DevConnect Program

Application Notes for Configuring Avaya IP Office Release 12.0 to support WorldNet Telecommunications SIP Trunking Service - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service on an enterprise solution consisting of Avaya IP Office 12.0 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 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) 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 the WorldNet Telecommunications 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.

User Data Protocol (UDP) and Real-time Transport Protocol (RTP) were used between the simulated enterprise site and the WorldNet Telecommunications network (public network side),

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Transport Layer Security (TLS) and Secure Real-Time Transport Protocol (SRTP) were used inside of the enterprise (private network side).

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).
- Dialing plans including local calls, international calls, outbound toll-free, etc.
- 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.).
- 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.
- G.711 pass-through fax.

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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.

- **Call transfer to the PSTN using the SIP REFER method** – Calls from the PSTN to the enterprise that were transferred back to the PSTN network using the SIP REFER method did not work properly. Due to this reason, REFER was left disabled in Avaya IP Office for the tests (refer to **Sections 5.4.2)**.
- **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 was observed from the PSTN to Workplace. The issue only occurs when **Media Security**, under **System → VoIP → VoIP Security** tab, is enabled in IP Office (e.g., set to "**Preferred**"). When set to "**Disabled**" there were no issues, audio was good in both directions. This issue is under investigation by Avaya. As a temporary work around, if TLS/SRTP is being use in IP Office, set **Media Security**, under **System → VoIP → VoIP Security** tab in IP Office, to **Disabled** (refer to **Section 5.2.3.2**).

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2.3. Support

For support on WorldNet Telecommunications SIP Trunking Service 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 testing. 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 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 used to connect to the public network.

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.

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

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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.

For inbound calls, the calls flowed from the WorldNet Telecommunications network 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 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 the 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 were masked.

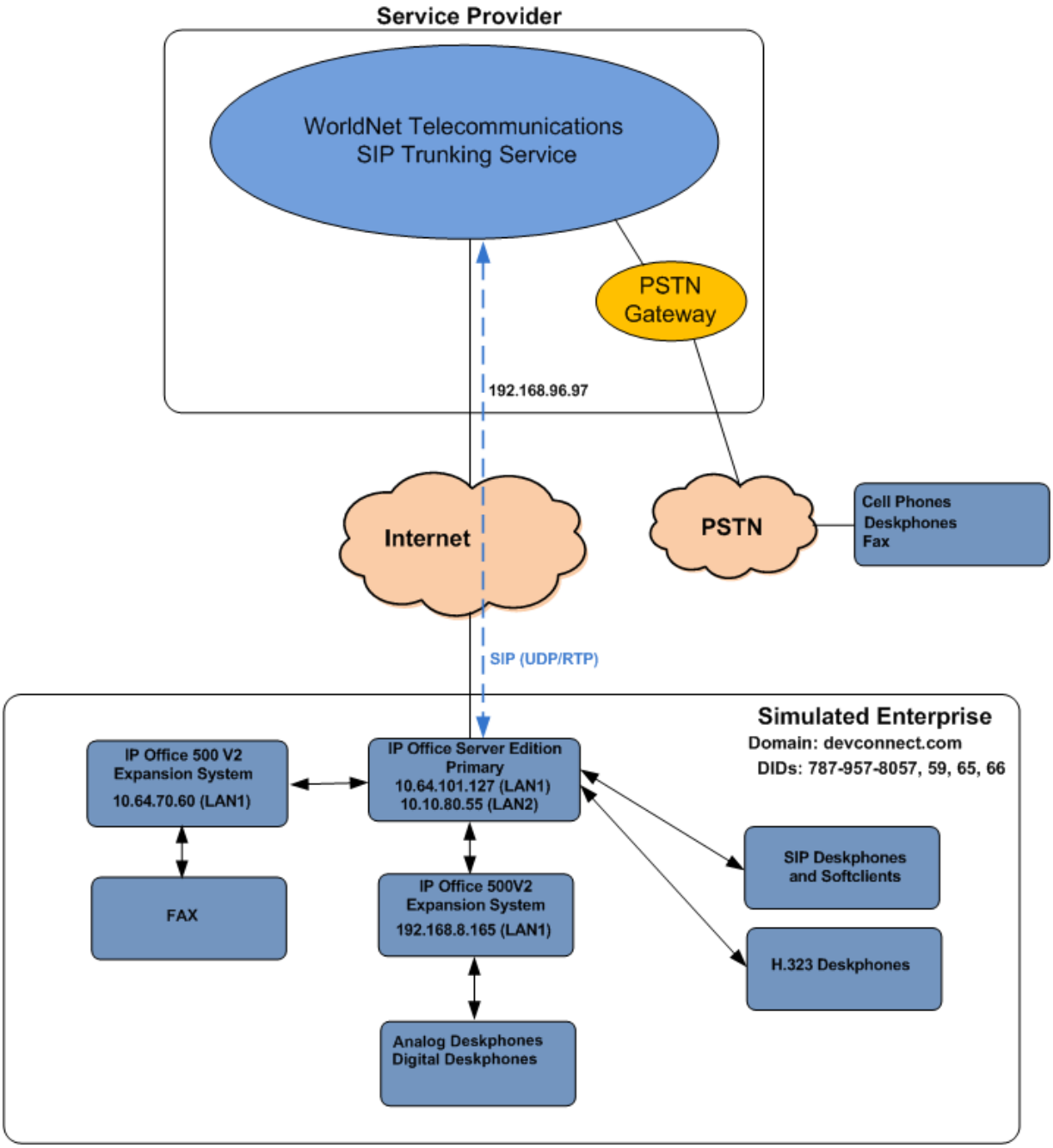

Figure 1: Avaya DevConnect lab Configuration

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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) • Avaya IP Office Voicemail Pro	12.0.0.0 Build 55 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 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.

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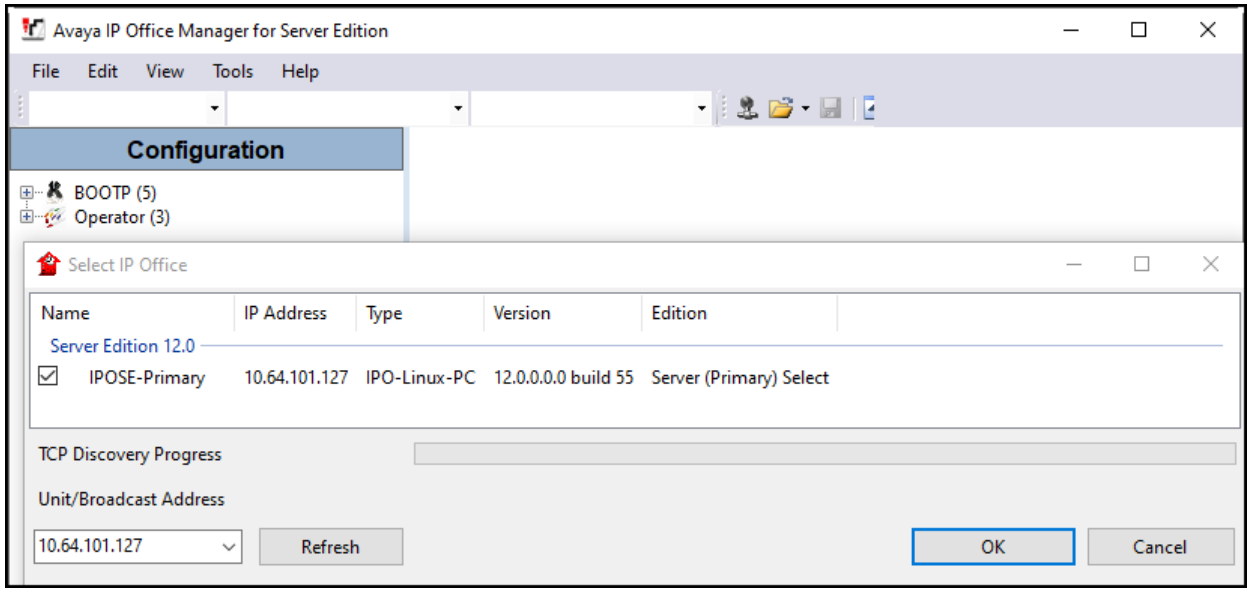
5. Avaya IP Office Primary Server Configuration

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.



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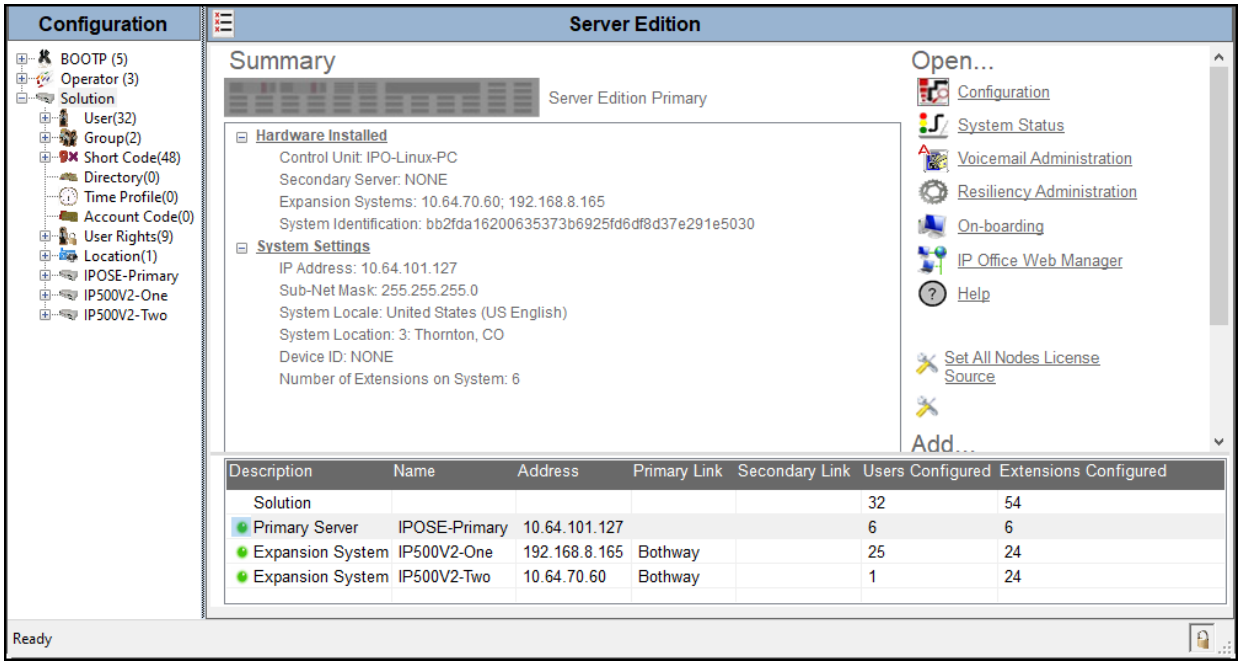
On Server Edition systems, the Solution View screen will appear, similar to the one shown below. All the Avaya IP Office configurable components are shown in the left pane, known as the Navigation Pane. Clicking the “plus” sign next to the Primary server system name, e.g., **IPOSE-Primary**, on the navigation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



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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.

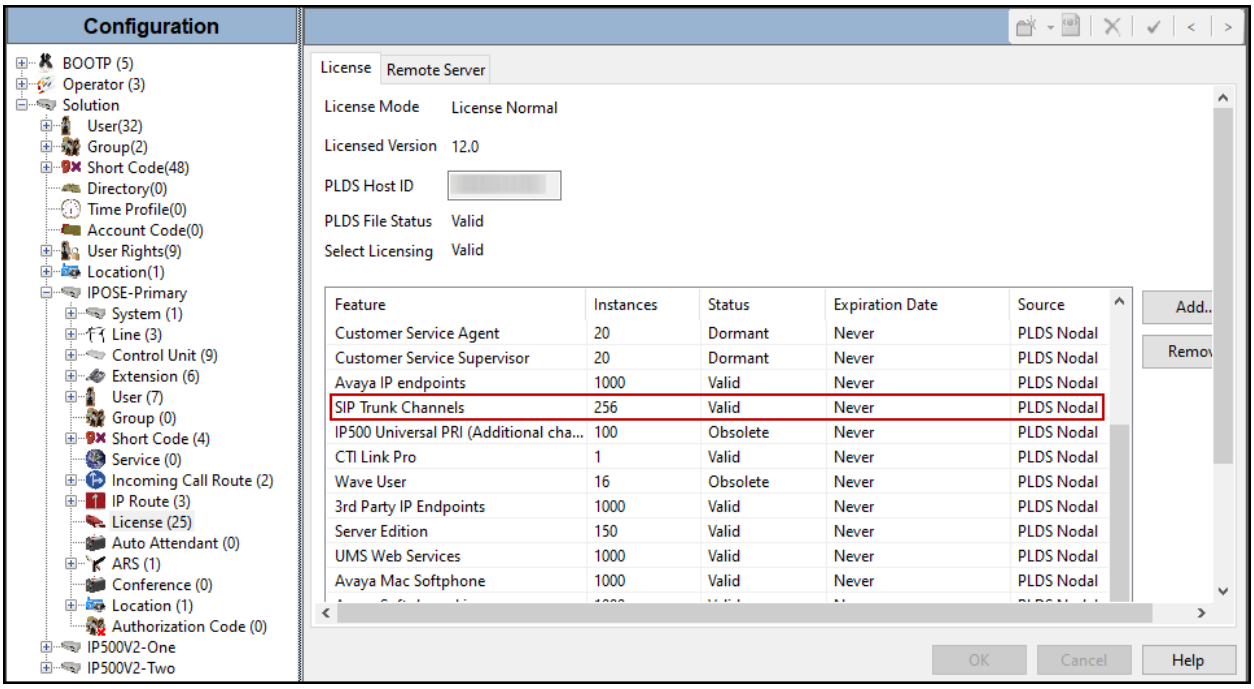
Standard feature configurations that are not directly related to the interfacing with the service provider are assumed to be already in place, and they are not part of these Application Notes.

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.



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5.2. System Settings

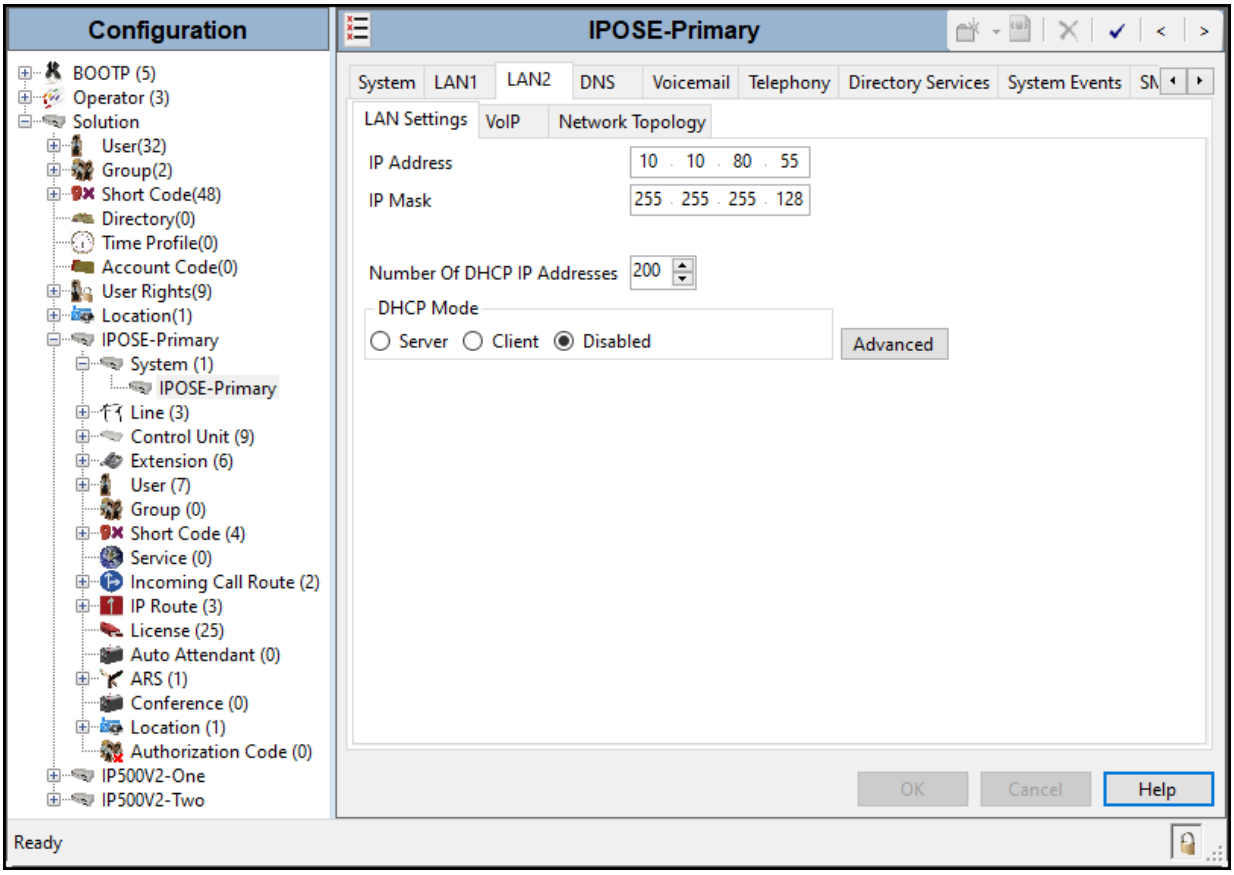
Configure the necessary system settings. In an Avaya IP Office, the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side).

