DevConnect Program

Application Notes for Configuring Avaya IP Office Release 11.1 to support Lumen Technologies Voice Complete SIP Trunking Service using TLS Transport - 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 11.1 to support Lumen Technologies Voice Complete SIP Trunking Service. Transport Layer Security (TLS) and Secure Real-time Transport Protocol (SRTP) were used between the simulated enterprise site and the Lumen Technologies network (public network side) and inside of the enterprise (private network side).

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.

Table of Contents

1. Introduction

These Application Notes describe the steps necessary for configuring Session Initiation Protocol (SIP) Trunking Service between Lumen Technologies and an Avaya SIP-enabled enterprise solution. Transport Layer Security (TLS) and Secure Real-time Transport Protocol (SRTP) were used between the simulated enterprise site and the Lumen Technologies Voice Complete network (public network side) and inside of the enterprise (private network side).

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 11.1 (hereafter referred to as IP Office) and various Avaya endpoints, listed in **Section 4**.

The Lumen Technologies Voice Complete 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”, “Lumen Technologies” or “Lumen” 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 Lumen Technologies Voice Complete 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.

Transport Layer Security (TLS) and Secure Real-time Transport Protocol (SRTP) were used between the simulated enterprise site and the Lumen Technologies Voice Complete network (public network side) and 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:

- Static IP SIP Trunk 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.711MU , G.711A and G.729, Lumen Technologies preferred codec order.
- 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.
- Signalling and Media encryption negotiation between Lumen and IP Office.

Items not supported or not tested included the following:

- The SIP REFER method for call redirection is not supported by Lumen, thus SIP REFER was not tested.
- Inbound toll-free calls were not tested.
- 0, 0+10 digits, 911 Emergency and Local Directory Assistance calls were not tested.

2.2. Test Results

Interoperability testing of Lumen Technologies Voice Complete SIP Trunking Service was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **Fax support** – Fax calls using the T.38 protocol failed during the compliance test. G.711 pass-through fax was also tested, G.711 pass-through fax was successful tested in both directions (inbound and outbound). The issue related to T.38 fax calls failing is related to the codec negotiation method used by IP Office, IP Office uses multiple m=lines in the SDP during codec re-negotiation, which is not supported by Lumen Technologies.
- **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 used in IP Office, set **Media Security** for the specific extension being used with Avaya Workplace Client for Windows softphone to "**Disabled**". This is done under **Extension → VoIP** tab in IP Office Manager.
- **SIP OPTIONS Messages** – During the compliance test Lumen Technologies did not send SIP OPTIONS messages to IP Office, IP Office did send SIP OPTIONS messages to Lumen Technologies, Lumen responded to the SIP OPTIONS messages sent by IP Office. This was sufficient to keep the SIP trunk up in-service.

2.3. Support

For support on Lumen Technologies Voice Complete SIP Trunking Service visit the corporate Web page at: https://www.lumen.com/en-us/collaboration/voice-complete.html

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 Lumen Technologies Voice Complete 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 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 Avaya IP Office and Lumen Technologies, across the public Internet, is SIP over TLS. SRTP was used to secure/encrypt the media (voice).

For inbound calls, the calls flowed from the Lumen Technologies 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 Lumen Technologies network.

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

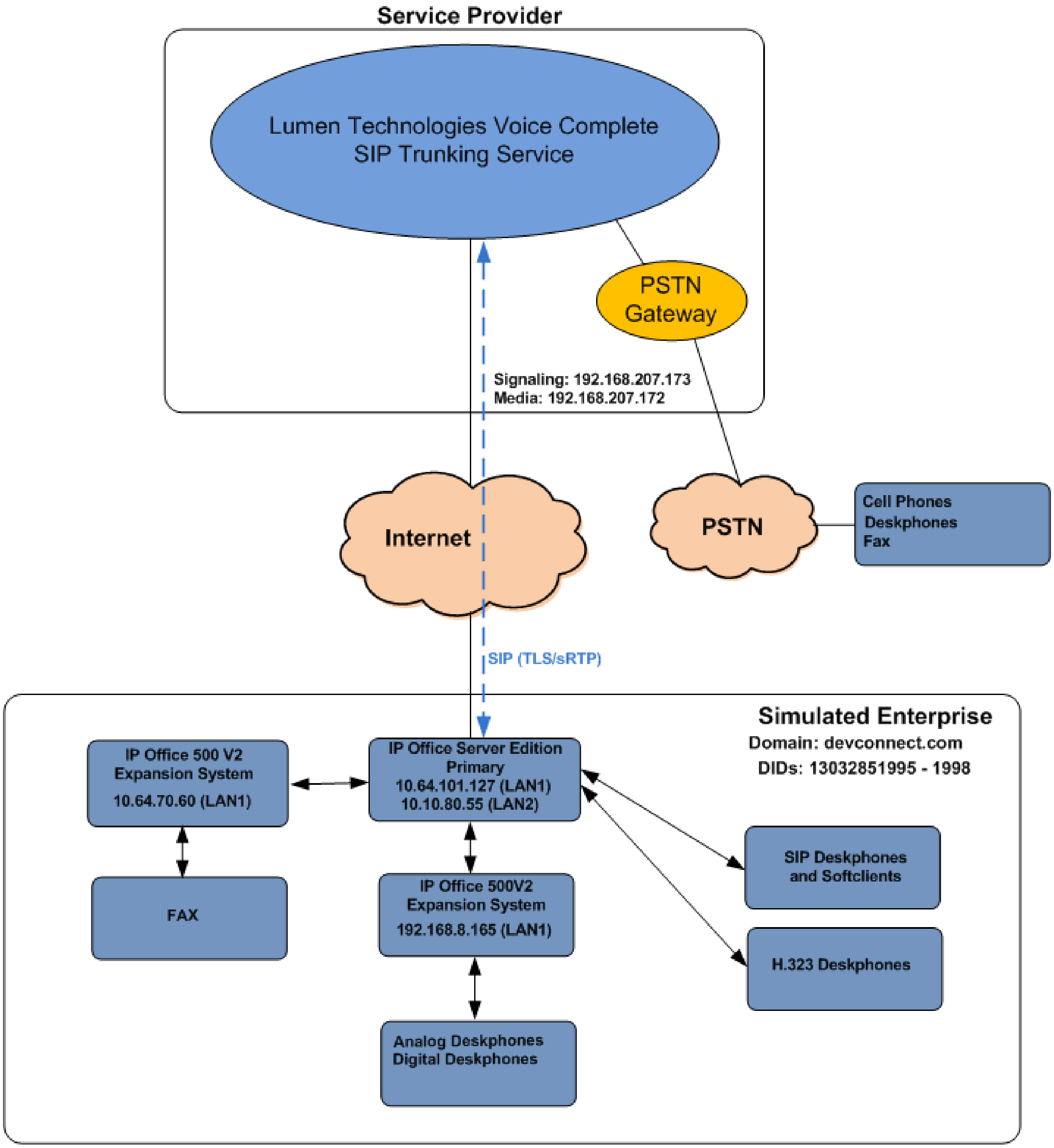


Figure 1: Avaya Interoperability Test Lab Configuration

4. Equipment and Software Validated

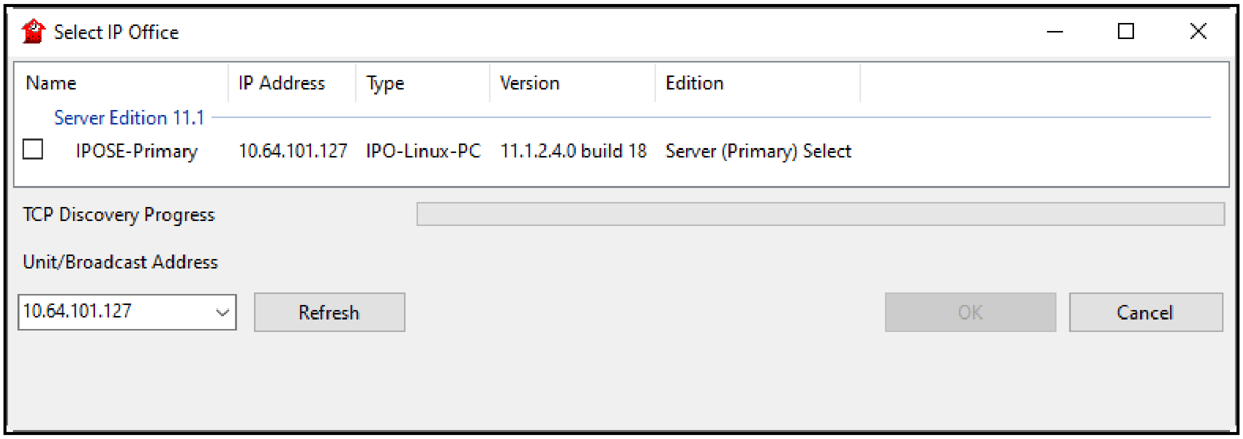
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office Server Edition (Primary Server) • Avaya IP Office Voicemail Pro	11.1.2.4.0 Build 18 11.1.2.4.0 Build 2
Avaya IP Office IP500 V2 (Expansion Systems)	11.1.2.4.0 Build 18
Avaya IP Office Manager	11.1.2.4.0 Build 18
Avaya J179 IP Telephone (H.323)	Version 6.8.5.4.10
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).	3.34.1.10
Analog Telephone	---
Lumen Technologies	
Cisco Broadworks	23.0_1.1075
Ribbon SBC7000	7.2.4

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.



Select IP Office

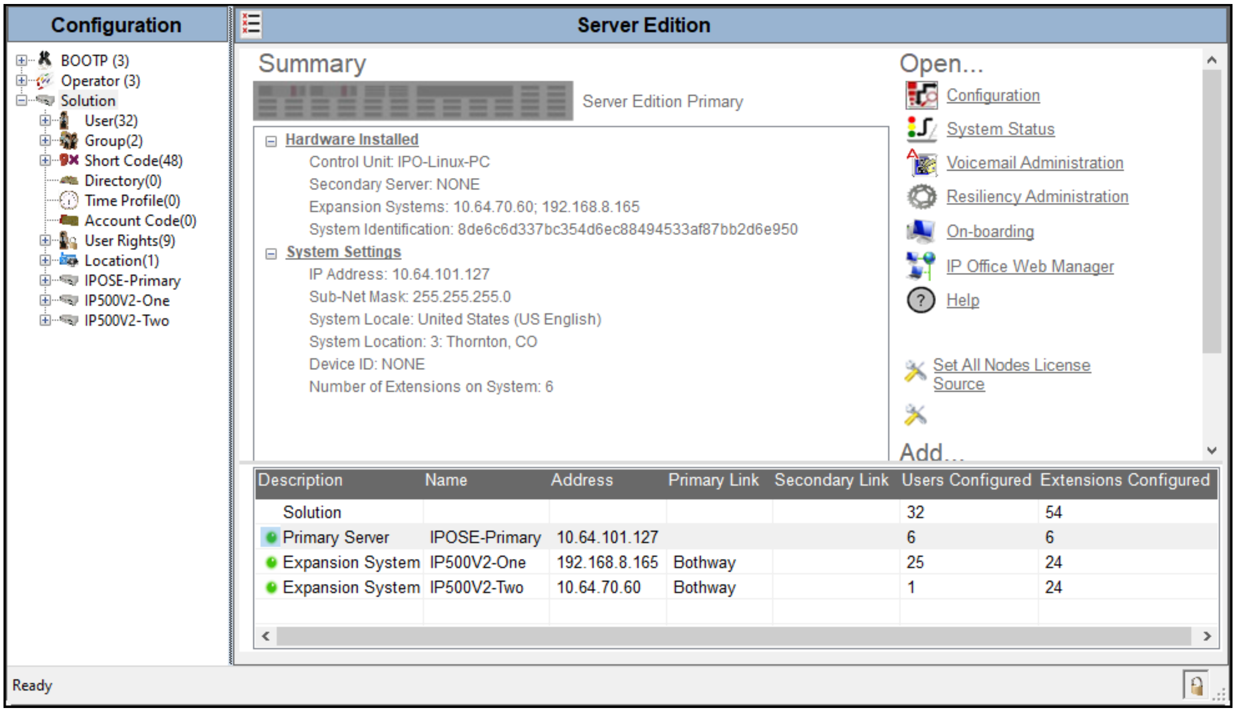
Name	IP Address	Type	Version	Edition
Server Edition 11.1				
□ IPOSE-Primary	10.64.101.127	IPO-Linux-PC	11.1.2.4.0 build 18	Server (Primary) Select

TCP Discovery Progress

Unit/Broadcast Address

10.64.101.127

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.



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: 8de6c6d337bc354d6ec88494533af87bb2d6e950

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

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.

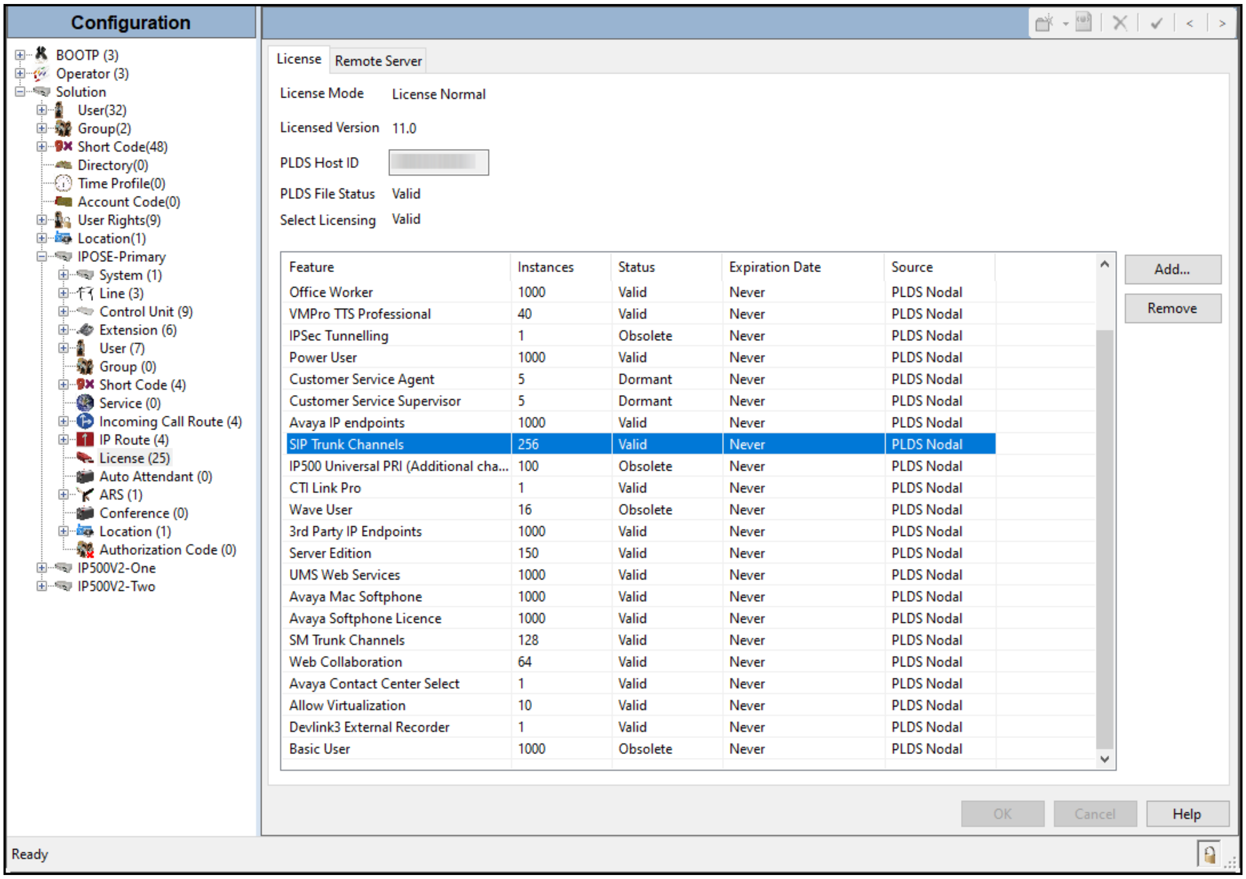
Standard feature configurations that are not directly related to the interfacing with the service provider are assumed to be already in place, and they are not part of these Application Notes.

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.



License Mode: License Normal

Licensed Version: 11.0

PLDS Host ID: [redacted]

PLDS File Status: Valid

Select Licensing: Valid

Feature	Instances	Status	Expiration Date	Source
Office Worker	1000	Valid	Never	PLDS Nodal
VMPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal
Power User	1000	Valid	Never	PLDS Nodal
Customer Service Agent	5	Dormant	Never	PLDS Nodal
Customer Service Supervisor	5	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
Avaya Softphone Licence	1000	Valid	Never	PLDS Nodal
SM Trunk Channels	128	Valid	Never	PLDS Nodal
Web Collaboration	64	Valid	Never	PLDS Nodal
Avaya Contact Center Select	1	Valid	Never	PLDS Nodal
Allow Virtualization	10	Valid	Never	PLDS Nodal
Devlink3 External Recorder	1	Valid	Never	PLDS Nodal
Basic User	1000	Obsolete	Never	PLDS Nodal

5.2. System Settings

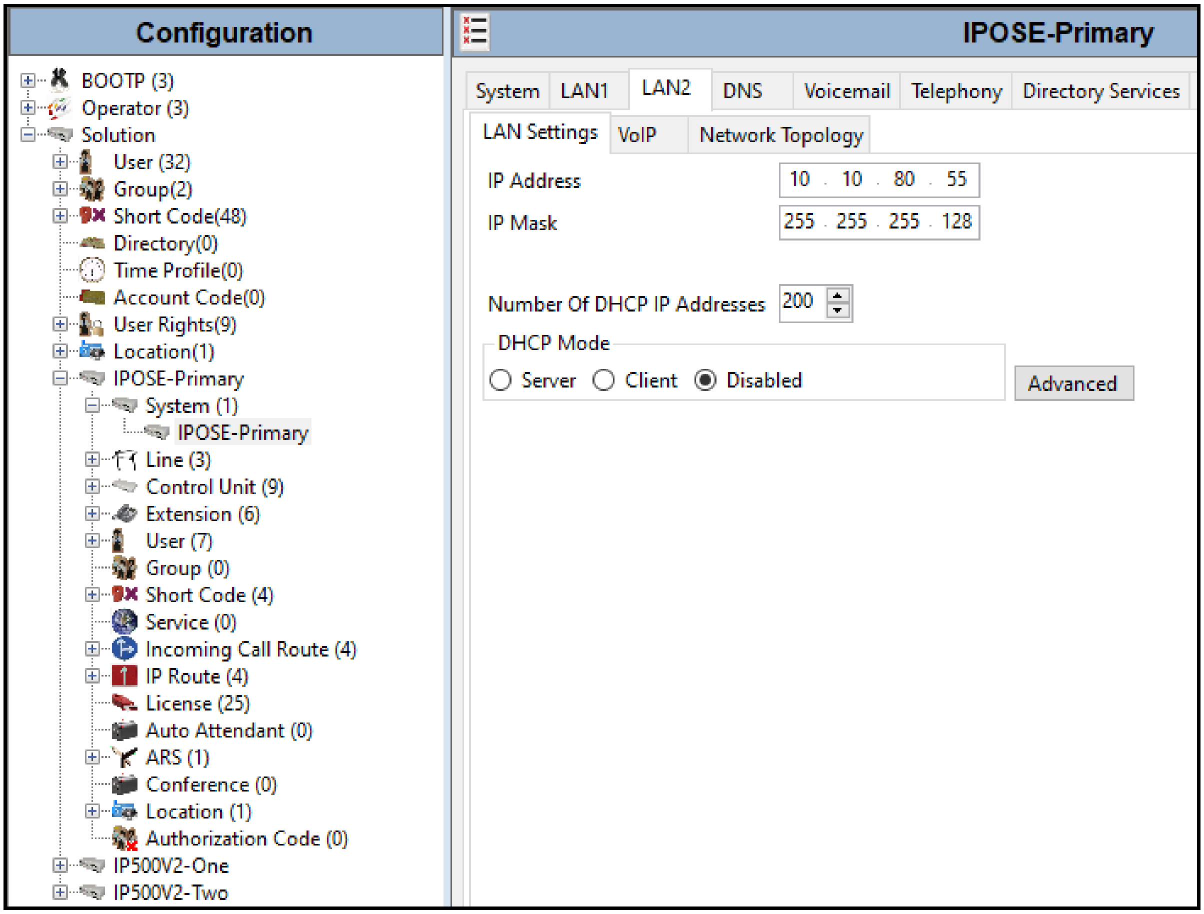
Configure the necessary system settings. In an Avaya IP Office, the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side).

