



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Vodafone NL SIP Trunking Service with Avaya IP Office 9.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between Vodafone NL SIP Trunking Service and Avaya IP Office. Vodafone NL SIP Trunking Service provides PSTN access via a SIP trunk connected to the Vodafone NL Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or Digital trunks. Vodafone NL is a member of the Avaya DevConnect Service Provider program.

Vodafone 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 Vodafone NL SIP Trunking Service and Avaya IP Office. Vodafone NL SIP Trunking Service provides PSTN access via a SIP trunk connected to the Vodafone NL network as an alternative to legacy Analogue or Digital trunks. This approach generally results in lower cost for customers. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 9.0, Avaya Voicemail Embedded, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

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 Vodafone SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

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

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

- FAX modes T.38 and G.711 Pass Through.
- Off-net call forwarding.
- Twinning to mobile phones on inbound calls.

2.2. Test Results

Vodafone NL SIP Trunking passed compliance testing.

Items not supported or not tested included the following:

- Inbound toll-free is not supported and not tested in the compliance test.

2.3. Support

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

For technical support on Vodafone NL SIP Trunking, contact Vodafone Netherlands at http://www.vodafone.nl/zakelijk/totaal_oplossingen/vast_en_mobiel/

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Vodafone NL SIP Trunking Service. Located at the enterprise site is an Avaya IP Office 500 V2. Endpoints include three Avaya 1608, 9620 (with H.323 firmware) and 1140E IP Telephones, an Avaya 2420 Digital Telephone, an Avaya Analogue Telephone and a fax machine. The site also has a Windows 7 PC running IP Office Manager to configure the 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 obscured and all phone numbers have been obscured beyond the city code.

