



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

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# **Application Notes for Configuring Tidal Communications tNet Business VoIP with Avaya IP Office using SIP Registration - Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Tidal Communications tNet Business VoIP and Avaya IP Office using SIP registration.

Tidal Communications tNet Business VoIP provides PSTN access via a SIP trunk connected to the Tidal Communications Voice Over Internet Protocol (VoIP) network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Tidal Communications 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 Tidal Communications tNet Business VoIP and Avaya IP Office with SIP registration.

Tidal Communications tNet Business VoIP provides PSTN access via a SIP trunk connected to the Tidal Communications Voice Over Internet Protocol (VoIP) network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

## 1.1. Interoperability Compliance Testing

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

- SIP Registration to the network
- Response to SIP OPTIONS queries
- Incoming PSTN calls to various phone types
- Phone types included 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 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 soft clients
- Avaya IP Office supports two soft clients: Avaya IP Office Phone Manager and Avaya IP Office Softphone. Avaya IP Office Phone Manager supports two modes (PC softphone and telecommuter). Both clients in each supported mode were tested.
- Various call types including: local, long distance, international, outbound toll-free, operator and directory assistance (411)
- Codecs G.729A, and G.711MU
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- 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

Items not supported or not tested included the following:

- Inbound toll-free and emergency calls (911) are supported but were not tested as part of the compliance test.
- Dialing 0 for Operator is supported but dialing 0 + 10 digits is not.

Please refer to **Section 6** for complete test results, observations, limitations and any necessary workarounds.

