



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Intelepeer SIP Trunking with Avaya IP Office 7.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Service Provider Intelepeer and Avaya IP Office 7.0.

Intelepeer SIP Trunking provides PSTN access via a SIP trunk between the enterprise and the Intelepeer network as an alternative to legacy analog or digital trunks. This approach generally results in lower costs for the enterprise.

Intelepeer 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 procedures for configuring Session Initiation Protocol (SIP) Trunking between Service Provider Intelpeer and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 7.0, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya SIP, H.323, digital, and analog endpoints.

The Intelpeer SIP Trunking service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

The Intelpeer SIP Trunking service does not require the enterprise IP PBX to register with the SIP network via SIP credentials. Customers are authenticated via IP authentication.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to Intelpeer SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**.

2.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office was connected to Intelpeer SIP Trunking service. To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various phone types. Phone types included SIP, H.323, digital, and analog 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 various phone types. Phone types included SIP, H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider.
- Inbound and outbound PSTN calls to/from the Avaya IP Office Softphone.
- Inbound and outbound long hold time call stability.
- Various call types including: local, long distance, international, outbound toll-free, operator service and directory assistance.
- Codec G.711MU and G.729A.
- Caller number/ID presentation.
- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- Telephony features such as hold and resume, transfer, and conference.
- Off-net call forwarding.
- Twinning to mobile phones on inbound calls.

2.2. Test Results

Intelepeer SIP Trunking passed compliance testing. Items not supported or not tested included the following:

- Inbound toll-free and outbound emergency calls (911) are supported but were not tested as part of the compliance test.
- Operator (0) calls, Operator Assisted (0+10 digits) dialing and 411 / 1411 services are not supported and therefore were not tested.

Interoperability testing of Intelepeer SIP Trunking was completed with successful results for all test cases.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>.

For technical support on Intelepeer SIP Trunking, contact Intelepeer at <http://www.Intelepeer.com/> or 1-866-780-8639.

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to Intelepeer SIP Trunking service through the public IP network. Located at the enterprise site is an Avaya IP Office 500v2 with the COMBO6210/ATM4 expansion card which provides connections for 6 digital stations, 2 analog stations, 4 analog trunks to the PSTN as well as 10-channel VCM (Voice Compression Module) for supporting VoIP codecs. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 1120e Series IP Telephone (with SIP firmware), an Avaya 1408 Digital Telephone, an Avaya Analog Telephone and an Avaya IP Office Softphone. The site also has a Windows 2003 Server running Avaya Voicemail Pro for providing voice messaging service to the Avaya IP Office users. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer the Avaya IP Office system. Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user's phones will also ring and can be answered at the configured mobile phones.

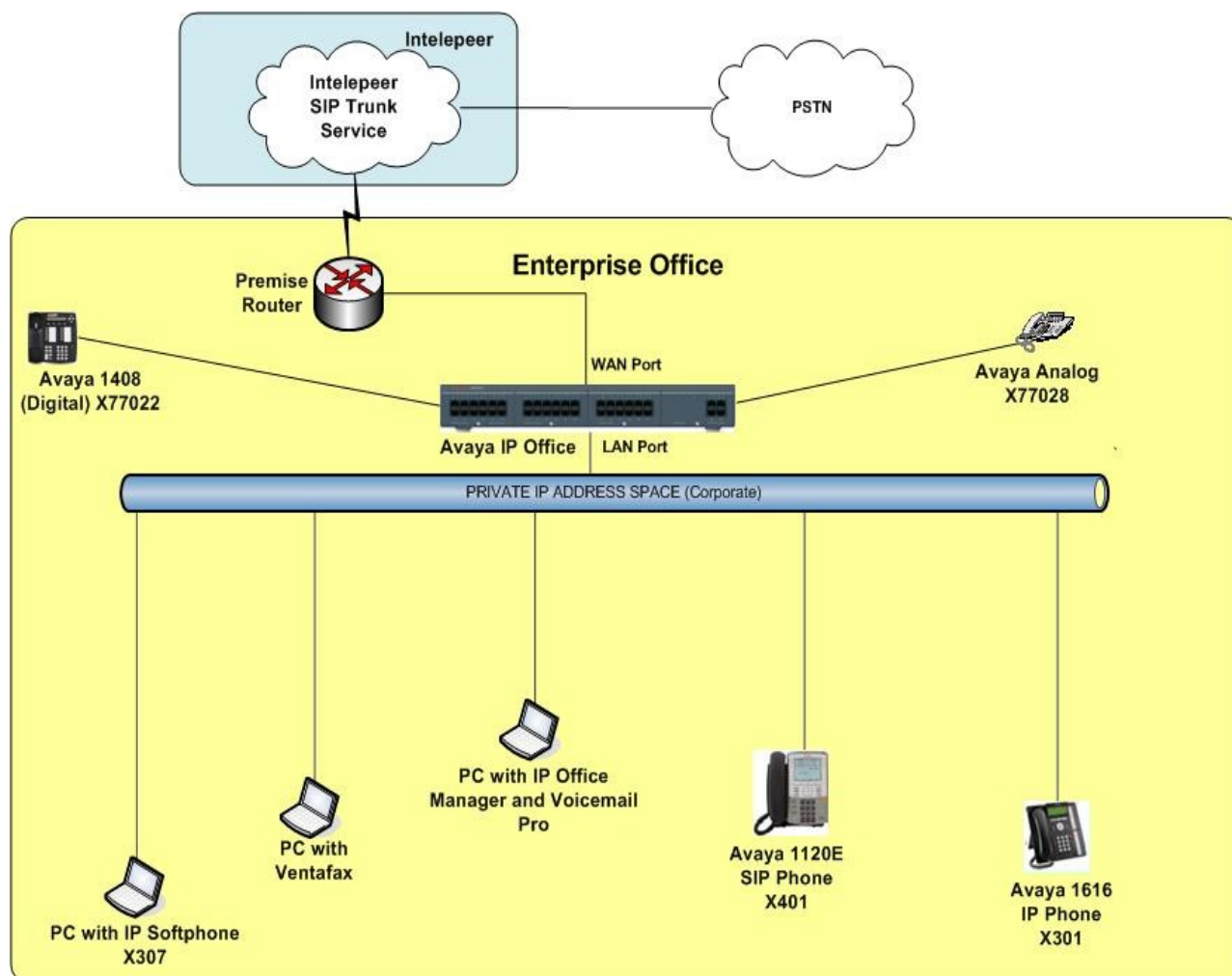


Figure 1: Test Configuration for Avaya IP Office with Intelepeer SIP Trunking Service

For security purposes, the real public IP addresses and PSTN routable phone numbers used in the compliance test are masked in these Application Notes.

