

#### **DevConnect Program**

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

## Abstract

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

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

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

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program.

## **Table of Contents**

1. Introduction	
2. General Test Approach and Test Results	
2.1. Interoperability Compliance Testing	4
2.2. Test Results	5
2.3. Support	6
3. Reference Configuration	6
4. Equipment and Software Validated	9
5. Avaya IP Office Primary Server Configuration	10
5.1. Licensing	12
5.2. System Settings	
5.2.1. System – LAN2 Tab	
5.2.2. System - Telephony Tab	17
5.2.3. System - VoIP Tab	
5.3. IP Route	
5.4. SIP Line	
5.4.1. Creating a SIP Trunk from an XML Template	
5.4.2. SIP Line – SIP Line Tab	
5.4.3. SIP Line - Transport Tab	
5.4.4. SIP Line – SIP Credentials Tab	
5.4.5. SIP Line – Call Details Tab	
5.4.6. SIP Line - VoIP Tab	
5.4.7. SIP Line – SIP Advanced Tab	
5.5. IP Office Line – Primary Server	
5.6. Incoming Call Route	
5.7. Outbound Call Routing	
5.7.1. Short Codes and Automatic Route Selection	
5.8. Save IP Office Primary Server Configuration	
6. Avaya IP Office Expansion System Configuration	
6.1. Physical Hardware	
6.2. LAN Settings	
6.3. IP Route	
6.4. IP Office Line – IP500 V2 Expansion System	
6.5. Short Codes	
6.6. Automatic Route Selection – ARS	
6.7. Save IP Office Expansion System Configuration	
7. WorldNet Telecommunications SIP Trunking Service Configuration	
8. Verification Steps	
8.1. IP Office System Status	
8.2. Monitor	
9. Conclusion	
10. Additional References	

## 1. Introduction

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

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

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

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

# 2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the WorldNet Telecommunications network via the public Internet, as depicted in **Figure 1**, and exercise the features and functionalities listed in **Section 2.1**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

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

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

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

HG; Reviewed: SPOC 10/11/2024 Avaya DevConnect Application Notes ©2024 Avaya Inc. All Rights Reserved. 3 of 52 WorldNet\_IPO120 Transport Layer Security (TLS) and Secure Real-Time Transport Protocol (SRTP) were used inside of the enterprise (private network side).

## 2.1. Interoperability Compliance Testing

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

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

Items not supported or not tested included the following:

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

## 2.2. Test Results

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

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

### 2.3. Support

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

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

## 3. Reference Configuration

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

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

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

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

In the sample configuration, the Primary server runs the Avaya IP Office Server Edition Linux software. Avaya Voicemail Pro runs as a service on the Primary Server. The LAN1 port of the Primary Server is connected to the enterprise LAN. The LAN2 port was used to connect to the public network.

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

IP endpoints at the enterprise included Avaya 1100 Series IP Deskphones (with SIP firmware), Avaya J100 Series IP Deskphones (with SIP and H.323 firmware), Avaya Workplace Client for Windows (SIP), Avaya Digital and Analog Deskphones. IP endpoints were registered to the Primary Server; non-IP endpoints (analog and digital) were registered to the Expansion Systems. The site also has a Windows PC running Avaya IP Office Manager to configure and administer

HG; Reviewed: SPOC 10/11/2024 Avaya DevConnect Application Notes ©2024 Avaya Inc. All Rights Reserved. 6 of 52 WorldNet\_IPO120 the system. Mobile Twinning is configured for some of the IP Office users so that calls to these user's extensions will also ring and can be answered at the configured mobile phones.

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

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

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

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

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

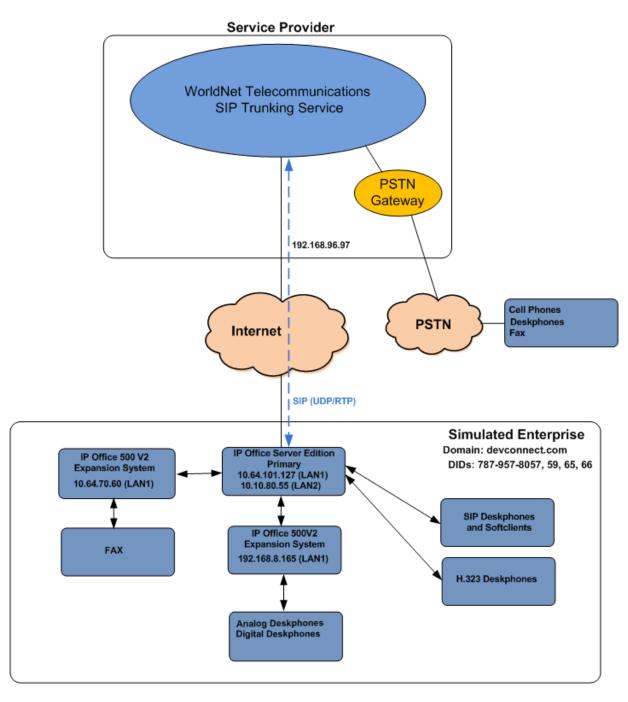


Figure 1: Avaya DevConnect lab Configuration

# 4. Equipment and Software Validated

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

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

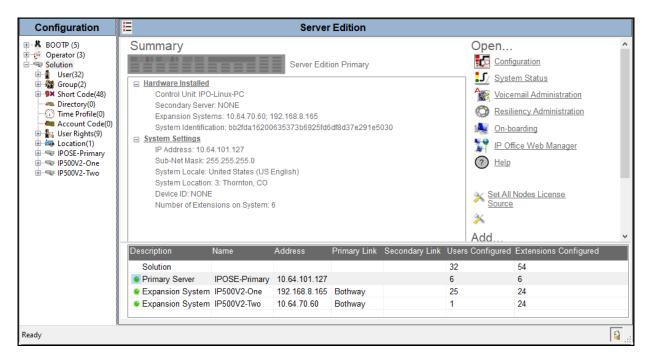
**Note**: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition. IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints.