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 Lumen.

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).



IP Address: 10 . 10 . 80 . 55

IP Mask: 255 . 255 . 255 . 128

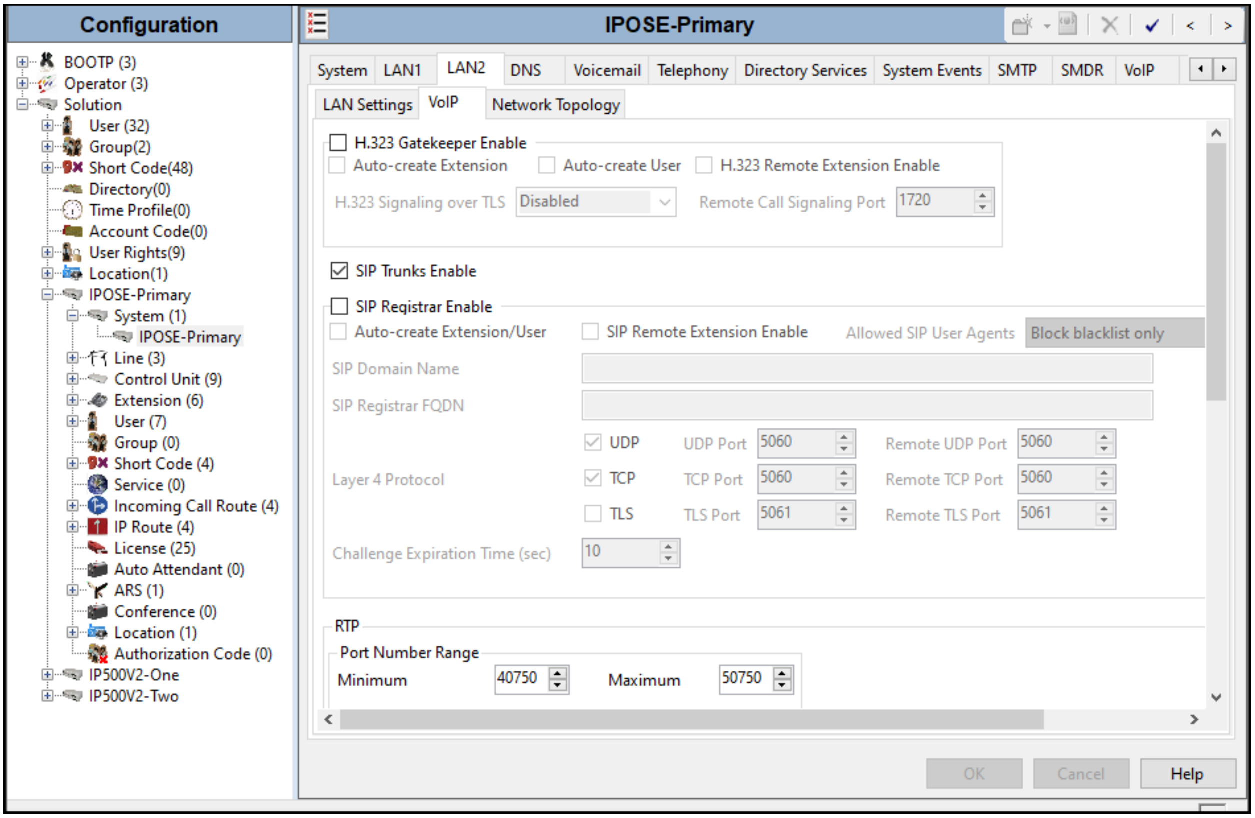
Number Of DHCP IP Addresses: 200

DHCP Mode: ○ Server ○ Client ● Disabled

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.



□ H.323 Gatekeeper Enable

□ Auto-create Extension □ Auto-create User □ H.323 Remote Extension Enable

H.323 Signaling over TLS: Disabled Remote Call Signaling Port: 1720

☑ SIP Trunks Enable

□ SIP Registrar Enable

□ Auto-create Extension/User □ SIP Remote Extension Enable Allowed SIP User Agents: Block blacklist only

SIP Domain Name:

SIP Registrar FQDN:

Layer 4 Protocol:

☑ UDP	UDP Port	5060	Remote UDP Port	5060
☑ TCP	TCP Port	5060	Remote TCP Port	5060
□ TLS	TLS Port	5061	Remote TLS Port	5061

Challenge Expiration Time (sec): 10

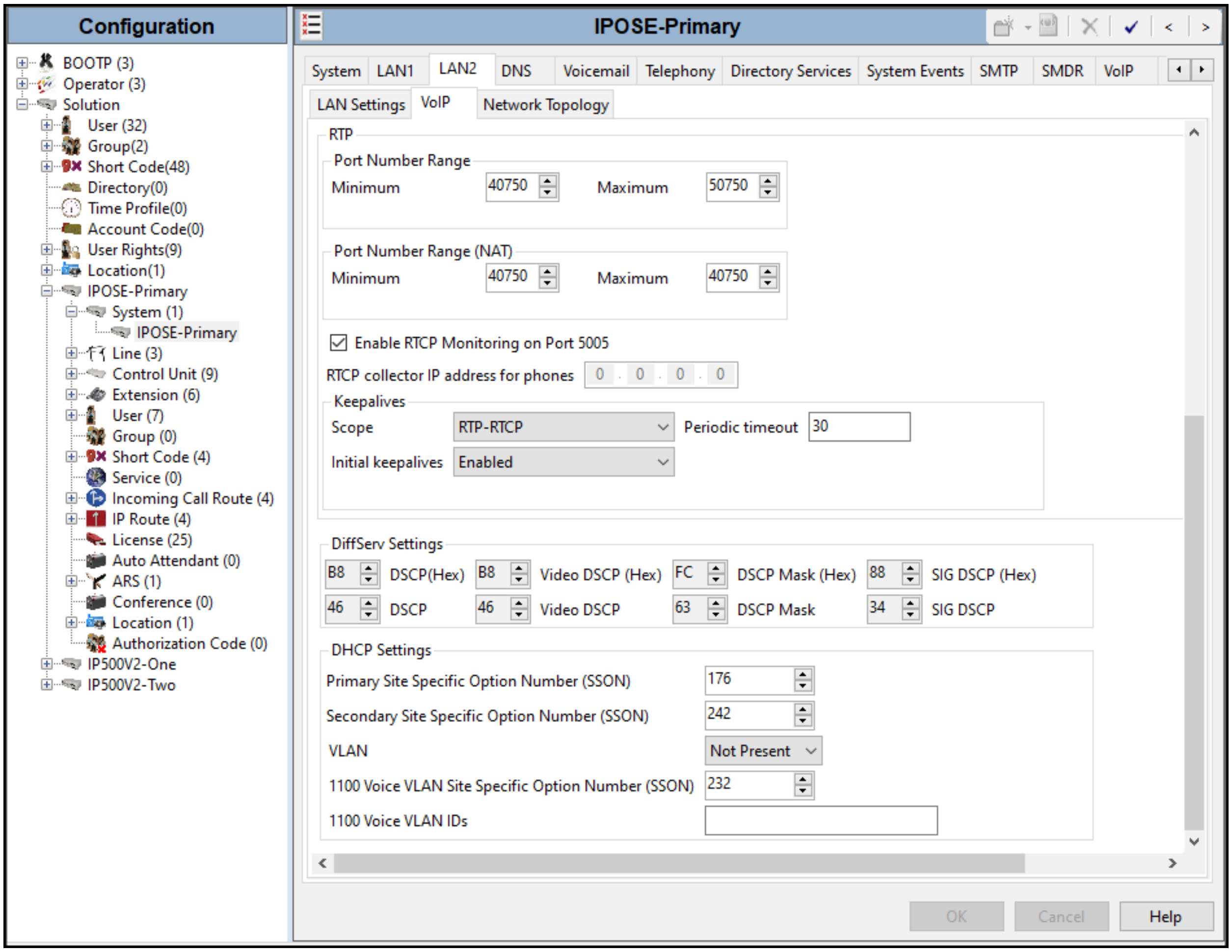
RTP

Port Number Range

Minimum: 40750 Maximum: 50750

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.



RTP

Port Number Range
Minimum 40750 Maximum 50750

Port Number Range (NAT)
Minimum 40750 Maximum 40750

☑ Enable RTCP Monitoring on Port 5005

RTCP collector IP address for phones 0 . 0 . 0 . 0

Keepalives
Scope RTP-RTCP Periodic timeout 30
Initial keepalives Enabled

DiffServ Settings

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

DHCP Settings

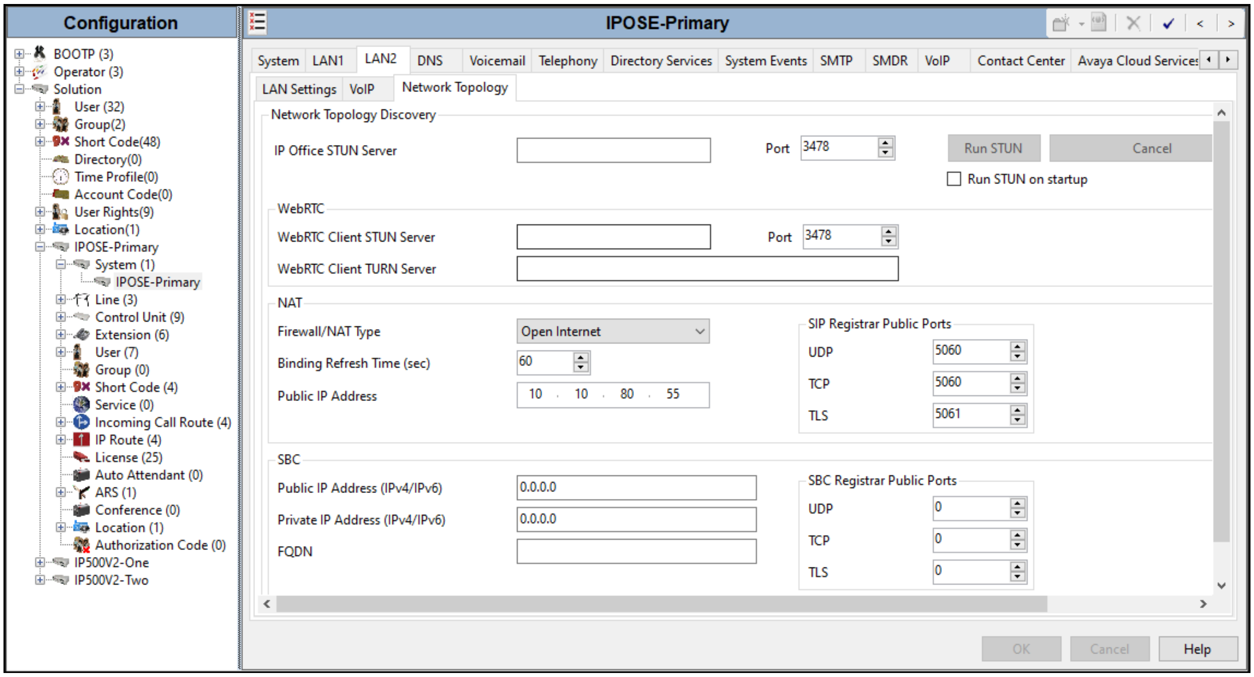
Primary Site Specific Option Number (SSON)	176
Secondary Site Specific Option Number (SSON)	242
VLAN	Not Present
1100 Voice VLAN Site Specific Option Number (SSON)	232
1100 Voice VLAN IDs	

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 Lumen Technologies Voice Complete 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.



Network Topology Discovery

IP Office STUN Server: Port 3478

□ Run STUN on startup

WebRTC

WebRTC Client STUN Server: Port 3478

WebRTC Client TURN Server:

NAT

Firewall/NAT Type: Open Internet

Binding Refresh Time (sec): 60

Public IP Address: 10 . 10 . 80 . 55

SIP Registrar Public Ports

UDP	5060
TCP	5060
TLS	5061

SBC

Public IP Address (IPv4/IPv6): 0.0.0.0

Private IP Address (IPv4/IPv6): 0.0.0.0

FQDN:

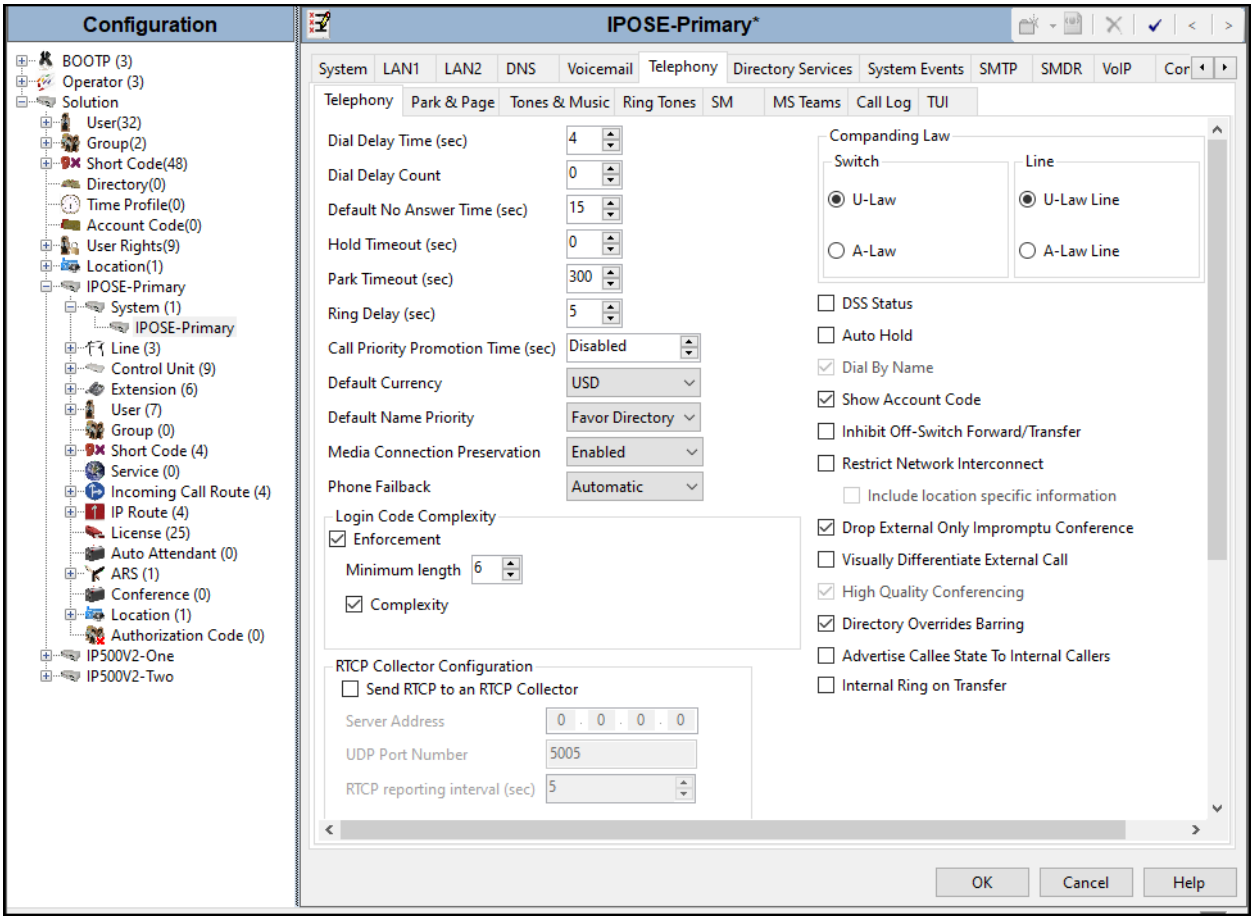
SBC Registrar Public Ports

UDP	0
TCP	0
TLS	0

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.



Dial Delay Time (sec): 4
Dial Delay Count: 0
Default No Answer Time (sec): 15
Hold Timeout (sec): 0
Park Timeout (sec): 300
Ring Delay (sec): 5
Call Priority Promotion Time (sec): Disabled
Default Currency: USD
Default Name Priority: Favor Directory
Media Connection Preservation: Enabled
Phone Failback: Automatic

Login Code Complexity
☑ Enforcement
Minimum length: 6
☑ Complexity

RTCP Collector Configuration
□ Send RTCP to an RTCP Collector
Server Address: 0 . 0 . 0 . 0
UDP Port Number: 5005
RTCP reporting interval (sec): 5

Companding Law
Switch: ● U-Law ○ A-Law
Line: ● U-Law Line ○ A-Law Line

□ DSS Status
□ Auto Hold
☑ Dial By Name
☑ Show Account Code
□ Inhibit Off-Switch Forward/Transfer
□ Restrict Network Interconnect
□ Include location specific information
☑ Drop External Only Impromptu Conference
□ Visually Differentiate External Call
☑ High Quality Conferencing
☑ Directory Overrides Barring
□ Advertise Callee State To Internal Callers
□ Internal Ring on Transfer

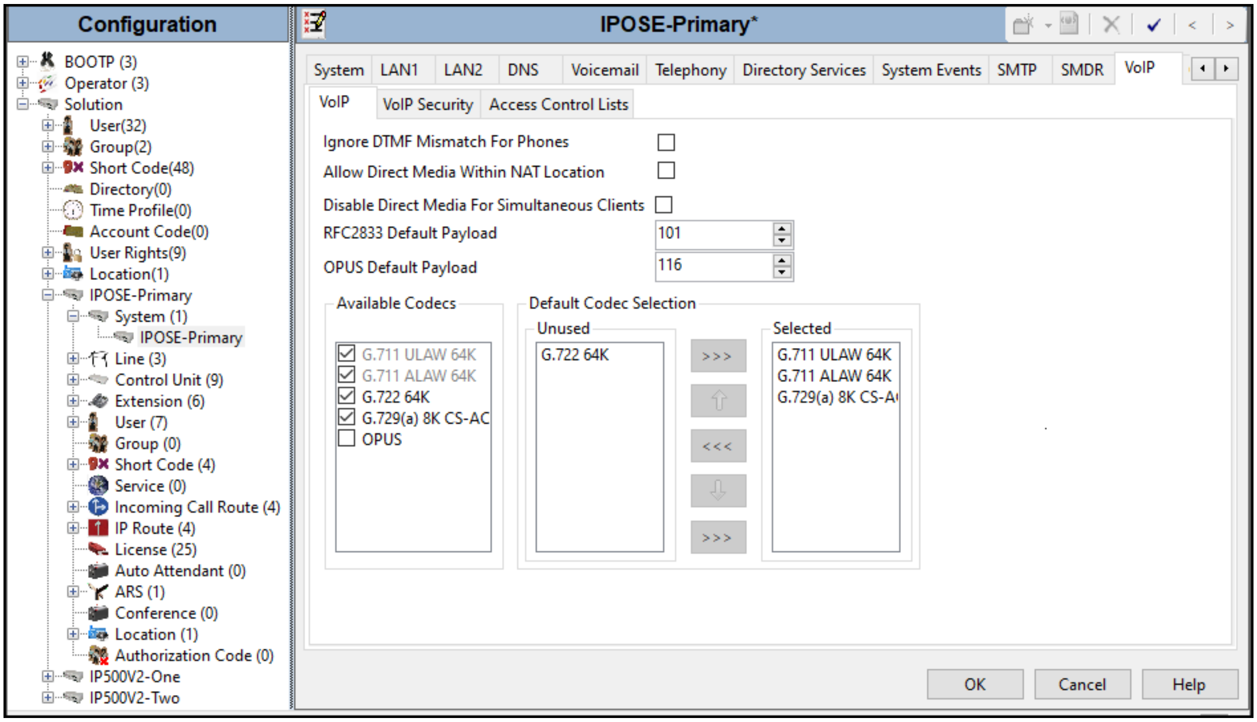
5.2.3. System - VoIP Tab

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), the system's default codecs and order were used.
- Click **OK** to commit.