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 9 + N digits to send digits across the SIP trunk to Intelepeer. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Intelepeer. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office would send 11 digits in the Request URI and the To headers of an outbound SIP INVITE message. IP Office was configured to send 10 digits in the From and PAI headers. For inbound calls, Intelepeer SIP Trunking sent 10 digits in the Request URI, To, and From headers of inbound SIP INVITE messages.

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

4. Equipment and Software Validated

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

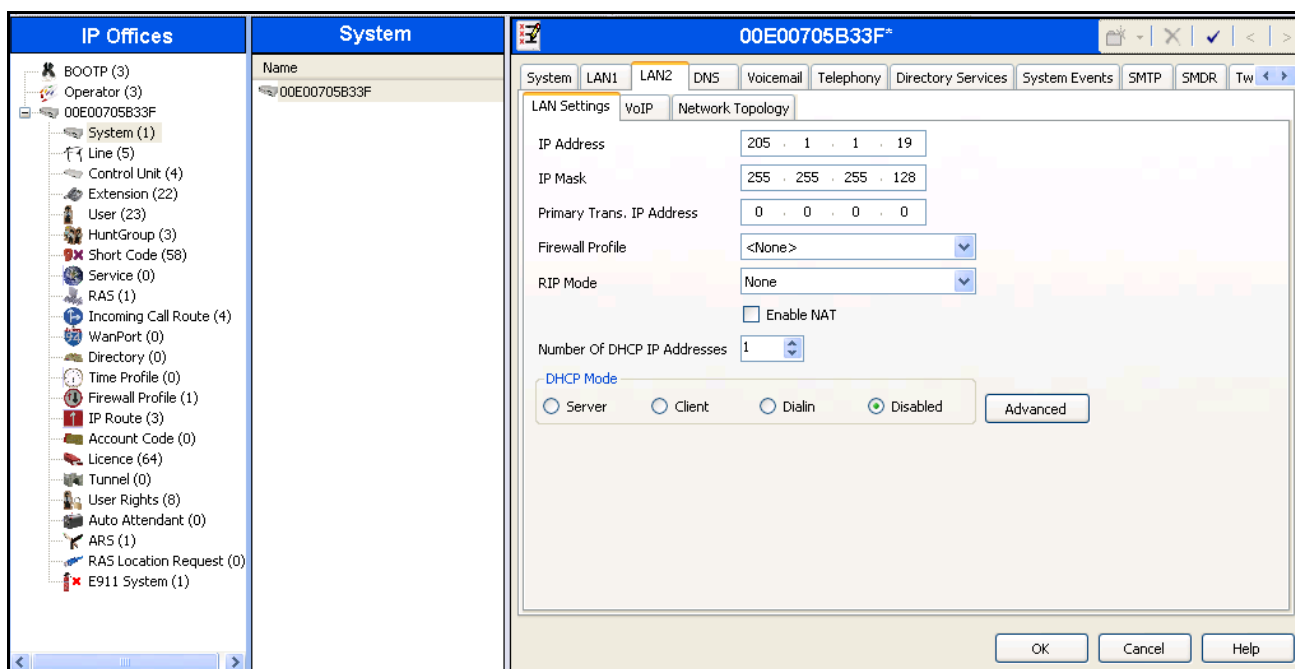
Avaya Telephony Components	
Equipment	Release
Avaya IP Office 500v2	7.0 (5)
Avaya IP Office COMBO6210/ATM4 Module	7.0 (5)
Avaya Voicemail Pro	7.0.17
Avaya IP Office Manager	9.0 (5)
Avaya 1616 IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.2.2
Avaya 1120E IP Telephone (SIP)	SIP1120 version 04.01.13.00
Avaya 1408 Digital Telephone	N/A
Avaya Analog Telephone	N/A
Avaya IP Office Softphone	3.1.2.17 59616
Intelepeer SIP Trunking	
Equipment	Release
Sonus GSX9000HD	Version: V07.03.04 S003

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to Intelpeer SIP Trunking service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one shown in the next section. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interface with the Service Provider (such as LAN interface to the enterprise site and IP Office Softphone support) is assumed to be already in place.

5.1. LAN2 Settings

In the sample configuration, the MAC address **00E00705B33F** was used as the system name and the WAN port (LAN2) 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 system. To access the LAN2 settings, first navigate to **System (1) → 00E00705B33F** in the Navigation and Group Panes respectively, and then 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. Click “OK” when finished and Save the configuration.



Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Intelepeer. 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. To prevent possible loss of audio path during some call forward off-net scenarios, it is recommended to set the following fields under **RTP Keepalives**: Set **Scope** to **RTP**, **Initial keepalives** to **Enable** and an appropriate **Periodic timeout** from **1** to **180** seconds. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements.

The screenshot displays the Avaya IP Office configuration window for system 00E00705B33F. The 'VoIP' tab is selected, showing the following configuration details:

- System:** 00E00705B33F
- LAN Settings:**
 - ☐ H323 Gatekeeper Enable
 - ☒ SIP Trunks Enable
 - ☐ SIP Registrar Enable
- RTP Port Number Range:**
 - ☒ H323 Auto-create Extn
 - ☐ H323 Auto-create User
 - Port Range (Minimum): 49152
 - Port Range (Maximum): 53246
- Enable RTP Monitoring On Port 5005:** ☒
- 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:
- RTP Keepalives:**
 - Scope: RTP
 - Initial keepalives: Enabled
 - Periodic timeout: 2

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**. With this configuration, STUN will not be used.
- Set **Binding Refresh Time (seconds)** to **60**. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the Service Provider. See **Section 5.10** for complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- All other parameters should be set according to customer requirements.
- Click “**OK**” when finished and Save the configuration.

