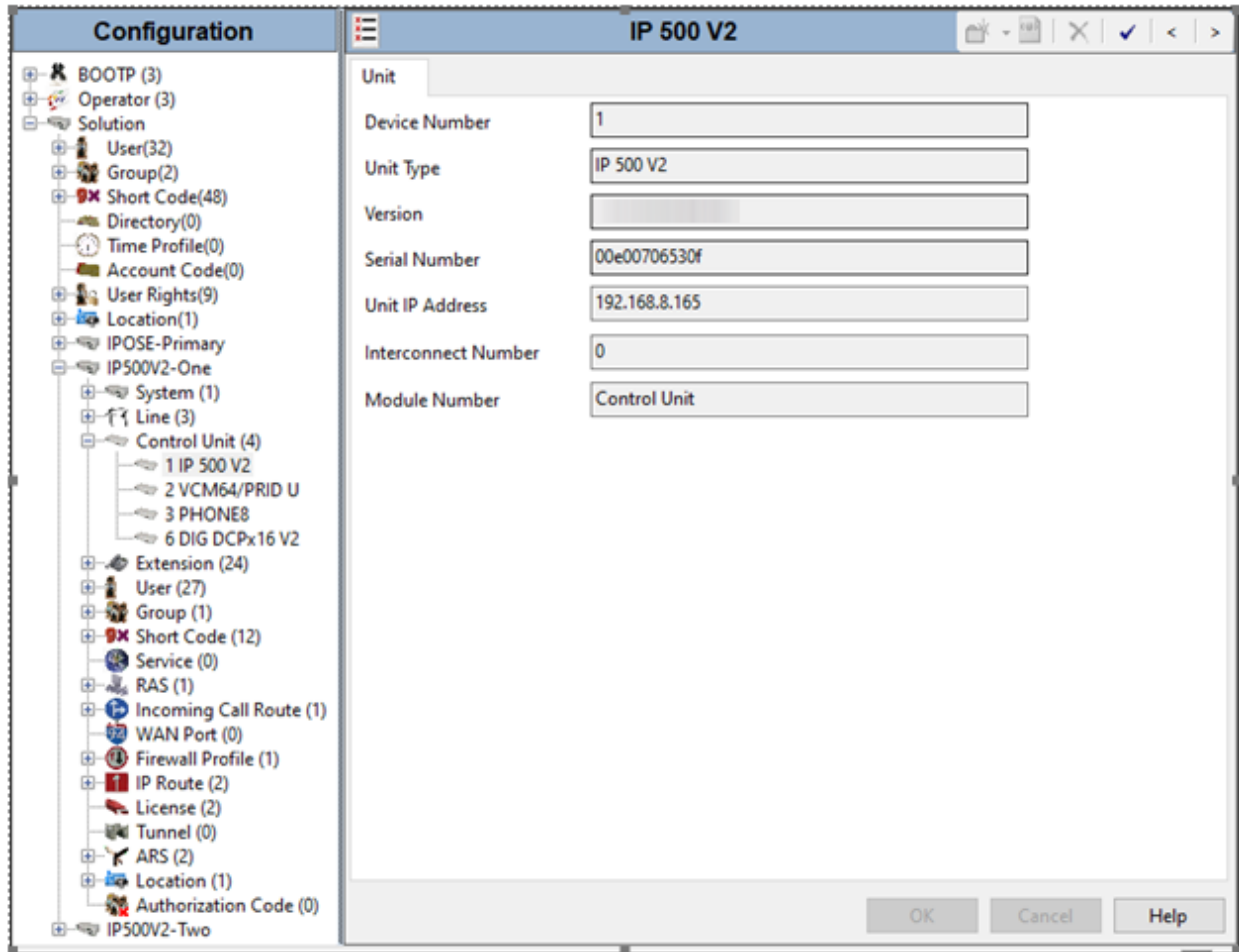
Configuration Pane: This pane shows a hierarchical tree structure of system components. The **IP500V2-One** system is selected and expanded, revealing its sub-components: System (1), Line (3), Control Unit (4), Extension (24), User (27), Group (1), Short Code (12), Service (0), RAS (1), Incoming Call Route (1), WAN Port (0), Firewall Profile (1), IP Route (4), License (2), Tunnel (0), ARS (2), Location (1), and Authorization Code (0). The **IP500V2-One** item is highlighted in blue.

System Inventory Pane: This pane displays the configuration details for the selected system, titled **Server Edition Expansion System**. It is organized into three sections:

- Hardware Installed:**
 - Control Unit: IP 500 V2
 - Internal Modules: VCM64/PRID U; PHONE8
 - Expansion Modules: DIG DCPx16 V2
- System Settings:**
 - IP Address: 192.168.8.165
 - Sub-Net Mask: 255.255.255.0
 - System Locale: United States (US English)
 - System Location: 3: Thornton, CO
 - Device ID: NONE
 - Number of Extensions on System: 24
- Features Configured:**
 - Licenses Installed: Server Edition(1); IP Office Select(1); Basic User(25)
 - Connected Extensions: 3043; 3044
 - Users NOT Configured for Voicemail: NONE
 - Users assigned as Ex-Directory: NONE
 - Users assigned for Twinning: NONE
 - Users barred from making Outgoing Calls: NONE
 - Music on Hold: WAV File

6.1. Physical Hardware

In the sample configuration, the IP500 V2 Expansion System contained a PHONE8 analog card, for the support of analog extensions, a DIG DCPx16 V2, for support of digital extensions. Also included is a VCM64 (Voice Compression Module). The VCM64 cards provide voice compression channels to the control unit. Voice compression channels are needed to support VoIP calls, including IP extensions and or IP trunks.



6.2. LAN Settings

In the sample configuration, LAN1 is used to connect the Expansion System to the enterprise network. To view or configure the LAN1 IP address, select **System** on the Navigation pane. Select the **LAN1 → LAN Settings** tab on the Details pane, and enter the following:

- **IP Address: 192.168.8.165** was used in the reference configuration.
- **IP Mask: 255.255.255.0** was used in the reference configuration.
- Click the **OK** button (not shown).

The screenshot displays the configuration interface for an IP500V2-One system. On the left is a navigation tree under 'Configuration' with 'IP500V2-One' selected. The main pane shows the 'LAN Settings' tab for 'LAN1'. The configuration fields are as follows:

Field	Value
IP Address	192 . 168 . 8 . 165
IP Mask	255 . 255 . 255 . 0
Primary Trans. IP Address	0 . 0 . 0 . 0
RIP Mode	None
Enable NAT	<input type="checkbox"/>
Number Of DHCP IP Addresses	200
DHCP Mode	<input type="radio"/> Server <input type="radio"/> Client <input type="radio"/> Dial In <input checked="" type="radio"/> Disabled

An 'Advanced' button is located at the bottom right of the configuration area.

Default values were used on the **VoIP** and **Network Topology** tabs (not shown).

6.3. IP Route

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

- Enter **0.0.0.0** on the **IP Address** and **IP Mask** fields to make this the default route.
- Set **Gateway IP Address** to the IP Address of the default router in the IP Office subnet. The default gateway in the reference configuration was **192.168.8.1**.
- Set **Destination** to **LAN1** from the pull-down menu.

The screenshot displays the Avaya IP Office configuration interface. On the left is a navigation tree under 'Configuration' for system 'IP500V2-One'. The 'IP Route' folder is expanded, showing four entries: '0.0.0.0' (highlighted in blue), '10.64.101.0', '192.168.8.0', and '192.168.99.0'. On the right, the configuration details for the selected '0.0.0.0' route are shown:

0.0.0.0	
IP Route	
IP Address	0 . 0 . 0 . 0
IP Mask	0 . 0 . 0 . 0
Gateway IP Address	192 . 168 . 8 . 1
Destination	LAN1
Metric	0
	<input type="checkbox"/> Proxy ARP

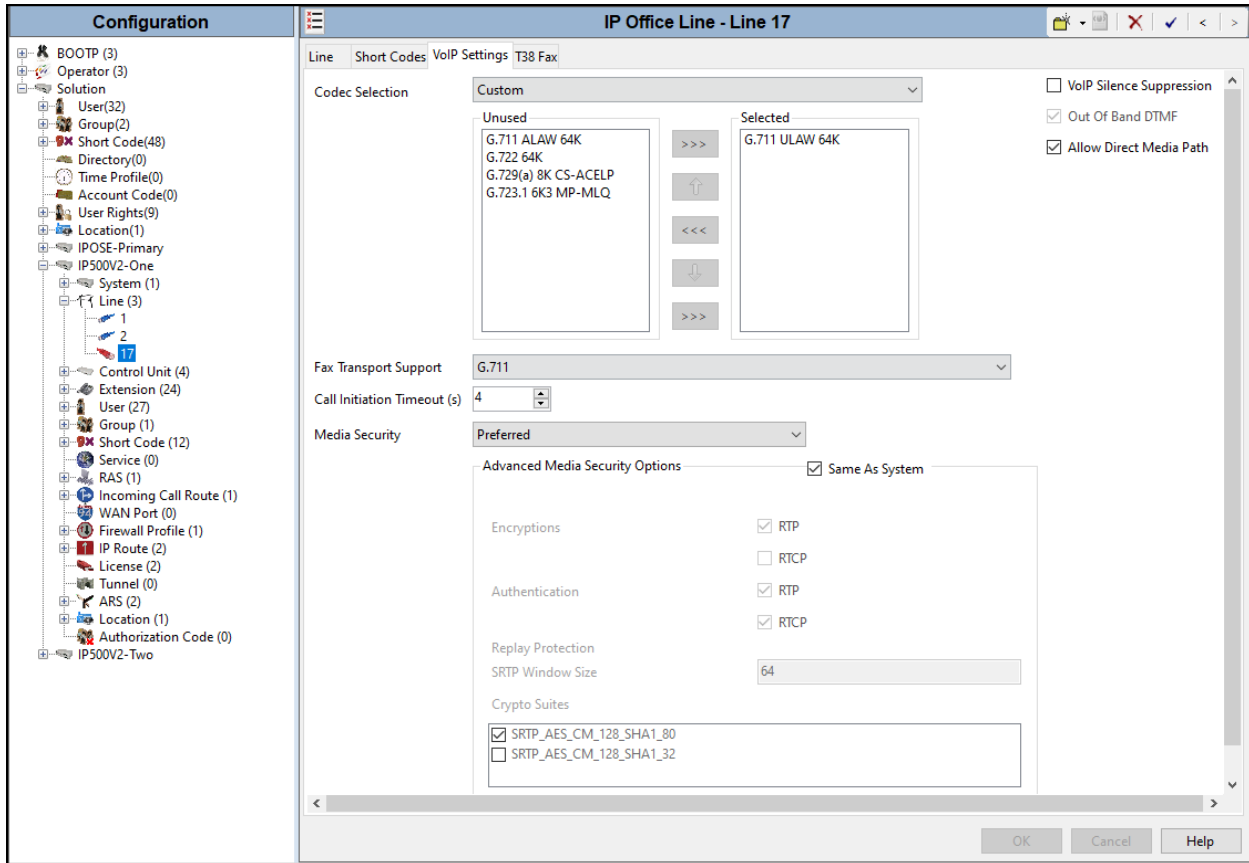
6.4. IP Office Line – IP500 V2 Expansion System

In IP Office Server Edition systems, IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate line to be configured in the Group pane. The screen below shows the IP Office Line to the Primary server.

Configuration	IP Office Line - Line 17*																																			
<ul style="list-style-type: none"> BOOTP (4) Operator (3) Solution <ul style="list-style-type: none"> User(32) Group(2) Short Code(48) Directory(0) Time Profile(0) Account Code(0) User Rights(9) Location(1) IPOSE-Primary IP500V2-One <ul style="list-style-type: none"> System (1) Line (3) <ul style="list-style-type: none"> 1 2 17 Control Unit (4) Extension (24) User (27) Group (1) Short Code (12) Service (0) RAS (1) Incoming Call Route (1) WAN Port (0) Firewall Profile (1) IP Route (4) License (2) Tunnel (0) ARS (2) Location (1) Authorization Code (0) IP500V2-Two 	<div style="border: 1px solid #ccc; padding: 5px;"> <div style="background-color: #e0e0e0; border-bottom: 1px solid #ccc; padding: 2px;">Line Short Codes VoIP Settings T38 Fax</div> <table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%;">Line Number</td> <td style="width: 35%;"><input type="text" value="17"/></td> <td style="width: 30%;">Telephone Number</td> <td><input type="text"/></td> </tr> <tr> <td>Transport Type</td> <td><input type="text" value="WebSocket Client"/></td> <td>Prefix</td> <td><input type="text"/></td> </tr> <tr> <td>Networking Level</td> <td><input type="text" value="SCN"/></td> <td>Outgoing Group ID</td> <td><input type="text" value="99999"/></td> </tr> <tr> <td>Security</td> <td><input type="text" value="Medium"/></td> <td>Number of Channels</td> <td><input type="text" value="250"/></td> </tr> <tr> <td></td> <td></td> <td>Outgoing Channels</td> <td><input type="text" value="250"/></td> </tr> </table> <div style="border: 1px solid #ccc; padding: 5px; margin-top: 5px;"> <table style="width: 100%; border-collapse: collapse;"> <tr> <td colspan="2">Gateway</td> </tr> <tr> <td style="width: 50%;">Address</td> <td style="width: 50%;"><input type="text" value="10 . 64 . 101 . 127"/></td> </tr> <tr> <td>Location</td> <td><input type="text" value="3: Thornton, CO"/></td> </tr> <tr> <td>Password</td> <td><input type="password" value="....."/></td> </tr> <tr> <td>Confirm Password</td> <td><input type="password" value="....."/></td> </tr> <tr> <td colspan="2">Port</td> </tr> <tr> <td colspan="2"><input type="text" value="443"/></td> </tr> </table> <div style="margin-top: 5px;"> <p>SCN Resiliency Options</p> <input type="checkbox"/> Supports Resiliency <input type="checkbox"/> Backs up my IP phones <input type="checkbox"/> Backs up my hunt groups <input type="checkbox"/> Backs up my IP DECT phones </div> </div> <div style="margin-top: 10px;"> <p>Description <input style="width: 150px;" type="text"/></p> </div> </div>		Line Number	<input type="text" value="17"/>	Telephone Number	<input type="text"/>	Transport Type	<input type="text" value="WebSocket Client"/>	Prefix	<input type="text"/>	Networking Level	<input type="text" value="SCN"/>	Outgoing Group ID	<input type="text" value="99999"/>	Security	<input type="text" value="Medium"/>	Number of Channels	<input type="text" value="250"/>			Outgoing Channels	<input type="text" value="250"/>	Gateway		Address	<input type="text" value="10 . 64 . 101 . 127"/>	Location	<input type="text" value="3: Thornton, CO"/>	Password	<input type="password" value="....."/>	Confirm Password	<input type="password" value="....."/>	Port		<input type="text" value="443"/>	
Line Number	<input type="text" value="17"/>	Telephone Number	<input type="text"/>																																	
Transport Type	<input type="text" value="WebSocket Client"/>	Prefix	<input type="text"/>																																	
Networking Level	<input type="text" value="SCN"/>	Outgoing Group ID	<input type="text" value="99999"/>																																	
Security	<input type="text" value="Medium"/>	Number of Channels	<input type="text" value="250"/>																																	
		Outgoing Channels	<input type="text" value="250"/>																																	
Gateway																																				
Address	<input type="text" value="10 . 64 . 101 . 127"/>																																			
Location	<input type="text" value="3: Thornton, CO"/>																																			
Password	<input type="password" value="....."/>																																			
Confirm Password	<input type="password" value="....."/>																																			
Port																																				
<input type="text" value="443"/>																																				

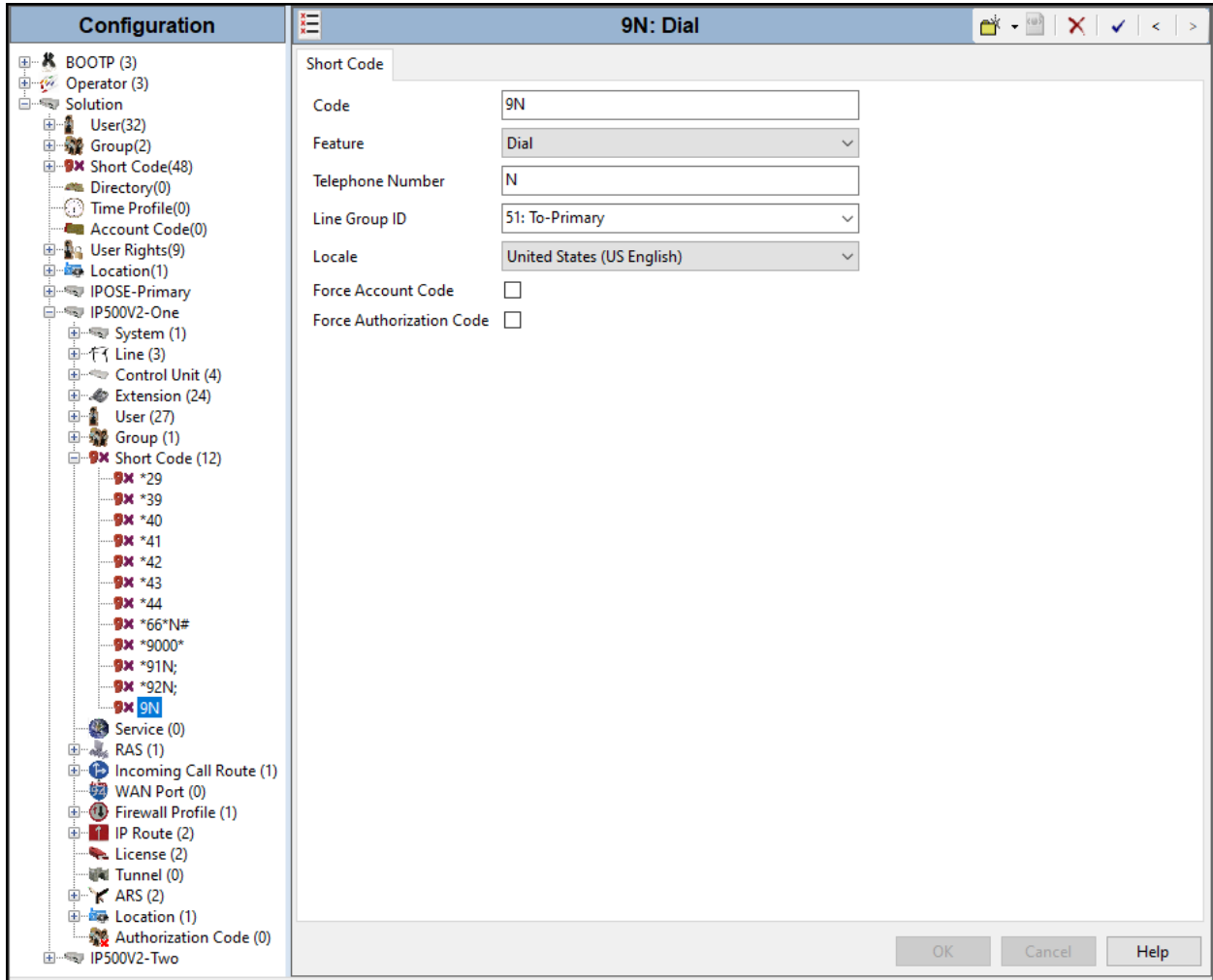
The screen below shows the IP Office Line, **VoIP Settings** tab:

- The **Codec Selection** was configured using the **Custom** option, allowing an explicit order of codecs to be specified for the SIP Line. The buttons allow setting the specific order of preference for the codecs to be used on the SIP Line, as shown. Only codec **G.711ULAW 64K** was selected for the IP500V2 expansion systems (G.722 64K was not included), this was required in order for G.711 pass-through fax to work properly.
- Select **G.711** for **Fax Transport Support** (refer to Section 2.1).
- Under **Media Security Preferred** was selected.
- On the **Advanced Media Security Options** check **Same As System**.