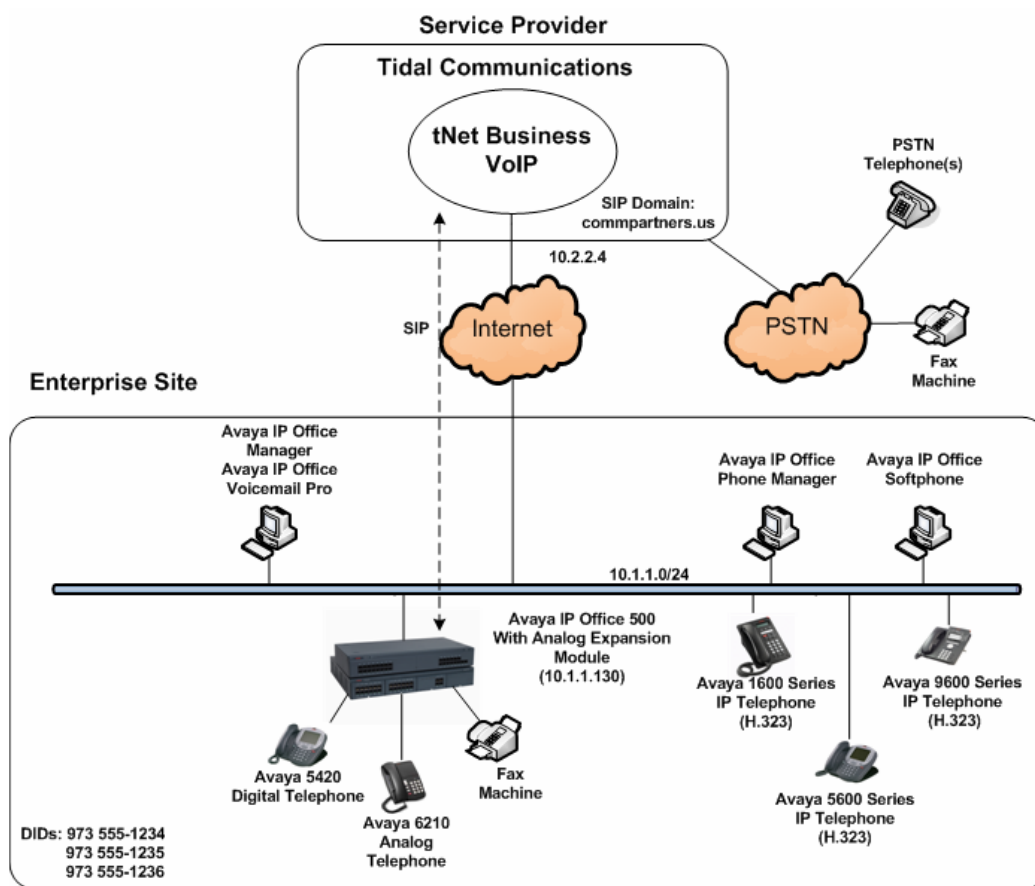
## 1.2. Support

For technical support on tNet Business VoIP, contact Tidal Communications using the contact and support links at [www.tidalcom.com](http://www.tidalcom.com).

## 2. Reference Configuration

**Figure 1** illustrates the test configuration. The test configuration shows an enterprise site connected to tNet Business VoIP.

Located at the enterprise site is an Avaya IP Office 500 with analog expansion module. 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 6211 Analog Telephone and 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.

Avaya IP Office registered with tNet Business VoIP using credentials provided by Tidal Communications. For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Tidal Communications. The short code of 9 is stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Tidal Communications. 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 field of an outbound SIP INVITE message. It was configured to send 10 digits in the From field. For inbound calls, tNet Business VoIP sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

Tidal Communications uses the phone number in the From header of a SIP INVITE message to authenticate the calling party. Thus, a call will be rejected by the network unless the From header contains a number known to Tidal Communications. This is especially important for calls inbound from the PSTN which are redirected back to the PSTN by call forwarding or twinning. For call forwarding, Avaya IP Office always sends the number of the forwarding phone in the From header. This is a number known to Tidal Communications. As a result, the call display on the destination phone shows the forwarding party not the original caller. For twinning, this behavior can be slightly altered through configuration. See **Section 4.3** and **4.4** for details.

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.

### 3. 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 500 with Analog Expansion Module	6.0 (8)
Avaya IP Office Analog Expansion Module	8.0 (8)
Avaya Voicemail Pro	6.0.22
Avaya IP Office Manager	8.0 (8)
Avaya 1608SW IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.2.2
Avaya 5620 IP Telephone (H.323)	2.9.1
Avaya 9640SW IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.1
Avaya IP Office Phone Manager	4.2.25
Avaya IP Office Softphone	3.0
Avaya 5420 Digital Telephone	N/A
Avaya 6211 Analog Telephone	N/A

Tidal Communications Components	
Equipment	Release
Acme Packet 4250 Net-Net Session Director Session Border Controller	v6.x
Broadsoft Broadworks	r14

## 4. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to tNet Business VoIP. 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 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 IP Office Softphone support) is assumed to already be in place.

### 4.1. LAN1 Settings

In the sample configuration, the MAC address **00E007026FBA** was used as the system name and the LAN1 port was used to connect the Avaya IP Office to the enterprise network. To access the LAN1 settings, first navigate to **System → 00E007026FBA** in the Navigation Pane and then navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are set from values shown in **Figure 1**. All other parameters should be set according to customer requirements.

The screenshot displays the Avaya IP Office Manager interface. On the left is the 'IP Offices' Navigation Pane, showing a tree structure with '00E007026FBA' selected. The main area on the right is the 'Details Pane' for system '00E007026FBA'. It features a tabbed interface with 'LAN1' selected, and within it, the 'LAN Settings' tab is active. The settings include:
 

- IP Address:** 10 . 1 . 1 . 130
- IP Mask:** 255 . 255 . 255 . 0
- Primary Trans. IP Address:** 0 . 0 . 0 . 0
- RIP Mode:** None (dropdown menu)
- Enable NAT:** Unchecked checkbox
- Number Of DHCP IP Addresses:** 200 (spin box)
- DHCP Mode:** Radio buttons for Server, Client, Dialin, and Disabled (Disabled is selected).
- Advanced:** A button to expand further settings.

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 LAN1. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements.

The screenshot shows the Avaya IP Office configuration window for system 00E007026FBA. The 'VoIP' tab is selected, and the 'SIP Registrar' sub-tab is active. The configuration includes several checkboxes and numeric input fields.

