



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring the Voice Carrier IntelliSIP Trunking Service with Avaya IP Office Release 8.1 – Issue 1.0**

### **Abstract**

These Application Notes describes the steps to configure Session Initiation Protocol (SIP) Trunking between Voice Carrier and Avaya IP Office Release 8.1.

The Voice Carrier IntelliSIP Trunking Service provides PSTN access via a SIP trunk between the enterprise and the Voice Carrier network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise. Voice Carrier is a member of the Avaya DevConnect Service Provider program.

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 steps to configure Session Initiation Protocol (SIP) Trunking between Voice Carrier and Avaya IP Office Release 8.1.

The Voice Carrier IntelliSIP Trunking Service referenced within these Application Notes is positioned for customers that have an IP-PBX or IP-based network equipment with SIP functionality, but need a form of IP transport and local services to complete their solution.

The Voice Carrier IntelliSIP Trunking Service will enable delivery of origination and termination of local, long-distance and toll-free traffic across a single broadband connection. A SIP signaling interface will be enabled to the Customer Premises Equipment (CPE).

## 2. General Test Approach and Test Results

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.

The general test approach was to connect a simulated enterprise site to the Voice Carrier IntelliSIP Trunking Service via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of Avaya IP Office and various Avaya endpoints.

The Voice Carrier IntelliSIP Trunking Service passed compliance testing with any observations or limitations described in **Section 2.2**.

### 2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- Use of UDP or TCP for the transport layer to the service provider
- Sending and receiving SIP OPTIONS queries to the service provider
- Incoming PSTN calls to SIP and H.323 telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from SIP and H.323 at the enterprise. All outgoing PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from soft clients. Avaya IP Office Video Softphone and Avaya Flare® Experience for Windows were tested.
- Various call types including: local, long-distance, outbound toll-free, international and local directory assistance.
- Codec G.711MU

- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation using DTMF for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and twinning
- G.711 pass-through fax
- REFER message for network call redirection

Inbound toll-free calls and 911 emergency calls are supported but were not tested as part of the compliance test.

The following items are not supported:

- G.729 codec
- Operator (dial 0) and operator-assisted ( dial 0 + 10 digits) services
- T.38 Fax

## 2.2. Test Results

Interoperability testing of the Voice Carrier IntelliSIP Trunking Service was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **Codec not locked down on outbound calls:** On outbound calls, Voice Carrier responds to the INVITE request with multiple codecs, instead of selecting one from the INVITE SDP list. IP Office uses the first compatible codec in the list. This behavior has no user impact. Calls were successful.
- **Inconsistent SIP headers on inbound Calling Party Number blocked calls:** On inbound Calling Party Number (CPN) blocked calls, Voice Carrier sends a SIP INVITE message containing two headers which contradict each other. The INVITE contains the Privacy = none header which indicates that the calling party number should not be blocked. The INVITE also contains the Remote-Party-ID header with parameter privacy=full which indicates that the calling party number should be blocked. In this scenario, IP Office blocked the calling party number as intended. The call was successful.
- **Blind transfer with Avaya 1100 IP Deskphones calling party number:** An Avaya 1100 Series IP Deskphone (SIP) places an outbound call to the PSTN, then performs a blind transfer of this call to another PSTN endpoint. In this scenario, the wrong calling party number is displayed on the second PSTN endpoint. The display shows the number dialed by the Avaya 1100 Series IP Deskphone for the initial outbound call including the short code (9) + 1 + 10 digits. The behavior is expected to be the same as with IP Office H.323 endpoints which is to display the transferring party number. This is an IP Office issue and is under investigation by the IP Office development team.