Ignore DTMF Mismatch For Phones ☐

Allow Direct Media Within NAT Location ☐

Disable Direct Media For Simultaneous Clients ☐

RFC2833 Default Payload 101

OPUS Default Payload 116

Available Codecs

☑ G.711 ULAW 64K
☑ G.711 ALAW 64K
☑ G.722 64K
☑ G.729(a) 8K CS-AC
☐ OPUS

Default Codec Selection

Unused

G.722 64K

Selected

G.711 ULAW 64K
G.711 ALAW 64K
G.729(a) 8K CS-A(

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 for VoIP calls (SIP and H.323). SRTP can be applied between telephones, between ends of an IP trunk or in various other combinations.

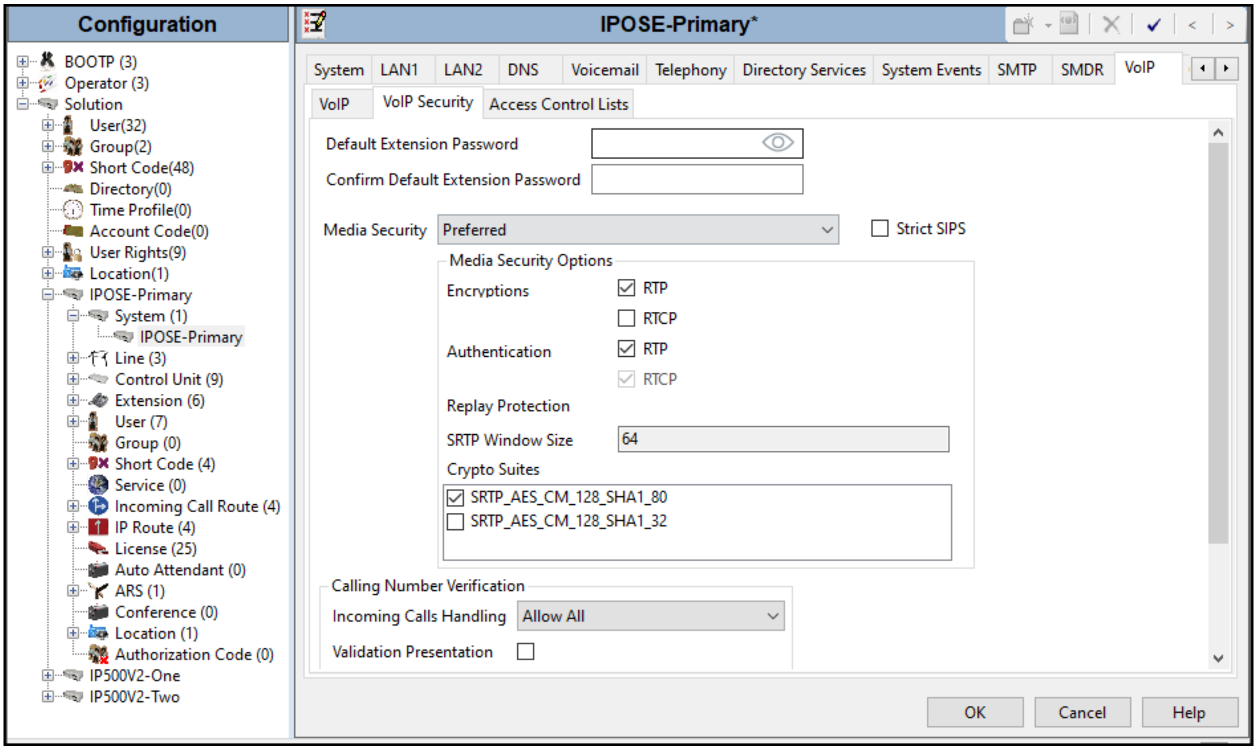
Configuring the use of SRTP at the system level is done on the **VoIP Security** tab using the Media Security setting. The options are:

- 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.

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.



Default Extension Password

Confirm Default Extension Password

Media Security: Preferred

☐ Strict SIPS

Media Security Options

Encryptions: ☑ RTP ☐ RTCP

Authentication: ☑ RTP ☑ RTCP

Replay Protection

SRTP Window Size: 64

Crypto Suites

☑ SRTP_AES_CM_128_SHA1_80

☐ SRTP_AES_CM_128_SHA1_32

Calling Number Verification

Incoming Calls Handling: Allow All

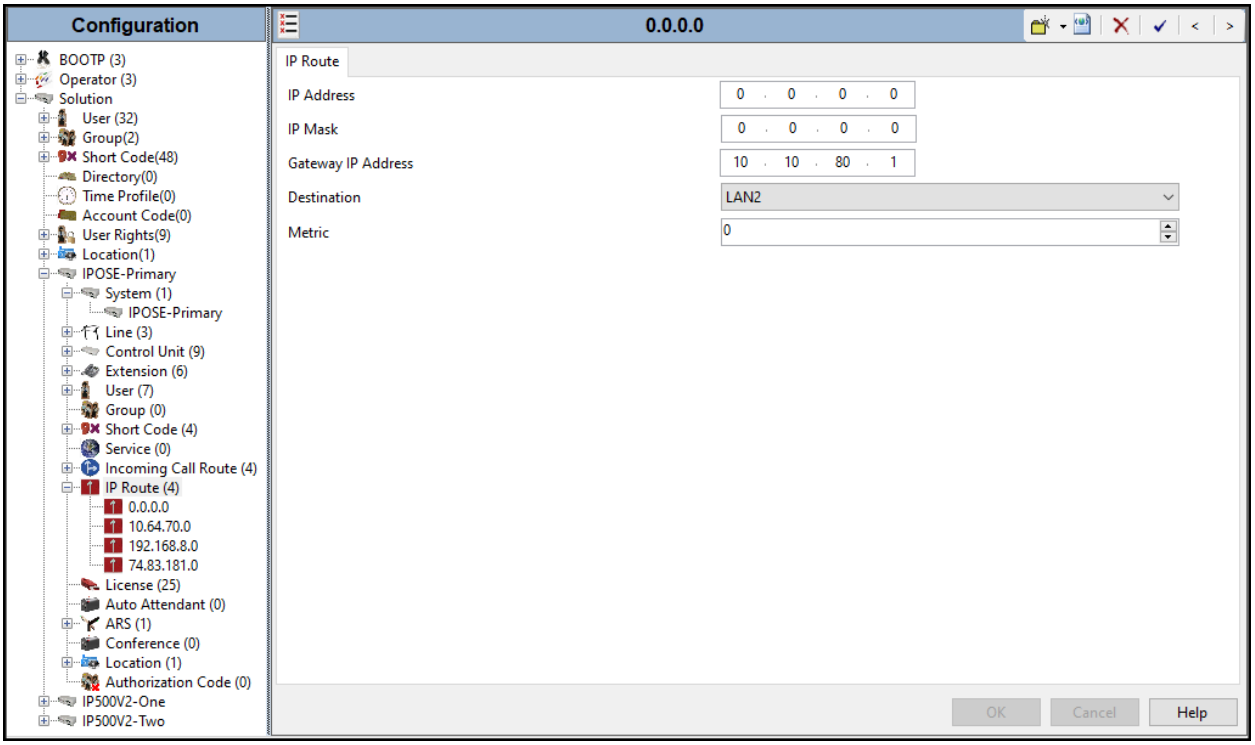
Validation Presentation ☐

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 Lumen Technologies 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.



IP Address: 0 . 0 . 0 . 0
IP Mask: 0 . 0 . 0 . 0
Gateway IP Address: 10 . 10 . 80 . 1
Destination: LAN2
Metric: 0

5.4. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Lumen Technologies. 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**.

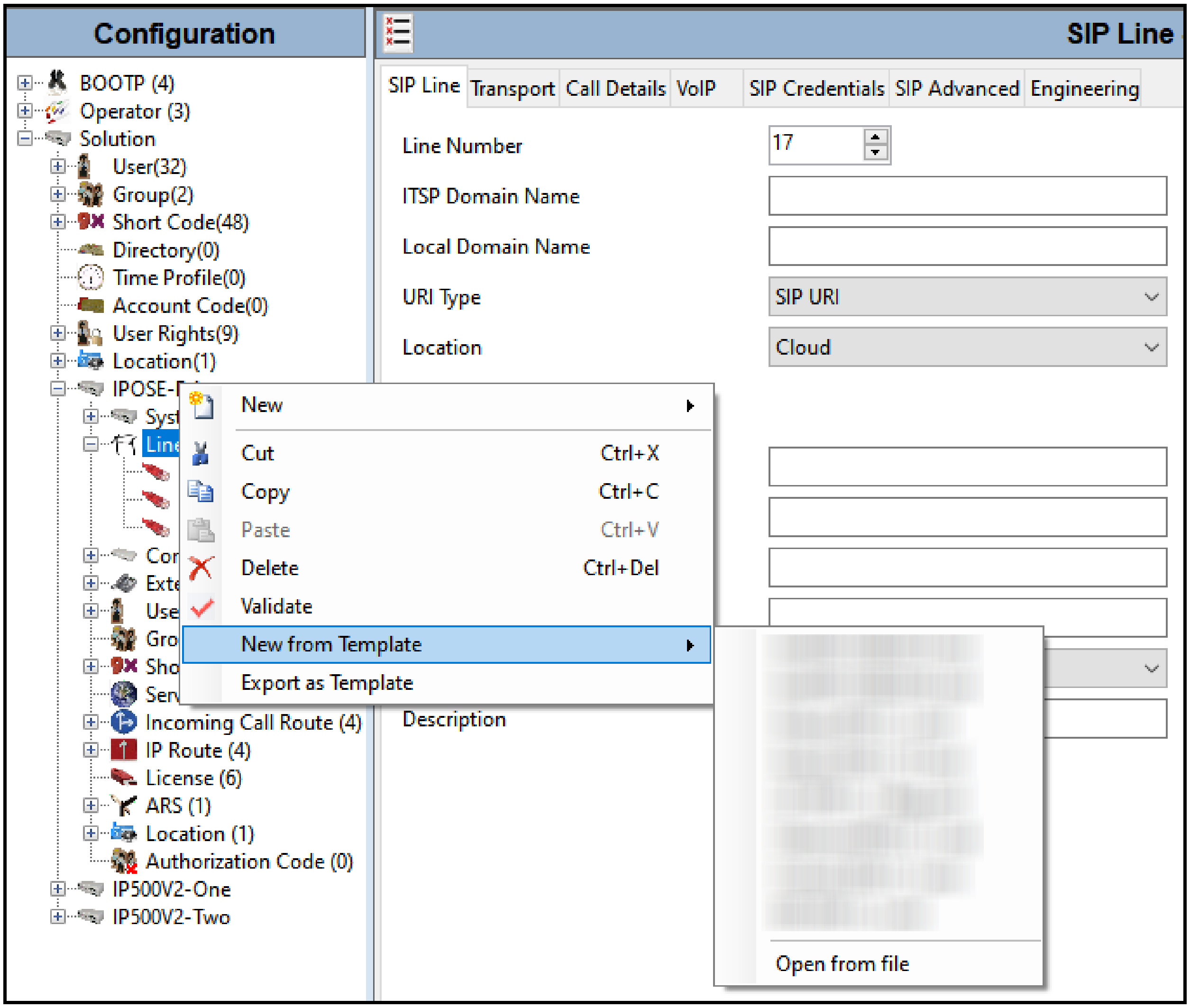
Alternatively, a SIP Line can be created manually. To do so, right-click on **Line** in the **Navigation** pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.2** to **5.4.6**.

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**.



Line Number: 17

ITSP Domain Name:

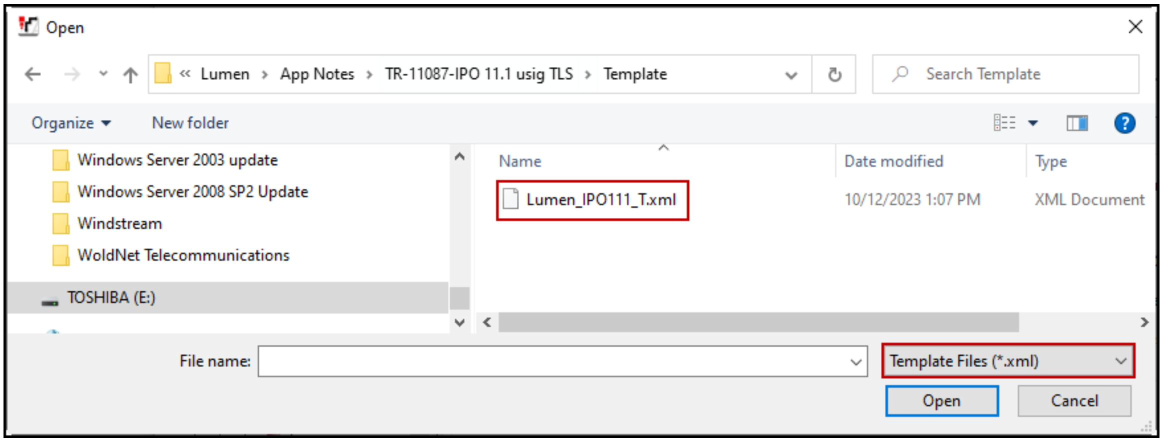
Local Domain Name:

URI Type: SIP URI

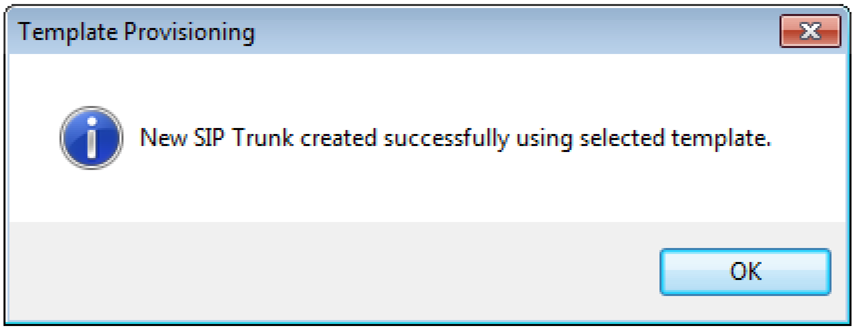
Location: Cloud

Description:

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

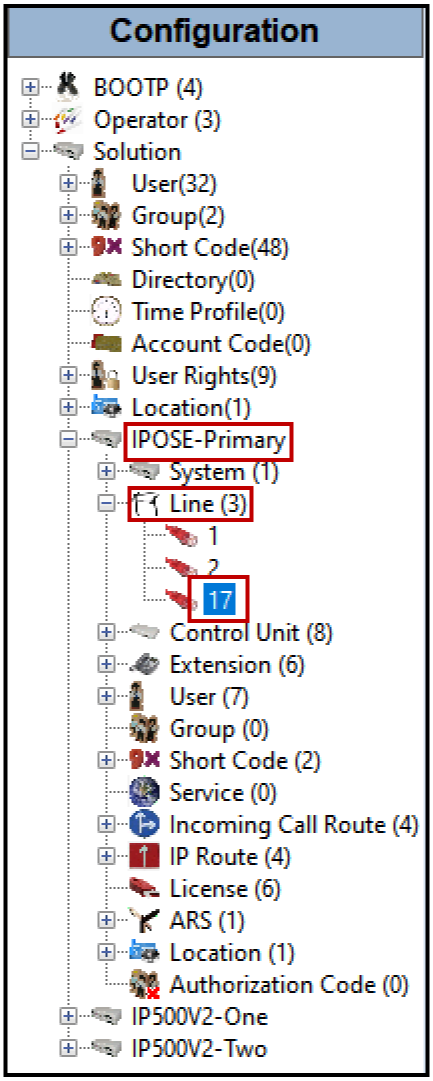


After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.


Template Provisioning

New SIP Trunk created successfully using selected template.

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

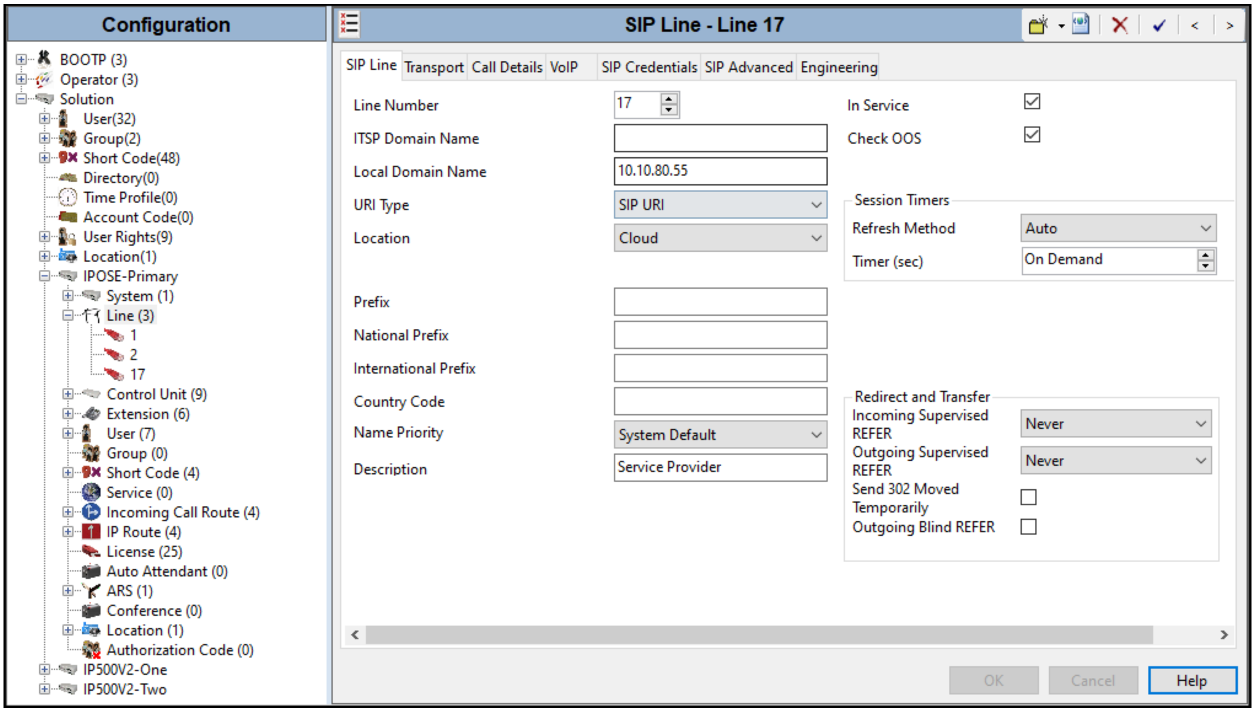


It is important that the SIP Line configuration be reviewed and updated, if necessary, after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2** to **5.4.6**.

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.
- 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 2.1)**.
- Click **OK** to commit.