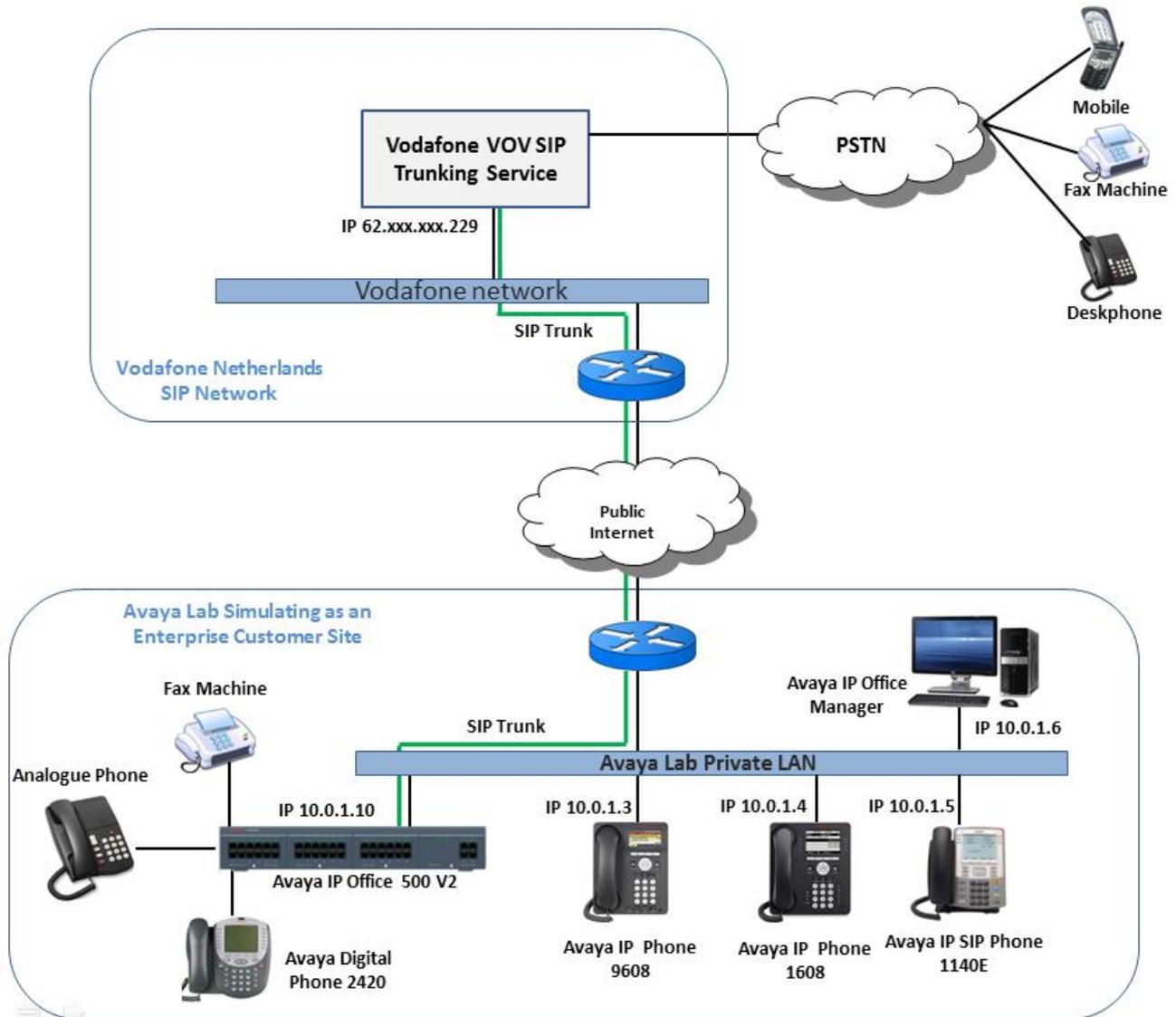


Figure 1: Test Configuration for Avaya IP Office with Vodafone NL SIP Trunking Service

Avaya IP Office was configured to connect to a Service Provider static IP address. For the purpose of the compliance test, users dialed a short code of N digits to send digits across the SIP trunk to Vodafone NL. The short code of 9 was not configured as required from Vodafone and the remaining N digits are sent in E.164 format.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the Avaya IP Officer such as a session border controller or data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the service provider and the IP Office must be allowed to pass through these devices.

4. Equipment and Software Validated

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

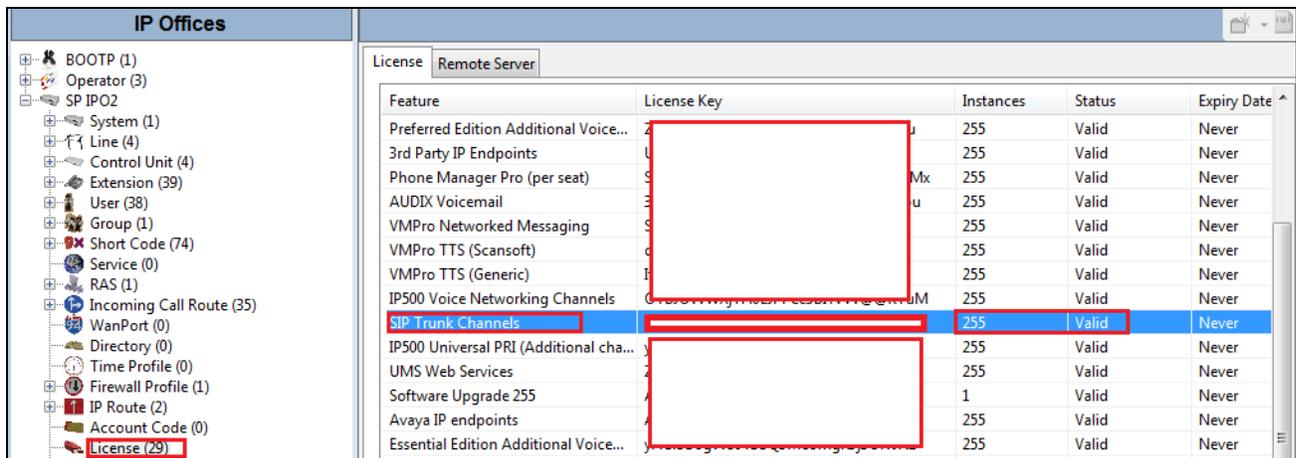
Avaya Telephony Components	
Equipment	Release
Avaya IP Office 500v2	9.0.2.0 Build 860
Avaya IP Office Manager	9.0.2.0 Build 860
Avaya 11x0 IP Telephone (SIP)	SIP11x0e04.4
Avaya 9620 IP Telephone (H.323)	Avaya one-X® Deskphone Edition 3.22A
Avaya 1608L Telephone (H.323)	1.343A
Avaya IP Office Softphone	3.2.3.20 64770
Avaya Digital Telephone 2420	N/A
Avaya Analog Telephone	N/A
Vodafone NL SIP Trunking Service Components	
Component	Release
Acme Packet Net-Net 4500 SBC	SCX6.2.0 MR-11 Patch 4
Cisco CPE/SIP Gateway	29xx