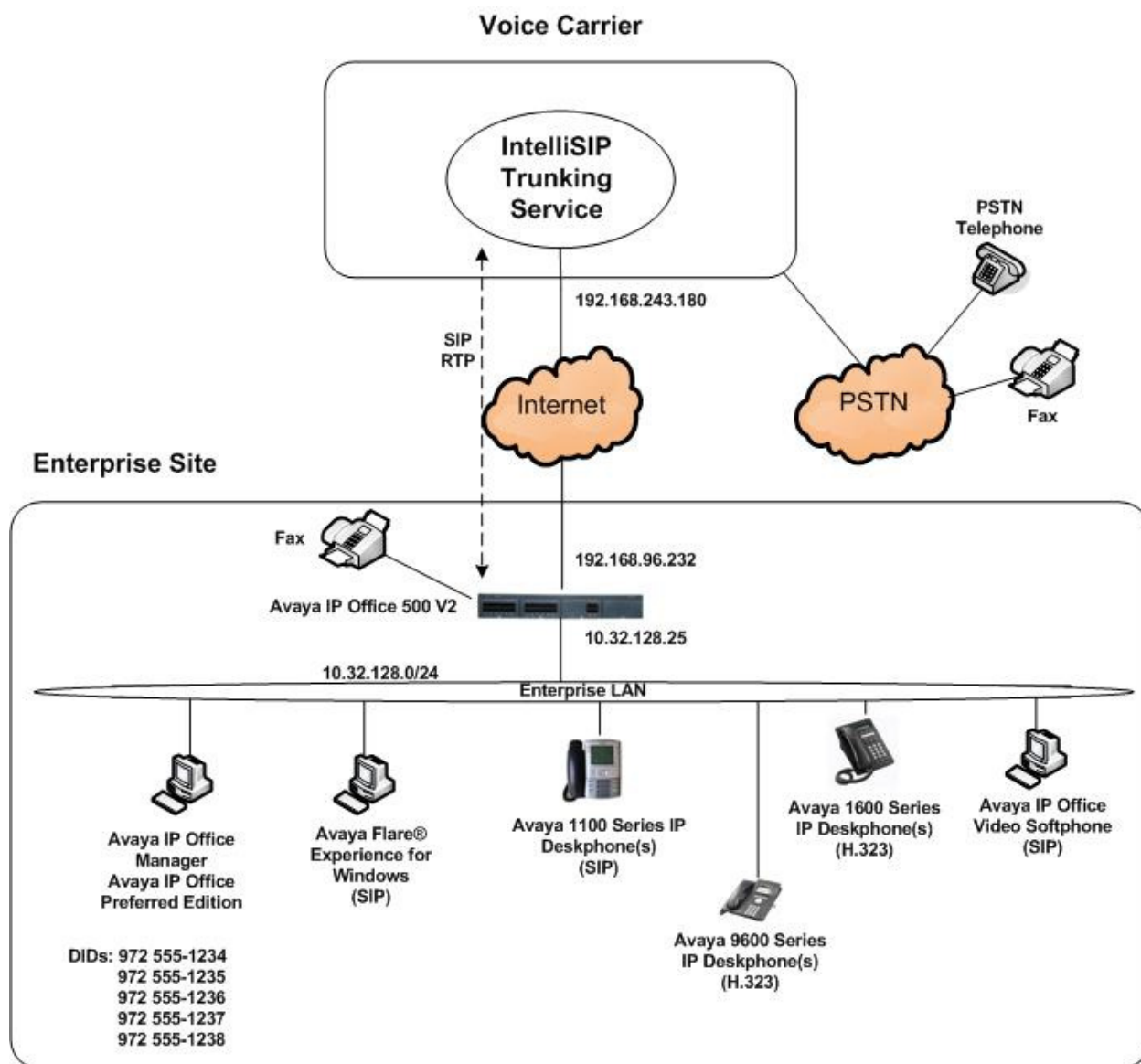
## 2.3. Support

For technical support on the Voice Carrier IntelliSIP Trunking Service, contact Voice Carrier using the Support links at [www.voicecarrier.com](http://www.voicecarrier.com) or by calling 1-888-830-6230.

## 3. Reference Configuration

**Figure 1** illustrates the sample configuration used for the DevConnect compliance testing. The sample configuration shows an enterprise site connected to the Voice Carrier IntelliSIP Trunking Service.

Located at the enterprise site is an Avaya IP Office 500 V2 with various endpoints. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public network. The site also has Avaya Preferred Edition for voicemail and Avaya IP Office Manager to configure the Avaya IP Office running on a single Windows 2003 Server.



**Figure 1: Avaya Interoperability Test Lab Configuration**

For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with numbers that cannot be routed over the PSTN.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Voice Carrier. The short code of 9 and any preceding 1 is stripped off by Avaya IP Office and the remaining 10 digits were sent unaltered to Voice Carrier. For outbound calls, Avaya IP Office sent 10 digits in all headers. For inbound calls, the Voice Carrier IntelliSIP Trunking Service sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the Avaya IP Office 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 SIP and RTP traffic between the service provider and the Avaya IP Office must be allowed to pass through these devices.

## 4. Equipment and Software Validated

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

Avaya Telephony Components	
Equipment	Software
Avaya IP Office 500 v2	8.1 (69)
Avaya IP Office Manager	10.1 (69)
Avaya IP Office Preferred Edition (Voicemail)	8.1 (9203)
Avaya 1140E IP Deskphone (SIP)	4.3 SP1 (04.03.12.00)
Avaya 1608 IP Deskphone (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP3 (1.3.3)
Avaya 9641G IP Deskphone (H.323) running Avaya one-X® Deskphone Edition	6.2 SP2 (6.2209)
Avaya IP Office Video Softphone (SIP)	3.2.3.48 (67009)
Avaya Flare® Experience for Windows	1.1.2.11

Voice Carrier Components	
Equipment	Software
IntelliSIP	2.1 – av1.6.2.18

Testing was performed with IP Office 500 V2 R8.1, but this testing also applies to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 R8.1 to support analog or digital endpoints or trunks. Also, IP Office 500 V2 does not support SIP Direct Media so Server Edition SIP Direct Media functionality was not compliance tested.

## 5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the Avaya IP Office Manager PC, select **Start → Programs → IP Office → Manager** to launch the application. A screen that includes the following in the center may be displayed:

### **WELCOME to IP Office Administration**

#### **What would you like to do ?**

[Create an Offline Configuration](#)

[Open Configuration from System](#)

[Read a Configuration from File](#)