6.5. Short Codes

Similar to the configuration of the Primary server in **Section 5.7**, create a Short Code to access ARS. In the reference configuration, the **Line Group ID** is set to the ARS route illustrated in the next section.



6.6. Automatic Route Selection – ARS

The following screen shows an example ARS configuration for the route named “**To-Primary**” on the Expansion System. The **Telephone Number** is set to **9N**. The **Line Group ID** is set to “**99999**” matching the number of the **Outgoing Group ID** configured on the IP Office Line 17 to the Primary server (**Section 6.4**).

The screenshot displays the configuration for an ARS route named "To-Primary". The interface includes a left-hand navigation tree and a main configuration area. The configuration area is divided into several sections:

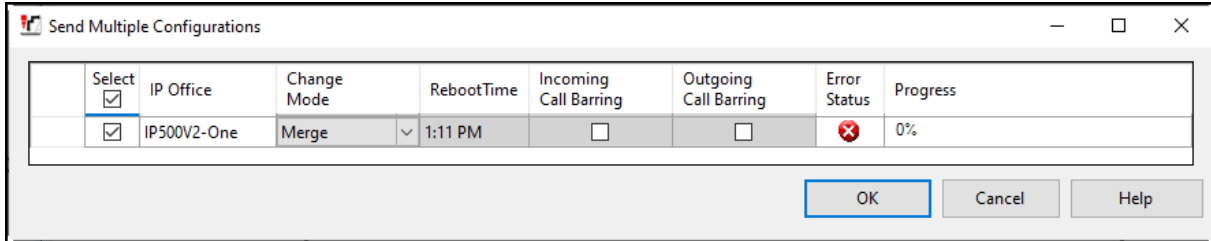
- ARS Section:** Contains fields for ARS Route ID (51), Route Name (To-Primary), Dial Delay Time (System Default (4)), and Description.
- Service and Time Profile Section:** Includes checkboxes for In Service and Secondary Dial tone, and dropdown menus for Out of Service Route, Out of Hours Route, and Check User Call Barring.
- Route Table:** A table with columns for Code, Telephone Number, Feature, and Line Group ID. It contains one entry: Code N, Telephone Number 9N, Feature Dial, and Line Group ID 99999. Action buttons (Add..., Remove, Edit...) are visible to the right of the table.
- Alternate Route Section:** Includes fields for Alternate Route Priority Level (3) and Alternate Route Wait Time (30), with a dropdown for Alternate Route (None).

Repeat the process described in **Section 6** on any additional Secondary server or Expansion Systems in the solution, as required.

6.7. Save IP Office Expansion System Configuration

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

The following will appear, with either **Merge** or **Reboot** 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.



7. Configure Avaya Session Border Controller

This section describes the required configuration of the Avaya SBC to connect to WorldNet Telecommunications SIP Trunking Service.

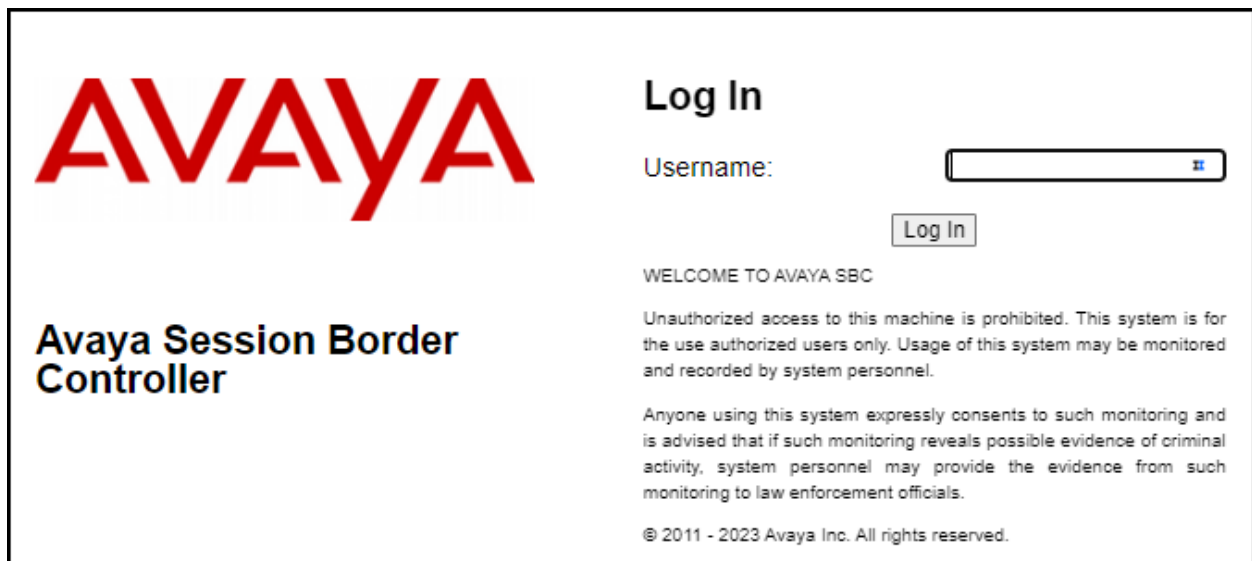
It is assumed that the Avaya SBC was provisioned and is ready to be used; the configuration shown here is accomplished using the Avaya SBC web interface.

Note: In the following pages, and for brevity in these Application Notes, not every provisioning step will have a screenshot associated with it. Some of the default information in the screenshots that follow may have been cut out (not included) for brevity.

7.1. Log in Avaya SBC

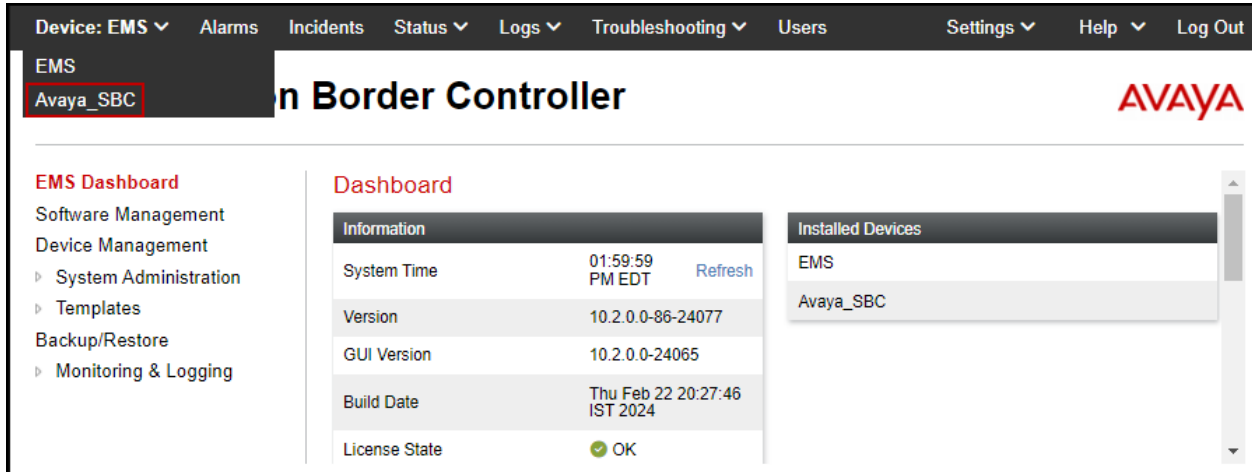
Use a Web browser to access the Avaya SBC Web interface. Enter `https://<ip-addr>/sbc` in the address field of the web browser, where `<ip-addr>` is the Avaya SBC management IP address.

Enter the appropriate credentials and click **Log In**.



The screenshot shows the Avaya Session Border Controller login interface. On the left, the Avaya logo is displayed in red, with the text "Avaya Session Border Controller" below it. On the right, the "Log In" section contains a "Username:" label, a text input field, and a "Log In" button. Below the login fields, there is a "WELCOME TO AVAYA SBC" message, a warning about unauthorized access, a consent statement, and a copyright notice: "© 2011 - 2023 Avaya Inc. All rights reserved."

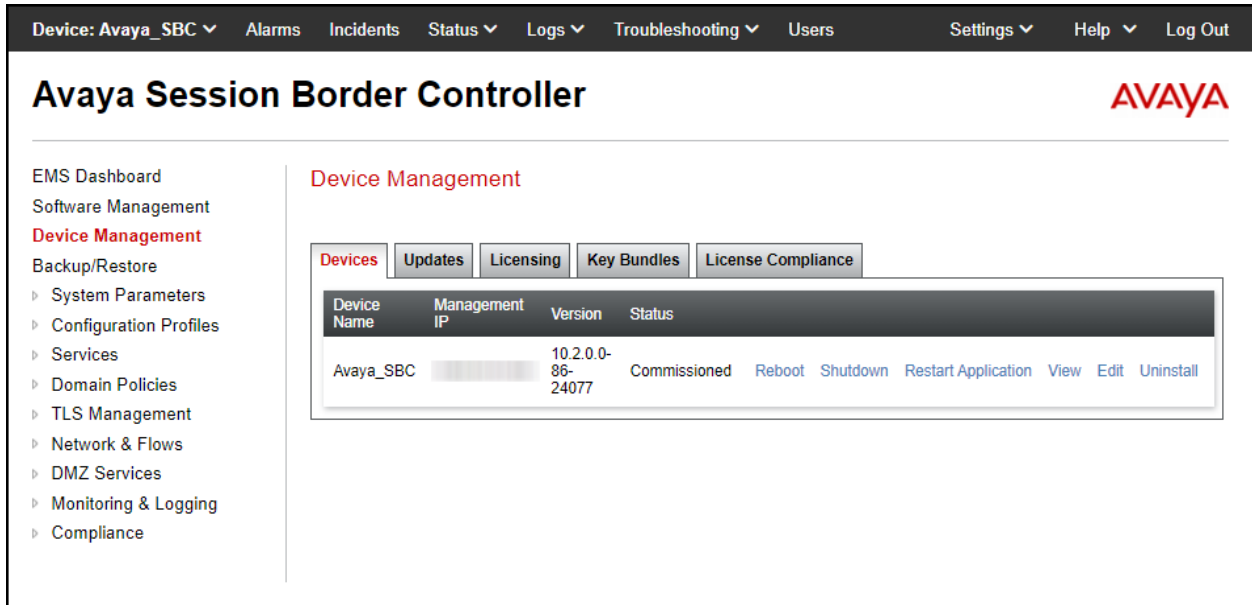
Once logged in, on the top left of the screen, under **Device:** select the device being managed, **Avaya_SBC** in the sample configuration.



The left navigation pane contains the different available menu items used for the configuration of the Avaya SBC. 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.

7.2. Device Management

To view current system information, select **Device Management** on the left navigation pane. In the reference configuration, the device named **Avaya_SBC** is shown. The management IP address that was configured during installation is blurred out for security reasons; the current software version is shown. The management IP address needs to be on a subnet separate from the ones used in all other interfaces of the Avaya SBC, 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.



The screenshot displays the Avaya Session Border Controller management interface. The top navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Avaya Session Border Controller' with the AVAYA logo. The left navigation pane lists various management options, with 'Device Management' highlighted. The main content area is titled 'Device Management' and features several tabs: 'Devices', 'Updates', 'Licensing', 'Key Bundles', and 'License Compliance'. The 'Devices' tab is active, showing a table with the following data:

Device Name	Management IP	Version	Status						
Avaya_SBC	[Blurred]	10.2.0.0-86-24077	Commissioned	Reboot	Shutdown	Restart Application	View	Edit	Uninstall

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

The **System Information** screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**. The **Box Type** was set to **SIP** and the **Deployment Mode** was set to **Proxy**. Default values were used for all other fields.

System Information: Avaya_SBC

General Configuration

Appliance Name	Avaya_SBC
Box Type	SIP
Deployment Mode	Proxy
HA Mode	No

Management IP(s)

IP #1 (IPv4) [Masked]

DNS Configuration

Primary DNS	8.8.8.8
Secondary DNS	8.8.4.4
DNS Location	DMZ
DNS Client IP	10.10.80.51

Dynamic License Allocation

	Min License Allocation	Max License Allocation
Standard Sessions	100	200
Advanced Sessions	100	200
Scopia Video Sessions	0	0
CES Sessions	0	0
Transcoding Sessions	75	100
AMR	<input type="checkbox"/>	
Premium Sessions	0	0
CLID	---	
Encryption Available: Yes	<input checked="" type="checkbox"/>	

Network Configuration

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.64.101.243	10.64.101.243	255.255.255.0	10.64.101.1	A1
[Masked]	[Masked]	[Masked]	[Masked]	A1
[Masked]	[Masked]	[Masked]	[Masked]	A1
[Masked]	[Masked]	[Masked]	[Masked]	B1
[Masked]	[Masked]	[Masked]	[Masked]	B1
10.10.80.51	10.10.80.51	255.255.255.128	10.10.80.1	B1

The IP addresses in the **System Information** screen shown above are the ones used for the SIP trunk to WorldNet Telecommunications and are the ones relevant to these Application Notes. Other IP addresses assigned to the Avaya SBC **A1** and **B1** interfaces are used to support remote workers and other SIP trunks, and they are not discussed in this document. Also note that for security purposes, any public IP addresses used during the compliance test have been masked in this document.

In the reference configuration, the private interface of the Avaya SBC (10.64.101.243) was used to connect to the enterprise network, while its public interface (10.10.80.51) was used to connect to the public network. See **Figure 1**.

On the **Dynamic 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.

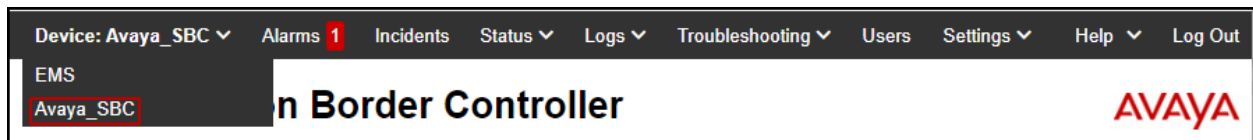
7.3. TLS Management

Note: Testing was done with System Manager signed identity certificates. The procedure to create and obtain these certificates is outside the scope of these Application Notes.

In the reference configuration, TLS transport is used for the communication between IP Office and Avaya SBC. The following procedures show how to create the client and server profiles to support the TLS connection.

7.3.1. Verify TLS Certificates – Avaya Session Border Controller

Once logged in, on the top left of the screen, under **Device:** select the device being managed, **Avaya_SBC** in the sample configuration.



Step 1 - Select **TLS Management** → **Certificates** from the left-hand menu. Verify the following:

- Verify the System Manager Root CA certificate is present in the **Installed CA Certificates** area, this certificate is required to enable TLS encryption inside of the enterprise (private network side). This Root CA certificate needs to be manually downloaded from System Manager and installed in the Avaya SBC; this Root CA certificate doesn't come pre-loaded in the Avaya SBC. Certificates from a 3rd party trusted Certificate Authority (CA) could be used for TLS encryption inside of the enterprise (private network side) instead of using Avaya System Manager as the Certificate Authority.
- Verify the identity certificate signed by the System Manager CA is present in the **Installed Certificates** area.
- Verify the Private key associated with the identity certificate signed by the System Manager CA is present in the **Installed Keys** area (not shown).

