



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Broadview Networks Broadspeed SIP Trunking with Avaya IP Office - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the Broadview Networks Broadspeed SIP Trunking Service and Avaya IP Office.

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

Broadview Networks 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 the Broadview Networks Broadspeed SIP Trunking service and Avaya IP Office.

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

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to Broadspeed SIP Trunking 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.

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.

2.1. Interoperability Compliance Testing

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

- Response to SIP OPTIONS queries
- Incoming PSTN calls to H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise over the SIP trunk from the service provider.
- Outgoing PSTN calls from H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise over the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from soft clients. Avaya IP Office supports two soft clients: Avaya IP Office Phone Manager and Avaya IP Office Video Softphone. Avaya IP Office Phone Manager supports two modes (PC softphone and telecommuter). Both clients were tested in their respective supported modes.
- Various call types including: local, long distance, international, and outbound toll-free
- Codec G.711MU and G.729A
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Response to incomplete call attempts and trunk errors
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning
- T.38 fax and fallback to G.711 fax
- REFER support for network transfer

Items not supported or not tested included the following:

- Inbound toll-free, operator (0), operator-assisted calls (0+10) and emergency calls (911) are supported but were not tested as part of the compliance test.

2.2. Test Results

Interoperability testing of BroadSpeed SIP Trunking was completed with successful results for all test cases with the exception of the observations/limitations described below. Testing passed using both the G.711Mu and G.729 codecs. However, it is recommended to use G.711Mu where possible due to the observations below related to the use of G.729.

- **403 Response to OPTIONS:** BroadSpeed SIP Trunking returned a “403 From URI not recognized” error in response to the OPTIONS message send by Avaya IP Office. This is due to the fact that Avaya IP Office does not send a Broadview Networks assigned DID in the From header of the message. This has no user impact since any response from the network indicates to Avaya IP Office that the trunk is in service.
- **RTP keepalives always use G.711:** If RTP keepalives are enabled, Avaya IP Office sends the keepalives using G.711Mu even if the call has negotiated the use of G.729.
- **G.729 Codec is not locked down:** On outbound G.729 calls from the enterprise, Broadview Networks return multiple codecs in the response to the initial INVITE instead of locking down the codec with a single codec in the response. Avaya IP Office selects G.729 from the list and the call is successful, thus this behavior has no user impact.
- **Use of silenceSupp parameter in the SDP:** Broadview Networks sends the **silenceSupp:off** in the SDP to indicate silence suppression off. Avaya IP Office currently does not support this parameter. In particular for G.729 calls, Avaya IP Office expects the **annexb=no** parameter to be used to indicate silence suppression. The inclusion of the **silenceSupp:off** parameter did not have an impact on user calls during the compliance testing. However, to avoid ambiguity that could lead to possible future failures, the best practice would be for Broadview Networks to send **annexb=no** to indicate silence suppression for G.729.
- **No error indication for outbound call with no matching codec:** If Avaya IP Office is misconfigured so that no supported codec is offered to Broadview Networks on an outbound call, Broadview Networks will not return a “488 Not Acceptable Here” response. Instead, Broadview Networks returns a 183 response with SDP including the G.711MU codec, followed by a 200 OK with the same SDP. The call gets established with no audio.
- **Voicemail Access:** Logging in to Voicemail Pro remotely from the PSTN to access a user’s voicemail sometimes took more than one attempt to be successful. This is currently being investigated by Avaya IP Office development.
- **T.38 Fax Version:** Avaya IP Office should be configured for T.38 version 0 to avoid bad checksum warnings that appeared in the network trace of these calls. These warnings did not impact the user’s fax calls. (See **Section 5.4**)
- **Transition to T.38 from a G.729 call:** On T.38 fax calls that transition from G.729, Broadview will begin with G.729 RTP then switch to G.711 RTP before transitioning to T.38. This does not impact the call and the fax proceeds normally as T.38.
- **Blind transfer with REFER:** Inbound calls from the PSTN to the enterprise which are blind transferred back to the PSTN are not supported by Broadview Networks if REFER is enabled.

2.3. Support

For technical support on BroadSpeed SIP Trunking, contact Broadview Networks via one of the following:

- Phone: 1-800-BROADVIEW (1-800-276-2384)
- Web: http://www.broadviewnet.com/contact_us/contact_us.asp

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. Selecting the **Support Contact Options** link followed by **Maintenance Support** provides the worldwide support directory for Avaya Global Services. Specific numbers are provided for both customers and partners based on the specific type of support or consultation services needed. Some services may require specific Avaya service support agreements. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to BroadSpeed SIP Trunking.

Located at the enterprise site is an Avaya IP Office 500. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public network. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 5600 Series IP Telephone (with H.323 firmware), an Avaya 9600 Series IP Telephone (with H.323 firmware), an Avaya IP Office Phone Manager, an Avaya IP Office Softphone, an Avaya 5420 Digital Telephone, an Avaya 6210 Analog Telephone and a fax machine. The site also has a Windows 2003 Server running Avaya Voicemail Pro for voicemail and running Avaya IP Office Manager to configure the Avaya IP Office.

