Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office Release 11.1 to support Keyyo Communications SIP Trunking Service - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking on an enterprise solution consisting of Avaya IP Office 11.1 to support Keyyo Communications SIP Trunking Service.

The Keyyo Communications SIP Trunking Service offer referenced within these Application Notes provides customers with PSTN access via a SIP trunk between the enterprise and the service provider network. The service provides local and/or long distance PSTN calling via standards-based SIP trunks directly as an alternative to legacy analog or digital trunks. The Keyyo Communications SIP Trunking Service provides a cost effective and flexible way to connect your business to the outside world. It helps your business use the internet bandwidth you already pay for in a more flexible way.

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

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

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service between the Keyyo Communications SIP Trunking Service offering and a simulated Avaya enterprise solution. User Datagram Protocol (UDP) transport was used to connect the simulated enterprise solution to the Keyyo Communications SIP Trunking Service offering (public 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 Keyyo Communications 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", "Keyyo" or "Keyyo Communications" 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 Keyyo's network via the public Internet, as depicted in **Figure 1**, and exercise the features and functionalities listed in **Section 2.1**.

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

For the testing associated with this Application Note, the interface between the simulated enterprise site (private network) and the Keyyo network (public network) did not include the use of any specific encryption features, UDP/RTP was used.

Encryption (TLS/sRTP) was used internal to the enterprise between Avaya products wherever possible.

2.1. Interoperability Compliance Testing

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

- Public DNS record queries to establish the SIP trunk connections.
- 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 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 network.
- Incoming and outgoing PSTN calls to/from Avaya Workplace Client for Windows (SIP).
- Dialing plans including local calls, outbound toll-free, etc.
- Caller ID presentation.
- Privacy (blocking calling party number).
- 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.711ALAW, G.729A and G722 64K, Keyyo preferred codec order.
- Proper response to no matching codecs.
- Proper early media transmissions.
- DTMF tone support using RFC 2833.
- 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.
- Simultaneous active calls.
- Long duration calls (over one hour).
- Proper response/error treatment to all trunks busy.
- Proper response/error treatment when disabling SIP connection.

Items that were not tested includes the following:

- 0, 0+10 digits, 411 and 911 (Emergency) calls were not tested.

2.2. Test Results

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

- **SIP OPTIONS Messages** – During the compliance test Keyyo did not send SIP OPTIONS messages to IP Office, IP Office did send SIP OPTIONS messages to Keyyo, this was sufficient to keep the SIP trunk up in service.
- **Fax support** – Fax calls using the T.38 protocol failed during the compliance test. G.711 pass-through fax was also tested, but it behaved unreliably. The issue related to G.711 pass-through fax failing during the compliance test may be related to the unpredictability of G.711 pass-through techniques, which only works well on networks with very few hops and with limited end-to-end delay. The issue related to T.38 fax calls failing is related to the method of codec re-negotiation involving T.38 is not supported by Keyyo.

2.3. Support

For technical support on Keyyo products please visit
https://www.keyyo.com/fr/support/contactsupport-partenaire/

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 compliance testing. The test configuration simulates an enterprise site with an Avaya SIP-enabled enterprise solution connected to the Keyyo Communications 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 V2s are equipped with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module). The LAN1 port of the Avaya IP Office IP500 V2 expansion systems was connected to the enterprise LAN, the LAN2 port was 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 protocols on the SIP trunk between IP Office and the Keyyo network, across the public Internet, is UDP for signaling and RTP for media. The transport protocol between Avaya components inside the enterprise private IP network (LAN) is TLS for signaling and SRTP for media.

For the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to the Keyyo network. The short code 9 was stripped off by Avaya IP Office, but the remaining N digits were sent unaltered to the Keyyo 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, domain names, and routable DID numbers used during the compliance testing have been masked.

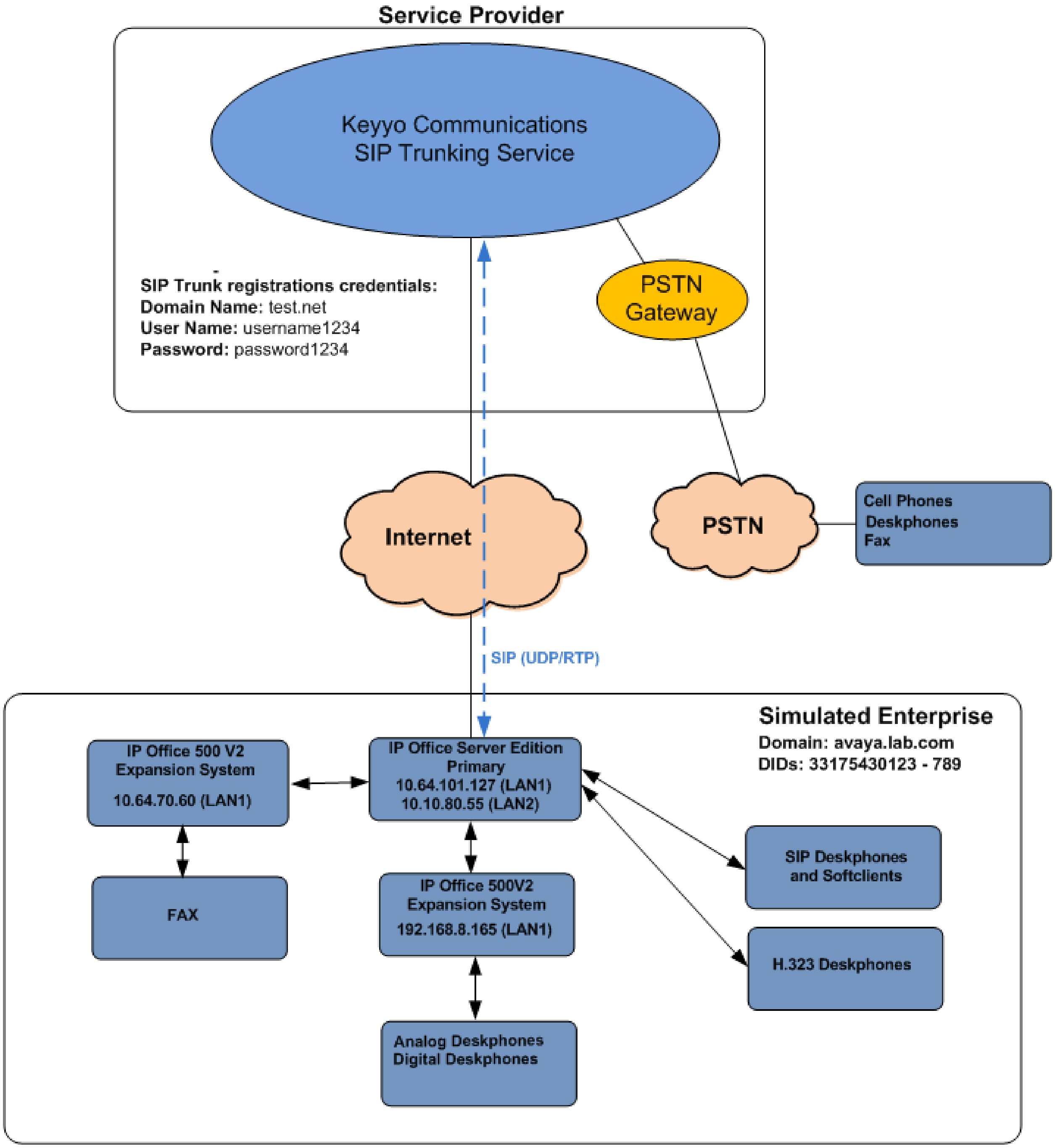


Figure 1: Avaya simulated enterprise site connected to the Keyyo Communications SIP Trunking Service offering

4. Equipment and Software Validated

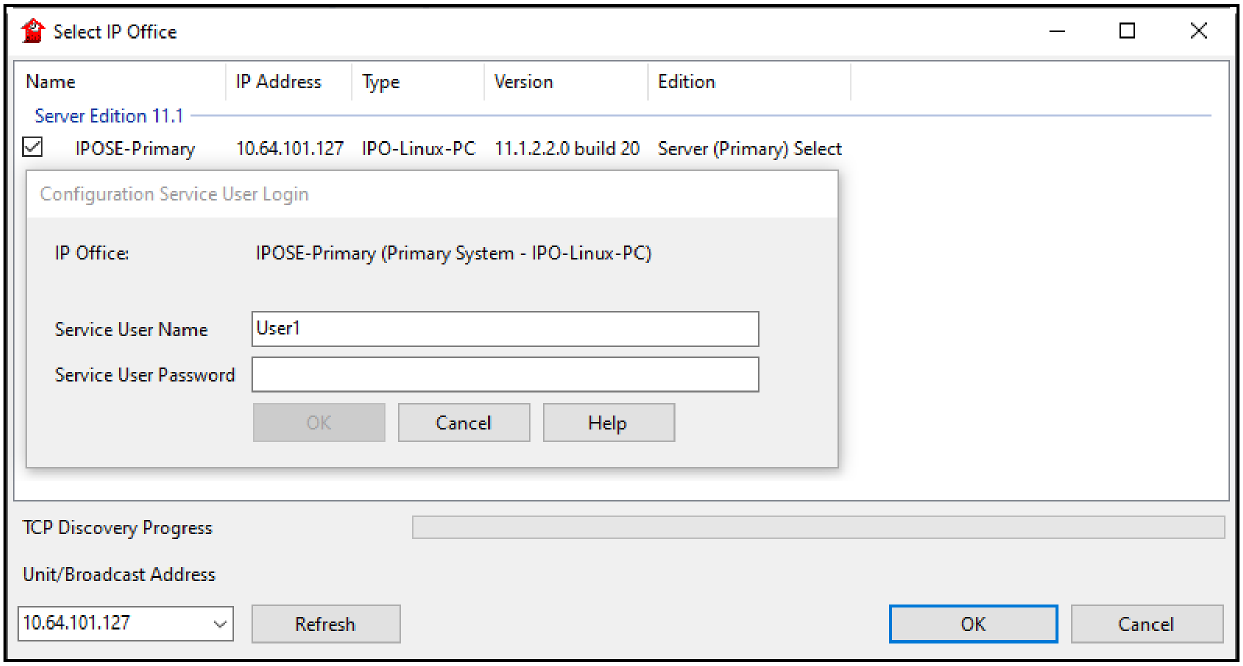
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office Server Edition (Primary Server) • Avaya IP Office Voicemail Pro	11.1.2.2.0 Build 20 11.1.2.2.0 Build 8
Avaya IP Office IP500 V2 (Expansion Systems)	11.1.2.2.0 Build 20
Avaya IP Office Manager	11.1.2.2.0 Build 20
Avaya J179 IP Telephone (H.323)	6.8304
Avaya 1140E IP Deskphones (SIP)	SIP1140e Ver. 04.04.23.00
Avaya J129 IP Deskphones (SIP)	4.0.7.0.7
Avaya 1408 Digital Telephone	48.02
Avaya Workplace Client for Windows (SIP)	3.24.0.84
Analog Telephone	---
Keyyo Communications	
SIP Platform	Proprietary

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.



Name	IP Address	Type	Version	Edition
Server Edition 11.1				
☑ IPOSE-Primary	10.64.101.127	IPO-Linux-PC	11.1.2.2.0 build 20	Server (Primary) Select

Configuration Service User Login

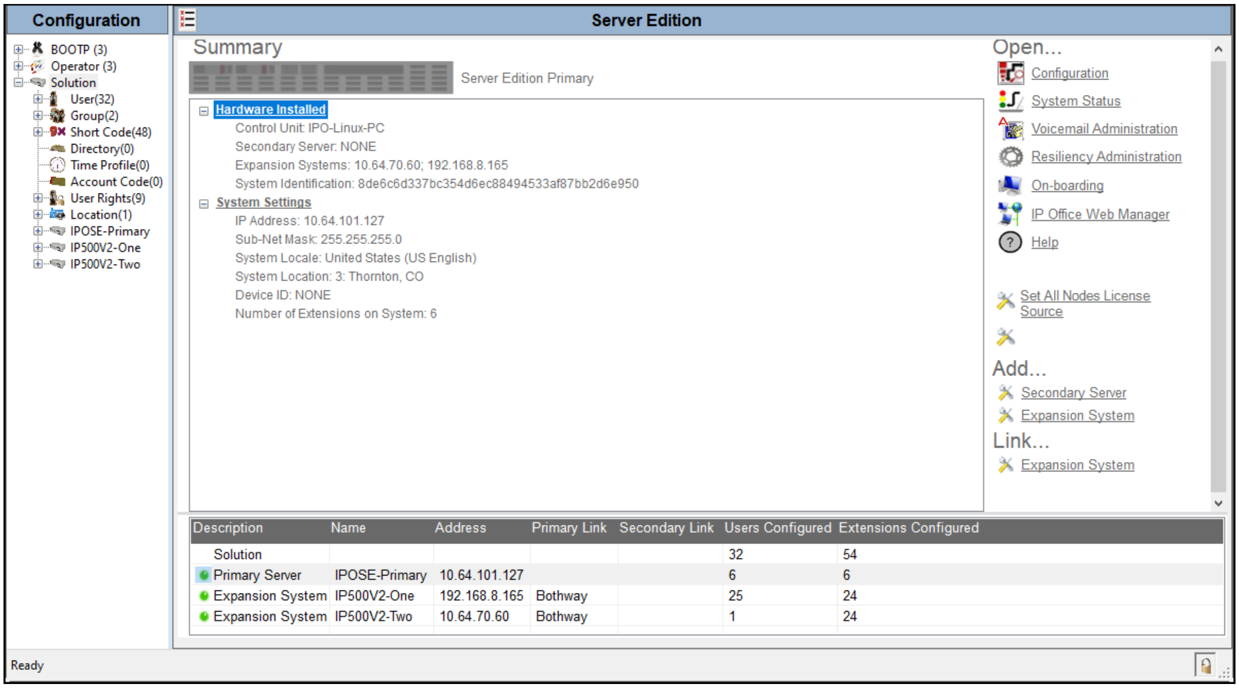
IP Office: IPOSE-Primary (Primary System - IPO-Linux-PC)

Service User Name: User1

Service User Password:

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.



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.

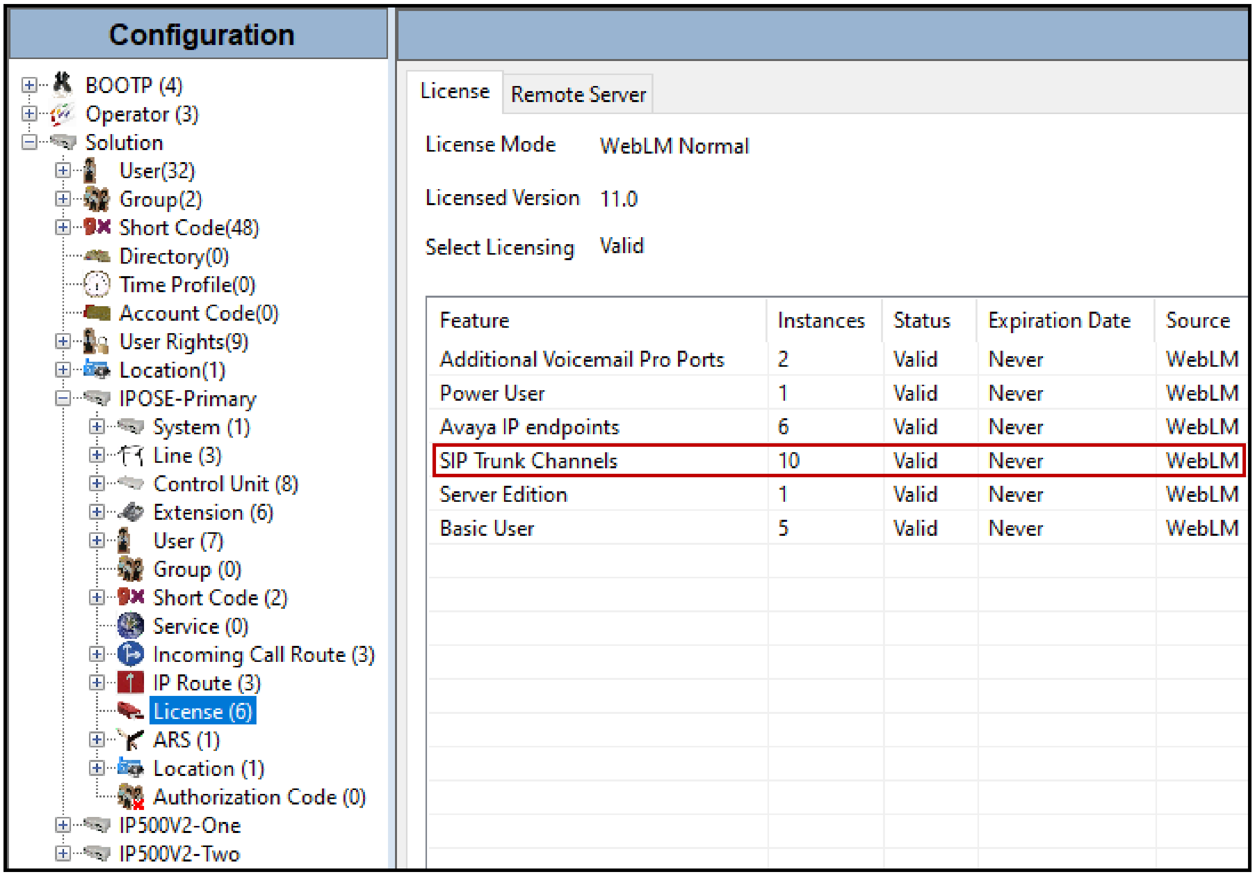
Standard feature configurations that are not directly related to the interfacing with the service provider are assumed to be already in place, and they are not part of these Application Notes.

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 and **IP500V2-One** and **IP500V2-Two** were used as the system names of the 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.