The screenshot shows the Avaya IP Office configuration window with the title bar '00E00705B33F*'. The 'Network Topology' tab is selected under the 'VoIP' category. The 'Network Topology Discovery' section contains the following fields and values:

Field	Value
STUN Server IP Address	69 . 90 . 168 . 13
STUN Port	3478
Firewall/NAT Type	Open Internet
Binding Refresh Time (seconds)	60
Public IP Address	205 . 1 . 1 . 19
Public Port	5060

Below the fields are two buttons: 'Run STUN' and 'Cancel'. At the bottom right of the window are 'OK', 'Cancel', and 'Help' buttons. A checkbox labeled 'Run STUN on startup' is located below the 'Run STUN' button and is currently unchecked.

In the compliance test, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with Intelpeer SIP Trunking service, and therefore is not described in these Application Notes.

5.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** Tab in the Details Pane. Set the **Automatic Codec Preference** for the default codec to be used for intra-enterprise traffic. Choose the **Companding Law** typical for the enterprise location. For North America, **ULAW** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the Service Provider across the SIP trunk.

00E00705B33F*

System LAN1 LAN2 DNS Voicemail **Telephony** Directory Services System Events SMTP SMDR Twir

Telephony Tones & Music Call Log

Analogue Extensions

Default Outside Call Sequence Normal

Default Inside Call Sequence Ring Type 1

Default Ring Back Sequence Ring Type 2

Restrict Analogue Extension Ringer Voltage ☐

Dial Delay Time (secs) 4

Dial Delay Count 0

Default No Answer Time (secs) 15

Hold Timeout (secs) 120

Park Timeout (secs) 300

Ring Delay (secs) 5

Call Priority Promotion Time (secs) Disabled

Default Currency USD

Automatic Codec Preference G.729(a) 8K CS-ACELP

Companding Law

Switch

☒ ULAW

☐ ALAW

Line

☒ UL

☐ AL

☐ DSS Status

☒ Auto Hold

☒ Dial By Name

☒ Show Account Code

☐ Inhibit Off-Switch Forward/T

☐ Restrict Network Interconne

☐ Drop External Only Impromptu

☐ Visually Differentiate External

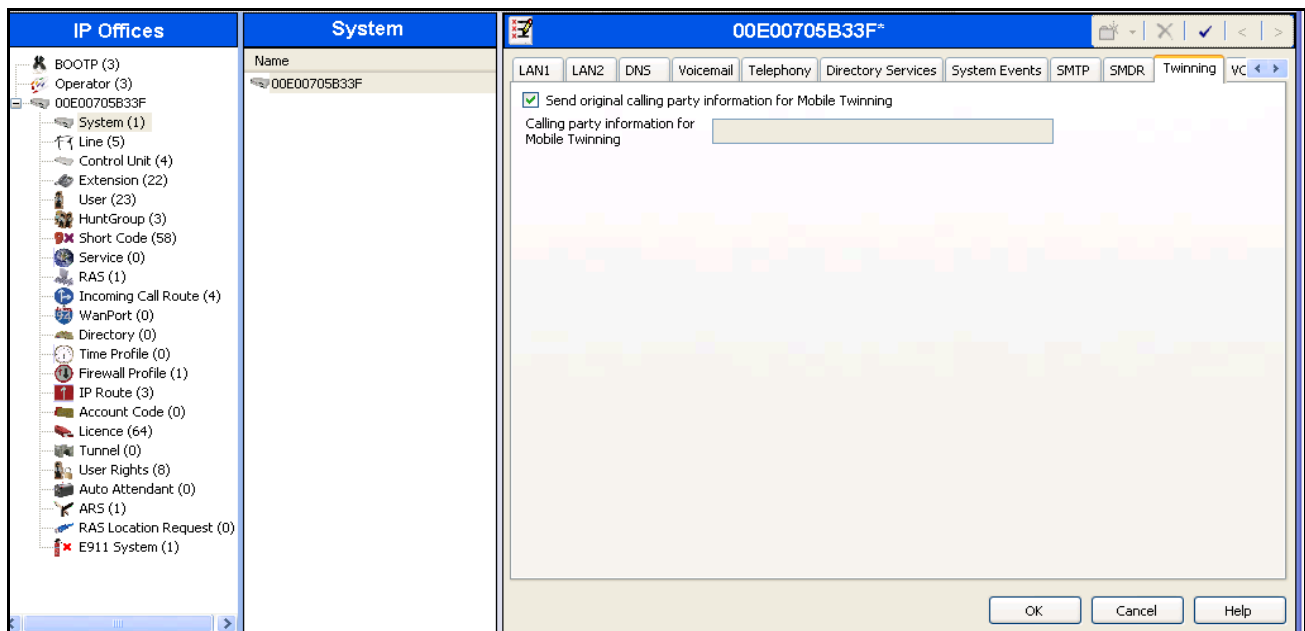
OK Cancel Help

5.3. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.4**). If **Send original calling party information for Mobile Twinning** on the **System→Twinning** tab is set, the setting of the second parameter is ignored and Avaya IP Office will send the following in the SIP From Header:

- On calls from an internal extension to a twinned phone, Avaya IP Office will send the calling party number of the originating extension.
- On calls from the PSTN to a twinned phone, Avaya IP Office will send the calling party number of the host phone associated with the twinned destination (instead of the number of the originating caller).

The above behavior in Avaya IP Office Release 7 is the same as in Avaya IP Office Release 6 and was tested and verified in the compliance test. Avaya IP Office Release 7 also provides an alternative method of sending caller ID through SIP Diversion header (configured via unchecking **Send original calling party information for Mobile Twinning** here then selecting **Diversion Header** for the **Send Caller ID** parameter on the **SIP Line** form in **Section 5.4**). This alternative configuration could provide more accurate caller ID information if the Service Provider supports the SIP Diversion header (not tested in the compliance test). For the compliance test, the **Send original calling party information for Mobile Twinning** box in the **System→Twinning** tab was checked which overrides any setting of the **Send Caller ID** parameter on the **SIP Line** form.



5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Intelepeer SIP Trunking service. To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the enterprise domain, or the IP address of the IP Office WAN interface, so that IP Office uses this domain / IP address as the host portion of SIP URI in SIP headers such as the From header.
- Set **Send Caller ID** to *None*. For the compliance test, this parameter was ignored since **Send original calling party information for Mobile Twinning** is optioned in **Section 5.3**.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, Avaya IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Default values may be used for all other parameters.

The area of the screen entitled **REFER Support** is used to enable/disable SIP REFER for call transfers. If “**REFER Support**” is left unchecked, or the fields for **Incoming** and **Outgoing** are set to “**Auto**”, this will effectively disable the use of SIP REFER. To enable SIP REFER, check the “**REFER Support**” button and select “Always” from the drop-down menu for **Incoming** and **Outgoing**.