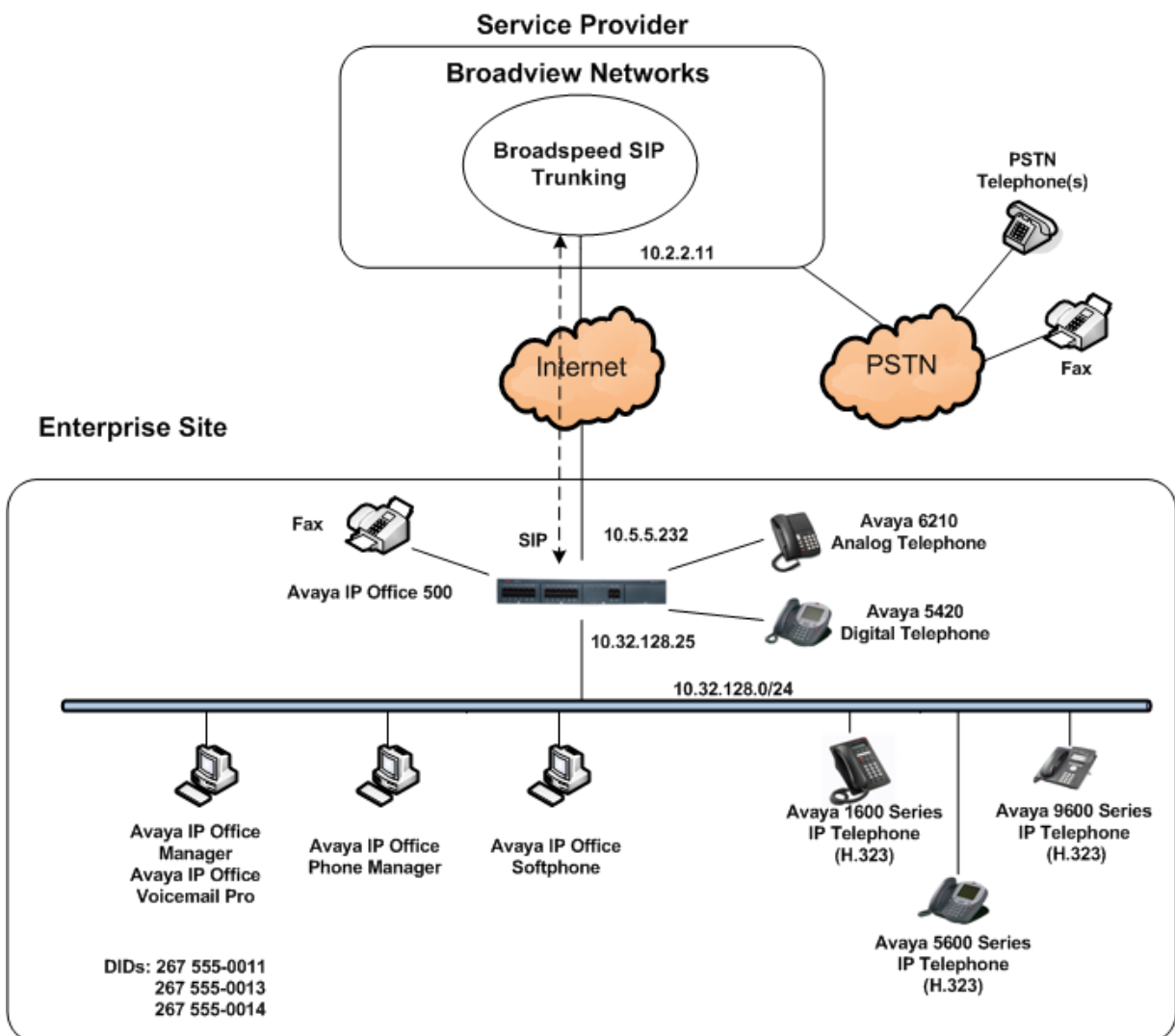


Figure 1: Test Configuration

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 can not be routed by 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 Broadview Networks. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Broadview Networks. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office sent 11 digits in the Request URI and the To header of an outbound SIP INVITE message. It was configured to send 10 digits in the From header. For inbound calls, Broadview Networks sent 10 digits in the Request URI and the To header 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/firmware were used for the sample configuration provided:

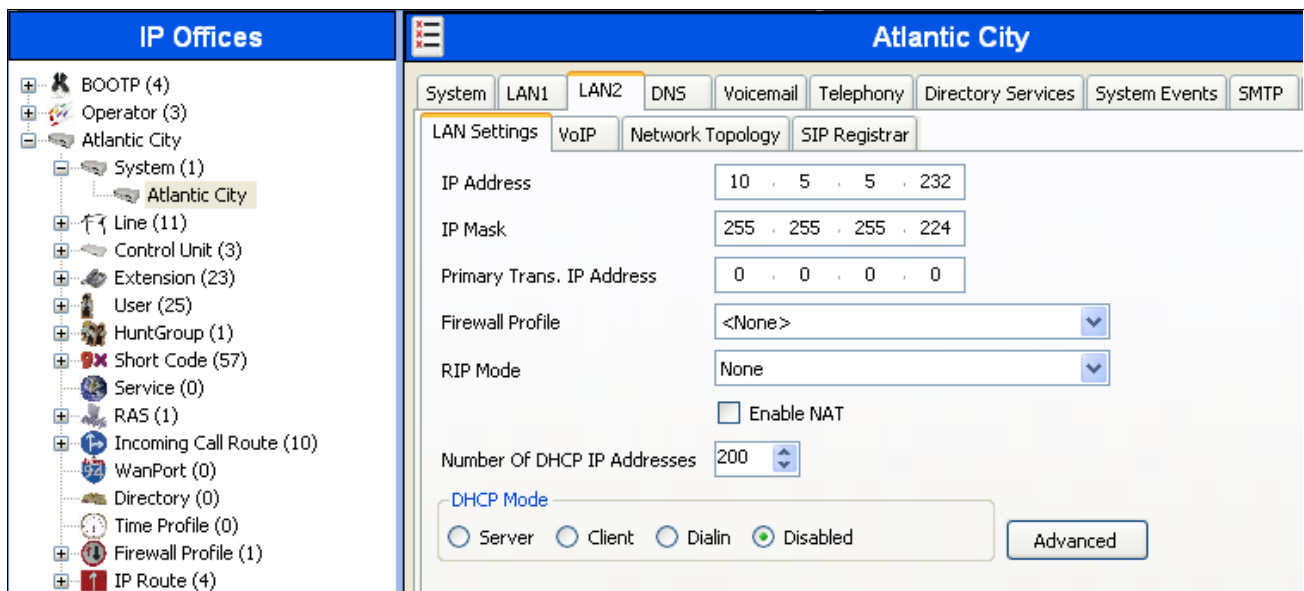
Avaya Telephony Components	
Equipment/Software	Release/Version
Avaya IP Office 500	8.0 (16)
Avaya IP Office Manager	10.0 (16)
Avaya Voicemail Pro	8.0 (Build 8.0.8.29)
Avaya 1608SW IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.300B
Avaya 5620 IP Telephone (H.323)	2.9.1
Avaya 9640SW IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.102S
Avaya IP Office Phone Manager	4.2.36
Avaya IP Office Softphone	3.2.3.15 (64595)
Avaya 5420 Digital Telephone	N/A
Avaya 6210 Analog Telephone	N/A
Broadspeed SIP Trunking Components	
Equipment	Release
Acme Packet Net-Net 4250 Session Border Controller	SC6.1.0 MR-9 Patch 2 (Build 954)
MetaSwitch Softswitch	7.3

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to Broadspeed SIP Trunking. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running the Avaya IP Office Manager application, select **Start → All Programs → IP Office → Manager** to launch the application. A pop-up window will appear with a list of Avaya IP Office systems reachable from the current Avaya IP Office Manager. Alternatively, navigate to **File → Open Configuration** to reach this window. 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 in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes 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 Avaya IP Office Softphone support) is assumed to already be in place.

5.1. LAN2 Settings

In the sample configuration, *Atlantic City* was used as the system name and 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. To access the LAN2 settings, first navigate to **Atlantic City → System → Atlantic City** in the Navigation Pane 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.



The screenshot displays the Avaya IP Office Manager configuration interface. On the left is the 'IP Offices' navigation pane, showing a tree structure with 'Atlantic City' selected. The main area is titled 'Atlantic City' and contains several tabs: 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', and 'SMTP'. The 'LAN2' tab is active, and within it, the 'LAN Settings' sub-tab is selected. The configuration fields are as follows:

Field	Value
IP Address	10 . 5 . 5 . 232
IP Mask	255 . 255 . 255 . 224
Primary Trans. IP Address	0 . 0 . 0 . 0
Firewall Profile	<None>
RIP Mode	None
Enable NAT	<input type="checkbox"/>
Number Of DHCP IP Addresses	200
DHCP Mode	Server <input type="radio"/> Client <input type="radio"/> Dialin <input type="radio"/> Disabled <input checked="" type="radio"/>

An 'Advanced' button is 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. In order to support inbound calls that are call forwarded back to the PSTN, RTP keepalives were necessary. In the RTP Keepalives section, set the **Scope** to **RTP**, set the **Periodic timeout** to the number of seconds requested between keepalives and set **Initial keepalives** to **Enabled**.

Atlantic City

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

LAN Settings **VoIP** Network Topology SIP Registrar

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

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

RTP Keepalives
 Scope RTP 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**. Since this parameter is set to **Open Internet**, then the **STUN Server IP Address** is not used.
- The **Binding Refresh Time** is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider if it has not received an OPTIONS request from the far-end. See **Section 5.9** for complete details. For the compliance test, the **Binding Refresh Time** was tested with values of **30** seconds and **120** seconds.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set the **Public Port** to **5060**.
- All other parameters should be set according to customer requirements.

The screenshot shows the 'Atlantic City' configuration interface. The 'Network Topology' tab is selected, and the 'Network Topology Discovery' section is visible. The settings are as follows:

Parameter	Value
STUN Server IP Address	10 . 90 . 168 . 13
STUN Port	3478
Firewall/NAT Type	Open Internet
Binding Refresh Time (seconds)	120
Public IP Address	10 . 5 . 5 . 232
Public Port	5060

Buttons: Run STUN, Cancel

Run STUN on startup

5.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For the compliance test, **ULAW** was 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.