Line Number: 17
ITSP Domain Name:
Local Domain Name: 10.10.80.55
URI Type: SIP URI
Location: Cloud
Prefix:
National Prefix:
International Prefix:
Country Code:
Name Priority: System Default
Description: Service Provider
In Service ☑
Check OOS ☑

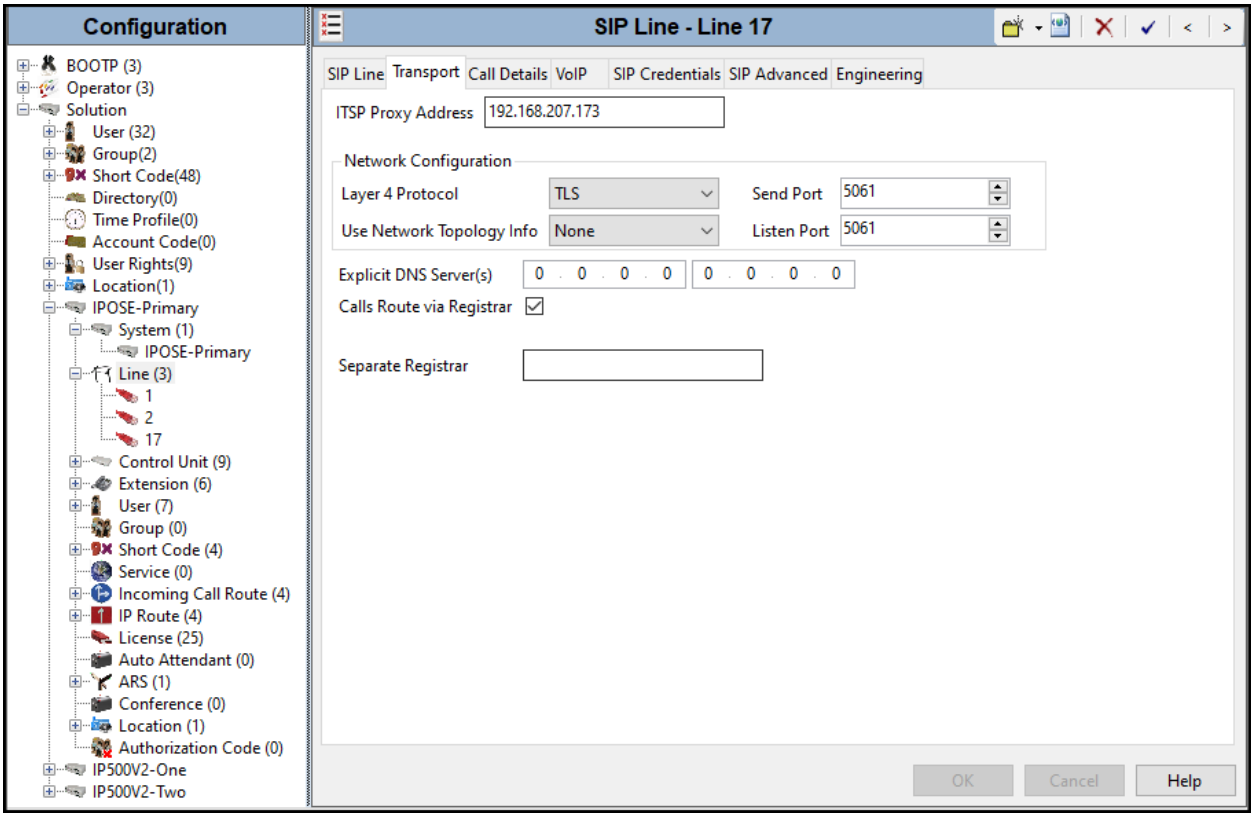
Session Timers
Refresh Method: Auto
Timer (sec): On Demand

Redirect and Transfer
Incoming Supervised REFER: Never
Outgoing Supervised REFER: Never
Send 302 Moved Temporarily ☐
Outgoing Blind REFER ☐

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 Lumen's SIP Proxy server (**192.168.207.173**) as shown in **Figure 1**. This information should be provided by Lumen.
- Set **Layer 4 Protocol** to **TLS**.
- Set **Use Network Topology Info** to **None** (see note below).
- Set the **Send** and **Listening Ports** to **5061**.
- Default values may be used for all other parameters.
- Click **OK** to commit.



SIP Line - Line 17

ITSP Proxy Address: 192.168.207.173

Network Configuration

Layer 4 Protocol: TLS

Send Port: 5061

Use Network Topology Info: None

Listen Port: 5061

Explicit DNS Server(s): 0 . 0 . 0 . 0   0 . 0 . 0 . 0

Calls Route via Registrar ☑

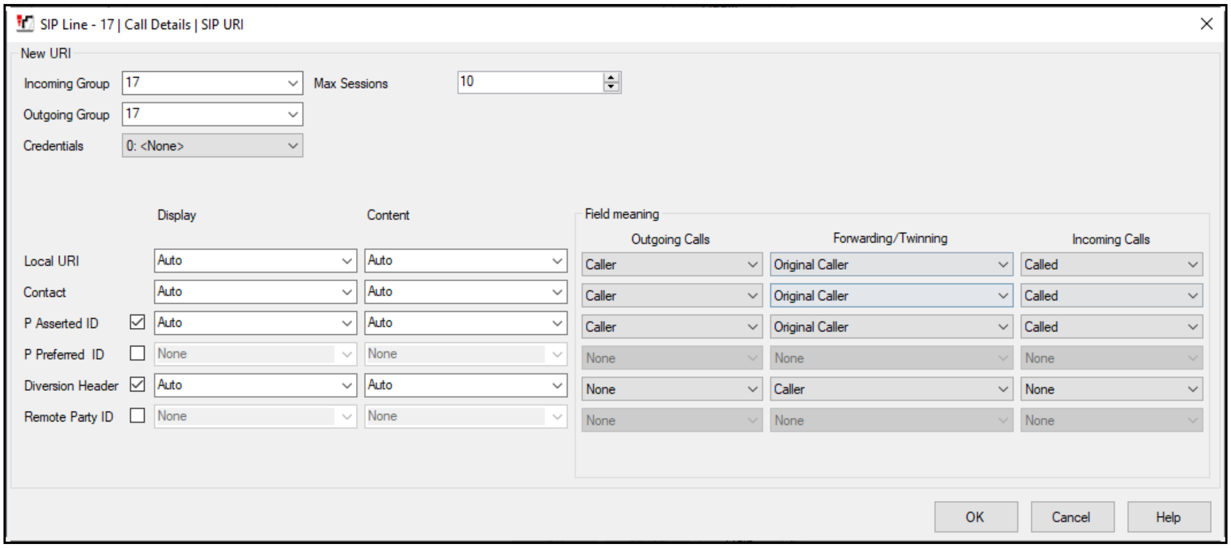
Separate Registrar:

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.

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 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 **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.
- 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.



SIP Line - 17 | Call Details | SIP URI

New URI

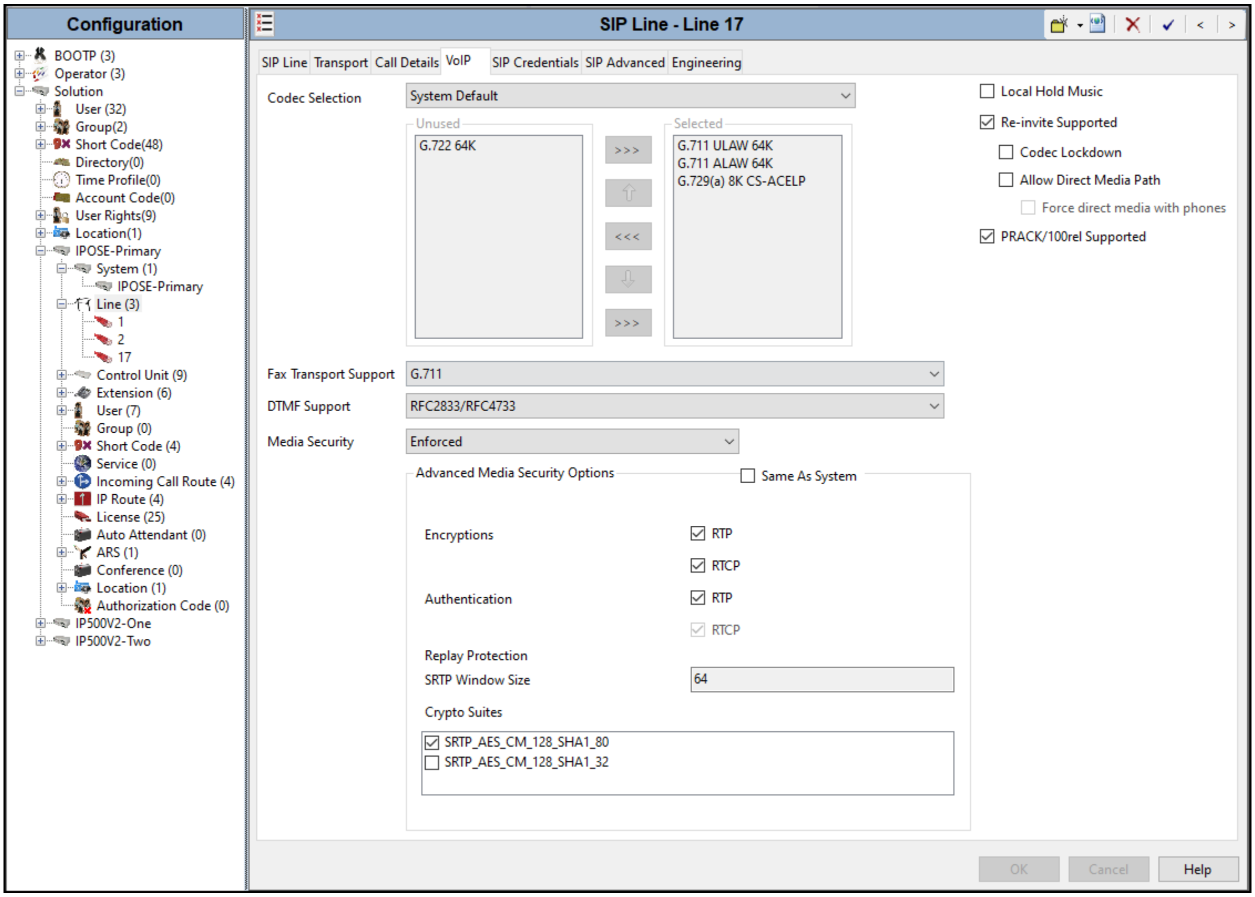
Incoming Group: 17
Max Sessions: 10
Outgoing Group: 17
Credentials: 0: <None>

		Display	Content	Field meaning: Outgoing Calls	Forwarding/Twinning	Incoming Calls
Local URI		Auto	Auto	Caller	Original Caller	Called
Contact		Auto	Auto	Caller	Original Caller	Called
P Asserted ID	☑	Auto	Auto	Caller	Original Caller	Called
P Preferred ID	□	None	None	None	None	None
Diversion Header	☑	Auto	Auto	None	Caller	None
Remote Party ID	□	None	None	None	None	None

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. The buttons allow setting the specific order of preference for the codecs to be used on the SIP Line, as shown. Lumen Technologies supports codecs **G.711ULAW**, **G.711ALAW and G.729(a)** for audio.
- Select **G.711** for **Fax Transport Support** (Refer to **Section 2.2**).
- 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 **Enforced**. **Note**: **Note**: Lumen requires **Media Security** to be set as **Enforced**, otherwise calls will fail.
- On the **Advanced Media Security Options**, under **Encryptions**, check **RTP** and **RTCP**. Under **Authentication** check **RTP** (RTCP is already checked by default). **Note**: These settings are required, otherwise calls will fail.
- 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.



Codec Selection: System Default

Unused: G.722 64K

Selected: G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP

□ Local Hold Music
☑ Re-invite Supported
□ Codec Lockdown
□ Allow Direct Media Path
□ Force direct media with phones
☑ PRACK/100rel Supported

Fax Transport Support: G.711

DTMF Support: RFC2833/RFC4733

Media Security: Enforced

Advanced Media Security Options □ Same As System

Encryptions ☑ RTP ☑ RTCP

Authentication ☑ RTP ☑ RTCP

Replay Protection

SRTP Window Size: 64

Crypto Suites
☑ SRTP_AES_CM_128_SHA1_80
□ SRTP_AES_CM_128_SHA1_32

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).

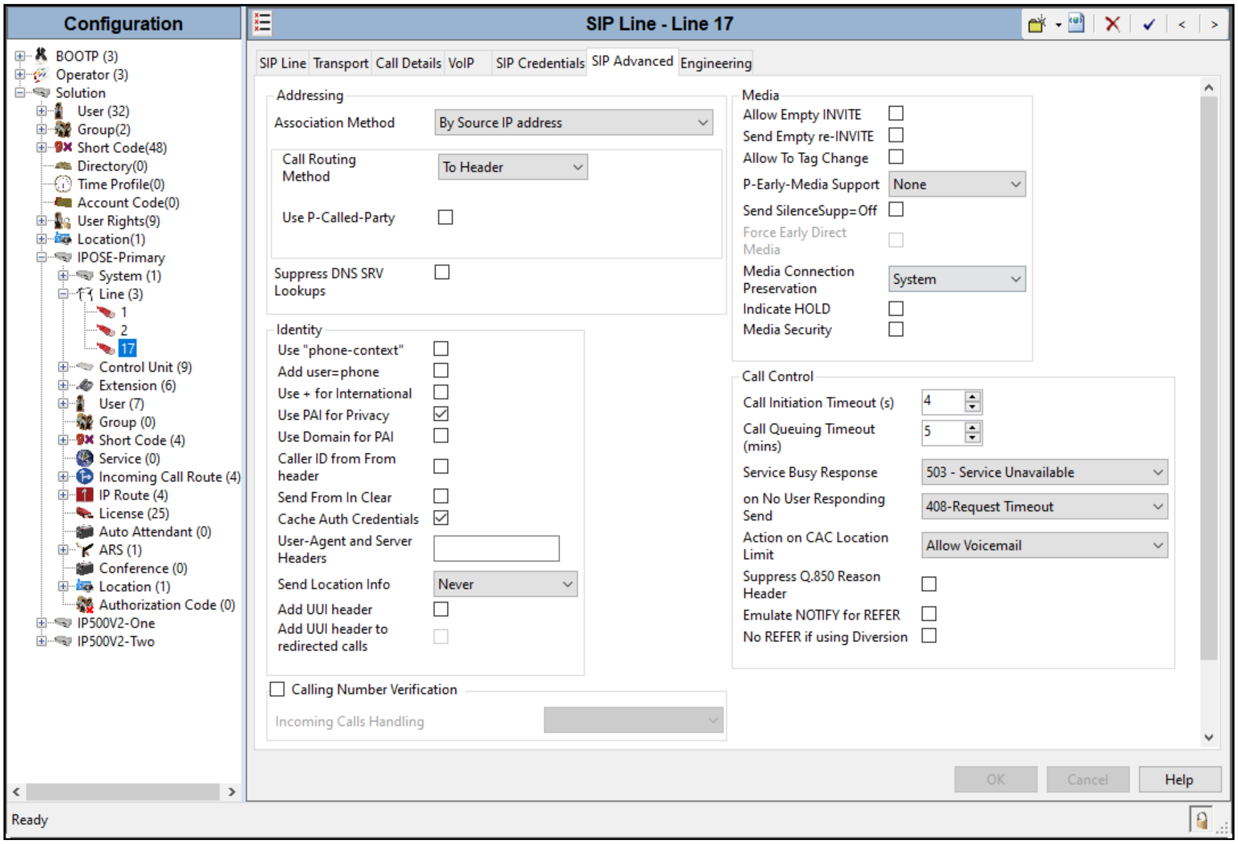
5.4.6. SIP Line – SIP Advanced Tab

In the **Addressing** area:

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

In the **Identity** area:

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



Configuration

- BOOTP (3)
- Operator (3)
- Solution
  - User (32)
  - Group(2)
  - Short Code(48)
  - Directory(0)
  - Time Profile(0)
  - Account Code(0)
  - User Rights(9)
  - Location(1)
  - IPOSE-Primary
    - System (1)
    - Line (3)
      - 1
      - 2
      - 17
    - Control Unit (9)
    - Extension (6)
    - User (7)
    - Group (0)
    - Short Code (4)
    - Service (0)
    - Incoming Call Route (4)
    - IP Route (4)
    - License (25)
    - Auto Attendant (0)
    - ARS (1)
    - Conference (0)
    - Location (1)
    - Authorization Code (0)
  - IP500V2-One
  - IP500V2-Two

Ready

SIP Line - Line 17

SIP Line | Transport | Call Details | VoIP | SIP Credentials | SIP Advanced | Engineering

Addressing

Association Method: By Source IP address

Call Routing Method: To Header

Use P-Called-Party ☐

Suppress DNS SRV Lookups ☐

Identity

Use "phone-context" ☐

Add user=phone ☐

Use + for International ☐

Use PAI for Privacy ☑

Use Domain for PAI ☐

Caller ID from From header ☐

Send From In Clear ☐

Cache Auth Credentials ☑

User-Agent and Server Headers:

Send Location Info: Never

Add UUI header ☐

Add UUI header to redirected calls ☐

☐ Calling Number Verification

Incoming Calls Handling:

Media

Allow Empty INVITE ☐

Send Empty re-INVITE ☐

Allow To Tag Change ☐

P-Early-Media Support: None

Send SilenceSupp=Off ☐

Force Early Direct Media ☐

Media Connection Preservation: System

Indicate HOLD ☐

Media Security ☐

Call Control

Call Initiation Timeout (s): 4

Call Queuing Timeout (mins): 5

Service Busy Response: 503 - Service Unavailable

on No User Responding Send: 408-Request Timeout

Action on CAC Location Limit: Allow Voicemail

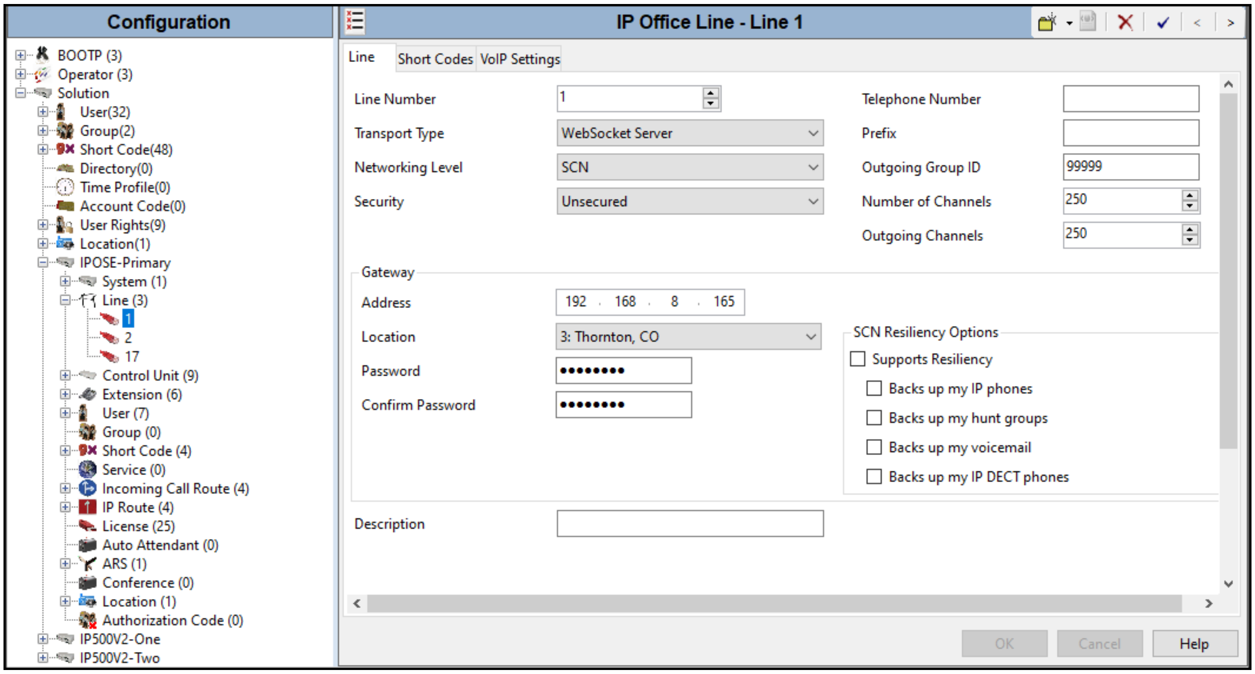
Suppress Q.850 Reason Header ☐

Emulate NOTIFY for REFER ☐

No REFER if using Diversion ☐

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.