The screenshot displays the Avaya IP Office configuration window. The left pane shows the 'Line' group selected. The main pane shows the 'SIP Line - Line 17' configuration. The 'SIP Line' tab is active, showing various configuration fields. The 'Line Number' is 17. The 'ITSP Domain Name' is 205.1.1.19. The 'In Service' checkbox is checked. The 'Use Tel URI' checkbox is unchecked. The 'Prefix' field is empty. The 'National Prefix' is 0. The 'Country Code' field is empty. The 'International Prefix' is 00. The 'Send Caller ID' is set to None. The 'Association Method' is set to By Source IP address. The 'Call Routing Method' is set to Request URI. The 'Originator number for forwarded and twinning calls' is 7133433763. The 'REFER Support' section is expanded, showing 'Incoming' and 'Outgoing' both set to Auto. The 'OK', 'Cancel', and 'Help' buttons are at the bottom right.

Line Number	Line Type
1	Analogue Trunk
2	Analogue Trunk
3	Analogue Trunk
4	Analogue Trunk
17	SIP Line

SIP Line - Line 17

Line Number: 17

ITSP Domain Name: 205.1.1.19

In Service: ☒

Use Tel URI: ☐

Prefix:

Check OOS: ☒

National Prefix: 0

Country Code:

Call Routing Method: Request URI

Originator number for forwarded and twinning calls: 7133433763

International Prefix: 00

Send Caller ID: None

Association Method: By Source IP address

☐ REFER Support

Incoming: Auto

Outgoing: Auto

OK Cancel Help

Select the **Transport** tab. The **ITSP Proxy Address** is set to the Intelpeer SIP Proxy IP Address provided by Intelpeer. As shown in **Figure 1**, this IP Address is **68.1.1.41**. In the **Network Configuration** area, **UDP** is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by Intelpeer. The **Use Network Topology Info** parameter is set to **LAN 2**. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab. Other parameters retain default values in the screen below.

The screenshot shows the 'SIP Line - Line 18*' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '68.1.1.41'. In the 'Network Configuration' section, 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', 'Use Network Topology Info' is set to 'LAN 2', and 'Listen Port' is '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. The window has 'OK', 'Cancel', and 'Help' buttons at the bottom right.

Field	Value
ITSP Proxy Address	68.1.1.41
Layer 4 Protocol	UDP
Send Port	5060
Use Network Topology Info	LAN 2
Listen Port	5060
Explicit DNS Server(s)	0 . 0 . 0 . 0
Explicit DNS Server(s)	0 . 0 . 0 . 0
Calls Route via Registrar	<input checked="" type="checkbox"/>
Separate Registrar	

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. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to *Use Internal Data*. This setting allows calls on this line for SIP URIs that match the number set in the **SIP** tab of any **User** as shown in **Section 5.6**.
- Set **PAI** to *Use Internal Data*. With this setting Avaya IP Office will populate the SIP P-Asserted-Identity header on outgoing calls with the data set in the **SIP** tab of the **User** initiating the call as shown in **Section 5.6**.
- For **Registration**, select *None* since Intelpeer doesn't require Registration. If Registration was required, then the account credentials that would have been configured on the line's **SIP Credentials** tab would be entered here.
- Associate this line with an incoming line group in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. For the compliance test, a new incoming and outgoing line group **18** was defined.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

The screenshot shows the 'SIP Line - Line 18' configuration window. The 'SIP URI' tab is selected. Below the tab, there is a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. The table contains one entry with Channel 1, Groups 18 18, Via 205.1.1.19, and Max Calls 10. To the right of the table are buttons: Add..., Remove, and Edit... The 'Edit...' button is highlighted. Below the table is the 'Edit Channel' dialog box. It contains the following fields: Via (205.1.1.19), Local URI (Use Internal Data), Contact (Use Internal Data), Display Name (Use Internal Data), PAI (Use Internal Data), Registration (0: <None>), Incoming Group (18), Outgoing Group (18), and Max Calls per Channel (10). At the bottom of the dialog are OK and Cancel buttons. At the bottom of the main window are OK, Cancel, and Help buttons.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	18 18	205.1.1.19				0: <Non...		10

Edit Channel

Via: 205.1.1.19

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 0: <None>

Incoming Group: 18

Outgoing Group: 18

Max Calls per Channel: 10

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

- The **Compression Mode** was configured using the **Advanced** button, allowing an explicit ordered list of codecs to be specified. Check marks next to **G.729(a) 8K CS-ACELP** and **G.711 ULAW** codecs cause Avaya IP Office to include these codes, supported by the Intelepeer SIP Trunking service, in the Session Description Protocol (SDP) offer, in that order.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP event messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box. By unchecking the **VoIP Silence Suppression** box, calls can be established with the G.729 codec but without silence suppression.
- Select **T.38** for **Fax Transport Support**. T.38 faxing is supported by Intelepeer and was successfully tested.
- Check the **Re-invite Supported** box.
- Default values may be used for all other parameters.

The screenshot shows the 'SIP Line - Line 18' configuration window with the 'VoIP' tab selected. The window has a blue title bar and a toolbar with icons for help, save, delete, confirm, and navigation. Below the title bar are tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'VoIP' tab is active, showing various configuration options. On the left, there is a 'Compression Mode' section with an 'Advanced' button. Below it are 'Fax Transport Support' (set to 'T38'), 'Call Initiation Timeout (s)' (set to '6'), and 'DTMF Support' (set to 'RFC2833'). In the center, a list of codecs is shown with checkboxes: 'G.729(a) 8K CS-ACELP' (checked), 'G.711 ULAW 64K' (checked), 'G.711 ALAW 64K' (unchecked), and 'G.723.1 6K3 MP-MLQ' (unchecked). On the right, there are checkboxes for 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), and 'Codec Lockdown' (unchecked). At the bottom right are 'OK', 'Cancel', and 'Help' buttons.