# 5. Avaya IP Office Primary Server Configuration

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

📶 Avaya IP Office Manager for Server Edition			_		×
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Configuration					
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🖀 Select IP Office			_	· 🗌	×
Name IP Address Type	e Version	Edition			
Server Edition 12.0 IPOSE-Primary 10.64.101.127 IPO-	Linux-PC 12.0.0.0 build 55	Server (Primary) Select			
TCP Discovery Progress					
Unit/Broadcast Address					
10.64.101.127 ~ Refresh		Ε	ОК	Canc	el

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.



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.

Configuration					📥 - 🖻	$\times$	√   <
🗄 📲 BOOTP (5)	License Remote Server						
Operator (3)     Solution     User(32)     Group(2)     Directory(0)     Time Profile(0)     Account Code(0)     User Rights(9)     Location(1)	License Mode License Normal Licensed Version 12.0 PLDS Host ID PLDS File Status Valid Select Licensing Valid						,
IPOSE-Primary     E	Feature	Instances	Status	Expiration Date	Source	^	Add
⊞ रि Line (3)	Customer Service Agent	20	Dormant	Never	PLDS Nodal	_	
🗄 🖘 Control Unit (9)	Customer Service Supervisor	20	Dormant	Never	PLDS Nodal		Remov
🗄 🛷 Extension (6)	Avaya IP endpoints	1000	Valid	Never	PLDS Nodal		
⊞¶ User (7) ∰ Group (0)	SIP Trunk Channels	256	Valid	Never	PLDS Nodal		
	IP500 Universal PRI (Additional cha	100	Obsolete	Never	PLDS Nodal		
Service (0)	CTI Link Pro	1	Valid	Never	PLDS Nodal		
🗉 🝺 Incoming Call Route (2)	Wave User	16	Obsolete	Never	PLDS Nodal		
IP Route (3)	3rd Party IP Endpoints	1000	Valid	Never	PLDS Nodal		
License (25)	Server Edition	150	Valid	Never	PLDS Nodal		
	UMS Web Services	1000	Valid	Never	PLDS Nodal		
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🗄 🖏 IP500V2-Two				(	OK Cance	1	Help

## 5.2. System Settings

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

### 5.2.1. System – LAN2 Tab

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

To view or configure the LAN2 port IP address and subnet mask, select the LAN2 $\rightarrow$  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., **255.255.128**.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit (not shown).

Configuration	E IPOSE-Primary	🚔 - 🔤   🗙   🗸   <   >
Configuration BOOTP (5) Solution Solut		Directory Services System Events Sh • •
∎		OK Cancel Help
Ready		🔒 🔐

## 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.

Configuration	E IPOSE-Primary	iii - □   ×   ✓   <   >
EOOTP (5)	System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events	SMTP SMDR VolP
🕀 🧐 Operator (3)		
i⊟	LAN Settings VolP Network Topology	
	H.323 Gatekeeper Enable	~
Email Group(2) Email Short Code(48)	Auto-create Extension Auto-create User H.323 Remote Extension Enable	
Directory(0)		-
Time Profile(0)	H.323 Signaling over TLS Disabled v Remote Call Signaling Port 1720	8
Account Code(0)		-
🗄 📲 User Rights(9)		
🕀 🏧 Location(1)	SIP Trunks Enable	
IPOSE-Primary	SIP Registrar Enable	
🖃 🖏 System (1)		
IPOSE-Primary	Auto-create Extension/User SIP Remote Extension Enable Allowed SIP User Ag	ents Block blacklist only
⊞ 17 Line (3)	SIP Domain Name	
E. Control Unit (9)		
⊞…≪ Extension (6) ⊞…⊉ User (7)	SIP Registrar FQDN	
Group (0)	✓ UDP UDP Port 5060 🚔 Remote UDP P	ort 5060 🐥
Short Code (4)	V ODP ODP Port 5000 V Remote ODP P	
Service (0)	Layer 4 Protocol 🗹 TCP TCP Port 5060 🌩 Remote TCP Po	ort 5060 🜲
🗈 🔁 Incoming Call Route (2)	TLS TLS Port 5061 🚔 Remote TLS Po	ort 5061 🐥
🗎 🖬 IP Route (3)		
License (25)	Challenge Expiration Time (sec) 10 🚖	
Auto Attendant (0)	<	
ARS (1)     Conference (0)		
E- Location (1)		
Authorization Code (0)	OK	Cancel Help

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.

Configuration	E IPOSE-Primary 🔐 📲	◙   ×   ✔   <   >
BOOTP (5)	System         LAN1         LAN2         DNS         Voicemail         Telephony         Directory Services         System Events         SMTP	SMDR VolP · ·
Solution	LAN Settings VolP Network Topology	
	□ TLS TLS Port 5061 + Remote TLS Port 506	1
Ender Short Code(48)		•
	Challenge Expiration Time (sec)	
🗄 📲 User Rights(9)	RTP	
⊞	Port Number Range	
🖶 🖘 System (1)	Minimum 40750 🗭 Maximum 50750 牵	
IPOSE-Primary		
। ●···◆ Control Unit (9)	Port Number Range (NAT)	
🗄 🛷 Extension (6)	Minimum 40750 ਦ Maximum 40750 ਦ	
⊕ <b>1</b> User (7)		
🗈 🖗 Short Code (4)	Enable RTCP Monitoring on Port 5005	
····· 🚱 Service (0) ····· 🏠 Incoming Call Route (2)	RTCP collector IP address for phones 0 . 0 . 0 . 0	
🖶 📶 IP Route (3)	Keepalives	
<b>%</b> License (25) <b>i</b> Auto Attendant (0)	Scope RTP-RTCP V Periodic timeout 30	
acto Attendant (0) ⊕`★ ARS (1)	Initial keepalives Enabled ~	
→		
Authorization Code (0)		
	DiffServ Settings	
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	46 🔹 DSCP 46 🖨 Video DSCP 63 🖨 DSCP Mask 34 🖨 SIG DSCP	
	- DHCP Settings-	
	Primary Site Specific Option Number (SSON)	
	Secondary Site Specific Option Number (SSON)	
	VLAN Not Present V	
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	<	>
	ОК	Cancel Help

• Click **OK** to commit.