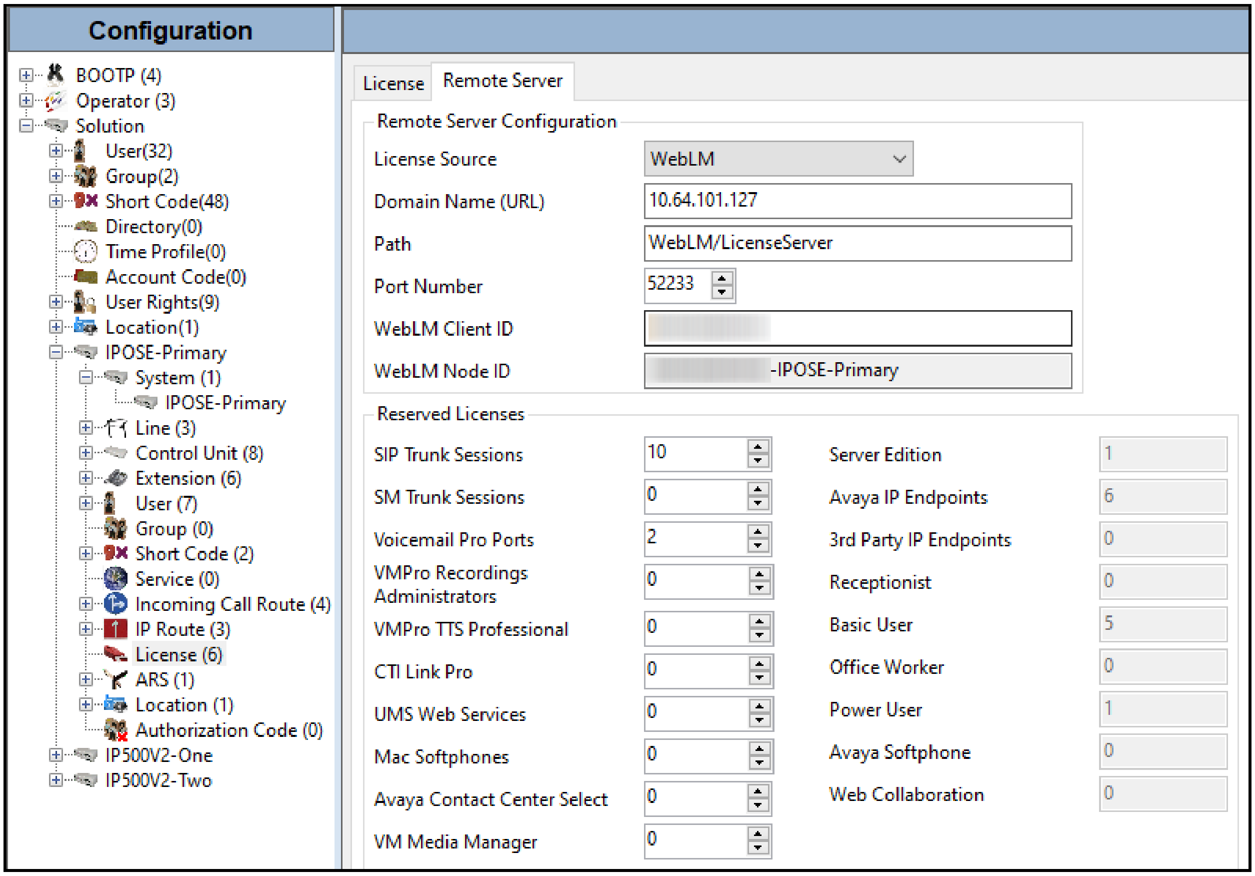
License Mode WebLM Normal

Licensed Version 11.0

Select Licensing Valid

Feature	Instances	Status	Expiration Date	Source
Additional Voicemail Pro Ports	2	Valid	Never	WebLM
Power User	1	Valid	Never	WebLM
Avaya IP endpoints	6	Valid	Never	WebLM
SIP Trunk Channels	10	Valid	Never	WebLM
Server Edition	1	Valid	Never	WebLM
Basic User	5	Valid	Never	WebLM

On Server Edition systems, the numbers of licenses to be assigned to the specific Server or Expansion Systems are reserved from the total pool of licenses present on the license server. On the screen below, **10 SIP Trunk Sessions** licenses were reserved to be used by the Primary Server.



Remote Server Configuration

License Source	WebLM
Domain Name (URL)	10.64.101.127
Path	WebLM/LicenseServer
Port Number	52233
WebLM Client ID	[redacted]
WebLM Node ID	[redacted]-IPOSE-Primary

Reserved Licenses

SIP Trunk Sessions	10	Server Edition	1
SM Trunk Sessions	0	Avaya IP Endpoints	6
Voicemail Pro Ports	2	3rd Party IP Endpoints	0
VMPro Recordings Administrators	0	Receptionist	0
VMPro TTS Professional	0	Basic User	5
CTI Link Pro	0	Office Worker	0
UMS Web Services	0	Power User	1
Mac Softphones	0	Avaya Softphone	0
Avaya Contact Center Select	0	Web Collaboration	0
VM Media Manager	0		

5.2. System Settings

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

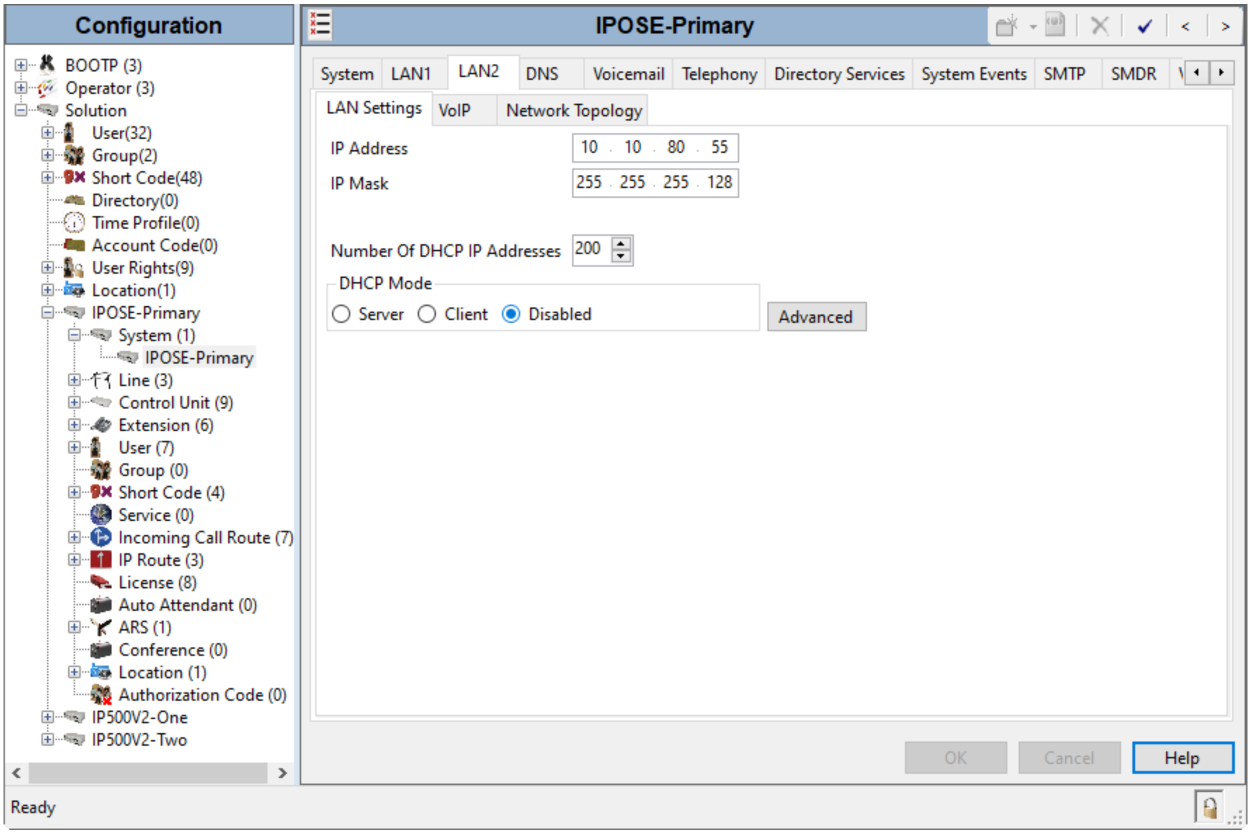
5.2.1. System – LAN2 Tab

In the sample configuration, the LAN2 interface is used for the SIP trunk connection to the Keyyo network.

5.2.1.1 LAN2 – LAN Settings Tab

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

- **IP Address: 10.10.80.55** was used in the reference configuration, this is the public IP address assigned to IP Office.
- **IP Mask: 255.255.255.128** was used in the reference configuration.
- Other parameters on this screen are set to the defaults.



IP Address: 10 . 10 . 80 . 55

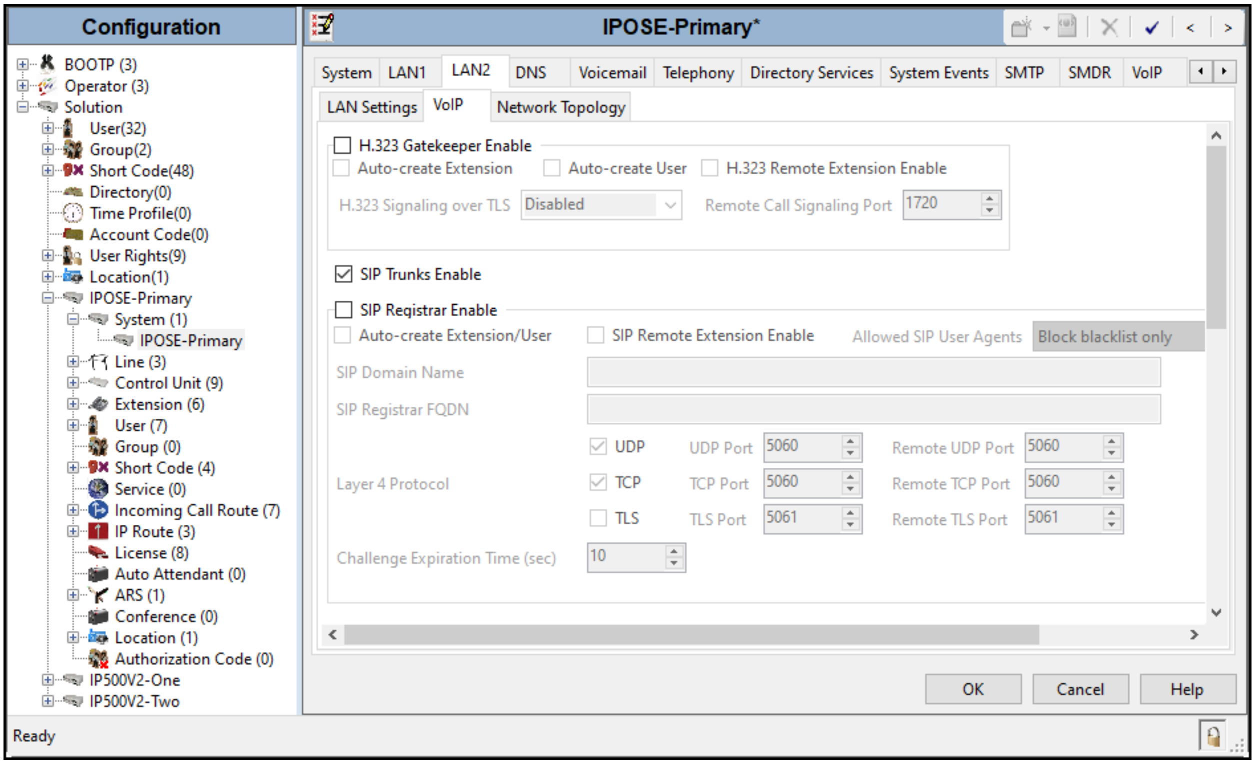
IP Mask: 255 . 255 . 255 . 128

Number Of DHCP IP Addresses: 200

DHCP Mode: ○ Server ○ Client ● Disabled

5.2.1.2 LAN2 – VoIP Tab

- 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, 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:

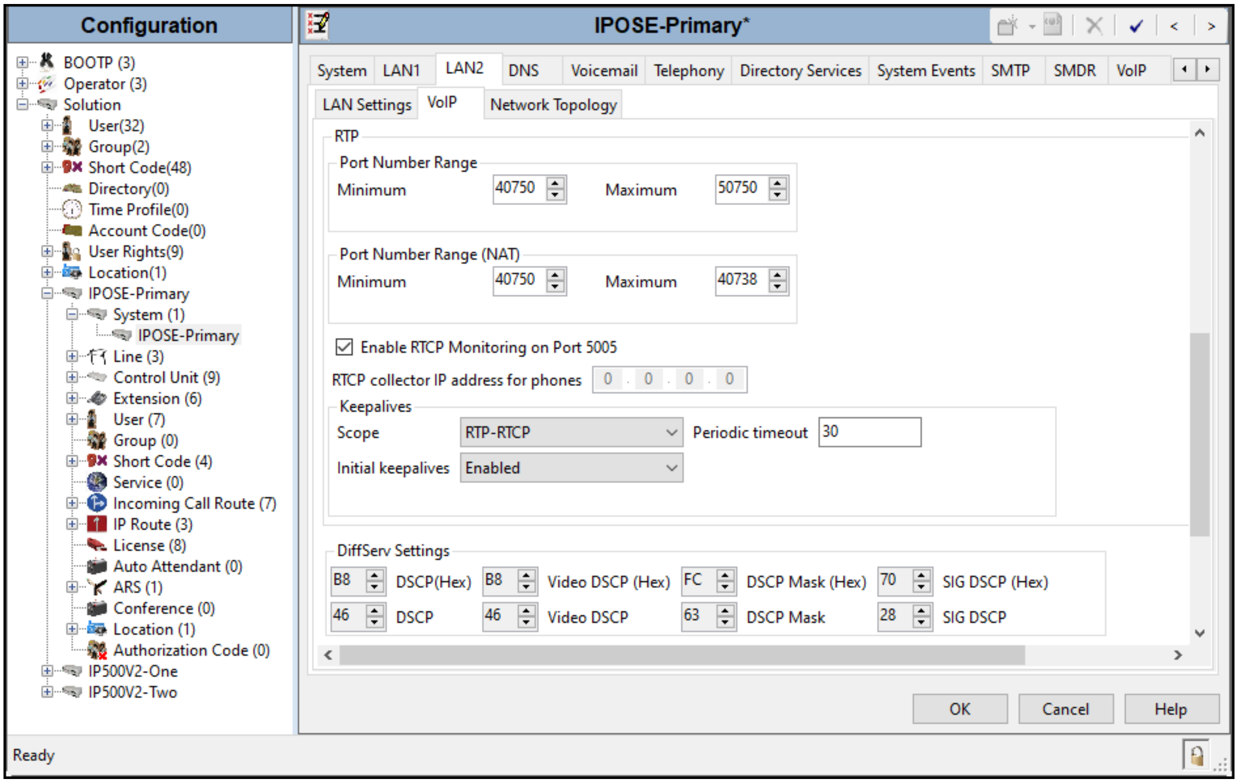
SIP Registrar FQDN:

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

Scroll down the page:

- Verify the **RTP Port Number Range**. Based on this setting, Avaya IP Office will request RTP media to be sent to a UDP port 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 40738

☑ 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

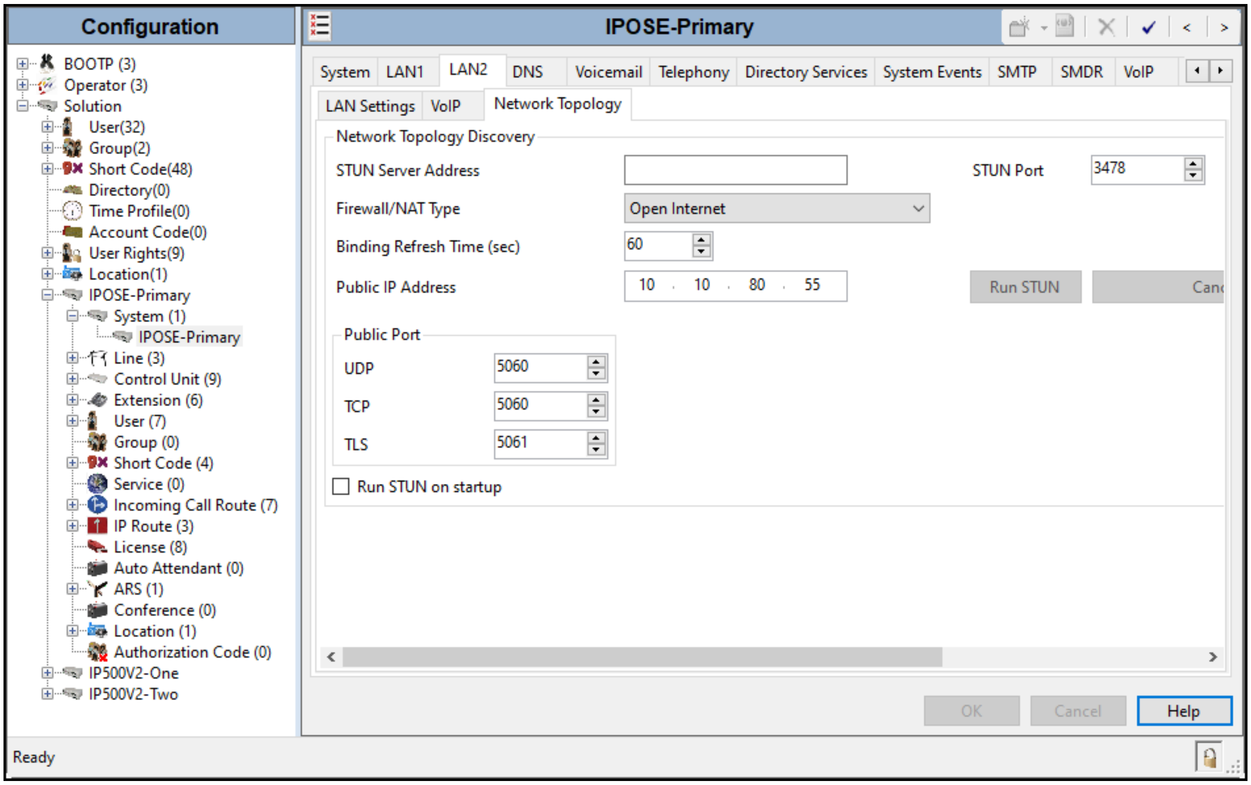
DiffServ Settings
B8 DSCP(Hex) B8 Video DSCP (Hex) FC DSCP Mask (Hex) 70 SIG DSCP (Hex)
46 DSCP 46 Video DSCP 63 DSCP Mask 28 SIG DSCP

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 Keyyo Communications SIP Trunking Service, and therefore is not described in these Application Notes.

5.2.1.3 LAN2 – Network Topology Tab

On the **LAN2 Network Topology** tab in the Details pane, set the following:

- Select the **Firewall/NAT Type** from the pull-down menu to **Open Internet**. With this configuration, the **STUN Server IP Address** and **STUN Port** are not used.
- Set **Binding Refresh Time (seconds)** to **60**. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the same public IP address assigned to the Avaya IP Office **LAN2**.
- Set **Public Port / UDP** to **5060**.
- Default values were used for all other parameters.
- Click the **OK** button.