5. Configure Avaya IP Office

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

5.1. Verify License Capacity

Navigate to **License → SIP Trunk Channels** in the Navigation Pane. In the Details Pane, 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 Vodafone NL.

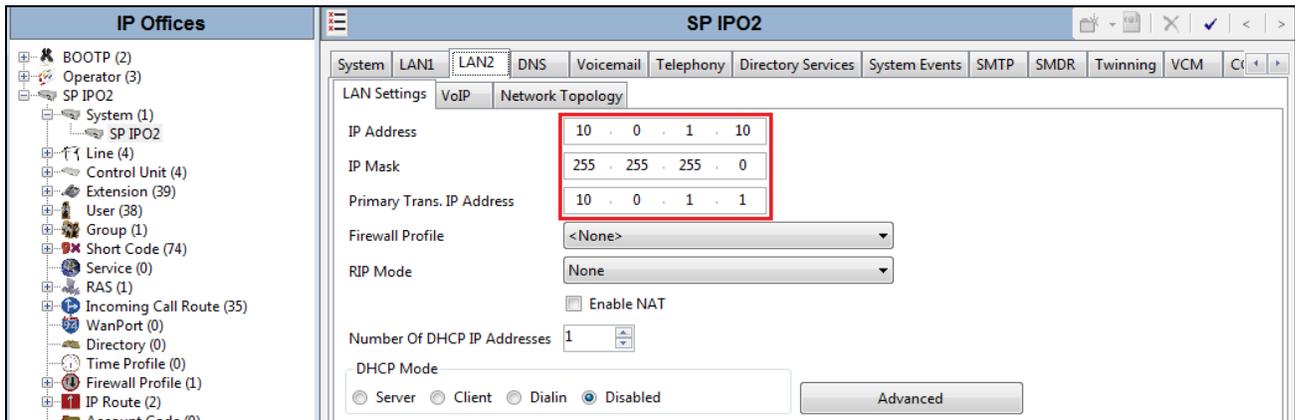


The screenshot displays the Avaya IP Office Manager interface. On the left is a navigation tree under 'IP Offices' with 'License (29)' selected. The main area shows a 'License' table with columns: Feature, License Key, Instances, Status, and Expiry Date. The 'SIP Trunk Channels' row is highlighted in blue, showing 255 instances and a 'Valid' status. Two red boxes highlight the 'License Key' column for the first and last rows of the table.