Setting	Value
H323 Gatekeeper Enable	<input checked="" type="checkbox"/>
SIP Trunks Enable	<input checked="" type="checkbox"/>
SIP Registrar Enable	<input checked="" type="checkbox"/>
H323 Auto-create Extn	<input checked="" type="checkbox"/>
H323 Auto-create User	<input type="checkbox"/>
Enable RTCP Monitoring On Port 5005	<input checked="" type="checkbox"/>
<b>DiffServ Settings</b>	
DSCP(Hex)	B8
DSCP Mask (Hex)	FC
SIG DSCP (Hex)	88
DSCP	46
DSCP Mask	63
SIG DSCP	34
<b>DHCP Settings</b>	
Primary Site Specific Option Number (SSON)	176
Secondary Site Specific Option Number (SSON)	242
VLAN	Not Present

The RTP Port Number Range is configured as follows:

Field	Value
Port Range (Minimum)	49152
Port Range (Maximum)	53246

Buttons at the bottom: OK, Cancel, Help.

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

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***.
- Set **Binding Refresh Time (seconds)** to **300**. 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 4.9** for complete details.
- Set **Public IP Address** to the IP address of LAN1.
- Set **Public Port** to **5060**.
- All other parameters should be set according to customer requirements.

The screenshot displays the Avaya IP Office configuration window with the title bar '00E007026FBA\*'. The 'Network Topology' tab is selected under the 'VoIP' category. The 'Network Topology Discovery' section contains the following settings:

- STUN Server IP Address: 0 . 0 . 0 . 0
- STUN Port: 3478
- Firewall/NAT Type: Open Internet (selected from a dropdown menu)
- Binding Refresh Time (seconds): 300
- Public IP Address: 10 . 1 . 1 . 130
- Public Port: 5060

At the bottom right of the configuration area, there are two buttons: 'Run STUN' and 'Cancel'. Below these buttons is a checkbox labeled 'Run STUN on startup', which is currently unchecked.

## 4.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** Tab on 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.

00E007026FBA

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

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

Dial Delay Time (secs) 4

Dial Delay Count 0

Default No Answer Time (secs) 15

Hold Timeout (secs) 0

Park Timeout (secs) 300

Ring Delay (secs) 5

Call Priority Promotion Time (secs) Disabled

Default Currency USD

Automatic Codec Preference G.711 ULAW 64K

Companding Law

Switch

ULAW

ALAW

Line

ULAW Line

ALAW Line

DSS Status

Auto Hold

Dial By Name

Show Account Code

Inhibit Off-Switch Forward/Transfer

Restrict Network Interconnect

Drop External Only Impromptu Conference

Visually Differentiate External Call

## 4.3. System Twinning Settings

Navigate to the **Twinning** tab, check the box labeled **Send original calling party information for Mobile Twinning**. With this setting, Avaya IP Office will send the original calling party number to the twinned phone in the SIP From header (not the associated desk phone number) for calls that originate from an internal extension. For calls to a twinned enabled phone that originate from the PSTN, Avaya IP Office will continue to send the associated host phone number in the SIP From header (used for the caller display) to the twinned destination. This setting only affects twinning and does not impact the messaging of other redirected calls such as forwarded calls. If this box is checked, it will also override any setting of the **Send Caller ID** parameter on the SIP line (**Section 4.4**).

00E007026FBA

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

☒ Send original calling party information for Mobile Twinning

Calling party information for Mobile Twinning



## 4.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and tNet Business VoIP. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New → SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters below to connect to tNet Business VoIP.