STUN Server Address:

STUN Port: 3478

Firewall/NAT Type: Open Internet

Binding Refresh Time (sec): 60

Public IP Address: 10 . 10 . 80 . 55

Public Port

UDP: 5060

TCP: 5060

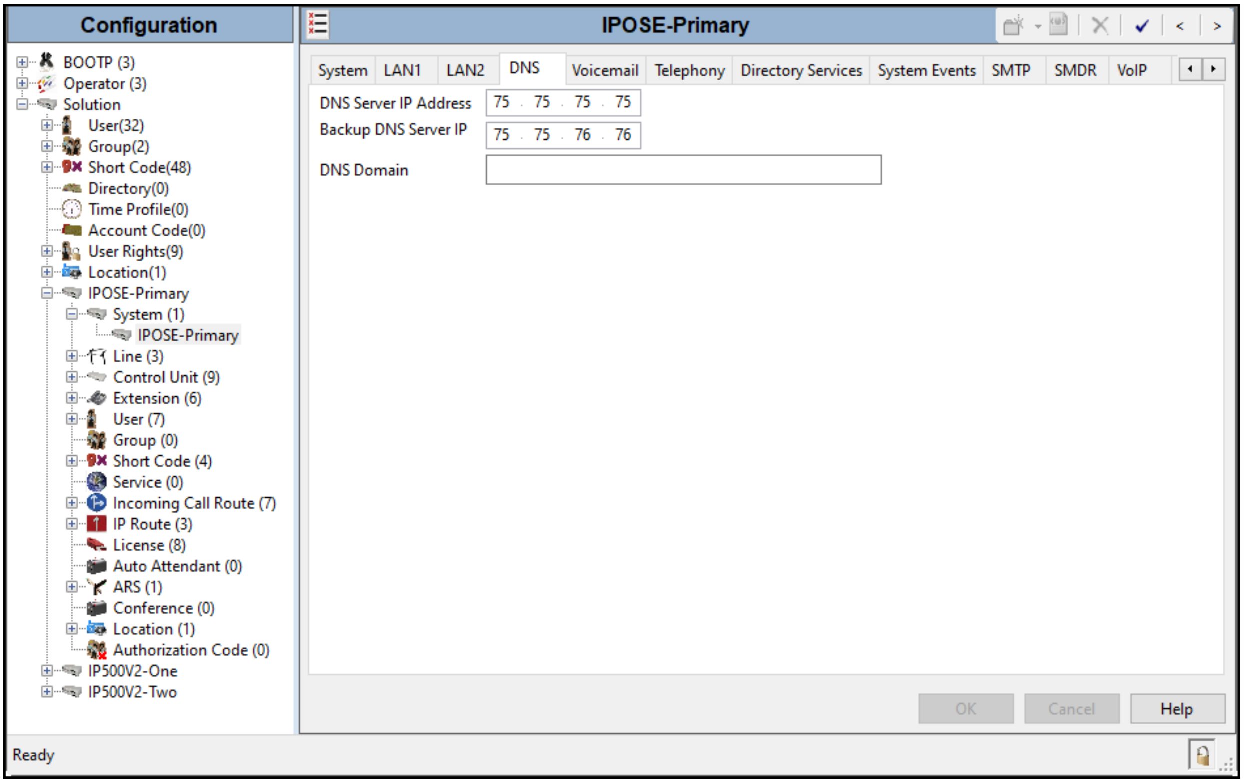
TLS: 5061

☐ Run STUN on startup

5.2.2. System – DNS Tab

Public DNS servers IP addresses are required to be configured; IP Office will retrieve the Keyyo Proxy IP Address via public DNS queries using the ISTP Domain Name configured under in **Section 5.4.2**. To access the System DNS settings, navigate to the **DNS** tab in the **Details** pane, configure the following parameters:

- Under DNS Server IP Address and Backup DNS Server IP Address enter the primary and backup public DNS servers IP addresses. These IP addresses should be provided by Keyyo.
- Click **OK** to commit.

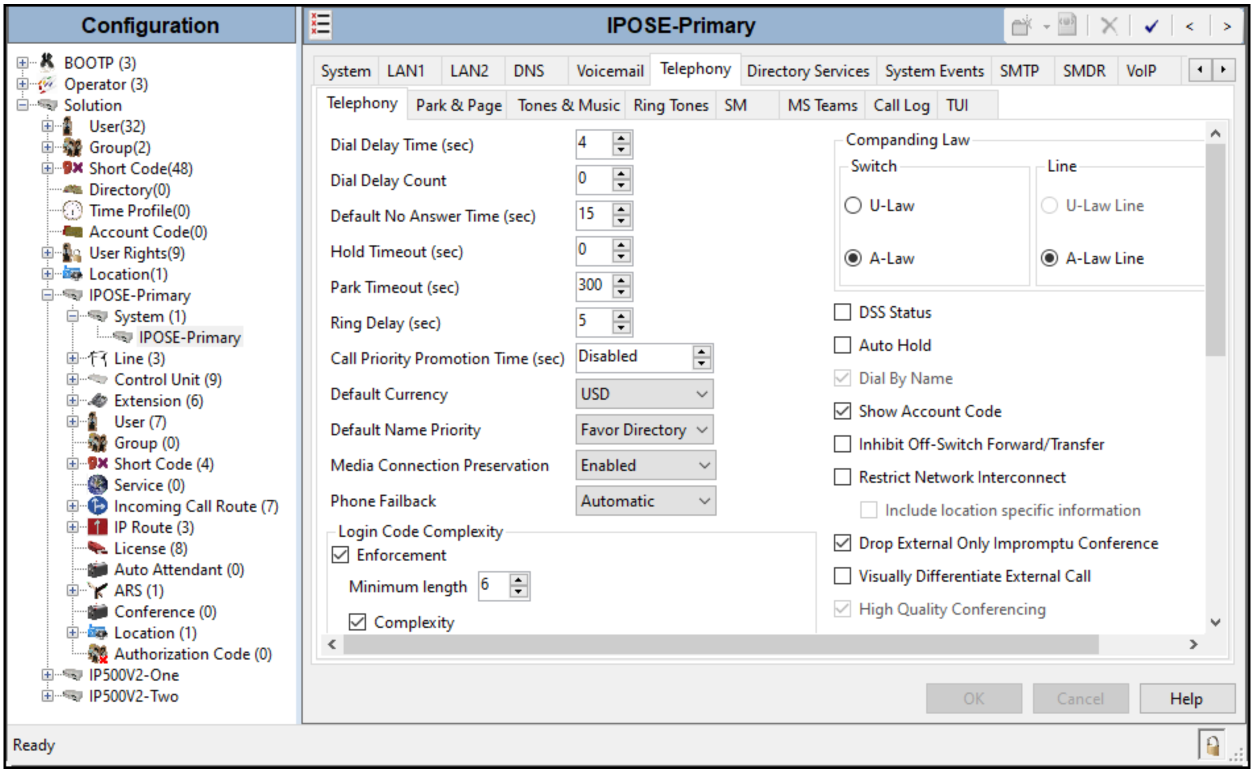


DNS Server IP Address 75 . 75 . 75 . 75
Backup DNS Server IP 75 . 75 . 76 . 76
DNS Domain

5.2.3. 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; **A-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

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

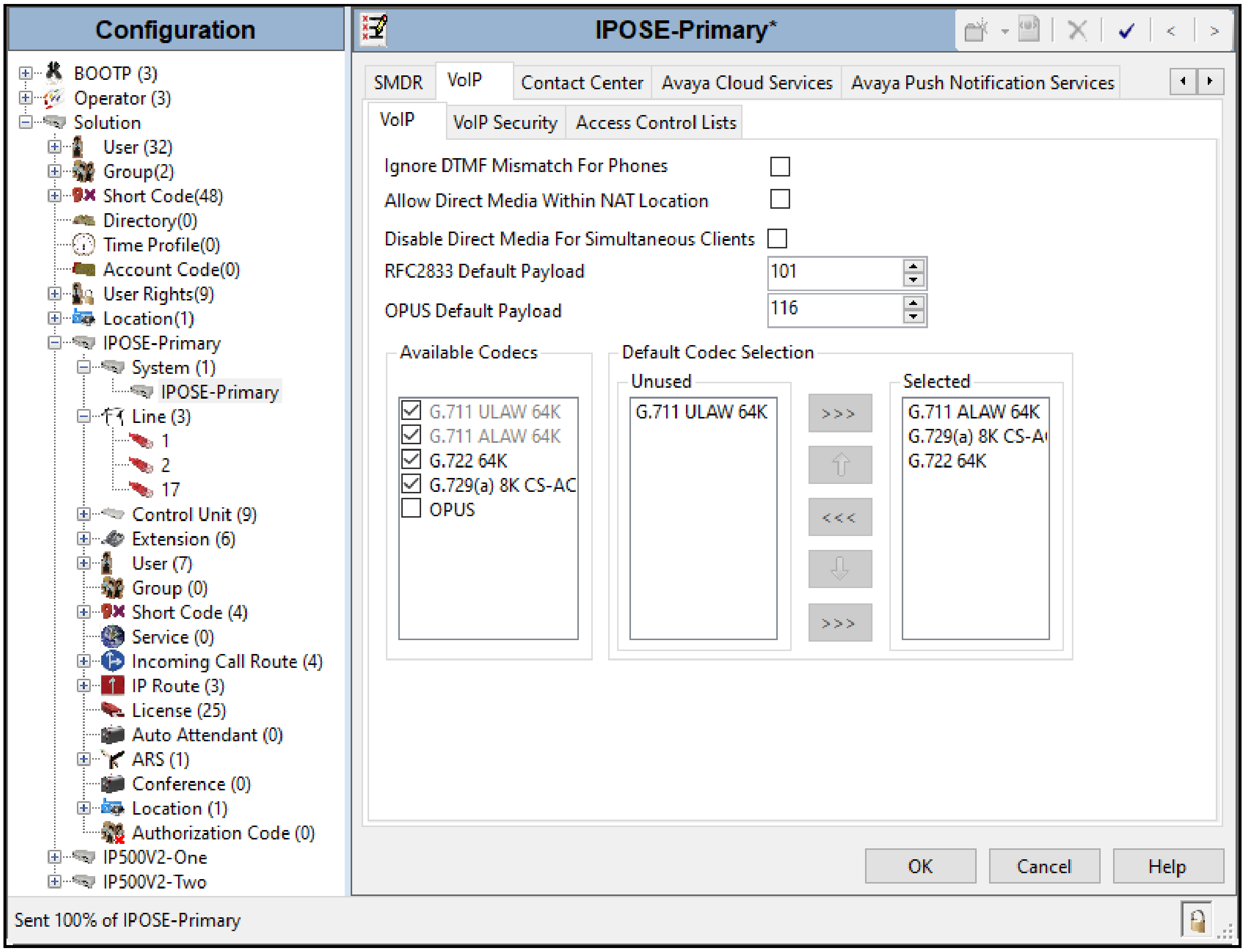
5.2.4. System – VoIP Tab

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

5.2.4.1 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.



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.711 ULAW 64K

Selected
G.711 ALAW 64K
G.729(a) 8K CS-A
G.722 64K

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).

5.2.4.2 VoIP – VoIP Security Tab

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

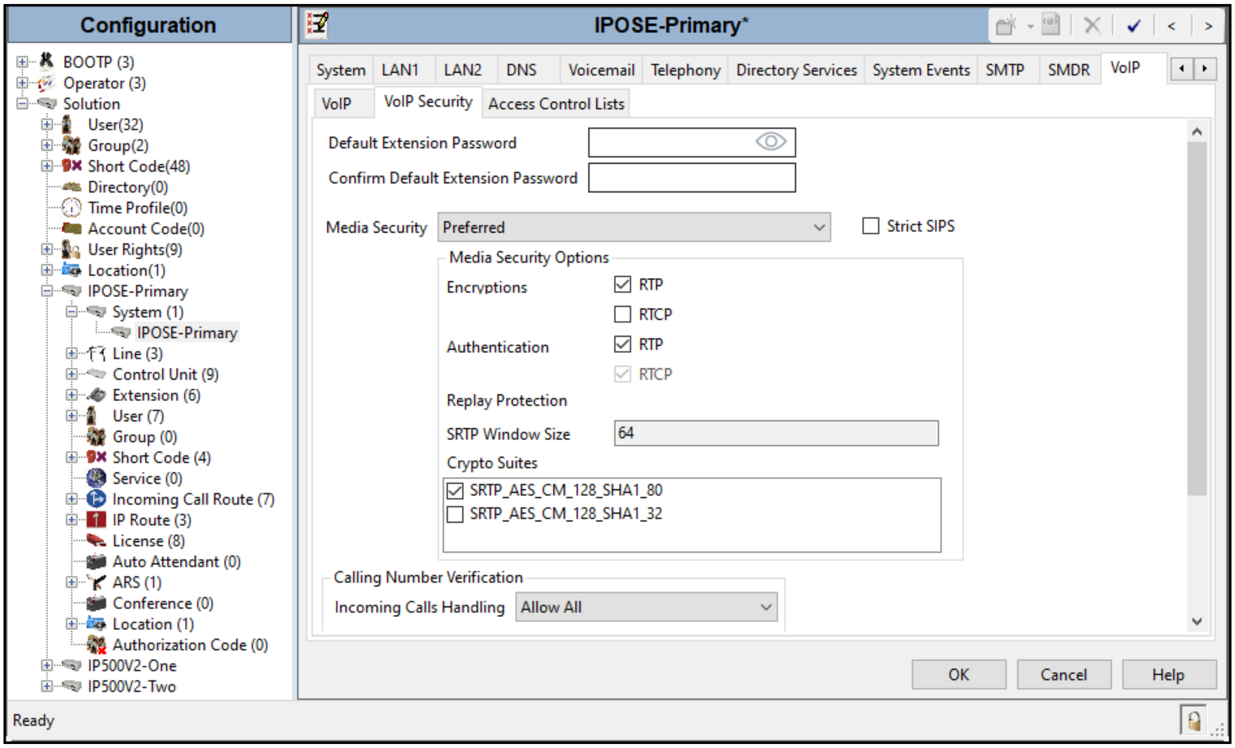
Configuring the use of SRTP at the system level is done on the **VoIP Security** tab using the Media Security setting. The options are:

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

When enabling SRTP on the system, the recommended setting is **Preferred**. In this scenario, IP Office uses SRTP if supported by the far-end, otherwise uses RTP. If the **Enforced** setting is used, and SRTP is not supported by the far-end, the call is not established.

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

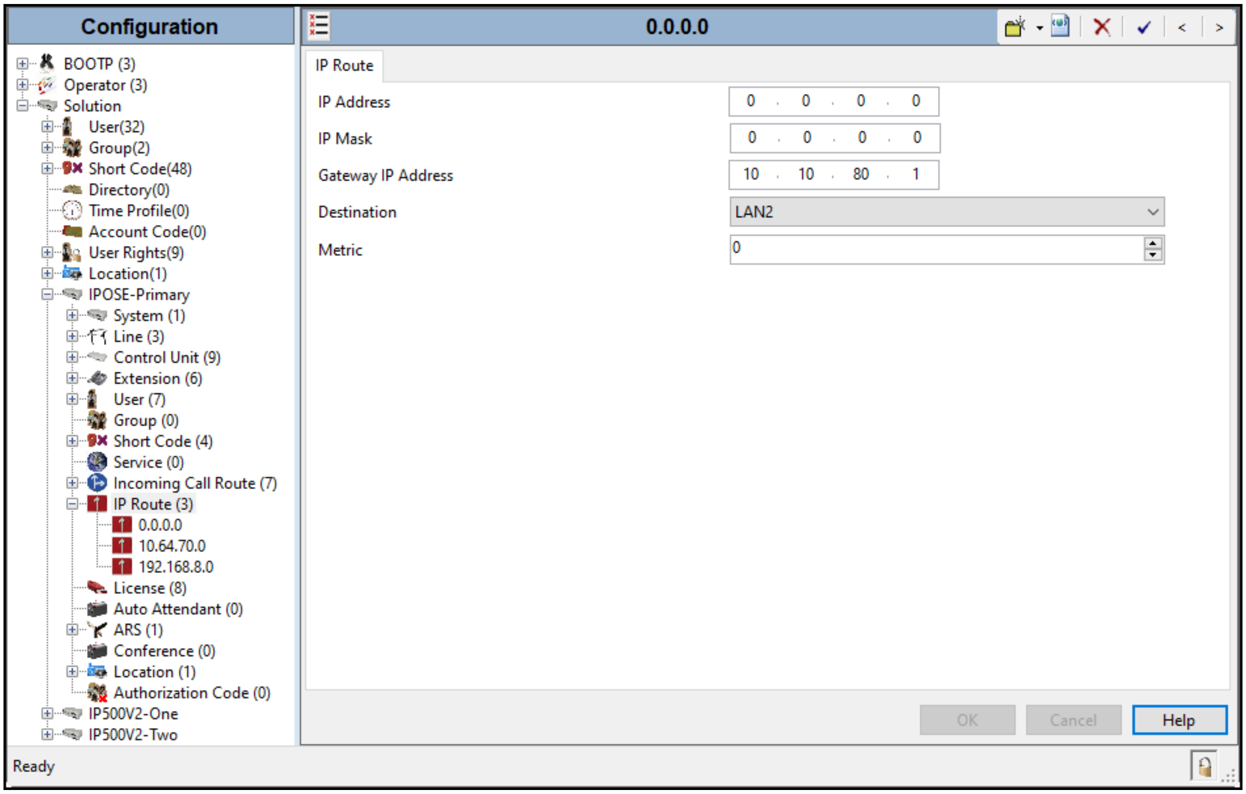
Incoming Calls Handling: Allow All

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 the Keyyo 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 the Keyyo network. 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
- Setting of the Use Network Topology Info field on the Transport tab

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary, after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 5.4.2** to **5.4.7**.

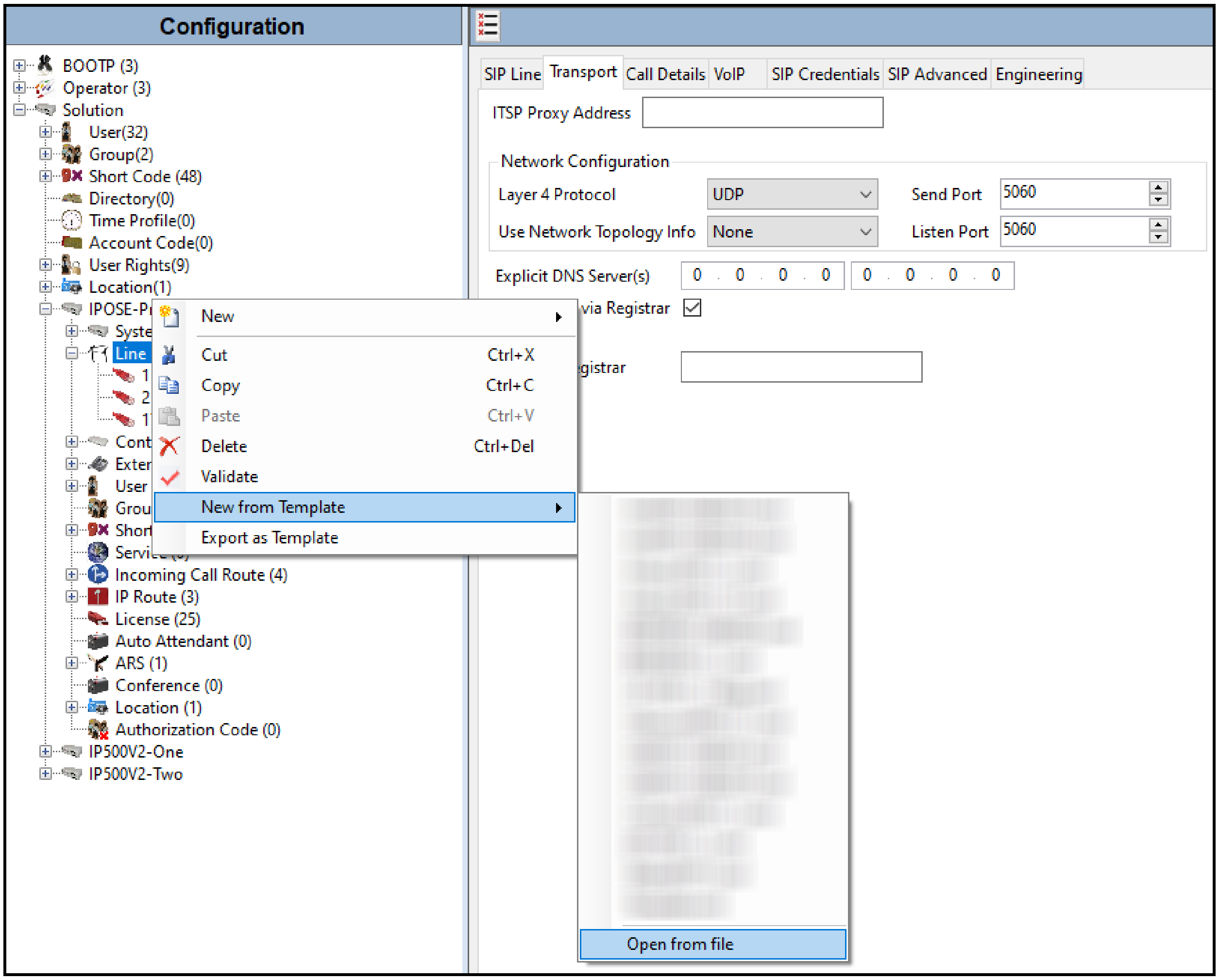
Alternatively, a SIP Line can be created manually. To do so, right-click on **Line** in the **Navigation** pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.2** to **5.4.7**.