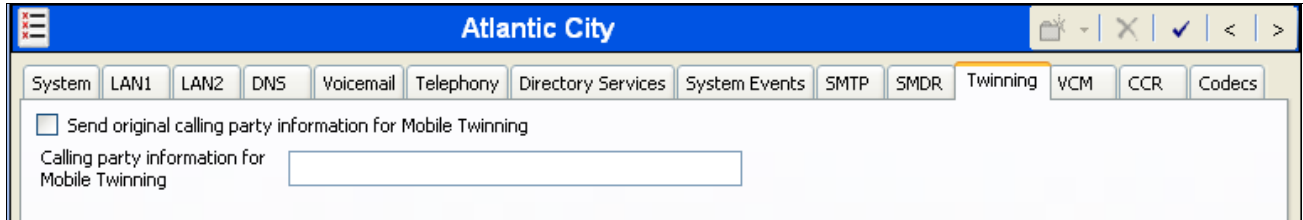
The screenshot shows the 'Atlantic City' configuration interface for 'Telephony'. The 'Companding Law' section is active, with 'U-Law' selected for both 'Switch' and 'Line'. The 'Inhibit Off-Switch Forward/Transfer' checkbox is unchecked. Other settings include 'Dial Delay Time (secs)' at 4, 'Dial Delay Count' at 0, 'Default No Answer Time (secs)' at 15, 'Hold Timeout (secs)' at 120, 'Park Timeout (secs)' at 300, 'Ring Delay (secs)' at 5, 'Call Priority Promotion Time (secs)' at Disabled, 'Default Currency' at USD, and 'Default Name Priority' at Favor Trunk. The 'Auto Hold', 'Dial By Name', 'Show Account Code', and 'High Quality Conferencing' checkboxes are checked.

Section	Parameter	Value
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	<input type="checkbox"/>
Companding Law	Switch	<input checked="" type="radio"/> U-Law <input type="radio"/> A-Law
	Line	<input checked="" type="radio"/> U-Law Line <input type="radio"/> A-Law Line
	DSS Status	<input type="checkbox"/>
	Auto Hold	<input checked="" type="checkbox"/>
	Dial By Name	<input checked="" type="checkbox"/>
	Show Account Code	<input checked="" type="checkbox"/>
	Inhibit Off-Switch Forward/Transfer	<input type="checkbox"/>
	Restrict Network Interconnect	<input type="checkbox"/>
	Drop External Only Impromptu Conference	<input type="checkbox"/>
	Visually Differentiate External Call	<input type="checkbox"/>
Unsupervised Analog Trunk Disconnect Handling	<input type="checkbox"/>	
High Quality Conferencing	<input checked="" type="checkbox"/>	
Other Settings	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
Default Name Priority	Favor Trunk	

5.3. Twinning Calling Party Settings

Navigate to the **Twining** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning and Call Forwarding to be controlled by the setting on the SIP Line (**Section 5.4**).

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



5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Broadspeed SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New** → **SIP Line** (not shown). On the **SIP Line** tab in the Details Pane, configure the parameters as shown below.

- Check the **In Service** box.
- Check the **Check OOS** box.
- Set the **Call Routing Method** to **Request URI**. Avaya IP Office will route inbound calls based on the number in the Request URI.
- Set **Send Caller ID** to **Diversion Header**. For forwarded or twinning calls, this setting results in the original calling party appearing in the SIP From header and the forwarding/twinning party in the Diversion header.
- Broadspeed SIP Trunking may be used with or without REFER for network call redirection. Using REFER allows Avaya IP Office to be removed from the middle of inbound calls that are redirected back to the PSTN. These redirected calls were tested with and without REFER enabled during the compliance test. Some redirected calls are not supported by Broadview Networks with REFER enabled. See **Section 2.2** for test results using REFER. To disable the use of REFER, uncheck the **REFER Support** box as shown below. To enable the use of REFER, check the **REFER Support** box and set both **Incoming** and **Outgoing** fields to **Always**.
- Default values may be used for all other parameters.

The screenshot displays the configuration for a SIP Line in the Avaya IP Office interface. The left-hand pane shows a tree view of IP Offices, with 'Line (11)' selected under 'Atlantic City'. The right-hand pane is titled 'SIP Line - Line 18' and contains the following configuration details:

- Line Number:** 18
- ITSP Domain Name:** (empty field)
- Prefix:** (empty field)
- National Prefix:** 0
- Country Code:** (empty field)
- International Prefix:** 00
- Send Caller ID:** Diversion Header
- Association Method:** By Source IP address
- REFER Support:** (unchecked box)
 - Incoming:** Always
 - Outgoing:** Always
- In Service:**
- Use Tel URI:**
- Check OOS:**
- Call Routing Method:** Request URI
- Originator number for forwarded and twinning calls:** (empty field)
- Name Priority:** System Default

Navigate to the **Transport** tab and set the following:

- Set the **ITSP Proxy Address** to the IP address of the Broadview Networks SIP Proxy provided by Broadview Networks.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN2** as configured in **Section 5.1**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.