Device: Avaya_SBC ▾ Alarms Incidents Status ▾ Logs ▾ Troubleshooting ▾ Users Settings ▾ Help ▾ Log Out

Avaya Session Border Controller

AVAYA

EMS Dashboard
Software Management
Device Management
Backup/Restore
▸ System Parameters
▸ Configuration Profiles
▸ Services
▸ Domain Policies
▸ TLS Management
 Certificates
 Client Profiles
 Server Profiles
 SNI Group
▸ Network & Flows
▸ DMZ Services
▸ Monitoring & Logging
▸ Compliance

Certificates

Install Generate CSR

Installed Certificates	
IPOSE_INTERNAL.pem	View Delete
SBC_Internal_new.pem	View Delete
IPOSE_11_1.pem	View Delete
IPOSE_12_0.pem	View Delete

Installed CA Certificates	
AvayaDeviceEnrollmentCAchain.crt	View Delete
DigiCertGlobalRootCA.cer	View Delete
default.pem	View Delete
GoDaddyRootCAClass2.crt	View Delete
Thornton_SMGR_ROOT_CA.pem	View Delete

7.3.2. Server Profiles

Step 1 - Select **TLS Management** → **Server Profiles** and click on **Add**. Enter the following:

- **Profile Name:** enter descriptive name, e.g., **IPO_12_0_Server_Profile**.
- **Certificate:** select the identity certificate, e.g., **IPOSE_12_0.pem**, from pull down menu
- **Peer Verification = None.**
- Click **Next**.

Step 2 - Accept default values for the next screen (not shown) and click **Finish**.

Edit Profile X

WARNING: Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems.

Changing the certificate in a TLS Profile which has SNI enabled may cause existing Reverse Proxy entries which utilize this TLS Profile to become invalid.

TLS Profile

Profile Name: IPO_12_0_Server_Profile

Certificate: IPOSE_12_0.pem

SNI Options: None

SNI Group: None

Certificate Verification

Peer Verification: None

Peer Certificate Authorities: AvayaDeviceEnrollmentCAchain.crt, DigiCertGlobalRootCA.cer, default.pem, GoDaddyRootCAClass2.crt

Peer Certificate Revocation Lists:

Verification Depth: 0

Next

The following screen shows the completed TLS **Server Profile** form:

The screenshot displays the Avaya Session Border Controller (SBC) web interface. At the top, a navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Avaya Session Border Controller' and the 'AVAYA' logo. A left-hand navigation menu lists various management options, with 'Server Profiles' highlighted under 'TLS Management'. The main content area is titled 'Server Profiles: IPO_12_0_Server_Profile' and features an 'Add' button and a 'Delete' button. A blue bar prompts the user to 'Click here to add a description.'. Below this, the 'Server Profile' configuration form is shown, containing several sections: 'TLS Profile' (Profile Name: IPO_12_0_Server_Profile, Certificate: IPOSE_12_0.pem, SNI Options: None), 'Certificate Verification' (Peer Verification: None, Extended Hostname Verification: unchecked), 'Renegotiation Parameters' (Renegotiation Time: 0, Renegotiation Byte Count: 0), and 'Handshake Options' (Version: TLS 1.3 and TLS 1.2 checked, Ciphers: Default selected, Value: DEFAULT:ISHA). An 'Edit' button is located at the bottom of the form.

7.3.3. Client Profiles

Step 1 - Select **TLS Management** → **Client Profiles** and click on **Add**. Enter the following:

- **Profile Name:** enter descriptive name, e.g., **IPO_12_0_Client_Profile**.
- **Certificate:** select the identity certificate, e.g., **IPOSE_12_0.pem**, from pull down menu.
- **Peer Verification = Required.**
- **Peer Certificate Authorities:** select the CA certificate used to verify the certificate received from Session Manager, e.g., **Thornton_SMGR_ROOT_CA.pem**.
- **Verification Depth:** enter **1**.
- Click **Next**.

Step 2 - Accept default values for the next screen (not shown) and click **Finish**.

Edit Profile X

WARNING: Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems.

Changing the certificate in a TLS Profile which has SNI enabled may cause existing Reverse Proxy entries which utilize this TLS Profile to become invalid.

TLS Profile

Profile Name:

Certificate:

SNI: Enabled

Certificate Verification

Peer Verification: Required

Peer Certificate Authorities:

Peer Certificate Revocation Lists:

Verification Depth:

Extended Hostname Verification:

Server Hostname:

The following screen shows the completed TLS **Client Profile** form:

The screenshot displays the Avaya Session Border Controller (SBC) web interface. The top navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Avaya Session Border Controller' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with categories like 'EMS Dashboard', 'Software Management', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Services', 'Domain Policies', 'TLS Management', 'Certificates', 'Client Profiles', 'Server Profiles', 'SNI Group', 'Network & Flows', 'DMZ Services', 'Monitoring & Logging', and 'Compliance'. The 'Client Profiles' section is currently selected.

The main content area is titled 'Client Profiles: IPO_12_0_Client_Profile'. It features an 'Add' button and a 'Delete' button. Below this is a blue bar with the text 'Click here to add a description.'. The main configuration area is titled 'Client Profile' and contains the following settings:

- Certificate:** IPOSE_12_0.pem
- SNI:** Enabled
- Certificate Verification:**
 - Peer Verification:** Required
 - Peer Certificate Authorities:** Thornton_SMGR_ROOT_CA.pem
 - Peer Certificate Revocation Lists:** ---
 - Verification Depth:** 1
 - Extended Hostname Verification:**
- Renegotiation Parameters:**
 - Renegotiation Time:** 0
 - Renegotiation Byte Count:** 0
- Handshake Options:**
 - Version:** TLS 1.3 TLS 1.2
 - Ciphers:** Default FIPS Custom
 - Value:** DEFAULT:!SHA

An 'Edit' button is located at the bottom of the configuration area.

7.4. Configuration Profiles

The Configuration Profiles Menu, on the left navigation pane, allows the configuration of parameters across all Avaya SBC appliances.

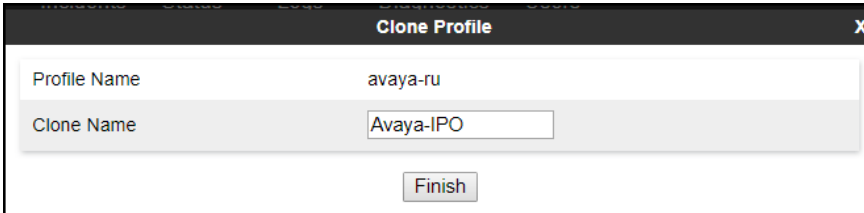
7.4.1. Server Interworking – Avaya-IPO

Interworking Profile features are configured to facilitate interoperability of implementations between enterprise SIP-enabled solutions and different SIP trunk service providers.

Several profiles have been already pre-defined and they populate the list under **Interworking Profiles** on the screen below. If a different profile is needed, a new Interworking Profile can be created, or an existing default profile can be modified or “cloned”. Since directly modifying a default profile is generally not recommended, for the test configuration the default **avaya-ru** profile was duplicated, or “cloned”. If needed, the profile can then be modified to meet specific requirements for the enterprise SIP-enabled solution. For WorldNet Telecommunications, this profile was left with the **avaya-ru** default values.

On the left navigation pane, select **Configuration Profiles** → **Server Interworking**. From the **Interworking Profiles** list, select **avaya-ru**. Click **Clone** on top right of the screen (not shown).

Enter the new profile name in the **Clone Name** field, the name of **Avaya-IPO** was chosen in this example. Click **Finish**.



Clone Profile	
Profile Name	avaya-ru
Clone Name	Avaya-IPO
<input type="button" value="Finish"/>	

The following screen capture shows the **General** tab of the newly created **Avaya-IPO** Server Interworking Profile.

The screenshot displays the Avaya Session Border Controller web interface. The top navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Avaya Session Border Controller' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with categories like 'EMS Dashboard', 'Software Management', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Services', 'Domain Policies', 'TLS Management', 'Network & Flows', 'DMZ Services', 'Monitoring & Logging', and 'Compliance'. Under 'Configuration Profiles', 'Server Interworking' is highlighted.

The main content area is titled 'Interworking Profiles: Avaya-IPO'. It features an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. A blue bar prompts 'Click here to add a description.' Below this are tabs for 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'General' tab is active, showing a table of settings:

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
Delayed Offer	Yes
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
Prack Handling	No
Allow 18X SDP	No
T.38 Support	No
URI Scheme	SIP
Via Header Format	RFC3261
SIPS Required	Yes
Mediasec	No

An 'Edit' button is located at the bottom right of the settings table.

The following screen capture shows the **Advanced** tab of the newly created **Avaya-IPO** Server Interworking Profile.

The screenshot displays the Avaya Session Border Controller configuration interface. At the top, there is a navigation bar with the following items: Device: Avaya_SBC, Alarms, Incidents, Status, Logs, Troubleshooting, Users, Settings, Help, and Log Out. The main header shows "Avaya Session Border Controller" and the AVAYA logo.

On the left side, there is a navigation menu with the following categories: EMS Dashboard, Software Management, Device Management, Backup/Restore, System Parameters, Configuration Profiles (expanded), Services, and Domain Policies. Under Configuration Profiles, the following items are listed: Domain DoS, Server, Interworking (highlighted), Media Forking, Routing, Topology Hiding, Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, FGDN Groups, Reverse Proxy Policy, URN Profile, Recording Profile, H248 Profile, and IP/URI Blocklist Profile.

The main content area is titled "Interworking Profiles: Avaya-IPO". It includes an "Add" button and "Rename", "Clone", and "Delete" buttons. Below this, there is a list of interworking profiles: avaya-ru, OCS-Edge-Ser..., cisco-ccm, cups, OCS-FrontEnd..., Avaya-SM, Avaya-IPO (highlighted), Avaya-CS1000, Avaya-CM, cs2100, and SP-General.

The configuration for the selected "Avaya-IPO" profile is shown in the "Advanced" tab. The configuration includes the following settings:

Setting	Value
Record Routes	Both Sides
Include End Point IP for Context Lookup	Yes
Extensions	Avaya
Diversion Manipulation	No
Has Remote SBC	Yes
Route Response on Via Port	No
MOBX Re-INVITE Handling	No
NATing for 301/302 Redirection	Yes
SIP Recording	
Relay INVITE Replace	No
Conference URI	
Include Called Participant	No
DTMF	
DTMF Support	None

An "Edit" button is located at the bottom of the configuration area.

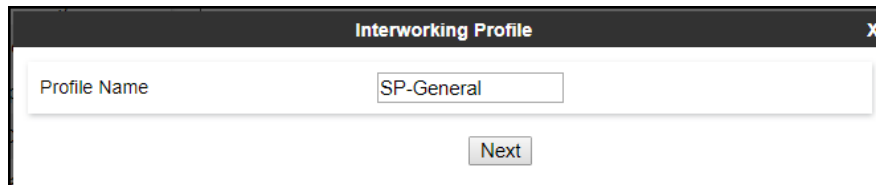
7.4.2. Server Interworking - SP-General

A second Server Interworking profile named **SP-General** was created for the Service Provider.

On the left navigation pane, select **Configuration Profiles → Server Interworking** (not shown). From the **Interworking Profiles** list, select **Add** (not shown) (note that **Add** is being used to create the SP-General profile instead of cloning the avaya-ru profile).

Enter the new profile name, the name of **SP-General** was chosen in this example.

- 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" containing the text "SP-General". Below the input field is a button labeled "Next".

On the **General** tab, click **Next** until the last tab is reached then click **Finish** on the last tab leaving remaining fields with default values (not shown).

The screenshot shows a dialog box titled "Editing Profile: SP-General" with a close button (X) in the top right corner. The "General" tab is active, displaying a list of configuration options. Each option has a radio button or checkbox next to it. The "Finish" button is located at the bottom center of the dialog.

Field	Value
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly <input type="radio"/> Microsoft Teams
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
URI Group	None
Send Hold	<input type="checkbox"/>
Delayed Offer	<input checked="" type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
Re-Invite Handling	<input type="checkbox"/>
Prack Handling	<input type="checkbox"/>
Allow 18X SDP	<input type="checkbox"/>
T.38 Support	<input type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543
SIPS Required	<input type="checkbox"/>
Mediasec Handling	<input type="checkbox"/>

Finish

The following screen capture shows the **General** tab of the newly created **SP-General** Server Interworking Profile.

The screenshot displays the Avaya Session Border Controller web interface. At the top, there is a navigation bar with options like 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Avaya Session Border Controller' and the 'AVAYA' logo.

On the left, there is a sidebar menu with categories such as 'EMS Dashboard', 'Software Management', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Server Interworking', 'Media Forking', 'Routing', 'Topology Hiding', 'Signaling Manipulation', 'URI Groups', 'SNMP Traps', 'Time of Day Rules', 'FGDN Groups', 'Reverse Proxy Policy', 'URN Profile', 'Recording Profile', 'H248 Profile', 'IP/URI Blocklist Profile', 'Services', 'Domain Policies', 'TLS Management', 'Network & Flows', 'DMZ Services', 'Monitoring & Logging', and 'Compliance'.

The main content area is titled 'Interworking Profiles: SP-General'. It includes an 'Add' button and action buttons for 'Rename', 'Clone', and 'Delete'. Below this is a blue bar with the text 'Click here to add a description.' and a list of tabs: 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'General' tab is selected, showing a table of configuration parameters.

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
Delayed Offer	Yes
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
Prack Handling	No
Allow 18X SDP	No
T.38 Support	No
URI Scheme	SIP
Via Header Format	RFC3261
SIPS Required	No
Mediasec	No

An 'Edit' button is located at the bottom right of the configuration table.

The following screen capture shows the **Advanced** tab of the newly created **SP-General** Server Interworking Profile.

The screenshot displays the Avaya Session Border Controller (SBC) web interface. At the top, there is a navigation bar with the following items: Device: Avaya_SBC, Alarms, Incidents, Status, Logs, Troubleshooting, Users, Settings, Help, and Log Out. The main header shows "Avaya Session Border Controller" and the Avaya logo.

The left sidebar contains a navigation menu with the following categories: EMS Dashboard, Software Management, Device Management, Backup/Restore, System Parameters, Configuration Profiles (with sub-items: Domain DoS, Server, Interworking, Media Forking, Routing, Topology Hiding, Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, FGDN Groups, Reverse Proxy Policy, URN Profile, Recording Profile, H248 Profile, IP/URI Blocklist Profile), Services, and Domain Policies. The "Interworking" option is highlighted in red.

The main content area is titled "Interworking Profiles: SP-General" and includes an "Add" button and "Rename", "Clone", and "Delete" buttons. Below this is a blue bar with the text "Click here to add a description." and a list of tabs: General, Timers, Privacy, URI Manipulation, Header Manipulation, and Advanced (which is selected and highlighted in red).

The configuration details for the Advanced tab are as follows:

Record Routes	Both Sides
Include End Point IP for Context Lookup	No
Extensions	None
Diversion Manipulation	No
Has Remote SBC	Yes
Route Response on Via Port	No
MOBX Re-INVITE Handling	No
NATing for 301/302 Redirection	Yes
SIP Recording	
Relay INVITE Replace	No
Conference URI	
Include Called Participant	No
DTMF	
DTMF Support	None

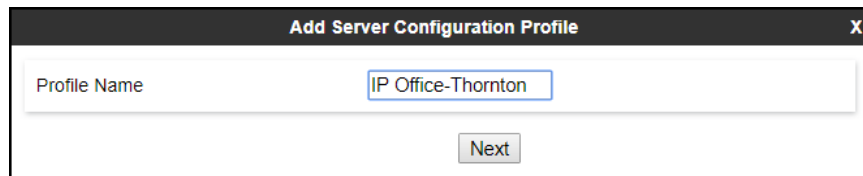
An "Edit" button is located at the bottom right of the configuration area.

7.4.3. SIP Server Configuration

SIP Server Profiles should be created for the Avaya SBC's two peers, the Call Server (IP Office) and the Trunk Server or SIP Proxy at the service provider's network.

To add the SIP Server profile for the Call Server, from the **Services** menu on the left-hand navigation pane, select **SIP Servers** (not shown). Click **Add** (not shown) and enter the profile name: **IP Office-Thornton**.

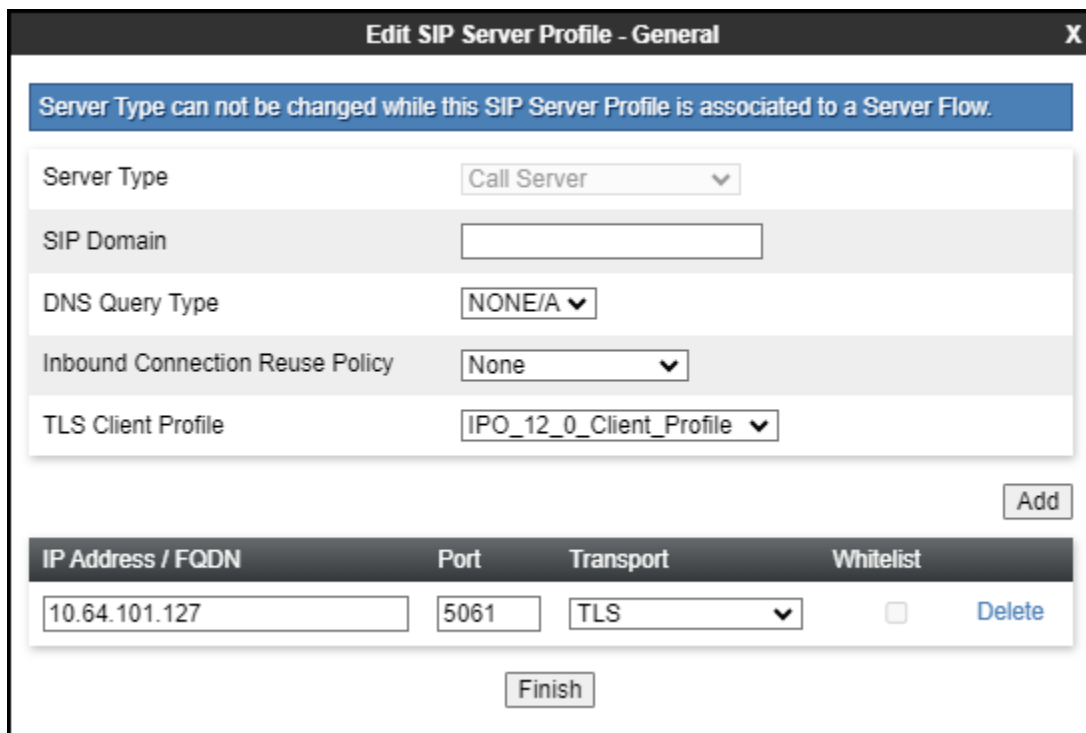
- Click **Next**.