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## 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.

Configuration	E IPOSE-Primary 💣 📲	$\times   \checkmark   <   >$
Configuration BOOTP (5) Configuration Config	System         LAN1         LAN2         DNS         Voicemail         Telephony         Directory Services         System Events         SMTP         SMDR         VolP         Contact Center         Avaya Cloud           LAN Settings         VolP         Network         Topology         Network         Topology         Network         Topology         Network         Network         Topology         Network         Topology         Network         Network <td></td>	
	NAT     SIP Registrar Public Ports       Binding Refresh Time (sec)     60       Public IP Address     10     10     80     55       TLS     5061	
	SBC       Public IP Address (IPv4)       0 . 0 . 0 . 0       OK       SBC Registrar Public Ports         Public IP Address (IPv6)       D       D       TCP       TCP       TCP         FQDN       D       D       D       TLS       TLS       TLS       OK       Cancel	el Help

#### 5.2.2. System - Telephony Tab

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

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

#### 5.2.3. System - VoIP Tab

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

#### 5.2.3.1 VoIP - VoIP Tab

Select the **VoIP**  $\rightarrow$  **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.

Configuration	E IPOSE-Primary	≝ - 🖻   🗙   🗸   <   >
BOOTP (5)     Operator (3)     Solution     User(32)     Group(2)     Short Code(48)     Directory(0)     Control Unit (9)     System (1)     User Rights(9)     System (1)     POSE-Primary     POSE-Primary     POSE-Primary     Control Unit (9)     System (1)     POSE-Primary     POSE-Primary     System (1)     POSE-Primary     Pose (1)     POSE-Primary     Pose (1)     POSE-Primary     Pose (2)     Pose		VolP Contact Center • •
Ready		<b>a</b>

**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).

### 5.2.3.2 VoIP – VoIP Security Tab

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

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

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

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

To configure the use of SRTP, select the VoIP  $\rightarrow$  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.

Configuration	×=		IPOSE-Prima	ry		<b>- -</b>	$\times   \checkmark$	<   >
Configuration  Config	System LAN1 VolP VolP See Default Extensio	LAN2 DNS curity Access Co on Password Extension Passw Media Security Encryptions Authentication Replay Protection SRTP Window Si Crypto Suites SRTP_AES_CI	Voicemail Telephony ontrol Lists ord Options RTP RTCP RTCP RTCP on ize 64 M_128_SHA1_80	-	System Events			
Auto Attendant (0) ARS (1) Location (1) Authorization Code (0) Figure IP500V2-One Figure IP500V2-Two	– Calling Numbe Incoming Calls Validation Pres	Handling Allow	v All	~	OK	Can	el	∽ Help

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### 5.3. IP Route

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

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

- Set the **IP Address** and **IP Mask** to **0.0.0.0** to make this the default route.
- Set **Gateway IP Address** to the IP address of the gateway/router used to route calls to the public network, e.g., **10.10.80.1**.
- Set **Destination** to **LAN2** from the pull-down menu.
- Click **OK** to commit.

Configuration		0.0.0.0		📸 • 🔛   🗙   🗸   <   >
BOOTP (5)	IP Route			
Operator (3)     Solution	IP Address	0	0 , 0 , 0 , 0	
User(32)				
🕀 🙀 Group(2)	IP Mask	0	0 , 0 , 0 , 0	
Short Code(48)	Gateway IP Address	10	0 · 10 · 80 · 1	
Directory(0) Time Profile(0)	Destination		N2	~
Account Code(0)			11 VL	
🗄 📲 User Rights(9)	Metric	0		▲ ▼
Location(1)				
i i i i i i i i i i i i i i i i i i i				
IPOSE-Primary				
⊞ fr Line (3)				
🗄 📲 User (7)				
Short Code (4)				
Incoming Call Route (2)				
IP Route (3)				
10.0.00 10.64.70.0				
1 192.168.8.0				
🔍 🍢 License (25)				
Auto Attendant (0)			ОК	Cancel Help

## 5.4. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and WorldNet Telecommunications. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Sections 5.4.1** to create the SIP Line from the template.

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

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

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

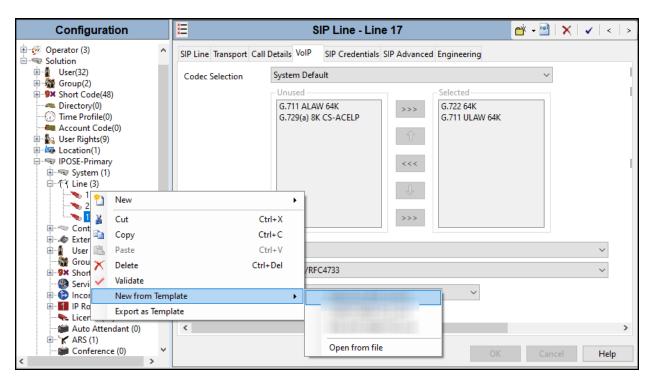
Alternatively, a SIP Line can be created manually. To do so, right-click on Line in the **Navigation** pane and select **New**  $\rightarrow$  **SIP Line**. Then, follow the steps outlined in **Sections 5.4.2** to **5.4.7**.

#### 5.4.1. Creating a SIP Trunk from an XML Template

DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

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

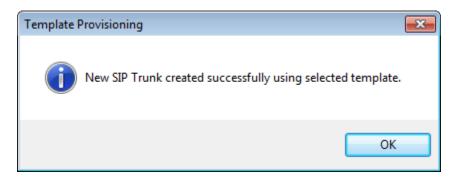
To create the SIP Trunk from the template, from the **Primary** server, right-click on **Line** in the Navigation Pane, then navigate to New  $\rightarrow$  New from Template $\rightarrow$ Open from file.



