



Application Notes for Configuring PAETEC Dynamic IP SIP Trunk Service using the BroadSoft Platform with Avaya IP Office and SIP Registration - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between PAETEC Dynamic IP SIP Trunk Service and Avaya IP Office using SIP registration. PAETEC can offer the Dynamic IP SIP Trunk Service using several different platform technologies in the PAETEC network. These Application Notes correspond to the Dynamic IP SIP Trunk Service offered using a BroadSoft platform in the network.

PAETEC Dynamic IP SIP Trunk Service provides PSTN access via a SIP trunk connected to the PAETEC 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.

PAETEC 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 PAETEC Dynamic IP SIP Trunk Service and Avaya IP Office using SIP registration. PAETEC can offer the Dynamic IP SIP Trunk Service using several different platform technologies in the PAETEC network. These Application Notes correspond to the Dynamic IP SIP Trunk Service offered using a BroadSoft platform in the network.

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1.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office was connected to the Dynamic IP SIP Trunk Service. 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 assisted calls and directory assistance (411)
- Codecs G.729A, G.711MU and G.711A.
- 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

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.
- T.38 Fax is not supported.

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

1.2. Support

For technical support on the Dynamic IP SIP Trunk Service, contact PAETEC using the Customer Care links at www.paetec.com.

2. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Dynamic IP SIP Trunk Service.

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, and an Avaya 6210 Analog Telephone. 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.