- Set the **ITSP Domain Name** to the domain name provided by Tidal Communications.
- Set **ITSP IP Address** to the IP address of the Tidal Communications SIP proxy.
- Set **Send Caller ID** to **None**. This parameter determines how the calling party number is sent in the SIP messaging for twinning if the box labeled **Send original calling party information for Mobile Twinning** is unchecked in **Section 4.3**. Choices include **None**, **Diversion Header**, **P Asserted Identity**, and **Remote Party ID**. For the compliance test, this parameter was set to **None** and the box in **Section 4.3** was checked.
- Check the **Registration Required** box.
- Check the **In Service** box.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to the network port configured in **Section 4.1**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane, showing a tree structure with 'Line (8)' selected. The main area is titled 'SIP Line - Line 21' and contains several tabs: 'SIP Line', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP Line' tab is active, showing the following configuration fields:

Field	Value	Field	Value
Line Number	21	Registration Required	<input checked="" type="checkbox"/>
ITSP Domain Name	commpartners.us	In Service	<input checked="" type="checkbox"/>
ITSP IP Address	10 . 2 . 2 . 4	Use Tel URI	<input type="checkbox"/>
Prefix		Check OOS	<input checked="" type="checkbox"/>
National Prefix	0	Call Routing Method	Request URI
Country Code			
International Prefix	00		
Send Caller ID	None		

Below these fields is a 'Network Configuration' section with the following settings:

Field	Value	Field	Value
Layer 4 Protocol	UDP	Send Port	5060
Use Network Topology Info	LAN 1	Listen Port	5060

Since Avaya IP Office will register with tNet Business VoIP, SIP credentials must be defined. To create a SIP credential entry, first select the **SIP Credentials** tab. Click the **Add** button and the **New SIP Credentials** area will appear at the bottom of the pane. For the compliance test, a single SIP credential was created with the parameters shown below.

- Set **User name**, **Authentication Name**, and **Password** to values provided by Tidal Communications. These values are usually one of the DID numbers provided by Tidal Communications and is not used for any other purpose than for registering the SIP trunk.
- Set the **Expiry** field, to the length of time (in seconds) that the registration is requested to be valid. The actual registration expiration time is negotiated as part of the registration exchange.

The screenshot shows the 'SIP Line - Line 21' configuration window with the 'SIP Credentials' tab selected. The window contains a table with columns: Index, UserName, Authentication Name, Password, and Expiry. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is a 'New SIP Credentials' section with input fields for 'User name', 'Authentication Name', 'Password', and 'Expiry'. The 'User name', 'Authentication Name', and 'Password' fields contain the value '9735551234'. The 'Expiry' field contains the value '600'. To the right of these fields are 'OK' and 'Cancel' buttons.

Index	UserName	Authentication Name	Password	Expiry
-------	----------	---------------------	----------	--------

New SIP Credentials

User name: 9735551234

Authentication Name: 9735551234

Password: 9735551234

Expiry: 600

After the SIP credentials are defined, each SIP URI that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. 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**, **Contact** and **Display Name** to *Internal Data*. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 4.6**.
- For **Registration**, select **1: 9735551234** from the pull-down menu. This matches the SIP credentials previously created on the **SIP Credentials** tab.
- 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 **21** was defined that only contains this line (line 21).
- 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 21' configuration window. The 'SIP URI' tab is selected, displaying a table with columns: Channel, Groups, Via, Local URI, and Contact. Below the table is a 'New Channel' section with the following fields and values:

- Via: 10.1.1.130
- Local URI: Use Internal Data
- Contact: Use Internal Data
- Display Name: Use Internal Data
- Registration: 1: 9735551234
- Incoming Group: 21
- Outgoing Group: 21
- Max Calls per Channel: 10

Buttons for 'Add...', 'Remove', 'Edit...', 'OK', and 'Cancel' are located on the right side 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.

- Configure the **Compression Mode** with the **Advanced** button to specify the preferred order of the offered codecs. Select the codecs and their order based on the needs of the customer. Click the box next to the codec with the highest preference first, followed by the second preference, etc. For the compliance test, **G.729(a) 8K CS-ACELP** was selected first followed by **G.711 ULAW 64K**.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Fax Transport Support** box to allow T.38 fax operation.
- Check the **Re-invite Supported** box, to allow for codec re-negotiation in cases where the target of the incoming call or transfer does not support the codec originally negotiated on the trunk.
- Default values may be used for all other parameters.