Parameter	Value
Compression Mode	Advanced
VoIP Silence Suppression	<input type="checkbox"/>
Re-invite Supported	<input checked="" type="checkbox"/>
Use Offerer's Preferred Codec	<input type="checkbox"/>
Codec Lockdown	<input type="checkbox"/>
Fax Transport Support	T38
Call Initiation Timeout (s)	6
DTMF Support	RFC2833
Codecs	<input checked="" type="checkbox"/> G.729(a) 8K CS-ACELP <input checked="" type="checkbox"/> G.711 ULAW 64K <input type="checkbox"/> G.711 ALAW 64K <input type="checkbox"/> G.723.1 6K3 MP-MLQ

5.5. Short Code

Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered “9N;” short code used in the test configuration.

- 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*68.1.1.41**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. The host part following the “@” is the domain or IP address of the Service Provider network.
- 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**. This short code will use this line group when placing the outbound call.

The screenshot displays the Avaya System Manager interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'Short Code (59)' selected. The main area is divided into two panes. The left pane, titled 'Short Code', contains a table listing various short codes. The right pane, titled '9N;: Dial', shows the configuration details for a specific short code.

Code	Telephone N...	Featu
9X*40	1	Relay
9X*41	1	Relay I
9X*42	2	Relay
9X*43	2	Relay
9X*44	2	Relay I
9X*45*N#	N	Call St
9X*46		Call St
9X*47		Confer
9X*48		Voicerr
9X*49		Voicerr
9X*50		Forwai
9X*51		Forwai
9X*52		Clear
9X*53*N#	N	Call Pic
9X*57*N#	N	Forwai
9X*70*N#	N	Dial Ph
9X*71*N#	N	Dial Ph
9X*9000*	"MAINTENAN...	Relay
9X*91N;	N".1"	Recorc
9X*92N;	N".2"	Recorc
9X*DSSN	";[0]151/ERR ...	Display
9X*SDN	";[0]151/ERR ...	Display
9X*SKN	";[0]151/ERR ...	Display
9X6N;	N*68.1.1.41	Dial
9X8N;	WN*68.1.1....	Dial
9X9N;	N*68.1.1.41"	Dial

The right pane, titled '9N;: Dial', shows the configuration details for a specific short code. The fields are as follows:

- Short Code**: 9N;
- Feature**: Dial
- Telephone Number**: N*68.1.1.41"
- Line Group Id**: 18
- Locale**: United States (US English)
- Force Account Code**: ☐

At the bottom of the right pane, there are buttons for 'OK', 'Cancel', and 'Help'.

The simple “9N;” short code illustrated above does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code 6N is illustrated for access to ARS. When the Avaya IP Office user dials 6 plus any number N, rather than being directed to a specific **Line Group Id**, the call is directed to **Line Group ID “50: Main”**, configurable via ARS. See **Section 5.8** for example ARS route configuration for “50: Main” as well as a backup route.

6N;: Dial

Short Code

Code: 6N;

Feature: Dial

Telephone Number: N"@68.1.1.41

Line Group Id: 50: Main

Locale:

Force Account Code: ☐

OK Cancel Help

5.6. 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 select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is “H323 1616”. 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 header for outgoing SIP trunk calls. They also 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**). The example below shows the settings for user H323 1616. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Intelpeer. 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.

The screenshot displays the Avaya SIP configuration interface. On the left, the 'IP Offices' pane shows a tree structure with 'User (24)' selected. The center pane, titled 'User', lists various users including 'Analog Phone', 'Digital 1408', and 'H323 1616' (highlighted). The right pane, titled 'H323 1616: 301', shows the 'SIP' tab selected. The 'SIP' tab contains the following fields:

Field	Value
SIP Name	3033289132
SIP Display Name (Alias)	H323 1616
Contact	3033289132

Below these fields is an 'Anonymous' checkbox, which is currently unchecked. At the bottom right of the details pane are 'OK', 'Cancel', and 'Help' buttons.

One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for User H323 1616. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone. Other options can be set according to customer requirements.

The screenshot shows a configuration window titled "H323 1616: 301*" with a blue header bar. Below the header is a tabbed interface with the following tabs: "Mobility", "Phone Manager Options", "Hunt Group Membership", "Announcements", "SIP", and "Personal Directory". The "Mobility" tab is selected. The "Internal Twinning" section is collapsed. The "Mobility Features" section is expanded and contains the following settings:

- ☒ **Mobility Features**
- ☒ **Mobile Twinning**
 - Twinned Mobile Number (including dial access code): 917205558022
 - Twinning Time Profile: <None>
 - Mobile Dial Delay (secs): 2
 - Mobile Answer Guard (secs): 0
 - ☒ Hunt group calls eligible for mobile twinning
 - ☐ Forwarded calls eligible for mobile twinning
 - ☐ Twin When Logged Out
- ☐ one-X Mobile Client
- ☐ Mobile Call Control
- ☐ Mobile Callback

At the bottom of the window are three buttons: "OK", "Cancel", and "Help".

5.7. 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, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New**. 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**.
- 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 left pane shows the 'IP Offices' tree with 'Incoming Call Route (9)' selected. The center pane displays a table of existing routes. The right pane shows the configuration for a new route with the title '18 3035559132'.

Line Group Id	Incoming Number	Destination
0		200 Main
0		DialIn
17	7135553764	100 DID Hunt Group
17	7135553765	301 H323 1616
18	3035559134	*17
18	3035559133	401 SIP 1120E
18	3035559132	301 H323 1616
18	3035559131	77022 Digital 1408
18	3035559130	77028 Analog Phone

Configuration fields for the new route:

- Bearer Capability: Any Voice
- Line Group Id: 18
- Incoming Number: 3035559132
- Incoming Sub Address:
- Incoming CLI:
- Locale:
- Priority: 1 - Low
- Tag:
- Hold Music Source: System Source

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to 303-555-9132 on line 18 are routed to user the Avaya 1616-I H.323 endpoint at extension 301.

The screenshot shows the 'Incoming Call Route' configuration window with the 'Destinations' tab selected. The left pane shows the 'Incoming Call Route' table. The right pane shows the configuration for a new route with the title '18 3035559132'.

Line Group Id	Incoming Number	Destination
0		200 Main
0		DialIn
17	7135553764	100 DID Hunt Group
17	7135553765	301 H323 1616
18	3035559134	*17
18	3035559133	401 SIP 1120E
18	3035559132	301 H323 1616
18	3035559131	77022 Digital 1408
18	3035559130	77028 Analog Phone

Configuration fields for the new route:

- TimeProfile: Default Value
- Destination: 301 H323 1616
- Fallback Extension: 301 H323 1616

5.8. ARS and Alternate Routing

While detailed coverage of Automatic Route Selection (ARS) is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations. ARS is illustrated here mainly to illustrate alternate routing should the SIP Line be out of service or temporarily not responding.