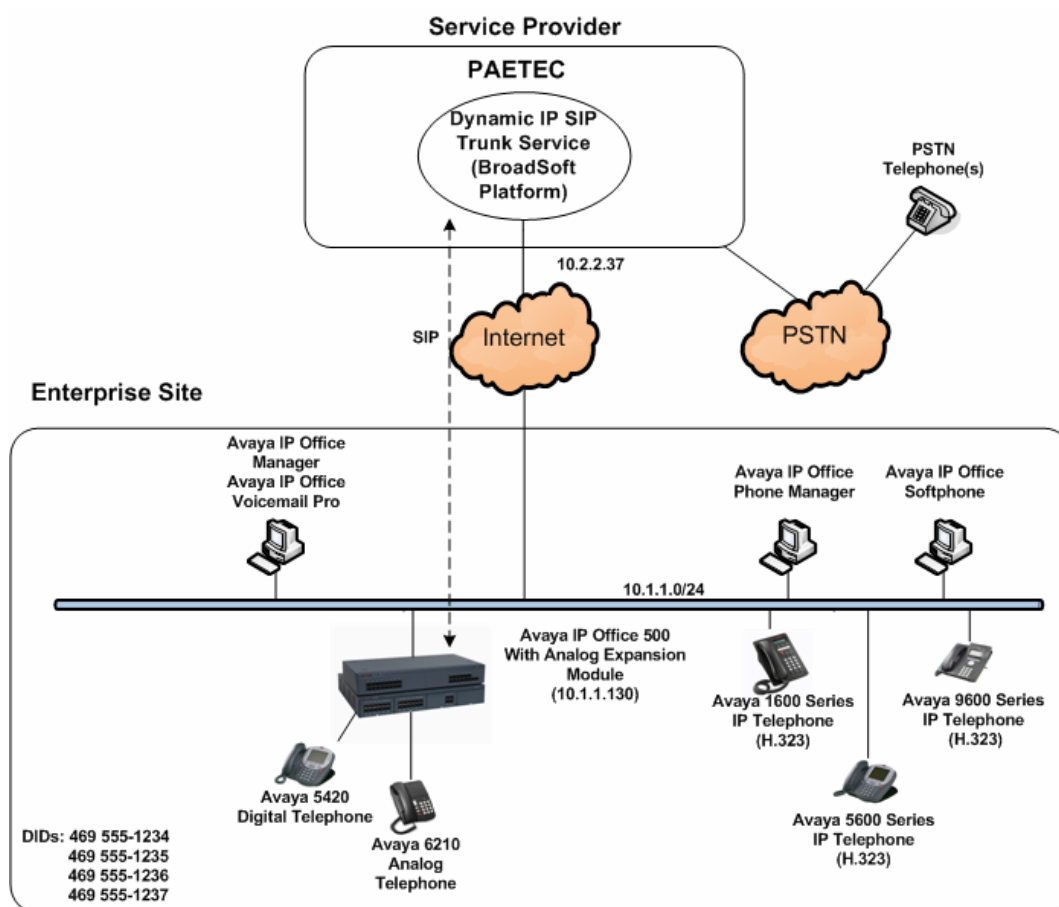


Figure 1: Test Configuration

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 the Dynamic IP SIP Trunk Service using credentials provided by PAETEC. For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to PAETEC. The short code of 9 is stripped off by Avaya IP Office but the remaining N digits were sent unaltered to PAETEC. 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, the Dynamic IP SIP Trunk Service sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

PAETEC 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 PAETEC. 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 PAETEC. 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 6210 Analog Telephone	N/A

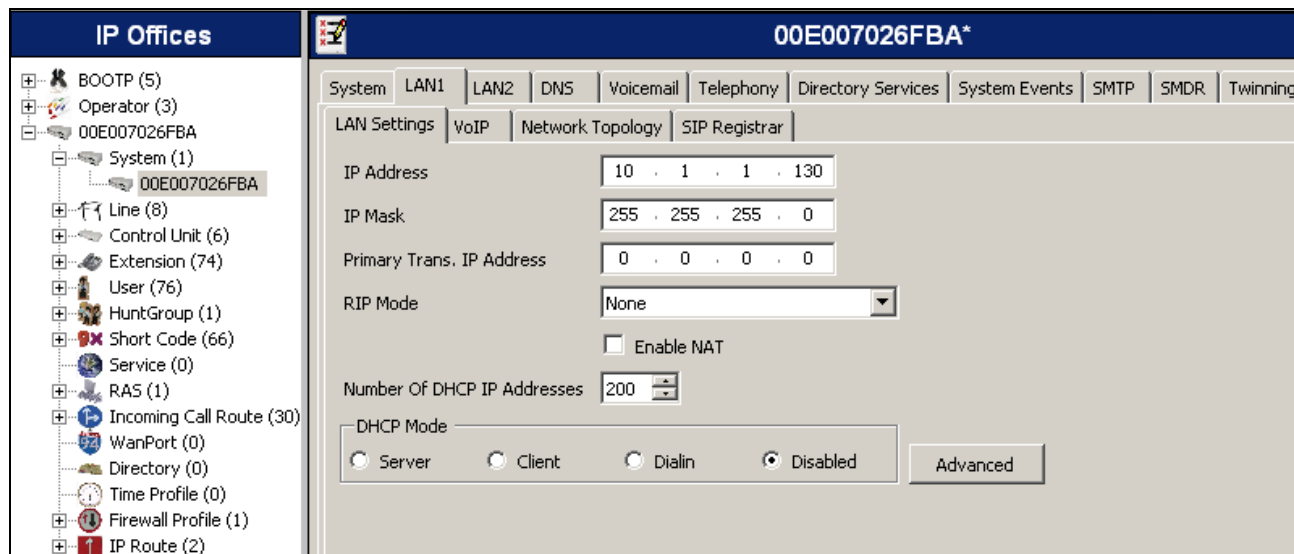
PAETEC Components	
Equipment	Release
Acme Packet Net-Net Session Director 4250 Session Border Controller	SC6.1.0
BroadSoft Platform	14sp9

4. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the Dynamic IP SIP Trunk 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 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.



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.

00E007026FBA

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

LAN Settings VoIP Network Topology SIP Registrar

☒ H323 Gatekeeper Enable
☒ SIP Trunks Enable
☒ SIP Registrar Enable

☒ H323 Auto-create Extn
☐ H323 Auto-create User

RTP Port Number Range
 Port Range (Minimum) 49152
 Port Range (Maximum) 53246

☒ Enable RTCP Monitoring On Port 5005

DiffServ Settings
 B8 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

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, showing the 'Network Topology Discovery' section. The configuration parameters are as follows:

Parameter	Value
STUN Server IP Address	0 . 0 . 0 . 0
STUN Port	3478
Firewall/NAT Type	Open Internet
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. Twinning Calling Party Settings