The screenshot shows the 'SIP Line - Line 21' configuration window with the 'VoIP' tab selected. The 'Compression Mode' is set to 'Advanced'. A list of codecs is shown with checkboxes: ☒ G.729(a) 8K CS-ACELP, ☒ G.711 ULAW 64K, ☐ G.711 ALAW 64K, and ☐ G.723.1 6K3 MP-MLQ. On the right, ☐ VoIP Silence Suppression is unchecked, ☒ Fax Transport Support is checked, ☒ Re-invite Supported is checked, and ☐ Use Offerer's Preferred Codec is unchecked. The 'Call Initiation Timeout (s)' is set to 4, and 'DTMF Support' is set to RFC2833.

Select the **T.38 Fax** tab, to set the T.38 parameters for the line. Verify that the **Use Default Values** box is checked. Click the **OK** button (not shown) to complete the SIP line configuration.

The screenshot shows the 'SIP Line - Line 21' configuration window with the 'T38 Fax' tab selected. 'T38 Fax Version' is 3, 'Transport' is UDPTL. Under 'Redundancy', 'Low Speed' and 'High Speed' are both 0. 'TCF Method' is Trans TCF, 'Max Bit Rate (bps)' is 14400, 'EFlag Start Timer (msecs)' is 2600, 'EFlag Stop Timer (msecs)' is 2300, and 'Tx Network Timeout (secs)' is 150. On the right, ☒ Scan Line Fix-up, ☒ TFOP Enhancement, ☐ Disable T30 ECM, ☐ Disable EFlags For First DIS, and ☐ Disable T30 MR Compression are all unchecked. Under 'NSF Override', 'Country Code' and 'Vendor Code' are both 0. At the bottom left, ☒ Use Default Values is checked.

## 4.5. 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**. 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"@commpartners.us"**. 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.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 4.4**. This short code will use this line group when placing the outbound call.

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

The screenshot displays the Avaya SIP Office configuration interface. On the left is the 'IP Offices' navigation pane, which includes a tree view with the following items: BOOTP (5), Operator (3), 00E007026FBA, System (1), Line (8), Control Unit (6), Extension (74), User (76), HuntGroup (1), Short Code (66) (highlighted with a blue selection bar), Service (0), and RAS (1). The main area on the right is titled '9N;: Dial' and contains a 'Short Code' configuration tab. The configuration fields are as follows:

Field	Value
Code	9N;
Feature	Dial
Telephone Number	N"@commpartners.us"
Line Group Id	21
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>

## 4.6. User

Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 4.4**. To configure these settings, first navigate to **User** in the Navigation Pane. 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 and allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 4.4**). As such, these fields should be set to one of the DID numbers assigned to the enterprise from Tidal Communications. In the example below, the DID number **9735551236** is used. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Click the **OK** button (not shown).

The screenshot displays the Avaya SIP configuration interface. On the left is the 'IP Offices' navigation pane with a tree structure including BOOTP (5), Operator (3), 00E007026FBA, System (1), Line (8), Control Unit (6), Extension (74), User (76) (highlighted), HuntGroup (1), Short Code (66), and Service (0). The main area is titled 'Ext370: 370' and contains a tabbed interface. The 'SIP' tab is selected, showing configuration fields: 'SIP Name' with value '9735551236', 'SIP Display Name (Alias)' with value 'Ext370', and 'Contact' with value '9735551236'. There is an 'Anonymous' checkbox which is unchecked. Other tabs visible include User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Buttons, Menu Programming, Mobility, Phone Manager Options, Hunt Group Membership, Announcements, and Personal.

## 4.7. 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 except the DID number used for trunk registration. To create an incoming call route, right-click **Incoming Call Routes** in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below.

- Set the **Bearer Capability** to *Any Voice*.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 4.4**.
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left.
- Default values can be used for all other fields.

The screenshot shows the 'IP Offices' navigation pane on the left with 'Incoming Call Route (31)' selected. The main pane displays the configuration for '21 9735551236' on the 'Standard' tab. The fields are as follows:

Field	Value
Bearer Capability	Any Voice
Line Group Id	21
Incoming Number	9735551236
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. Click the **OK** button (not shown). In this example, incoming calls to 9735551236 on line 21 are routed to extension 370.