The screenshot shows a dialog box titled "Add Server Configuration Profile". It has a close button (X) in the top right corner. The "Profile Name" field is filled with "IP Office-Thornton". Below the field is a "Next" button.

On the **Edit SIP Server Profile – General** window:

- **Server Type:** Select **Call Server**.
- **IP Address / FQDN:** **10.64.101.127** (IP Address of IP Office).
- **Port:** **5061** (This port must match the port number defined in **Section 5.2.1**).
- **Transport:** Select **TLS**.
- Select a **TLS Client Profile** (**Section 7.3.3**).
- Click **Next** (not shown).



The screenshot shows the "Edit SIP Server Profile - General" window. At the top, a blue banner reads: "Server Type can not be changed while this SIP Server Profile is associated to a Server Flow." Below this, several fields are visible: "Server Type" (Call Server), "SIP Domain" (empty), "DNS Query Type" (NONE/A), "Inbound Connection Reuse Policy" (None), and "TLS Client Profile" (IPO_12_0_Client_Profile). An "Add" button is at the bottom right of this section. Below is a table with columns: IP Address / FQDN, Port, Transport, and Whitelist. The table contains one row with values: 10.64.101.127, 5061, TLS, and a checkbox. A "Delete" button is next to the checkbox. At the bottom of the window is a "Finish" button.

IP Address / FQDN	Port	Transport	Whitelist
10.64.101.127	5061	TLS	<input type="checkbox"/>

- Click **Next** until the **Add SIP Server Profile - Advanced** tab is reached (not shown).
- On the **Add SIP Server Profile - Advanced** tab:
- Verify that **Enable Grooming** is checked (required for TLS transport).
- Select **Avaya-IPO** from the **Interworking Profile** drop down menu (**Section 7.4.1**).
- Leave the **Signaling Manipulation Script** at the default **None**.
- Click **Finish**.

Add SIP Server Profile - Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input checked="" type="checkbox"/>
Interworking Profile	Avaya-IPO
Signaling Manipulation Script	None
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	5060
TLS Failover Port	5061
Tolerant	<input type="checkbox"/>
URI Group	None
NG911 Support	<input type="checkbox"/>
Back Finish	

The following screen capture shows the **General** tab of the newly created **IP Office-Thornton** SIP Server Configuration Profile.

The screenshot shows the Avaya Session Border Controller interface. The top navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header displays 'Avaya Session Border Controller' and the 'AVAYA' logo. On the left, a navigation menu lists various management options, with 'Services' expanded to show 'SIP Servers'. The main content area is titled 'SIP Servers: IP Office-Thornton' and features an 'Add' button and 'Rename', 'Clone', and 'Delete' options. Below this, there are tabs for 'General', 'Authentication', 'Heartbeat', 'Registration', 'Ping', and 'Advanced'. The 'General' tab is active, showing a configuration table:

Server Type	Call Server		
TLS Client Profile	IPO_12_0_Client_Profile		
DNS Query Type	NONE/A		
Inbound Connection Reuse Policy	None		
IP Address / FQDN	Port	Transport	Whitelist
10.64.101.127	5061	TLS	<input type="checkbox"/>

An 'Edit' button is located below the table.

The following screen capture shows the **Advanced** tab of the newly created **IP Office-Thornton** SIP Server Configuration Profile.

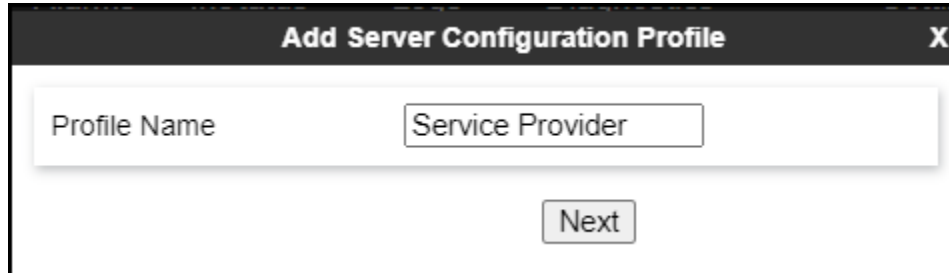
The screenshot shows the Avaya Session Border Controller interface with the 'Advanced' tab selected. The navigation and header elements are identical to the previous screenshot. The 'Advanced' tab is active, displaying a configuration table with various options:

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input checked="" type="checkbox"/>
Interworking Profile	Avaya-IPO
Signaling Manipulation Script	None
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
Tolerant	<input type="checkbox"/>
URI Group	None
NG911 Support	<input type="checkbox"/>

An 'Edit' button is located below the table.

To add the SIP Server profile for the Trunk Server, from the **Services** menu on the left-hand navigation pane, select **SIP Servers** (not shown). Click **Add** (not shown) and enter the profile name: **Service Provider**.

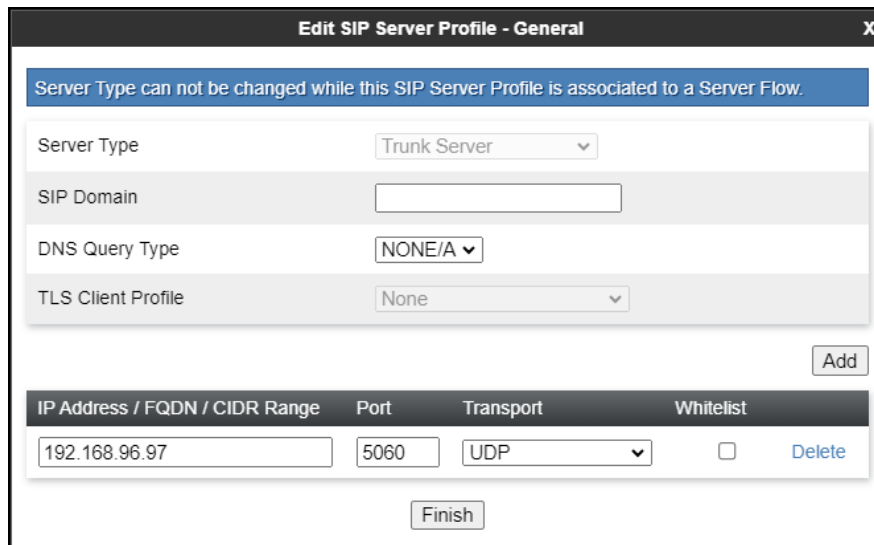
- Click **Next**.



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 "Service Provider". Below the input field is a "Next" button.

On the **Edit SIP Server Profile – General** window:

- **Server Type:** Select **Trunk Server**.
- Click on **Add** and under **IP Address / FQDN** enter: **192.168.96.97** (WorldNet SIP proxy server IP address, this information was provided by WorldNet).
- Enter **5060** under **Port** and select **UDP** for **Transport**.
- Click **Next** (not shown).



The screenshot shows the "Edit SIP Server Profile - General" window. At the top, there is a blue warning message: "Server Type can not be changed while this SIP Server Profile is associated to a Server Flow." Below this, there are several configuration fields:

- Server Type:** Trunk Server (dropdown menu)
- SIP Domain:** (empty text input field)
- DNS Query Type:** NONE/A (dropdown menu)
- TLS Client Profile:** None (dropdown menu)

Below these fields is an "Add" button. Underneath is a table with the following columns: "IP Address / FQDN / CIDR Range", "Port", "Transport", and "Whitelist".

IP Address / FQDN / CIDR Range	Port	Transport	Whitelist
192.168.96.97	5060	UDP	<input type="checkbox"/>

At the bottom of the window is a "Finish" button.

On the **Add SIP Server Profile - Authentication** tab:

- Check the **Enable Authentication** box.
- Enter the **User Name** credential provided by WorldNet for SIP trunk registration.
- Leave **Realm** blank.
- Enter **Password** credential provided by WorldNet for SIP trunk registration.
- Click **Next**.

The screenshot shows a web-based configuration window titled "Add SIP Server Profile - Authentication". The window has a close button in the top right corner. The form contains the following fields and controls:

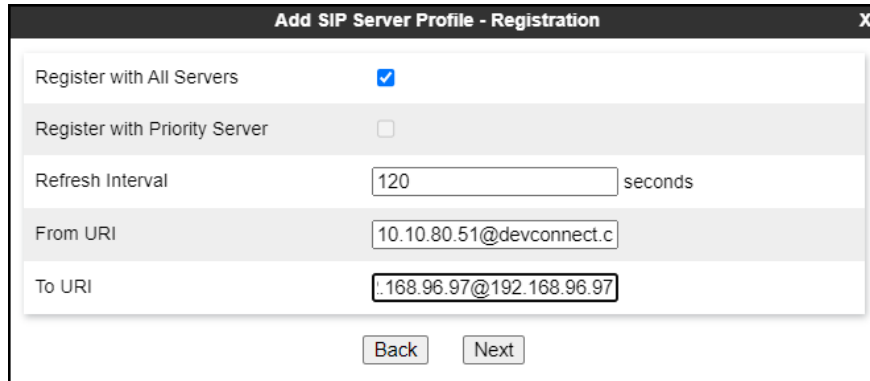
- Enable Authentication:** A checkbox that is checked.
- User Name:** A text input field containing the value "user123".
- Realm:** A text input field that is currently empty. Below the field is the text "(Leave blank to detect from server challenge)".
- Password:** A text input field with a masked password (represented by dots) and a small icon to toggle visibility.
- Confirm Password:** A text input field with a masked password (represented by dots) and a small icon to toggle visibility.

At the bottom of the form, there are two buttons: "Back" and "Next".

- Click **Next** on the **Add Server Configuration Profile - Heartbeat** window (not shown).

On the **Add SIP Server Profile - Registration** tab.

- Check the **Register with All Servers** box.
- **Frequency**: Enter the amount of time (in seconds) between REGISTER messages that will be sent from the enterprise to the Service Provider Proxy Server to refresh the registration binding of the SIP trunk. This value should be chosen in consultation with the service provider. **120** seconds was the value used during the compliance test.
- The **From URI** and **To URI** entries for the REGISTER messages are built using the following:
 - **From URI**: Use the Avaya SBC public IP address and the enterprise domain (10.10.80.51@devconnect.com), as shown on the screen below.
 - **To URI**: Use WorldNet's SIP proxy IP address (192.168.96.97@(192.168.96.97)), as shown on the screen below.
 - Click **Next**.



The screenshot shows a configuration window titled "Add SIP Server Profile - Registration". It contains the following fields and controls:

Register with All Servers	<input checked="" type="checkbox"/>
Register with Priority Server	<input type="checkbox"/>
Refresh Interval	<input type="text" value="120"/> seconds
From URI	<input type="text" value="10.10.80.51@devconnect.c"/>
To URI	<input type="text" value=":168.96.97@192.168.96.97"/>

At the bottom of the window, there are two buttons: "Back" and "Next".

- Click **Next** on the **Add SIP Server Profile - Ping** window (not shown).

On the **Add SIP Server Profile – Advanced** tab:

- Uncheck **Enable Grooming** (not required for UDP transport).
- Select **SP-General** from the **Interworking Profile** drop-down menu (**Section 7.4.2**).
- Click **Finish**.

The screenshot shows a configuration window titled "Add SIP Server Profile - Advanced" with a close button (X) in the top right corner. The window contains several settings:

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SP-General
Signaling Manipulation Script	None
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	5060
TLS Failover Port	5061
Tolerant	<input type="checkbox"/>
URI Group	None
NG911 Support	<input type="checkbox"/>

At the bottom of the window, there are two buttons: "Back" and "Finish".

The following screen capture shows the **General** tab of the newly created **Service Provider** SIP Server Configuration Profile.

The screenshot displays the Avaya Session Border Controller (SBC) configuration interface. At the top, a navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Avaya Session Border Controller' with the AVAYA logo on the right. A left-hand navigation menu lists various management options, with 'SIP Servers' expanded to show 'Service Providers'. The main content area is titled 'SIP Servers: Service Provider' and features an 'Add' button and 'Rename', 'Clone', and 'Delete' actions. Below this are tabs for 'General', 'Authentication', 'Heartbeat', 'Registration', 'Ping', and 'Advanced'. The 'General' tab is active, showing configuration details for a 'Trunk Server'. The 'DNS Query Type' is set to 'NONE/A' and the 'Inbound Connection Reuse Policy' is 'None'. A table lists the IP address and port configuration:

IP Address / FQDN /CIDR Range	Port	Transport	Whitelist
192.168.96.97	5060	UDP	<input type="checkbox"/>

An 'Edit' button is located below the table.

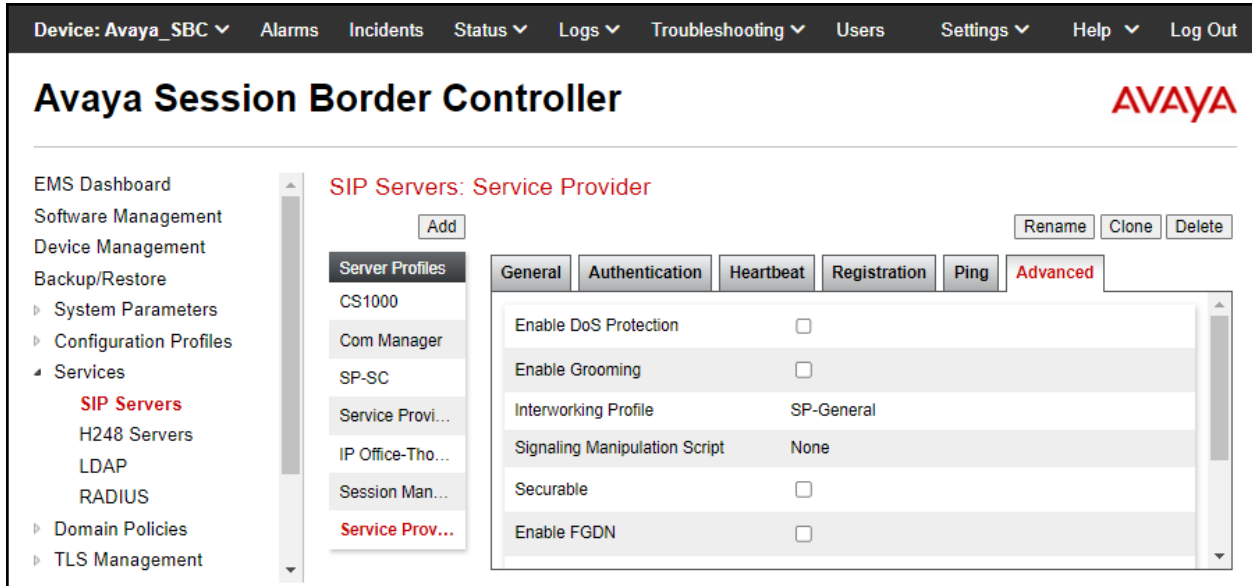
The following screen capture shows the **Authentication** tab of the newly created **Service Provider** Server Configuration Profile.

The screenshot displays the Avaya Session Border Controller web interface. At the top, a navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Avaya Session Border Controller' and the 'AVAYA' logo. On the left, a sidebar menu lists various management options, with 'SIP Servers' expanded to show 'Service Provider' selected. The main content area is titled 'SIP Servers: Service Provider' and includes an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. Below this, there are tabs for 'General', 'Authentication', 'Heartbeat', 'Registration', 'Ping', and 'Advanced'. The 'Authentication' tab is active, showing a form with the following fields: 'Enable Authentication' (checked), 'User Name' (user123), and 'Realm' (---). An 'Edit' button is located at the bottom right of the form.

The following screen capture shows the **Registration** tab of the newly created **Service Provider** Server Configuration Profile.

The screenshot displays the Avaya Session Border Controller web interface, similar to the previous one. The navigation bar and sidebar are identical. The main content area is titled 'SIP Servers: Service Provider' and includes an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. Below this, there are tabs for 'General', 'Authentication', 'Heartbeat', 'Registration', 'Ping', and 'Advanced'. The 'Registration' tab is active, showing a form with the following fields: 'Register with All Servers' (checked), 'Register with Priority Server' (unchecked), 'Refresh Interval' (120 seconds), 'From URI' (10.10.80.51@devconnect.com), and 'To URI' (192.168.96.97@192.168.96.97). An 'Edit' button is located at the bottom right of the form.

The following screen capture shows the **Advanced** tab of the newly created **Service Provider** SIP Server Configuration Profile.



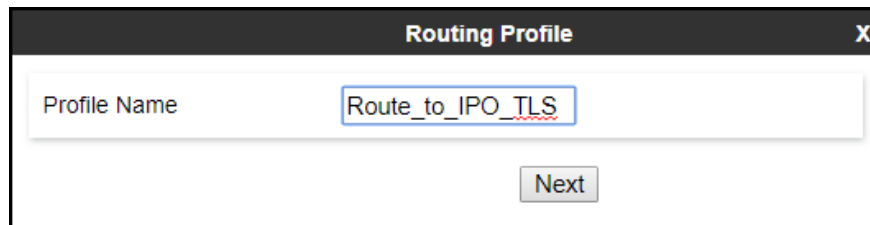
7.4.4. Routing Profiles

Routing profiles define a specific set of routing criteria that are used, in conjunction with other types of domain policies, to determine the route that SIP packets should follow to arrive at their intended destination.

Two Routing profiles were created, one for inbound calls, with IP Office as the destination, and the second one for outbound calls, which are sent to the Service Provider SIP trunk.

To create the inbound route, from the **Configuration Profiles** menu on the left-hand side (not shown):

- Select **Routing** (not shown).
- Click **Add** in the **Routing Profiles** section (not shown).
- Enter Profile Name: **Route_to_IPO_TLS**.
- Click **Next**.