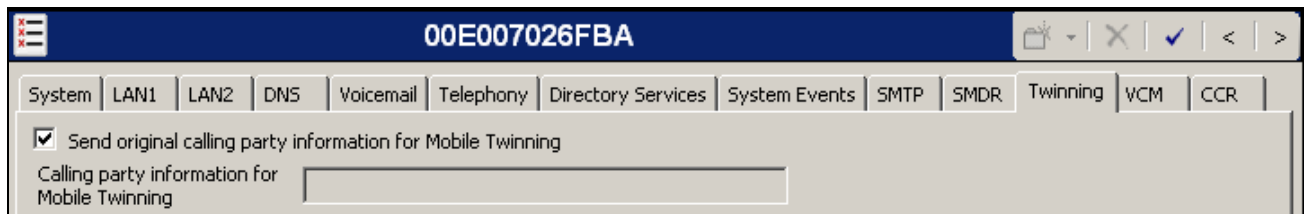
When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affects twinning and do not impact the messaging or operation of other redirected calls such as forwarded calls. The first parameter is the **Send original calling party 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 4.4**).

If this box (representing the first parameter) is checked, 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).

If this box is unchecked, the value sent in the SIP From header is determined by the setting of the second parameter (See **Section 4.4**).

PAETEC supports the SIP Diversion header for re-directed calls which is one of the choices available for setting the **Send Caller ID** parameter in **Section 4.4**. Thus, there are multiple ways the calling party can be configured for twinning, depending on the call display desired on the twinned phone. For the compliance test, the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab was checked which overrides any setting of the **Send Caller ID** parameter on the **SIP Line** form.



The screenshot shows the Avaya IP Office configuration window with the title bar '00E007026FBA'. The 'Twinning' tab is selected among several other tabs like System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VCM, and CCR. In the Twinning tab, the checkbox 'Send original calling party information for Mobile Twinning' is checked. Below this checkbox is a text field labeled 'Calling party information for Mobile Twinning' which is currently empty.

4.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Dynamic IP SIP Trunk Service. 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 the Dynamic IP SIP Trunk Service.

- Set the **ITSP Domain Name** to the IP address of the PAETEC SIP proxy.
- Set **ITSP IP Address** to the IP address of the PAETEC SIP proxy.
- Set **Send Caller ID** to *None*. For the compliance test, this parameter was ignored since the **Send original calling party information for Mobile Twinning** box is checked in **Section 4.3**. Since PAETEC supports the SIP Diversion header, alternatively, this parameter can be set to *Diversion Header* and the box in **Section 4.3** would then be unchecked. This alternate configuration allows Avaya IP Office to send the calling party number of the originating caller in the SIP From header for calls originating from the PSTN. It will also send the SIP Diversion header with the number of the host phone of the twinned destination. The number in the Diversion Header is known to PAETEC (even though the number in the From header is not) so the call is allowed to complete.
- 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 shows the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree view containing items like BOOTP (5), Operator (3), 00E007026FBA, System (1), Line (8), Control Unit (6), Extension (74), User (76), HuntGroup (1), Short Code (66), Service (0), RAS (1), Incoming Call Route (30), WanPort (0), Directory (0), Time Profile (0), Firewall Profile (1), IP Route (2), Account Code (0), License (70), Tunnel (0), and Logical LAN (0). The 'Line (8)' item is selected. The main pane is titled 'SIP Line - Line 22' and has tabs for 'SIP Line', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP Line' tab is active. It contains the following fields and controls:

- Line Number: 22
- ITSP Domain Name: 10.2.2.37
- ITSP IP Address: 10 . 2 . 2 . 37
- Prefix: (empty)
- National Prefix: 0
- Country Code: (empty)
- International Prefix: 00
- Send Caller ID: None
- Registration Required: ☒
- In Service: ☒
- Use Tel URI: ☐
- Check OOS: ☒
- Call Routing Method: Request URI
- Network Configuration section:
 - Layer 4 Protocol: UDP
 - Send Port: 5060
 - Use Network Topology Info: LAN 1
 - Listen Port: 5060

Since Avaya IP Office will register with the Dynamic IP SIP Trunk Service, 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 PAETEC. The **User name** and **Authentication Name** is a DID number provided by PAETEC and is not used for any purpose other 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 22' 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' (4695551234), 'Authentication Name' (4695551234), 'Password' (ipoffice), and 'Expiry' (10). At the bottom right are 'OK' and 'Cancel' buttons.