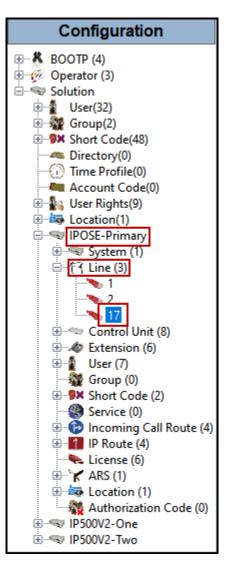
Navigate to the directory on the local machine where the template was copied and select the template.

忆 Open			Х
← → × ↑ 📙 « WoldNet Telecommuni	cations > App Notes > WorldNet IPO 12.0 > Template	✓ Õ Search Template	م
Organize 🔻 New folder			
E Pictures	^ Name	Date modified Type	
📕 Videos	WorldNet_IPO120.xml	9/4/2024 12:21 PM XML I	Document
🎬 Avaya eSOE (C:)			
TOSHIBA (D:)			
👝 TOSHIBA (E:)	~ <		>
File name:		✓ Template Files (*.xml)	~
		Open	Cancel

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



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



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

#### 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:

- Leave the **ITSP Domain Name** blank. Note that if this field is left blank, then IP Office inserts the ITSP Proxy Address from the Transport tab as the ITSP Domain in the SIP messaging.
- Under Local Domain Name enter the public IP address assigned to the LAN2 interface (Section 5.2.1).
- Verify that **In Service** box is checked, the default value. This makes the trunk available to incoming and outgoing calls.
- Verify that **Check OOS** box is checked, the default value. IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Verify that **Refresh Method** is set to **Auto**.
- Verify that **Timer** (sec) is set to **On Demand**.
- Under Redirect and Transfer, set Incoming Supervised REFER and Outgoing Supervised REFER to Never (refer to Section 2.2).
- Click **OK** to commit.

Configuration	E	SIP Line - Line 17		📸 • 🔛   🗙   •   <   >
Operator (3)	SIP Line Transport Call Details VolP	SIP Credentials SIP Advanced Engi	neering	
	Line Number	17	In Service	
Short Code(48)	ITSP Domain Name		Check OOS	
Directory(0) 	Local Domain Name	10.10.80.55		
Account Code(0)	URI Type	SIP URI V	Session Timers	
E	Location	Cloud ~	Refresh Method	Auto ~
iPOSE-Primary i∃≪ System (1)			Timer (sec)	On Demand 韋
□(†-7 Line (3)	Prefix			
**• 2	National Prefix			
E	International Prefix			
±	Country Code		Redirect and Transfer	
	Name Priority	System Default ~	Incoming Supervised REFER	Never ~
Short Code (4)	Description	Service Provider	Outgoing Supervised REFER	Never ~
••• Service (0) ••• P Incoming Call Route (2)	Description		Send 302 Moved	
IP Route (3)     License (25)			Temporarily Outgoing Blind REFER	
ARS (1)     Conference (0)				
⊕ Location (1)				
Authorization Code (0)	<			>
				OK Cancel Help
< >				Cancer Help

#### 5.4.3. SIP Line - Transport Tab

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

- Set the **ITSP Proxy Address** to the public IP address (signaling IP address) of WorldNet's SIP Proxy server (**192.168.96.97**) as shown in **Figure 1**. This information should be provided by WorldNet.
- Set Layer 4 Protocol to UDP.
- Set Use Network Topology Info to None (see note below).
- Set the **Send** and **Listening Ports** to **5060**.
- Default values may be used for all other parameters.
- Click **OK** to commit.

Configuration	SIP Line - Line 17	📸 • 🔛   🗙   🗸   <   >
Configuration	SIP Line - Line 17         SIP Line Transport Call Details VolP SIP Credentials SIP Advanced Engineering         ITSP Proxy Address       192.168.96.97         Network Configuration       Layer 4 Protocol         Layer 4 Protocol       UDP         Use Network Topology Info       None         Explicit DNS Server(s)       0       0       0         Calls Route via Registrar       Separate Registrar	
Location (1)     Authorization Code (0)     Posov2-One     Posov2-Two		OK Cancel Help

**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**  $\rightarrow$  **LAN1 (or 2)**  $\rightarrow$  **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 credential provided by WorldNet for SIP Trunk registration.
- Leave Authentication Name blank.
- Leave the **Contact** blank.
- For **Password** and **Confirm Password**, add the password credential provided by WorldNet for SIP Trunk registration.
- Set **Expiry** (mins) to a value acceptable to the enterprise. This setting defines how often registration with Clearcom is required following any previous registration. For the compliance test **60** minutes was used. This value should be chosen in consultation with the service provider.
- Verify that **Registration required** is checked.
- Click the OK to commit.

Configuration	SIP Line - Line 17*	📸 - 🔛   🗙   🗸   <   >
	SIP Line Transport Call Details VoIP SIP Credentials SIP Advanced Engineering	
⊕…∰ Operator (3) ⊡…≪ Solution	Index User Name Authentication Name Contact Expiration (mins) Register	Add
🗄 📲 User(32)	1 user123 60 True	
🗄 📲 Group(2)		Remove
Short Code(48) Directory(0)		
Time Profile(0)		Edit
Account Code(0)		
🗄 📲 User Rights(9)		
Eccation(1)		
IPOSE-Primary		
□ 17 Line (3)		
* 17 ⊞≪ Control Unit (9)		
Extension (6)		
User (7)		
📲 Group (0)		
Short Code (4)	Edit SIP Credentials	
Service (0) for the coming Call Route (2)	User name user123	ОК
IP Route (3) License (25)	Authentication Name	Cancel
Auto Attendant (0)	Contact	
Conference (0)	Password	
🗄 🚟 Location (1)	Confirm Password	
	Expiration (mins) 60	
	Registration required	
		~
		OK Cancel Help

#### 5.4.5. SIP Line – Call Details Tab

Select the **Call Details** tab, and then click the **Add...** button (not shown) and the screen shown below will appear. To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button. In the example screen below one new entry was added to handle incoming and outgoing calls, both.