The screenshot shows the configuration page for 'SIP Line - Line 18'. The 'Transport' tab is selected. The 'ITSP Proxy Address' is set to '10.2.2.11'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', 'Use Network Topology Info' is 'LAN 2', and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are both '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty.

ITSP Proxy Address	10.2.2.11
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
Explicit DNS Server(s)	0 . 0 . 0 . 0
Calls Route via Registrar	<input checked="" type="checkbox"/>
Separate Registrar	

Select the **SIP URI** tab, to create a SIP URI entry. A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. For the compliance test, a single SIP URI entry was created that matched any number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

- Set **Local URI** to **Use Internal Data**. This setting allows calls on this line whose SIP URI matches the **SIP Name** set on the **SIP** tab of any **User** as shown in **Section 5.6**.
- Set **Contact** and **Display Name** to **Use Internal Data**. This setting will cause the Contact and Display Name data to be set from the corresponding fields on the **SIP** tab of the individual **User** as shown in **Section 5.6**.
- Set **PAI** to **None**.
- For the **Registration** field, 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. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **18** was defined that only contains this line (line 18).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Default values may be used for all other parameters.

Click **OK**.

The screenshot shows the 'SIP Line - Line 18' 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, Max Calls. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is the 'New Channel' configuration form with the following fields:

Via	10.5.5.232
Local URI	Use Internal Data
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	None
Registration	0: <None>
Incoming Group	18
Outgoing Group	18
Max Calls per Channel	10

Buttons for 'OK', 'Cancel', and 'Help' are visible at the bottom of the window.

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

- Set the **Codec Selection** to **System Default**. The default codec list and their order are shown in the **Selected** box.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Set **Fax Transport Support** to **T.38 Fallback**. Avaya IP Office will use T.38 fax if possible. Otherwise, it will fallback to G.711 fax.
- 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 configuration window for 'SIP Line - Line 18'. The 'VoIP' tab is selected. The 'Codec Selection' dropdown is set to 'System Default'. Below this, there are two lists: 'Unused' (empty) and 'Selected' (containing 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 lists are navigation buttons: '>>', '<<', '<', '>', and '>>'. On the far right, there are five checkboxes: 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), 'Codec Lockdown' (unchecked), and 'PRACK/100rel Supported' (unchecked). Below the codec lists, the 'Fax Transport Support' dropdown is set to 'T38 Fallback', 'Call Initiation Timeout (s)' is set to '4', and 'DTMF Support' is set to 'RFC2833'.

On the **T38 Fax** tab, set the **T38 Fax Version** to **0**. This will avoid checksum warnings that appear when performing a network trace of the T.38 protocol (See **Section 2.2**). Leave the remaining parameters set to their default values.

The screenshot shows the configuration window for 'SIP Line - Line 18'. The 'T38 Fax' tab is selected. The configuration includes:

- T38 Fax Version:** 0 (dropdown)
- Transport:** UDPTL (dropdown)
- Redundancy:**
 - Low Speed: 0 (spin box)
 - High Speed: 0 (spin box)
- TCF Method:** Trans TCF (dropdown)
- Max Bit Rate (bps):** 14400 (dropdown)
- EFlag Start Timer (msecs):** 2600 (spin box)
- EFlag Stop Timer (msecs):** 2300 (spin box)
- Tx Network Timeout (secs):** 150 (spin box)
- Use Default Values

On the right side, there are several checkboxes and two spin boxes:

- Scan Line Fix-up
- TFOP Enhancement
- Disable T30 ECM
- Disable EFlags For First DIS
- Disable T30 MR Compression
- NSF Override
- Country Code:** 0 (spin box)
- Vendor Code:** 0 (spin box)

5.5. Short Code

Define a short code to route outbound traffic to the SIP line. 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"@10.2.2.11"**. 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 Broadview Networks SIP IP address follows the **@** sign in the above expression.
- 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.
- Default values may be used for all other parameters.

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

The screenshot displays the configuration interface for a Short Code. On the left is the 'IP Offices' navigation pane, and on the right is the 'Short Code' configuration pane for the '9N;; Dial' short code.

IP Offices	9N;; Dial
<ul style="list-style-type: none">BOOTP (4)Operator (3)Atlantic City<ul style="list-style-type: none">System (1)Line (11)Control Unit (3)Extension (23)User (25)HuntGroup (1)Short Code (57)Service (0)RAS (1)Incoming Call Route (10)	<p>Short Code</p> <p>Code: 9N;</p> <p>Feature: Dial</p> <p>Telephone Number: N"@10.2.2.11"</p> <p>Line Group ID: 18</p> <p>Locale: [Dropdown]</p> <p>Force Account Code: <input type="checkbox"/></p>

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 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 **Extn240**. Select the **SIP** tab in the Details Pane. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Broadview Networks. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. The value entered for the **Contact** field will be used in the SIP INVITE for outgoing SIP trunk calls. The value entered for the **SIP Name** is used as the user part of the SIP URI in the From header for outgoing SIP trunk calls.

