



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

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# **Application Notes for Phonect SIP Trunk Service and Avaya IP Office 7.0 – Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the Phonect SIP Trunk Service and Avaya IP Office.

The Phonect SIP Trunk Service provides PSTN access via a SIP trunk connected to the Phonect Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or digital trunks. This approach generally results in lower cost for the enterprise. Phonect 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 a Session Initiation Protocol (SIP) trunk between the Phonect SIP Trunk Service and Avaya IP Office. Phonect SIP Trunk service provides PSTN access via a SIP trunk connected to the Phonect network as an alternative to legacy Analogue 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 configure a simulated enterprise site using Avaya IP Office to connect to the Phonect SIP Trunk service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

### 2.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office was connected to the Phonect SIP Trunk service. To verify SIP trunk interoperability the following features and functionality were exercised during the interoperability compliance test.

- Incoming PSTN calls to various phone types. Phone types included H.323, digital, and Analogue 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 Analogue telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- T38 Inbound and Outbound Fax.
- Inbound and outbound PSTN calls to/from Phone Manager Lite clients.
- Various call types including: local, long distance, international, toll free(outbound) and directory assistance
- Codecs G.729A 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

## **2.2. Test Results**

Interoperability testing of the sample configuration was completed with successful results for the Phonect SIP Trunk Service with the following observations:

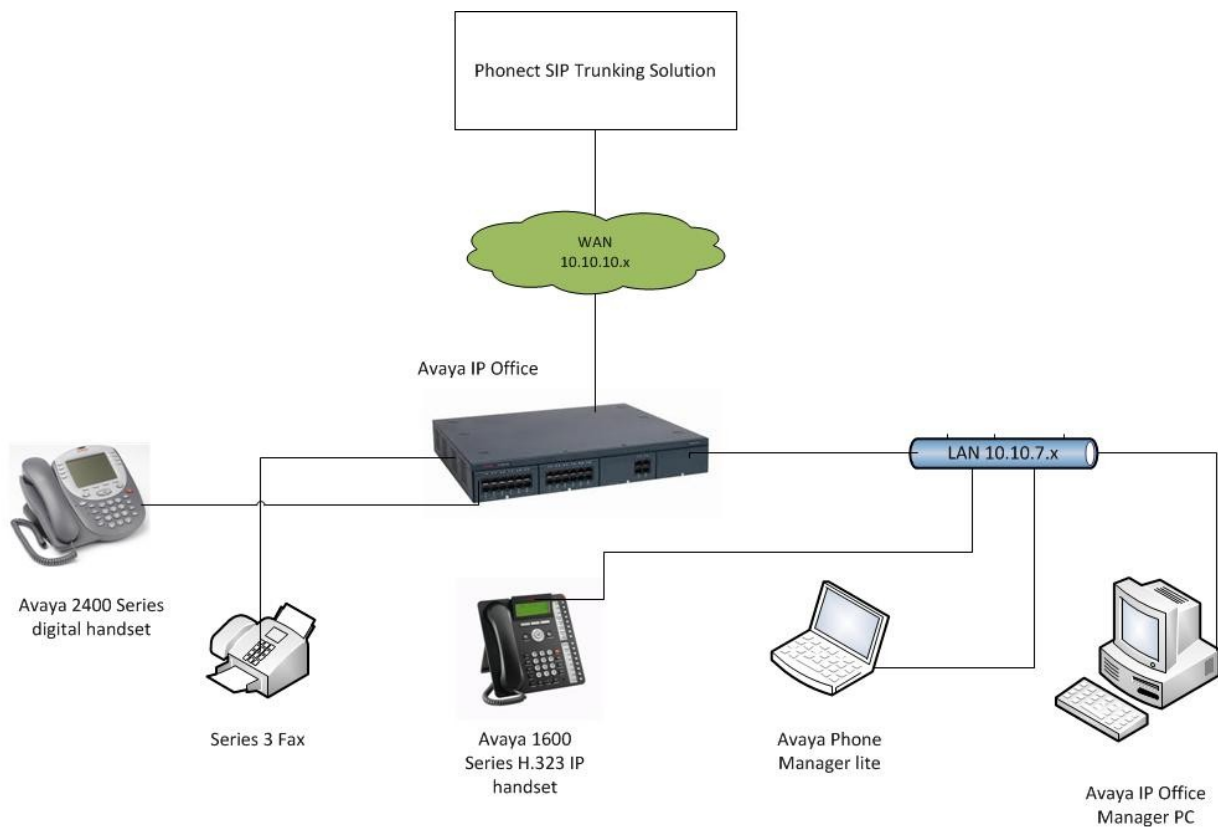
- No inbound toll free numbers were tested, however routing of inbound DID numbers, and the relevant number translation, was successfully tested.