The screenshot shows the 'Destinations' tab for the configuration '21 9735551236'. It contains a table with the following data:

TimeProfile	Destination	Fallback Extension
Default Value	370 Extn370	

## 4.8. 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. tNet Business VoIP

does not support PPI or PAI for the purposes of privacy. However, this functionality can be achieved by dialing feature codes that are interpreted by tNet Business VoIP. To place a privacy/anonymous call, the user dials 9 (Avaya IP Office short code) + \*67 (network feature code) + destination number. The following network feature codes provided by Tidal Communications are available to Avaya IP Office users to control the delivery of the calling party number to the far-end.

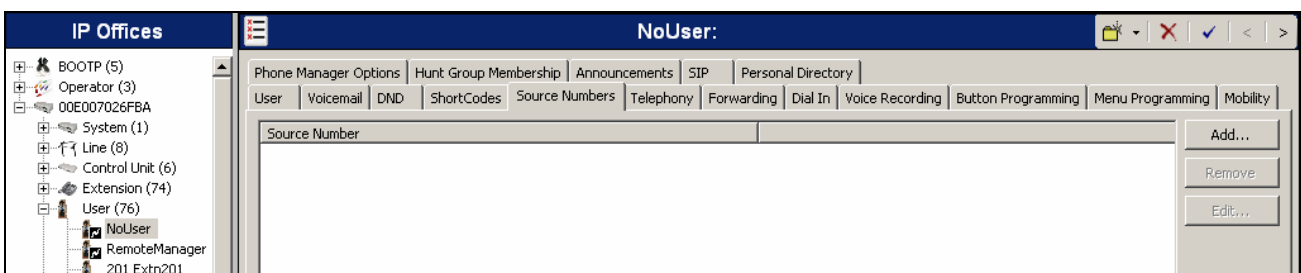
- \*67 Calling Line ID Delivery Blocking per Call
- \*65 Calling Line ID Delivery per Call
- \*31 Calling Line ID Delivery Blocking Persistent Activation
- #31 Calling Line ID Delivery Blocking Persistent Deactivation

## 4.9. SIP Options

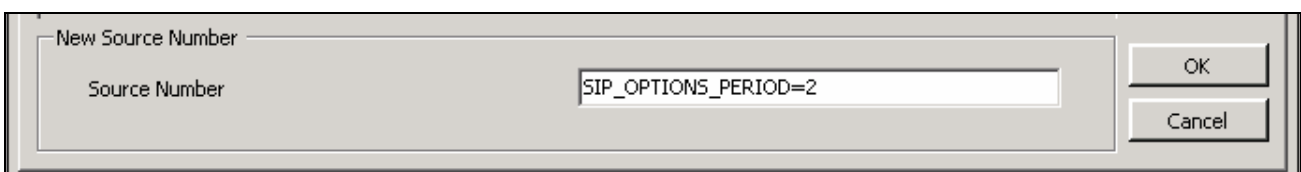
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 4.1** and the **SIP\_OPTIONS\_PERIOD** 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 44 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** and the **SIP\_OPTIONS\_PERIOD**.

To configure the **SIP\_OPTIONS\_PERIOD** parameter, navigate to **User** → **noUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.

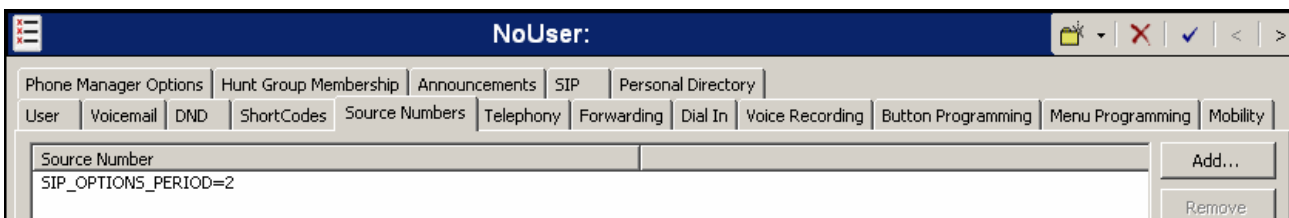


At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP\_OPTIONS\_PERIOD=X**, where **X** is the desired value in minutes. Click **OK**.