- Associate this entry to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic from this line. For the compliance test outgoing group **17** was used. Leave the **Incoming Group** field as 0.
- Under **Credentials**, select **1: user123** from the pull-down menu (this field will default to the **User name** used under the **SIP Credentials** tab in **Section 5.4.4**).
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Check the **P** Asserted **ID** and **Diversion Header**.
- Set the Local URI, Contact, P Asserted ID and Diversion Header fields to the values shown in the screenshot below.
- Set all remaining fields as shown on the screenshot below.
- Click **OK** to Commit.

🐮 SIP Line - 17   Call Details   SIP URI X								
New URI								
Incoming Group 17 V Max Sessions 10								
Outgoing Group	17	~						
Credentials	1: us	er123 v						
		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	$\checkmark$	Auto ~	Auto ~	Caller ~	Original Caller	Called	~	
P Preferred ID		None ~	None ~	None ~	None	None	$\sim$	
Diversion Header	$\checkmark$	Auto ~	Auto ~	None ~	Caller	None	~	
Remote Party ID		None ~	None ~	None ~	None	None	$\sim$	
	OK Cancel Help							

#### 5.4.6. SIP Line - VoIP Tab

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

- The Codec Selection was configured using the System Default option. The buttons allow setting the specific order of preference for the codecs to be used on the SIP Line, as shown. WorldNet Telecommunications supports codecs G.722 64K and G.711 ULAW 64K for audio.
- Select G.711 for Fax Transport Support (Refer to Section 2.1).
- Set the **DTMF Support** field to **RFC2833/RFC4733**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Set the **Media Security** field to **Disabled**.
- Check the **Re-invite Supported** box.
- Check the **PRACK/100rel Supported** box.
- Default values may be used for all other parameters.
- Click the **OK** to commit.

Configuration	X	SIP Line - Line 17	<mark>☆ - ×   ×   ×   ×   &gt;</mark>
BOOTP (5)	SIP Line Transport Call	Details VolP SIP Credentials SIP Advanced Engineering	
⊕ • • • • • • • • • • • • • • • • • • •	Codec Selection	System Default         Selected           Unused         G.711 ALAW 64K         Selected           G.729(a) 8K CS-ACELP         G.711 ULAW 64K         G.711 ULAW 64K           Image: Selected default of the selected defa	<ul> <li>Local Hold Music</li> <li>Re-invite Supported</li> <li>Codec Lockdown</li> <li>Allow Direct Media Path</li> <li>Force direct media with phones</li> <li>PRACK/100rel Supported</li> </ul>
	Fax Transport Support	G.711	~
🖓 Group (0)	DTMF Support	RFC2833/RFC4733	~
	Media Security	Disabled ~	
IP500V2-One IP500V2-Two			OK Cancel Help

**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).

#### 5.4.7. SIP Line – SIP Advanced Tab

In the **Addressing** area:

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

In the **Identity** area:

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

Configuration	E	P Line - Line 17	📸 - 🔛   🗙   🗸   <   >
BOOTP (5)     Operator (3)     Solution     User(32)     Solution     Directory(0)     Directory(0)     Code(48)     Code(48)     Code(0)     Solution(1)     User Rights(9)     Solution(1)     DOSE-Primary     Grow System (1)	SIP Line Transport Call Details VolP       SIP Credentials       SI         Addressing       Addressing       Addressing         Association Method       By Source IP address         Call Routing       To Header       V         Use P-Called-Party          Suppress DNS SRV	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 Surtage	
	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	Preservation Indicate HOLD Indicate HOLD Call Control Call Initiation Timeout (s) 4 Call Queuing Timeout (s) 4 Call Queuing Timeout (s) 5 Service Busy Response a on No User Responding 2 Send Attice of CAC Learning 4	4 3 486 - Busy Here 408-Request Timeout Allow Voicemail
< >			OK Cancel Help

## 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.

Configuration	X.	IP Office Line - Line	1		📸 • 🔤   🗙   🗸   <   >
BOOTP (3)	Line Short Codes VolP Settin	ngs			
ie∲⁄ Operator (3) ie≪ Solution ie∰ User(32)	Line Number	1		Telephone Number	^ `
Group(2)	Transport Type	WebSocket Server	$\sim$	Prefix	
Directory(0)	Networking Level	SCN	$\sim$	Outgoing Group ID	99999
	Security	Unsecured	$\sim$	Number of Channels	250
⊕ <b>‱</b> User Rights(9) ⊕				Outgoing Channels	250
ie≪ IPOSE-Primary ie≪ System (1)	Gateway				
=77 Line (3)	Address	192 · 168 · 8 · 165			
	Location	3: Thornton, CO	$\sim$	SCN Resiliency Options	
E Control Unit (9)	Password	•••••		<ul> <li>Supports Resiliency</li> <li>Backs up my IP phones</li> </ul>	
⊞…≪ Extension (6) ⊕…⊉ User (7)	Confirm Password	•••••		Backs up my hunt grou	
				Backs up my voicemail	
Service (0)     Incoming Call Route (4)				Backs up my IP DECT p	hones
IP Route (4)	Description				
License (25)	Description				
ARS (1)					~
Location (1)     Authorization Code (0)	<				>
IP500V2-One IP500V2-Two				OK	Cancel Help

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

- Under Codec Selection verify System Default is selected (default value).
- Select G.711 for Fax Transport Support (refer to Section 2.1).
- Under Media Security verify Same as System (Preferred) is selected (default value).
- On the Advanced Media Security Options check Same As System.
- Click **OK** to commit.