Line Number: 1

Transport Type: WebSocket Server

Networking Level: SCN

Security: Unsecured

Telephone Number:

Prefix:

Outgoing Group ID: 99999

Number of Channels: 250

Outgoing Channels: 250

Gateway

Address: 192 . 168 . 8 . 165

Location: 3: Thornton, CO

Password: ••••••••

Confirm Password: ••••••••

SCN Resiliency Options

□ Supports Resiliency

□ Backs up my IP phones

□ Backs up my hunt groups

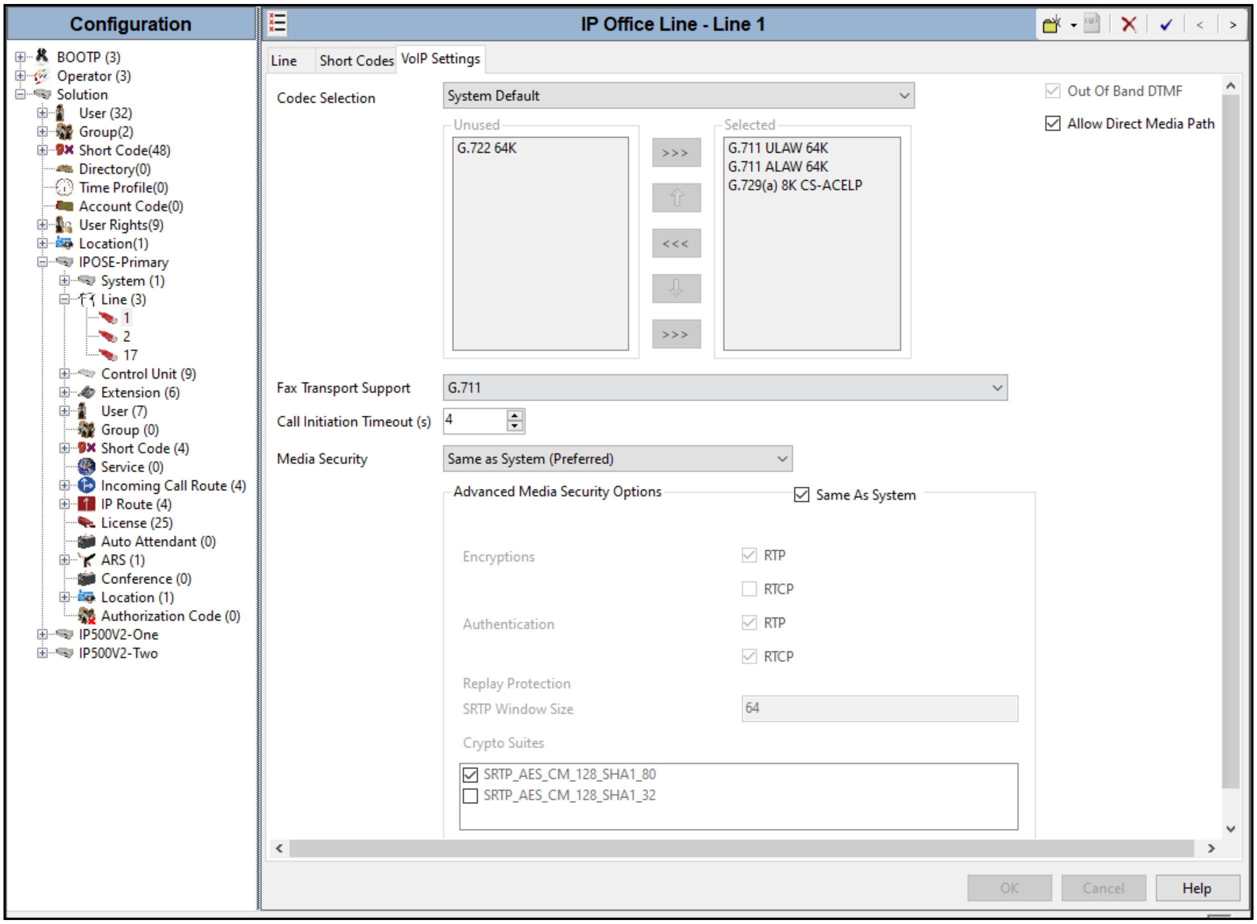
□ Backs up my voicemail

□ Backs up my IP DECT phones

Description:

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 2.2**).
- 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.



IP Office Line - Line 1

Line | Short Codes | VoIP Settings

Codec Selection: System Default

Unused: G.722 64K

Selected: G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP

☑ Out Of Band DTMF

☑ Allow Direct Media Path

Fax Transport Support: G.711

Call Initiation Timeout (s): 4

Media Security: Same as System (Preferred)

Advanced Media Security Options ☑ Same As System

Encryptions: ☑ RTP ☐ RTCP

Authentication: ☑ RTP ☑ RTCP

Replay Protection

SRTP Window Size: 64

Crypto Suites

☑ SRTP_AES_CM_128_SHA1_80

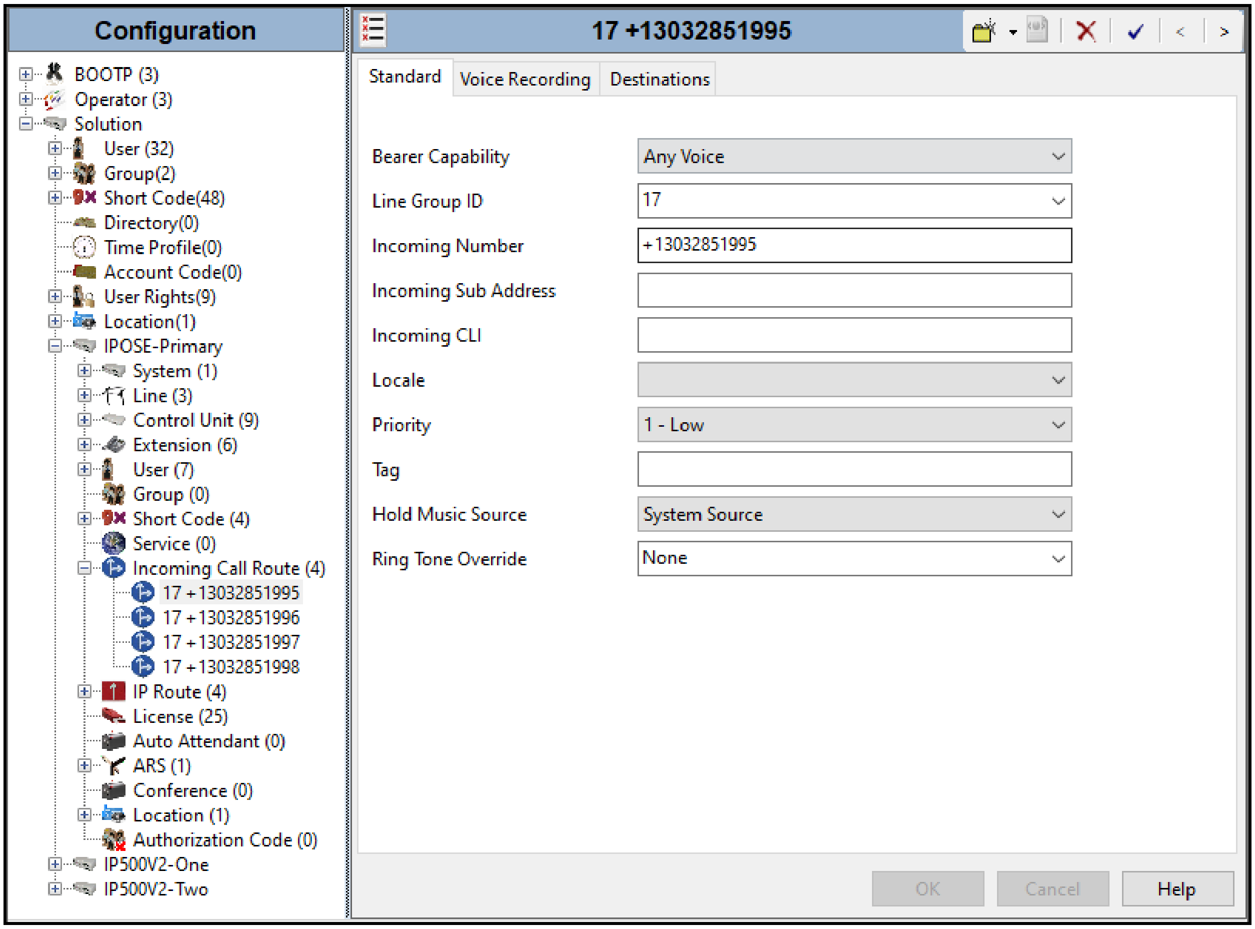
☐ SRTP_AES_CM_128_SHA1_32

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

5.6. Incoming Call Route

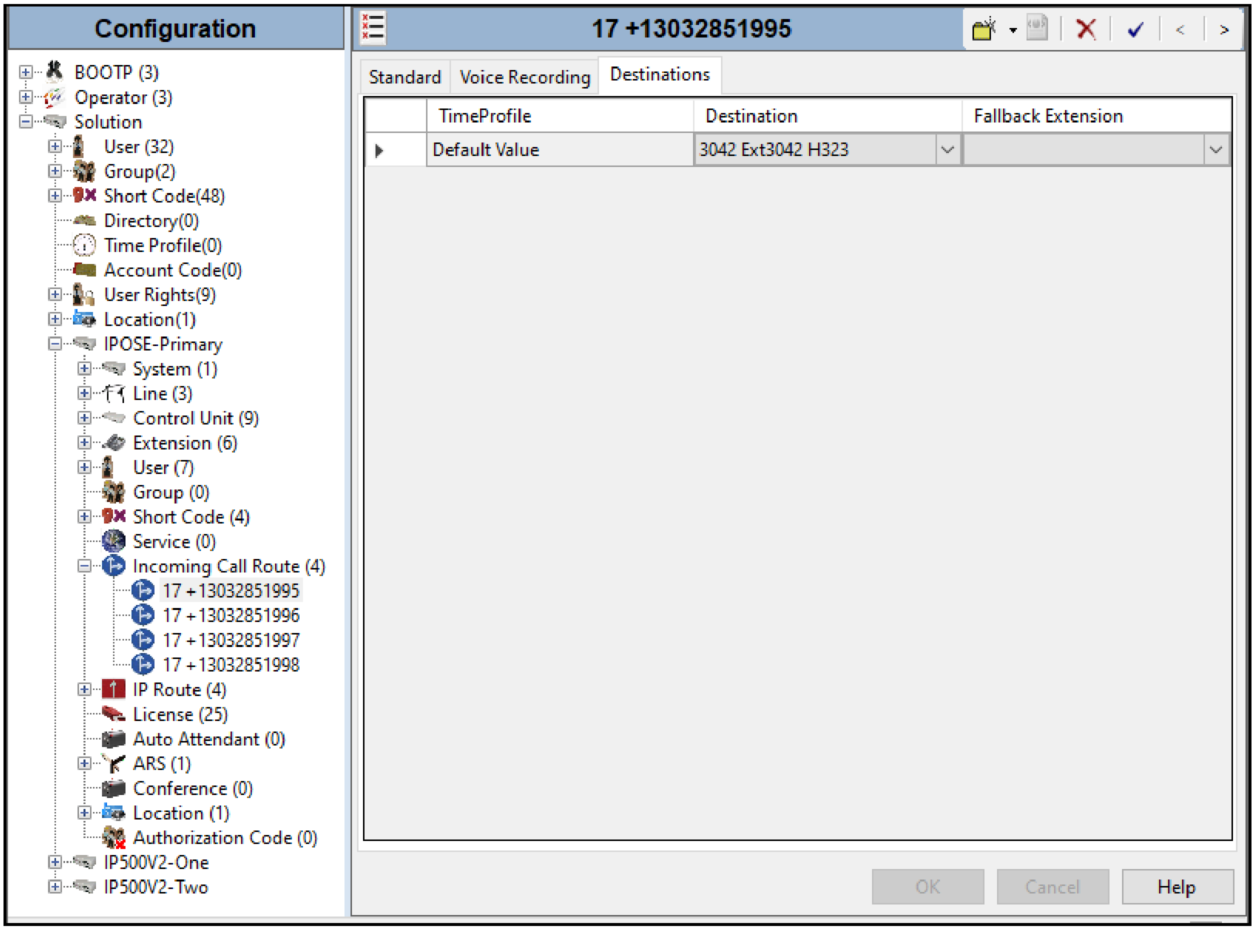
Incoming call routes map inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc., within the IP Office system. To add an incoming call route, right click on **Incoming Call Route** in the **Navigation** pane and select **New** (not shown). On the Details Pane, under 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 Lumen Technologies. Notice that a **plus** sign (+) was added preceding the DID number, this is required to comply with the E.164 numbering format.
- Default values may be used for all other parameters.
- Click **OK** to commit.



Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	+13032851995
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

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 +13032851995 provided by Lumen Technologies was associated with the Avaya IP Office extension **3042**.



17 +13032851995

	TimeProfile	Destination	Fallback Extension
▶	Default Value	3042 Ext3042 H323	

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.).

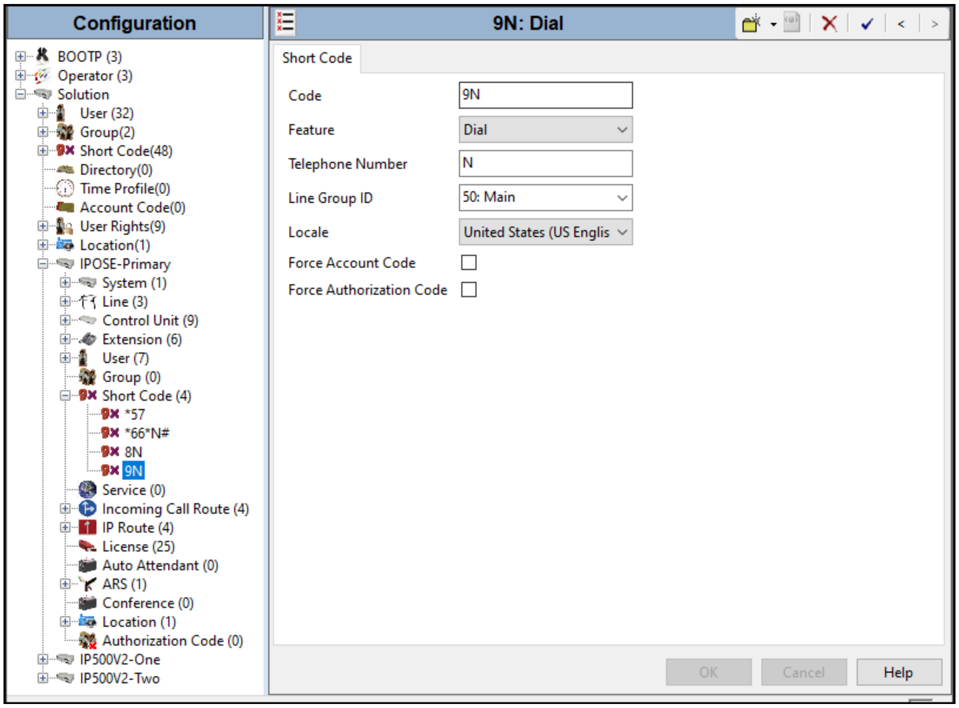
5.7. Outbound Call Routing

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.



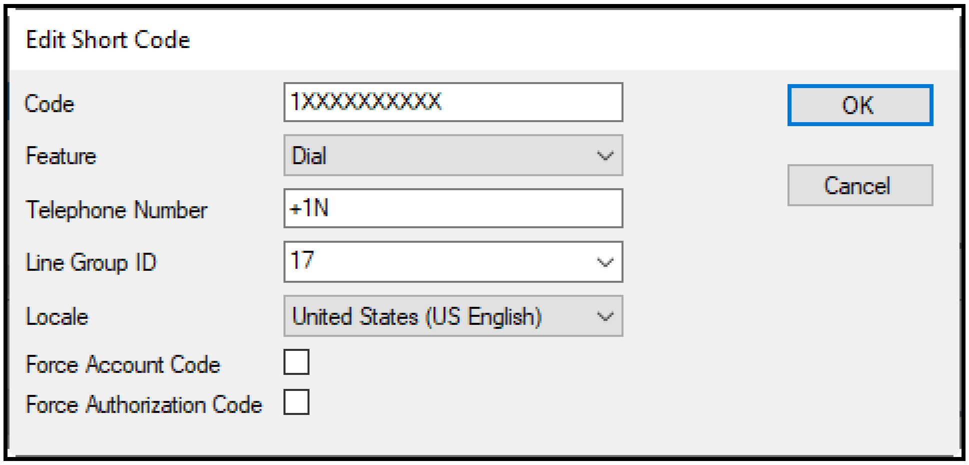
Code	9N
Feature	Dial
Telephone Number	N
Line Group ID	50: Main
Locale	United States (US Englis
Force Account Code	□
Force Authorization Code	□

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). The **plus** (+) was added to comply with the E.164 numbering format, it will be included preceding the number in SIP headers (e.g., Request-URI, From, To, etc.) destined for Lumen (e.g., +17863311234).
- 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.



Edit Short Code

Code: 1XXXXXXXXXX

Feature: Dial

Telephone Number: +1N

Line Group ID: 17

Locale: United States (US English)

Force Account Code ☐

Force Authorization Code ☐

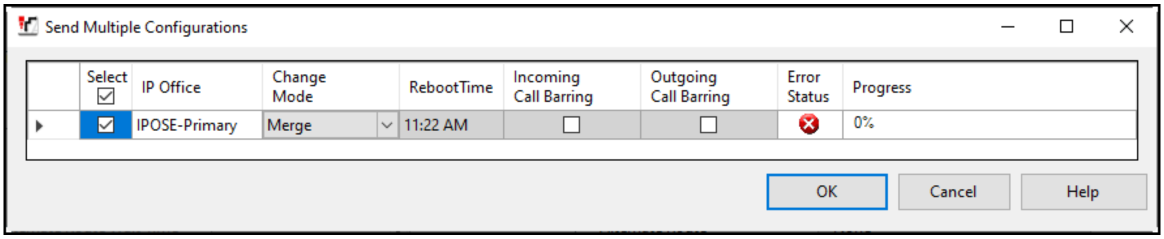
Repeat the above procedure for additional dial patterns to be used by the enterprise to dial out from IP Office, including route patterns for calls to special numbers (411, etc.).

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.



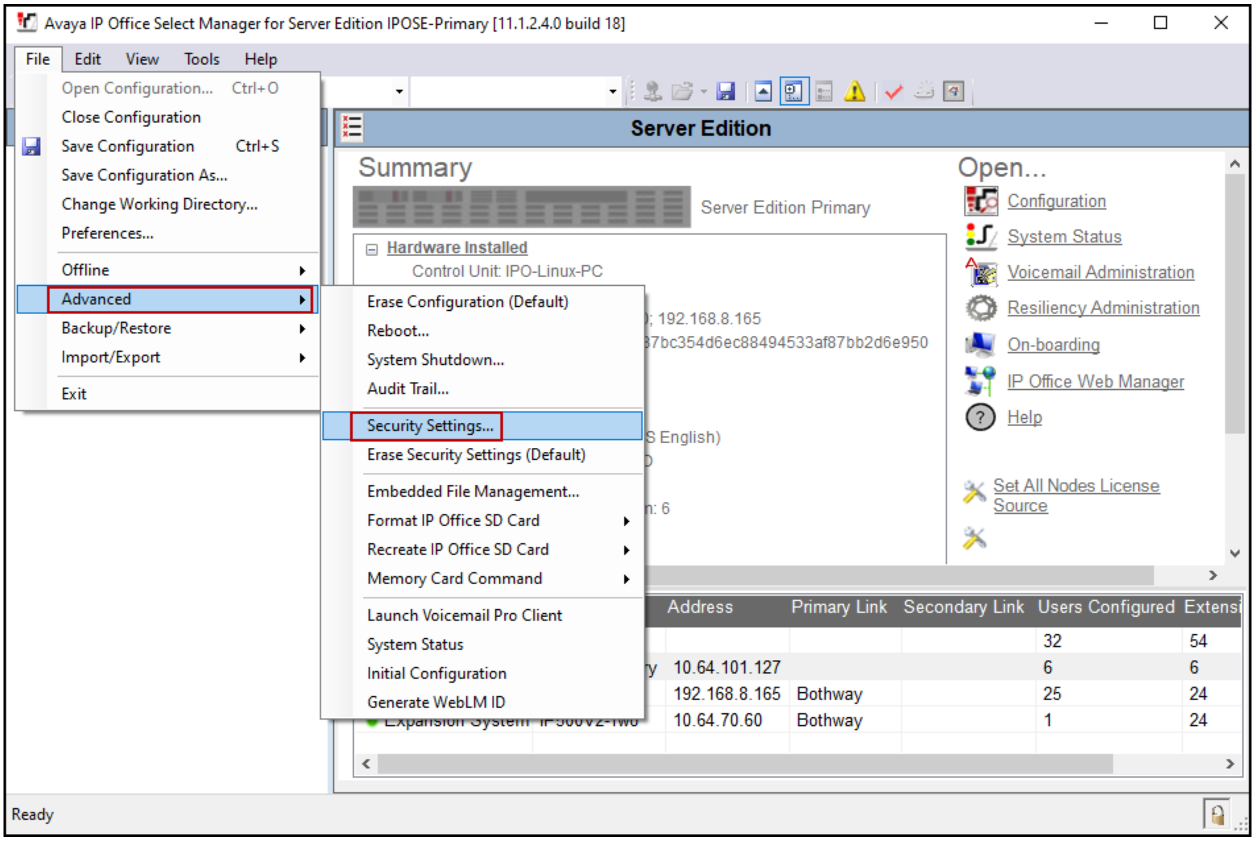
	Select ☑	IP Office	Change Mode	RebootTime	Incoming Call Barring	Outgoing Call Barring	Error Status	Progress
▸	☑	IPOSE-Primary	Merge	11:22 AM	□	□	☒	0%

5.9. TLS Management

Note – Testing was done with identity certificates signed by a 3rd party trusted certificate authority (CA) for enhanced security to enable TLS encryption outside of the enterprise (public network side, between Lumen and the Enterprise). Verify the Root CA certificate for the trusted certificate authority being used by the Service Provider is present in the **IP Office Trusted Certificate Store**, required to enable TLS encryption outside of the enterprise (public network side, between Lumen and the Enterprise). This Root CA certificate needs to be manually loaded/installed in Avaya IP Office; this Root CA certificate doesn't come pre-loaded in IP Office. The Service Provider could provide the Root CA certificate to the customer, or the customer could download it directly from the 3rd party trusted Certificate Authority web/home page. The name of the 3rd party trusted Certificate Authority (CA) being used by the Service Provider is required when downloading it from the 3rd party trusted Certificate Authority web/home page. The Service provider can guide the customer on how to obtain the necessary certificate(s).