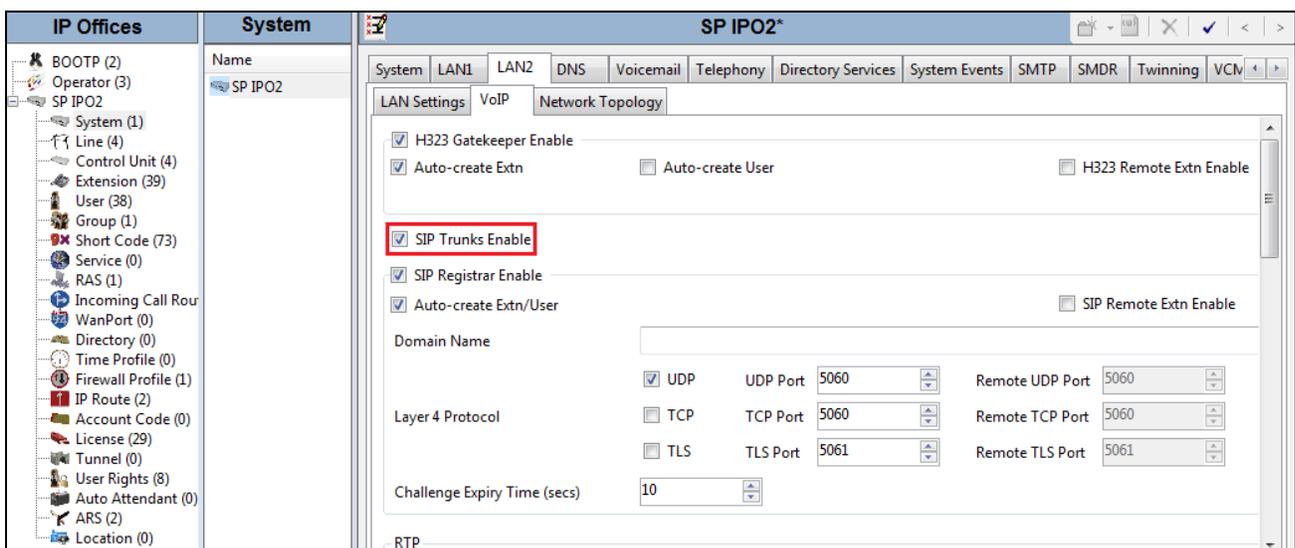
Feature	License Key	Instances	Status	Expiry Date
Preferred Edition Additional Voice...		255	Valid	Never
3rd Party IP Endpoints		255	Valid	Never
Phone Manager Pro (per seat)		255	Valid	Never
AUDIX Voicemail		255	Valid	Never
VMPPro Networked Messaging		255	Valid	Never
VMPPro TTS (Scansoft)		255	Valid	Never
VMPPro TTS (Generic)		255	Valid	Never
IP500 Voice Networking Channels		255	Valid	Never
SIP Trunk Channels		255	Valid	Never
IP500 Universal PRI (Additional cha...		255	Valid	Never
UMS Web Services		255	Valid	Never
Software Upgrade 255		1	Valid	Never
Avaya IP endpoints		255	Valid	Never
Essential Edition Additional Voice...		255	Valid	Never