## **2.3. Support**

For technical support on Phonect products, please contact the Phonect authorized representative at [support@phonect.no](mailto:support@phonect.no).

### 3. Reference Configuration

**Figure 1** illustrates the test configuration. The test configuration shows an enterprise site connected to the Phonect SIP Trunking service. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 2420 Digital Telephone, an Avaya Analogue Telephone, fax machine and a Phone Manager lite client PC. The site also has a Windows PC running Avaya IP Office Manager to configure the Avaya IP Office. 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 arbitrary numbers that bear no relevance to the test configuration.



**Figure 1: Phonect SIP Trunk Solution to Avaya IP Office Topology**

IP Office was configured to connect to a static IP address at the service provider allowing the SIP line on IP Office to Register with Phonect. For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Phonect. The short code of 9 is stripped off by Avaya IP Office but the remaining N digits were sent with the SIP domain provided by Phonect added.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and 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 IP Office must be allowed to pass through these devices. Phonect sends SIP signaling from one IP address, however, RTP traffic may originate from a different IP address and ports which may vary from customer to customer. Customers will need to work with Phonect to determine the proper IP addresses and ports required to access the Phonect network.

## 4. Equipment and Software Validated

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

Equipment	Software
Avaya IP Office 500 V2	7.0(23)
Avaya 1620 Phone (H.323)	1.030
Avaya 2420 Digital Phone	NA
Avaya 98390 Analogue Phone	NA
Phonect SIP Trunk	435 - Setup Avaya Porta MR22 - version 1.0

## 5. Configure Avaya IP Office

This section describes the IP Office configuration necessary to support connectivity to the Phonect SIP Trunk service. IP Office is configured through the IP Office Manager PC application. From a PC running the 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 (Not Shown). A management window will appear similar to the one in the next section. All the 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 IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning) is assumed to already be in place.

### 5.1. Verify System Capacity

Navigate to **License → SIP Trunk Channels** in the Navigation Pane. In the **Licences** tab verify that the **License Status** is **Valid** and that the number of **Instances** is sufficient to support the number of SIP trunk channels provisioned by Phonect.

IP Offices	SIP Trunk Channels
CCR CCC UPG	Licences
CCR Designer	Licence Key: unXMB6x9dJKGKJ73uEpof7JrpF4smme
CCR SUP	Licence Type: SIP Trunk Channels
Compact Business Centre	Licence Status: Valid
Conferencing Center	Instances: 255
CTI Link Pro	Expiry Date: Never
Customer Service Agent	
Customer Service Supervisor	
DECT Integration (ports)	
eBLF	
Enterprise Branch User	
Essential Edition Additional V	
Essential to Branch Edition M	