5.4.1. Creating a SIP Trunk from an XML Template

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 (**IPOSE-Primary**), right-click on **Line** in the Navigation Pane, then navigate to **New → New from Template → Open from file**.



ITSP Proxy Address

Network Configuration

Layer 4 Protocol: UDP

Send Port: 5060

Use Network Topology Info: None

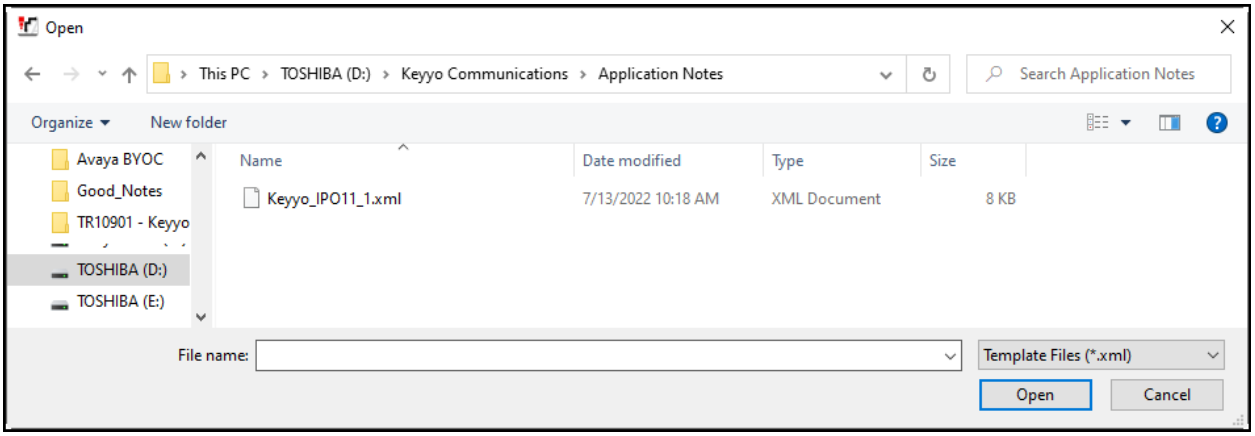
Listen Port: 5060

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

via Registrar ☑

gistrar

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

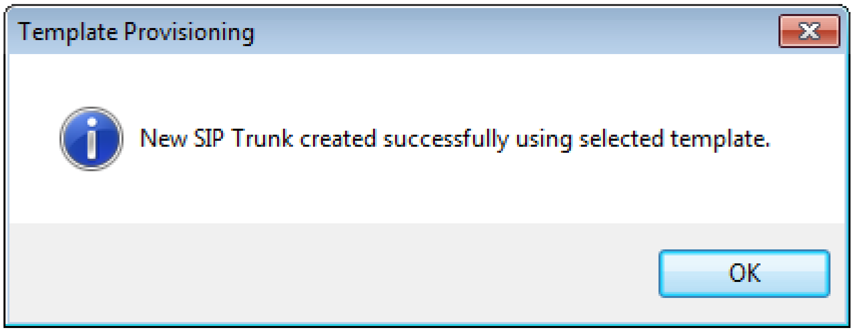


Name	Date modified	Type	Size
Keyyo_IPO11_1.xml	7/13/2022 10:18 AM	XML Document	8 KB

File name:

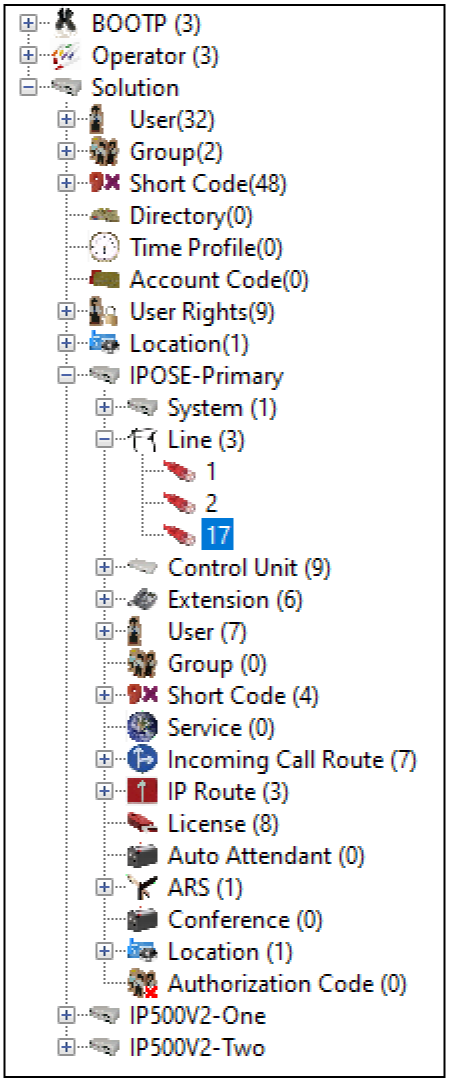
Template Files (*.xml)

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



New SIP Trunk created successfully using selected template.

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

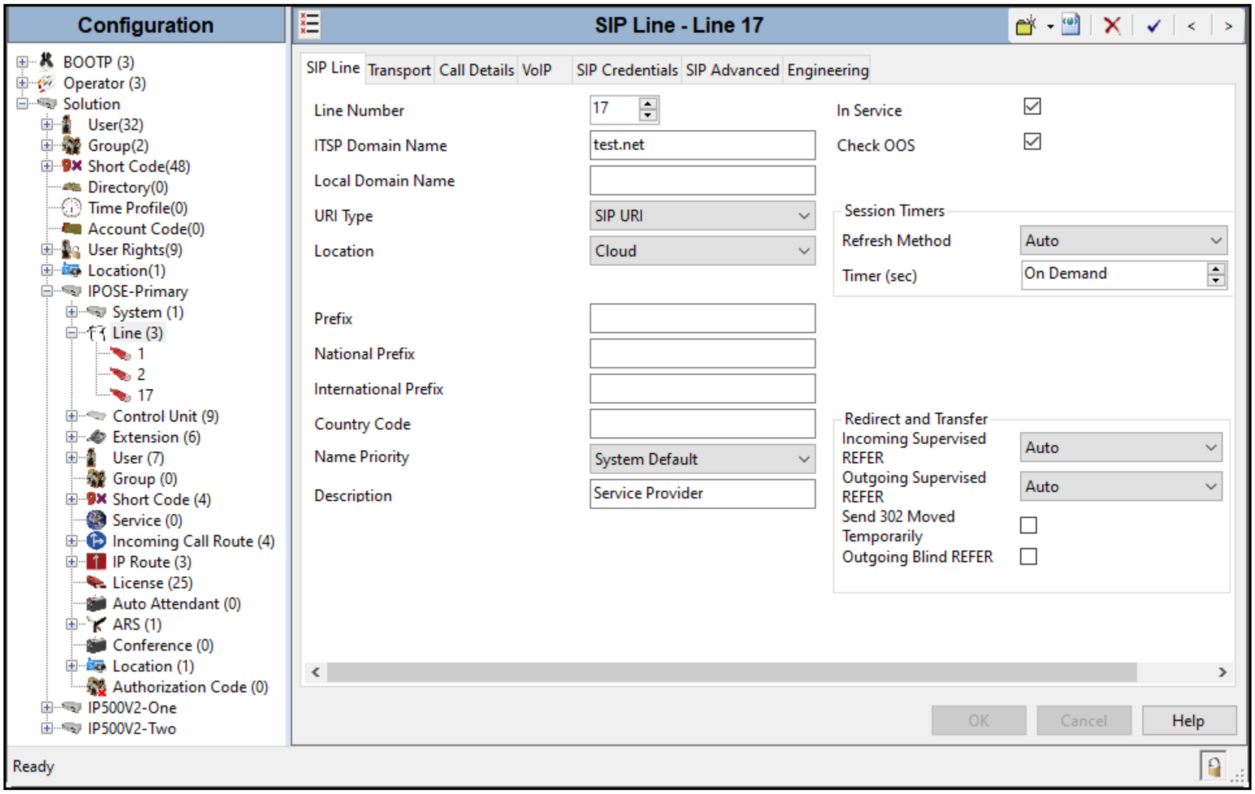


It is important that the SIP Line configuration be reviewed and updated if necessary, after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2** to **5.4.7**.

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:

- Set **ITSP Domain Name** to **test.net**, the domain name provided by Keyyo. **Note**: The Domain Name shown here and throughout this document has been masked for confidentiality and privacy purposes. Set the ITSP Domain Name to the domain name provided by Keyyo instead of the domain shown here.
- 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**.
- Set **Incoming Supervised REFER** and **Outgoing Supervised REFER** to **Auto**.
- Click **OK** to commit.



SIP Line - Line 17

Line Number: 17
ITSP Domain Name: test.net
Local Domain Name:
URI Type: SIP URI
Location: Cloud
Prefix:
National Prefix:
International Prefix:
Country Code:
Name Priority: System Default
Description: Service Provider

In Service ☑
Check OOS ☑

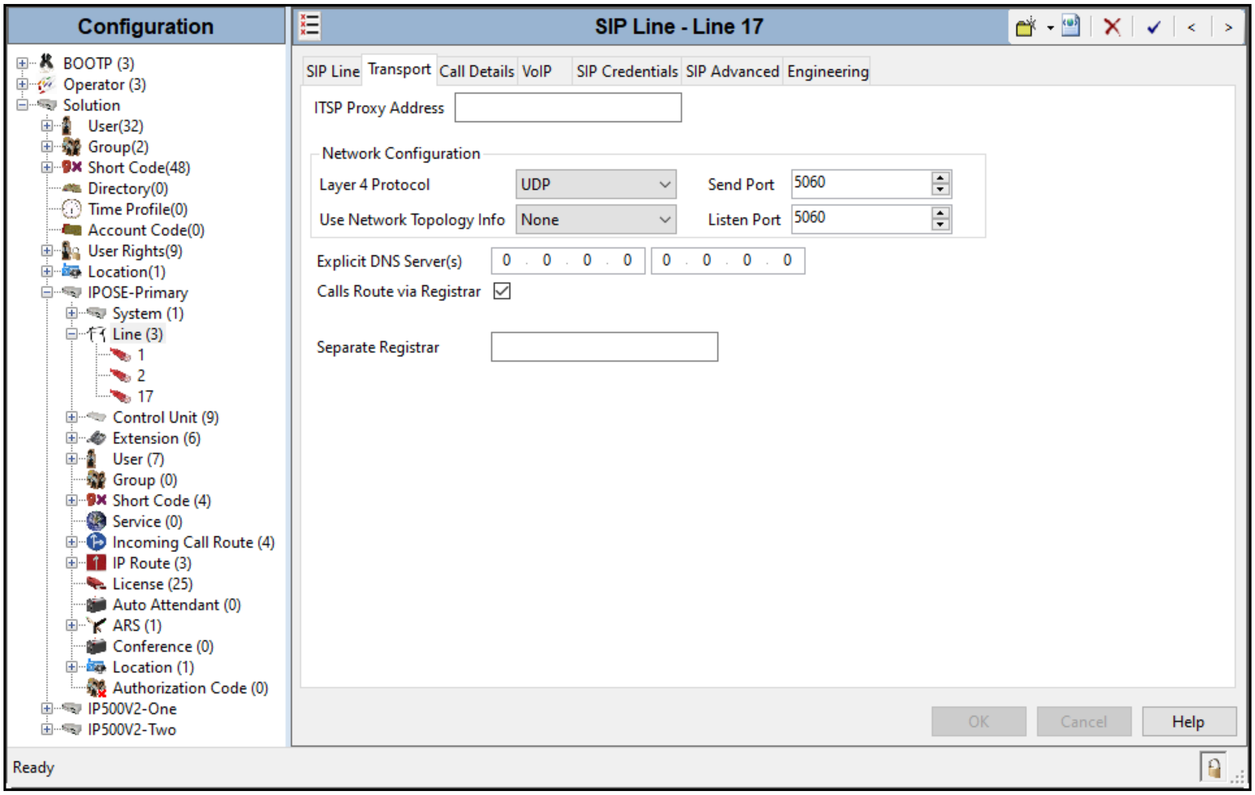
Session Timers
Refresh Method: Auto
Timer (sec): On Demand

Redirect and Transfer
Incoming Supervised REFER: Auto
Outgoing Supervised REFER: Auto
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:

- Leave the **ITSP Proxy Address** blank (IP Office will retrieve the ITSP Proxy Address via public DNS queries using the ISTP Domain Name provided under in **Section 5.4.2**). The public DNS IP addresses were configured under **Section 5.2.2**.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **None** (refer to the note below).
- Set the **Send Port** and **Listen Port** to **5060**.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).



ITSP Proxy Address:

Layer 4 Protocol: UDP

Send Port: 5060

Use Network Topology Info: None

Listen Port: 5060

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

Calls Route via Registrar: ☑

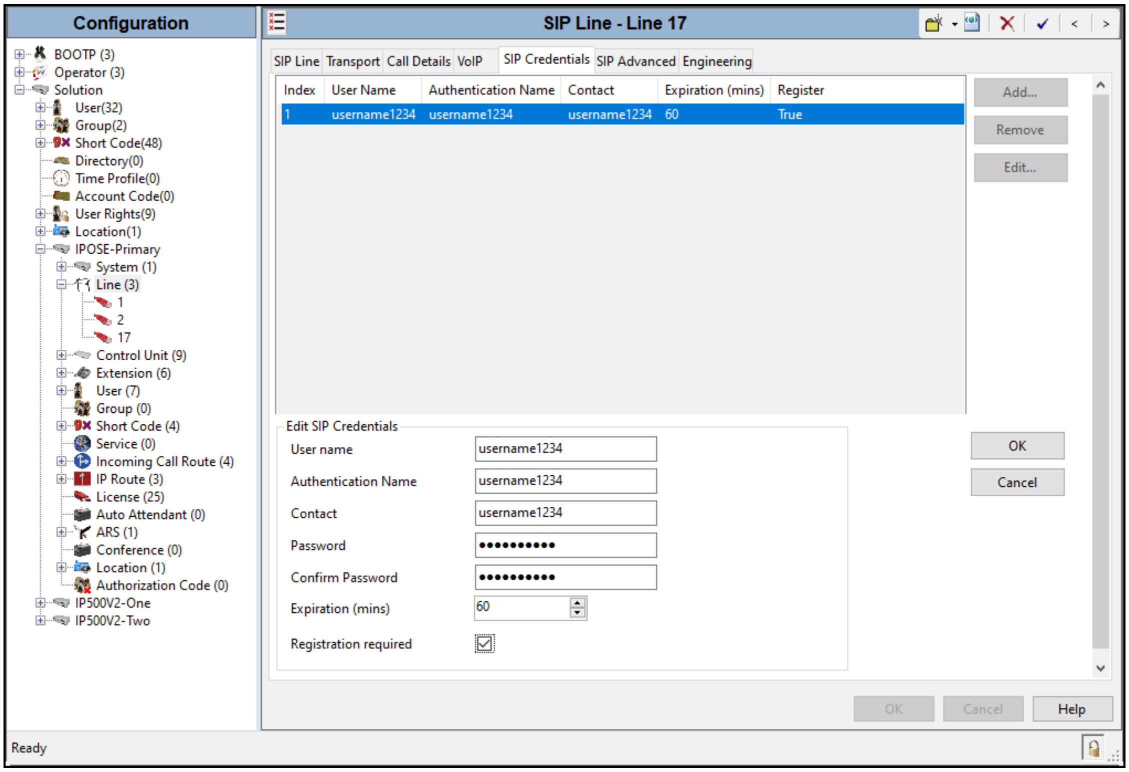
Separate Registrar:

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 – 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 provided by Keyyo.
- For **Authentication Name** and **Contact** enter the username credential provided by Keyyo for SIP Trunk registration, same as above.
- For **Password** and **Confirm Password**, enter the password credential provided by Keyyo for SIP Trunk registration.
- Set **Expiry (mins)** to a value acceptable to the enterprise. This setting defines how often registration with Keyyo 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 **Registration required** was checked. When selected, the credentials fields above are included in the SIP INVITE messages sent to Keyyo when making calls, required by Keyyo.



Index	User Name	Authentication Name	Contact	Expiration (mins)	Register
1	username1234	username1234	username1234	60	True

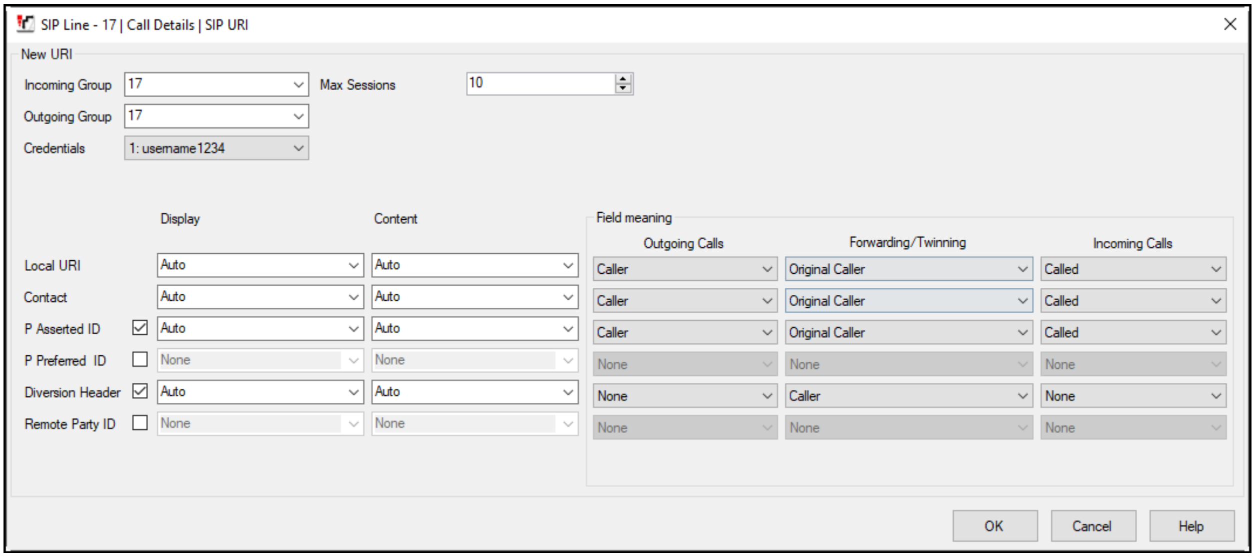
Edit SIP Credentials

User name: username1234
Authentication Name: username1234
Contact: username1234
Password: ••••••••••
Confirm Password: ••••••••••
Expiration (mins): 60
Registration required: ☑

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 a new entry was created with the parameters shown below:

- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **17** was defined that only contains this line (line 17).
- Under **Credentials**, select **1: username1234** 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.
- Verify **P Asserted ID** and **Diversion Header** are checked.
- Set the **Local URI**, **Contact**, **P Asserted ID** and **Diversion Header** fields to the values shown in the screenshot below.
- Set all remaining fields as shown on the screenshot below.
- Click **OK**.