Configuration	<u></u>	IP Office Line	- Line 1	📸 🕶 🛛 🗙 🛛 🖌 🗠
	Line Short Codes VolP S Codec Selection Fax Transport Support Call Initiation Timeout (s) Media Security		- Line 1  Selected  G.722 64K  G.711 ULAW 64K  Selected  Comparison  Selected  Selected Selec	<ul> <li>✓ Out Of Band DTMF</li> <li>✓ Allow Direct Media Path</li> </ul>
ARS (1)     Gonference (0)	<	Encryptions Authentication Replay Protection SRTP Window Size Crypto Suites SRTP_AES_CM_128_SHA1_80 SRTP_AES_CM_128_SHA1_32		OK Cancel Help

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

Configuration	1	7 7879578057	🚔 - 🔤   🗙   🗸   <   >
<ul> <li>BOOTP (3)</li> <li>Operator (3)</li> <li>Solution</li> <li>Solution</li> <li>Solution</li> <li>Solution</li> <li>Short Code(48)</li> <li>Directory(0)</li> <li>Time Profile(0)</li> <li>Account Code(0)</li> <li>User Rights(9)</li> <li>Location(1)</li> <li>User Rights(9)</li> <li>Location(1)</li> <li>IPOSE-Primary</li> <li>System (1)</li> <li>T? Line (3)</li> <li>Control Unit (9)</li> <li>Extension (6)</li> <li>User (7)</li> <li>Group (0)</li> <li>Service (0)</li> <li>Incoming Call Route (4)</li> <li>T 7879578055</li> <li>T 7879578065</li> <li>T 7879578066</li> <li>IP Route (4)</li> </ul>	Standard       Voice Recording       D         Bearer Capability       Line Group ID       Incoming Number         Incoming Sub Address       Incoming CLI         Locale       Priority         Tag       Hold Music Source         Ring Tone Override		
IP Route (4)     Auto Attendant (0)     Arts (1)     Conference (0)     Location (1)     Authorization Code (0)     IP500V2-One     IP500V2-Two		OK	Cancel Help
Ready			<b>a</b>

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

Configuration	×		17 7879	578057		📸 • 🗐   🗙   🗸   <   >
BOOTP (3) 	Standa	rd Voice Recording	Destination	s		
Solution		TimeProfile		Destination		Fallback Extension
🗄 📲 User (32)	•	Default Value		3042 Ext3042 H323	~	~
⊞¶¥ Group(2) ⊞¶¥ Short Code(48)						
Directory(0)						
Time Profile(0)						
····· <b>···</b> ■ Account Code(0) ⊕··· <b>§</b> User Rights(9)						
Location(1)						
E IPOSE-Primary						
⊞… रू System (1) ⊞… रि Line (3)						
Control Unit (9)						
Extension (6)						
🗄 📲 User (7)						
Short Code (4)						
Incoming Call Route (4) 17 7879578057						
17 7879578057						
17 7879578066						
⊡ IP Route (4)						
🗰 Auto Attendant (0)						
Conference (0)						
Authorization Code (0)						
					ОК	Cancel Help
⊞					U.S.	

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.

Configuration	XIX	9N: Dial		📸 - 🔤   🗙   🗸   <   >
	Short Code			
Solution	Code	9N		
i∰∰ User (32) ∰∰ Group(2)	Feature	Dial $\checkmark$		
Short Code(48)     Directory(0)	Telephone Number	Ν		
Time Profile(0) Account Code(0)	Line Group ID	50: Main 🗸 🗸		
user Rights(9) ⊡…∰ Location(1)	Locale	United States (US Englis $ \smallsetminus $		
🖃 🖘 IPOSE-Primary	Force Account Code			
●	Force Authorization Code			
⊞ <b>1</b> User (7)				
Group (0)				
<b>9</b> × *57 <b>-9</b> × *66*N#				
<b>9×</b> 8N				
9N 9N 9N 9N				
🗉 💮 Incoming Call Route (4)				
IP Route (4)     License (25)				
will Auto Attendant (0)				
ARS (1)     Conference (0)				
Location (1)				
Authorization Code (0) E			OK	Cancel Help
⊞			UK	Cancel Help

Avaya DevConnect Application Notes ©2024 Avaya Inc. All Rights Reserved. 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  $\rightarrow$  50: Main on the Navigation Pane and click Add (not shown). Configure the following parameters:

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

The following example shows the dial pattern for calls within North America.

Edit Short Code			
Code	1XXXXXXXXXXX		ОК
Feature	Dial	$\sim$	
Telephone Number	1N		Cancel
Line Group ID	17	~	
Locale	United States (US English)	$\sim$	
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**  $\rightarrow$  **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.

1	Send	l Multipl	le Configurations							-		×
		Select	IP Office	Change Mode	RebootTime	Incoming Call Barring	Outgoing Call Barring	Error Status	Progress			
	•		IPOSE-Primary	Merge 🗸 🗸	11:22 AM			8	0%			
								ОК	Cancel		Help	

# 6. Avaya IP Office Expansion System Configuration

Navigate to File  $\rightarrow$  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.

Configuration	😚 System Inventory
BOOTP (4) Grant Code(3) Group(2) Group(2) Group(2) Group(2) Group(0)	Server Edition Expansion System           Image: 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
<ul> <li>Fi Line (3)</li> <li>Control Unit (4)</li> <li>Extension (24)</li> <li>User (27)</li> <li>Group (1)</li> <li>Short Code (12)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Incoming Call Route (1)</li> <li>WAN Port (0)</li> <li>Firewall Profile (1)</li> <li>Firewall Profile (1)</li> <li>Firewall Profile (2)</li> <li>License (2)</li> <li>Tunnel (0)</li> <li>ARS (2)</li> <li>Location (1)</li> <li>Authorization Code (0)</li> </ul>	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.