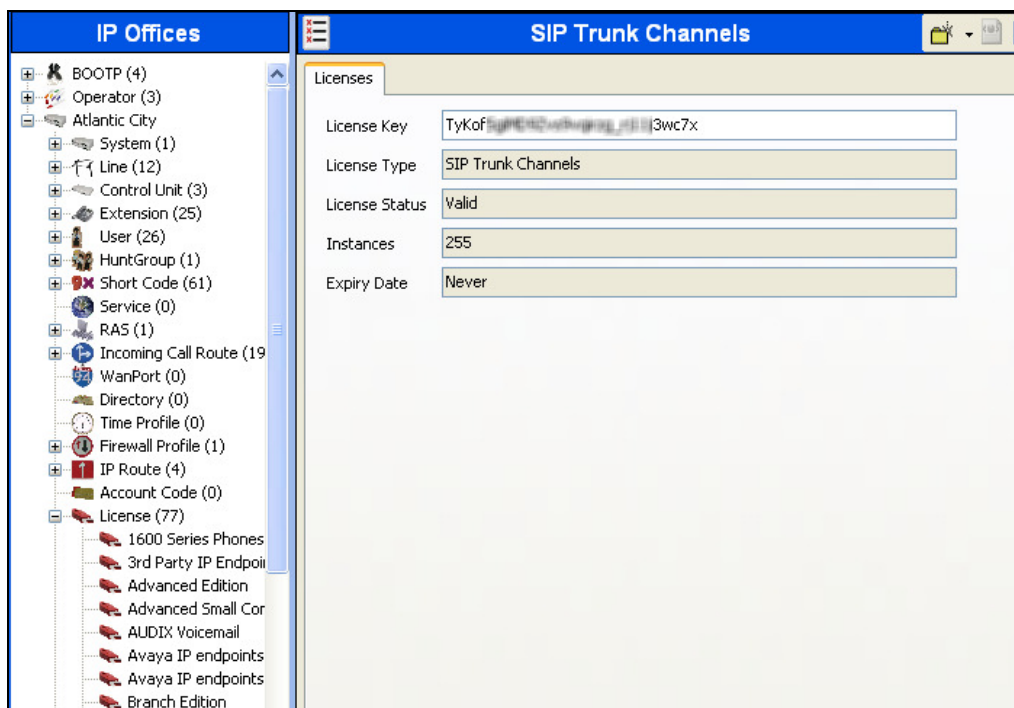
Select **Open Configuration from System**. If the above screen does not appear, the configuration may be alternatively opened by navigating to **File → Open Configuration** at the top of the Avaya IP Office Manager window. Select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and show the Details pane on the right side. Since the Group Pane has been omitted, its content is shown as submenus in the Navigation pane. These panes (Navigation, Group and Details) will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and IP Office Video Softphone support) is assumed to already be in place.

In the sample configuration, **Atlantic City** was used as the system name. All navigation described in the following sections (e.g., **License → SIP Trunk Channels**) appears as submenus underneath the system name **Atlantic City** in the Navigation Pane.

## 5.1. Licensing and Physical Hardware

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.

To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License** → **SIP Trunk Channels** in the Navigation pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details pane.





To view the details of the component, select the component in the Navigation pane. The following screen shows the details of the **IP 500 V2**.

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VCSipTkIPO81

## 5.2. System

Configure the necessary system settings.

### 5.2.1. System - LAN2 Tab

In the sample configuration, the WAN port was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office 500. To access the LAN2 settings, first navigate to **System** → **<Name>**, where **<Name>** is the system name assigned to the IP Office. In the case of the compliance test, the system name is **Atlantic City**. Next, navigate to the **LAN2** → **LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree shows a hierarchy starting with 'Atlantic City', which includes a 'System (1)' entry. The main panel on the right is titled 'Atlantic City' and contains several tabs: 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', 'SMTP', and 'SMDR'. The 'LAN2' tab is selected, and within it, the 'LAN Settings' sub-tab is active. The configuration fields are as follows:

- IP Address:** 192 . 168 . 96 . 232
- IP Mask:** 255 . 255 . 255 . 224
- Primary Trans. IP Address:** 0 . 0 . 0 . 0
- Firewall Profile:** <None>
- RIP Mode:** None
- Enable NAT:** ☐
- Number Of DHCP IP Addresses:** 200
- DHCP Mode:** Radio buttons for Server, Client, Dialin, and Disabled. The 'Disabled' option is selected.
- Advanced:** A button located at the bottom right of the configuration area.

On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values will be provided by Voice Carrier. All other parameters should be set according to customer requirements.

The screenshot shows the Avaya IP Office configuration interface for 'Atlantic City'. The 'VoIP' tab is selected, and the 'SIP Registrar' sub-tab is active. The interface includes several sections with checkboxes and input fields:

- General Settings:**
  - ☒ H.323 Gatekeeper Enable
  - ☒ SIP Trunks Enable
  - ☒ SIP Registrar Enable
  - ☐ H.323 Auto-create Extn
  - ☐ H.323 Auto-create User
  - ☐ H.323 Remote Extn Enable
  - ☒ Enable RTCP Monitoring On Port 5005
- RTP Port Number Range:**
  - Port Range (Minimum): 49152
  - Port Range (Maximum): 53246
- DiffServ Settings:**
  - DSCP(Hex): 88, DSCP Mask (Hex): FC, SIG DSCP (Hex): 88
  - DSCP: 46, DSCP Mask: 63, SIG DSCP: 34
- DHCP Settings:**
  - Primary Site Specific Option Number (SSON): 176
  - Secondary Site Specific Option Number (SSON): 242
  - VLAN: Not Present
  - 1100 Voice VLAN Site Specific Option Number (SSON): 232
  - 1100 Voice VLAN IDs: (empty field)
- RTP Keepalives:**
  - Scope: Disabled, Periodic timeout: 30
  - Initial keepalives: Enabled

On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**.
- 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 IP address of the Avaya IP Office WAN port.
- Set the **Public Port** to the port Avaya IP Office will listen on.
- All other parameters should be set according to customer requirements.