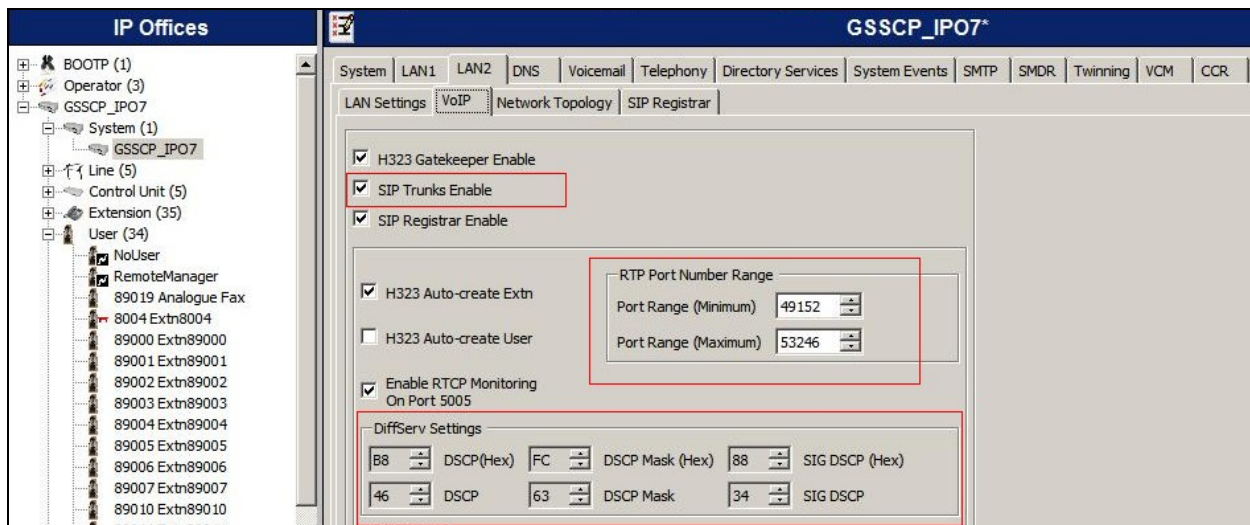
## 5.2. LAN2 Settings

In the sample configuration, the LAN2 port was used to connect the IP Office to the external intranet. To access the LAN2 settings, first navigate to **System → GSSCP\_IPO7** in the Navigation Pane where GSSCP\_IPO7 is the name of the IP Office system. Navigate to the **LAN2 → LAN Settings** tab in the details plane. 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 IP Office configuration interface. On the left is the 'IP Offices' navigation pane, showing a tree structure with 'GSSCP\_IPO7' selected. The main area is titled 'GSSCP\_IPO7\*' and contains several tabs: 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', 'SMTP', 'SMDR', 'Twinning', 'VCM', and 'CCR'. The 'LAN2' tab is active, and within it, the 'LAN Settings' sub-tab is selected. The 'IP Address' and 'IP Mask' fields are highlighted with a red box. The 'IP Address' is set to '10 . 10 . 10 . 10' and the 'IP Mask' is set to '255 . 255 . 255 . 128'. Other visible settings include 'Primary Trans. IP Address' (10 . 10 . 10 . 1), 'Firewall Profile' (<None>), 'RIP Mode' (None), 'Enable NAT' (unchecked), 'Number Of DHCP IP Addresses' (200), and 'DHCP Mode' (Disabled). An 'Advanced' button is located at the bottom right of the settings area.

Field	Value
IP Address	10 . 10 . 10 . 10
IP Mask	255 . 255 . 255 . 128
Primary Trans. IP Address	10 . 10 . 10 . 1
Firewall Profile	<None>
RIP Mode	None
Enable NAT	<input type="checkbox"/>
Number Of DHCP IP Addresses	200
DHCP Mode	Disabled

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, IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2. 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.





### 5.3. 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 Europe, **ALAW** 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.

The screenshot shows the Avaya IP Office configuration interface for GSSCP\_IPO7. The left pane shows the system hierarchy. The right pane is the 'Telephony' tab. The 'Automatic Codec Preference' is set to 'G.711 ALAW 64K'. The 'Companding Law' section shows 'ALAW' selected for the Switch and 'ALAW Line' selected for the Line. The 'Inhibit Off-Switch Forward/Transfer' checkbox is unchecked.

### 5.4. System Twinning Settings

Navigate to the **Twining** 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 the PSTN and are extended to the twinned mobile. For inbound PSTN calls to a twinned enabled phone, IP Office will continue to send the associated host phone number in the SIP From header (used for the caller display). 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 5.5).

The screenshot shows the Avaya IP Office configuration interface for GSSCP\_IPO7, Twining tab. The 'Send original calling party information for Mobile Twinning' checkbox is checked. Below it, there is a text field for 'Calling party information for Mobile Twinning'.

## 5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between IP Office and the Phonect SIP Trunking service. 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 below to connect to the SIP Trunking service.

- Set the **ITSP Domain Name** to the domain name provided by Phonect.
- 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 5.4**. This parameter was set to **None** and the box in **Section 5.4** was checked.
- Ensure the **In Service** box is checked.
- Default values may be used for all other parameters.