The screenshot 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_IPO_TLS". Below the input field is a "Next" button.

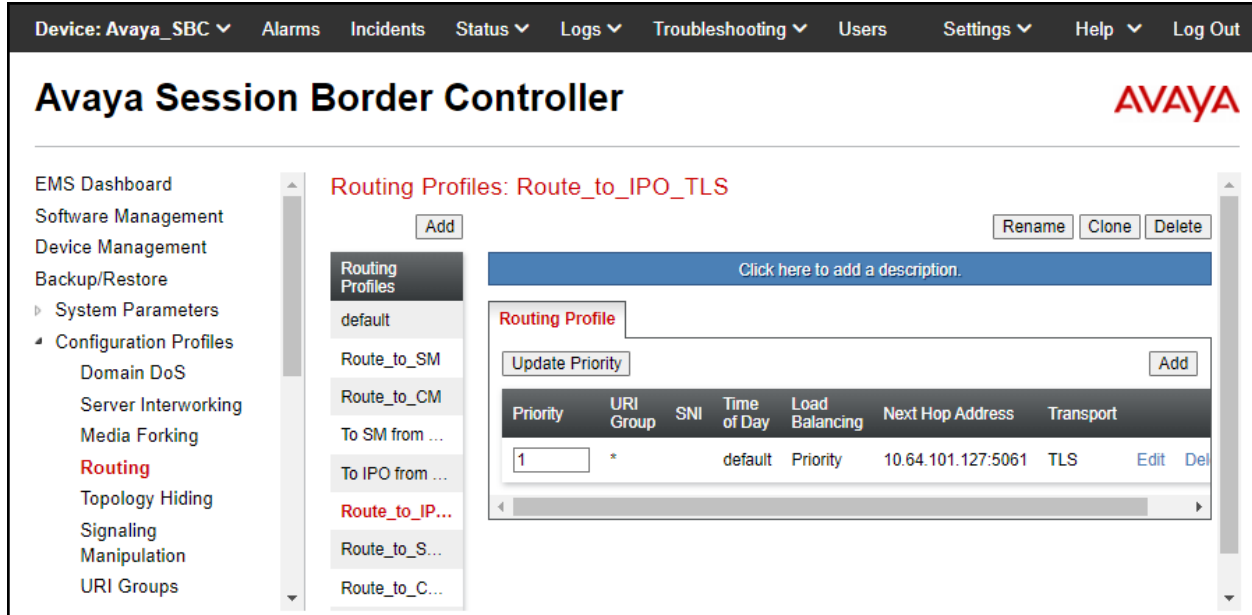
On the **Routing Profile** screen complete the following:

- Click on the **Add** button to add a **Next-Hop Address**.
- **Priority / Weight: 1**
- **SIP Server Profile: Select IP Office Thornton.**
- **Next Hop Address** is populated automatically with **10.64.101.127:5061 (TLS)** (IP Office IP address, Port and Transport).
- Click **Finish**.

The screenshot shows the 'Profile : Route_to_IPO_TLS - Edit Rule' configuration window. The main configuration area includes fields for URI Group, Time of Day, Load Balancing, Transport, LDAP Server Profile, LDAP Base DN (Search), Matched Attribute Priority, Next Hop Priority (checked), Ignore Route Header, ENUM, and ENUM Suffix. An 'Add' button is located at the bottom right of this section. Below the main configuration is a table with the following columns: Priority / Weight, LDAP Search Attribute, LDAP Search Regex Pattern, LDAP Search Regex Result, SIP Server Profile, Next Hop Address, and Transport. The table contains one entry with a priority of 1, an empty LDAP Search Attribute, empty LDAP Search Regex Pattern and Result, 'IP Office-' as the SIP Server Profile, '10.64.101.127:5061' as the Next Hop Address, and 'None' as the Transport. A 'Delete' button is next to this entry. A 'Finish' button is located at the bottom center of the window.

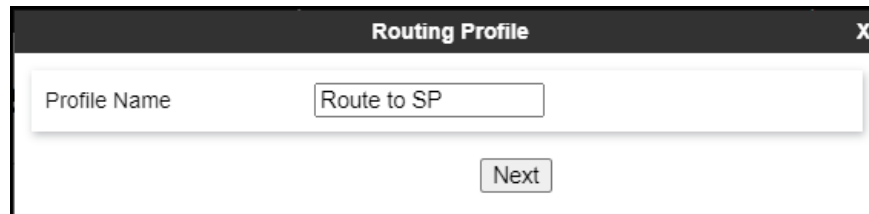
Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport
1				IP Office-	10.64.101.127:5061	None

The following screen shows the newly created **Route_to_IPO_TLS** Routing Profile.



Similarly, for the outbound route:

- Select **Routing** (not shown).
- Click **Add** in the **Routing Profiles** section (not shown).
- Enter Profile Name: **Route to SP**.
- Click **Next**.



On the Routing Profile screen complete the following:

- **Load Balancing:** Select **Priority**.
- Click on the **Add** button to add a **Next-Hop Address**.
- **SIP Server Profile:** Select **Service Provider**.
- The **Next Hop Address** is populated automatically with **192.168.96.97:5060 (UDP)** (WorldNet's SIP Proxy IP address, port and transport).
- Click **Finish**.

Profile : Route to SP - Edit Rule

URI Group	*	Time of Day	default
Load Balancing	Priority	NAPTR	<input type="checkbox"/>
Transport	None	LDAP Routing	<input type="checkbox"/>
LDAP Server Profile	None	LDAP Base DN (Search)	None
Matched Attribute Priority	<input type="checkbox"/>	Alternate Routing	<input type="checkbox"/>
Next Hop Priority	<input checked="" type="checkbox"/>	Next Hop In-Dialog	<input type="checkbox"/>
Ignore Route Header	<input type="checkbox"/>		
ENUM	<input type="checkbox"/>	ENUM Suffix	

Add

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport	
1				Service P	192.168.96.97:5060	None	Delete

Finish

The following screen capture shows the newly created **Route to SP** Routing Profile.

The screenshot displays the Avaya Session Border Controller (SBC) web interface. At the top, a navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Avaya Session Border Controller' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with categories like 'EMS Dashboard', 'Software Management', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Domain DoS', 'Server Interworking', 'Media Forking', 'Routing' (highlighted), 'Topology Hiding', 'Signaling Manipulation', 'URI Groups', 'SNMP Traps', and 'Time of Day Rules'.

The main content area is titled 'Routing Profiles: Route to SP'. It features an 'Add' button and a list of routing profiles: 'default', 'Route_to_SM', 'Route_to_CM', 'To SM from R...', 'To IPO from R...', 'Route_to_IPO...', 'Route_to_SP_...', 'Route_to_CS...', and 'Route to SP' (highlighted). Above the list are 'Rename', 'Clone', and 'Delete' buttons.

The 'Route to SP' profile is expanded, showing a description field with the text 'Click here to add a description.' Below this is a 'Routing Profile' configuration table with an 'Update Priority' button and an 'Add' button.

Priority	URI Group	SNI	Time of Day	Load Balancing	Next Hop Address	Transport	
1	*		default	Priority	192.168.96.97:5060	UDP	Edit Delete

7.4.5. Topology Hiding

Topology Hiding is a security feature which allows changing several parameters of the SIP packets, 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 SIP headers like To, From, Request-URI, Via, Record-Route and SDP to the IP addresses or domains expected by IP Office and the SIP trunk service provider, allowing the call to be accepted in each case.

For the compliance test, 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.

To add the Topology Hiding Profile in the Enterprise direction, select **Topology Hiding** from the **Configuration Profiles** menu on the left-hand side (not shown):

- Click on **default** profile and select **Clone Profile** (not shown).
- Enter the **Profile Name: IP Office**.
- Click **Finish** (not shown).



The screenshot shows a dialog box titled "Topology Hiding 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 the input field is a "Next" button.

The following screen capture shows the newly added **IP Office** Topology Hiding Profile. Note that for IP Office no values were overwritten (left with default values).

The screenshot displays the Avaya Session Border Controller web interface. The top navigation bar shows the device name 'Avaya_SBC' and various menu options like Alarms, Incidents, Status, Logs, Troubleshooting, Users, Settings, Help, and Log Out. The main header reads 'Avaya Session Border Controller' with the AVAYA logo on the right.

The left sidebar contains a navigation menu with categories such as EMS Dashboard, Software Management, Device Management, Backup/Restore, System Parameters, Configuration Profiles, and others. The 'Topology Hiding' option is highlighted in red.

The main content area is titled 'Topology Hiding Profiles: IP Office'. It features an 'Add' button and three action buttons: 'Rename', 'Clone', and 'Delete'. Below this is a blue bar with the text 'Click here to add a description.' and a tab labeled 'Topology Hiding'.

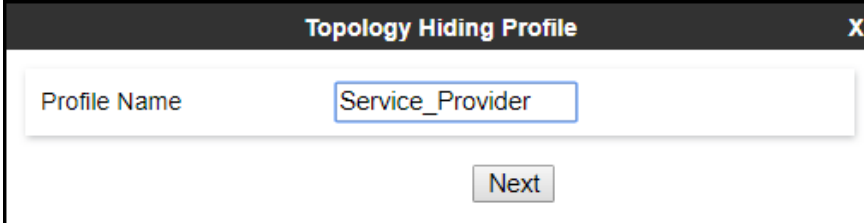
The central part of the interface contains a table with the following data:

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

An 'Edit' button is located at the bottom right of the table.

To add the Topology Hiding Profile in the Service Provider direction, select **Topology Hiding** from the **Configuration Profiles** menu on the left-hand side (not shown):

- Click on **default** profile and select **Clone Profile** (not shown).
- Enter the **Profile Name: Service_Provider**.
- Click **Finish** (not shown).



The screenshot shows a dialog box titled "Topology Hiding Profile" with a close button (X) in the top right corner. The dialog contains a text input field labeled "Profile Name" with the text "Service_Provider" entered. Below the input field is a "Next" button.

The following screen capture shows the newly added **Service_Provider** Topology Hiding Profile (left with default values).

The screenshot shows the Avaya Session Border Controller web interface. The top navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header displays 'Avaya Session Border Controller' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with categories like 'EMS Dashboard', 'Software Management', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Routing', 'Topology Hiding', 'Signaling Manipulation', 'URI Groups', 'SNMP Traps', 'Time of Day Rules', 'FGDN Groups', 'Reverse Proxy', and 'Policy'. The 'Topology Hiding' option is highlighted in red.

The main content area is titled 'Topology Hiding Profiles: Service_Provider'. It features an 'Add' button and three buttons: 'Rename', 'Clone', and 'Delete'. Below this is a blue bar with the text 'Click here to add a description.' and a tab labeled 'Topology Hiding'.

The 'Topology Hiding' tab contains a table with the following data:

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

An 'Edit' button is located at the bottom right of the table.

7.5. Domain Policies

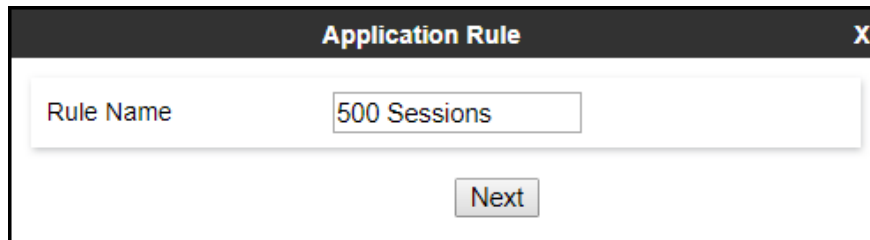
Domain Policies allow configuring, managing and applying various sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise.

7.5.1. Application Rules

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

From the menu on the left-hand side, select **Domain Policies** → **Application Rules** (not shown).

- Click on the **Add** button to add a new rule (not shown).
- **Rule Name:** enter the name of the profile, e.g., **500 Session**.
- Click **Next**.



The screenshot shows a dialog box titled "Application Rule" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Rule Name" containing the text "500 Sessions". Below the input field is a "Next" button.

- Under **Audio** check **In** and **Out** and set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** to recommended values; the value of **500** was used in the sample configuration.
- Under **Video** check **In** and **Out** and set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** to recommended values; the value of **100** was used in the sample configuration.
- Click **Finish**.

Editing Rule: 500 Sessions X

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Audio	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	500	500
Video	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	100	100

Miscellaneous

CDR Support Off
 RADIUS
 CDR Adjunct

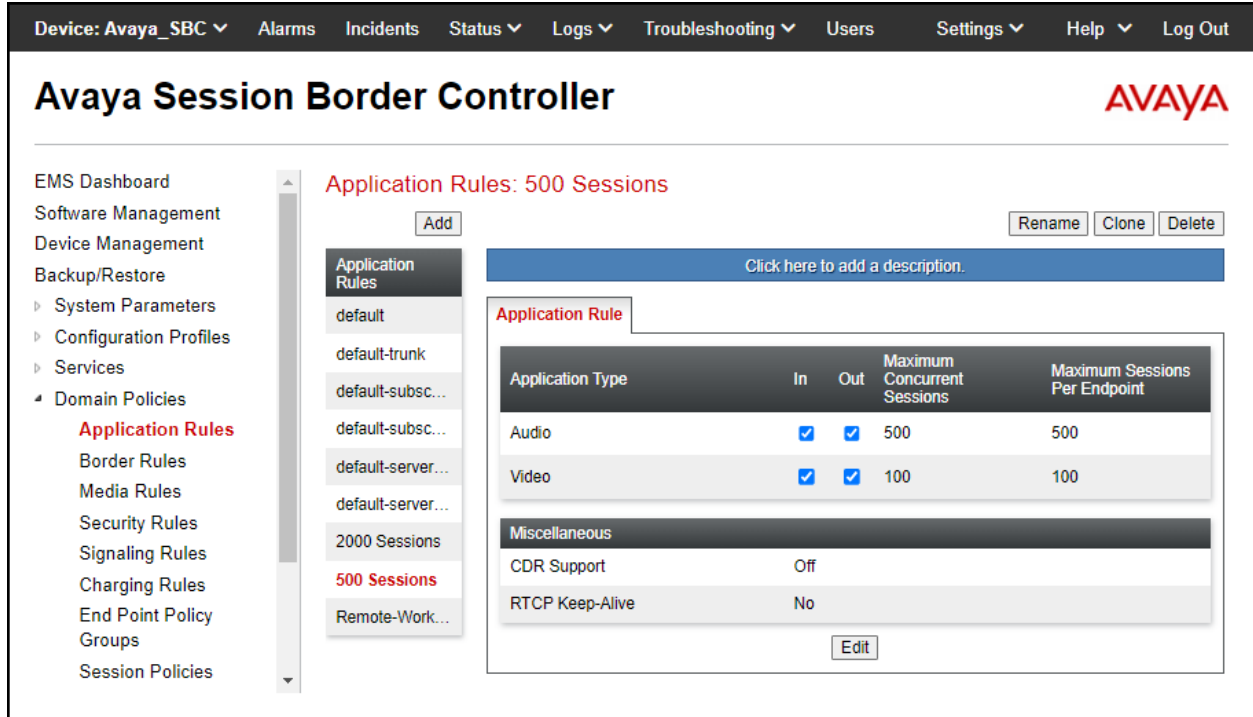
RADIUS Profile None ▾

Media Statistics Support

Call Duration Setup
 Connect

RTCP Keep-Alive

The following screen capture shows the newly created **500 Sessions** Application Rule.

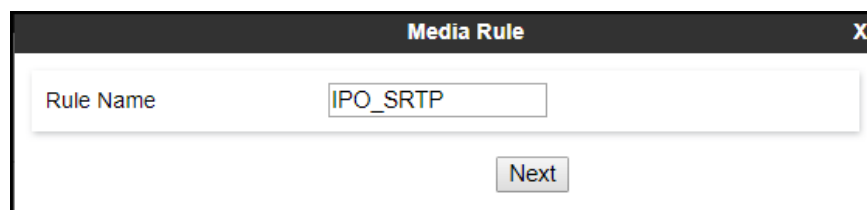


7.5.2. Media Rules

Media Rules allow one to define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBC security product. For the compliance test one media rule was created toward IP Office, the existing **default-low-med** media rule was used toward the Service Provider.

To add a media rule in the IP Office direction, from the menu on the left-hand side, select **Domain Policies → Media Rules**.

- Click on the **Add** button to add a new media rule (not shown).
- Under **Rule Name** enter **IPO_SRTP**.
- Click Next.



- Under Audio Encryption, **Preferred Format #1**, select **SRTP_AES_CM_128_HMAC_SHA1_80**.
- Under Audio Encryption, **Preferred Format #2**, select **RTP**.
- Under Audio Encryption, uncheck **Encrypted RTCP**.
- Under Audio Encryption, check **Interworking**.
- Repeat the above steps under Video Encryption.
- Under Miscellaneous check **Capability Negotiation**.
- Click **Next** (not shown).

Media Encryption	
Audio Encryption	
Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80 ▼
Preferred Format #2	RTP ▼
Preferred Format #3	NONE ▼
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime Leave blank to match any value.	2^ <input type="text"/>
Interworking	<input checked="" type="checkbox"/>
Symmetric Context Reset	<input checked="" type="checkbox"/>
Key Change in New Offer	<input type="checkbox"/>
Video Encryption	
Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80 ▼
Preferred Format #2	RTP ▼
Preferred Format #3	NONE ▼
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime Leave blank to match any value.	2^ <input type="text"/>
Interworking	<input checked="" type="checkbox"/>
Symmetric Context Reset	<input checked="" type="checkbox"/>
Key Change in New Offer	<input type="checkbox"/>
Miscellaneous	
Capability Negotiation	<input checked="" type="checkbox"/>
Finish	

- Accept default values in the remaining sections by clicking **Next** (not shown), and then click **Finish** (not shown).

For the compliance test, the **default-low-med** Media Rule was used in the Service Provider direction.