Index	UserName	Authentication Name	Password	Expiry
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New SIP Credentials

User name: 4695551234

Authentication Name: 4695551234

Password: ipoffice

Expiry: 10

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: 4695551234** 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 **22** was defined that only contains this line (line 22).
- 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 22' 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: 4695551234
- Incoming Group: 22
- Outgoing Group: 22
- Max Calls per Channel: 5

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** then **G.711 ALAW 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.
- Uncheck the **Fax Transport Support** box.
- 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 22' configuration window with the 'VoIP' tab selected. The window has a title bar with standard icons and a toolbar with icons for help, cancel, OK, and navigation. The main area contains several sections:

- Compression Mode:** A button labeled 'Advanced' is next to a list of codecs. The list has four items: 'G.729(a) 8K CS-ACELP' (checked), 'G.711 ULAW 64K' (checked), 'G.711 ALAW 64K' (checked), and 'G.723.1 6K3 MP-MLQ' (unchecked).
- Call Initiation Timeout (s):** A numeric field with the value '4' and a spinner control.
- DTMF Support:** A dropdown menu showing 'RFC2833'.
- Checkboxes:** On the right side, there are four checkboxes: 'VoIP Silence Suppression' (unchecked), 'Fax Transport Support' (unchecked), 'Re-invite Supported' (checked), and 'Use Offerer's Preferred Codec' (unchecked).

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"@10.2.2.37"**. 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 Management System interface. On the left is the 'IP Offices' navigation pane, which includes a tree view with categories like BOOTP (5), Operator (3), and Short Code (66). The 'Short Code (66)' item is selected. On the right is the '9N;; Dial' configuration pane. It contains a 'Short Code' tab and several input fields: 'Code' (9N;;), 'Feature' (Dial), 'Telephone Number' (N"@10.2.2.37"), 'Line Group Id' (22), 'Locale' (United States (US English)), and 'Force Account Code' (unchecked).

9N;; Dial	
Short Code	
Code	9N;;
Feature	Dial
Telephone Number	N"@10.2.2.37"
Line Group Id	22
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>

4.6. User

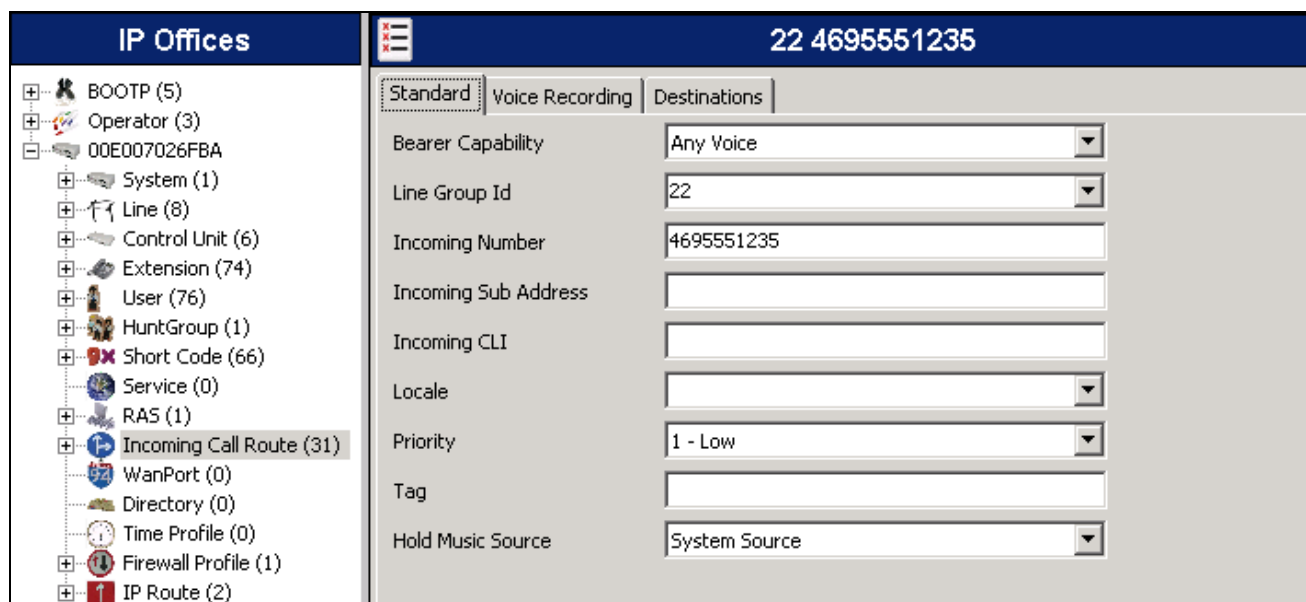
Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 4.4**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **SIP** tab in the Details Pane. The values entered for the **SIP Name** and **Contact** 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 PAETEC. In the example below, the DID number **4695551235** is used. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Click the **OK** button (not shown).