The screenshot shows the 'SIP Line - Line 17' configuration window. The left pane shows the 'IP Offices' tree with 'Line (6)' expanded, highlighting line 17. The main pane has tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP Line' tab is active, showing fields for Line Number (17), ITSP Domain Name (sip.phonect.no), Prefix, National Prefix (0), Country Code (47), International Prefix (00), Send Caller ID (None), Association Method (By Source IP address), In Service (checked), Use Tel URI (unchecked), Check OOS (unchecked), Call Routing Method (Request URI), and Originator number for forwarded and twinning calls (4721082440). There is also a 'REFER Support' section with 'Incoming' and 'Outgoing' set to 'Auto'.

Field	Value
Line Number	17
ITSP Domain Name	sip.phonect.no
Prefix	
National Prefix	0
Country Code	47
International Prefix	00
Send Caller ID	None
Association Method	By Source IP address
In Service	<input checked="" type="checkbox"/>
Use Tel URI	<input type="checkbox"/>
Check OOS	<input type="checkbox"/>
Call Routing Method	Request URI
Originator number for forwarded and twinning calls	4721082440
REFER Support Incoming	Auto
REFER Support Outgoing	Auto

Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the IP address of the Phonect SIP proxy.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Send Port** and **Listen Port** to **5060**.
- Set the **Use Network Topology Info** to the appropriate interface, in this case **LAN 2**.

The screenshot shows the 'SIP Line - Line 17\*' configuration window with the 'Transport' tab selected. A red box highlights the 'ITSP Proxy Address' field (10.10.10.20) and the 'Network Configuration' section. In the 'Network Configuration' section, 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', 'Use Network Topology Info' is 'LAN 2', and 'Listen Port' is '5060'. Below this, 'Explicit DNS Server(s)' are set to '0.0.0.0'. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

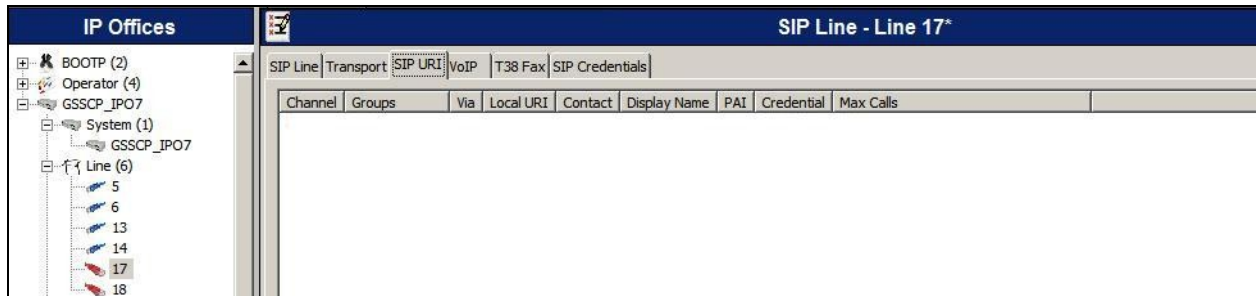
The configuration requires that the SIP Trunk registers with the Phonect network. The registration credential must be entered under the **SIP Credentials** tab. Click on the **Add** button to administer the registration details

The screenshot shows the 'SIP Credentials' tab in the 'SIP Line - Line 17\*' configuration window. It features a table with the following columns: Index, UserName, Authentication Name, Contact, Expiry, and Register. The table is currently empty. To the right of the table are three buttons: 'Add...', 'Remove', and 'Edit...'.

Add the **User name**, **Authentication Name**, **Contact** and **Password** provided by Phonect. Click on the **OK** button to save.

The screenshot shows the 'Edit SIP Credentials' dialog box. It contains the following fields: 'User name' (joooooooooooo), 'Authentication Name' (joooooooooooo), 'Contact' (joooooooooooo), and 'Password' (masked with asterisks). Below these is an 'Expiry' field set to '60' and a 'Registration required' checkbox which is checked. 'OK' and 'Cancel' buttons are located on the right side of the dialog.

After the SIP line parameters are defined, each SIP URI that 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 (not shown) 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 IP Office user. The entry was created with the parameters shown below.