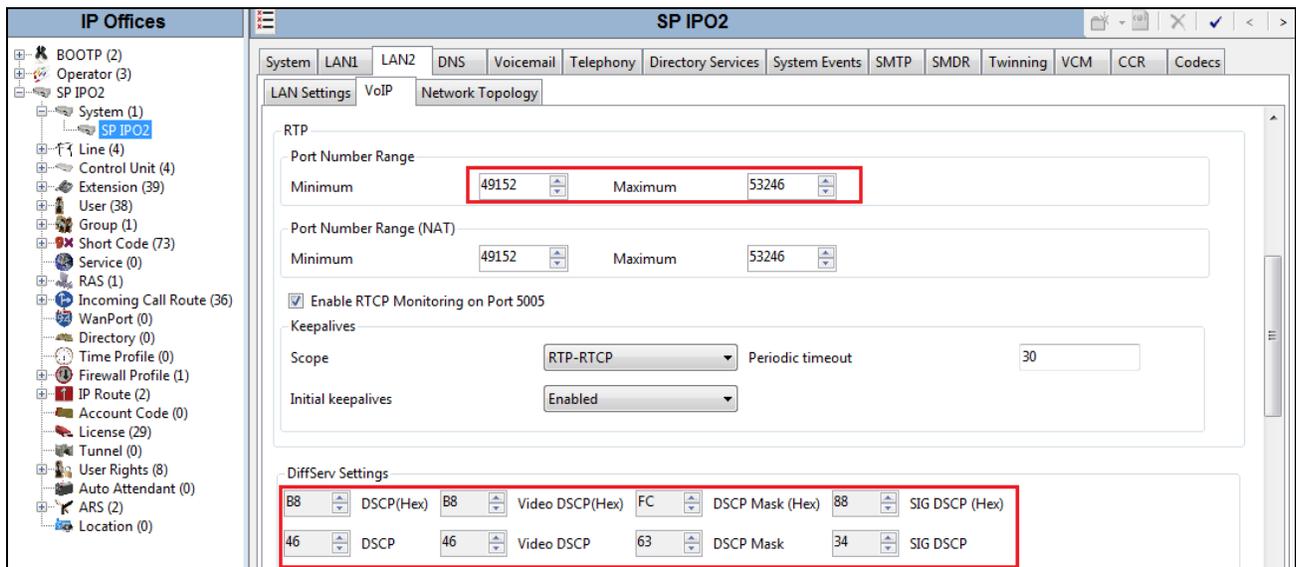
5.2. LAN2 Settings

In the sample configuration, the **LAN2** port was used to connect IP Office to the external internet. To access the **LAN2** settings, first navigate to **System** → **SP IPO2** in the left Navigation Pane where **SP IPO2** is the name of the IP Office. Navigate to the **LAN2** → **LAN Settings** tab in the Details Pane. The IP Address and IP Mask fields are the public interface of IP Office; Primary Trans. IP Address is the next hop, usually the default gateway address. All other parameters should be set according to customer requirements. On completion, click the OK button (not shown).

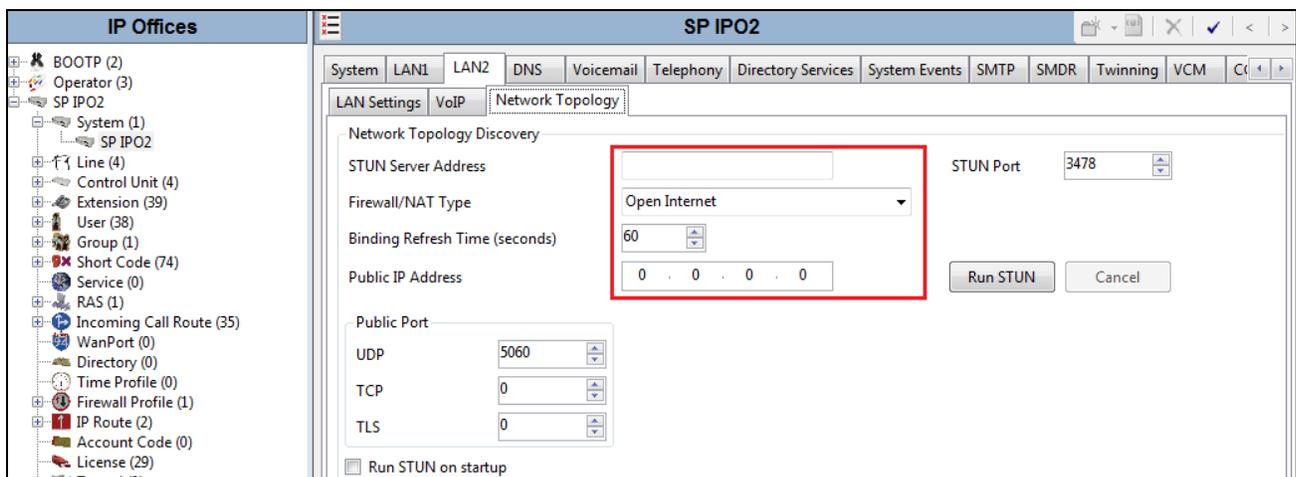


On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The RTP Port Number Range can be customized to a specific range of receive ports for the RTP media. Based on this setting, IP Office would request RTP media be sent to a UDP port in 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 signalling and media. The DSCP field is the value used for media and the SIG DSCP is the value used for signalling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements. On completion, click the OK button (not shown).