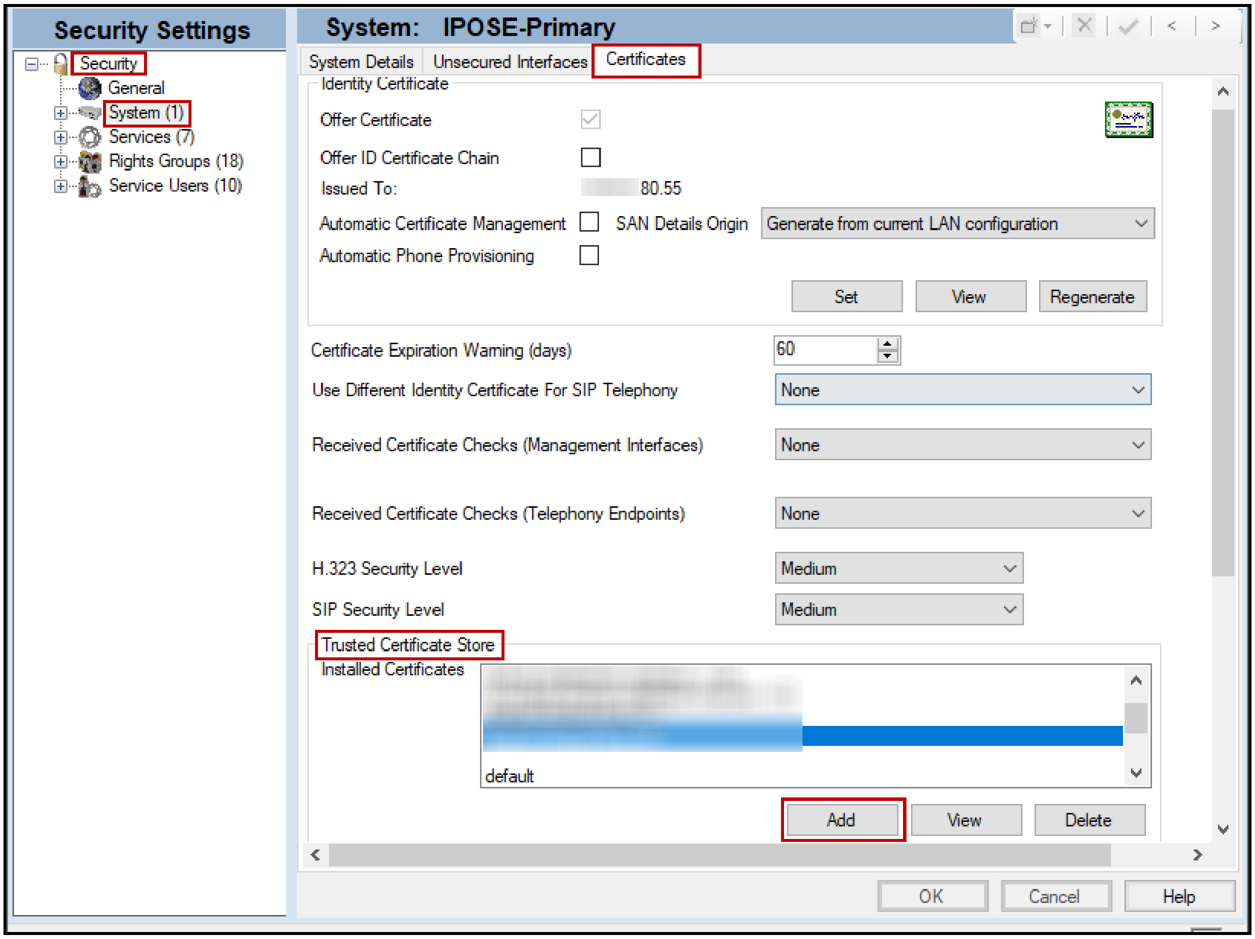
The following procedure shows how to add a 3rd party trusted certificate authority (CA) Root CA to the IP Office **Trusted Certificate Store**.

Access the IP Office **Security Settings** by selecting **File → Advanced** on the top-left of the IP Office Manager screen:



Under **Security Settings**, Select **System** (**1**), then select the **Certificates** tab.

- Under the **Trusted Certificate Store** add the 3rd party trusted certificate authority (CA) Root CA certificate by selecting **Add** and following the prompts given, as shown below:



System: IPOSE-Primary

Identity Certificate

Offer Certificate ☑

Offer ID Certificate Chain □

Issued To: [redacted]80.55

Automatic Certificate Management □ SAN Details Origin: Generate from current LAN configuration

Automatic Phone Provisioning □

Certificate Expiration Warning (days): 60

Use Different Identity Certificate For SIP Telephony: None

Received Certificate Checks (Management Interfaces): None

Received Certificate Checks (Telephony Endpoints): None

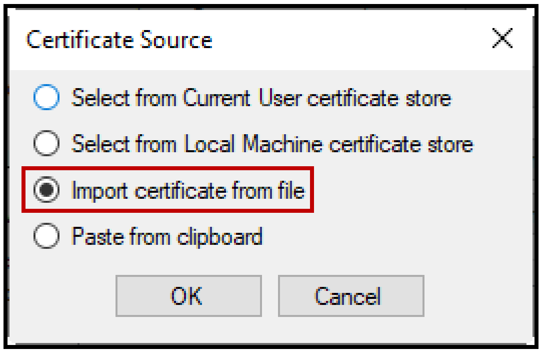
H.323 Security Level: Medium

SIP Security Level: Medium

Trusted Certificate Store

Installed Certificates: [redacted] default

The 3rd party trusted certificate authority (CA) Root CA certificate was imported into IP Office in **.pem** file format:



Certificate Source

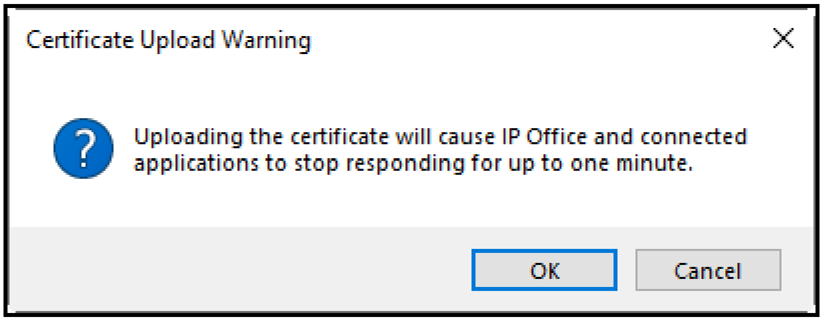
○ Select from Current User certificate store

○ Select from Local Machine certificate store

● Import certificate from file

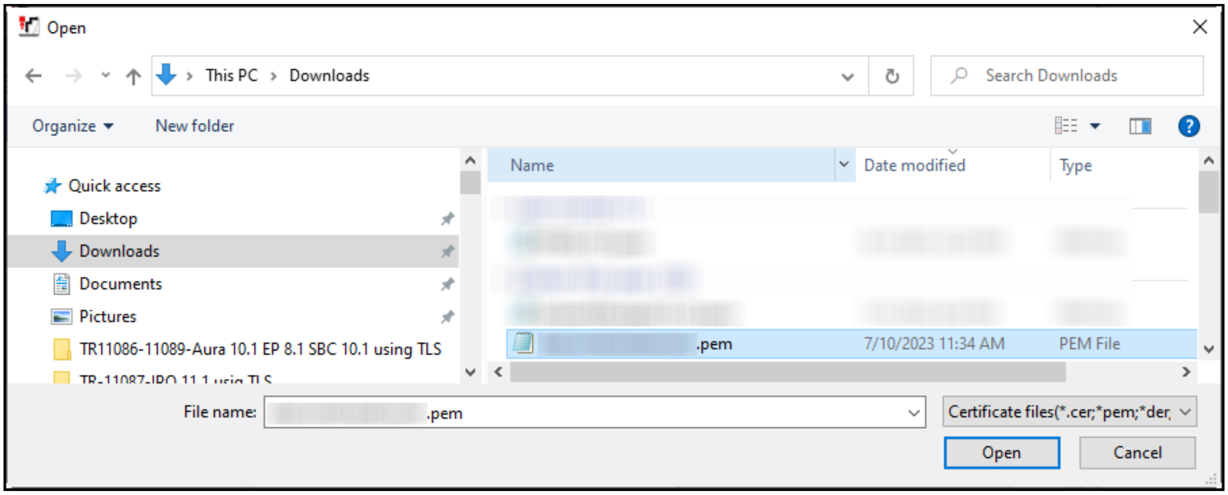
○ Paste from clipboard

Important: Notice the service impact warning given:


Certificate Upload Warning

Uploading the certificate will cause IP Office and connected applications to stop responding for up to one minute.

Select the Root CA certificate file and click **Open**:


Open

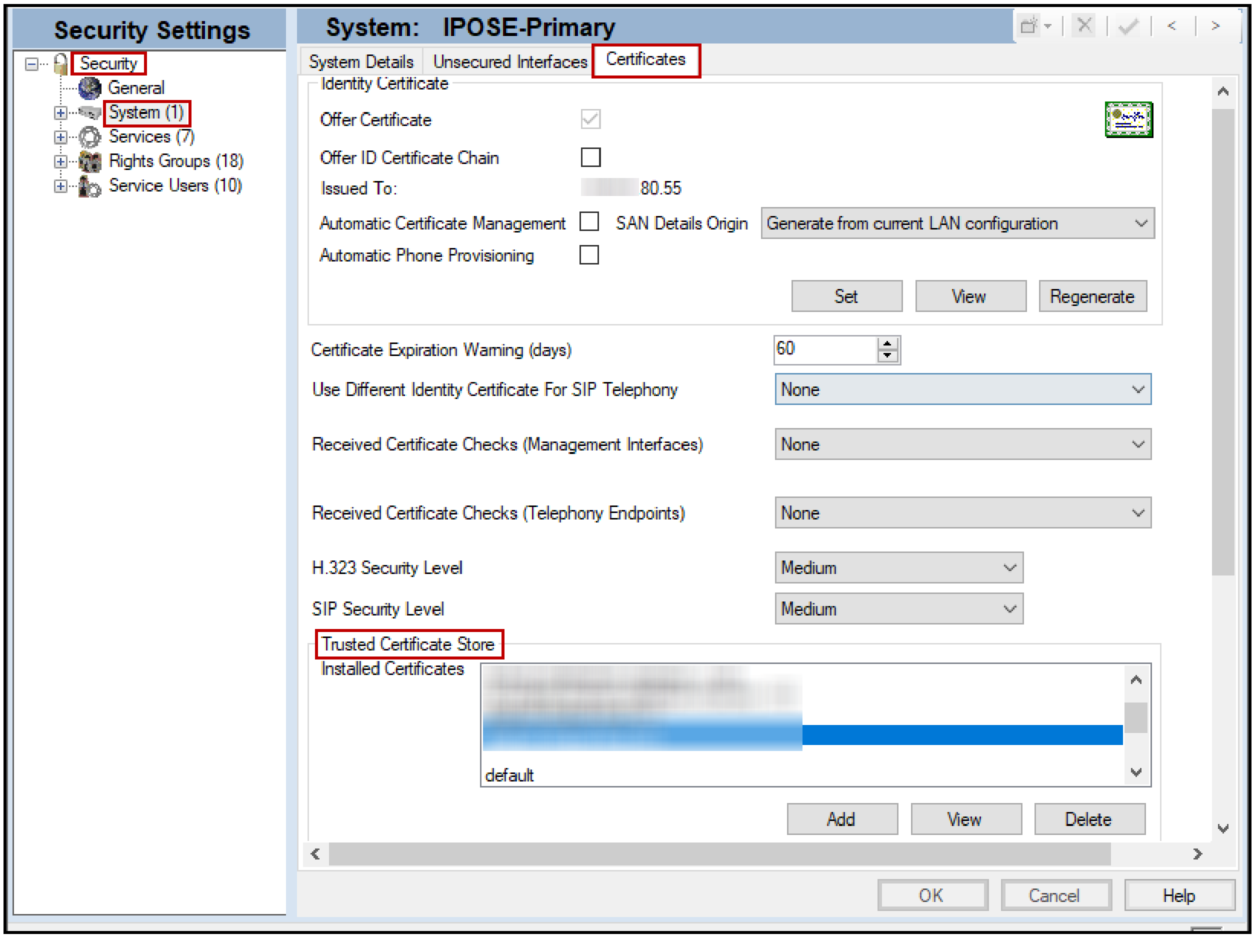
This PC > Downloads

Name	Date modified	Type
[redacted]	[redacted]	[redacted]
[redacted].pem	7/10/2023 11:34 AM	PEM File

File name: [redacted].pem

Certificate files(*.cer;*pem;*der,

- Click **OK** to commit.
- Under **File** select **Save Security Settings** to save the changes. **Note**: Clicking on **OK** to **Commit** will not save the changes, changes are only saved by selecting **Save Security Settings** under **File**.



Identity Certificate

Offer Certificate ☑

Offer ID Certificate Chain □

Issued To: [redacted]80.55

Automatic Certificate Management □ SAN Details Origin: Generate from current LAN configuration

Automatic Phone Provisioning □

Certificate Expiration Warning (days): 60

Use Different Identity Certificate For SIP Telephony: None

Received Certificate Checks (Management Interfaces): None

Received Certificate Checks (Telephony Endpoints): None

H.323 Security Level: Medium

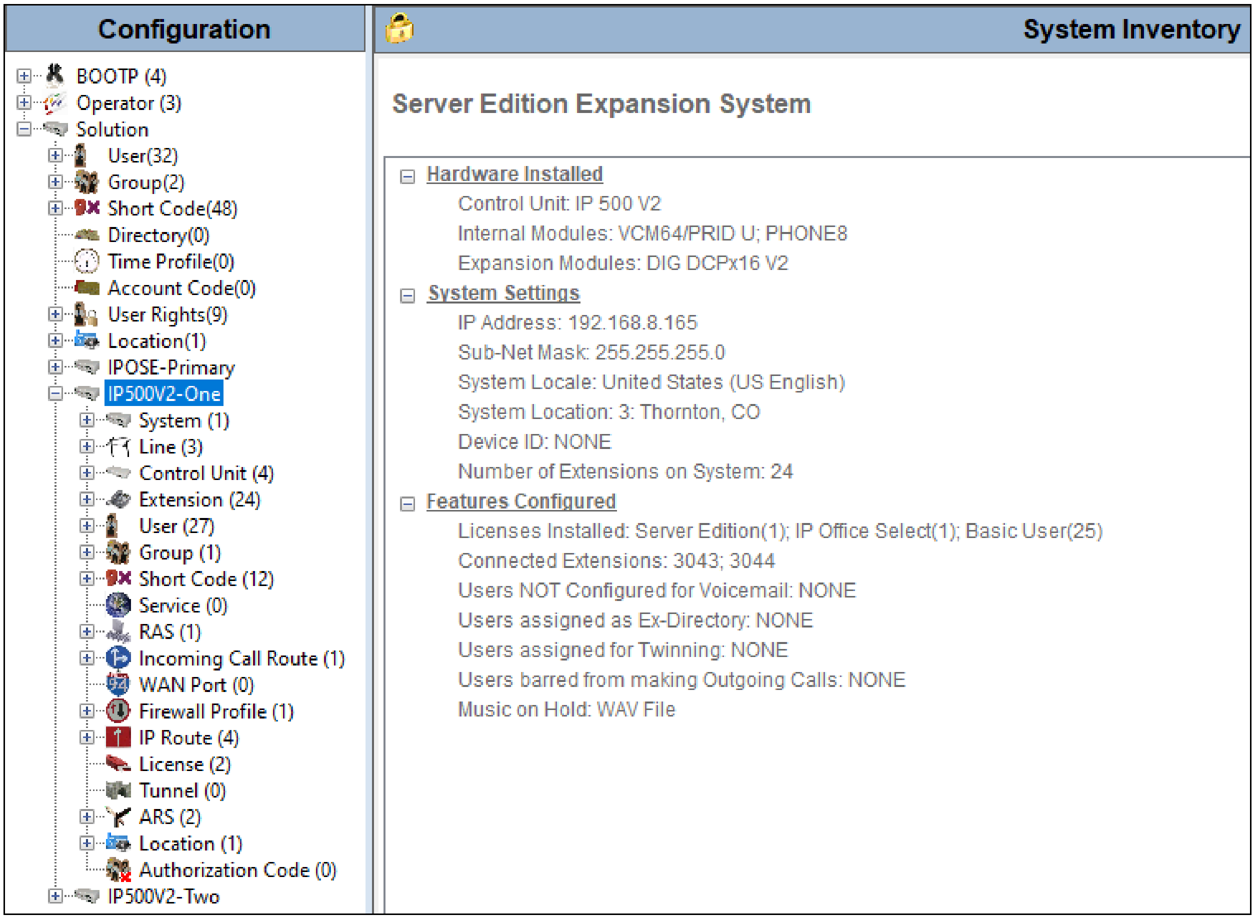
SIP Security Level: Medium

Trusted Certificate Store

Installed Certificates: [redacted]
default

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.