Audio Encryption	
Preferred Format #1	RTP
Preferred Format #2	NONE
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime <small>Leave blank to match any value.</small>	2^ <input type="text"/>
Interworking	<input checked="" type="checkbox"/>
Symmetric Context Reset	<input checked="" type="checkbox"/>
Key Change in New Offer	<input type="checkbox"/>

Video Encryption	
Preferred Format #1	RTP
Preferred Format #2	NONE
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime <small>Leave blank to match any value.</small>	2^ <input type="text"/>
Interworking	<input checked="" type="checkbox"/>
Symmetric Context Reset	<input checked="" type="checkbox"/>
Key Change in New Offer	<input type="checkbox"/>

Miscellaneous	
Capability Negotiation	<input type="checkbox"/>

The following screen capture shows the newly created **IPO_SRTP** Media Rule.

The screenshot displays the Avaya Session Border Controller (SBC) configuration interface. At the top, there is a navigation bar with the following items: Device: Avaya_SBC, Alarms, Incidents, Status, Logs, Troubleshooting, Users, Settings, Help, and Log Out. The main header reads "Avaya Session Border Controller" with the AVAYA logo on the right.

On the left side, there is a navigation menu with the following categories and sub-items:

- EMS Dashboard
- Software Management
- Device Management
- Backup/Restore
- System Parameters
- Configuration Profiles
- Services
- Domain Policies
 - Application Rules
 - Border Rules
 - Media Rules**
 - Security Rules
 - Signaling Rules
 - Charging Rules
 - End Point Policy Groups
 - Session Policies
- TLS Management
- Network & Flows
- DMZ Services
- Monitoring & Logging
- Compliance

The main content area is titled "Media Rules: IPO_SRTP". It features an "Add" button and three action buttons: "Rename", "Clone", and "Delete". Below this is a blue bar with the text "Click here to add a description." and a list of media rule templates: "default-low-med", "default-low-m...", "default-high", "default-high-enc", "avaya-low-me...", "Rem_Worker...", **IPO_SRTP** (highlighted in red), "ServiceProvid...", and "SM_SRTP".

The configuration for the **IPO_SRTP** rule is shown in a tabbed interface with the following settings:

- Encryption** (selected tab):
 - Audio Encryption:
 - Preferred Formats: SRTP_AES_CM_128_HMAC_SHA1_80 RTP
 - Encrypted RTCP:
 - MKI:
 - Lifetime: Any
 - Interworking:
 - Symmetric Context Reset:
 - Key Change in New Offer:
 - Video Encryption:
 - Preferred Formats: SRTP_AES_CM_128_HMAC_SHA1_80 RTP
 - Encrypted RTCP:
 - MKI:
 - Lifetime: Any
 - Interworking:
 - Symmetric Context Reset:
 - Key Change in New Offer:
 - Miscellaneous:
 - Capability Negotiation:
- Codec Prioritization
- Advanced
- QoS

An "Edit" button is located at the bottom right of the configuration panel.

The following screen capture shows the default-low-med Media Rule.

The screenshot displays the Avaya Session Border Controller (SBC) configuration interface. At the top, a navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Avaya Session Border Controller' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with categories like 'EMS Dashboard', 'Software Management', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Services', 'Domain Policies', 'Application Rules', 'Border Rules', 'Media Rules', 'Security Rules', 'Signaling Rules', 'Charging Rules', 'End Point Policy Groups', 'Session Policies', 'TLS Management', 'Network & Flows', 'DMZ Services', 'Monitoring & Logging', and 'Compliance'. The 'Media Rules' section is highlighted.

The main content area is titled 'Media Rules: default-low-med'. It features an 'Add' button and a 'Clone' button. A warning message states: 'It is not recommended to edit the defaults. Try cloning or adding a new rule instead.' Below this, there are tabs for 'Encryption', 'Codec Prioritization', 'Advanced', and 'QoS'. The 'Encryption' tab is active, showing settings for 'Audio Encryption' and 'Video Encryption'. The 'Miscellaneous' section is also visible.

Audio Encryption	
Preferred Formats	RTP
Interworking	<input checked="" type="checkbox"/>
Symmetric Context Reset	<input checked="" type="checkbox"/>
Key Change in New Offer	<input type="checkbox"/>

Video Encryption	
Preferred Formats	RTP
Interworking	<input checked="" type="checkbox"/>
Symmetric Context Reset	<input checked="" type="checkbox"/>
Key Change in New Offer	<input type="checkbox"/>

Miscellaneous	
Capability Negotiation	<input type="checkbox"/>

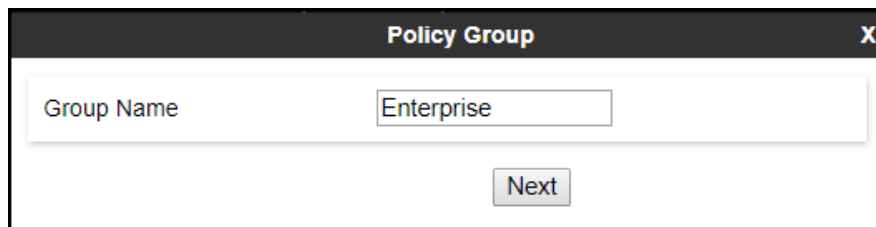
An 'Edit' button is located at the bottom right of the configuration area.

7.5.3. End Point Policy Groups

End Point Policy Groups are associations of different sets of rules (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBC.

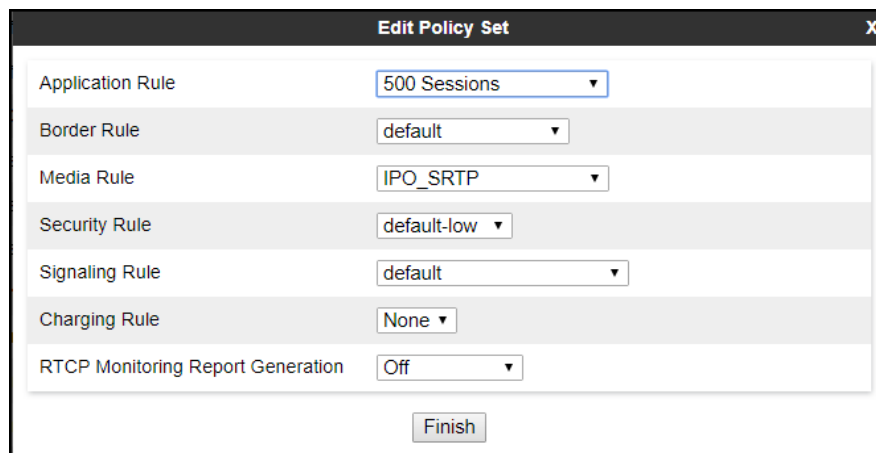
To create an End Point Policy Group for the Enterprise, from the **Domain Policies** menu, select **End Point Policy Groups** (not shown).

- Click on the **Add** button to add a new policy group (not shown).
- **Group Name: Enterprise.**
- Click **Next**.



The screenshot shows a dialog box titled "Policy Group" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Group Name" containing the text "Enterprise". Below the input field is a "Next" button.

- **Application Rule: 500 Sessions.**
- **Border Rule: default.**
- **Media Rule: IPO_SRTP (Section 7.5.2).**
- **Security Rule: default-low.**
- **Signaling Rule: default.**
- Click **Finish**.



The screenshot shows a dialog box titled "Edit Policy Set" with a close button (X) in the top right corner. The dialog contains several configuration options, each with a dropdown menu:

- Application Rule: 500 Sessions
- Border Rule: default
- Media Rule: IPO_SRTP
- Security Rule: default-low
- Signaling Rule: default
- Charging Rule: None
- RTCP Monitoring Report Generation: Off

A "Finish" button is located at the bottom center of the dialog.

The following screen capture shows the newly created **Enterprise** End Point Policy Group.

The screenshot displays the Avaya Session Border Controller (SBC) management interface. At the top, a navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Avaya Session Border Controller' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with categories like 'EMS Dashboard', 'Software Management', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Services', 'Domain Policies', 'Application Rules', 'Border Rules', 'Media Rules', 'Security Rules', 'Signaling Rules', 'Charging Rules', 'End Point Policy Groups' (highlighted in red), 'Session Policies', and 'TLS Management'.

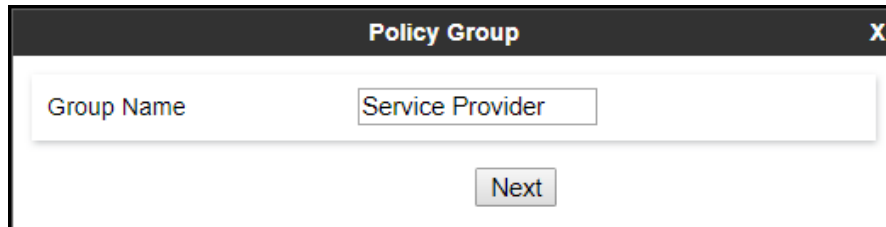
The main content area is titled 'Policy Groups: Enterprise'. It features an 'Add' button and three action buttons: 'Rename', 'Clone', and 'Delete'. Below these are two blue boxes with text: 'Click here to add a description.' and 'Click here to add a row description.'.

A 'Policy Group' configuration window is open, showing a table with the following data:

Order	Application	Border	Media	Security	Signaling	Charging	RTCP Mon Gen	
1	500 Sessions	default	IPO_SRTP	default-low	default	None	Off	Edit

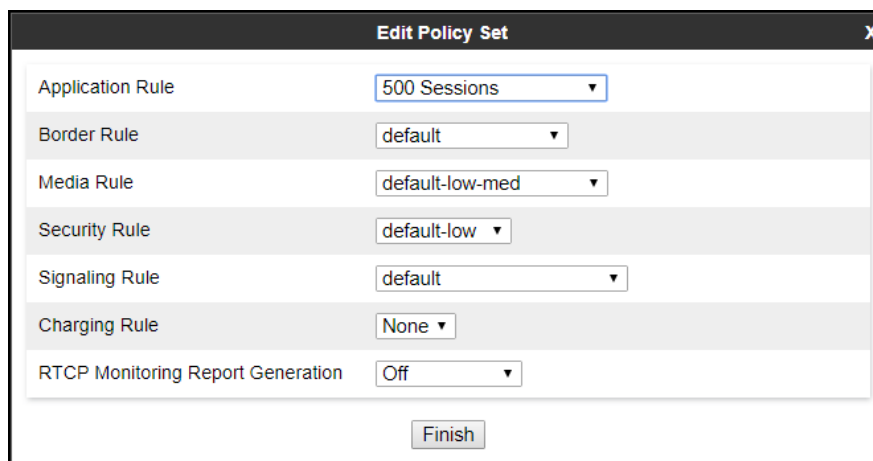
Similarly, to create an End Point Policy Group for the Service Provider SIP Trunk.

- Click on the **Add** button to add a new policy group (not shown).
- **Group Name: Service Provider.**
- Click **Next**.



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

- **Application Rule: 500 Sessions**
- **Border Rule: default.**
- **Media Rule: default-low-med.**
- **Security Rule: default-low.**
- **Signaling Rule: default.**
- Click **Finish**.



The screenshot shows a dialog box titled "Edit Policy Set" with a close button (X) in the top right corner. The dialog contains several rows, each with a label and a dropdown menu:

Application Rule	500 Sessions
Border Rule	default
Media Rule	default-low-med
Security Rule	default-low
Signaling Rule	default
Charging Rule	None
RTCP Monitoring Report Generation	Off

At the bottom of the dialog is a button labeled "Finish".

The following screen capture shows the newly created **Service Provider** End Point Policy Group.

The screenshot displays the Avaya Session Border Controller interface. At the top, there is a navigation bar with options like 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Avaya Session Border Controller' with the AVAYA logo on the right.

On the left, a navigation menu lists various management options, including 'Domain Policies' and 'End Point Policy Groups'. The main content area is titled 'Policy Groups: Service Provider' and features an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. Below this, there are two blue boxes with text: 'Click here to add a description.' and 'Click here to add a row description.'

A 'Policy Group' section is expanded, showing a table with the following data:

Order	Application	Border	Media	Security	Signaling	Charging	RTCP Mon Gen
1	500 Sessions	default	default-low-med	default-low	default	None	Off

An 'Edit' button is visible next to the last row of the table. A 'Summary' button is also present in the top right of the policy group view.

7.6. Network & Flows Settings

The **Network & Flows** settings allow the management of various device-specific parameters, which determine how a particular device will function when deployed in the network. Specific server parameters, like network and interface settings, as well as call flows, etc. are defined here.

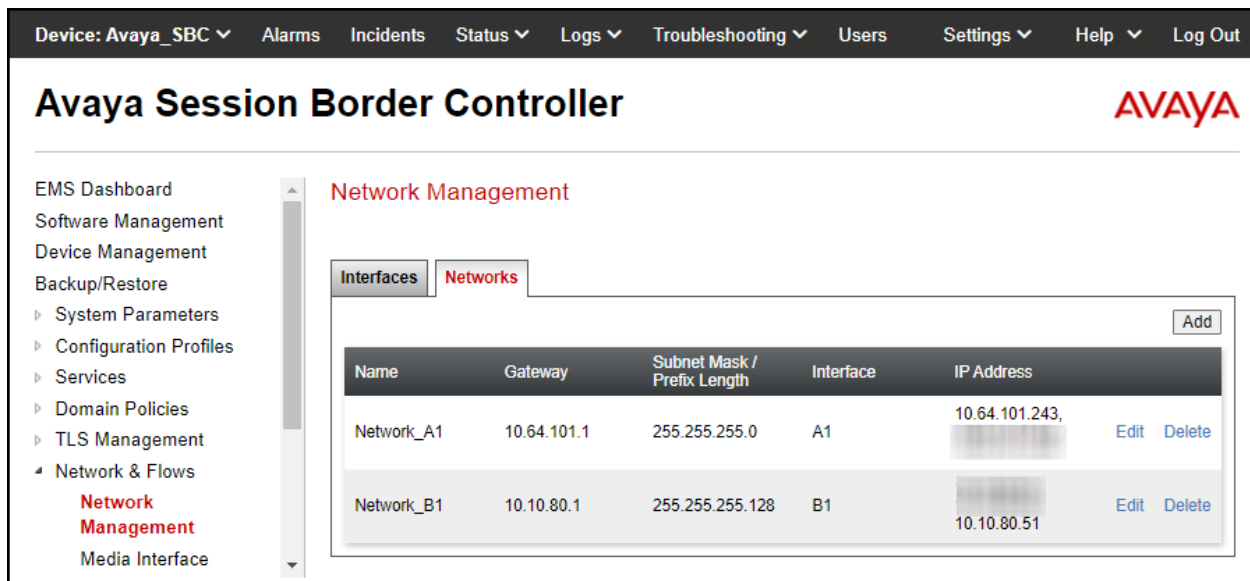
7.6.1. Network Management

The network information should have been previously completed. To verify the network configuration, from the **Network & Flows** on the left-hand side, select **Network Management**. Select the **Networks** tab.

In the event that changes need to be made to the network configuration information, they can be entered here.

Use **Figure 1** as reference for IP address assignments.

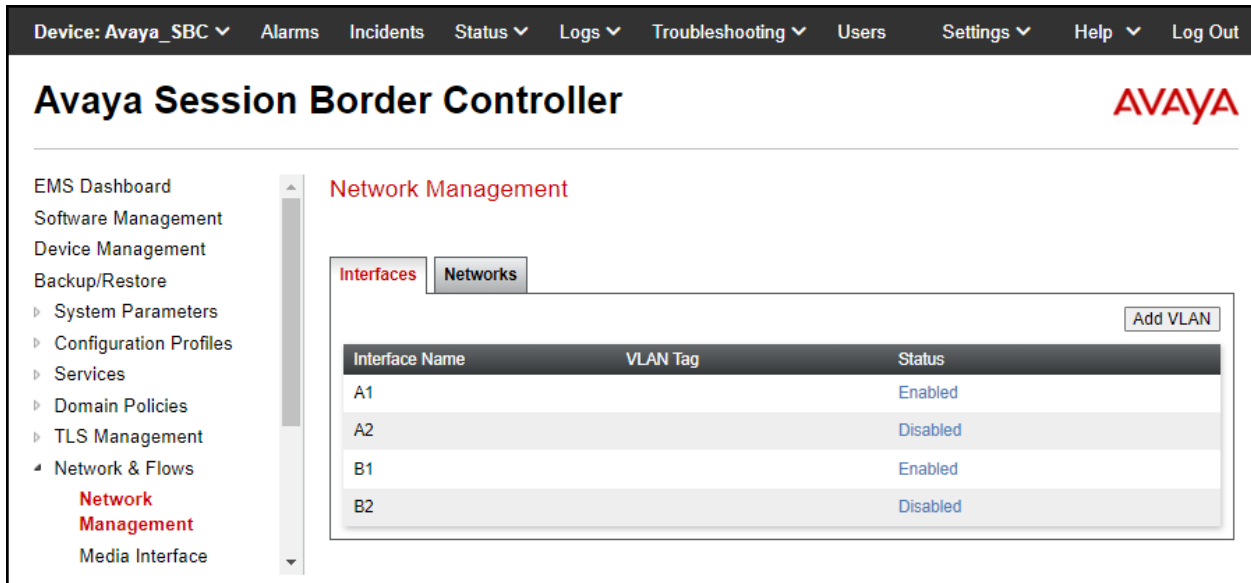
Note: Only the highlighted entity items were created for the compliance test and are the ones relevant to these Application Notes. Blurred out items are part of the Remote Worker configuration, which is not discussed in these Application Notes.