The screenshot shows the Avaya SIP configuration interface. On the left is the 'IP Offices' navigation pane with a tree structure: BOOTP (5), Operator (3), 00E007026FBA, System (1), Line (8), Control Unit (6), Extension (74), and User (76). Under 'User (76)', there are sub-items: NoUser, RemoteManager, 201 Extn201, and 202 Extn202. The main pane is titled 'Extn370: 370' and contains several tabs: User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Menu Programming, Mobility, Phone Manager Options, Hunt Group Membership, Announcements, SIP, and Pers. The 'SIP' tab is selected. It contains three text input fields: 'SIP Name' with the value '4695551235', 'SIP Display Name (Alias)' with the value 'Extn370', and 'Contact' with the value '4695551235'. Below these fields is an unchecked checkbox labeled 'Anonymous'.

4.7. Incoming Call Route

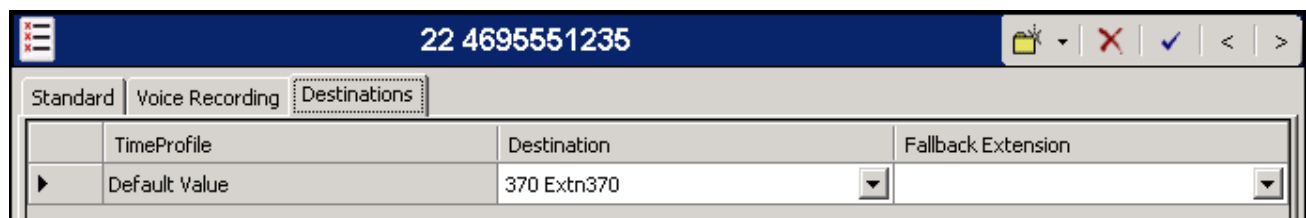
An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider except the DID number used for trunk registration. To create an incoming call route, right-click **Incoming Call Routes** in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below.

- Set the **Bearer Capacity** 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.



IP Offices		22 4695551235	
+ BOOTP (5)		Standard Voice Recording Destinations	
+ Operator (3)		Bearer Capacity: Any Voice	
- 00E007026FBA		Line Group Id: 22	
+ System (1)		Incoming Number: 4695551235	
+ Line (8)		Incoming Sub Address:	
+ Control Unit (6)		Incoming CLI:	
+ Extension (74)		Locale:	
+ User (76)		Priority: 1 - Low	
+ HuntGroup (1)		Tag:	
+ Short Code (66)		Hold Music Source: System Source	
+ Service (0)			
+ RAS (1)			
+ Incoming Call Route (31)			
+ WanPort (0)			
+ Directory (0)			
+ Time Profile (0)			
+ Firewall Profile (1)			
+ IP Route (2)			

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 4695551235 on line 22 are routed to extension 370.