The screenshot shows the 'Atlantic City' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following settings:

Parameter	Value
STUN Server IP Address	10 . 90 . 168 . 13
STUN Port	3478
Firewall/NAT Type	Open Internet
Binding Refresh Time (seconds)	60
Public IP Address	192 . 168 . 96 . 232
Public Port	5060

Buttons: Run STUN, Cancel

☐ Run STUN on startup

### 5.2.2. System - Telephony Tab

To access the System Telephony settings, navigate to the **Telephony** → **Telephony** tab in the Details Pane. 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.

The screenshot shows the 'Atlantic City' configuration window with the 'Telephony' tab selected. The interface is divided into several sections:

- System Navigation:** A top bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony (selected), Directory Services, System Events, SMTP, SMDR, Twinning, VCM, CCR, and Codecs.
- Sub-Tabs:** Under the 'Telephony' tab, there are sub-tabs for Telephony, Tones & Music, and Call Log.
- Analogue Extensions:** A section on the left containing:
  - Default Outside Call Sequence: Normal (dropdown)
  - Default Inside Call Sequence: Ring Type 1 (dropdown)
  - Default Ring Back Sequence: Ring Type 2 (dropdown)
  - Restrict Analogue Extension Ringer Voltage: ☐
  - Dial Delay Time (secs): 4 (spinner)
  - Dial Delay Count: 0 (spinner)
  - Default No Answer Time (secs): 25 (spinner)
  - Hold Timeout (secs): 0 (spinner)
  - Park Timeout (secs): 300 (spinner)
  - Ring Delay (secs): 5 (spinner)
  - Call Priority Promotion Time (secs): Disabled (dropdown)
  - Default Currency: USD (dropdown)
  - Default Name Priority: Favor Trunk (dropdown)
- Companding Law:** A section on the right with two columns:
  - Switch:** U-Law (selected), A-Law (radio button).
  - Line:** U-Law Line (selected), A-Law Line (radio button).
- Advanced Settings:** A list of checkboxes on the right:
  - ☐ DSS Status
  - ☒ Auto Hold
  - ☒ Dial By Name
  - ☒ Show Account Code
  - ☐ Inhibit Off-Switch Forward/Transfer
  - ☐ Restrict Network Interconnect
  - ☐ Drop External Only Impromptu Conference
  - ☐ Visually Differentiate External Call
  - ☐ Unsupervised Analog Trunk Disconnect Handling
  - ☒ High Quality Conferencing

### 5.2.3. System - Twinning Tab

To view or change the System Twinning settings, navigate to the **Twining** tab in the Details Pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. Click the **OK** button at the bottom of the page (not shown).

The screenshot shows the 'Atlantic City' configuration window with the 'Twining' tab selected. The 'Send original calling party information for Mobile Twinning' checkbox is unchecked. Below it, the 'Calling party information for Mobile Twinning' field is empty.

### 5.3. IP Route

Navigate to **IP Route** → **0.0.0.0** in the left Navigation Pane if a default route already exists. Otherwise, to create the default route, right-click on **IP Route** and select **New** (not shown). Create/verify a default route with the following parameters:

- Set **IP Address** and **IP Mask** to **0.0.0.0**.
- Set **Gateway IP Address** to the IP Address of the default router to reach Voice Carrier.
- Set **Destination** to **LAN2** from the drop-down list.

Click the **OK** button at the bottom of the page (not shown).

The screenshot shows the 'IP Route' configuration window for the '0.0.0.0' route. The left pane shows the 'IP Offices' tree with 'IP Route (4)' expanded, showing '0.0.0.0' and '10.32.0.0'. The right pane shows the configuration for '0.0.0.0':

Field	Value
IP Address	0 . 0 . 0 . 0
IP Mask	0 . 0 . 0 . 0
Gateway IP Address	192 . 168 . 96 . 254
Destination	LAN2
Metric	0
Proxy ARP	<input type="checkbox"/>

## 5.4. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Voice Carrier IntelliSIP Trunking Service. To create a SIP line, right-click **Line** in the Navigation Pane and select **New → SIP Line** (not shown).

### 5.4.1. SIP Line – SIP Line Tab

On the **SIP Line** tab in the Details Pane, configure the parameters as shown below.

- Set **ITSP Domain Name** to the domain provided by Voice Carrier.
- Set **Send Caller ID** to **Diversion Header**. With this setting and the related configuration in **Section 5.2.3**, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Voice Carrier. It will also include the Diversion Header for calls that are call forwarded out the SIP Line.
- Check **REFER Support**.
- Check the **In Service** box. This makes the trunk available to incoming and outgoing calls.
- Check the **Check OOS** box. IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN2, as shown in **Section 5.2.1**.
- Default values may be used for all other parameters.

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree view containing various system components like BOOTP, Operator, Atlantic City, System, Line, Control Unit, Extension, User, HuntGroup, Short Code, Service, RAS, Incoming Call Route, WanPort, Directory, Time Profile, Firewall Profile, IP Route, Account Code, License, Tunnel, User Rights, ARS, RAS Location Request, and E911 System. The main area is titled 'SIP Line - Line 22' and contains several tabs: 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP Line' tab is active, showing configuration fields for Line Number (22), ITSP Domain Name (avayatest.callingcloud.net), Prefix, National Prefix (0), Country Code, International Prefix (00), In Service (checked), Use Tel URI (unchecked), Check OOS (checked), Call Routing Method (Request URI), Originator number for forwarded and twinning calls, Name Priority (System Default), Caller ID from From header (unchecked), Send From In Clear (unchecked), User-Agent and Server Headers, Send Caller ID (Diversion Header), Association Method (By Source IP address), REFER Support (checked), Incoming (Auto), Outgoing (Auto), and UPDATE Supported (Never).