- Set **Local URI**, **Contact**, **Display Name** and to **Use Internal Data**. This setting allows calls on this line for any SIP URI that matches the number set in the **SIP** tab of any **User** as shown in **Section 5.7**.
- Set **PAI** to none.
- For **Registration**, select **1:** from the pull-down menu since this configuration requires SIP registration administered in the previous step.
- 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 line group **17** was defined that was associated to a single line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

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, and so on. For the compliance test, **G.711 ALAW 64K** was selected first followed by **G.729(a) 8K CS-ACELP**.
- 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.
- Select the **T38** option for **Fax Transport Support**.
- 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.

Select the **T38** tab and set the **T38 Fax Version** to **2**. All other settings are default.

**Note:** It is advisable at this stage to save the configuration as described in **Section 5.10** to make the Line Group ID available in **Section 5.5**.



## 5.6. Short Codes

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click **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**. 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 5.5**. This short code will use this line group when placing the outbound call

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

The screenshot shows a software interface for configuring a short code. On the left, a list of IP Offices is visible, including \*44, \*45\*N#, \*46, \*47, \*48, \*49, \*50, \*51, \*52, \*53\*N#, \*57\*N#, and \*66. The main configuration area is titled 'Short Code' and contains the following fields:

- Code:** 9N;;
- Feature:** Dial (selected from a dropdown menu)
- Telephone Number:** N
- Line Group Id:** 17 (selected from a dropdown menu)
- Locale:** (empty dropdown menu)
- Force Account Code:** ☐

## 5.7. User

Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 5.5**. 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 allows matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.5**). As such, these fields should be set to one of the DID numbers assigned to the enterprise from Phonect.

In the example below, DID number **4712345678** 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 User configuration interface. On the left, a list of IP Offices includes extensions 89016 through 89023, a Mailbox, and two Occ (Clean and Dirty) entries. The main area is titled 'Extn89101: 89101\*' and contains several tabs: Telephony, Forwarding, Dial In, Voice Recording, Button Programming, Menu Programming, Mobility, Phone Manager Options, Hunt Group Membership, Announcements, and SIP. The SIP tab is active, showing three text input fields: 'SIP Name' with the value '4712345678', 'SIP Display Name (Alias)' with the value 'Extn89101', and 'Contact' with the value '4712345678'. Below these fields is an unchecked checkbox labeled 'Anonymous'.

## 5.8. Incoming Call Routing

An incoming call route maps an inbound DID number on a specific line to an internal extension. 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 5.5**.
- Set the **Incoming Number** to the incoming DID that this route should match on.  
Matching is right to left
- Default values can be used for all other fields

The screenshot shows the 'Standard' tab of the 'Incoming Call Routes' configuration window. The 'Line Group Id' is set to 17 and the 'Incoming Number' is 4712345678. A red box highlights these two fields. The 'Bearer Capability' is set to 'Any Voice'. The 'Incoming Sub Address', 'Incoming CLI', 'Locale', 'Priority', 'Tag', and 'Hold Music Source' fields are empty or set to default values.

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 4712345678 on line 17 are routed to extension 89101.

The screenshot shows the 'Destinations' tab of the 'Incoming Call Routes' configuration window. The 'Destination' field is set to 89101 Extn89101. The 'Fallback Extension' field is empty. The 'TimeProfile' field is set to 'Default Value'.