5.2.1. System – LAN2 Tab

In the sample configuration, **IPOSE-Primary** was used as the system name, the **LAN2** port is used for the SIP trunk connection to WorldNet Telecommunications.

To view or configure the LAN2 port IP address and subnet mask, select the **LAN2→ LAN Settings** tab, and enter the information as needed, according to the customer network requirements:

- Set the **IP Address** field to the LAN IP address, e.g., **10.10.80.55**.
- Set the **IP Mask** field to the subnet mask of the enterprise private network, e.g., **255.255.255.128**.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit (not shown).



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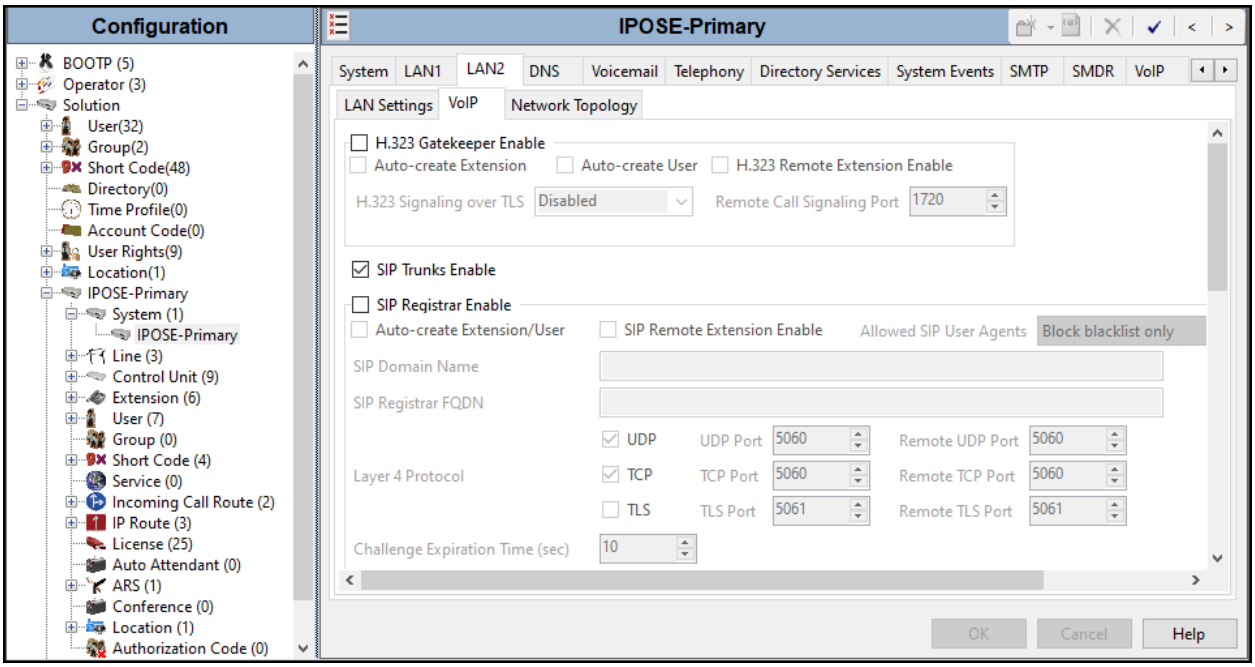
5.2.1.1 LAN2 VoIP Tab

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

- Select the **LAN2 → VoIP** tab in the Details Pane. Check the **SIP Trunks Enable** box to allow the configuration of SIP trunks. Since no SIP endpoints are to register on this interface (SIP endpoints register to the **LAN1**), leave the **SIP Registrar Enable** box unchecked.



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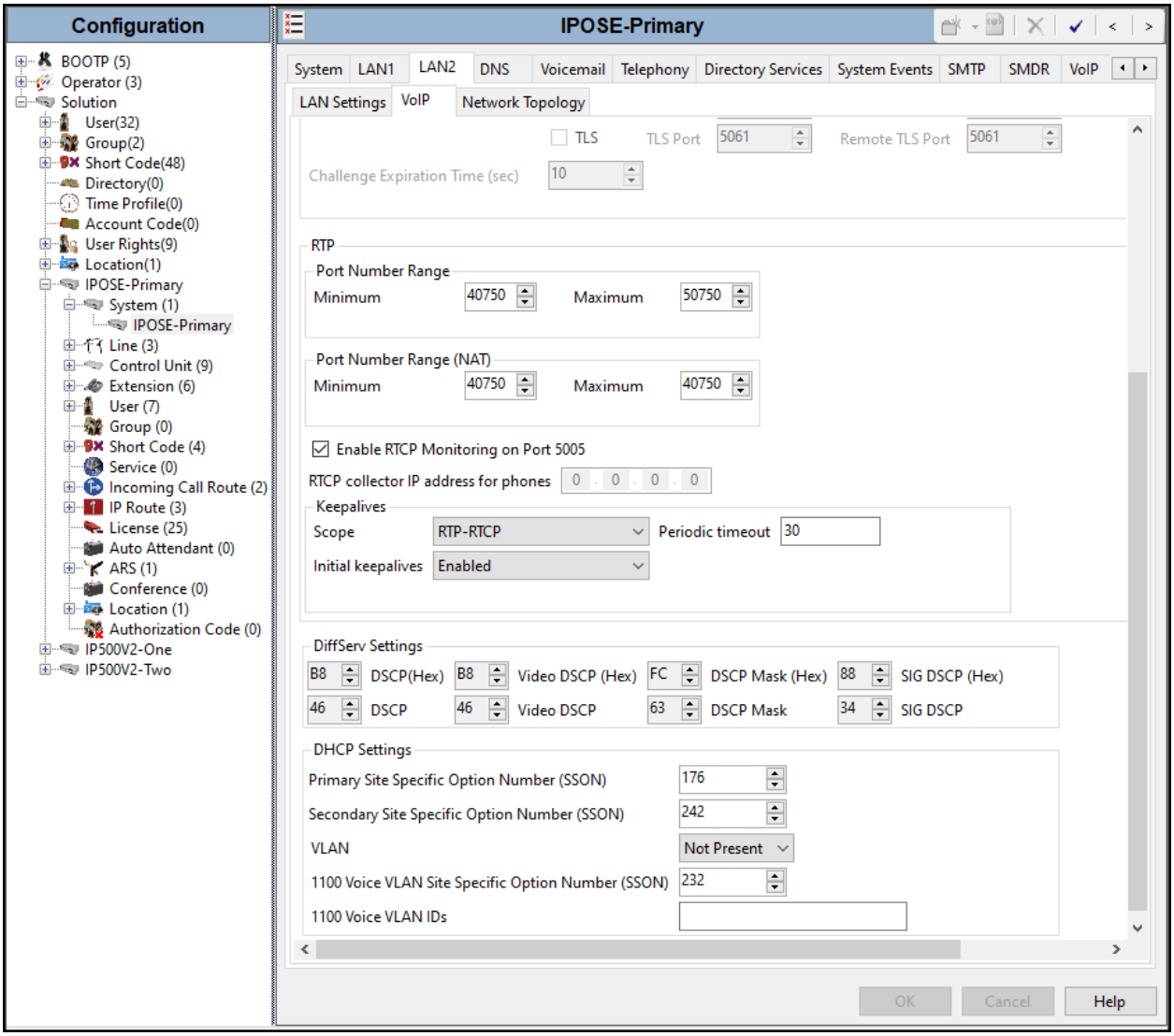
Scroll down the page:

- Verify the **RTP Port Number Range**. Based on this setting, Avaya IP Office will request RTP media in the configurable range for calls using LAN2. The **Minimum** and **Maximum** port numbers were kept at their default values in the reference configuration.
- In the **Keepalives** section, set the **Scope** to **RTP-RTCP**. Set the **Periodic timeout** to **30** and the **Initial keepalives** parameter to **Enabled**. This is done to prevent possible issues with network firewalls closing idle RTP channels.
- In the **DiffServ Settings** section, IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services (QoS) policies for both signaling and media. The **DSCP** field is the value used for media, while the **SIG DSCP** is the value used for signaling. These settings should be set according to the customer's QoS policies in place. The default values used during the compliance test are shown.
- Click **OK** to commit.



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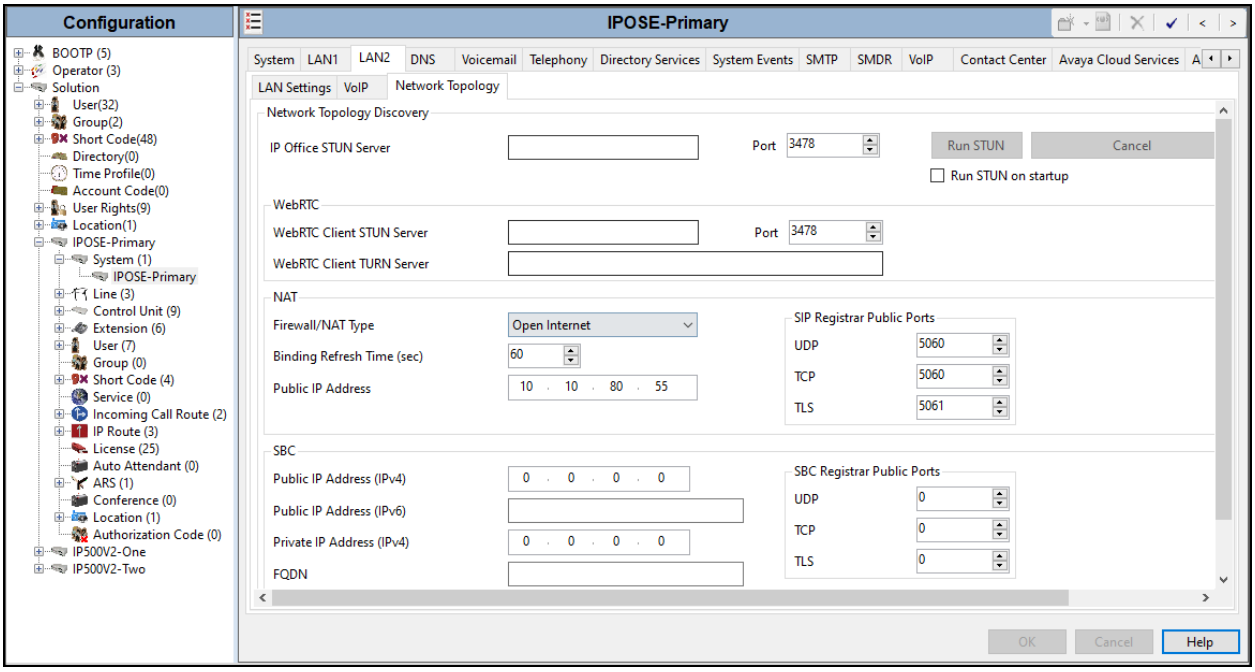
Note: In the compliance test, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network (private network). The LAN1 interface configuration is not directly relevant to the interface with the WorldNet Telecommunications SIP Trunking Service, and therefore is not described in these Application Notes.

5.2.1.2 LAN2 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** was set to the public IP address of **LAN2** (e.g., **10.10.80.55**) and **Public Port** and the standard UDP, TCP and TLS ports, as shown.
- Click **OK** to commit.



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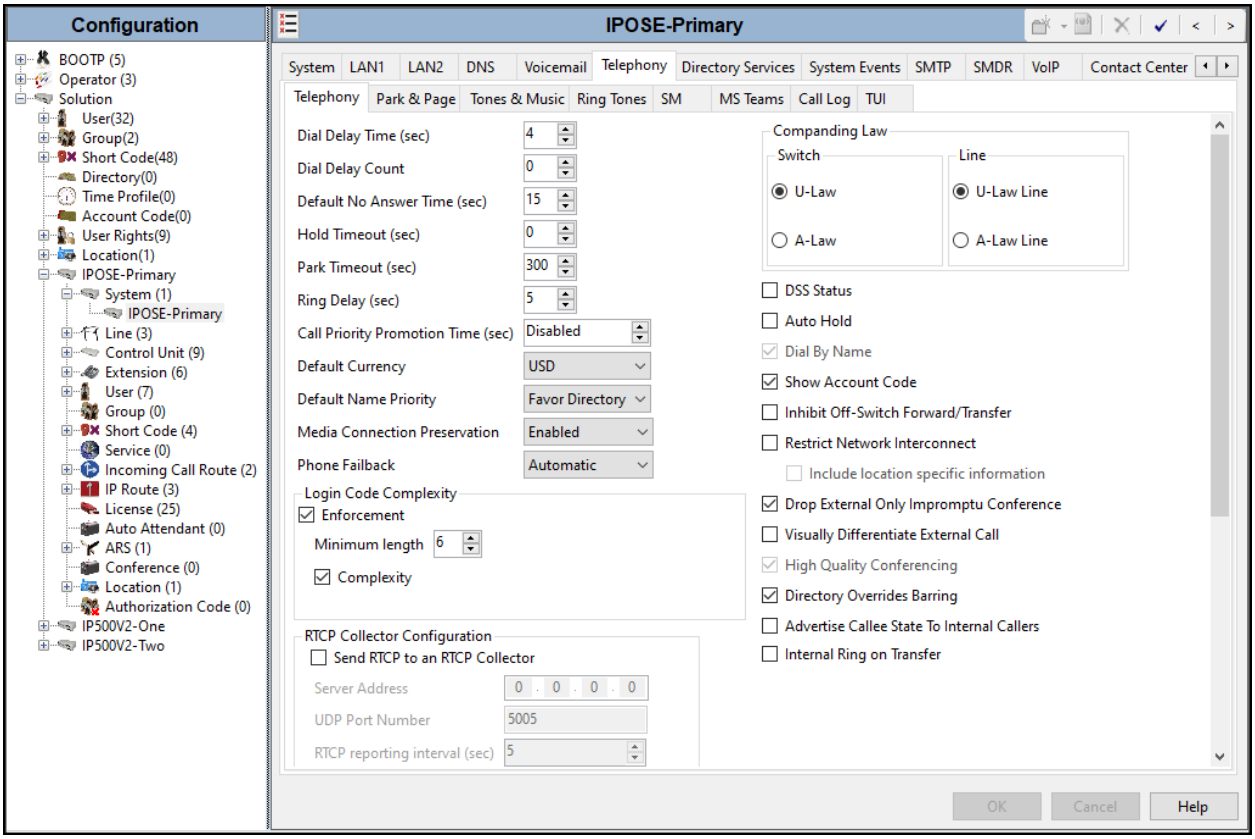
5.2.2. System - Telephony Tab

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.