Configuration	E	IP 500 V2	<u> - 1</u>	≞   ×   ✔   < [ > ]
Configuration     Configuration     Configuration     Configuration     Control User(32)     Concord Unit (4)     Control Unit (4)	Unit Device Number Unit Type Version Serial Number Unit IP Address Interconnect Number Module Number	1 IP 500 V2 00e00706530f 192.168.8.165 0 Control Unit		
Authorization Code (0)			ОК С	iancel Help

#### 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  $\rightarrow$  LAN Settings tab on the Details pane, and enter the following:

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

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

#### 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** 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.

Configuration	***	0.0.0.0				
н <b>ВООТР</b> (4)	IP Route					
⊕	IP Address		0.0.	0	. 0	1
	IP Mask		0 · 0 ·	0	. 0	
Short Code(48)     Directory(0)	Gateway IP Ac	ldress	192 - 168 -	8	· 1	
Time Profile(0)	Destination		LAN1			
🗄 📲 User Rights(9)	Metric		0			
⊞…‱ Location(1) ⊞…≪ IPOSE-Primary			Proxy ARP			
iP500V2-One i±≪ System (1)						
● 行了 Line (3) ● 一句 Control Unit (4)						
Extension (24)						
🕀 🎆 Group (1)						
· · · · · · · · · · · · · · · · · · ·						
WAN Port (0)						
□ 1 IP Route (4)						
10.64.101.0						
192.168.8.0 192.168.99.0						
····· <b>%</b> License (2) ····· <b>i</b> i Tunnel (0)						
Authorization Code (0)						
i⊞						

### 6.4. IP Office Line – IP500 V2 Expansion System

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

Configuration	12	IP	Offic	ce Line - Line 17*		
BOOTP (4)	Line Short Codes VolP Setti	ngs T38 Fax				
ianian Solution ianian User(32)	Line Number	17		Telephone Number		
⊕∰ Group(2) ⊕¶≭ Short Code(48)	Transport Type	WebSocket Client	$\sim$	Prefix		
Directory(0)     Time Profile(0)	Networking Level	SCN	$\sim$	Outgoing Group ID	99999	
Account Code(0)	Security	Medium	$\sim$	Number of Channels	250	•
●劉 User Rights(9) ●				Outgoing Channels	250	* *
IPOSE-Primary	Gateway					
। ●… <sup>…</sup> 气 Line (3)	Address	10 · 64 · 101 · 127		Port	443	▲ ▼
- 🖝 1	Location	3: Thornton, CO	$\sim$	SCN Resiliency Options		
2	Password	•••••		Supports Resiliency		
	Confirm Password	•••••		Backs up my IP phones Backs up my hunt group		
🗉 📲 User (27)				Backs up my hunt group Backs up my IP DECT photogram		
⊕∰ Group (1) ⊕					ones	
	Description					
🗉 🈳 Incoming Call Route (1)						
······································						
⊞ IP Route (4) ► License (2)						
Authorization Code (0)						

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

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

Configuration	X III	IP Office Line - Line 17		📸 • 🔤   🗙   •   <   >
н ВООТР (3)	Line Short Codes VolP S	ettings T38 Fax		
<ul> <li>BOOTP (3)</li> <li>Operator (3)</li> <li>Solution</li> <li>User (32)</li> <li>Directory(0)</li> <li>Time Profile(0)</li> <li>Account Code(0)</li> <li>User Rights(9)</li> <li>Location(1)</li> <li>POSE-Primary</li> <li>IPSOV2-One</li> <li>System (1)</li> <li>1</li> <li>1</li> <li>2</li> <li>17</li> <li>Control Unit (4)</li> <li>Extension (24)</li> <li>User Rights(0)</li> <li>Service (0)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Plonoming Call Route (1)</li> <li>Ple Route (2)</li> <li>License (2)</li></ul>	Line Short Codes VolP S Codec Selection Fax Transport Support Call Initiation Timeout (s) Media Security	Custom         Selected           G.711 ALW 64K         >>>           G.722 64K         >>>           G.729(a) 8K CS-ACELP         Image: Comparison of the second se	Same As System	<ul> <li>VoIP Silence Suppression</li> <li>Out Of Band DTMF</li> <li>Allow Direct Media Path</li> </ul>
			OK	Cancel Help

#### 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.

Configuration		9N: Dial		📸 + 🗐	X   ✓   <   >
	Short Code				
<ul> <li>Operator (3)</li> <li>Solution</li> </ul>	Code	9N			
i∎…i User (32) i∎…i group(2)	Feature	Dial	~		
Short Code(48)     Directory(0)	Telephone Number	N			
Time Profile(0)	Line Group ID	51: To-Primary	~		
🗄 📲 User Rights(9)	Locale	United States (US English)	~		
	Force Account Code				
	Force Authorization Code				
⊞…रीने Line (3)					
i≘…1 User (27) i∃…1 Group (1)					
Short Code (12)					
<b>9×</b> *29 <b>9×</b> *39					
<b>9×</b> *40 <b>9×</b> *41					
<b>9</b> × *42					
<b>9×</b> *43 <b>9×</b> *44					
<b>9</b> × *66*N# <b>9</b> × *9000*					
<b>9×</b> *91N;					
9≭ *92N; 9× 9N					
🖶 💑 RAS (1) 🖶 🍅 Incoming Call Route (1)					
WAN Port (0)					
IP Route (2)					
<b>%</b> License (2) <b>ii</b> i Tunnel (0)					
Authorization Code (0)			OK	Cance	Help
Ready					 