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

The screenshot displays the Avaya user configuration interface. On the left is a navigation pane titled "IP Offices" with a tree view containing: BOOTP (4), Operator (3), Atlantic City, System (1), Line (11), Control Unit (3), Extension (23), User (25) (highlighted), HuntGroup (1), Short Code (57), and Service (0). The main pane is titled "Extn240: 240" and features a tabbed interface. The "SIP" tab is selected, showing the following configuration fields:

SIP Name	2675550011
SIP Display Name (Alias)	Extn240
Contact	2675550011

Below these fields is an unchecked checkbox labeled "Anonymous".

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, right-click **Incoming Call Routes** in the Navigation Pane and select **New** (not shown). 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. Matching is right to left.
- Default values can be used for all other fields.

IP Offices		18 2675550011	
+	BOOTP (4)	Standard	Voice Recording
+	Operator (3)	Destinations	
-	Atlantic City	Bearer Capability	Any Voice
+	System (1)	Line Group ID	18
+	Line (11)	Incoming Number	2675550011
+	Control Unit (3)	Incoming Sub Address	
+	Extension (23)	Incoming CLI	
+	User (25)	Locale	
+	HuntGroup (1)	Priority	1 - Low
+	Short Code (57)	Tag	
+	Service (0)	Hold Music Source	System Source
+	RAS (1)		
+	Incoming Call Route (10)		
+	WanPort (0)		
+	Directory (0)		
+	Time Profile (0)		
+	Firewall Profile (1)		
+	IP Route (4)		

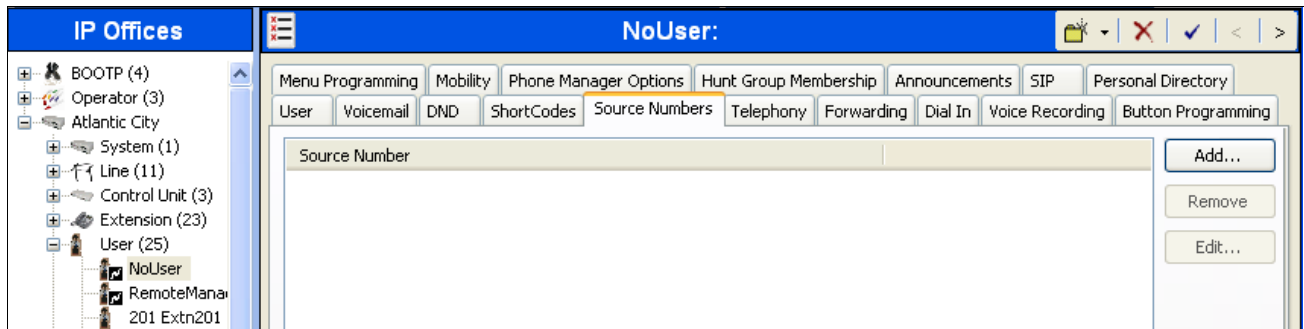
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 2675550011 on line 18 are routed to extension 240.

18 2675550011		
Standard	Voice Recording	
TimeProfile	Destination	Fallback Extension
▶ Default Value	240 Extn240	

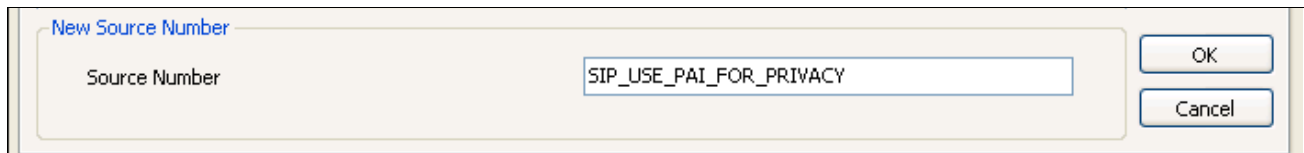
5.8. Privacy/Anonymous Calls

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. Broadview Networks supports both PPI and PAI for purposes of privacy. For the compliance test, both PPI and PAI were tested for the purposes of privacy.

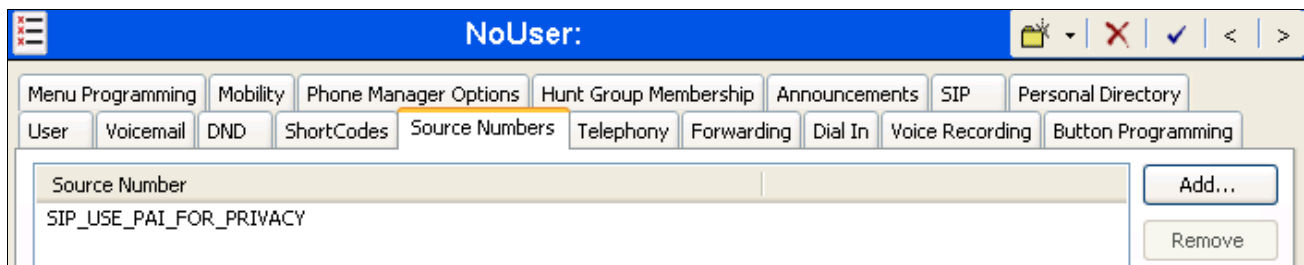
By default, Avaya IP Office will use PPI for privacy. If Avaya IP Office is to be configured to use PAI for privacy calls, then perform the following. Navigate to **User** → **NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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



The **SIP_USE_PA1_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.