### 5.4.2. SIP Line - Transport Tab

Select the **Transport** tab. Set the parameters as shown below.

- Set **ITSP Proxy Address** to the IP address of the Voice Carrier SIP proxy.
- Set **Layer 4 Protocol** to **UDP** or **TCP** as defined by Voice Carrier.
- Set **Use Network Topology Info** to the network port used by the SIP line to access the far-end and configured in **Section 5.2.1**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.

The screenshot shows the 'SIP Line - Line 22' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '192.168.243.180'. The 'Network Configuration' section shows 'Layer 4 Protocol' set to 'UDP', 'Send Port' set to '5060', 'Use Network Topology Info' set to 'LAN 2', and 'Listen Port' set to '5060'. The 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

SIP Line - Line 22	
SIP Line   <b>Transport</b>   SIP URI   VoIP   T38 Fax   SIP Credentials	
ITSP Proxy Address: 192.168.243.180	
Network Configuration	
Layer 4 Protocol: UDP	Send Port: 5060
Use Network Topology Info: LAN 2	Listen Port: 5060
Explicit DNS Server(s): 0 . 0 . 0 . 0    0 . 0 . 0 . 0	
Calls Route via Registrar: <input checked="" type="checkbox"/>	
Separate Registrar:	



### 5.4.3. SIP Line - SIP URI Tab

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit** button (not shown). In the example screen below, a new entry is created. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact**, **Display Name** and **PAI** to **Use Internal Data**. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any User as shown in **Section 5.7**.
- For **Registration**, select **0: <None>** from the pull-down menu.
- 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 in **Section 5.8**. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining ARS entries for routing outbound traffic to this line in **Section 5.6**. For the compliance test, a new incoming and outgoing group **22** was defined that only contained this line (line 22).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

Click **OK**.

The screenshot shows the 'SIP Line - Line 22\*' configuration window. The 'SIP URI' tab is selected. The main area contains a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is the 'New Channel' section with the following fields:

Via	192.168.96.232
Local URI	Use Internal Data
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	Use Internal Data
Registration	0: <None>
Incoming Group	22
Outgoing Group	22
Max Calls per Channel	10

At the bottom right of the 'New Channel' section are 'OK' and 'Cancel' buttons.

Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user such as a short code. These URIs are created in the same manner as shown above with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, **Display Name** and **PAI** fields.

#### 5.4.4. SIP Line - VoIP Tab

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

- For **Codec Selection**, select **System Default** from the pull-down menu. A list of the codecs in their current order of preference is shown on the right in the **Selected** column. The compliance test used the default codec list. To use a custom list of codecs, select **Custom** for **Codec Selection**. Next, move unwanted codecs from the **Selected** column to the **Unused** column. Lastly, move the codecs up or down the list in the **Selected** column to achieve the desired order of preference.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Set the **Fax Transport Support** to **G.711**.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters.

Click the **OK** button at the bottom of the page (not shown).

The screenshot shows the 'SIP Line - Line 22' configuration window with the 'VoIP' tab selected. The 'Codec Selection' dropdown is set to 'System Default'. Below it, there are two columns: 'Unused' and 'Selected'. The 'Selected' column contains a list of codecs: G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ. To the right of these columns are buttons for moving codecs between columns and within the 'Selected' column. On the right side of the window, there are several checkboxes: 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), 'Codec Lockdown' (unchecked), and 'PRACK/100rel Supported' (unchecked). At the bottom, there are three fields: 'Fax Transport Support' set to 'G.711', 'Call Initiation Timeout (s)' set to '4', and 'DTMF Support' set to 'RFC2833'.

## 5.5. Short Codes

ARS is used to route outbound traffic to the SIP line. A short code is used to route outbound traffic to ARS. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

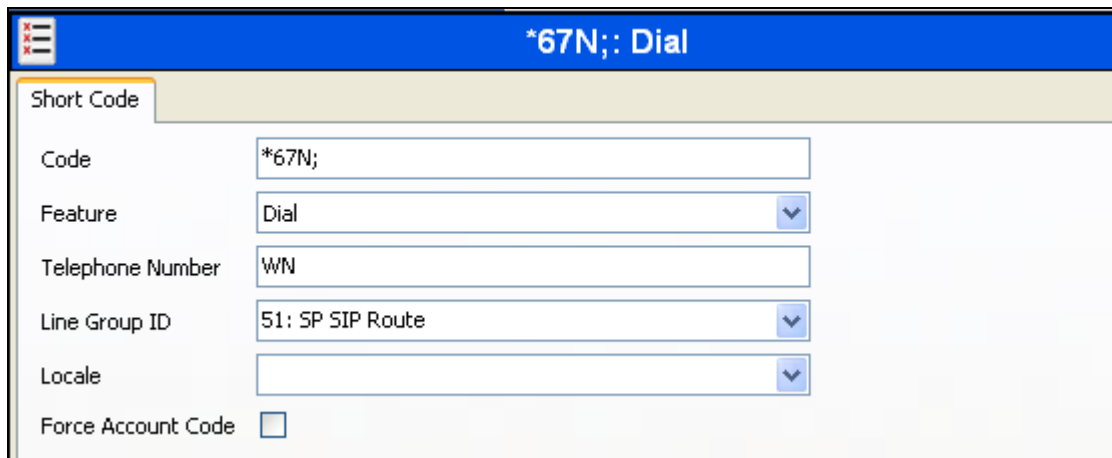
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;**. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone 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 the ARS route to be used which is defined in **Section 5.6**.