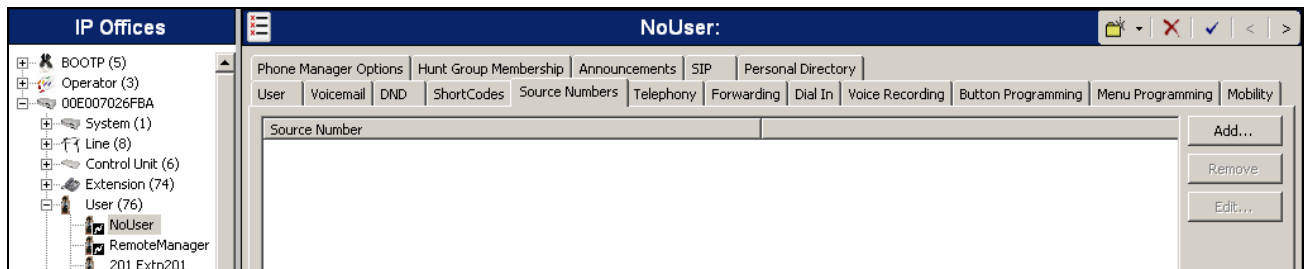


22 4695551235		
Standard Voice Recording Destinations		
TimeProfile	Destination	Fallback Extension
▶ Default Value	370 Extn370	

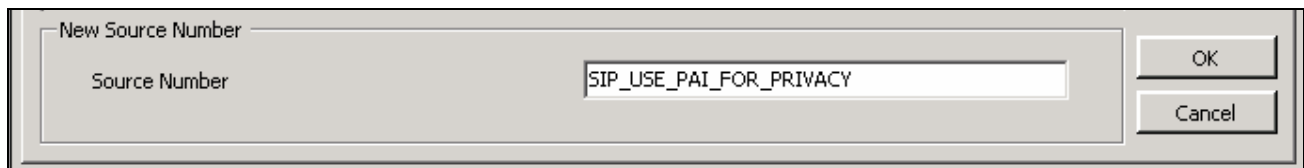
4.8. Privacy/Anonymous Calls

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 for authentication and billing. The Dynamic IP SIP Trunk Service supports PAI for the purposes of privacy.

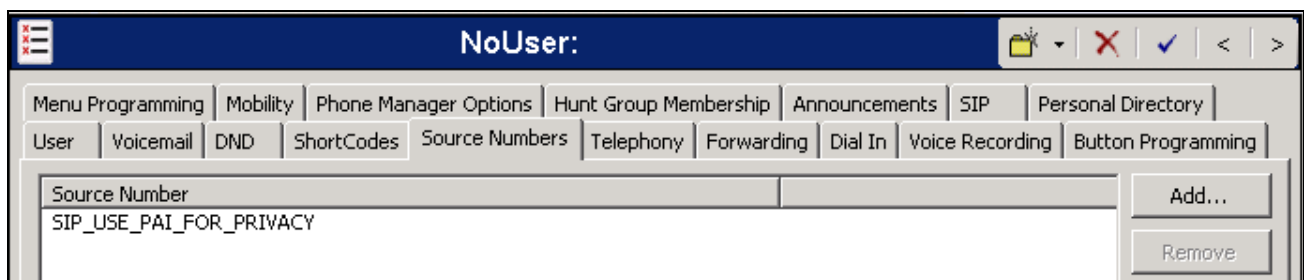
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → noUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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.