### 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**).

Configuration	H		٦	To-Primary		
BOOTP (4)	ARS					
Solution	ARS Route ID	51		Secondary Dial tone		
<ul> <li>Group(2)</li> <li>Short Code(48)</li> <li>→ Directory(0)</li> </ul>	Route Name	To-Primary		SystemTone	$\sim$	
Time Profile(0)	Dial Delay Time	System Default (4)	* *	Check User Call Barring		
User Rights(9)     Location(1)     Source IPOSE-Primary	Description					
ि-ज्ज् IP500V2-One अञ्च System (1) ख्र−र्नर Line (3)	In Service	☑		Out of Service Route	<none></none>	$\checkmark$
· ← ← Control Unit (4) · ← ← ↓ Extension (24) · ← ↓ User (27)	Time Profile	<none></none>	$\sim$	Out of Hours Route	<none></none>	~
in ∰ Group (1) in ∰ Short Code (12)		Ļ				
	Code	Telephone Number	Feature	Line Group ID		Add
Incoming Call Route (1)     WAN Port (0)	N	9N	Dial	99999		Remove
						Edit
License (2)						
Authorization Code (0)     IP500V2-Two		Ļ				
'⊞	Alternate Route Priority	Level 3	$\sim$			
		Ţ				Ļ
	Alternate Route Wait Tir	ne 30	* *	Alternate Route	<none></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  $\rightarrow$  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.

1	🖸 Send	Multipl	e Configurations							-		×
		Select ☑	IP Office	Change Mode	RebootTime	Incoming Call Barring	Outgoing Call Barring	Error Status	Progress			
			IP500V2-One	Merge ~	1:11 PM			8	0%			
	L						[	ОК	Cancel		Help	,

# 7. WorldNet Telecommunications SIP Trunking Service Configuration

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

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

WorldNet Telecommunications is responsible for the configuration SIP Trunking Service. The customer will need to provide the public IP address used to reach Avaya IP Office at the enterprise, the public IP address assigned to LAN2.

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

WorldNet Telecommunications will provide the following information:

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

# 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  $\rightarrow$  Programs  $\rightarrow$  IP Office  $\rightarrow$  System Status on the PC where IP Office Manager is installed, log in with the proper credentials.

AVAYA	IP Of	fice System Stat	tus
Help Exit About			
	Online Offline		
	Control Unit Address:	10.64.101.127	~
	SCN Gateway Address:	<none></none>	$\sim$
	Services Base TCP Port:	50804	
	Local IP Address:	Automatic	$\sim$
	User Name:	Administrator	
	Password:		
	Auto reconnect		
	Secure connection		Logon
	Websocket connectio	n	Lugon
IP Office System Status Ver1.0.0 build 20	9		

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

AVAYA		1	(P Of	fice S	Syst	em :	Stat	tus				
Help Snapshot LogOff Exi	t About											
<ul> <li>System</li> <li>Alarms (4)</li> <li>Extensions (4)</li> </ul>	Status	Utilization Sumn	nary	Alarms F	Registra	tion						_
Trunks (3)				SIP T	runk	Summar	У					
Line: 1	Line Servi	ce State:		In Ser	vice							
Line: 2	Peer Dom	ain Name:		sip://	.96	.97						
Line: 17 Active Calls	Resolved	Address:			96.97							
E Resources	Line Numb	er:		17								
E Voicemail	Number of	f Administered (	Channels:	10								
IP Networking	Number of	f Channels in Us	e:	0								
Locations	Administer	red Compression	n:	G722,	G711 M	u						
	Silence Su	ppression:		Off								
	Media Stre	Media Stream:			RTP							
	Layer 4 Pr	Layer 4 Protocol:			UDP							
	SIP Trunk	Channel Licens	es:	256	256 0%							
	SIP Trunk	Channel Licens	es in Use:	0								
	SIP Device	e Features:		UPDATE (Incoming and Outgoing)								
	Ch U	Call Cur Tin Ref in .		. C Co		Other Party		. Ro	Rec	Rec	Tra	Tra
	1	Idle 10										
	2	Idle 1 d										
	3		ł				_					
	5	Idle 10										
	6	Idle 10										
	7		ł									
	8		ł									
	9	Idle 1 o	ł									
	10											
	Trace	Trace All	Pause	e Ping	)	Call Deta	ils	Grace	eful Shu	tdown		
	Force Out	t of Service	Print	Sav	e As							
	-								2:36:5	7 PM	0	Online

#### 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  $\rightarrow$  Programs  $\rightarrow$  IP Office  $\rightarrow$  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.

All Settings		
T1 VComp VPN WAN SCN SSI Jade ATM Call DTE EConf Frame Relay GOD H.323 Interface ISDN Key/Lamp Directory Media PPP R2 Routing Services SIP System		
Events		
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	
I⊽ Sip Rx I⊽ Sip Tx	☐ hex IP Filter (nnn. ☐ hex	nnn.nnn.nnn)
Default All Clear All	Tab Clear All Tab Set All	OK Cancel
Save File Load File	Load Partial File Select File	•

# 9. Conclusion

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

The WorldNet Telecommunications SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks.

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

# 10. Additional References

This section references the documentation relevant to these Application Notes. Product documentation for Avaya IP Office, including the following, is available at: <a href="http://support.avaya.com/">http://support.avaya.com/</a>

- [1] Deploying IP Office Server Edition and Application Servers, Release 12.0, Issue 31, April 2024
- [2] IP Office Platform 12.0, Deploying Avaya IP Office Servers as Virtual Machines, June 2024
- [3] Avaya IP Office Platform Server Edition Reference Configuration Release 12.0, Issue 22, May 2024
- [4] IP Office Platform 12.0, Deploying an IP500 V2 IP Office Basic Edition System, Issue 41e, May 29, 2024
- [5] IP Office Platform 12.0, Deploying an IP500 V2 IP Office Essential Edition System, Issue 41e, May 29, 2024
- [6] Administering Avaya IP Office using Manager, Release 12.0, Issue 51.1.2, June 2024.
- [7] Administering Avaya IP Office with Web Manager, Release 12.0, Issue 46.1.1, May 2024.
- [8] Avaya IP Office Platform Feature Description, Release 12.0, Issue 21.1.1, May 2024.

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