Click the **OK** button (not shown).

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'Short Code (61)' selected. The main pane is titled '9N;: Dial' and contains the 'Short Code' configuration tab. The configuration fields are as follows:

Field	Value
Code	9N;
Feature	Dial
Telephone Number	N
Line Group ID	51: SP SIP Route
Locale	
Force Account Code	<input type="checkbox"/>

Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code **\*67N;** is illustrated. This short code is similar to the **9N;** short code except that the **Telephone Number** field begins with the letter **W**, which means “withhold the outgoing calling line identification”. In the case of the SIP Line to Voice Carrier documented in these Application Notes, when a user dials \*67 plus the number, IP Office will include the user’s telephone number in the P-Asserted-Identity (PAI) header and will include the Privacy: Id header. Voice Carrier will allow the call due to the presence of a valid DID in the PAI header, but will prevent presentation of the caller id to the called PSTN destination.



*67N;; Dial	
Short Code	
Code	<input type="text" value="*67N;"/>
Feature	<input type="text" value="Dial"/>
Telephone Number	<input type="text" value="WN"/>
Line Group ID	<input type="text" value="51: SP SIP Route"/>
Locale	<input type="text"/>
Force Account Code	<input type="checkbox"/>

## 5.6. ARS

ARS is used to route outbound traffic to the SIP line. To define a new ARS route, right-click **ARS** in the Navigation pane and select **New** (not shown). In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

For the compliance test, two entries were created. The first entry matches on any 1 + 10 digit number (**1XXXXXXXXXX**) and then sends only 10 digits (**N**) in the SIP INVITE message on the line group defined in **Section 5.4.3** (e.g., line group 22). The second entry matches on any other number (**N**) and passes it unaltered to the line group 22.

To create an entry, click the **Add** button and enter the following in the pop-up window.

- In the **Code** field, enter the pattern to match the number passed to ARS from the short code in **Section 5.5** followed by a semi-colon.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@avayatest.callingcloud.net"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. In the first entry, **N** represents the remaining 10 digits after removing the preceding 1. In the second entry, the value **N** represents the complete number passed to ARS. The domain **avayatest.callingcloud.net** is the service provider domain provided by Voice Carrier.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4.3**. This short code will use this line group when placing the outbound call.

Click the **OK** button (not shown).

Code	Telephone Number	Feature	Line Group ID
1XXXXXXXXXX;	N"@avayatest.callingcloud.net"	Dial	22
N;	N"@avayatest.callingcloud.net"	Dial	22

## 5.7. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. To configure these settings, first navigate to **User** → *Name* in the Navigation Pane where *Name* is the name of the user to be modified. In the example below, the name of the user is **Extn243**. Select the **SIP** tab in the Details Pane. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls and allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.4.3**). The example below shows the settings for User **Extn243**. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Voice Carrier. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network.

Click the **OK** button (not shown).

IP Offices	
+	BOOTP (4)
+	Operator (3)
+	Atlantic City
+	System (1)
+	Line (14)
+	Control Unit (3)
+	Extension (25)
+	User (27)
+	HuntGroup (1)
+	Short Code (61)
+	Service (0)
+	RAS (1)

Extn243: 243								
User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording
Menu Programming	Mobility	Phone Manager Options	Hunt Group Membership	Announcements	SIP	Pers		
SIP Name		9725551236						
SIP Display Name (Alias)		Extn243						
Contact		9725551236						
<input type="checkbox"/> Anonymous								

## 5.8. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, right-click **Incoming Call Routes** in the Navigation Pane and select **New** (not shown).

### 5.8.1. Incoming Call Route – Standard Tab

On the **Standard** tab of the Details Pane, enter the parameters as shown below.

- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4.3**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Default values can be used for all other fields.

The screenshot shows the 'Incoming Call Route' configuration window with the 'Standard' tab selected. The window title is '22 9725551236'. The left navigation pane shows a tree structure with 'Incoming Call Route (28)' selected. The main area contains the following fields:

Field	Value
Bearer Capacity	Any Voice
Line Group ID	22
Incoming Number	9725551236
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

### 5.8.2. Incoming Call Route – Destinations Tab

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 9725551236 on line 22 are routed to extension 243.

The screenshot shows the 'Incoming Call Route' configuration window with the 'Destinations' tab selected. The window title is '22 9725551236'. The main area contains a table with the following columns: TimeProfile, Destination, and Fallback Extension.

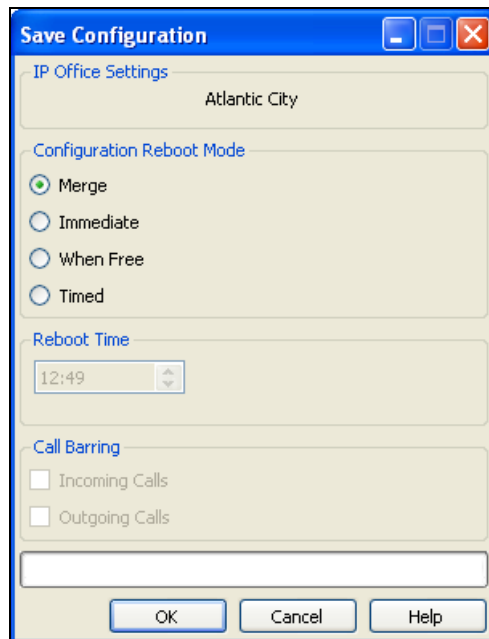
TimeProfile	Destination	Fallback Extension
Default Value	243 Extn243	

Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Figure 1** are omitted here, but can be configured in the same fashion.