Server Edition Expansion System

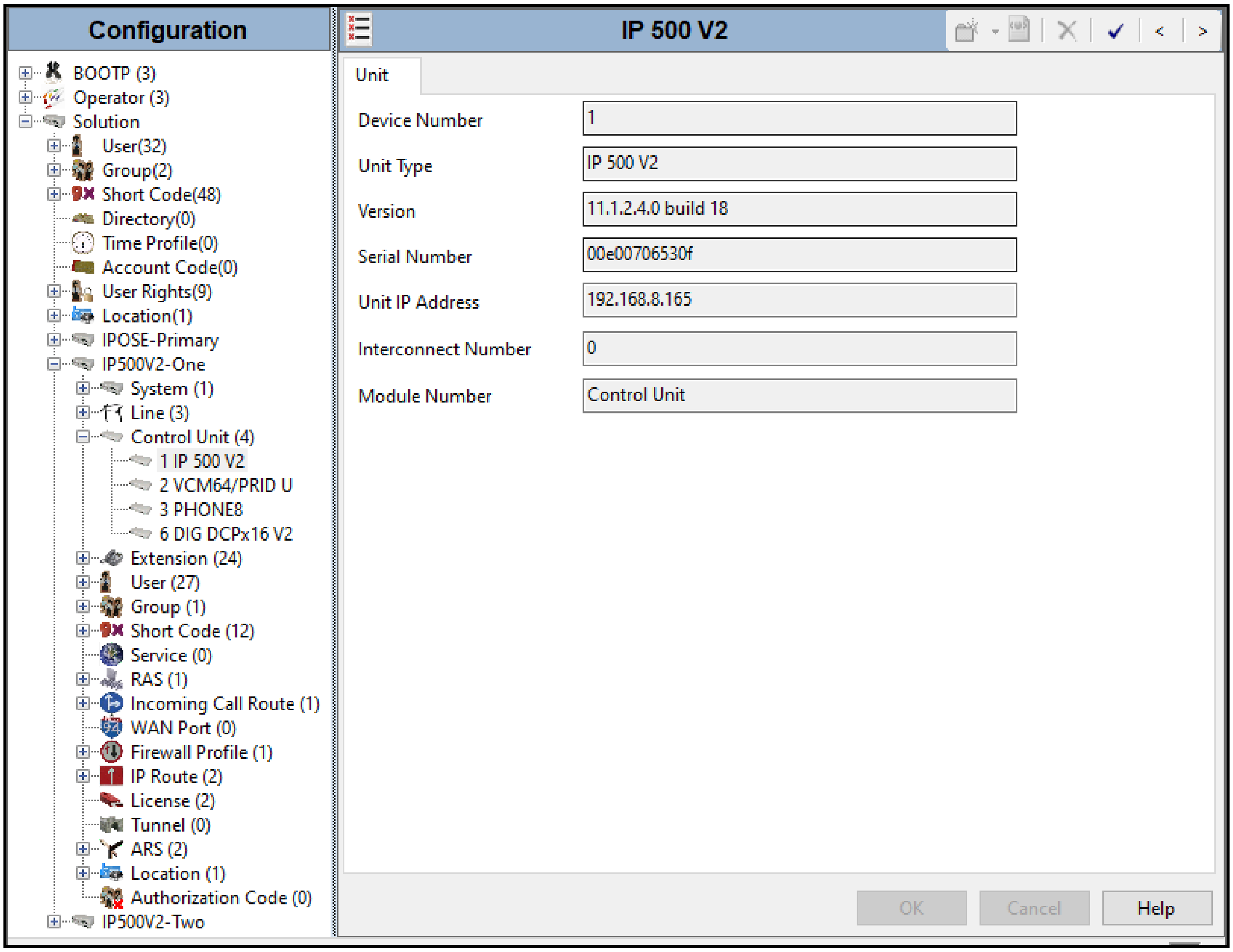
Hardware Installed
Control Unit: IP 500 V2
Internal Modules: VCM64/PRID U; PHONE8
Expansion Modules: DIG DCPx16 V2

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.

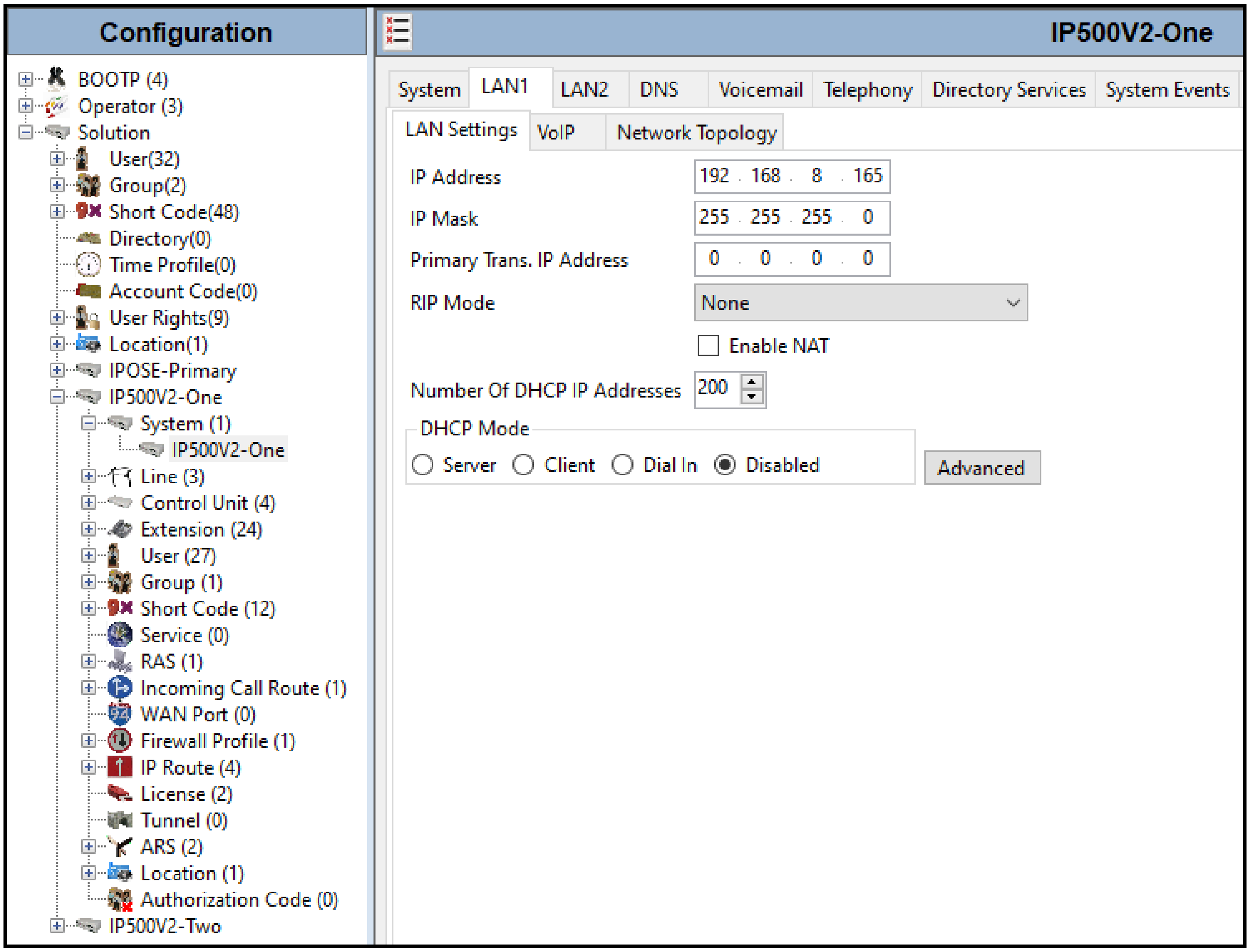


Device Number	1
Unit Type	IP 500 V2
Version	11.1.2.4.0 build 18
Serial Number	00e00706530f
Unit IP Address	192.168.8.165
Interconnect Number	0
Module Number	Control Unit

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).



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

Number Of DHCP IP Addresses: 200

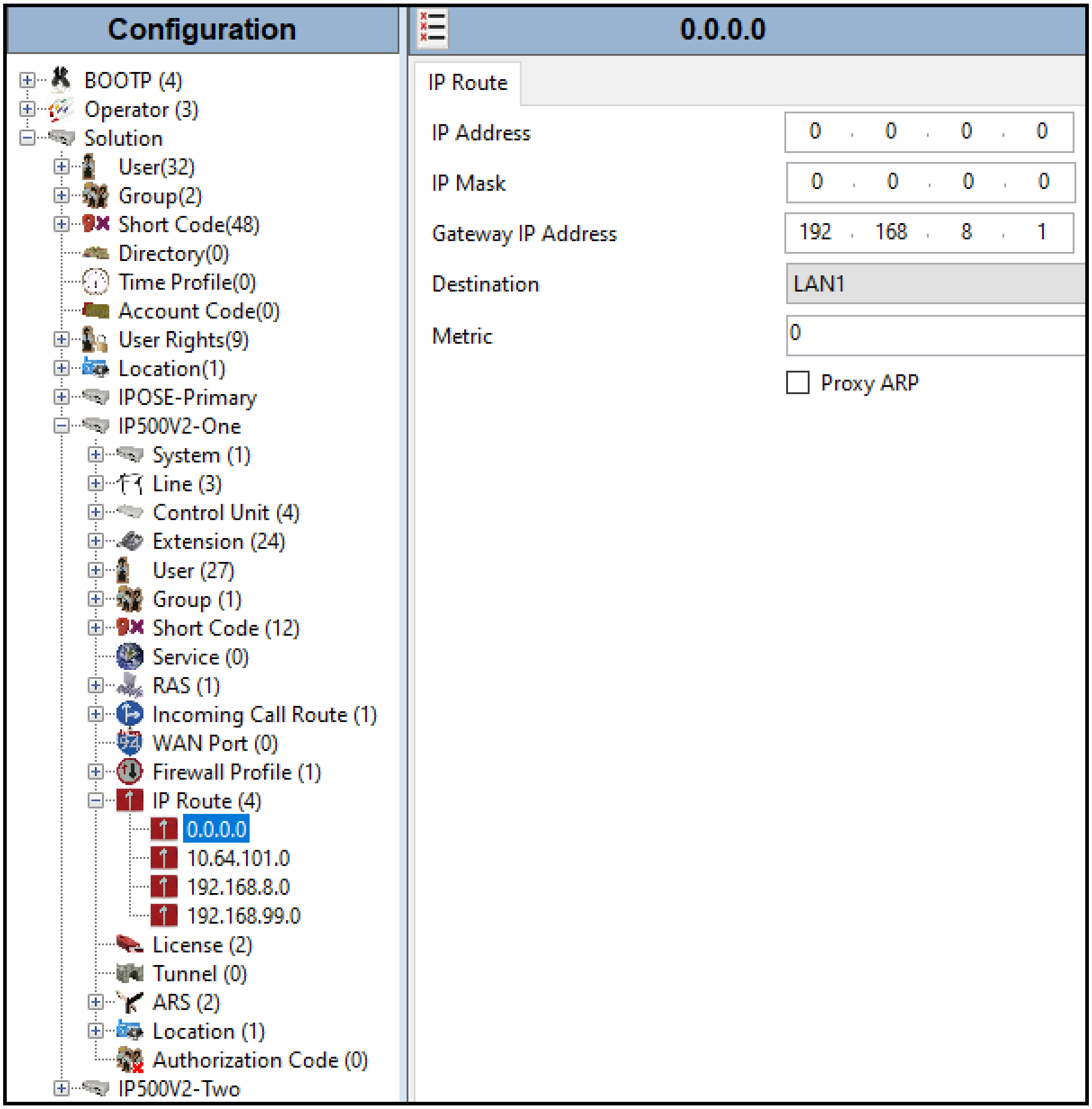
DHCP Mode: ○ Server ○ Client ○ Dial In ● Disabled

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.



IP Address: 0 . 0 . 0 . 0

IP Mask: 0 . 0 . 0 . 0

Gateway IP Address: 192 . 168 . 8 . 1

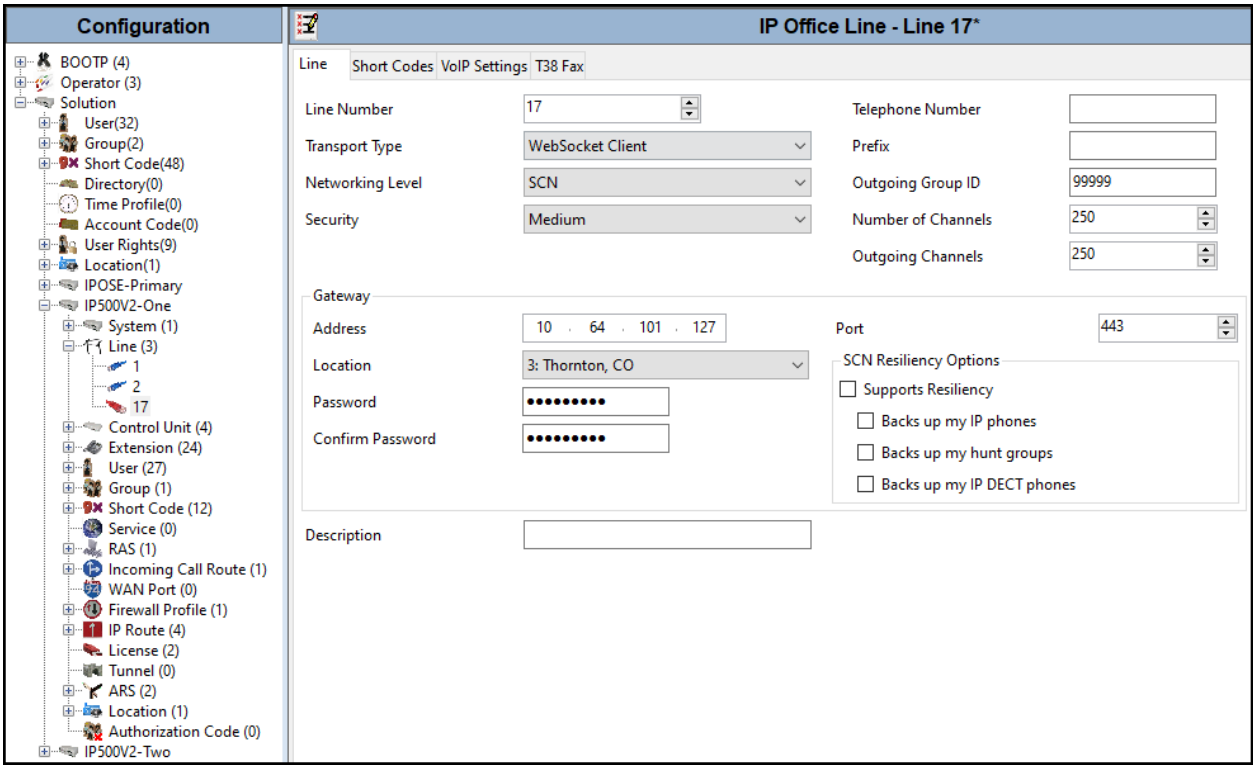
Destination: LAN1

Metric: 0

☐ 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.



IP Office Line - Line 17*

Line Number	17	Telephone Number	
Transport Type	WebSocket Client	Prefix	
Networking Level	SCN	Outgoing Group ID	99999
Security	Medium	Number of Channels	250
		Outgoing Channels	250

Gateway

Address	10 . 64 . 101 . 127	Port	443
Location	3: Thornton, CO		
Password	•••••••••		
Confirm Password	•••••••••		

SCN Resiliency Options

☐ Supports Resiliency

☐ Backs up my IP phones

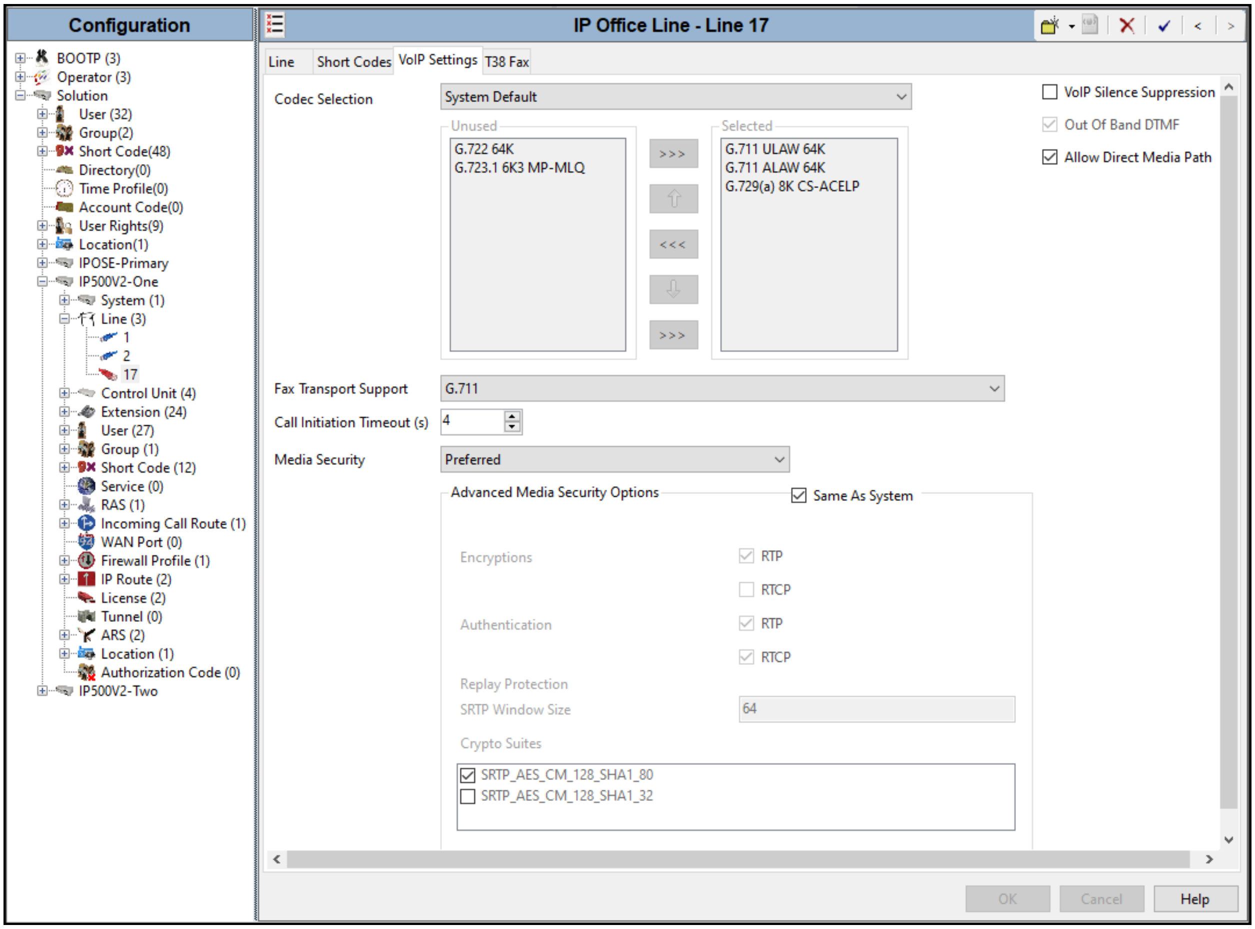
☐ Backs up my hunt groups

☐ Backs up my IP DECT phones

Description

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 2.2**).
- Under **Media Security Preferred** was selected.



IP Office Line - Line 17

Codec Selection: System Default

Unused: G.722 64K, G.723.1 6K3 MP-MLQ

Selected: G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP

□ VoIP Silence Suppression
☑ Out Of Band DTMF
☑ Allow Direct Media Path

Fax Transport Support: G.711

Call Initiation Timeout (s): 4

Media Security: Preferred

Advanced Media Security Options ☑ Same As System

Encryptions: ☑ RTP □ RTCP

Authentication: ☑ RTP ☑ RTCP

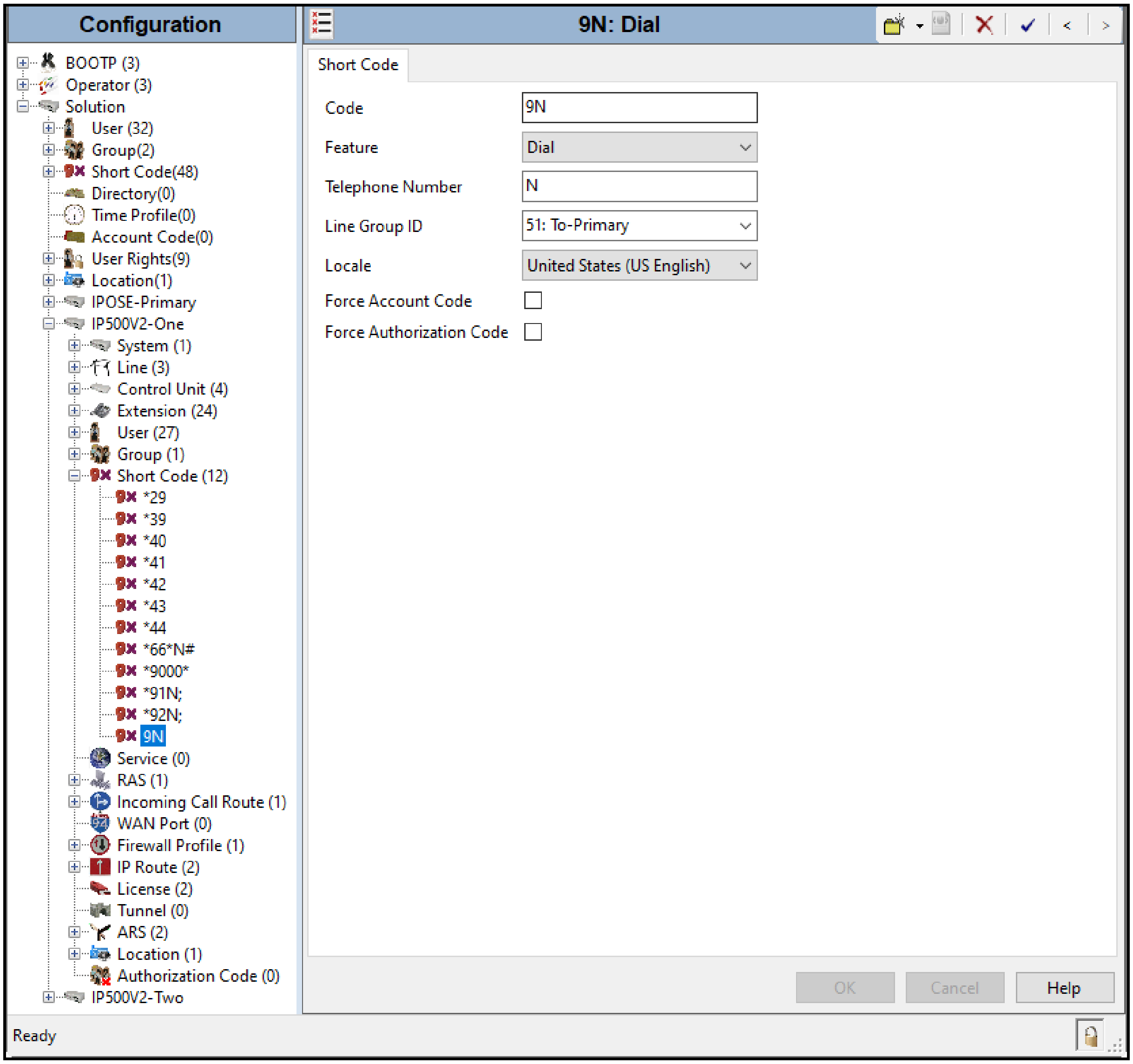
Replay Protection

SRTP Window Size: 64

Crypto Suites
☑ SRTP_AES_CM_128_SHA1_80
□ SRTP_AES_CM_128_SHA1_32

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.