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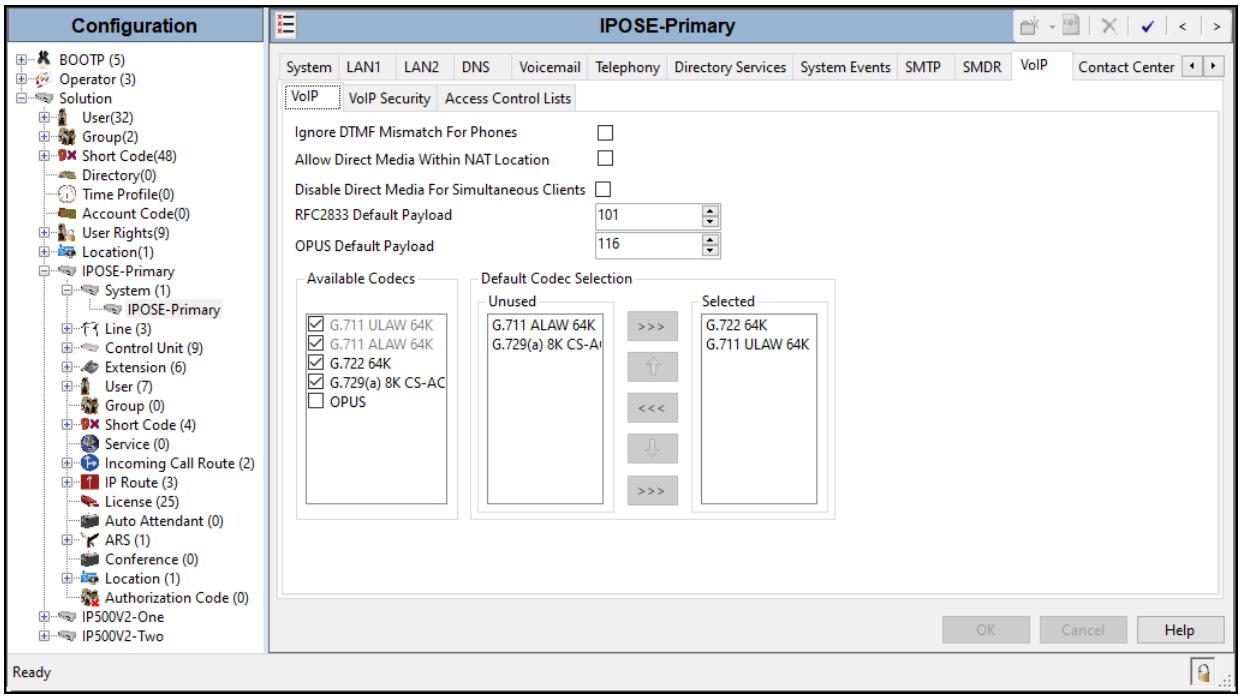
5.2.3. System - VoIP Tab

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 default 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).
- 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 5.4.6** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

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5.2.3.2 VoIP – VoIP Security Tab

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

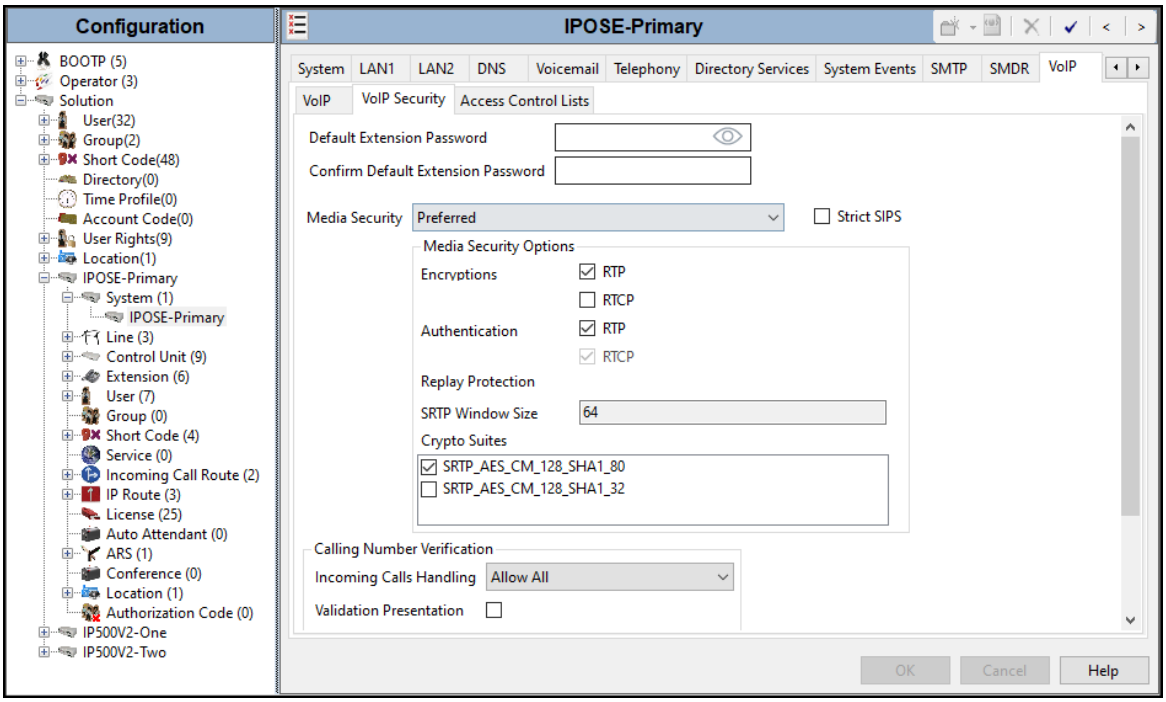
Configuring the use of SRTP at the system level is done on the **VoIP Security** tab using the Media Security setting. The options are:

- 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.



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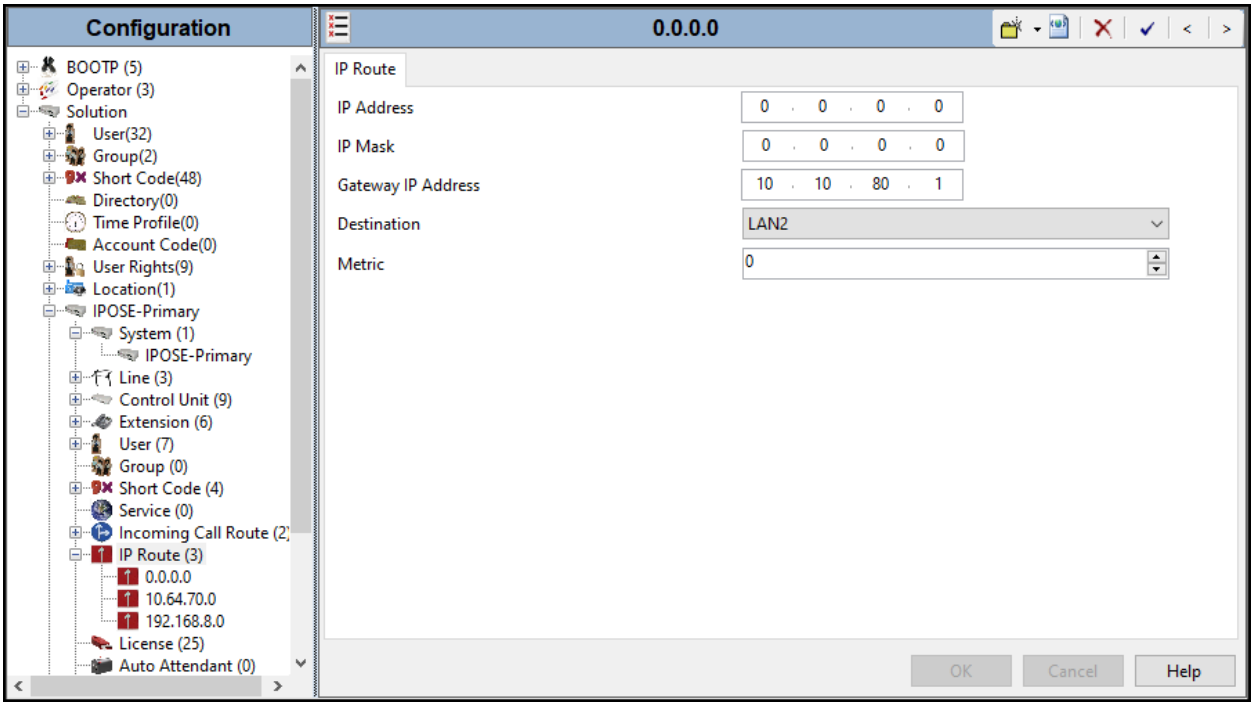
5.3. IP Route

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., **10.10.80.1**.
- Set **Destination** to **LAN2** from the pull-down menu.
- Click **OK** to commit.



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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 **5.4.7**.

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 **5.4.7**.

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.

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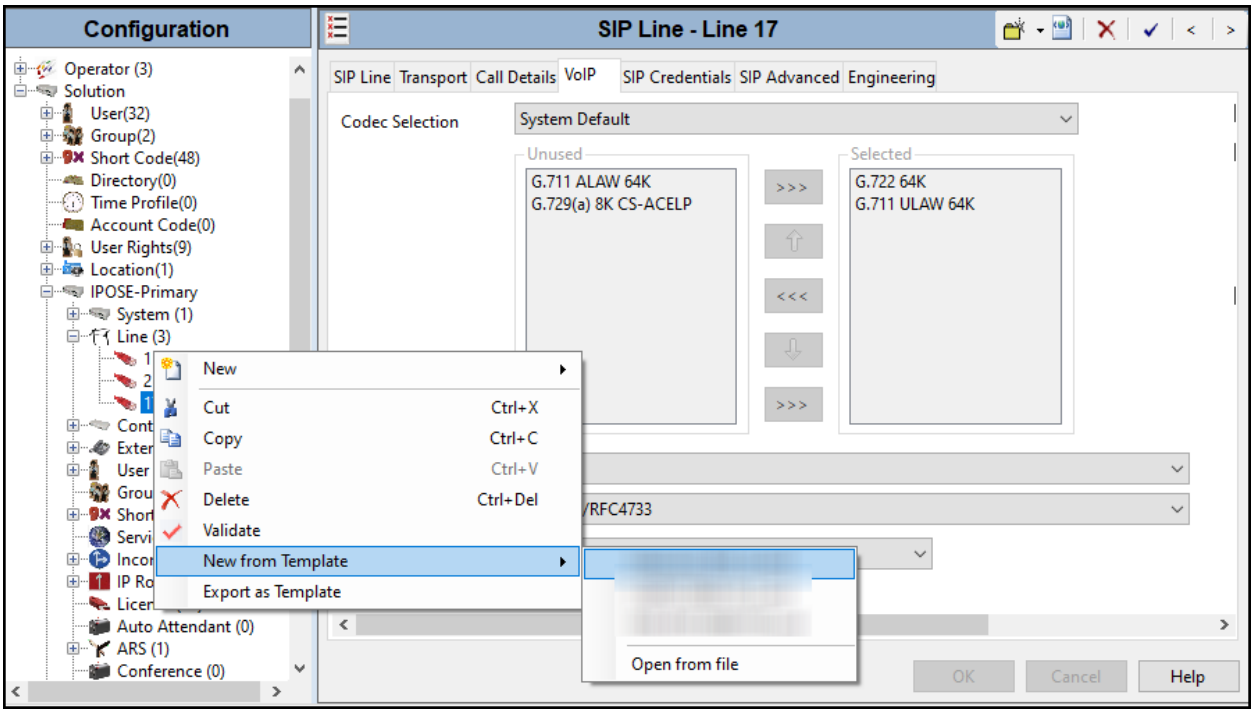
To create the SIP Trunk from the template, from the **Primary** server, right-click on **Line** in the Navigation Pane, then navigate to **New → New from Template→Open from file**.



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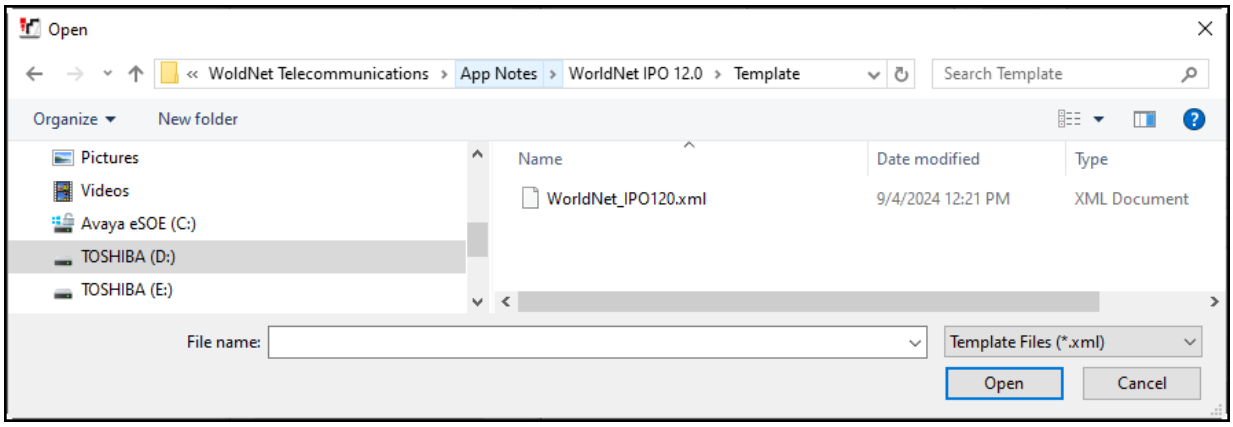
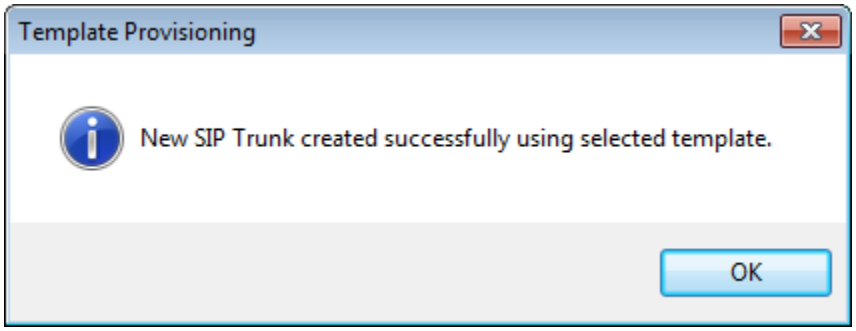
Navigate to the directory on the local machine where the template was copied and select the template.



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



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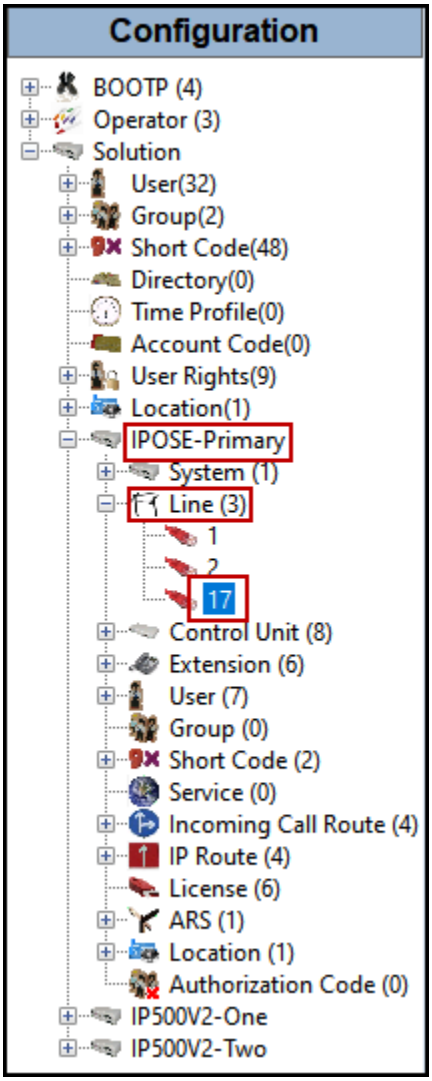
The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line **17**).