Optionally, ARS can be used rather than the simple “9N;” short code approach documented in **Section 5.5**. With ARS, a secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, 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. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish between these call behaviors.

To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named “Main”. The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter. IP Office short codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service Route, and the means to manually activate the Out of Service Route, can be helpful for scheduled maintenance or other known service-affecting events for the primary route.

The screenshot displays the Avaya IP Office configuration interface. The left pane shows the 'IP Offices' tree with 'ARS' selected. The middle pane shows the 'ARS' configuration for the 'Main' route. The right pane shows the 'Main' route configuration details.

ARS Configuration:

- Name: Main
- Time Profile: <None>

Main Route Configuration:

- ARS Route Id: 50
- Route Name: Main
- Dial Delay Time: System Default (4)
- Secondary Dial tone: SystemTone
- Check User Call Barring: ☒
- In Service: ☒ (Out of Service Route: <None>)
- Time Profile: <None> (Out of Hours Route: <None>)

Code Table:

Code	Telephone Number	Feature	Line Group Id
11	911	Dial Emergency	0
911	911	Dial Emergency	0
0N;	0N	Dial 3K1	18
1N;	1N@68.1.1.41"	Dial 3K1	18
XN;	N	Dial 3K1	0
XXXXXXXXXXN	N	Dial 3K1	0

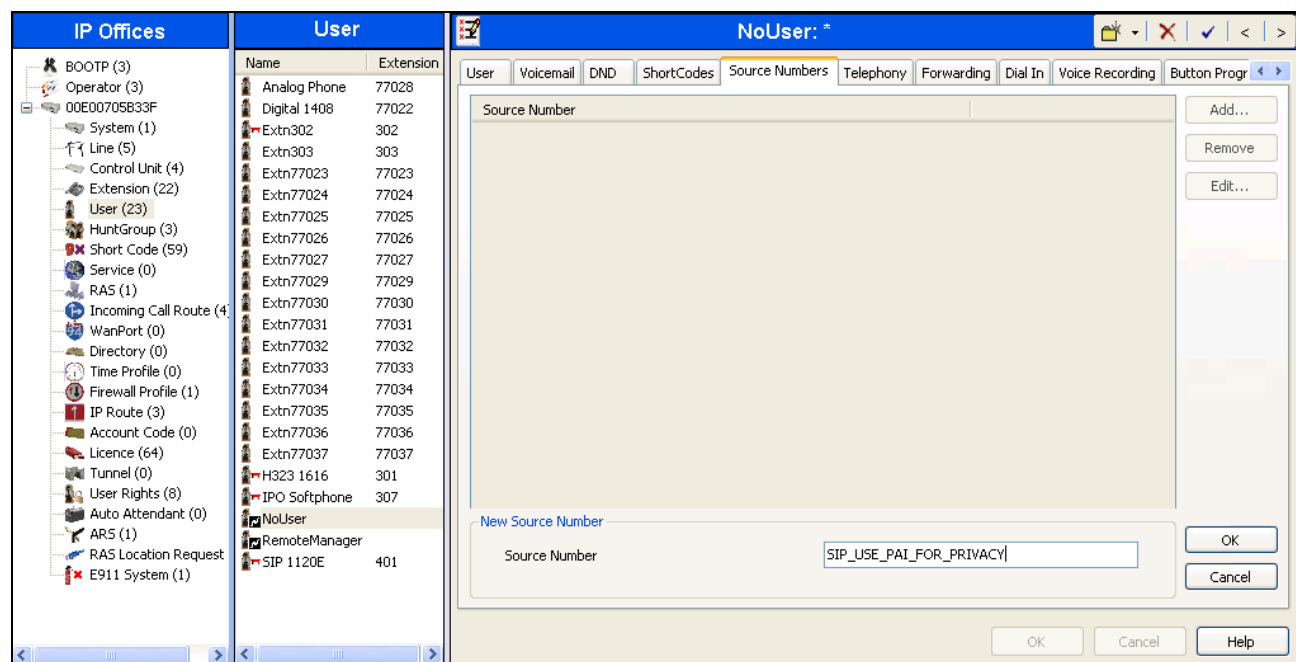
Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., 6N in **Section 5.5**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 6-1-303-555-1234, the call would be directed to Line Group 18, the SIP Line configured and described in these Application Notes. If Line Group 18 cannot be used, the call can automatically route to the route name configured in the **Additional Route** parameter in the lower right of the screen (not configured for the compliance test). Since alternate routing can be considered a privilege not available to all callers, Avaya IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

5.9. Privacy/Anonymous Calls

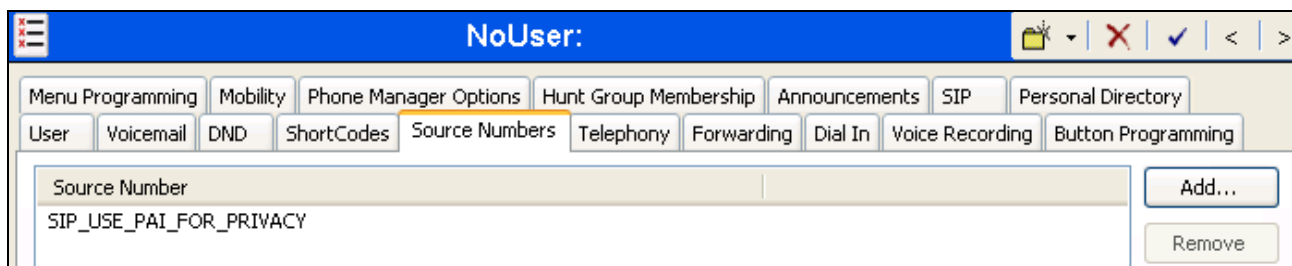
For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "restricted" and "anonymous" respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. For the compliance test, PAI was used for the purposes of privacy.

To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → NoUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.

At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_USE_PA1_FOR_PRIVACY**. Click **OK**.



The **SIP_USE_PAID_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.