## 5.9. Save 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 **Immediate** 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.





## 6. Voice Carrier IntelliSIP Trunking Configuration

Voice Carrier is responsible for the configuration of the Voice Carrier IntelliSIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. In the case of the compliance test, this is the Avaya IP Office LAN2 public IP address. Voice Carrier will provide the customer the necessary information to configure the Avaya IP Office SIP connection to Voice Carrier including:

- Voice Carrier SIP domain
- IP address of the Voice Carrier SIP proxy
- Supported codecs
- DID numbers
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices

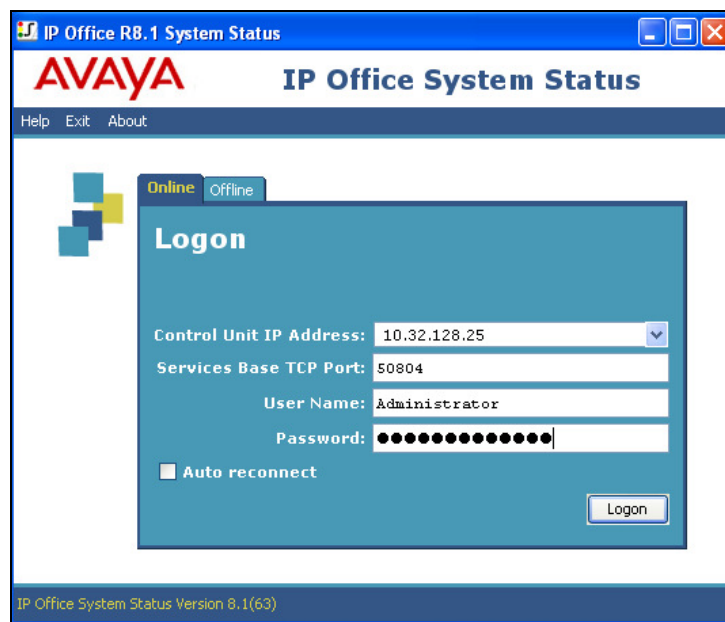
## 7. Verification Steps

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

### 7.1. System Status

The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → Programs → IP Office → System Status**.

The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



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

**AVAYA IP Office System Status**

Help Snapshot LogOff Exit About

**System**  
**Alarms (10)**  
**Extensions (20)**  
**Trunks (14)**  
 Lines: 1 - 4  
 Lines: 5 - 8  
 Line: 17  
 Line: 18  
 Line: 19  
 Line: 20  
 Line: 21  
**Line: 22**  
 Active Calls  
 Resources  
 Voicemail  
 IP Networking

**Status** Utilization Summary Alarms

**SIP Trunk Summary**

Peer Domain Name: avayatest.callingcloud.net  
 Resolved Address: 192.168.243.180  
 Line Number: 22  
 Number of Administered Channels: 10  
 Number of Channels in Use: 0  
 Administered Compression: G711 Mu, G711 A, G729 A, G7231  
 Silence Suppression: Off  
 SIP Trunk Channel Licenses: Unlimited  
 SIP Trunk Channel Licenses in Use: 0  
 SIP Device Features: REFER (Incoming and Outgoing)

0%

Channel Number	U...	Call Ref	Current State	Time in State	Remote Media A...	Co...	Conne...	Caller ID or ...	Other Party on Call	Directi...	Round Trip D...	Receive Jitter	Receive Packe...	Transmit Jitter	Trans...
1			Idle	21:39...											
2			Idle	21:46...											
3			Idle	1 day ...											
4			Idle	1 day ...											

Select the **Alarms** tab and verify that no alarms are active on the SIP line.

**Status** Utilization Summary **Alarms**

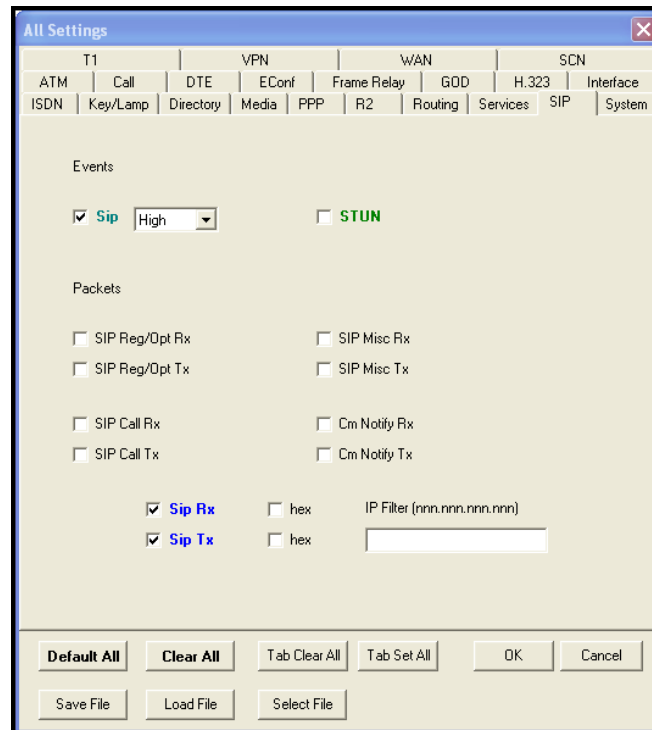
**Alarms for Line: 22 SIP avayatest.callingcloud.net**

Last Date Of Error	Occurrences	Error Description
--------------------	-------------	-------------------

## 7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select **Filters → Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked.