SIP Line - 17 | Call Details | SIP URI

New URI

Incoming Group: 17

Max Sessions: 10

Outgoing Group: 17

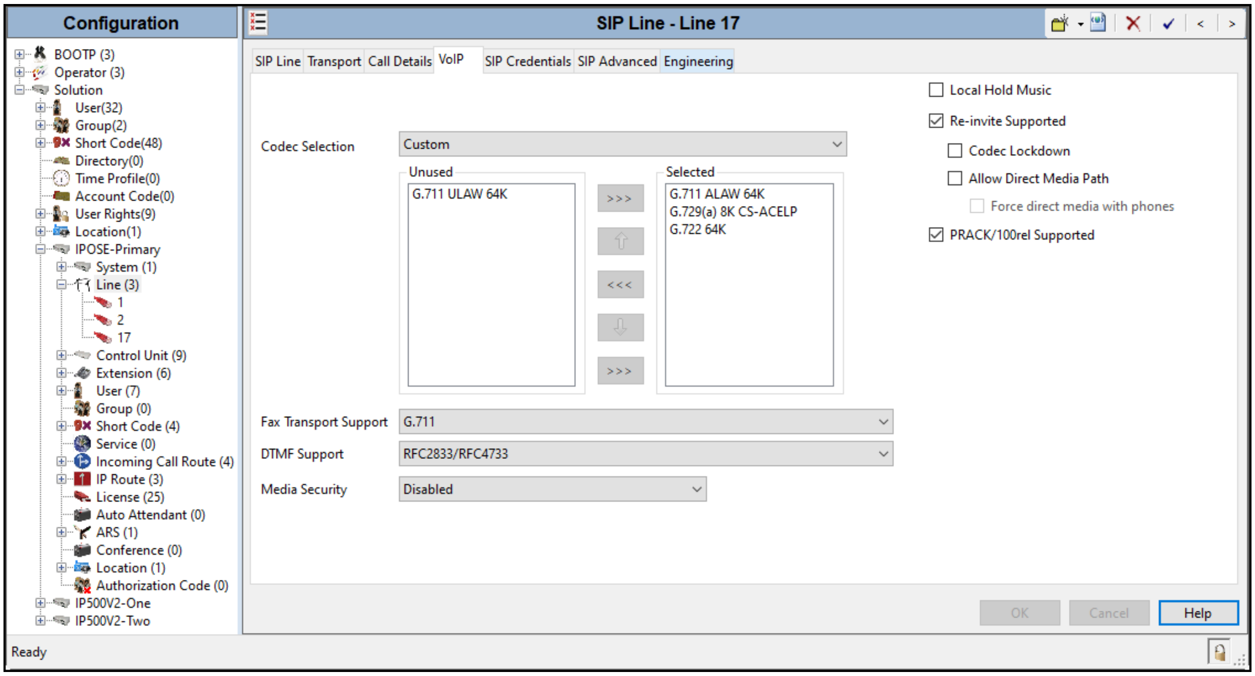
Credentials: 1: username1234

				Field meaning		
		Display	Content	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

5.4.6. 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 **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. Keyyo supports codecs **G.711ALAW**, **G.729(a)** and **G.722 64K** 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.
- Check the **Re-invite Supported** box.
- Check the **PRACK/100rel Supported** box.
- Default values may be used for all other parameters.



Codec Selection: Custom

Unused: G.711 ULAW 64K

Selected: G.711 ALAW 64K, G.729(a) 8K CS-ACELP, G.722 64K

Fax Transport Support: G.711

DTMF Support: RFC2833/RFC4733

Media Security: Disabled

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

Note: The codec selections defined under this section are the codecs selected for the SIP Line (Trunk). The codec selections defined under **Section 5.2.4.1** are the codecs selected for the IP phones/extension (H.323 and SIP).

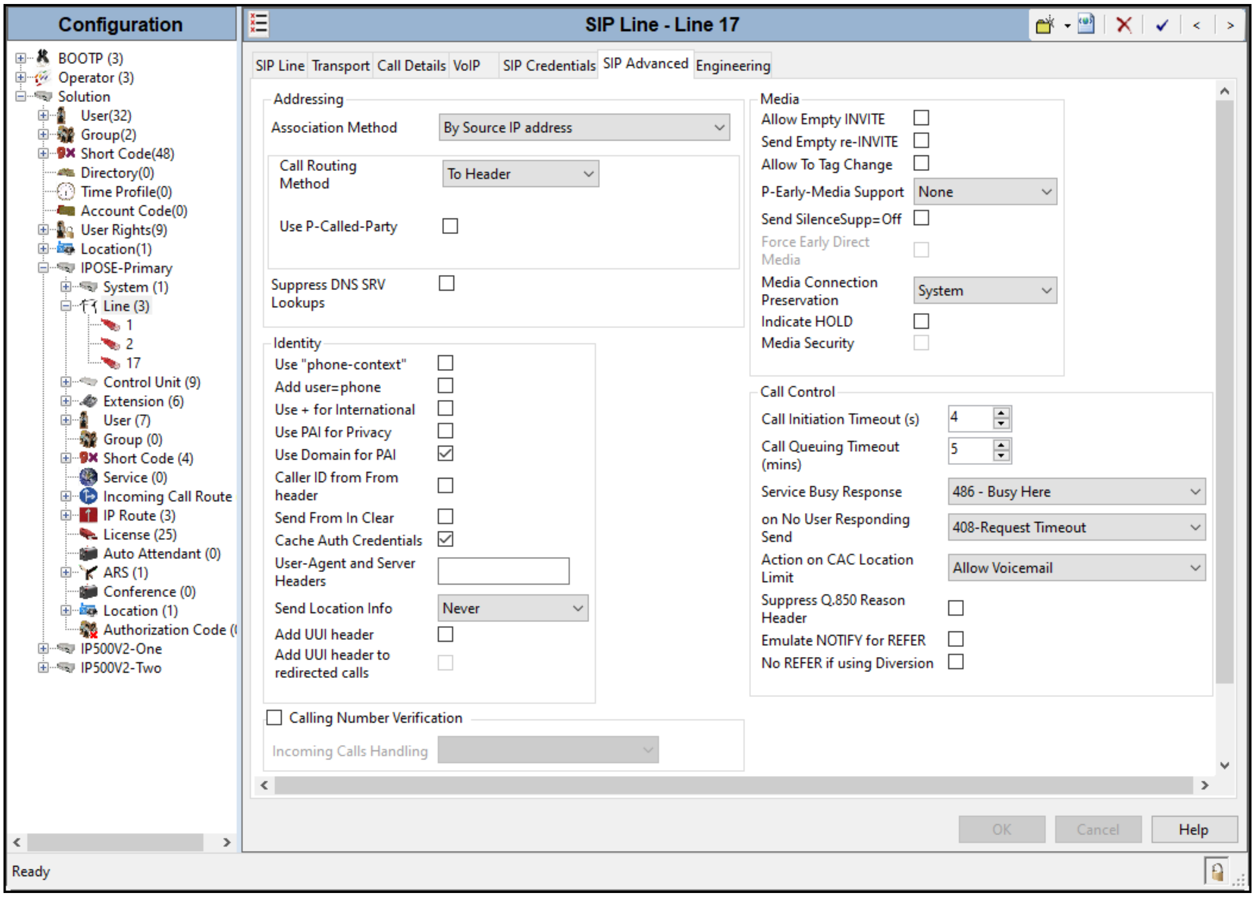
5.4.7. SIP Line – SIP Advanced Tab

In the **Addressing** area:

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

In the **Identity** area:

- Verify that **Cache Auth Credentials** box is checked (Default = On). When set to On, allows the credentials challenge and response from a registration transaction to be automatically inserted into later SIP messages without waiting for a subsequent challenge.
- Default values may be used for all other parameters.
- Click **OK** to commit.



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:

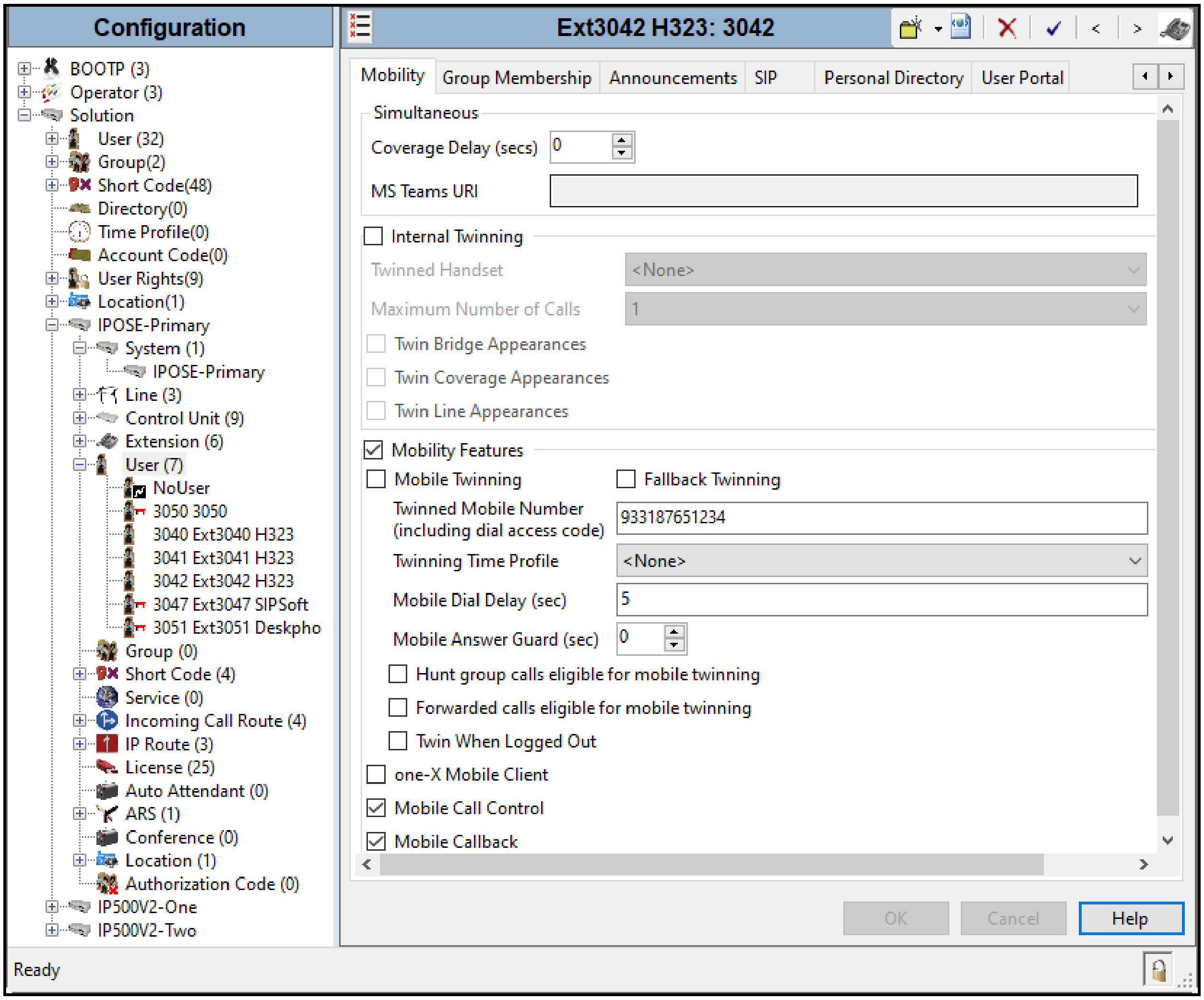
Media
Allow Empty INVITE ☐
Send Empty re-INVITE ☐
Allow To Tag Change ☐
P-Early-Media Support: None
Send SilenceSupp=Off ☐
Force Early Direct Media ☐
Media Connection Preservation: System
Indicate HOLD ☐
Media Security ☐

Call Control
Call Initiation Timeout (s): 4
Call Queuing Timeout (mins): 5
Service Busy Response: 486 - Busy Here
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. Mobility

Select the **Mobility** tab for the user. In the sample configuration user 3042 was one of the users configured to test the Mobile Twinning feature. The following screen shows the Mobility tab for user 3042. The Mobility Features and Mobile Call Control boxes are checked. The Twinned Mobile Number field is configured with the number to dial to reach the twinned telephone, in this case 933187651234 (including dial access code “9”). Other options can be set according to customer requirements.

Note: Checking the **Mobile Call Control** box allows a user receiving a call on their twinned device to access system dial tone and then perform dialing action including making calls and activating short codes.



Simultaneous

Coverage Delay (secs) 0

MS Teams URI

☐ Internal Twinning

Twinned Handset <None>

Maximum Number of Calls 1

☐ Twin Bridge Appearances

☐ Twin Coverage Appearances

☐ Twin Line Appearances

☑ Mobility Features

☐ Mobile Twinning ☐ Fallback Twinning

Twinned Mobile Number (including dial access code) 933187651234

Twinning Time Profile <None>

Mobile Dial Delay (sec) 5

Mobile Answer Guard (sec) 0

☐ Hunt group calls eligible for mobile twinning

☐ Forwarded calls eligible for mobile twinning

☐ Twin When Logged Out

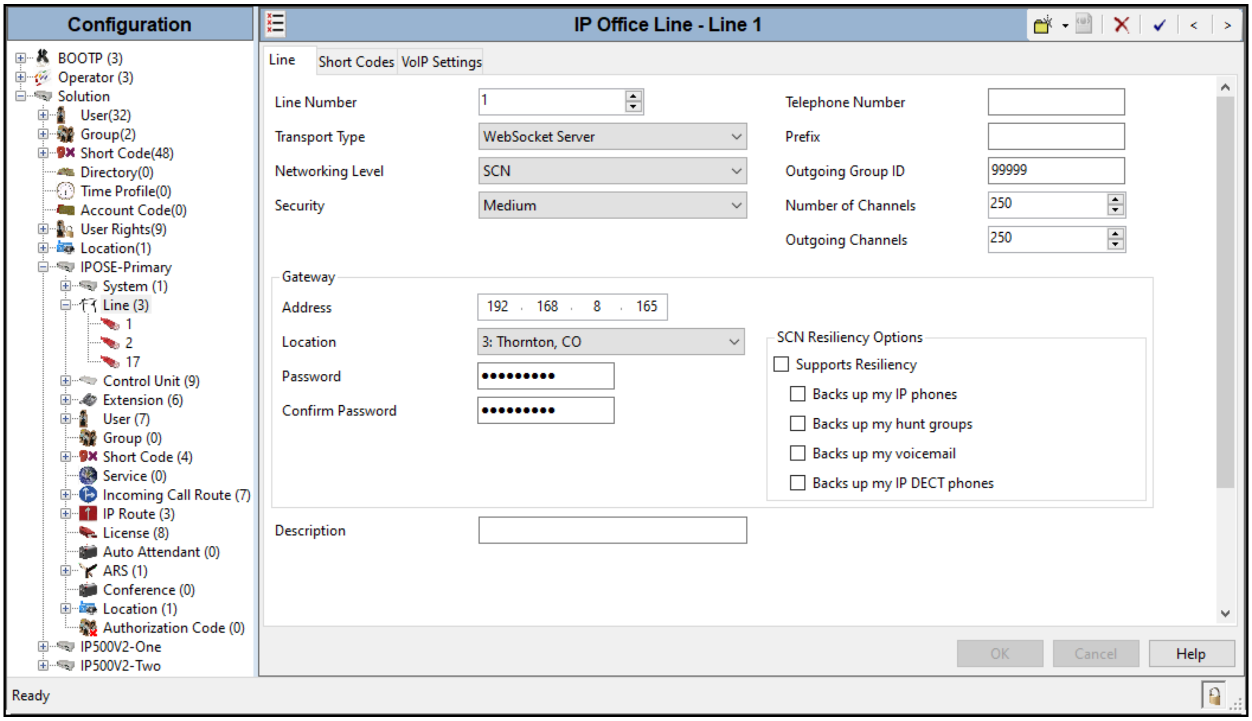
☐ one-X Mobile Client

☑ Mobile Call Control

☑ Mobile Callback

5.6. 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 P500V2-One Expansion System.



Line Number: 1

Transport Type: WebSocket Server

Networking Level: SCN

Security: Medium

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

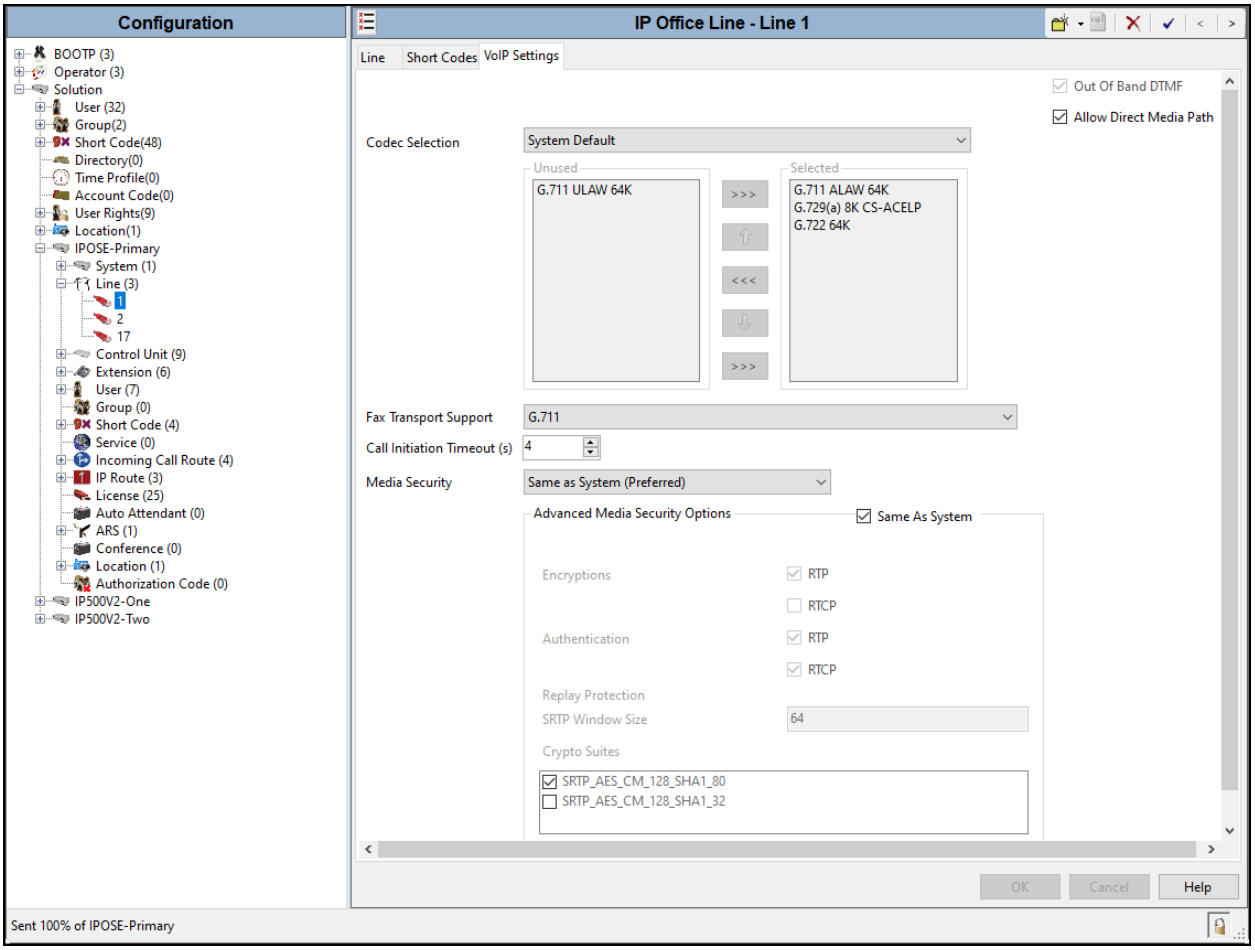
☐ Backs up my voicemail

☐ 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.