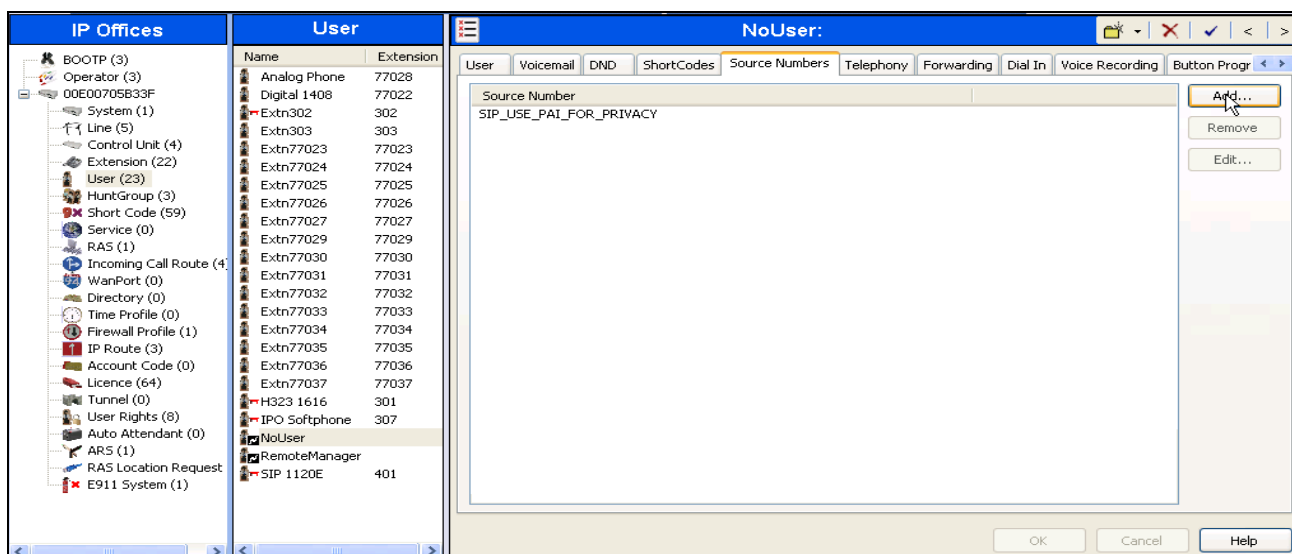


5.10. SIP Options

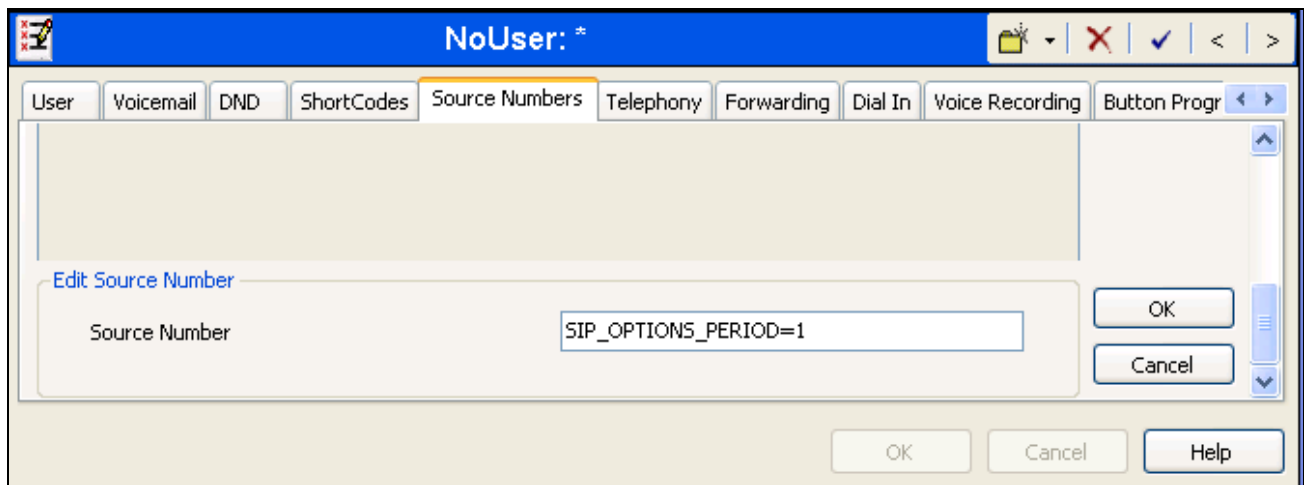
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.1** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **NoUser** user. The OPTIONS period is determined in the following manner:

- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 42 seconds is used.
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** or the **SIP_OPTIONS_PERIOD**.

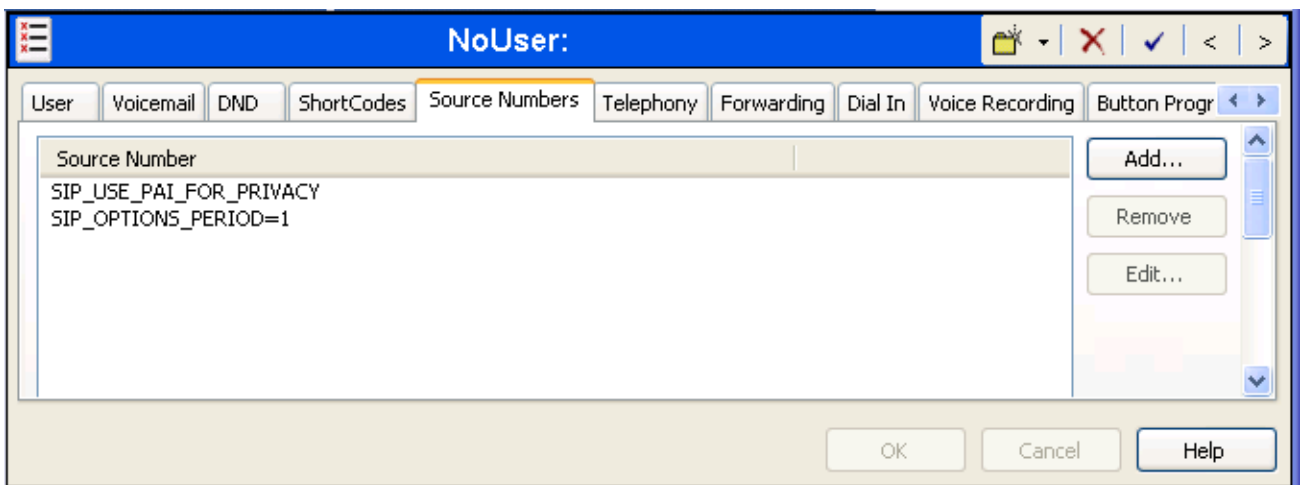
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → noUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_OPTIONS_PERIOD=X**, where *X* is the desired value in minutes. Click **OK**.