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 **5.4.7**.

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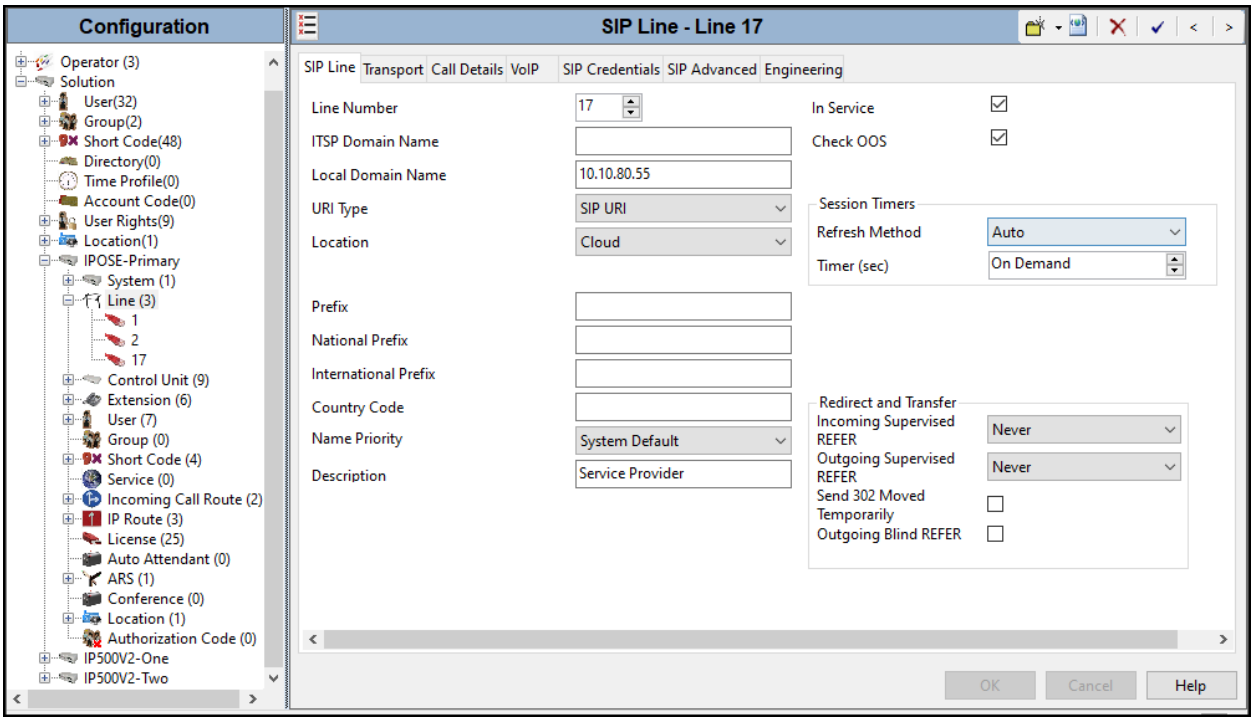
5.4.2. SIP Line – SIP Line Tab

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.
- Under **Local Domain Name** enter the public IP address assigned to the LAN2 interface (**Section 5.2.1**).
- 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 **Never** (refer to **Section 2.2)**.
- Click **OK** to commit.



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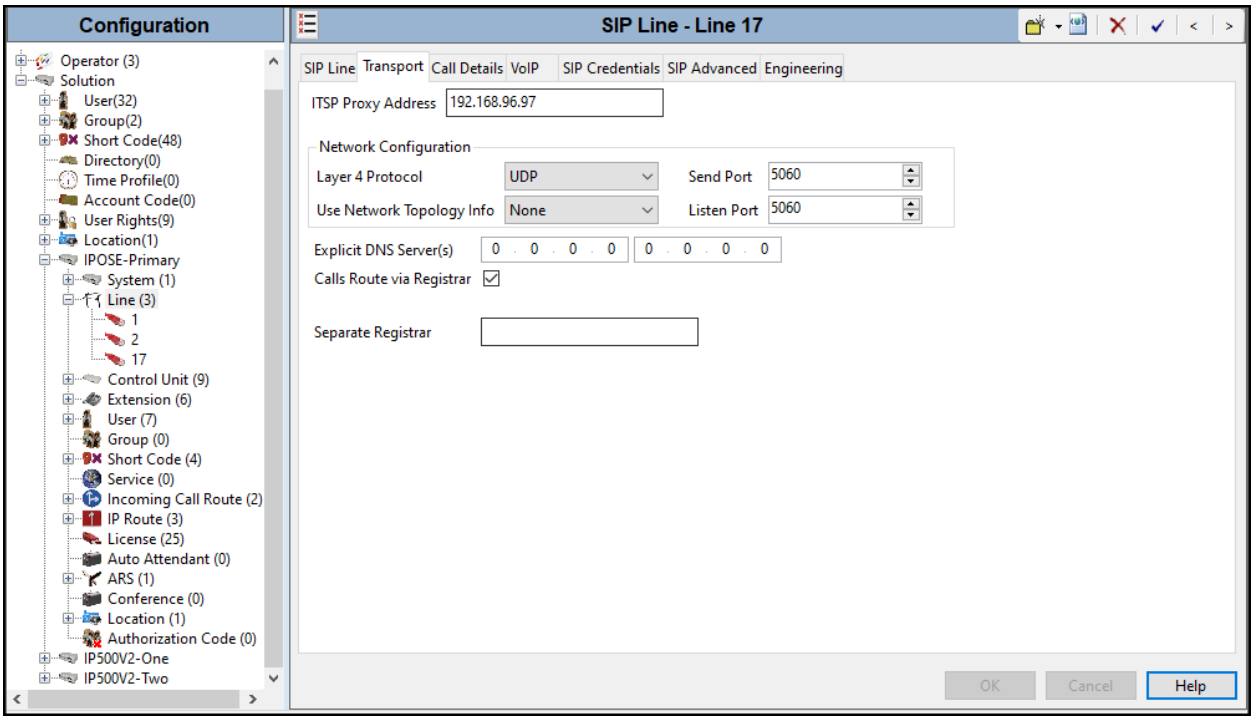
5.4.3. SIP Line - Transport Tab

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

- Set the **ITSP Proxy Address** to the public IP address (signaling IP address) of WorldNet's SIP Proxy server (**192.168.96.97**) as shown in **Figure 1**. This information should be provided by WorldNet.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **None** (see note below).
- Set the **Send** and **Listening Ports** to **5060**.
- 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. 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 or LAN2) used by the trunk and the **System → LAN1 (or 2) → Network Topology** tab needs to be configured with the details of the NAT device.

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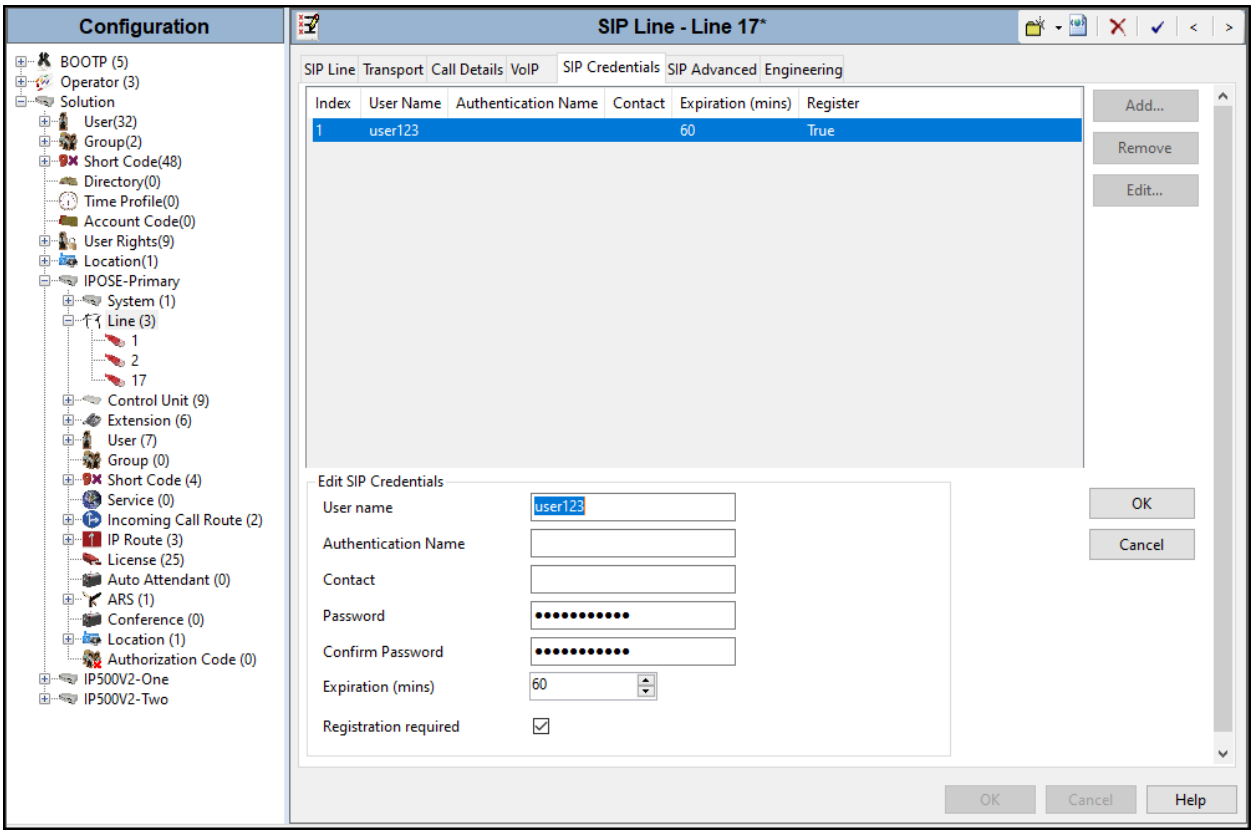
5.4.4. SIP Line – SIP Credentials Tab

Select the **SIP Credentials** tab, and then click the **Add** button to add the SIP Trunk registration credentials. Set the parameters as show below:

- For **User name**, enter the user name credential provided by WorldNet for SIP Trunk registration.
- Leave **Authentication Name** blank.
- Leave the **Contact** blank.
- For **Password** and **Confirm Password**, add the password credential provided by WorldNet for SIP Trunk registration.
- Set **Expiry (mins)** to a value acceptable to the enterprise. This setting defines how often registration with Clearcom is required following any previous registration. For the compliance test **60** minutes was used. This value should be chosen in consultation with the service provider.
- Verify that **Registration required** is checked.
- Click the OK to commit.



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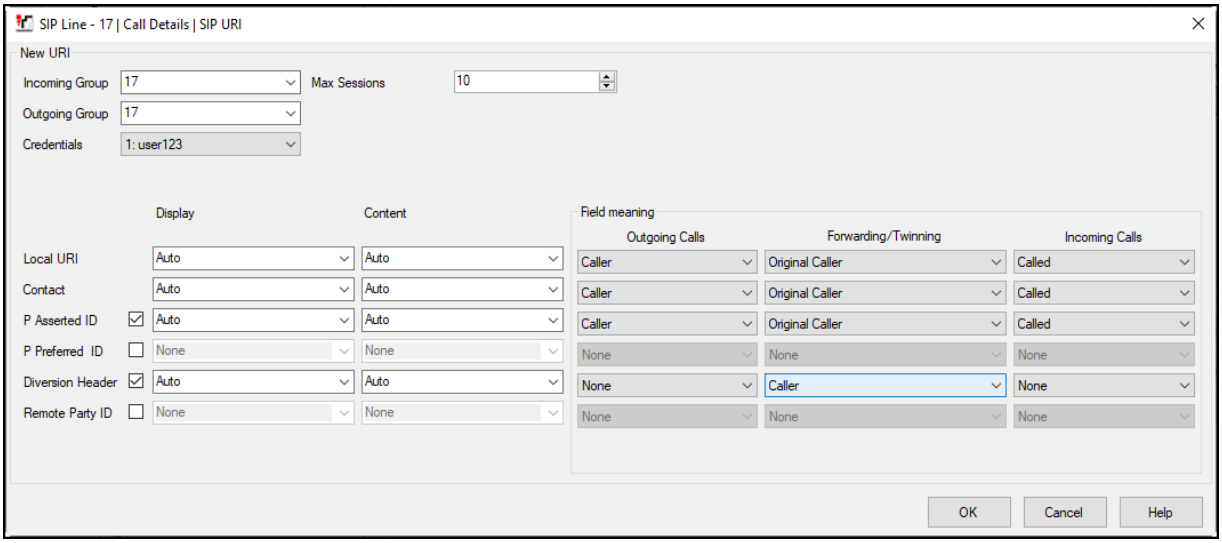
5.4.5. SIP Line – Call Details Tab

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 one new entry was added to handle incoming and outgoing calls, both.

- Associate this entry 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 from this line. For the compliance test outgoing group **17** was used. Leave the **Incoming Group** field as 0.
- Under **Credentials**, select **1: user123** from the pull-down menu (this field will default to the **User name** used under the **SIP Credentials** tab in **Section 5.4.4**).
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Check the **P Asserted ID** and **Diversion Header**.
- 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** to Commit.



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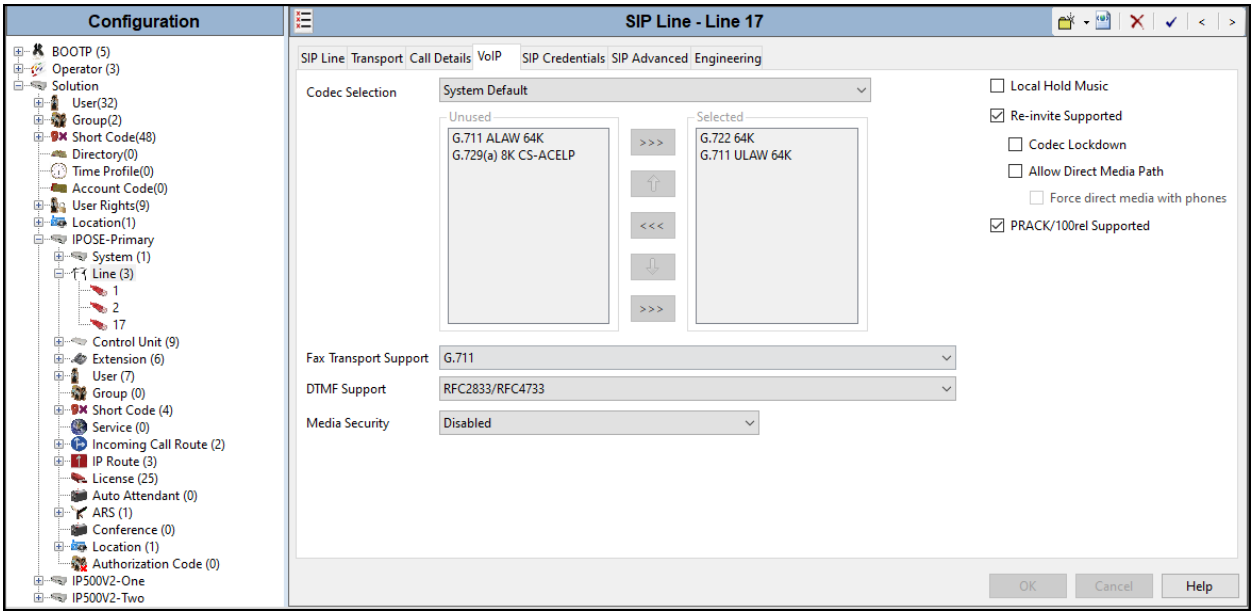
5.4.6. SIP Line - VoIP Tab

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. The buttons allow setting the specific order of preference for the codecs to be used on the SIP Line, as shown. WorldNet Telecommunications supports codecs **G.722 64K** and **G.711 ULAW 64K** for audio.
- 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).