☑ Out Of Band DTMF
☑ Allow Direct Media Path

Codec Selection: System Default

Unused: G.711 ULAW 64K

Selected:
G.711 ALAW 64K
G.729(a) 8K CS-ACELP
G.722 64K

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

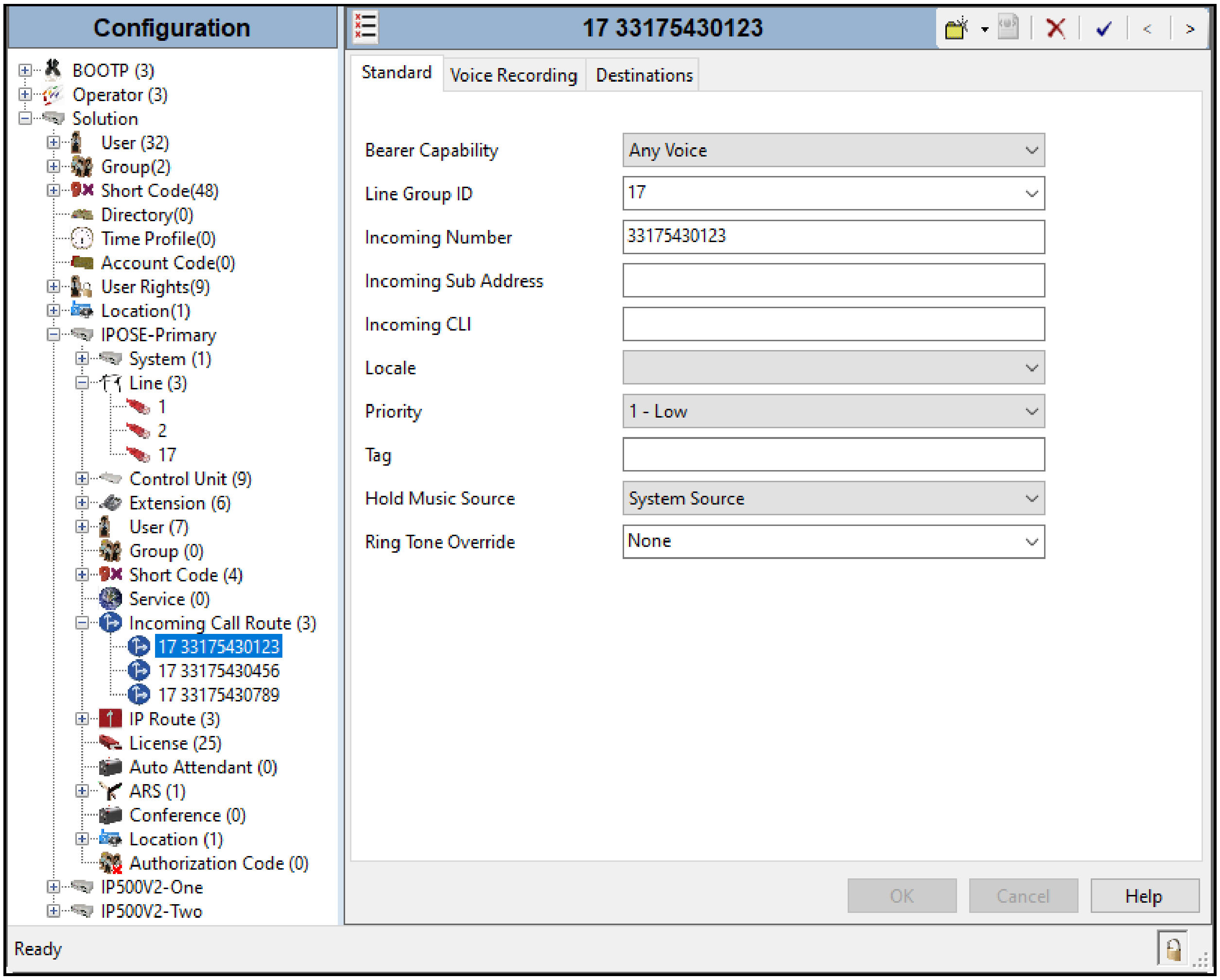
Sent 100% of IPOSE-Primary

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

5.7. Incoming Call Route

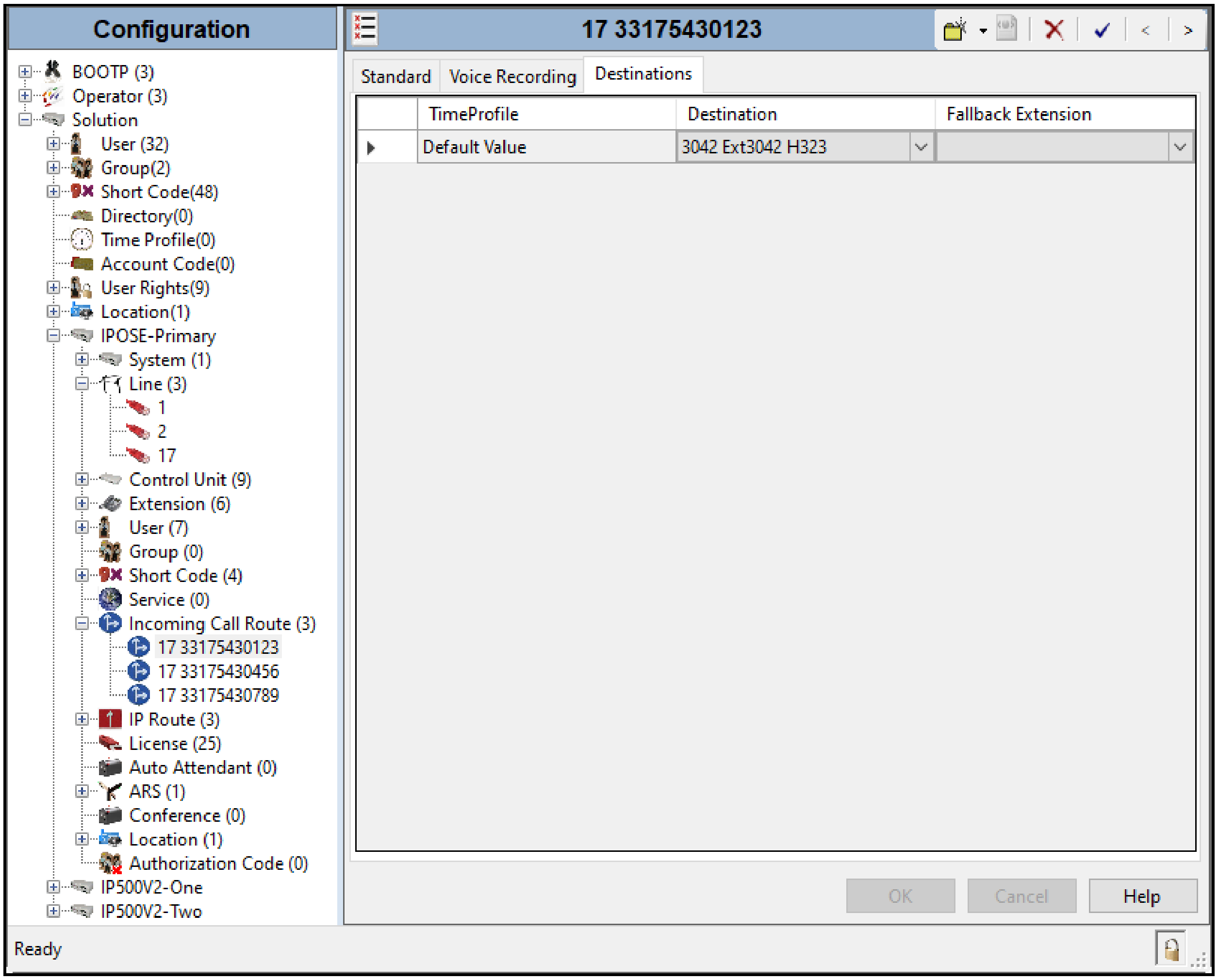
Incoming call routes map inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc., within the IP Office system. To add an incoming call route, right click on **Incoming Call Route** in the **Navigation** pane and select **New** (not shown). On the Details Pane, under 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 Keyyo.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).



Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	33175430123
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 33175430123 provided by Keyyo was associated with the Avaya IP Office extension **3042 Ext3042 H323**.



17 33175430123

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.).

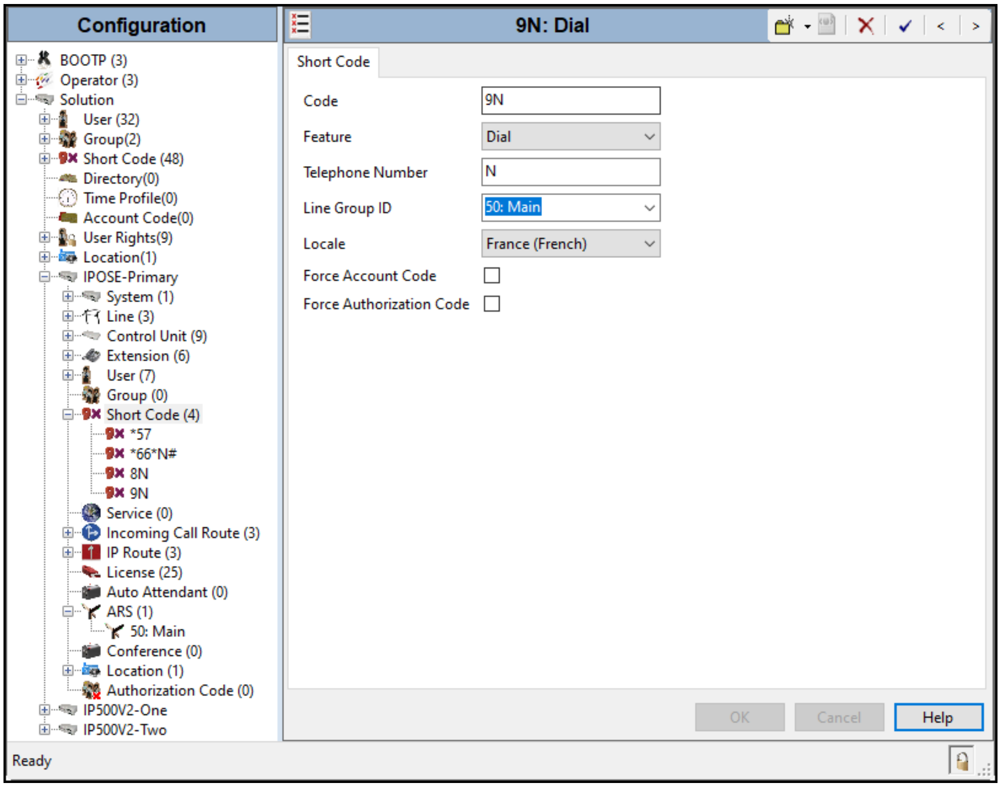
5.8. Outbound Call Routing

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.

5.8.1. 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**, **France** (**French**) was used.
- Click the **OK** to commit (not shown).



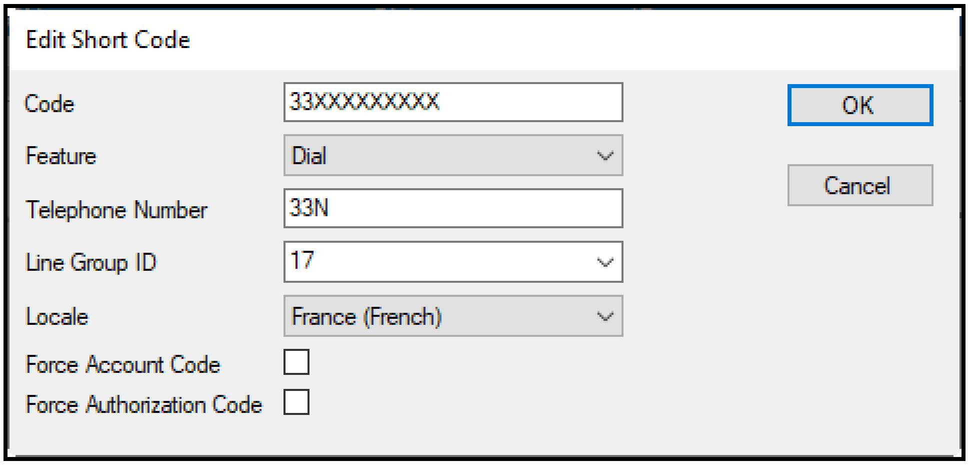
Code: 9N
Feature: Dial
Telephone Number: N
Line Group ID: 50: Main
Locale: France (French)
Force Account Code ☐
Force Authorization Code ☐

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, **33** followed by **9 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 **33N**. The value **N** represents the additional number of digits dialed by the user after dialing **33** (The **9** will be stripped off). With this setting **33** and the dialed **9** digits number will be sent to the trunk, with **33** preceding the dialed **9**-digits number, **33** is the country code (France).
- 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**, **France (French)** was used
- Click **OK** to commit.

The following examples shows the dial pattern used during the compliance test.



Code: 33XXXXXXXXX
Feature: Dial
Telephone Number: 33N
Line Group ID: 17
Locale: France (French)
Force Account Code ☐
Force Authorization Code ☐

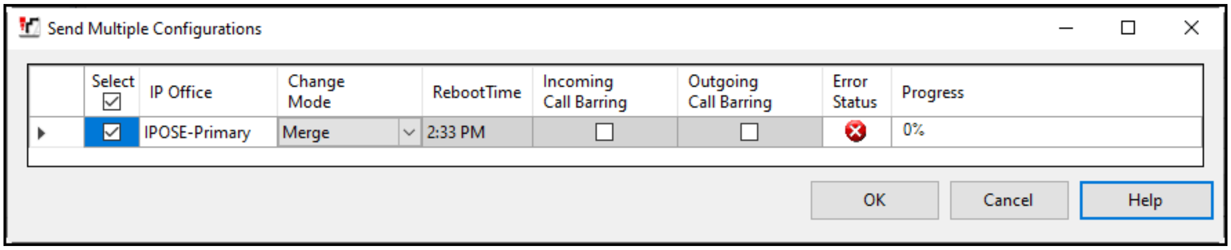
Repeat the above procedure for additional dial patterns to be used by the enterprise to dial out from IP Office.

5.9. 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.

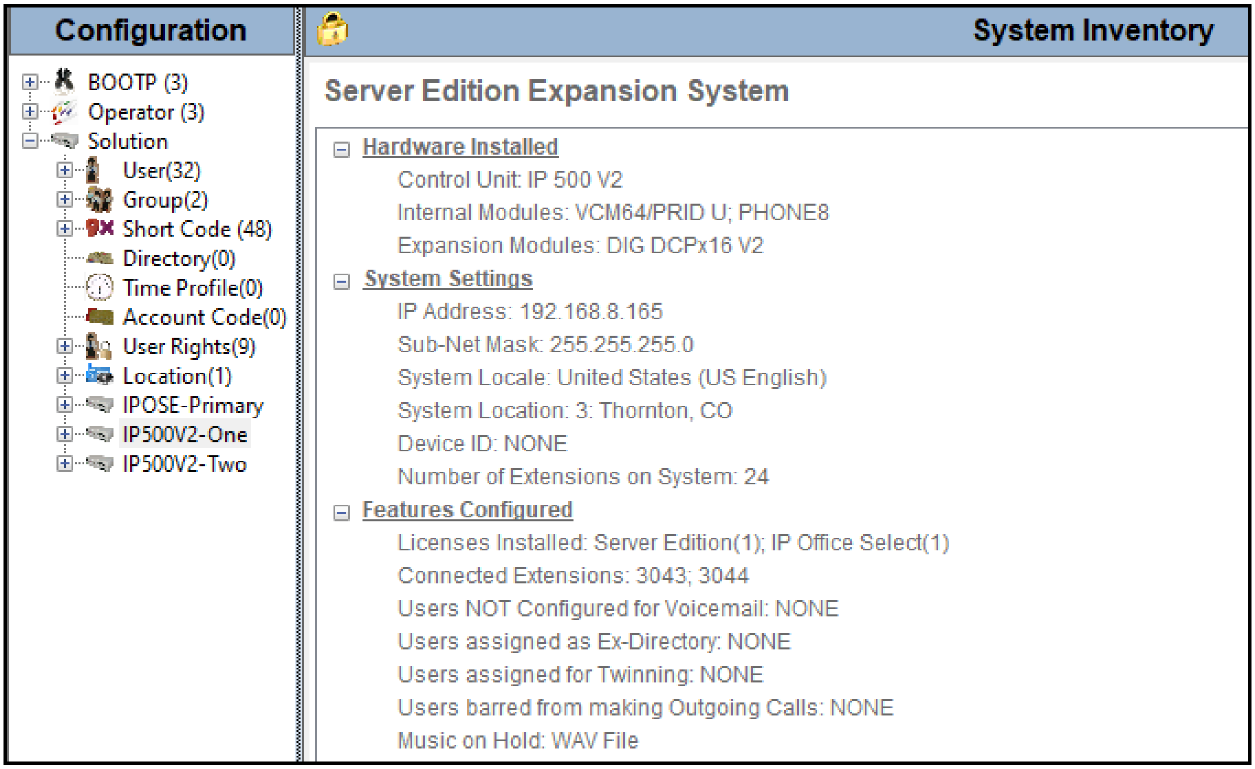
Click **OK** to execute the save.



	Select ☑	IP Office	Change Mode	RebootTime	Incoming Call Barring	Outgoing Call Barring	Error Status	Progress
▸	☑	IPOSE-Primary	Merge	2:33 PM	☐	☐	✖	0%

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 IP Office Expansion system.