The **SIP\_OPTIONS\_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 2 minutes was desired. The **Binding Refresh Time** was set to **300** seconds (5 minutes) in **Section 4.1**. The **SIP\_OPTIONS\_PERIOD** was set to 2 minutes. Avaya IP Office chose the OPTIONS period as the smaller of these two values (2 minutes). Click the **OK** button (not shown).



## 4.10. 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.

## 5. tNet Business VoIP Configuration

Tidal Communications is responsible for the configuration of tNet Business VoIP. The customer will need to provide the public IP address used to reach the Avaya IP Office at the enterprise. Tidal Communications will provide the customer the necessary information to configure the SIP connection to tNet Business VoIP including:

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

## 6. General Test Approach and Test Results

This section describes the general test approach used during compliance testing and the test results.

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to tNet Business VoIP. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 1.1**.

tNet Business VoIP passed compliance testing. The following observations were made during the compliance test:

- **SIP Registration Interval:** As part of the SIP REGISTER message exchange, the network provides an interval in seconds that the current registration is valid (Expires parameter). In order to ensure that this interval is not exceeded, Avaya IP Office will re-register at ½ this time. Tidal Communications would prefer that Avaya IP Office did not re-register until near the end of this interval.

- **Privacy Calls:** To support privacy calls (calling party number blocking), Avaya IP Office sends “anonymous” as the calling number in the SIP From header and uses either the PPI or PAI header to pass the actual calling party number. tNet Business VoIP does not support PPI or PAI to provide this feature from Avaya IP Office. Instead, this functionality is made available directly from Tidal Communications using network feature codes. See **Section 4.8** for full details.

## 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.

**AVAYA IP Office System Status**

Help Snapshot LogOff Exit About

**System**  
**Alarms (7)**  
**Extensions (42)**  
**Trunks (9)**  
 Line: 9  
 Line: 10  
 Line: 17  
 Line: 18  
 Line: 19  
 Line: 20  
**Line: 21**  
 Line: 22  
 Line: 23  
**Active Calls**  
**Resources**  
**Voicemail**  
**IP Networking**

**Status** Utilization Summary Alarms Registration

**SIP Trunk Summary**

Peer Domain Name: commpartners.us  
 Gateway Address: 10.2.2.4  
 Line Number: 21  
 Number of Administered Channels: 4  
 Number of Channels in Use: 0  
 Administered Compression: Auto  
 Silence Suppression: Off  
 SIP Trunk Channel Licences: Unlimited  
 SIP Trunk Channel Licences in Use: 0  
 SIP Device Features:

Channel Number	UR	Call Grc	Call Ref	Current State	Time in State	Remote Address	Code	Connex Type	Caller I Dialed	Other Party on Call	Directic of Call	Round Delay	Receiv Jitter	Receiv Loss Fr	Transm Jitter	Trar Loss
1				Idle	00:01:24											
2				Idle	00:01:24											
3				Idle	00:01:24											
4				Idle	00:01:24											

Trace Trace All Pause Ping Call Details Print... Save As...

7:30:11 PM Online

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

Status	Utilization Summary	Alarms	Registration
Alarms for Line: 21 SIP commpartners.us			
Last Date Of Error	Occurrences	Error Description	

- 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

tNet Business VoIP passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and tNet Business VoIP as shown in **Figure 1**.

## 9. Additional References

- [1] *IP Office 6.0 Documentation CD*, February 2010.
- [2] *IP Office Installation*, Document number15-601042, May 2010.
- [3] *IP Office Manager*, Document number15-601011, May 2010.
- [4] *System Status Application*, Document number15-601758, February 2010.

Product documentation for Avaya products may be found at <http://support.avaya.com>.  
Product documentation for tNet Business VoIP is available from Tidal Communications.

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