The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 1 minute was desired. The **Binding Refresh Time** was set to **60** seconds (1 minute) in **Section 5.1** and the **SIP_OPTIONS_PERIOD** was set to **1** minute. Avaya IP Office chose the OPTIONS period as the smaller of these two values (1 minute was the OPTIONS period since they were both set equal to 1 minute). Click the **OK** button (not shown).



5.11. 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.

6. Intelepeer SIP Trunking Configuration

Intelepeer is responsible for the configuration of Intelepeer SIP Trunking service. The customer will need to provide the IP address and / or FQDN used to reach the Avaya IP Office at the enterprise. Intelepeer will provide the customer the necessary information to configure the Avaya IP Office SIP connection to Intelepeer. The provided information from Intelepeer includes:

- IP address of the Intelepeer SIP proxy
- Intelepeer SIP domain
- Supported codecs
- DID numbers
- IP addresses, port numbers and transport protocol used for signaling or media through any security devices

7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is *Idle* for each channel (assuming no active calls at present time).

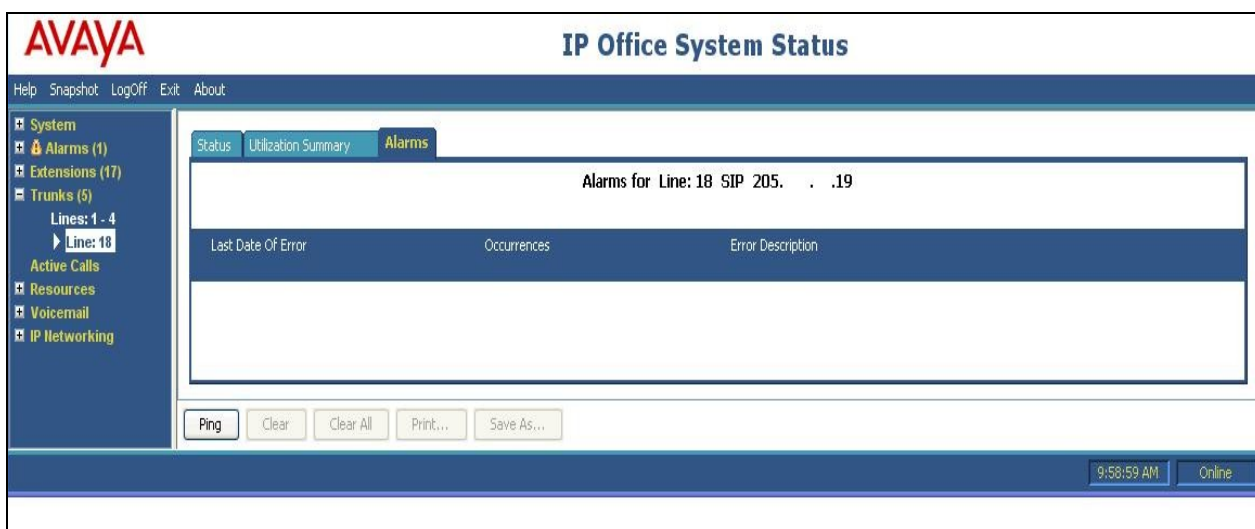
The screenshot shows the Avaya IP Office System Status application. The left pane displays a tree view with 'System' expanded, showing 'Alarms (1)', 'Extensions (17)', 'Trunks (5)', 'Lines: 1 - 4', 'Line: 18', 'Active Calls', 'Resources', 'Voicemail', and 'IP Networking'. The right pane shows the 'Status' tab for 'Line: 18'. The 'SIP Trunk Summary' section displays the following information:

- Peer Domain Name: 205. . .19
- Resolved Address: 68. . .41
- Line Number: 18
- Number of Administered Channels: 10
- Number of Channels in Use: 0
- Administered Compression: G729A
- Silence Suppression: Off
- SIP Trunk Channel Licences: Unlimited
- SIP Trunk Channel Licences in Use: 0
- SIP Device Features: 0%

Below the summary is a table with 15 columns: Channel Number, URI, Call Group Ref, Current State, Time in State, Remote RTP Address, Codec, Connection Type, Caller ID or Dialed Digits, Other Party on Call, Direction of Call, Round Trip Delay, Receive Jitter, Receive Pack Loss Fraction, Transmit Jitter, and Transmit Loss Fraction. The table contains 10 rows, all with 'Idle' as the 'Current State' and '5 days 18:18:01' as the 'Time in State'.

At the bottom of the application, there are buttons for 'Trace', 'Trace All', 'Pause', 'Ping', 'Call Details', 'Print...', and 'Save As...'. The status bar at the bottom right shows '9:51:22 AM' and 'Online'.

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



- Verify that a phone connected to the PSTN can successfully place a call to the Avaya IP Office with two-way audio.
- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
 1. Using a network sniffing tool (e.g., Wireshark), monitor the SIP signaling messages between Intelepeer and Avaya IP Office on an outbound call from the enterprise to the PSTN.

8. Conclusion

Intelepeer SIP Trunking passed compliance testing. These Application Notes describe the procedures required to configure the SIP trunk connection between Avaya IP Office and the Intelepeer SIP Trunking service as shown in **Figure 1**. Please refer to **Section 2.2** above for Test Results and any Limitations that were observed.

9. Additional References

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

- [1] IP Office 7.0 IP Office Standard Version Installation, Document number15-601042, March 2011.
- [2] IP Office Release 7.0 Manager 9.0, Document number15-601011, April 2011.
- [3] IP Office Release 7.0 Voicemail Pro Installation and Maintenance, Document Number 15-601063, March 2011.
- [4] IP Office Release 6.0 System Status Application, Document number15-601758, February 2010.
- [5] IP Office System Monitor, Document Number 15-601019, November 28, 2008

Product documentation for Avaya products may be found at <http://support.avaya.com>. Additional IP Office documentation can be found at:
<http://marketingtools.avaya.com/knowledgebase/>

Product documentation for Intelepeer SIP Trunking is available from Intelepeer.

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