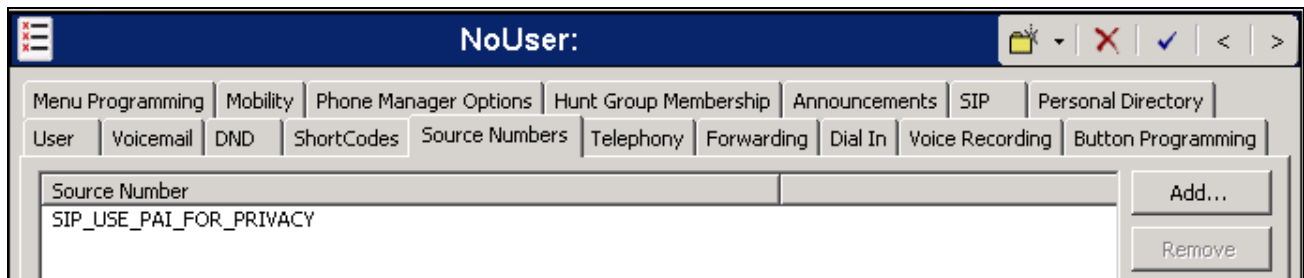


4.9. SIP Options

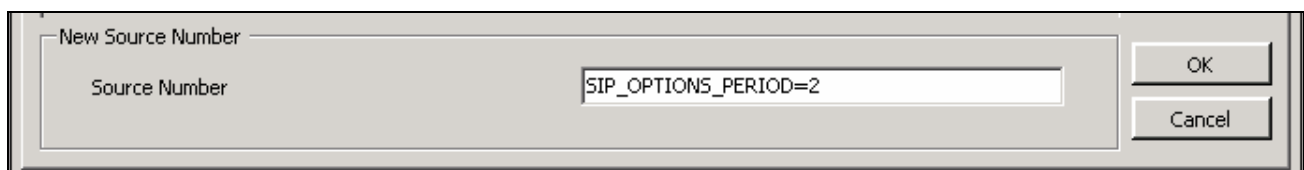
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 4.1** and the **SIP_OPTIONS_PERIOD**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

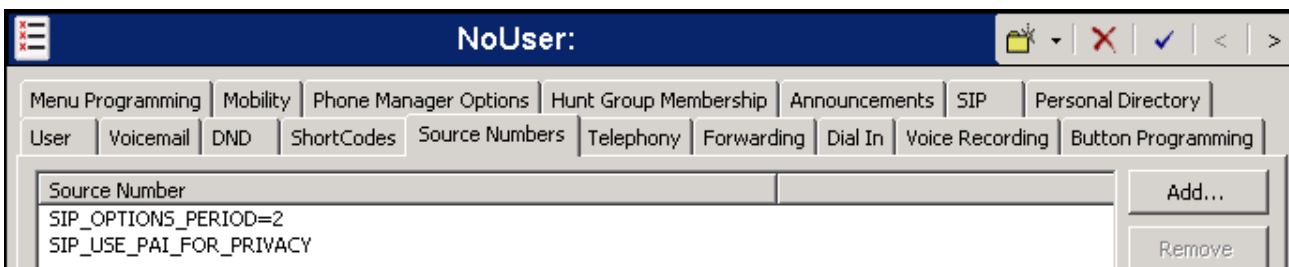
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → noUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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. Dynamic IP SIP Trunk Service Configuration

PAETEC is responsible for the configuration of the Dynamic IP SIP Trunk Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. PAETEC will provide the customer the necessary information to configure the SIP connection to the Dynamic IP SIP Trunk Service including:

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

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 the Dynamic IP SIP Trunk Service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 1.1**.

The Dynamic IP SIP Trunk Service passed compliance testing.

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 pane lists various system components, with 'Line: 22' selected under 'Trunks (9)'. The right pane displays the 'SIP Trunk Summary' for Line 22. The summary includes fields for Peer Domain Name, Gateway Address, Line Number, Number of Administered Channels, Number of Channels in Use, Administered Compression, Silence Suppression, SIP Trunk Channel Licences, and SIP Trunk Channel Licences in Use. A green progress indicator shows 0% utilization. Below the summary is a table of channels.

Channel Number	UR	Call Grp	Ref	Current State	Time in State	Remote Address	Code	Connec Type	Caller Dialed	I on Call	Other Party	Directio of Call	Round Delay	Receiv Jitter	Receiv Loss	Transm Fr	Tr Lo
1				Idle	13 days...												
2				Idle	13 days...												
3				Idle	13 days...												
4				Idle	13 days...												
5				Idle	13 days...												

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

The screenshot shows the 'Alarms' tab selected in the application. The title is 'Alarms for Line: 22 SIP 10.2.2.37'. Below the title is a table with columns for Last Date Of Error, Occurrences, and Error Description.

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

The Dynamic IP SIP Trunk Service passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and the PAETEC Dynamic IP SIP Trunk Service 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 the Dynamic IP SIP Trunk Service is available from PAETEC.

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