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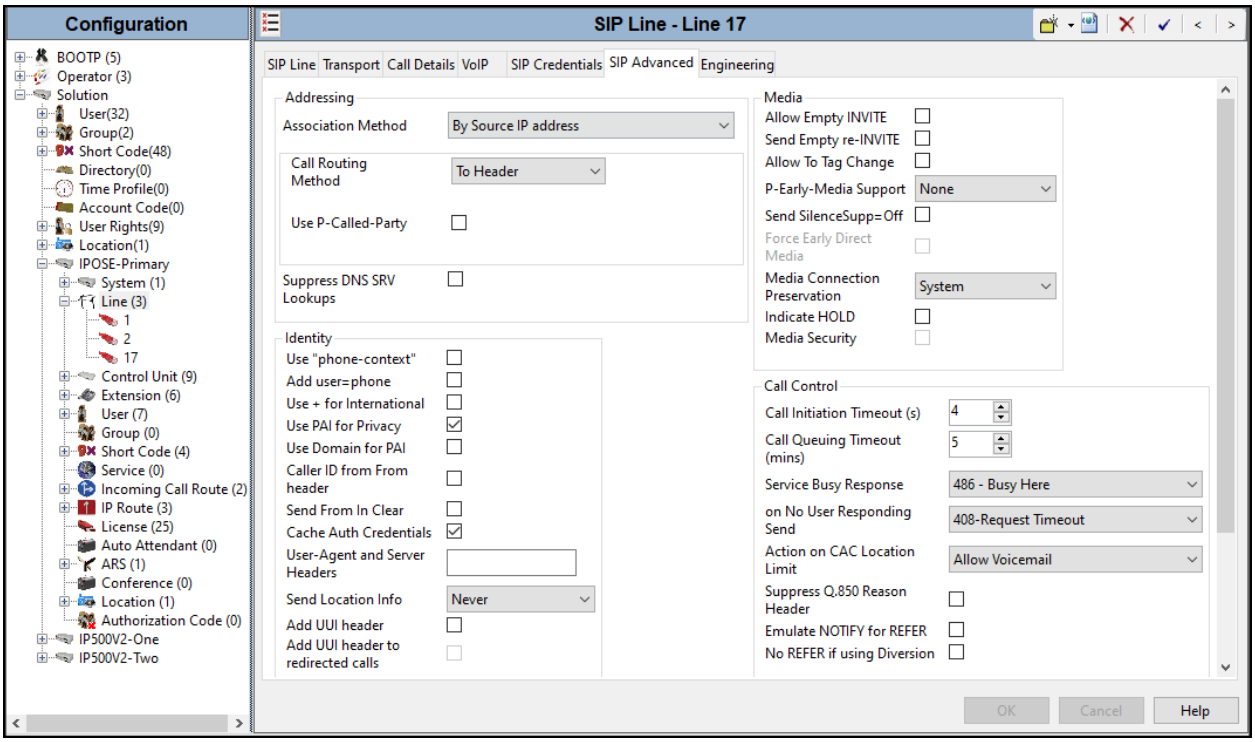
5.4.7. SIP Line – SIP Advanced Tab

In the **Addressing** area:

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

In the **Identity** area:

- Check the box for **Use PAI for Privacy**.
- Default values may be used for all other parameters.
- Click **OK** to commit.



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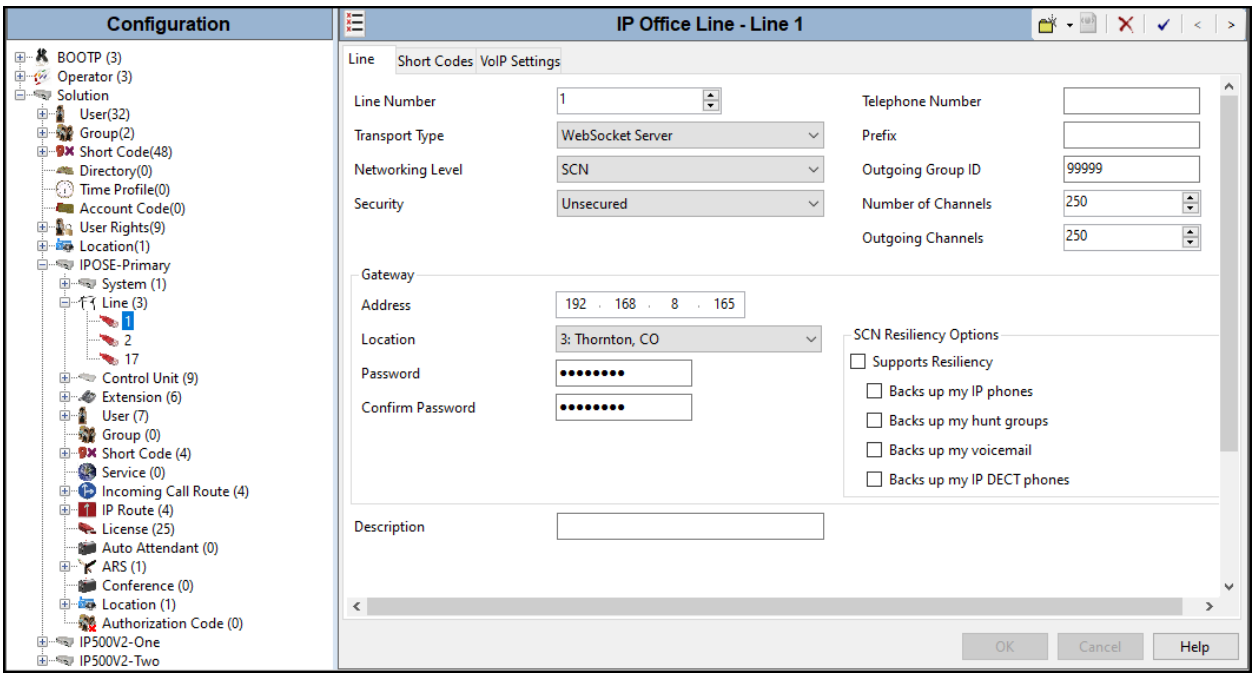
5.5. IP Office Line – Primary Server

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.



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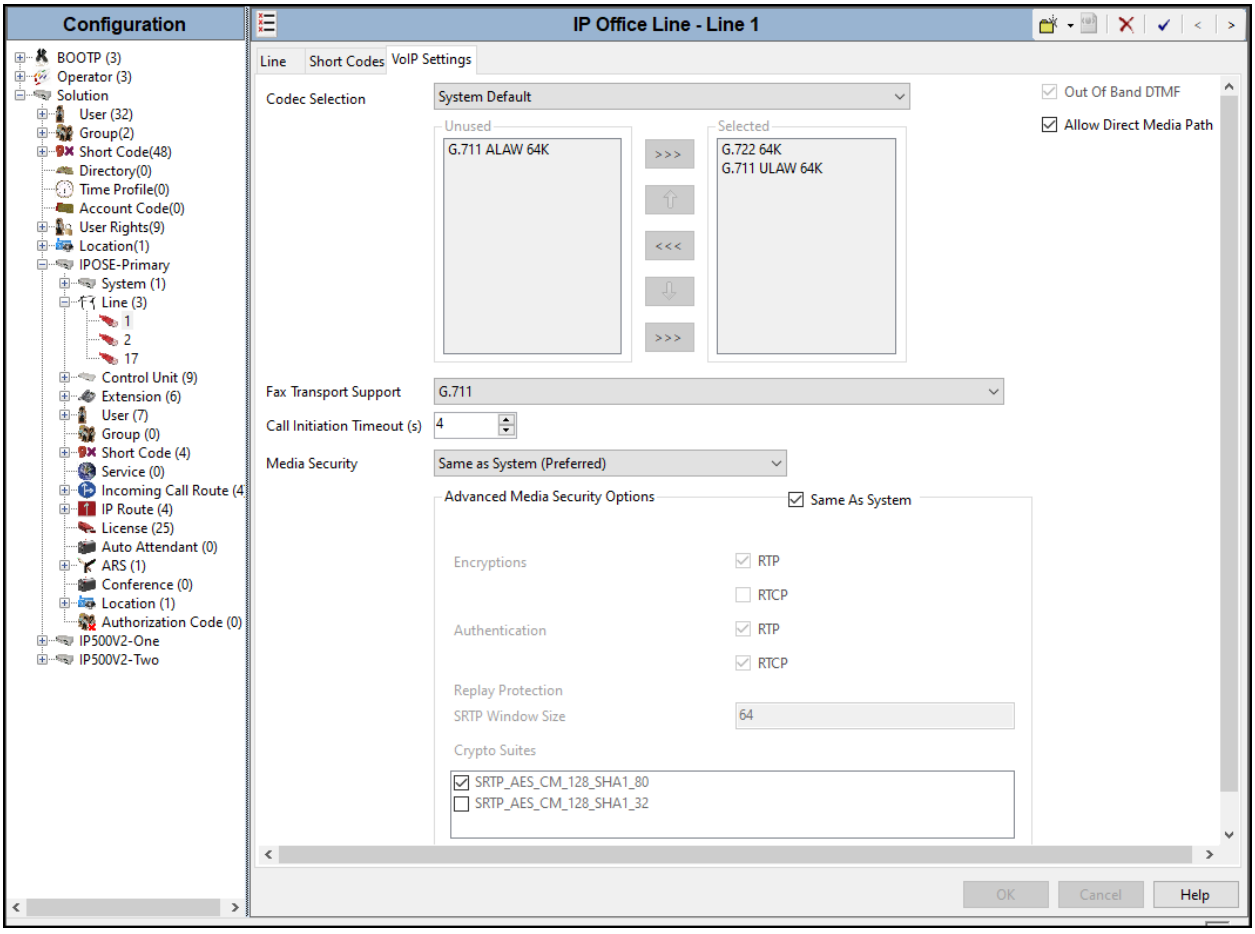
The screen below shows the IP Office Line, **VoIP Settings** tab:

- Under **Codec Selection** verify **System Default** is selected (default value).
- 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**.
- Click **OK** to commit.



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

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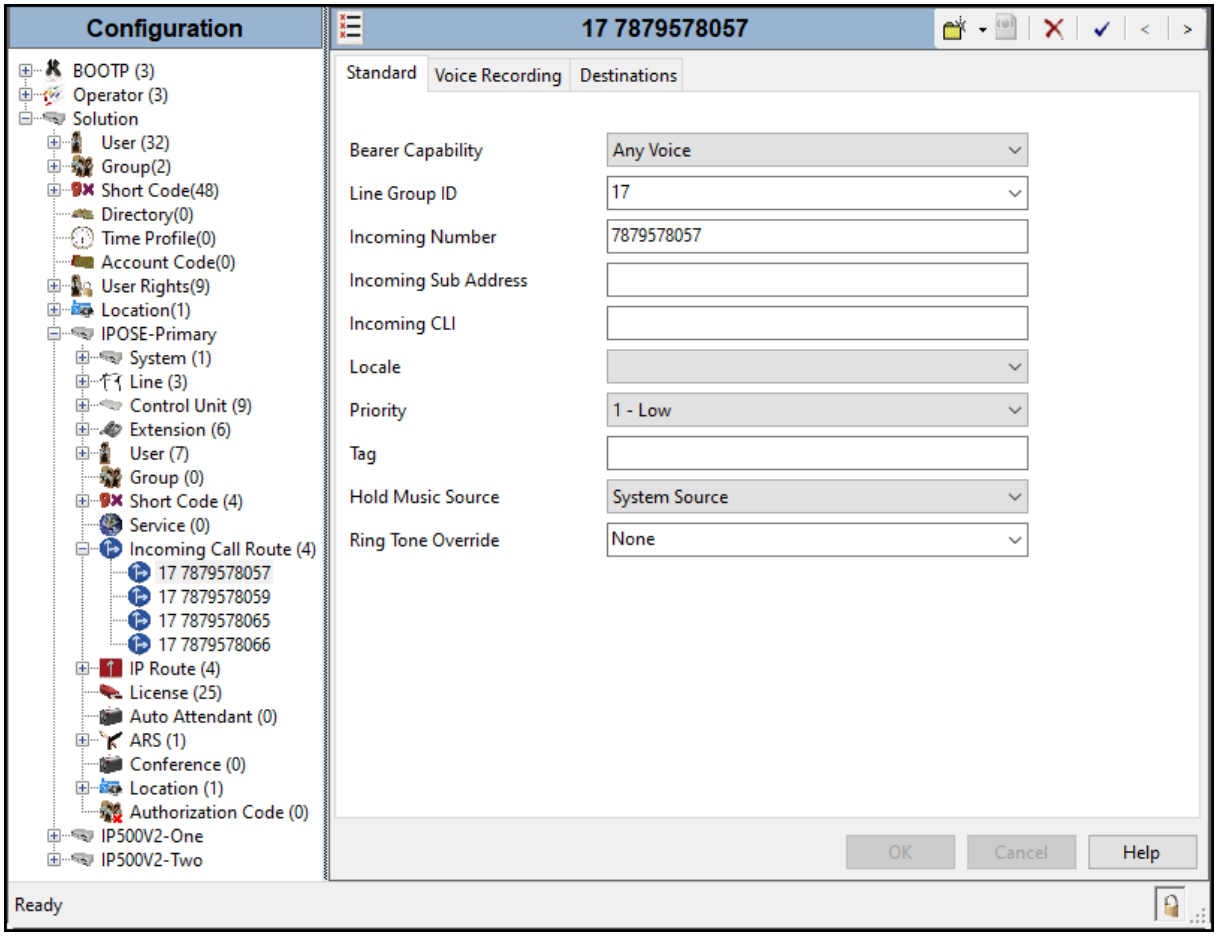
5.6. Incoming Call Route

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 5.4.5**.
- On the **Incoming Number**, enter one of the DID numbers provided by WorldNet Telecommunications.
- Default values may be used for all other parameters.
- Click **OK** to commit.