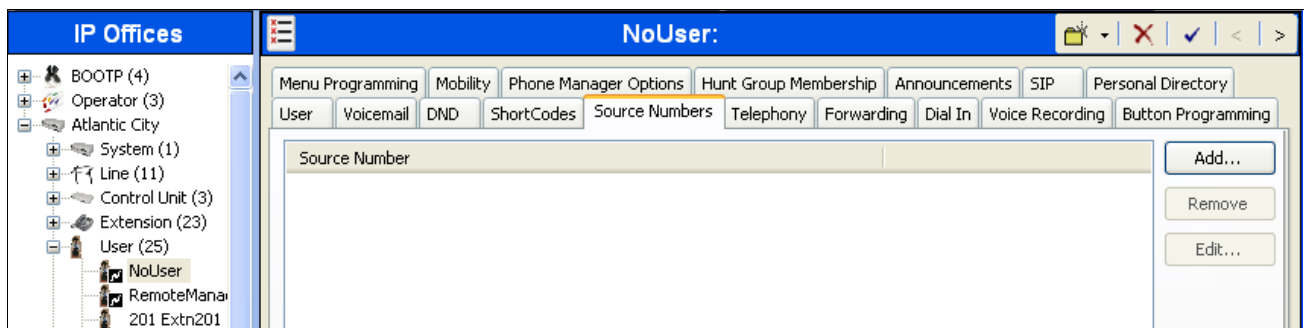
5.9. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. Avaya IP Office will send a SIP OPTIONS message only if an OPTIONS request is not received from the far-end during the defined interval. 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:

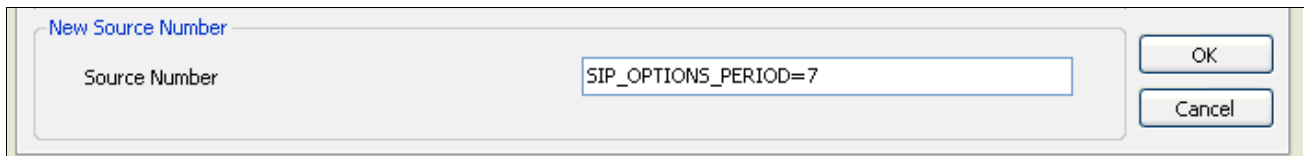
- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 300 seconds is used.
- To establish a period less than 300 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 300 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 seconds. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

For the compliance test, an OPTIONS period of 30 seconds and 2 minutes was tried. Thus, the **Binding Refresh Time** was set to **30** seconds or **120** seconds (2 minutes) in **Section 5.1** and no **SIP_OPTIONS_PERIOD** parameter was defined.

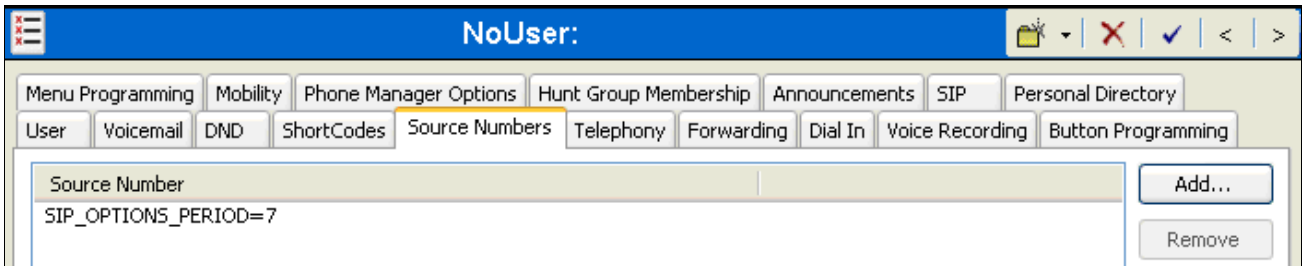
Alternatively, if an OPTIONS period greater than 300 seconds is desired then set the **Binding Refresh Time** to a value greater than 300 seconds and define the **SIP_OPTIONS_PERIOD** by doing the following. Navigate to **User** → **NoUser** in the Navigation Pane. Select the **Source Numbers** tab in 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. Click the **OK** button (not shown).



The resulting OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD** parameter.

5.10. 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. Broadspeed SIP Trunking Configuration

Broadview Networks is responsible for the configuration of Broadspeed SIP Trunking. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. Broadview Networks will provide the customer the necessary information to configure the Avaya IP Office SIP connection to Broadview Networks including:

- IP address of the Broadview Networks SIP proxy
- DID numbers to assign to users
- Supported codecs
- 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

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. 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.