9N: Dial

Short Code

Code: 9N

Feature: Dial

Telephone Number: N

Line Group ID: 51: To-Primary

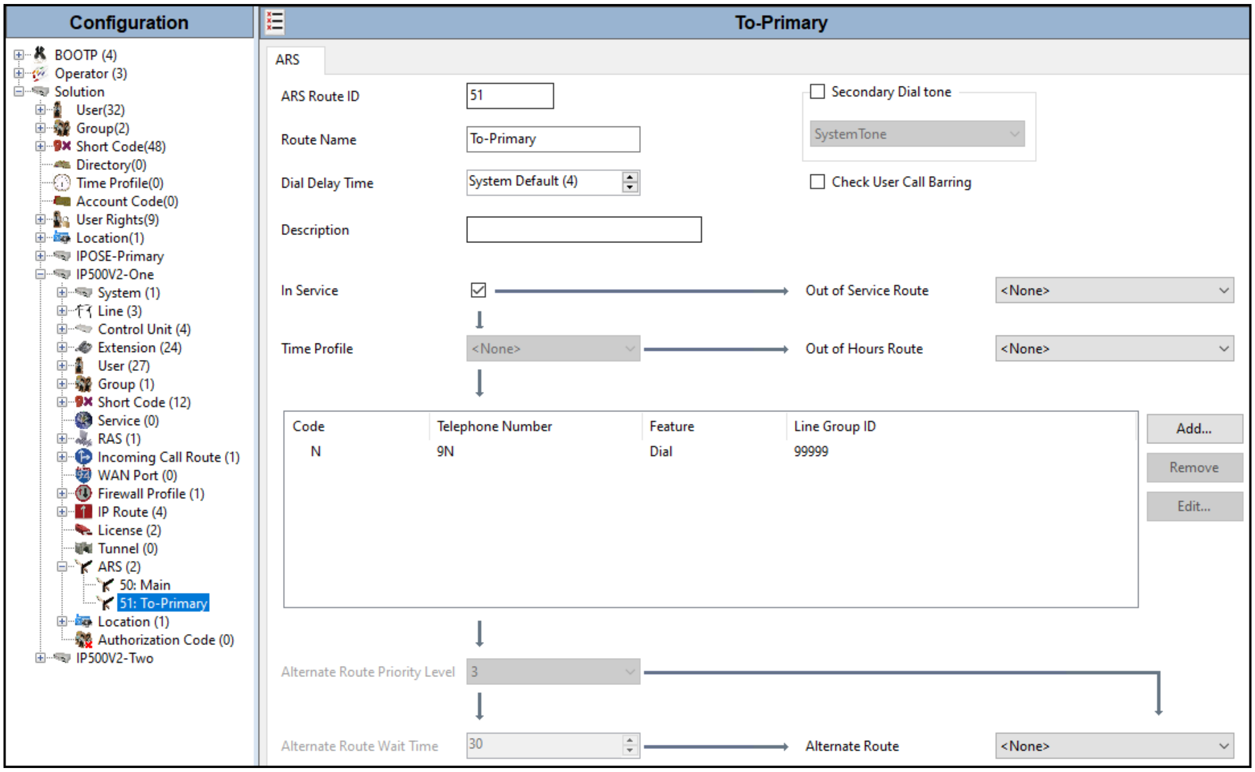
Locale: United States (US English)

Force Account Code ☐

Force Authorization Code ☐

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**).



ARS Route ID: 51

Route Name: To-Primary

Dial Delay Time: System Default (4)

Description:

□ Secondary Dial tone: SystemTone

□ Check User Call Barring

In Service: ☑

Out of Service Route: <None>

Time Profile: <None>

Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group ID
N	9N	Dial	99999

Alternate Route Priority Level: 3

Alternate Route Wait Time: 30

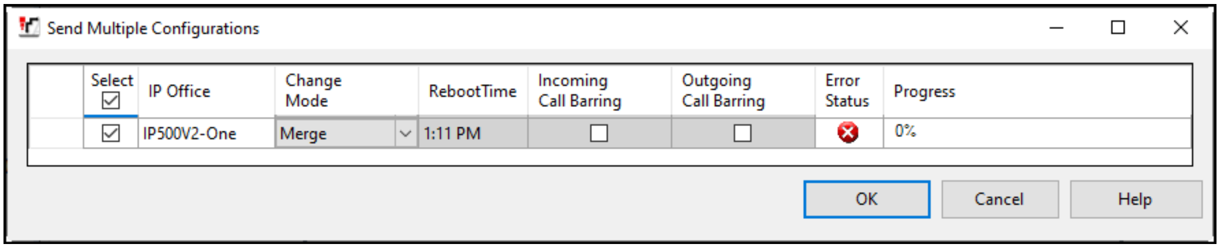
Alternate Route: <None>

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.



Select ☑	IP Office	Change Mode	RebootTime	Incoming Call Barring	Outgoing Call Barring	Error Status	Progress
☑	IP500V2-One	Merge	1:11 PM	□	□	☒	0%

7. Lumen Technologies Voice Complete SIP Trunking Service Configuration

To use the Lumen Technologies Voice Complete SIP Trunking Service, a customer must request the service from Lumen Technologies using the established sales processes. The process can be started by contacting Lumen Technologies via the corporate web site at: https://www.lumen.com/en-us/collaboration/voice-complete.html and requesting information.

During the signup process, Lumen Technologies and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to Lumen Technologies Voice Complete network.

Lumen Technologies 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**.

Lumen Technologies will provide the customer the necessary information to configure Avaya IP Office following the steps discussed in the previous sections, including:

Lumen Technologies will provide the following information:

- Lumen Technologies SIP Proxy Signaling IP address.
- 3rd party trusted certificate authority (CA) being used by Lumen for enhanced security, to enable TLS encryption outside of the enterprise (public network side, between Lumen and the Enterprise), as described in **Section 5.9**.
- DID numbers, etc.

8. Verification Steps

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

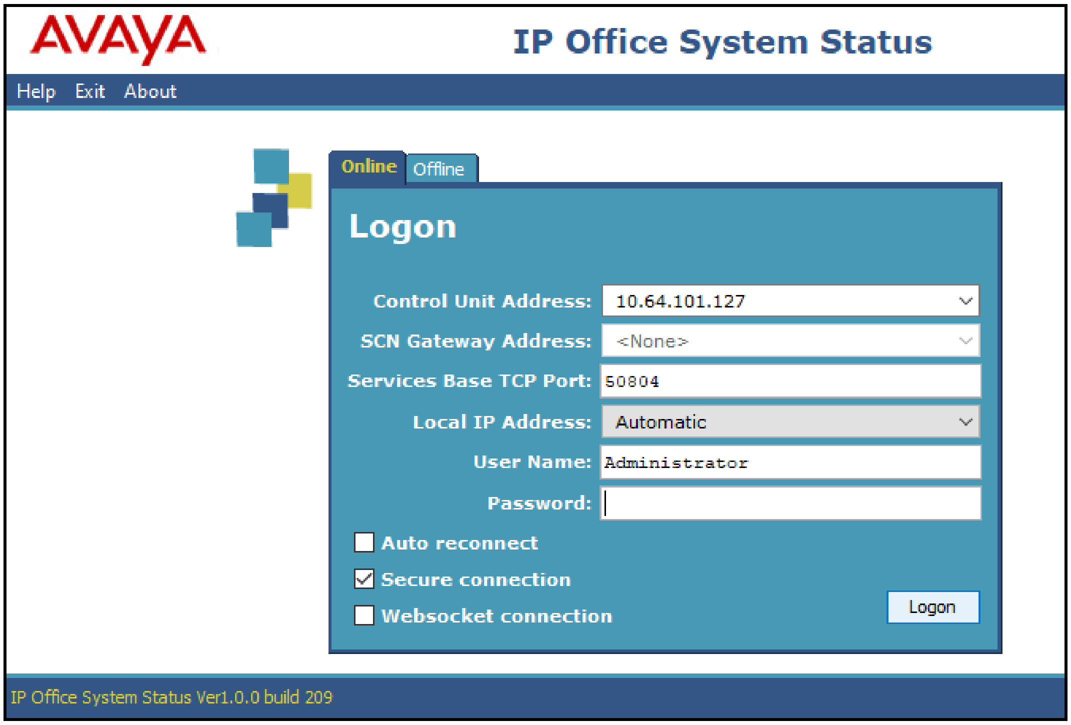
The following steps may be used to verify the configuration:

- 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.



Control Unit Address: 10.64.101.127

SCN Gateway Address: <None>

Services Base TCP Port: 50804

Local IP Address: Automatic

User Name: Administrator

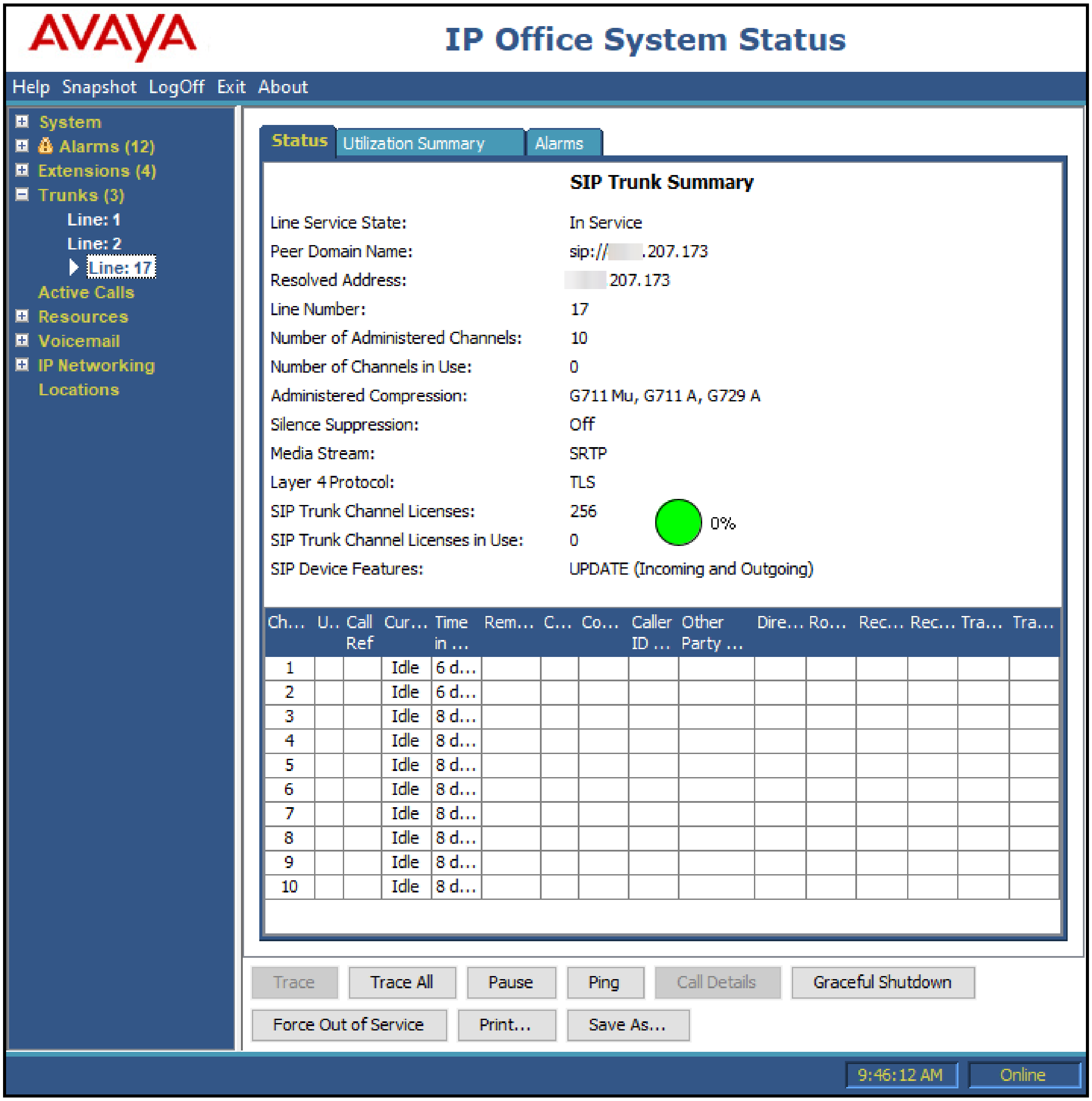
Password:

□ Auto reconnect

☑ Secure connection

□ Websocket connection

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.



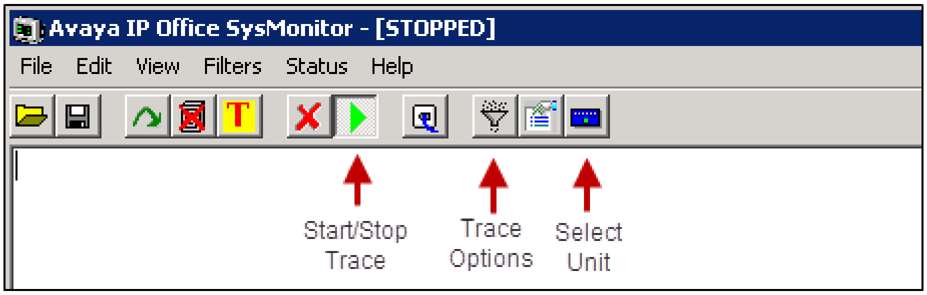
SIP Trunk Summary

Line Service State: In Service
Peer Domain Name: sip://[redacted].207.173
Resolved Address: [redacted]207.173
Line Number: 17
Number of Administered Channels: 10
Number of Channels in Use: 0
Administered Compression: G711 Mu, G711 A, G729 A
Silence Suppression: Off
Media Stream: SRTP
Layer 4 Protocol: TLS
SIP Trunk Channel Licenses: 256 ● 0%
SIP Trunk Channel Licenses in Use: 0
SIP Device Features: UPDATE (Incoming and Outgoing)

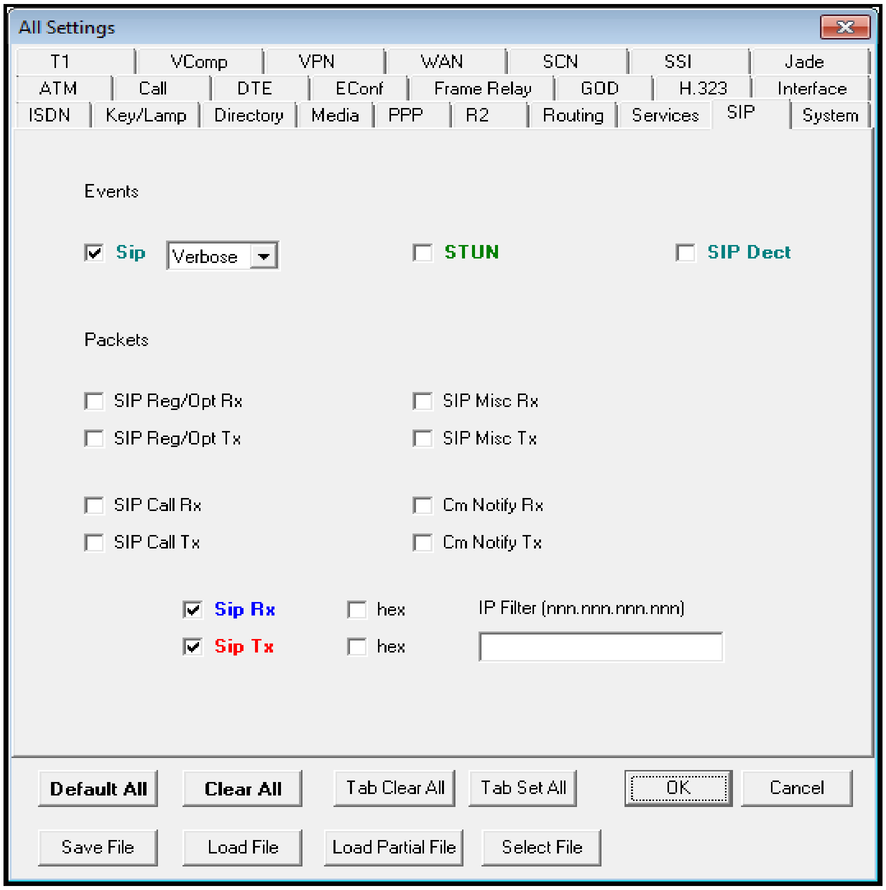
Ch...	U..	Call Ref	Cur...	Time in ...	Rem...	C...	Co...	Caller ID ...	Other Party ...	Dire...	Ro...	Rec...	Rec...	Tra...	Tra...
1			Idle	6 d...											
2			Idle	6 d...											
3			Idle	8 d...											
4			Idle	8 d...											
5			Idle	8 d...											
6			Idle	8 d...											
7			Idle	8 d...											
8			Idle	8 d...											
9			Idle	8 d...											
10			Idle	8 d...											

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.



Events

☑ Sip Verbose ☐ STUN ☐ SIP Dect

Packets

☐ SIP Reg/Opt Rx ☐ SIP Misc Rx
☐ SIP Reg/Opt Tx ☐ SIP Misc Tx

☐ SIP Call Rx ☐ Cm Notify Rx
☐ SIP Call Tx ☐ Cm Notify Tx

☑ Sip Rx ☐ hex IP Filter (nnn.nnn.nnn.nnn)
☑ Sip Tx ☐ hex

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 11.1 to support Lumen Technologies Voice Complete SIP Trunking Service. Transport Layer Security (TLS) and Secure Real-time Transport Protocol (SRTP) were used between the simulated enterprise site and the Lumen Technologies Voice Complete network (public network side) and inside of the enterprise (private network side, between Avaya servers and end-points).

The Lumen Technologies Voice Complete 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 Platform Server Edition, Release 11.1 FP2, Issue 26, January 2023.

[2] IP Office Platform 11.1.2.4, Deploying Avaya IP Office Servers as Virtual Machines, Issue 13, January 2023.

[3] Avaya IP Office Platform Server Edition Reference Configuration Release 11.1 FP2, Issue 18, January 2023.

[4] IP Office Platform 11.1 FP2, Deploying an IP500 V2 IP Office Essential Edition System, Issue 39b, March 3, 2023.

[5] Administering Avaya IP Office using Manager, Release 11.1.2.4, Issue 43, March 2023.

[6] Avaya IP Office Platform Feature Description, Release 11.1 FP2, Issue 18, January 2023.

Additional Avaya IP Office documentation can be found at:
https://ipofficekb.avaya.com/

The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.