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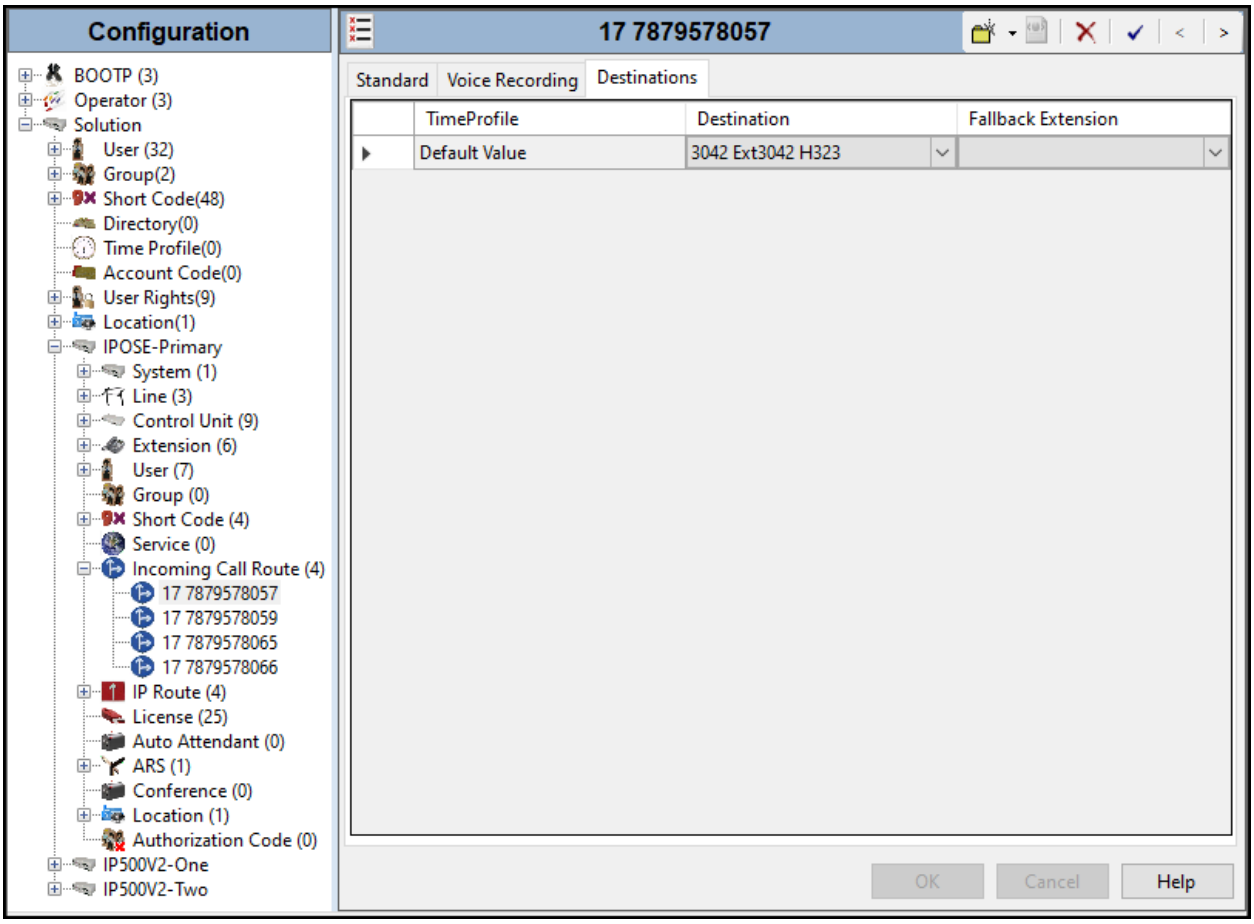
Select the **Destinations** tab. From the **Destination** drop-down menu, select the IP Office extension associated with this DID number. In the reference configuration, the DID number 7879578057 provided by WorldNet Telecommunications was associated with the Avaya IP Office extension **3042**.



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.).

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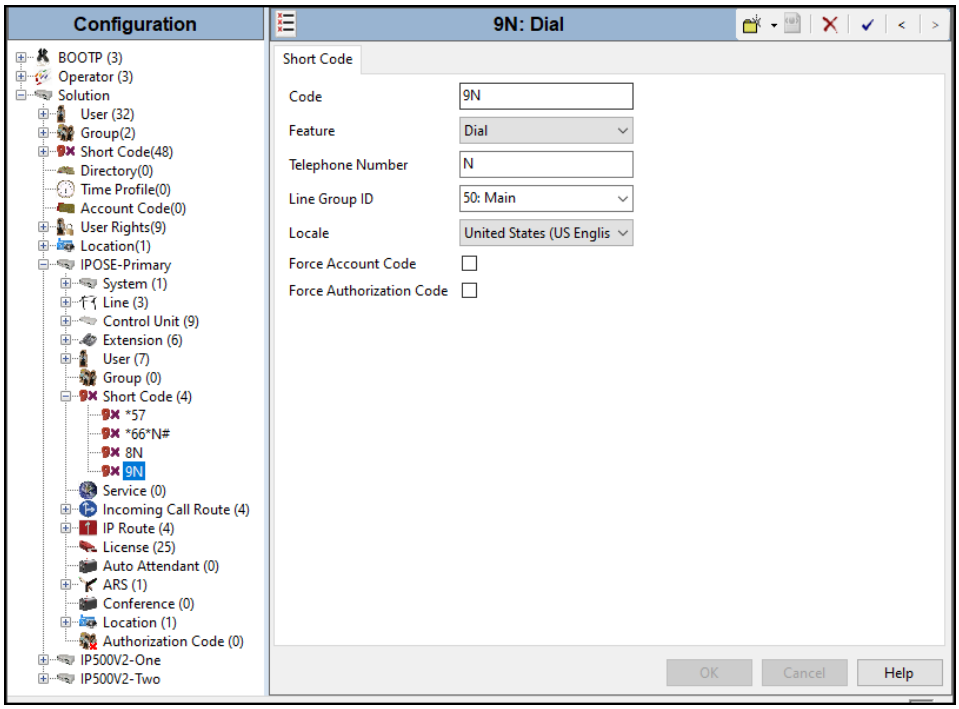
5.7. Outbound Call Routing

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.



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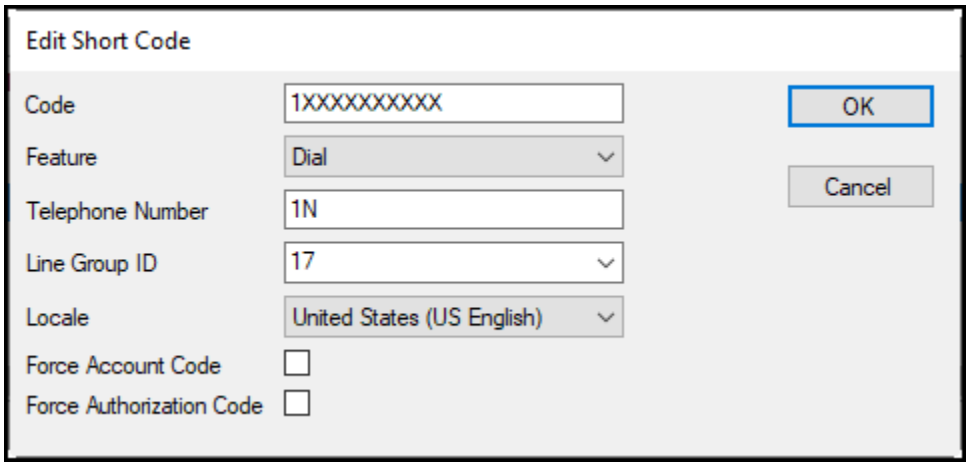
The following screen shows the example ARS configuration for the route **Main**. Note the sequence of **X**s used in the **Code** column of the entries to specify the exact number of digits to be expected, following the access code and the first set of digits on the string. This type of setting results 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 North America.



Repeat the above procedure for additional dial patterns to be used by the enterprise to dial out from IP Office, including route patterns for calls to special numbers (411, etc.).

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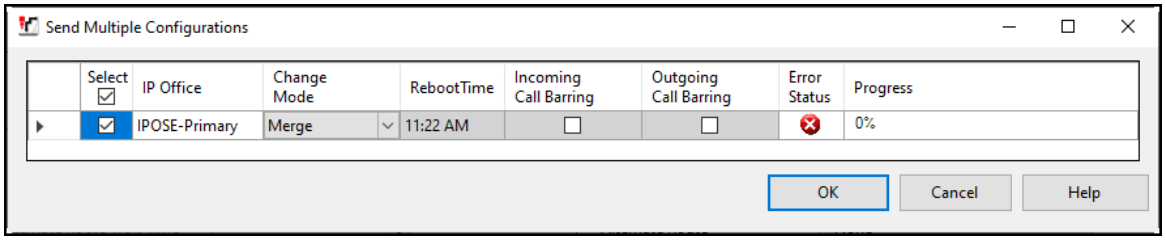
5.8. Save IP Office Primary Server Configuration

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.



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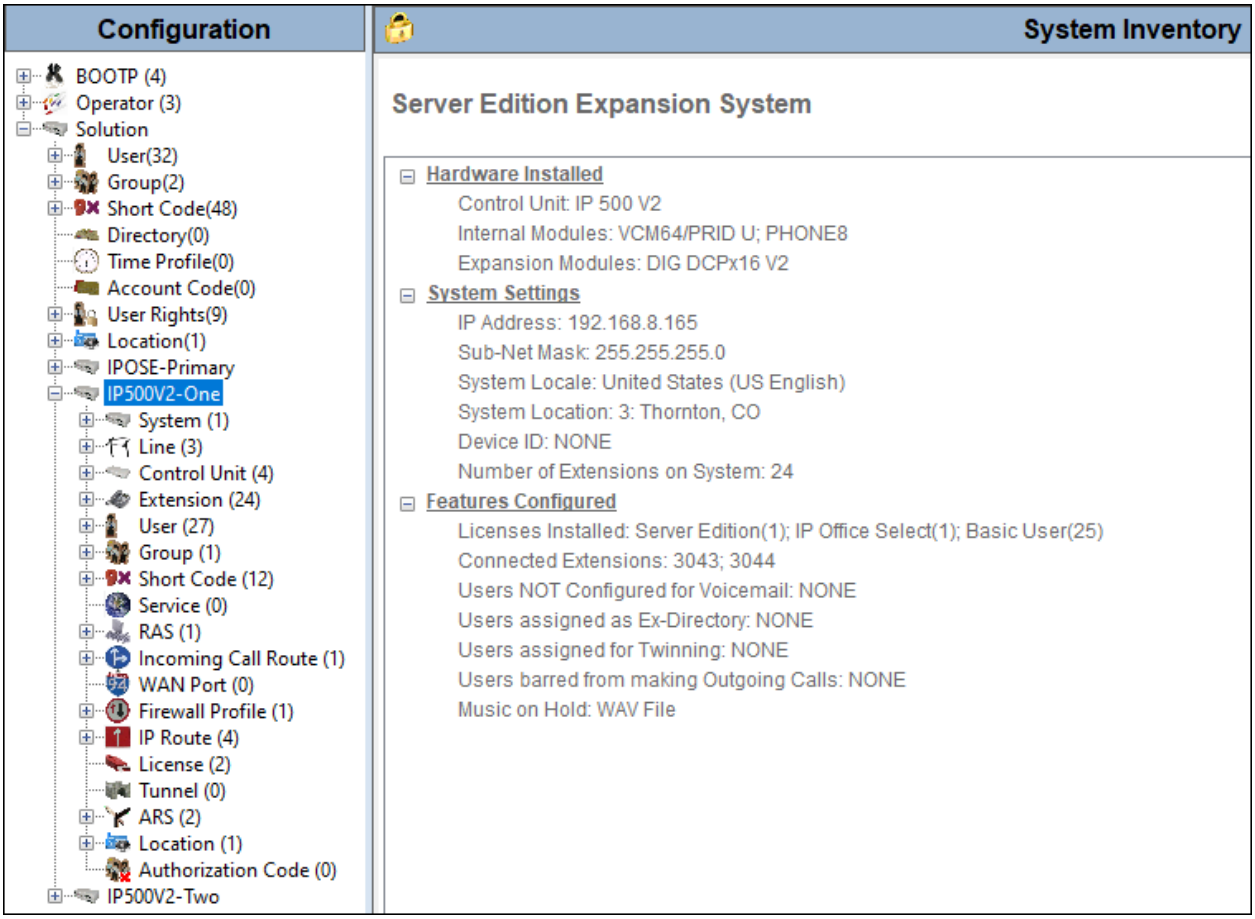
6. Avaya IP Office Expansion System Configuration

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.



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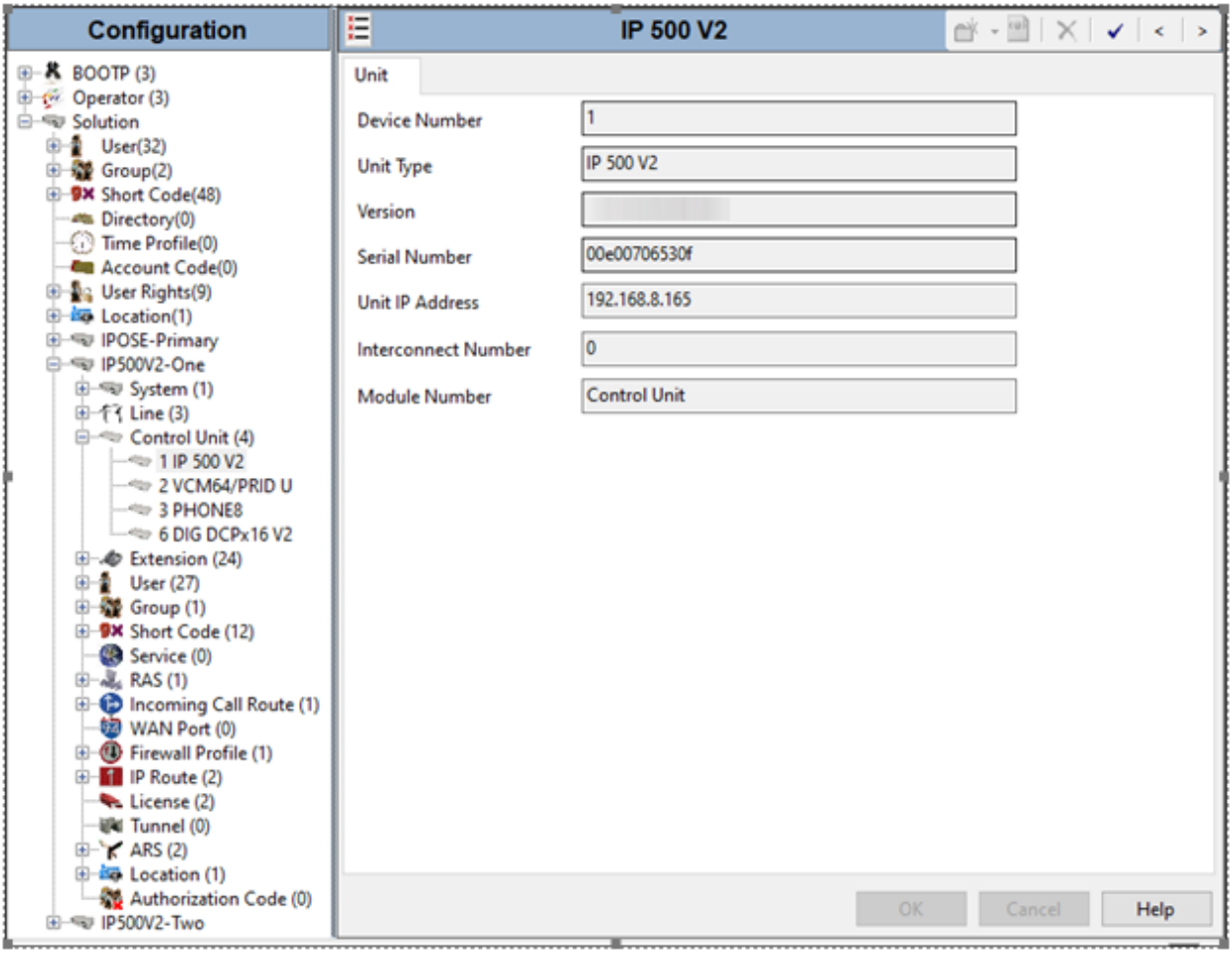
6.1. Physical Hardware

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.