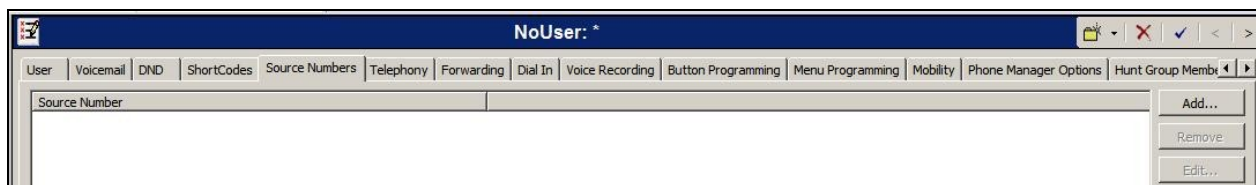


## 5.9. SIP Options

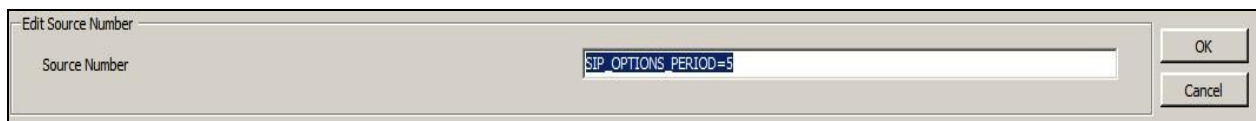
IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2** (not shown) and the **SIP\_OPTIONS\_PERIOD** 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 44 seconds, do not define a **SIP\_OPTIONS\_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 44 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**
- To establish a period greater than 44 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**

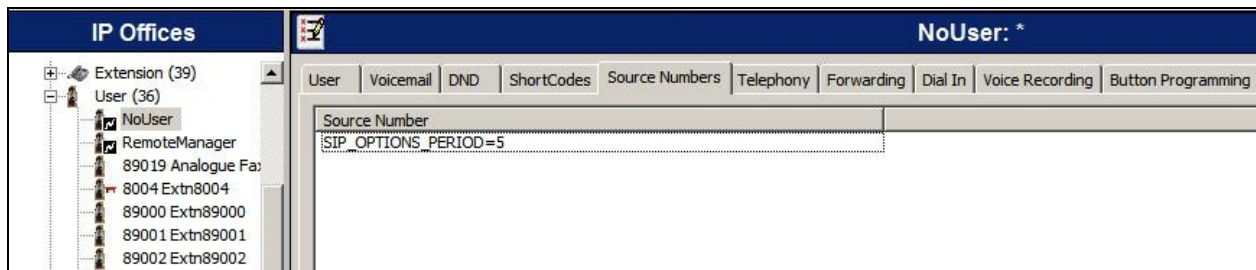
To configure the **SIP\_OPTIONS\_PERIOD** parameter, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the subsequent 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 5 minutes was desired. The **SIP\_OPTIONS\_PERIOD** was set to **5** minutes. Click the **OK** button (not shown).



## 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. Phonect SIP Trunk Service Configuration

Phonect is responsible for the configuration of the SIP Trunk service. The customer will need to provide the public IP address or FQDN used to reach the IP Office at the enterprise. Phonect will provide the customer with the necessary information to configure the SIP connection to the SIP Trunking service 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

## 7. Verification Steps

This section includes steps that can be used to verify that the configuration has been done correctly.

### 7.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. This is found under **Start → All Programs → IP Office → System Status**. From the left hand menu expand **Trunks** and choose the SIP line (17 in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.

The screenshot shows the 'Status' window for a SIP trunk. The left-hand menu is expanded to 'Trunks (2)', showing 'Line: 13 (1)' and 'Line: 17 (0)'. The main area displays the status for Line 17. The 'Status' tab is selected, showing a green circle indicating 0% utilization. Below this is a table with 8 columns: Channel Number, URI, Call Group, Ref, Current State, Time in State, Remote RTP Address, Codec, and Connection Type. The table lists 10 channels, all in an 'Idle' state.

Channel Number	URI	Call Group	Ref	Current State	Time in State	Remote RTP Address	Codec	Connection Type
1				Idle	2 days 18:...			
2				Idle	3 days 00:...			
3				Idle	3 days 00:...			
4				Idle	3 days 00:...			
5				Idle	3 days 00:...			
6				Idle	3 days 00:...			
7				Idle	3 days 00:...			
8				Idle	3 days 00:...			
9				Idle	3 days 00:...			
10				Idle	3 days 00:...			

## 8. Conclusion

These Application Notes describe the configuration necessary to connect IP Office 7.0 to Phonect SIP Trunk Service. Phonect SIP Trunk Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks.

## 9. Additional References

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

- [1] *IP Office 7 Documentation CD*, 4<sup>th</sup> May 2011.
- [2] *IP Office Installation*, Document number15-601042, 22<sup>nd</sup> May 2011.
- [3] *IP Office Manager*, Document number15-601011, 22<sup>nd</sup> May 2011.
- [4] *System Status Application*, Document number15-601758, 12<sup>th</sup> February 2010.

Product documentation for the Phonect SIP Trunking service is available from the local authorized Phonect representative.

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