The screenshot displays the Avaya Session Border Controller interface. The top navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Avaya Session Border Controller' with the AVAYA logo. A left sidebar lists navigation options: EMS Dashboard, Software Management, Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, TLS Management, and Network & Flows (selected). Under 'Network & Flows', 'Network Management' is highlighted. The main content area is titled 'Network Management' and has two tabs: 'Interfaces' and 'Networks' (selected). An 'Add' button is in the top right of the table area. The table lists network configurations:

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	
Network_A1	10.64.101.1	255.255.255.0	A1	10.64.101.243	Edit Delete
Network_B1	10.10.80.1	255.255.255.128	B1	10.10.80.51	Edit Delete

On the Interfaces tab, click the **Status** control for interfaces **A1** and **B1** to change the status to **Enabled**. It should be noted that the default state for all interfaces is **Disabled**, so it is important to perform this step, or the Avaya SBC will not be able to communicate on any of its interfaces.



The screenshot displays the Avaya Session Border Controller (SBC) web interface. At the top, a navigation bar includes 'Device: Avaya_SBC' and various menu items like 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Avaya Session Border Controller' and the 'AVAYA' logo. A left sidebar contains a navigation menu with categories like 'EMS Dashboard', 'Software Management', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Services', 'Domain Policies', 'TLS Management', and 'Network & Flows'. Under 'Network & Flows', 'Network Management' is highlighted. The main content area is titled 'Network Management' and has two tabs: 'Interfaces' (selected) and 'Networks'. An 'Add VLAN' button is located in the top right of the interface section. Below the tabs is a table with the following data:

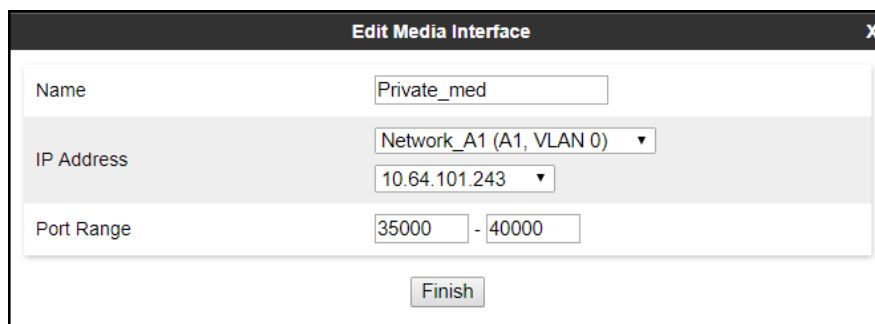
Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Disabled

7.6.2. Media Interface

Media Interfaces are created to specify the IP address and port range in which the Avaya SBC will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBC will advertise this IP address, and one of the ports in this range as the listening IP address and port in which the SBC will accept media from the connected server. Create a SIP Media Interface for both the inside and outside IP interfaces. On the Private and Public interfaces of the Avaya SBC, the port range 35000 to 40000 was used.

From the **Network & Flows** menu on the left-hand side, select **Media Interface** (not shown).

- Select **Add** in the **Media Interface** area (not shown).
- **Name: Private_med.**
- Under **IP Address** select: **Network_A1 (A1, VLAN 0)**
- Select **IP Address: 10.64.101.243** (Inside IP Address of the Avaya SBC, toward IP Office).
- **Port Range: 35000-40000.**
- Click **Finish**.



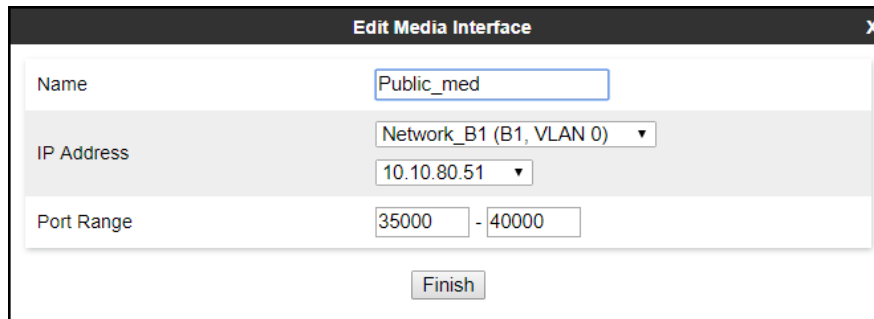
The screenshot shows a window titled "Edit Media Interface" with a close button (X) in the top right corner. The window contains the following configuration fields:

Name	Private_med
IP Address	Network_A1 (A1, VLAN 0) ▼ 10.64.101.243 ▼
Port Range	35000 - 40000

At the bottom center of the window is a button labeled "Finish".

Select **Add** in the **Media Interface** area (not shown).

- **Name: Public_med.**
- Under **IP Address** select: **Network_B1 (B1, VLAN 0)**
- Select **IP Address: 10.10.80.51** (Outside IP Address of the Avaya SBC, toward the Service Provider).
- **Port Range: 35000-40000.**
- Click **Finish**.



Edit Media Interface

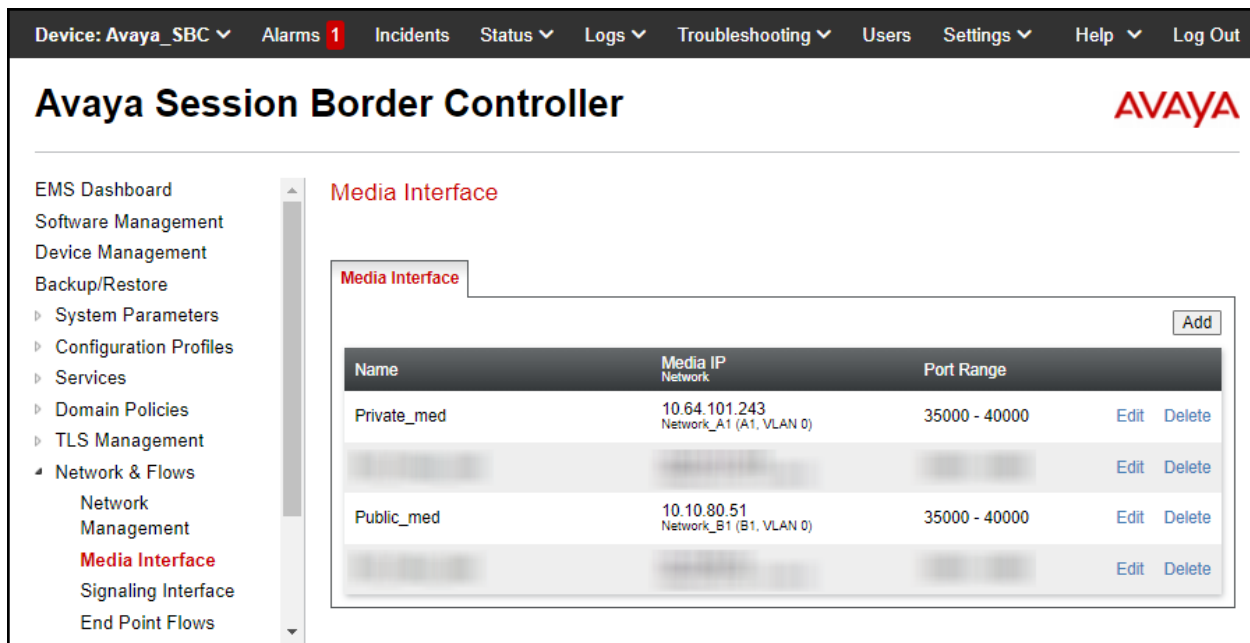
Name: Public_med

IP Address: Network_B1 (B1, VLAN 0)
10.10.80.51

Port Range: 35000 - 40000

Finish

The following screen capture shows the newly created Media Interfaces.



Device: Avaya_SBC Alarms 1 Incidents Status Logs Troubleshooting Users Settings Help Log Out

Avaya Session Border Controller

AVAYA

EMS Dashboard
Software Management
Device Management
Backup/Restore
System Parameters
Configuration Profiles
Services
Domain Policies
TLS Management
Network & Flows
Network Management
Media Interface
Signaling Interface
End Point Flows

Media Interface

Media Interface Add

Name	Media IP Network	Port Range	Edit	Delete
Private_med	10.64.101.243 Network_A1 (A1, VLAN 0)	35000 - 40000	Edit	Delete
Public_med	10.10.80.51 Network_B1 (B1, VLAN 0)	35000 - 40000	Edit	Delete

7.6.3. Signaling Interface

To create the Signaling Interface toward IP Office, from the **Network & Flows** menu on the left-hand side, select **Signaling Interface** (not shown).

- Select **Add** in the **Signaling Interface** area (not shown).
- **Name:** **Private_sig**.
- Under **IP Address** select: **Network_A1 (A1, VLAN 0)**
- Select **IP Address:** **10.64.101.243** (Inside IP Address of the Avaya SBC, toward IP Office).
- **TLS Port:** **5061**.
- Select a **TLS Profile** (**Section 7.3.2**).
- Click **Finish**.

Name	Private_sig
IP Address	Network_A1 (A1, VLAN 0) 10.64.101.243
TCP Port <small>Leave blank to disable</small>	
UDP Port <small>Leave blank to disable</small>	
TLS Port <small>Leave blank to disable</small>	5061
TLS Profile	IPO_12_0_Server_Profile
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

Finish

- Select **Add** in the **Signaling Interface** area (not shown).
- **Name: Public_sig.**
- Under **IP Address** select: **Network_B1 (B1, VLAN 0)**
- Select **IP Address: 10.10.80.51** (outside or public IP Address of the Avaya SBC, toward the Service Provider).
- **UDP Port: 5060.**
- Click **Finish.**

Name	Public_sig
IP Address	Network_B1 (B1, VLAN 0) 10.10.80.51
TCP Port <small>Leave blank to disable</small>	
UDP Port <small>Leave blank to disable</small>	5060
TLS Port <small>Leave blank to disable</small>	
TLS Profile	None
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

Finish

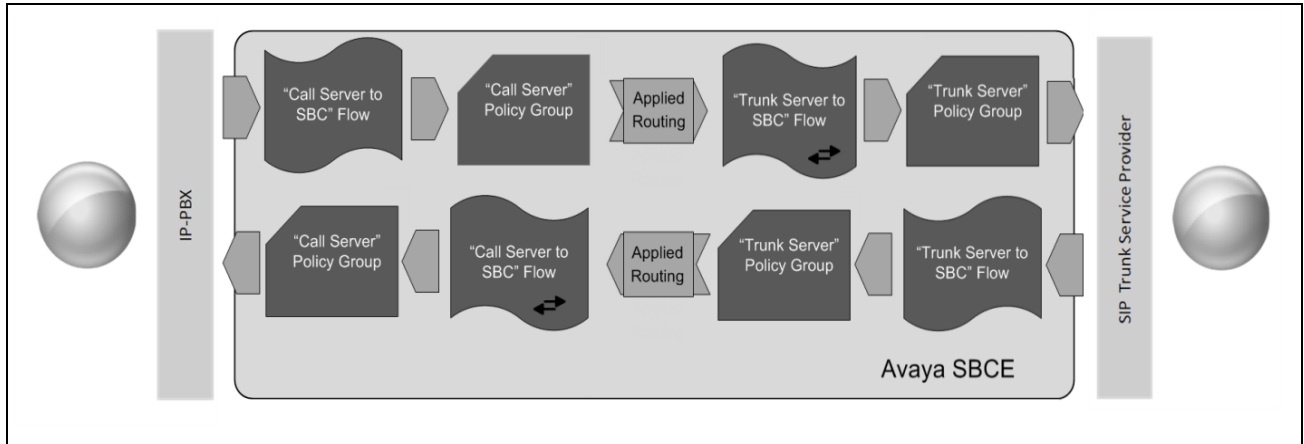
The following screen capture shows the newly created Signaling Interfaces.

The screenshot shows the Avaya Session Border Controller web interface. The top navigation bar includes: Device: Avaya_SBC, Alarms 1, Incidents, Status, Logs, Troubleshooting, Users, Settings, Help, and Log Out. The main header displays "Avaya Session Border Controller" and the AVAYA logo. The left sidebar menu lists various management options, with "Signaling Interface" highlighted in red. The main content area is titled "Signaling Interface" and features a table with the following data:

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile	
Private_sig	10.64.101.243 Network_A1 (A1, VLAN 0)	---	---	5061	IPO_12_0_Server_Profile	Edit Delete
Public_sig	10.10.80.51 Network_B1 (B1, VLAN 0)	---	5060	---	None	Edit Delete

7.6.4. End Point Flows

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



The **End-Point Flows** define certain parameters that pertain to the signaling and media portions of a call, whether it originates from within the enterprise or outside of the enterprise.

To create the call flow toward the Service Provider SIP trunk, from the **Network & Flows** menu, select **End Point Flows** (not shown), then the **Server Flows** tab. Click **Add** (not shown).

- **Name: SP to IPO Flow**
- **Server Configuration: Service Provider (Section 7.4.3).**
- **URI Group: ***
- **Transport: ***
- **Remote Subnet: ***
- **Received Interface: Private_sig (Section 7.6.3).**
- **Signaling Interface: Public_sig (Section 7.6.3).**
- **Media Interface: Public_med (Section 7.6.2).**
- **Secondary Media Interface: None.**
- **End Point Policy Group: Service Provider (Section 7.5.3).**
- **Routing Profile: Route_to_IPO_TLS (Section 7.4.4).**
- **Topology Hiding Profile: Service_Provider (Section 7.4.5).**
- **Check Link Monitoring from Peer.**
- Click **Finish**.

Note – Ensure “Link Monitor from Peer” is checked. Selecting **Link Monitoring from Peer** enables Avaya SBC to send a 200 OK response for a match of the SIP OPTIONS request with a server flow. If you don’t enable **Link Monitoring from Peer**, then OPTIONS request will be relayed to the destination server (IP Office).

Edit Flow: SP to IPO Flow X

Flow Name	SP to IPO Flow
SIP Server Profile	Service Provider
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Private_sig
Signaling Interface	Public_sig
Media Interface	Public_med
Secondary Media Interface	None
End Point Policy Group	Service Provider
Routing Profile	Route_to_IPO_TLS
Topology Hiding Profile	Service_Provider
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input checked="" type="checkbox"/>
FQDN Support	<input type="checkbox"/>
FQDN	

To create the call flow toward IP Office, click **Add** (not shown).

- **Name: IPO to SP Flow.**
- **Server Configuration: IP Office-Thornton (Section 7.4.3).**
- **URI Group: ***
- **Transport: ***
- **Remote Subnet: ***
- **Received Interface: Public_sig (Section 7.6.3).**
- **Signaling Interface: Private_sig (Section 7.6.3).**
- **Media Interface: Private_med (Section 7.6.2).**
- **Secondary Media Interface: None.**
- **End Point Policy Group: Enterprise (Section 7.5.3).**
- **Routing Profile: Route to SP (Section 7.4.4).**
- **Topology Hiding Profile: IP Office (Section 7.4.5).**
- **Click Finish.**

Edit Flow: IPO to SP Flow	
Flow Name	IPO to SP Flow
SIP Server Profile	IP Office-Thornton
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Public_sig
Signaling Interface	Private_sig
Media Interface	Private_med
Secondary Media Interface	None
End Point Policy Group	Enterprise
Routing Profile	Route to SP
Topology Hiding Profile	IP Office
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>
FQDN Support	<input type="checkbox"/>
FQDN	
Finish	

The following screen capture shows the newly created **End Point Flows**.

Avaya Session Border Controller

Device: Avaya_SBC | Alarms 1 | Incidents | Status | Logs | Troubleshooting | Users | Settings | Help | Log Out

End Point Flows

Subscriber Flows | **Server Flows**

Filter Add

Modifications made to a Server Flow will only take effect on new sessions.

Click here to add a row description.

SIP Server: IP Office-Thornton

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	
1	IPO to SP Flow	*	Public_sig	Private_sig	Enterprise	Route to SP	View Clone Edit Delete

SIP Server: Service Provider

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	
1	SP to IPO Flow	*	Private_sig	Public_sig	Service Provider	Route_to_IPO_TLS	View Clone Edit Delete

8. WorldNet Telecommunications SIP Trunking Service Configuration

To use WorldNet Telecommunications SIP Trunking Service, a customer must request the service from WorldNet Telecommunications using the established sales processes. The process can be started by contacting WorldNet Telecommunications via the corporate web site at: <https://www.worldnetpr.com/en/voice-service/> and requesting information.

During the signup process, WorldNet Telecommunications and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to WorldNet Telecommunications network.

WorldNet Telecommunications is responsible for the configuration of WorldNet Telecommunications SIP Trunking Service. The customer will need to provide the public IP address used to reach the Avaya Session Border Controller at the enterprise, the public IP address assigned to interface B1.

WorldNet Telecommunications will provide the customer the necessary information to configure Avaya IP Office and the Avaya Session Border Controller following the steps discussed in the previous sections, including:

WorldNet Telecommunications will provide the following information:

- SIP Trunk registration credentials (User Name, Password, etc.).
- WorldNet's SIP Proxy IP address.
- Supported audio codecs and their prefer order.
- DID numbers, etc.