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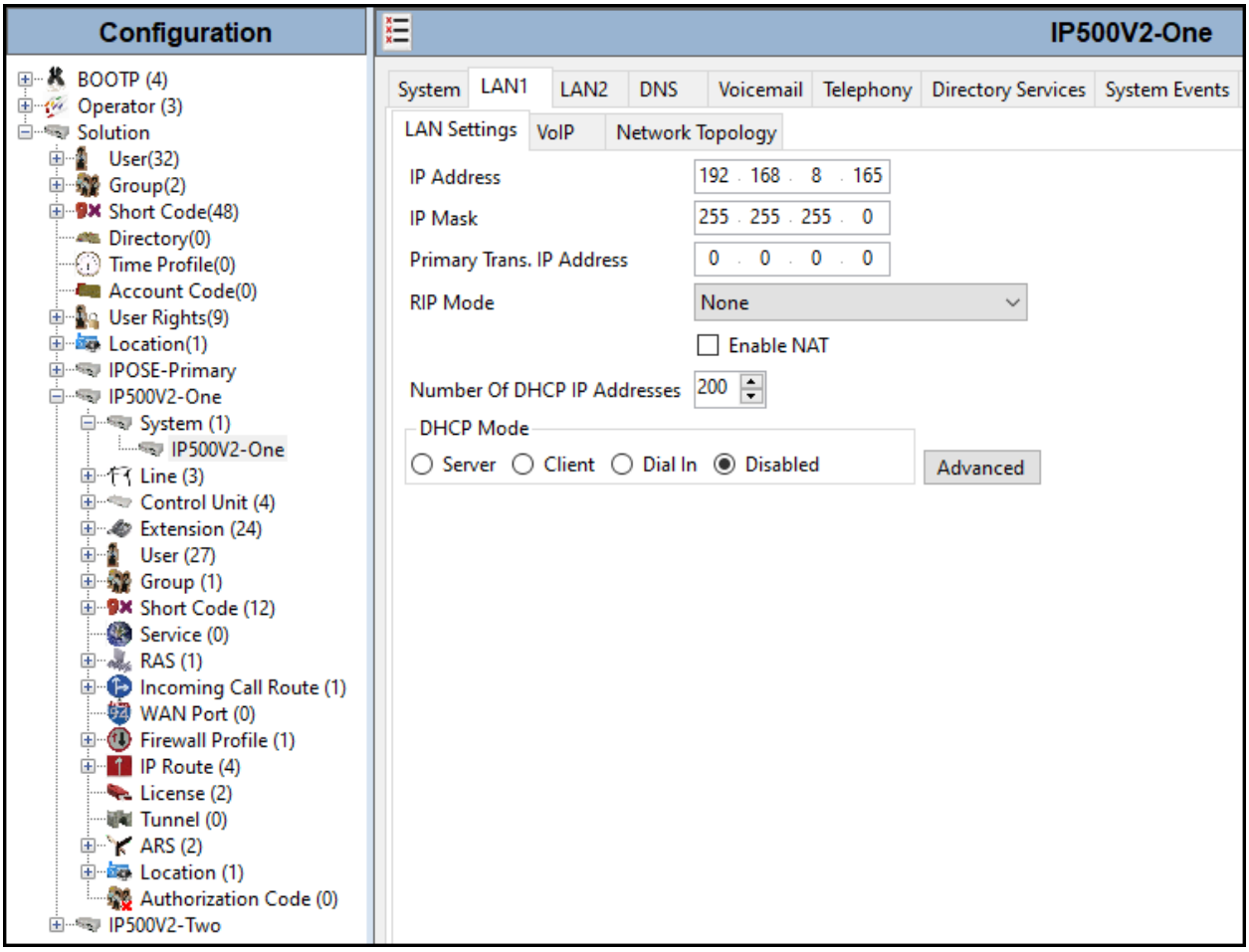
6.2. LAN Settings

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).



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

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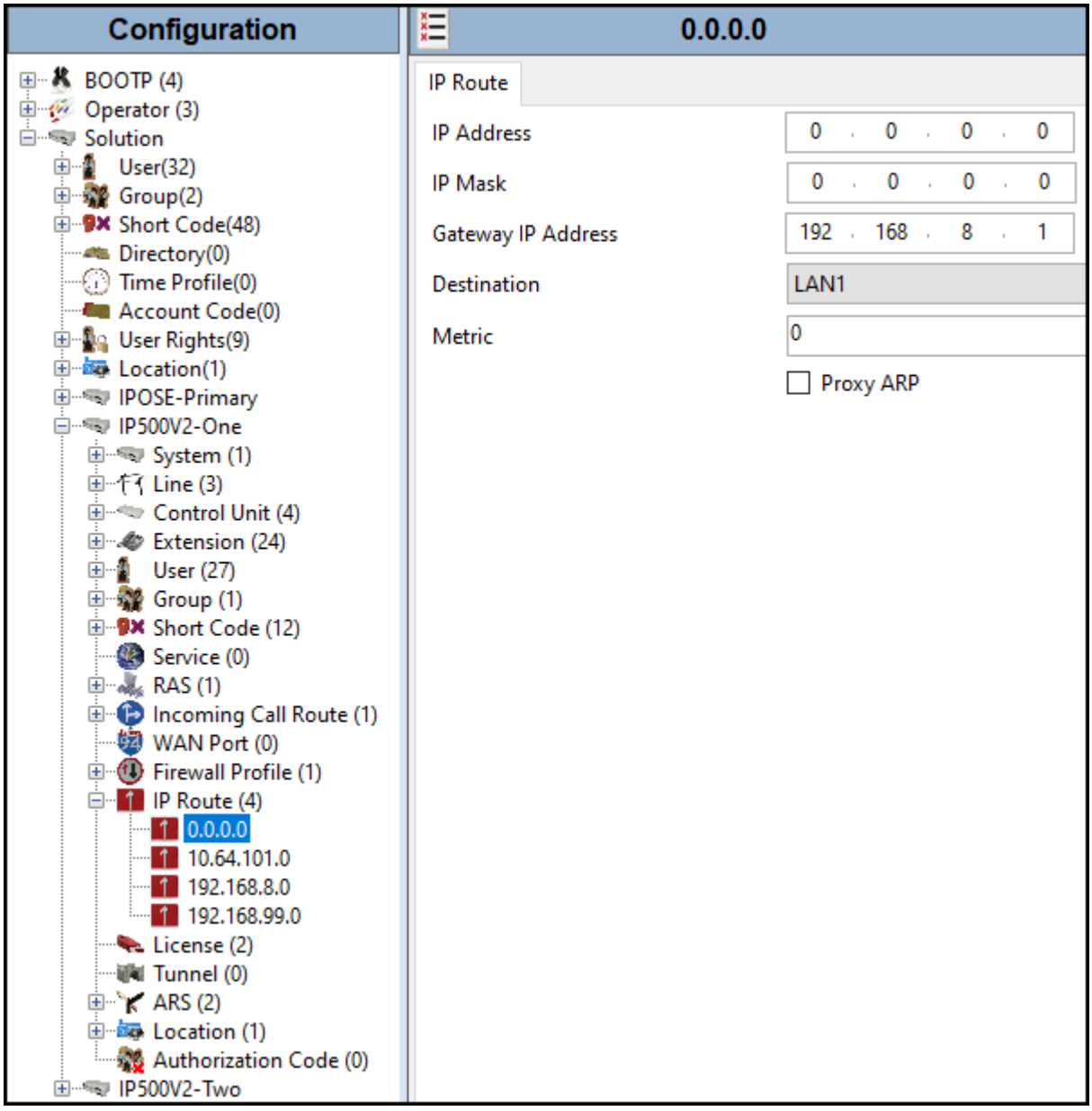
6.3. IP Route

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.



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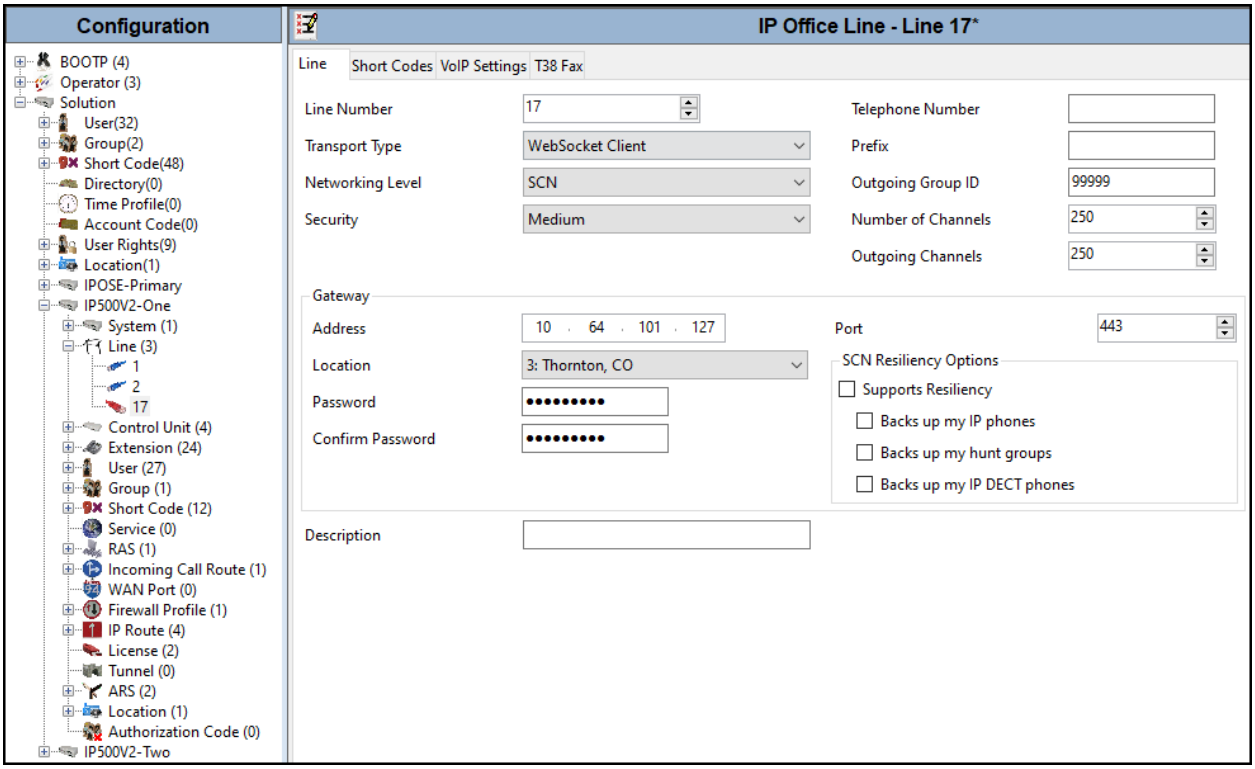
6.4. IP Office Line – IP500 V2 Expansion System

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.



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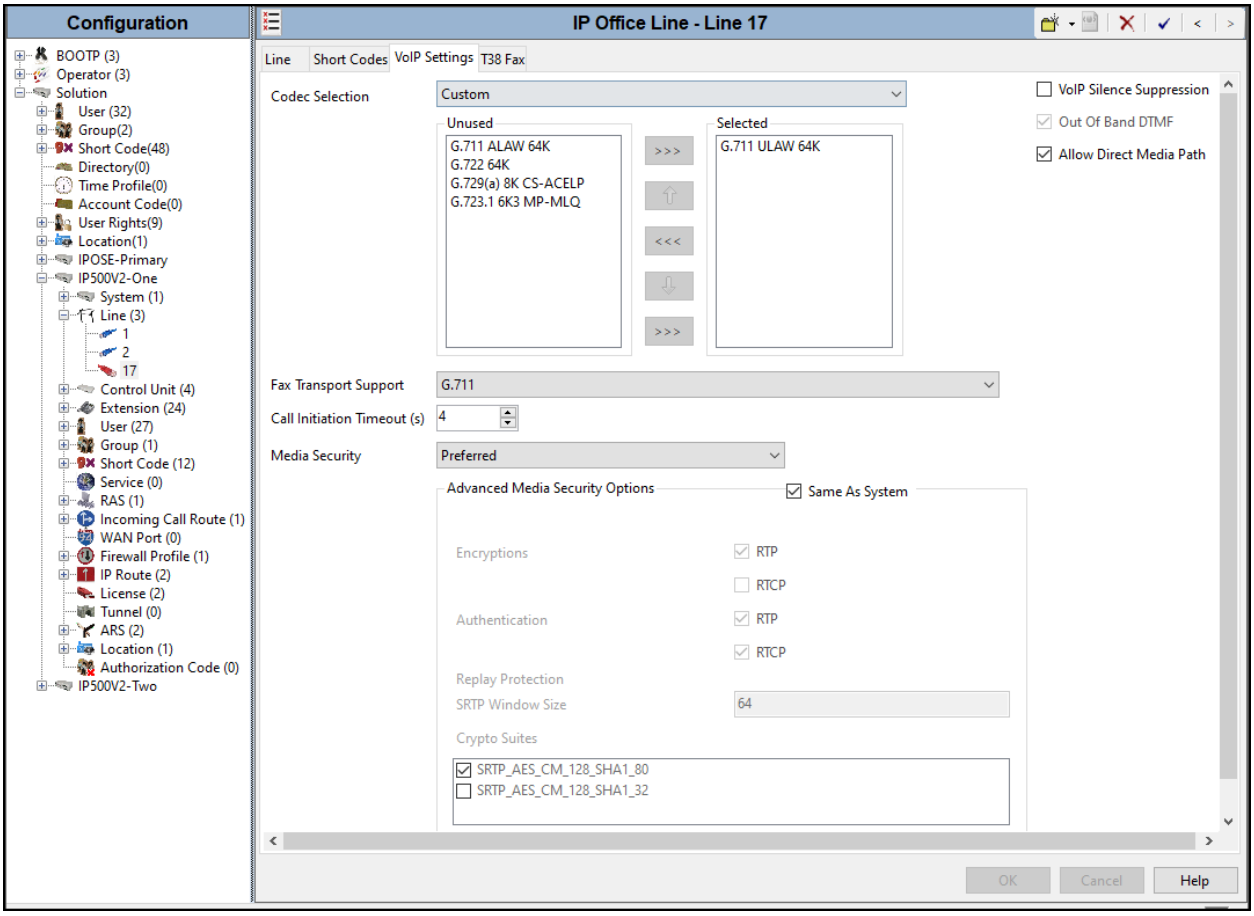
The screen below shows the IP Office Line, **VoIP Settings** tab:

- 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**.



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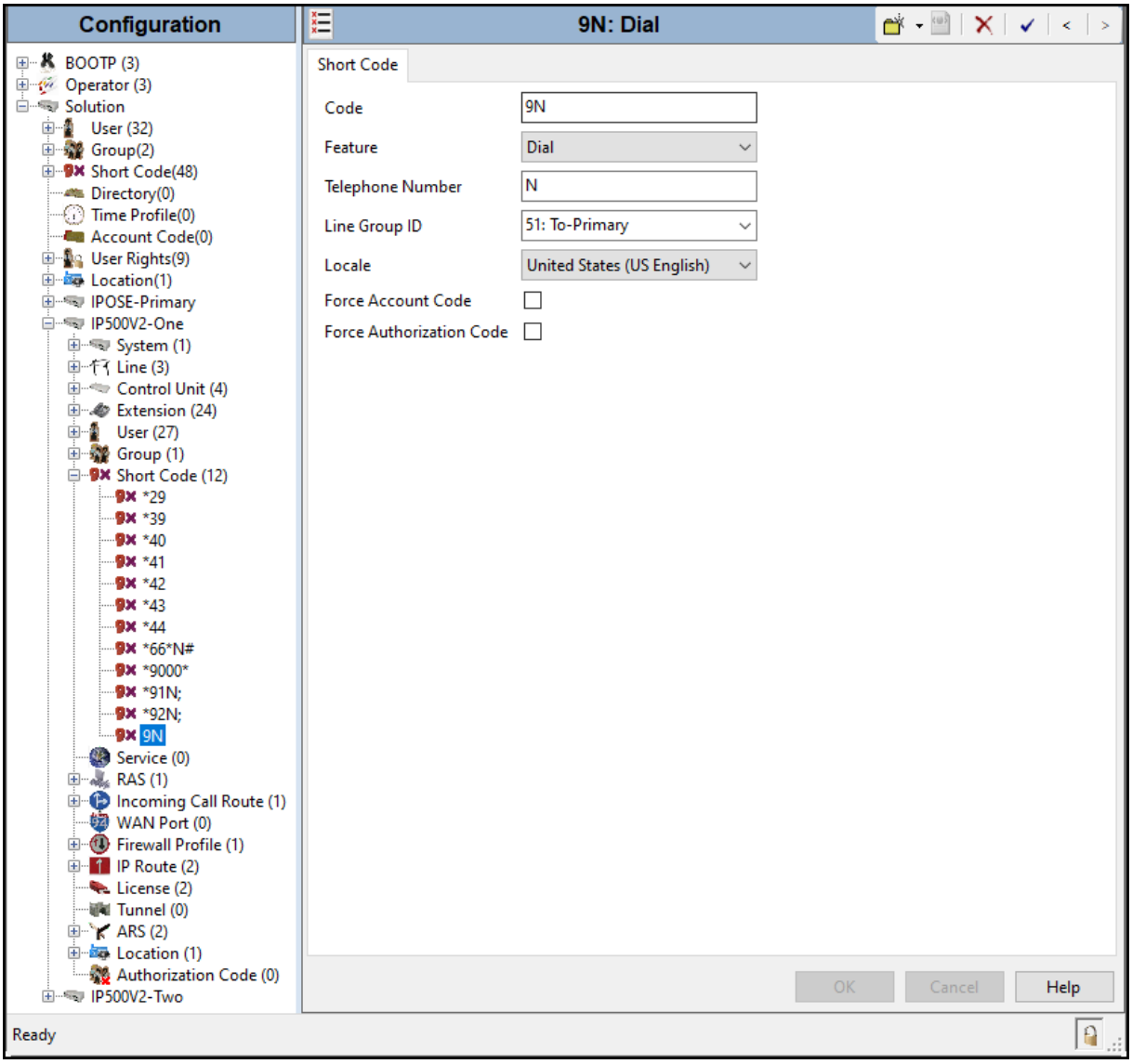
6.5. Short Codes

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.