Server Edition Expansion System

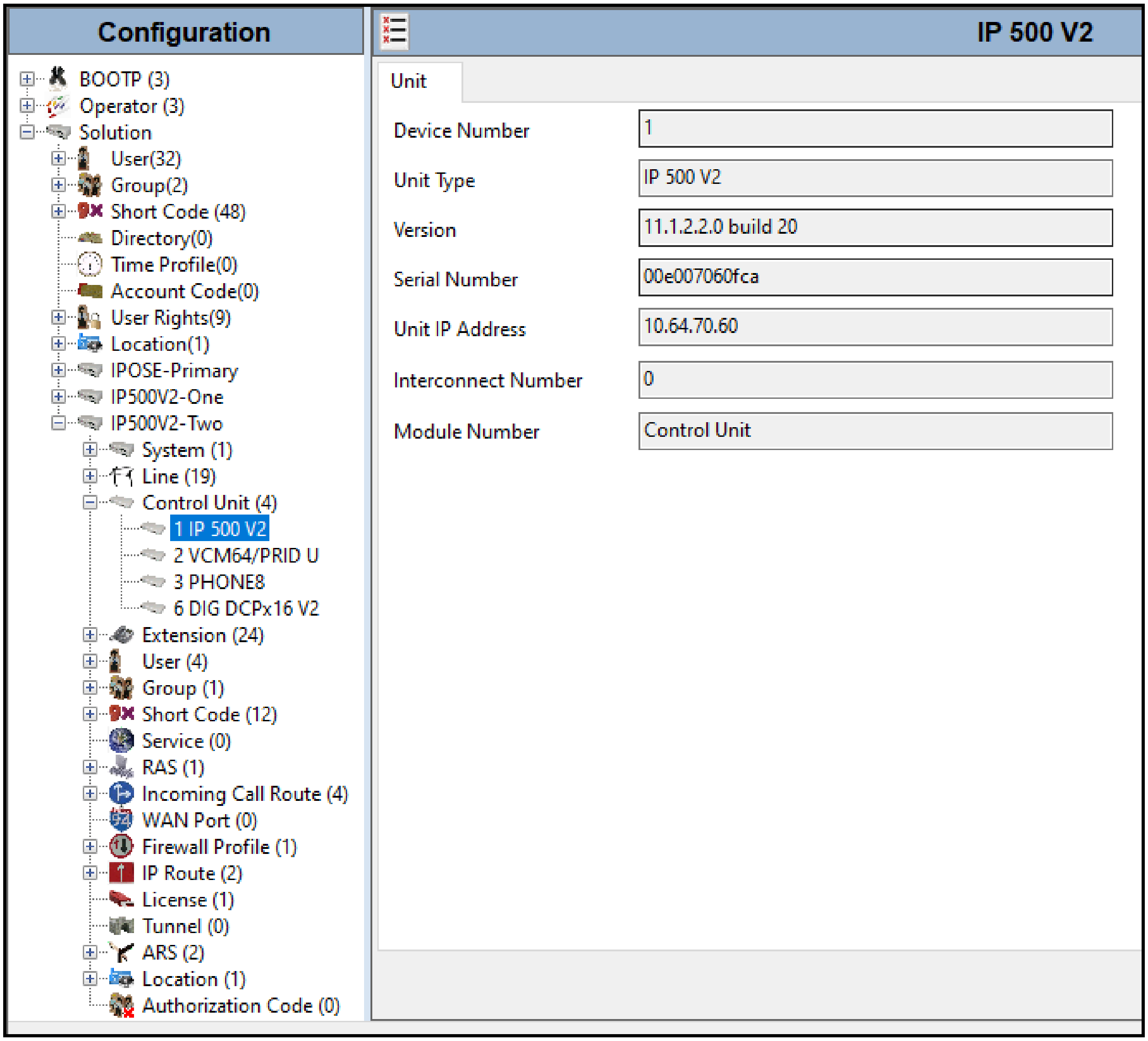
Hardware Installed
Control Unit: IP 500 V2
Internal Modules: VCM64/PRID U; PHONE8
Expansion Modules: DIG DCPx16 V2

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)
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.

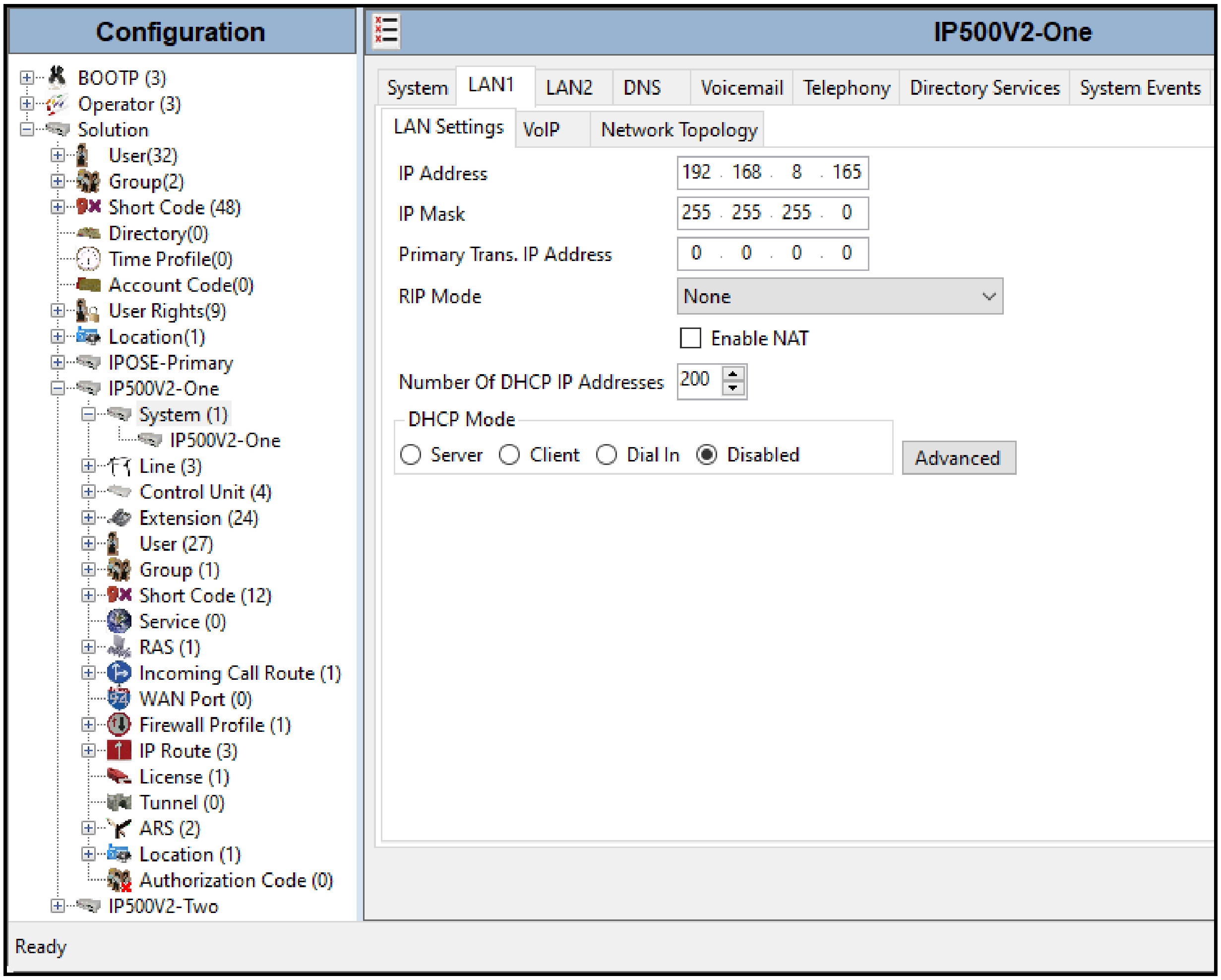


Device Number	1
Unit Type	IP 500 V2
Version	11.1.2.2.0 build 20
Serial Number	00e007060fca
Unit IP Address	10.64.70.60
Interconnect Number	0
Module Number	Control Unit

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).



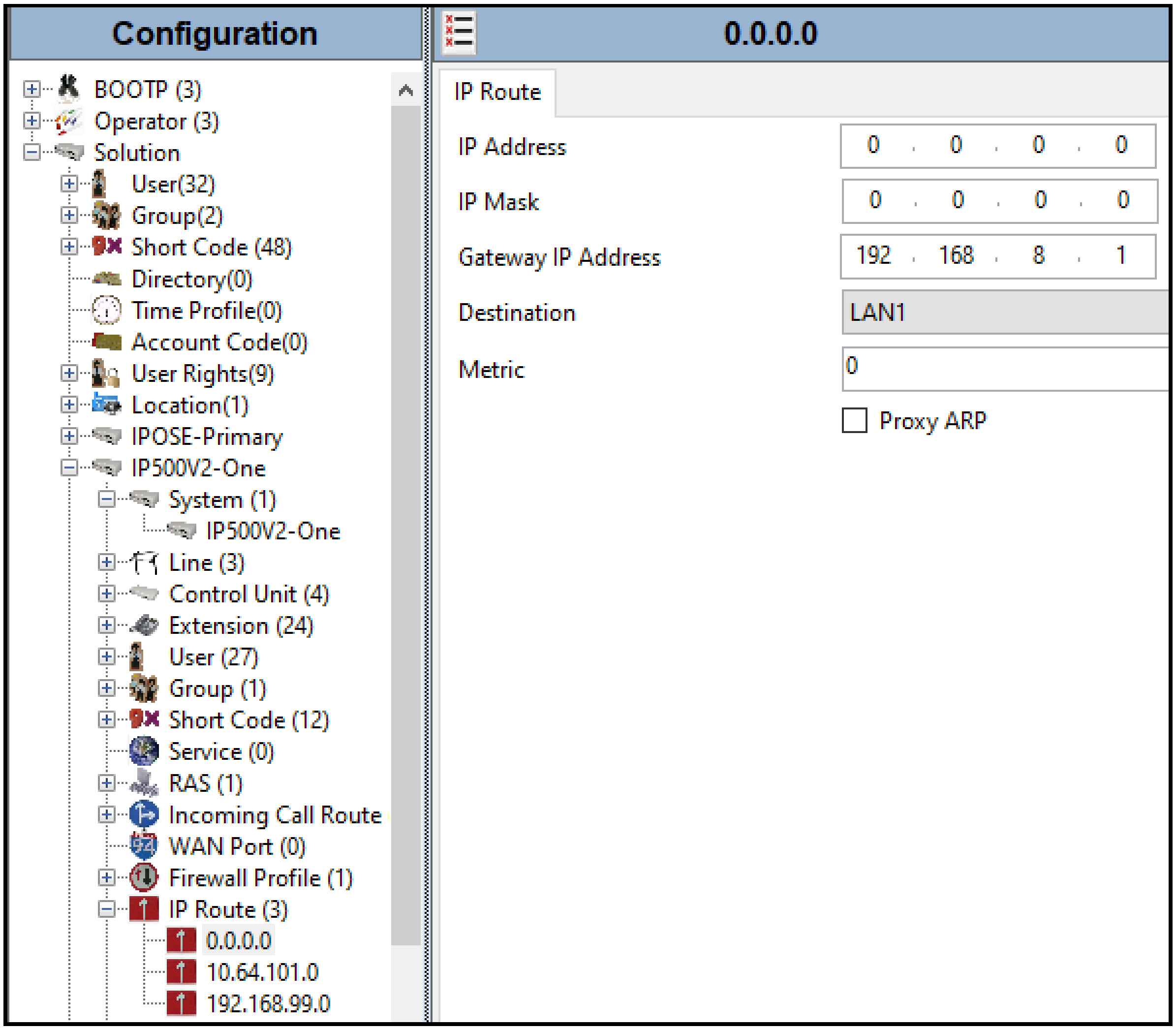
IP Address: 192 . 168 . 8 . 165
IP Mask: 255 . 255 . 255 . 0
Primary Trans. IP Address: 0 . 0 . 0 . 0
RIP Mode: None
☐ Enable NAT
Number Of DHCP IP Addresses: 200
DHCP Mode: ○ Server ○ Client ○ Dial In ● Disabled

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).

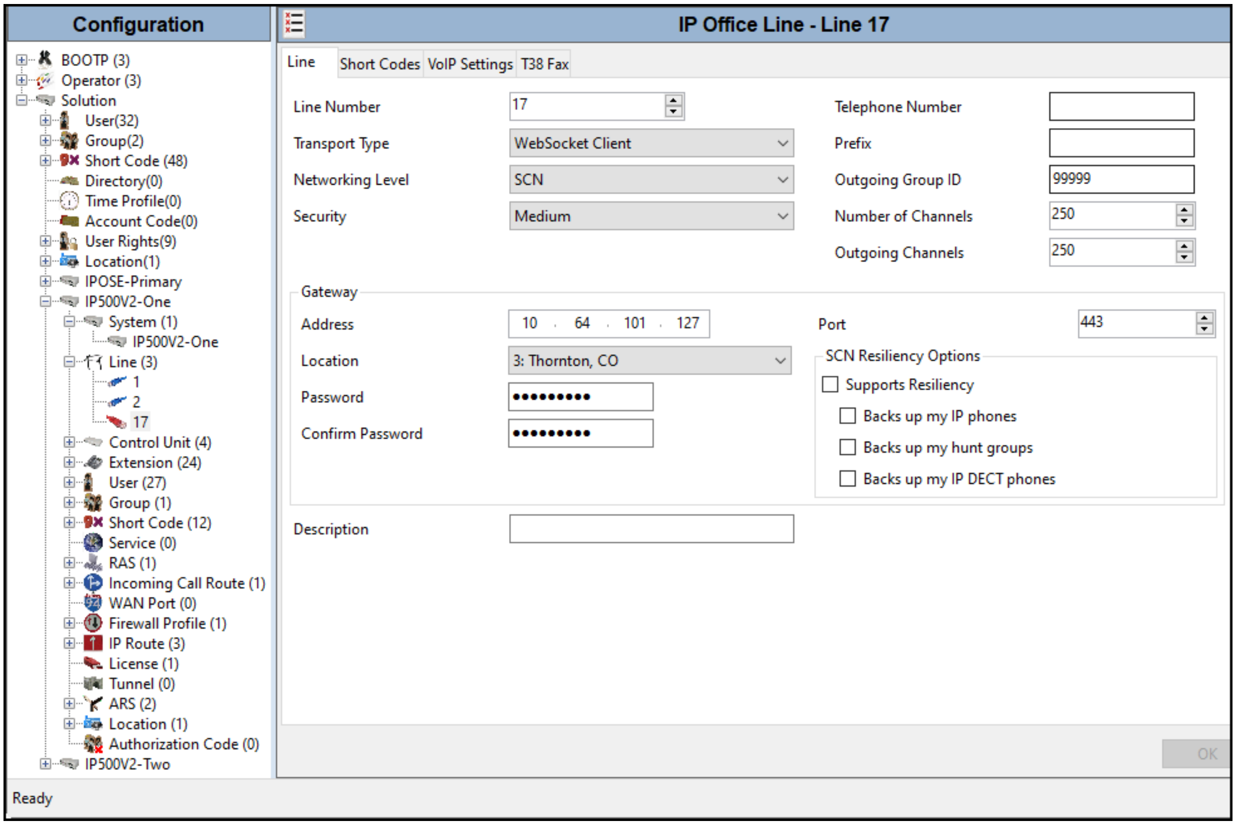
- Enter **0.0.0.0** on the **IP Address** and **IP Mask** fields to make this the default route.
- Set **Gateway IP Address** to the IP Address of the default router in the IP Office subnet. The default gateway in the reference configuration was **192.168.8.1**
- Set **Destination** to **LAN1** from the pull-down menu.
- Click **OK** to commit (not shown).



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.



Line Number: 17

Transport Type: WebSocket Client

Networking Level: SCN

Security: Medium

Telephone Number:

Prefix:

Outgoing Group ID: 99999

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

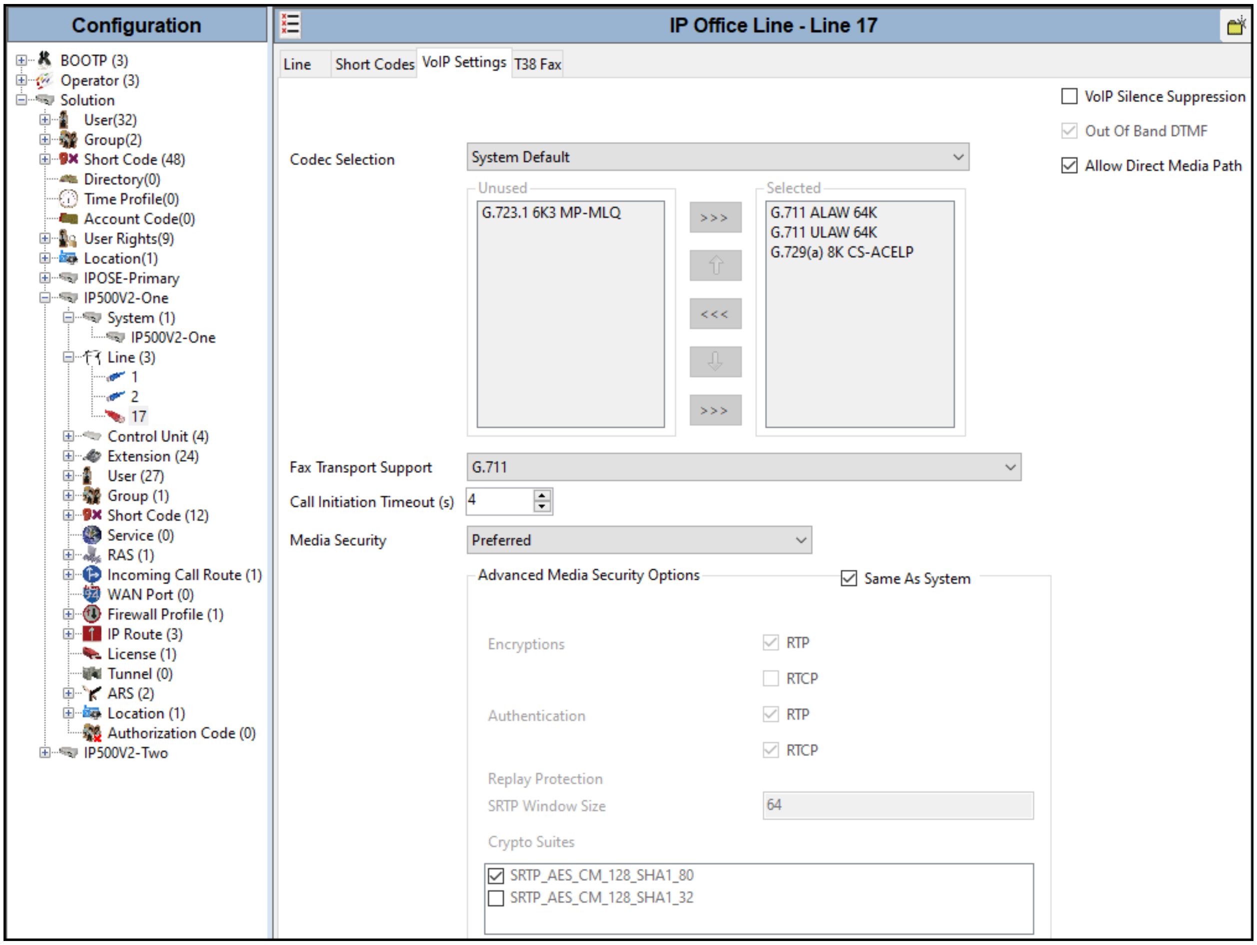
☐ Backs up my hunt groups

☐ 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.
- Under **Advanced Media Security Options Same as System** was selected.
- Click **OK** to commit (not shown).