The screenshot shows the Avaya IP Office System Status application. The left sidebar contains a tree view with categories: System, Alarms (3), Extensions (19), Trunks (14), Active Calls, Resources, Voicemail, and IP Networking. Under Trunks, lines 1-4, 5-8, 17, 18, and 19 are listed, with Line 18 selected. The main window displays the 'Status' tab for Line 18. The 'SIP Trunk Summary' section shows the following details:

- Peer Domain Name: sip://10.2.2.11
- Resolved Address: 10.2.2.11
- Line Number: 18
- 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: (indicated by a green circle and 0%)

Below the summary is a table with the following columns: Channel Number, URI, Call Ref, Current State, Time in State, Remote Media, Co..., Conn..., Caller ID or ..., Other Party on Call, Directi..., Round Trip ..., Receive Jitter, Receive Pack..., and Trans... T. The table contains 8 rows, all with 'Idle' as the current state and '00:50:54' as the time in state.

At the bottom of the window are buttons for Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As...

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

The screenshot shows the 'Alarms' tab for Line 18. The title is 'Alarms for Line: 18 SIP sip://10.2.2.11'. Below the title is a table with the following columns: Last Date Of Error, Occurrences, and Error Description. The table is currently empty, indicating no active alarms.

- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Verify that a phone connected to PSTN can successfully place a call to the Avaya IP Office with two-way audio.

8. Conclusion

Broadspeed SIP Trunking passed compliance testing with observations noted in **Section 2.2**. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and Broadview Networks Broadspeed SIP Trunking as shown in **Figure 1**.

9. Additional References

[1] *IP Office Documentation CD*, December 2011

[2] *IP Office Installation Manual*, Document number 15-601042, December 2011.

[3] *IP Office Manager Manual*, Document number 15-601011, December 2011.

[4] *IP Office System Status Application*, Document number 15-601758, November 2011.

Product documentation for Avaya products may be found at <http://support.avaya.com>.

Product documentation for Broadspeed SIP Trunking is available from Broadview Networks.

Appendix: SIP Line Template

Avaya IP Office Release 8.0 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.

Not all of the configuration information 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>20120305</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Broadview</DescriptiveName>
  <ITSPDomainName>10.2.2.11</ITSPDomainName>
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>false</ReferSupport>
  <ReferSupportIncoming>1</ReferSupportIncoming>
  <ReferSupportOutgoing>1</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <ITSPProxy>10.2.2.11</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_T38FB</FaxTransportSupport>
```

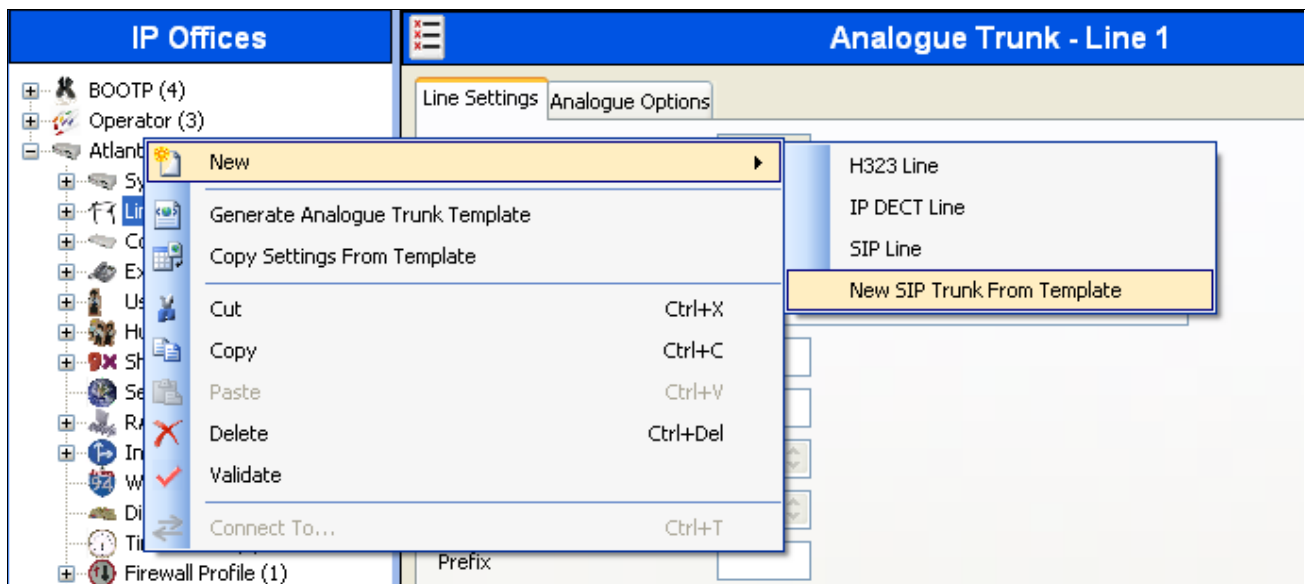
```

<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown>
<Rel100Supported>false</Rel100Supported>
<T38FaxVersion>0</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>false</UseDefaultValues>
<ScanLineFixup>true</ScanLineFixup>
<TFOPEnhancement>true</TFOPEnhancement>
<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 Avaya IP Office Manager is installed, copy and paste the above template into a text document named **US_Broadview_SIPTrunk.xml**. Move the .xml file to the Avaya IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates).
2. Import the template into an Avaya 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 **Broadview** 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: Broadview Display All

Create new SIP Trunk Cancel

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