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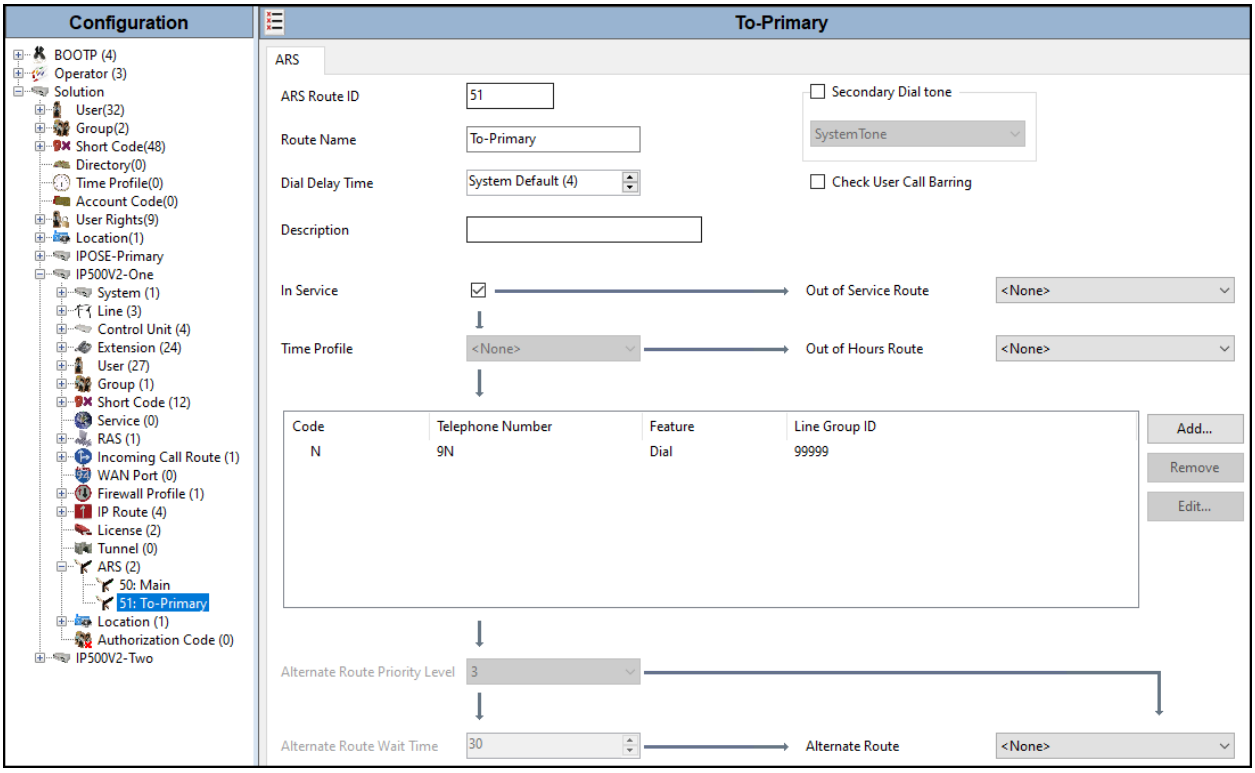
6.6. Automatic Route Selection – ARS

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**).



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

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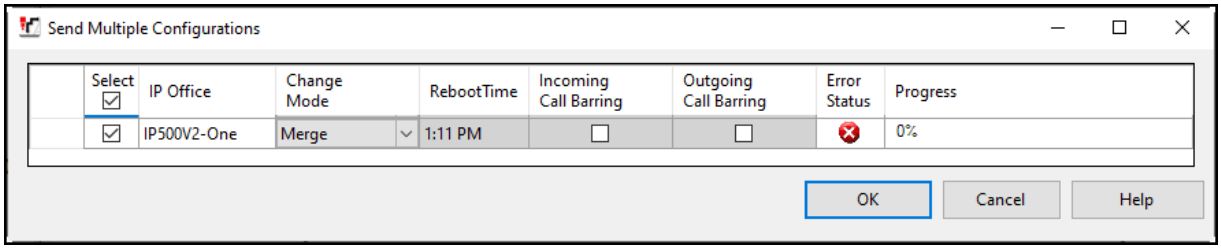
6.7. Save IP Office Expansion System Configuration

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.



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7. WorldNet Telecommunications SIP Trunking Service Configuration

To use the 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 SIP Trunking Service. The customer will need to provide the public IP address used to reach Avaya IP Office at the enterprise, the public IP address assigned to **LAN2**.

WorldNet Telecommunications will provide the customer the necessary information to configure Avaya IP Office 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.

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8. Verification Steps

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

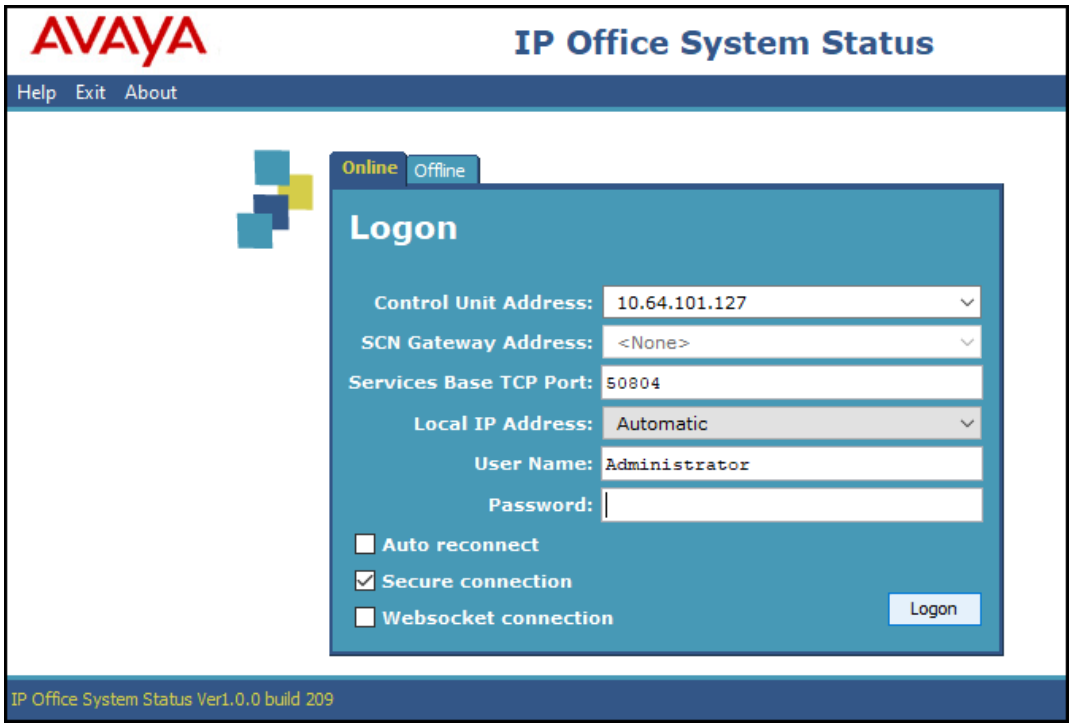
The following steps may be used to verify the configuration:

- 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.

8.1. 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.



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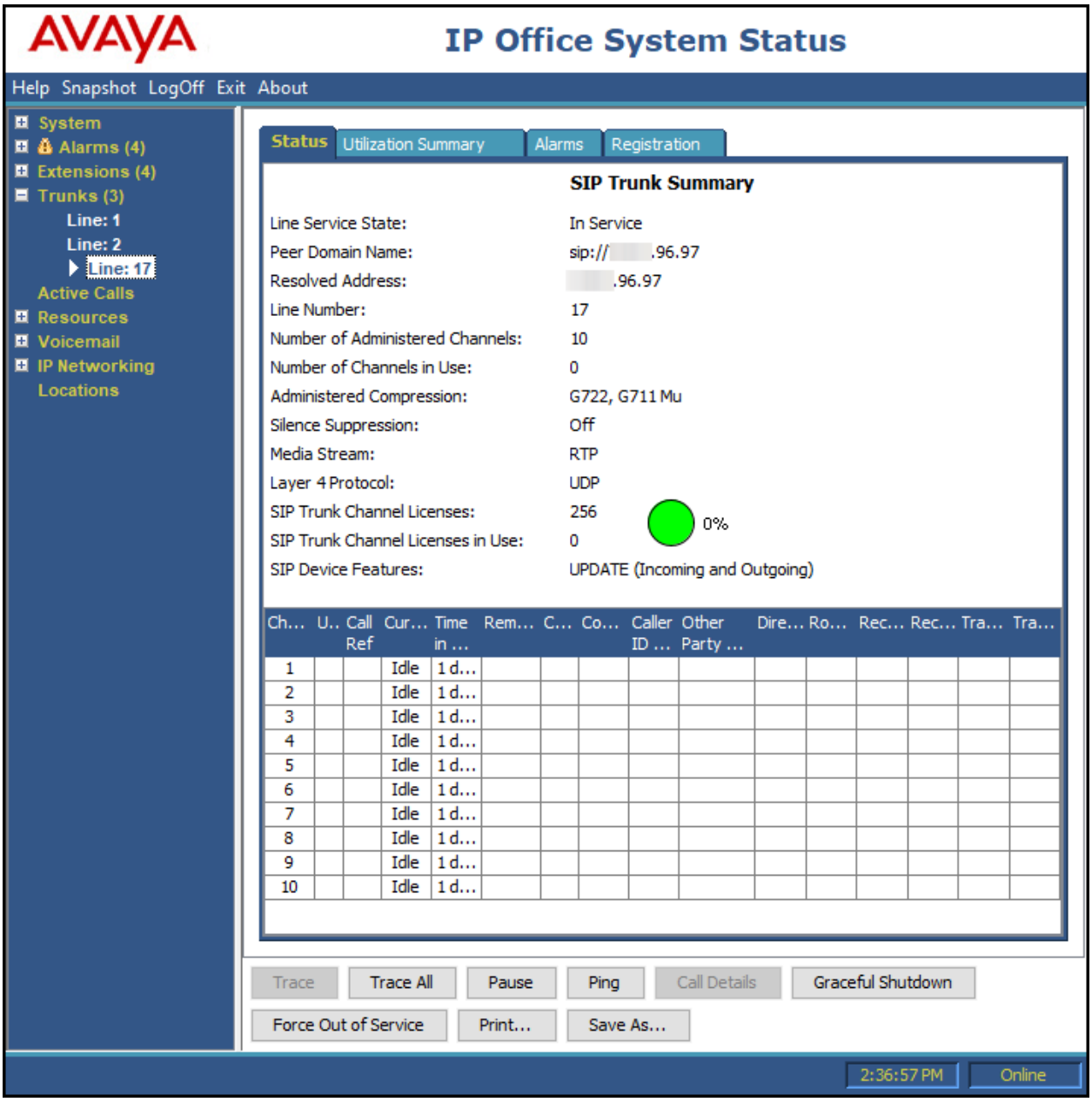
Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current Stat**e is **Idle** for each channel.



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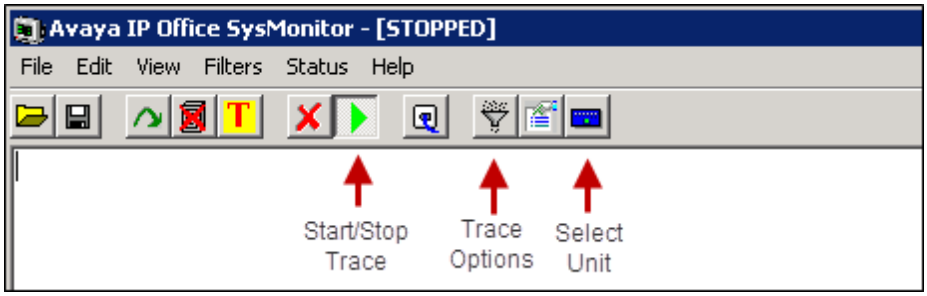
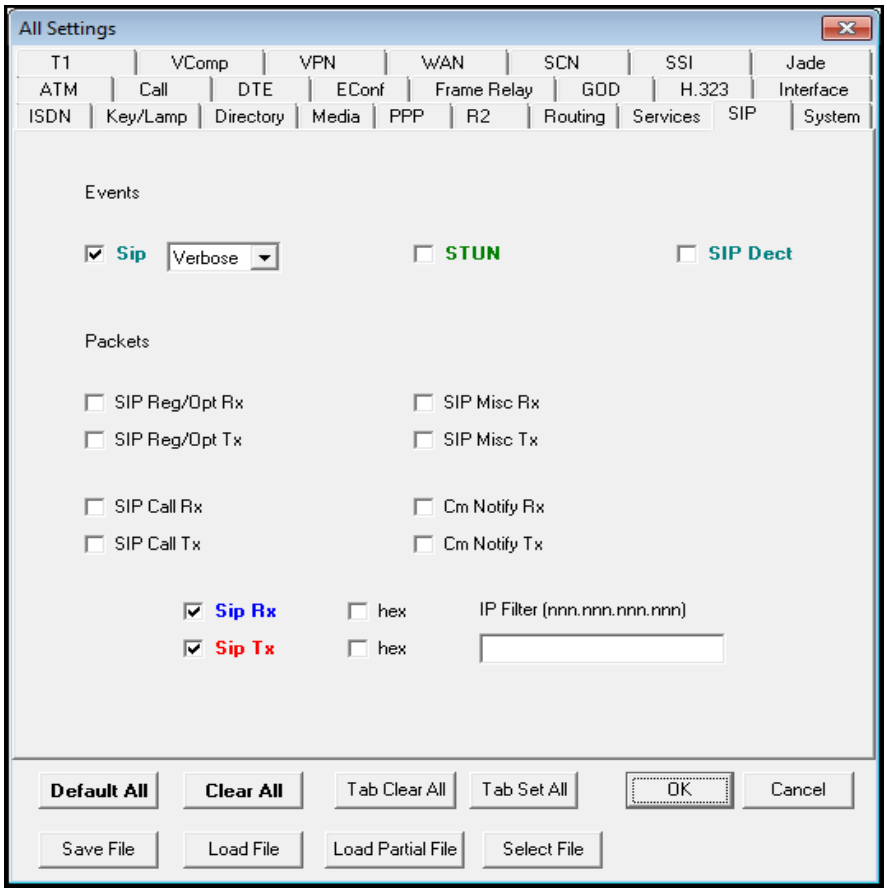
8.2. Monitor

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.



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9. Conclusion

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service on an enterprise solution consisting of Avaya IP Office 12.0 to support WorldNet Telecommunications SIP Trunking Service.

The 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**.

10. 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.*

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