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

Codec Selection: System Default

Unused: G.723.1 6K3 MP-MLQ

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

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

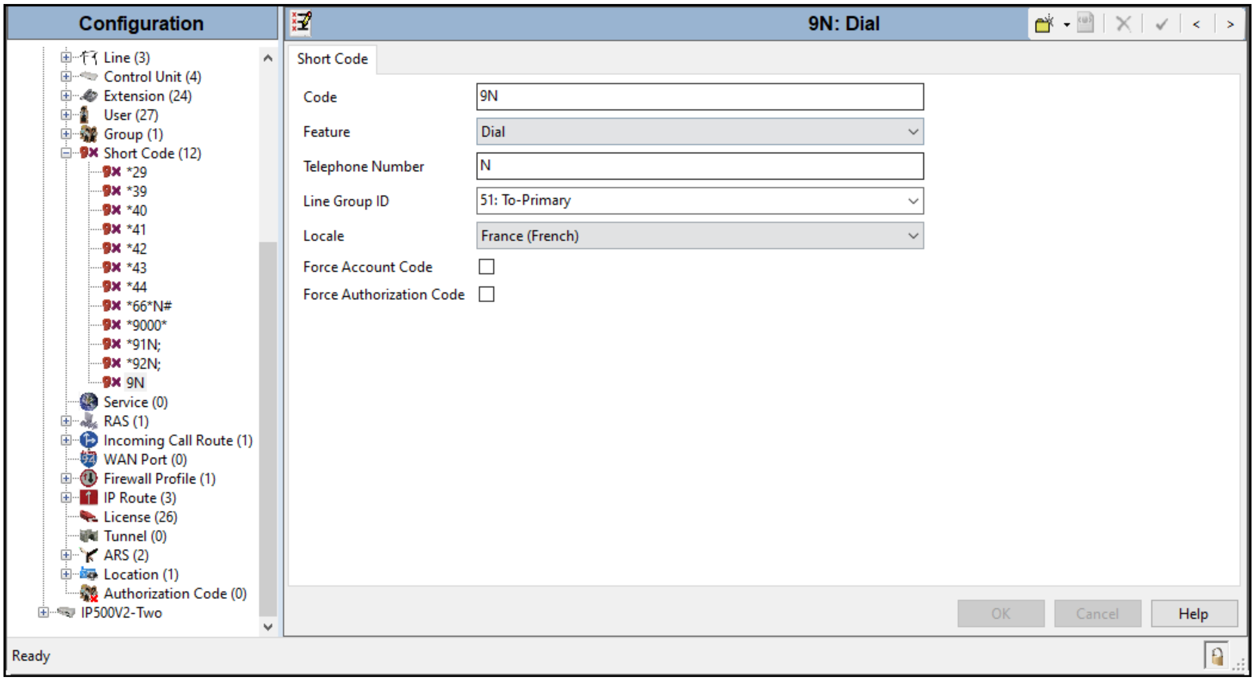
Replay Protection

SRTP Window Size: 64

Crypto Suites:
☑ SRTP_AES_CM_128_SHA1_80
☐ SRTP_AES_CM_128_SHA1_32

6.5. Short Codes

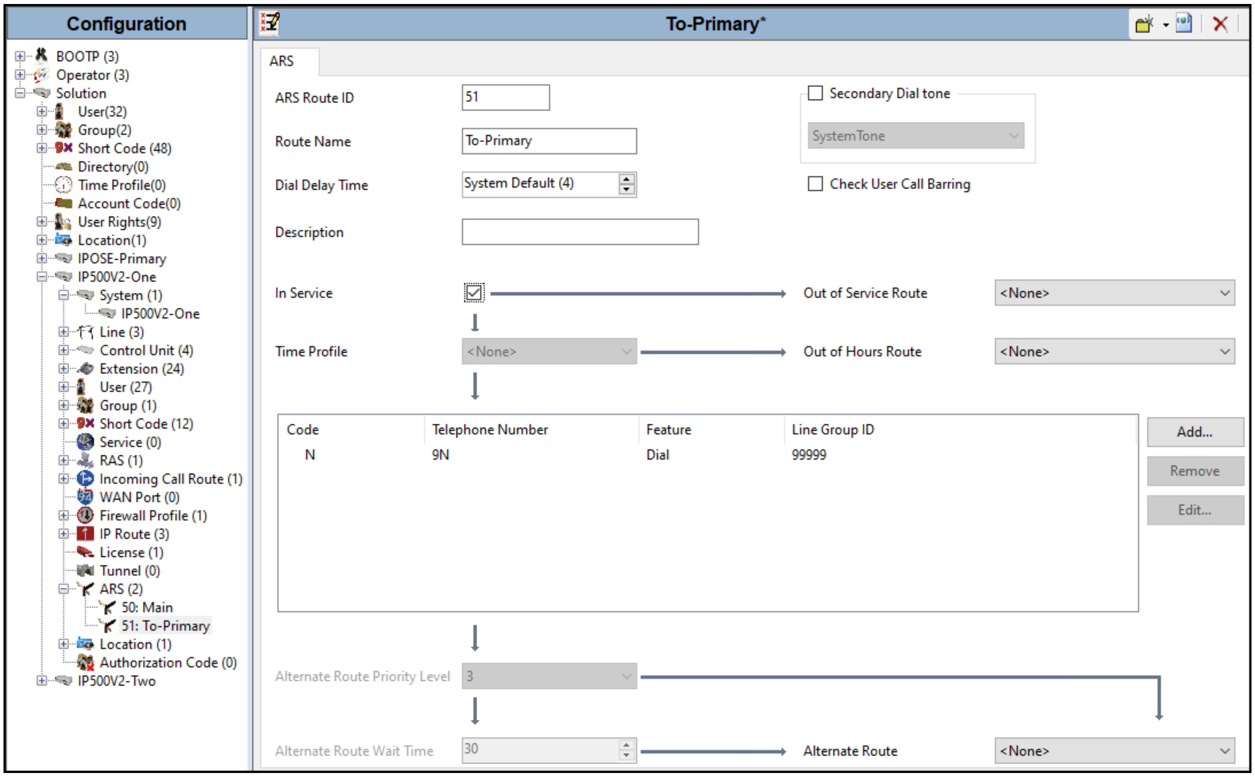
Similar to the configuration of the Primary server in **Section 5.8.1**, create a Short Code to access ARS. In the reference configuration, the **Line Group ID** is set to the ARS route illustrated in the next section.



Code: 9N
Feature: Dial
Telephone Number: N
Line Group ID: 51: To-Primary
Locale: France (French)
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

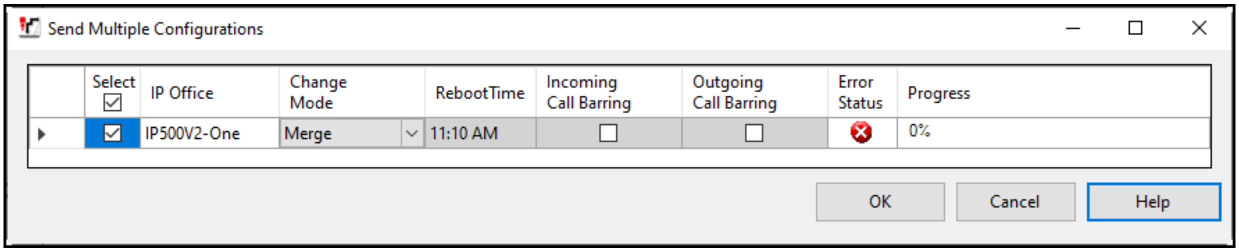
Alternate Route: <None>

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	11:10 AM	☐	☐	☒	0%

7. Keyyo Communications SIP Trunking Service Configuration

To use Keyyo Communications SIP Trunking Service, a customer must request the service from Keyyo using the established sales processes. The process can be started by contacting Keyyo via the corporate web site at: https://www.keyyo.com/fr/support/contactsupport-partenaire/

During the signup process, Keyyo and the customer will discuss details about the preferred method to be used to connect the customer's Avaya enterprise network to the Keyyo Communications SIP Trunking Service network.

Keyyo will provide the following information:

- SIP domain name.
- SIP Trunk registration credentials (User Name, Password, etc.).
- DID numbers.
- Public DNS IP addresses, etc.

Note: The SIP Trunk registration credentials, Domain Name, DIDs, etc., shown in this document, were masked for confidentiality and privacy purposes. During the signup process Keyyo will provide the customer the necessary information to configure Avaya IP Office.

8. Verification Steps

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

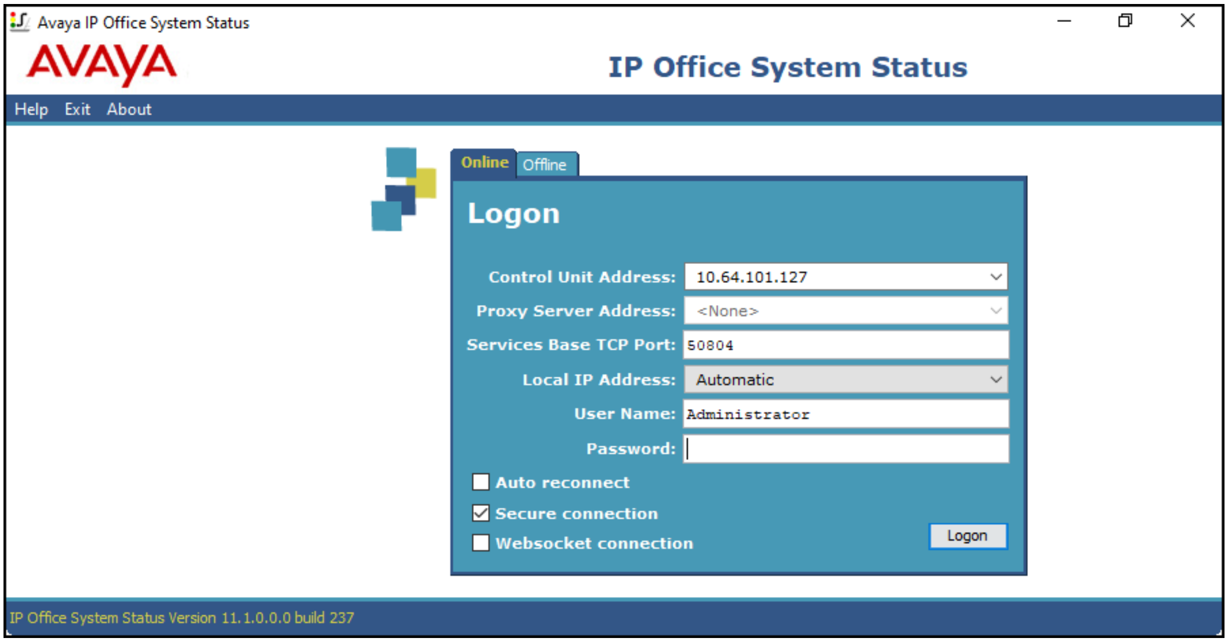
The following steps may be used to verify the configuration:

- 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.

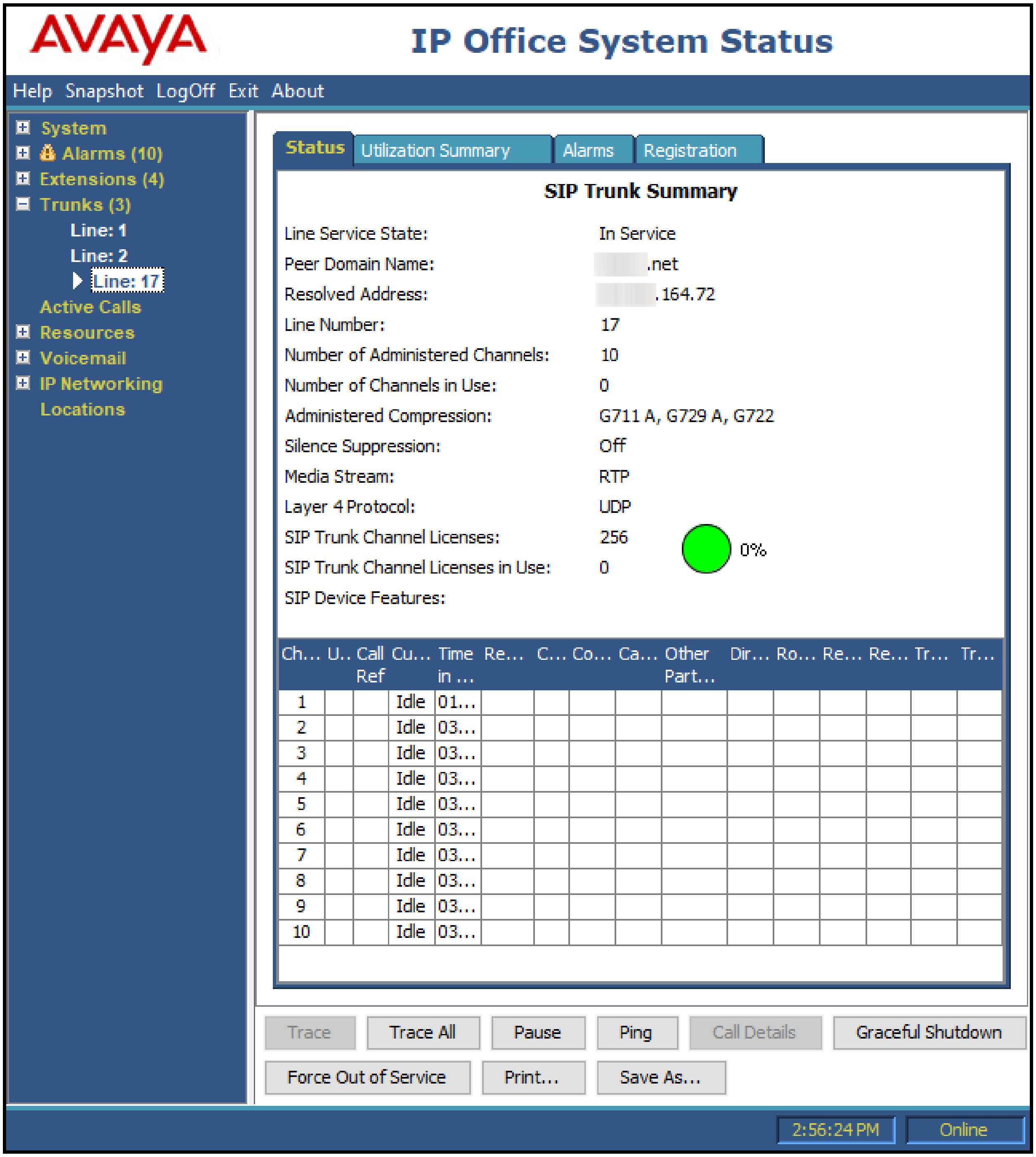
Use the IP Office **System Status** application to verify the state of SIP connections. Launch the application from **Start** → **Programs** → **IP Office** → **System Status** on the PC where IP Office Manager is installed, log in with the proper credentials.



Control Unit Address: 10.64.101.127
Proxy Server Address: <None>
Services Base TCP Port: 50804
Local IP Address: Automatic
User Name: Administrator
Password:
☐ Auto reconnect
☑ Secure connection
☐ Websocket connection

IP Office System Status Version 11.1.0.0.0 build 237

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.



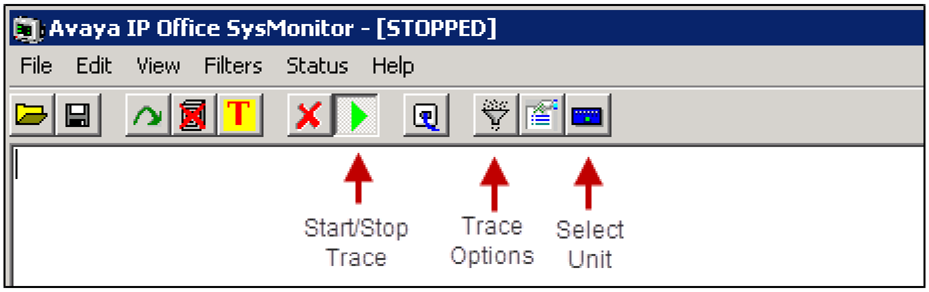
SIP Trunk Summary

Line Service State:	In Service
Peer Domain Name:	[redacted].net
Resolved Address:	[redacted].164.72
Line Number:	17
Number of Administered Channels:	10
Number of Channels in Use:	0
Administered Compression:	G711 A, G729 A, G722
Silence Suppression:	Off
Media Stream:	RTP
Layer 4 Protocol:	UDP
SIP Trunk Channel Licenses:	256 ● 0%
SIP Trunk Channel Licenses in Use:	0
SIP Device Features:	

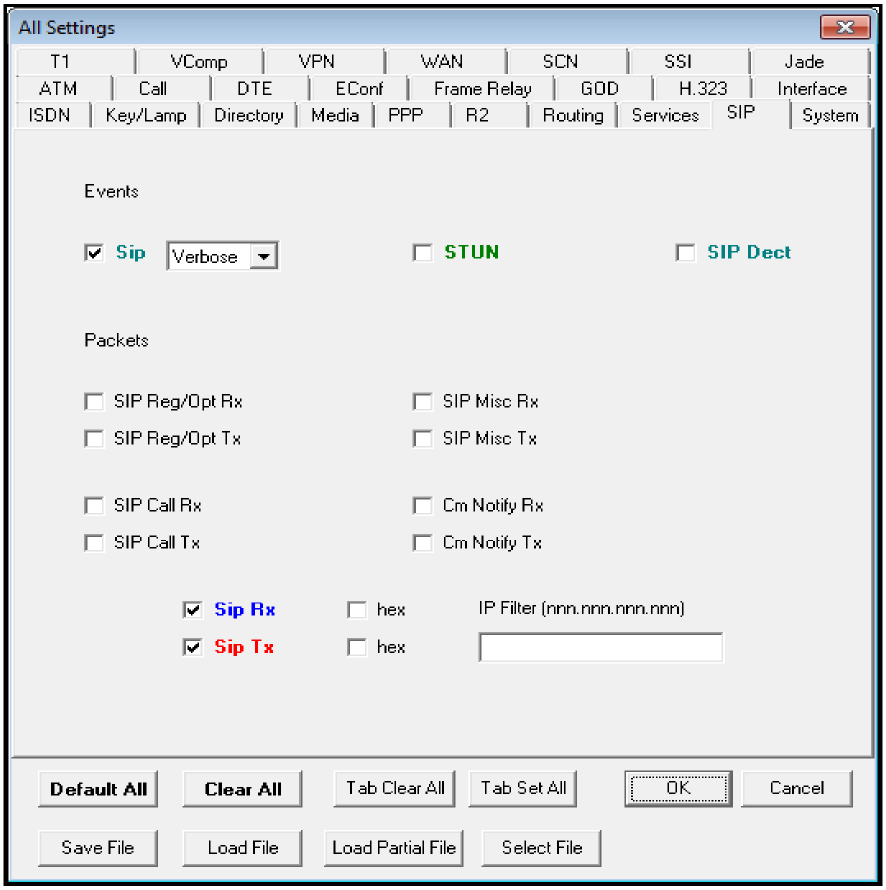
Ch…	U..	Call Ref	Cu…	Time in …	Re…	C…	Co…	Ca…	Other Part…	Dir…	Ro…	Re…	Re…	Tr…	Tr…
1			Idle	01…											
2			Idle	03…											
3			Idle	03…											
4			Idle	03…											
5			Idle	03…											
6			Idle	03…											
7			Idle	03…											
8			Idle	03…											
9			Idle	03…											
10			Idle	03…											

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 Reg/Opt Tx
☐ SIP Call Rx
☐ SIP Call Tx
☐ SIP Misc Rx
☐ SIP Misc Tx
☐ Cm Notify Rx
☐ Cm Notify Tx

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

9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office Release 11.1 to Keyyo Communications SIP Trunking Service. Keyyo Communications 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*, Release 11.1 FP1, Issue 16, February 2021
[2] *Administering Avaya IP Office with IP Office Manager*, November 15, 2021.
[3] *Administering Avaya IP Office with Web Manager*, August 2021.

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

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.