## 8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office 8.1 to the Voice Carrier IntelliSIP Trunking Service. The Voice Carrier IntelliSIP 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. The Voice Carrier IntelliSIP Trunking Service passed compliance testing. Please refer to **Section 2.2** for any exceptions.

## 9. Additional References

This section references documentation relevant to these Application Notes. In general, Avaya product documentation is available at <http://support.avaya.com>.

- [1] *IP Office 8.1 IP500/IP500 V2 Installation*, Document Number 15-601042, Issue 27m, July 2, 2013.
- [2] *IP Office Release 8.1 Manager 10.1*, Document Number 15-601011, Issue 29u, April 5, 2013.
- [3] *IP Office System Status Application*, Document Number 15-601758, Issue 07a, November 26, 2012.
- [4] *IP Office Release 8.1 Administering Voicemail Pro*, Document Number 15-601063, Issue 8b, December 11, 2012.
- [5] *IP Office System Monitor*, Document Number 15-601019, Issue 03c, March 1, 2013.

Additional IP Office documentation can be found at:

<http://marketingtools.avaya.com/knowledgebase/>

## 10. Appendix A: SIP Line Template

Avaya IP Office supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Note that not all of the configuration information, particularly items relevant to a specific installation environment, is included in the SIP Line Template. Therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported and additional configuration be supplemented using **Section 5.4** in these Application Notes as a reference.

The SIP Line Template created from the configuration as documented in these Application Notes is as follows:

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130710</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Voice Carrier</DescriptiveName>
  <ITSPDomainName>avayatest.callingcloud.net</ITSPDomainName>
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>2</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateNever</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>192.168.243.180</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>false</UseAdvVoiceCodecPrefs>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_G711</FaxTransportSupport>
  <UseOffererPreferredCodec>false</UseOffererPreferredCodec>
```

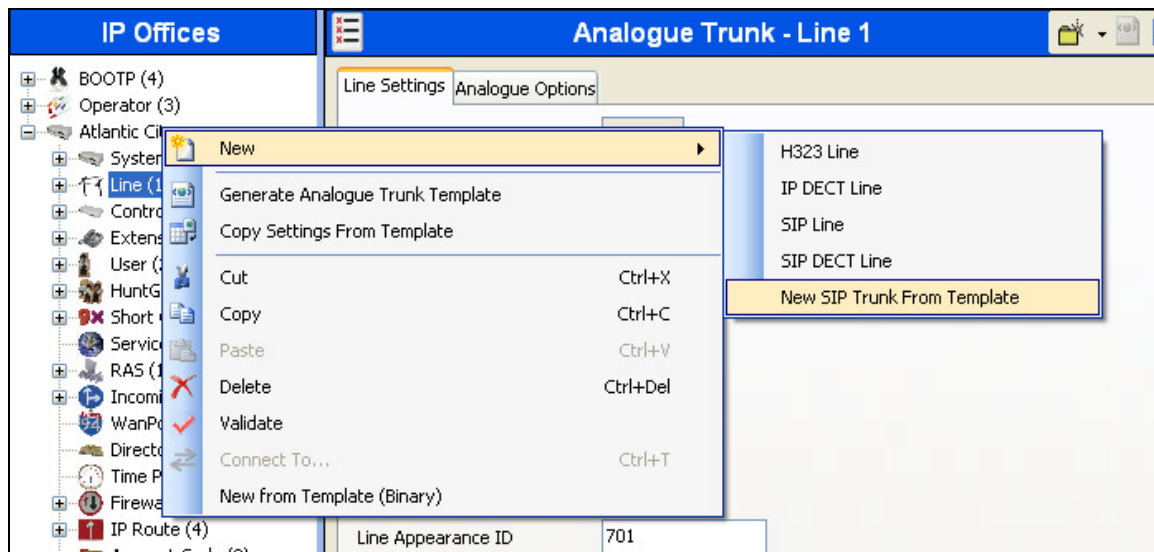
```

<CodecLockdown>>false</CodecLockdown>
<Rel100Supported>>false</Rel100Supported>
<T38FaxVersion>3</T38FaxVersion>
<Transport>UDPTL</Transport>
<LowSpeed>0</LowSpeed>
<HighSpeed>0</HighSpeed>
<TCFMethod>Trans_TCF</TCFMethod>
<MaxBitRate>FaxRate_14400</MaxBitRate>
<EflagStartTimer>2600</EflagStartTimer>
<EflagStopTimer>2300</EflagStopTimer>
<UseDefaultValues>>true</UseDefaultValues>
<ScanLineFixup>true</ScanLineFixup>
<TFOPENenhancement>true</TFOPENenhancement>
<DisableT30ECM>>false</DisableT30ECM>
<DisableEflagsForFirstDIS>>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>>false</DisableT30MRCompression>
<NSFOVERRIDE>>false</NSFOVERRIDE>
</Template>

```

To import the above template into a new installation:

1. On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US\_VoiceCarrier\_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates). It may be necessary to create this directory.
2. Import the template into an IP Office installation by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New → New SIP Trunk From Template**:



3. Verify that **United States** is automatically populated for **Country** and **VoiceCarrier** is automatically populated for **Service Provider** in the resulting Template Type Selection screen as shown below. Click **Create new SIP Trunk** to finish the importing process.

**Template Type Selection**

Locale: United States (US English) ▼

Country: United States ▼

Service Provider: VoiceCarrier ▼ ☐ Display All

Create new SIP Trunk Cancel

---

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