9. Verification Steps

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

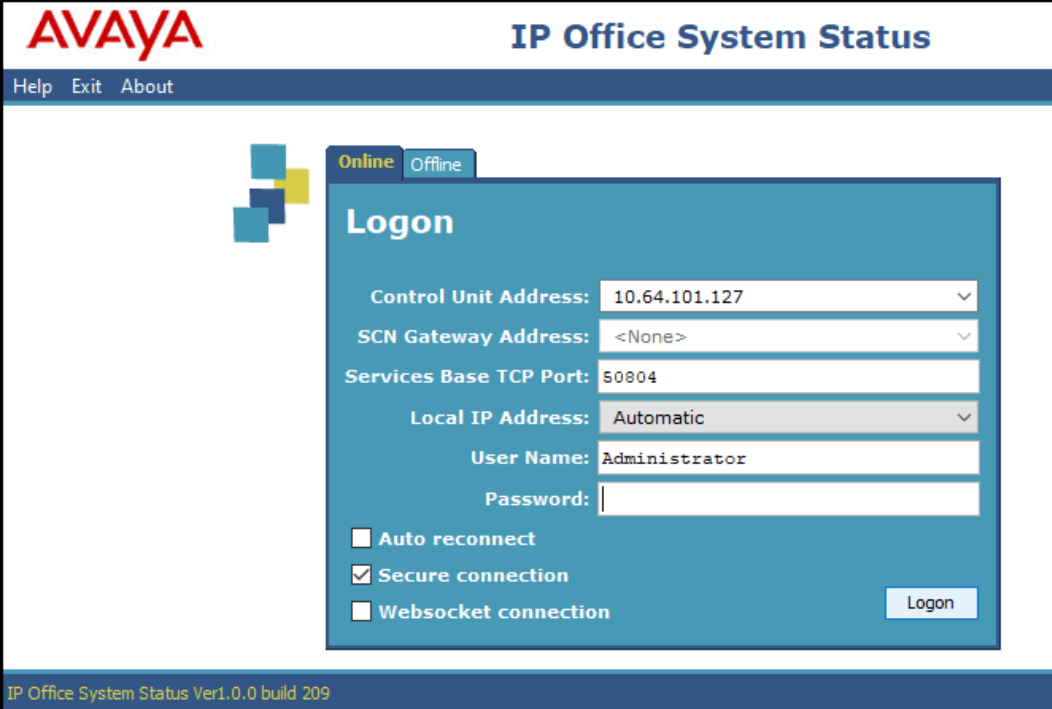
The following steps may be used to verify the configuration:

- Verify that endpoints at the enterprise site can place calls to the PSTN.
- Verify that endpoints at the enterprise site can receive calls from the PSTN.
- Verify that users at the PSTN can end active calls to endpoints at the enterprise by hanging up.
- Verify that endpoints at the enterprise can end active calls to PSTN users by hanging up.

9.1. IP Office System Status

The following steps can also be used to verify the configuration.

Use the IP Office **System Status** application to verify the state of SIP connections. Launch the application from **Start** → **Programs** → **IP Office** → **System Status** on the PC where IP Office Manager is installed, log in with the proper credentials.



The screenshot shows the AVAYA IP Office System Status application interface. At the top left is the AVAYA logo, and at the top right is the title "IP Office System Status". Below the title is a menu bar with "Help", "Exit", and "About". The main area features a "Logon" dialog box with a status indicator showing "Online" and "Offline" tabs. The dialog box contains the following fields and options:

- Control Unit Address: 10.64.101.127
- SCN Gateway Address: <None>
- Services Base TCP Port: 50804
- Local IP Address: Automatic
- User Name: Administrator
- Password: (empty field)
- Auto reconnect:
- Secure connection:
- Websocket connection:

A "Logon" button is located at the bottom right of the dialog box. At the bottom of the application window, the text "IP Office System Status Ver1.0.0 build 209" is displayed.


Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current State** is **Idle** for each channel.

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (24)
- Extensions (3)
- Trunks (3)
 - Line: 1
 - Line: 2
 - Line: 17**
- Active Calls
- Resources
- Voicemail
- IP Networking
- Locations

SIP Trunk Summary

Line Service State: In Service
 Peer Domain Name: sip://10.64.101.243
 Resolved Address: 10.64.101.243
 Line Number: 17
 Number of Administered Channels: 10
 Number of Channels in Use: 0
 Administered Compression: G722, G711 Mu
 Silence Suppression: Off
 Media Stream: RTP
 Layer 4 Protocol: TLS
 SIP Trunk Channel Licenses: 256  0%
 SIP Trunk Channel Licenses in Use: 0
 SIP Device Features: REFER (Incoming and Outgoing), UPDATE (Incoming and Outgoing)

Cha...	U..	Call	Cur...	Time	Rem...	C...	Con...	Caller	Other	Dire...	Rou...	Rec...	Rec...	Tra...	Tra...
Ref			in S...					ID ...	Party o...						
1			Idle	1 d...											
2			Idle	1 d...											
3			Idle	7 d...											
4			Idle	7 d...											
5			Idle	7 d...											
6			Idle	7 d...											
7			Idle	7 d...											
8			Idle	7 d...											
9			Idle	7 d...											
10			Idle	7 d...											

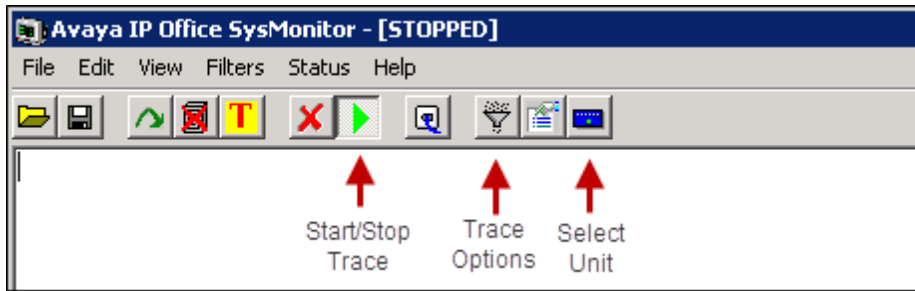
Trace Trace All Pause Ping Call Details Graceful Shutdown

Force Out of Service Print... Save As...

2:37:07 PM Online

9.2. Monitor

The Avaya IP Office Monitor application can be used to monitor and troubleshoot signaling messaging on the SIP trunk. Launch the application from **Start → Programs → IP Office → Monitor** on the PC where 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.



Clicking the **Trace Options** icon on the taskbar, selecting the **SIP** tab allows modifying 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.



9.3. Avaya Session Border Controller

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 Avaya SBC.

The screenshot shows the Avaya Session Border Controller web interface. At the top, there is a navigation bar with the following items: Device: Avaya_SBC, Alarms, Incidents, Status, Logs, Troubleshooting, Users, Settings, Help, and Log Out. A red arrow points to the 'Alarms' link. Below the navigation bar, the main header reads 'Avaya Session Border Controller' with the AVAYA logo on the right. On the left side, there is a sidebar menu with the following items: EMS Dashboard, Software Management, Device Management (highlighted in red), Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, TLS Management, Network & Flows, DMZ Services, Monitoring & Logging, and Compliance. The main content area is titled 'Device Management' and contains several tabs: Devices, Updates, Licensing, Key Bundles, and License Compliance. The 'Devices' tab is active, displaying a table with the following data:

Device Name	Management IP	Version	Status						
Avaya_SBC	10.64.101.242	10.2.0.0-86-24077	Commissioned	Reboot	Shutdown	Restart Application	View	Edit	Uninstall

The following screen shows the **Alarm Viewer** page.

The screenshot shows the Avaya Alarm Viewer web interface. At the top, there is a navigation bar with the following items: Device: Avaya_SBC and Help. The main header reads 'Alarm Viewer' with the AVAYA logo on the right. Below the header, there is a section titled 'Alarms' with a table containing the following columns: ID, Details, State, Time, and Device. The table is currently empty, displaying the message 'No alarms found for this device.' Below the table, there are two buttons: 'Clear Selected' and 'Clear All'.

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

The screenshot shows the Avaya Session Border Controller web interface. At the top, a navigation bar includes 'Device: Avaya_SBC', 'Alarms', 'Incidents' (highlighted with a red arrow), 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header displays 'Avaya Session Border Controller' and the 'AVAYA' logo. On the left, a sidebar menu lists various management options, with 'Device Management' selected. The main content area is titled 'Device Management' and contains several tabs: 'Devices', 'Updates', 'Licensing', 'Key Bundles', and 'License Compliance'. The 'Devices' tab is active, showing a table with the following data:

Device Name	Management IP	Version	Status						
Avaya_SBC	10.64.101.242	10.2.0.0-86-24077	Commissioned	Reboot	Shutdown	Restart Application	View	Edit	Uninstall

The following screen shows the Incident Viewer page.

The screenshot displays the 'Incident Viewer' page in the Avaya Session Border Controller interface. The top navigation bar shows 'Device: Avaya_SBC' and a 'Help' link. The page title is 'Incident Viewer' with the 'AVAYA' logo. Below the title, there is a 'Category' dropdown menu set to 'All', a 'Clear Filters' button, and 'Refresh' and 'Generate Report' buttons. A 'Summary' tab is selected, and the page indicates 'Displaying entries 1 to 15 of 2000'. A table lists the following incidents:

ID	Date & Time	Category	Type	Cause
861479021261184	Aug 6, 2024 11:27:22 AM	Policy	Routing Failure	Timeout while contacting DNS serverssip.clearcom.mx
861478979865000	Aug 6, 2024 11:25:59 AM	Policy	Message Dropped	No Subscriber Flow Matched

Status: Provides the status for each server handling calls to/from the PSTN.

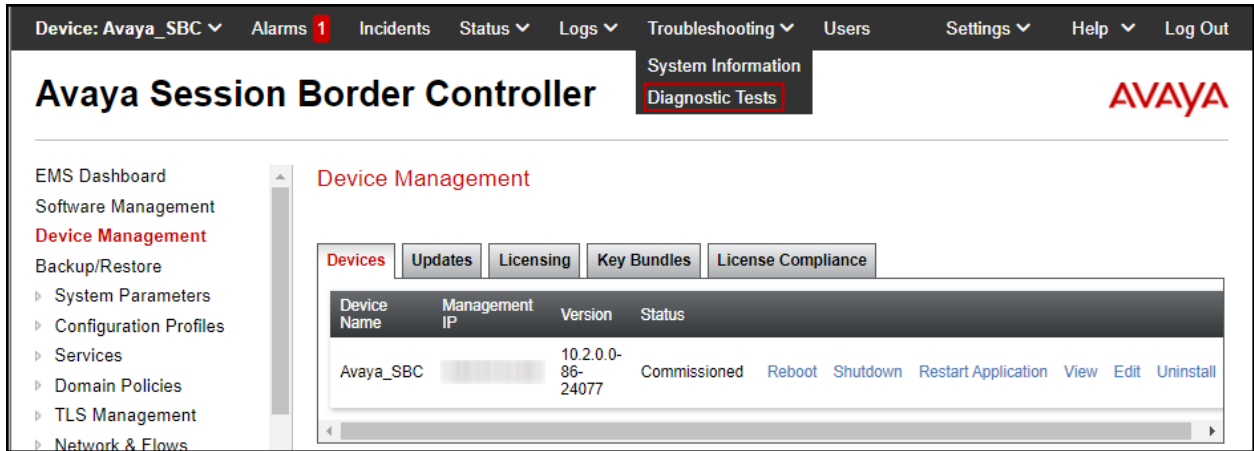
The screenshot shows the Avaya SBC management interface. At the top, there is a navigation bar with the following items: Device: Avaya_SBC, Alarms 1, Incidents, Status (highlighted with a red arrow), Logs, Troubleshooting, Users, Settings, Help, and Log Out. Below the navigation bar is the main header 'Avaya Session Border Controller' with the AVAYA logo on the right. On the left side, there is a sidebar menu with the following items: EMS Dashboard, Software Management, Device Management (highlighted in red), Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, TLS Management, Network & Flows, and DMZ Services. The main content area is titled 'Device Management' and contains several tabs: Devices (selected), Updates, Licensing, Key Bundles, and License Compliance. Below the tabs is a table with the following data:

Device Name	Management IP	Version	Status						
Avaya_SBC	10.64.101.242	10.2.0.0-86-24077	Commissioned	Reboot	Shutdown	Restart Application	View	Edit	Uninstall

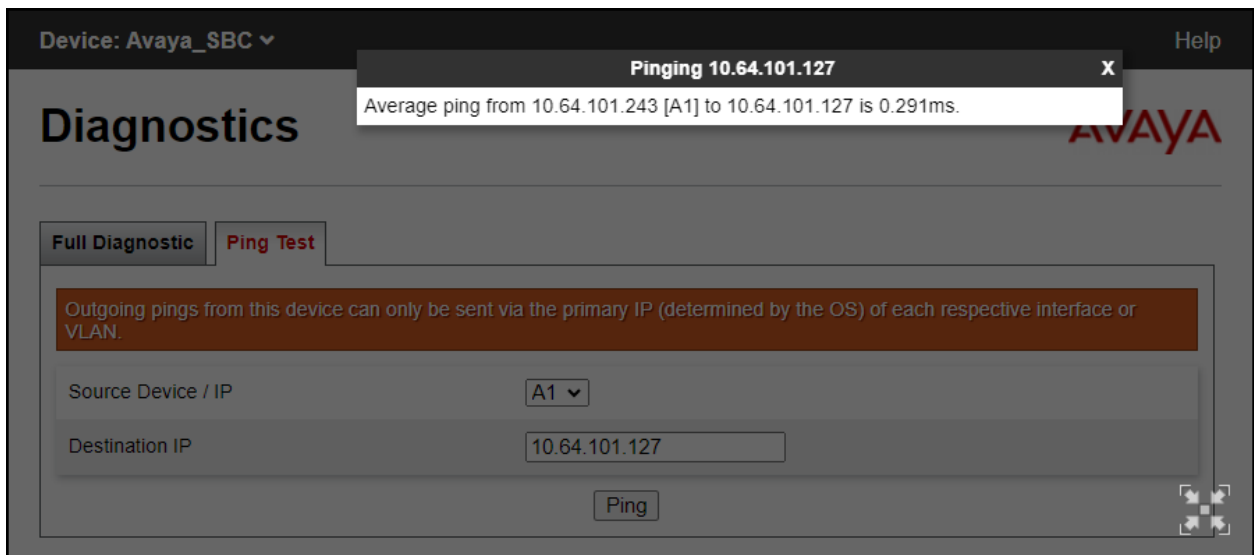
The screenshot shows the 'Status' page in the Avaya SBC management interface. At the top, there is a navigation bar with the following items: Device: Avaya_SBC and Help. Below the navigation bar is the main header 'Status' with the AVAYA logo on the right. Below the header is a section titled 'Server Status' which contains a table with the following data:

Server Profile	Server FQDN	Server IP	Server Port	Server Transport	Heartbeat Status	Registration Status	TimeStamp
Service Provider	96.97	96.97	5060	UDP	UNKNOWN	REGISTERED	08/21/2024 15:50:37 EDT

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



The following screen shows the Diagnostics page with the results of a ping test.



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

The screenshot displays the Avaya Session Border Controller (SBC) web interface. At the top, a navigation bar includes 'Device: Avaya SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Avaya Session Border Controller' with the AVAYA logo on the right. A left-hand navigation menu lists various management options, with 'Monitoring & Logging' expanded to show 'Trace' as the active selection. The main content area is titled 'Trace: Avaya SBC' and features two tabs: 'Packet Capture' (active) and 'Captures'. Below the tabs is a 'Packet Capture Configuration' form with the following fields:

Packet Capture Configuration	
Status	Ready
Interface	Any
Local Address <small>IP[:Port]</small>	All : <input type="text"/>
Remote Address <small>*, *:Port, IP, IP:Port</small>	<input type="text"/>
Protocol	All
Maximum Number of Packets to Capture	10000
Capture Filename <small>Using the name of an existing capture will overwrite it.</small>	Test1.pcap

At the bottom of the configuration form are two buttons: 'Start Capture' and 'Clear'.

Once the capture is stopped, click on 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.

The screenshot shows the Avaya Session Border Controller web interface. The top navigation bar includes 'Device: Avaya SBC', 'Alarms', 'Incidents', 'Status', 'Logs', 'Troubleshooting', 'Users', 'Settings', 'Help', and 'Log Out'. The main header displays 'Avaya Session Border Controller' and the 'AVAYA' logo. A left sidebar lists various management options, with 'Trace' highlighted in red. The main content area is titled 'Trace: Avaya SBC' and features two tabs: 'Packet Capture' and 'Captures'. The 'Captures' tab is active, showing a table of capture files with columns for 'File Name', 'File Size (bytes)', and 'Last Modified'. Each row includes a 'Delete' link.

File Name	File Size (bytes)	Last Modified	
Test1_20240723155527	221,184	July 23, 2024 at 3:55:41 PM MDT	Delete
Test1_20240501132013	376,832	May 1, 2024 at 1:20:40 PM MDT	Delete
OPTIONS1	2,975	August 4, 2023 at 7:56:59 AM MDT	Delete
test2	4,362	August 4, 2023 at 6:51:03 AM MDT	Delete
test1	6,188	August 4, 2023 at 6:48:20 AM MDT	Delete

Also, the **traceSBC** tool can be used to monitor the SIP signaling messages between the Service provider and the Avaya SBC.

10. Conclusion

These Application Notes describe the procedures required to configure Avaya IP Office Release 12.0 and Avaya Session Border Controller Release 10.2 to connect to WorldNet Telecommunications SIP Trunking Service. WorldNet Telecommunications SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks.

Interoperability testing was completed successfully with the observations/limitations outlined in the scope of testing in **Section 2.1** as well as under test results in **Section 2.2**.

11. Additional References

This section references the documentation relevant to these Application Notes. Product documentation for Avaya IP Office, including the following, is available at:

<http://support.avaya.com/>

- [1] *Deploying IP Office Server Edition and Application Servers, Release 12.0, Issue 31, April 2024*
- [2] *IP Office Platform 12.0, Deploying Avaya IP Office Servers as Virtual Machines, June 2024*
- [3] *Avaya IP Office Platform Server Edition Reference Configuration Release 12.0, Issue 22, May 2024*
- [4] *IP Office Platform 12.0, Deploying an IP500 V2 IP Office Basic Edition System, Issue 41e, May 29, 2024*
- [5] *IP Office Platform 12.0, Deploying an IP500 V2 IP Office Essential Edition System, Issue 41e, May 29, 2024*
- [6] *Administering Avaya IP Office using Manager, Release 12.0, Issue 51.1.2, June 2024.*
- [7] *Administering Avaya IP Office with Web Manager, Release 12.0, Issue 46.1.1, May 2024.*
- [8] *Avaya IP Office Platform Feature Description, Release 12.0, Issue 21.1.1, May 2024.*
- [9] *Planning for and Administering Avaya Workplace Client for Android, iOS, Mac and Windows, September 2020*
- [10] *Deploying Avaya Session Border Controller on a Virtualized Environment Platform, Release 10.2, Issue 1, March 2024.*

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