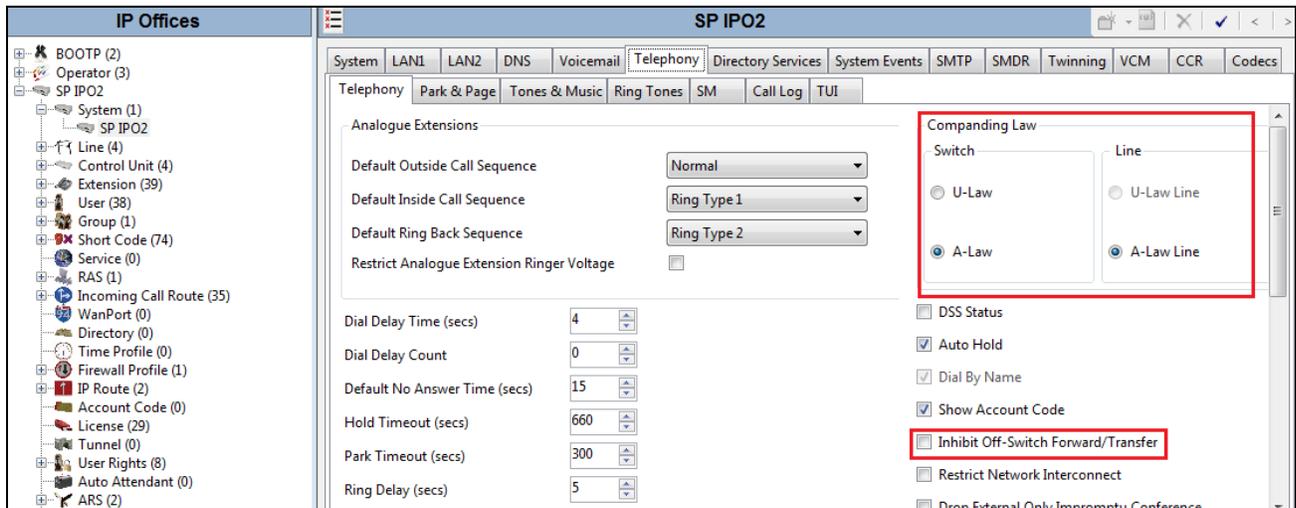


On the **Network Topology** tab in the Details Pane, leave the **STUN Server Address** field as blank and enter the Public IP Address as 0.0.0.0 for IP Office since the STUN server was not used for this test. It is important that the Binding Refresh Time is set to the correct value. IP Office sends SIP 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; see **Section 5.10** for more details. Below is a sample configuration. On completion, click the OK button (not shown).



5.3. System Telephony Settings

Navigate to the **Telephony** → **Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For Europe, A-Law 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. On completion, click the OK button (not shown).

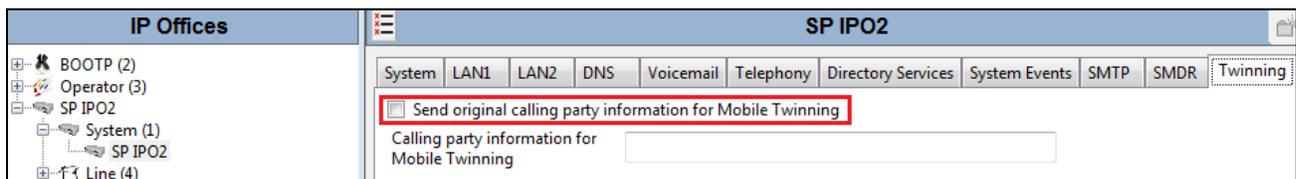


5.4. System Twinning Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affect 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 5.6.2).

For the compliance testing, the **Send original calling party information for Mobile Twinning** as shown below was unchecked. This setting allows **Send Caller ID** parameter that was set to **None** in Section 5.6.2 to be used. IP Office will send the following in the “From” header:

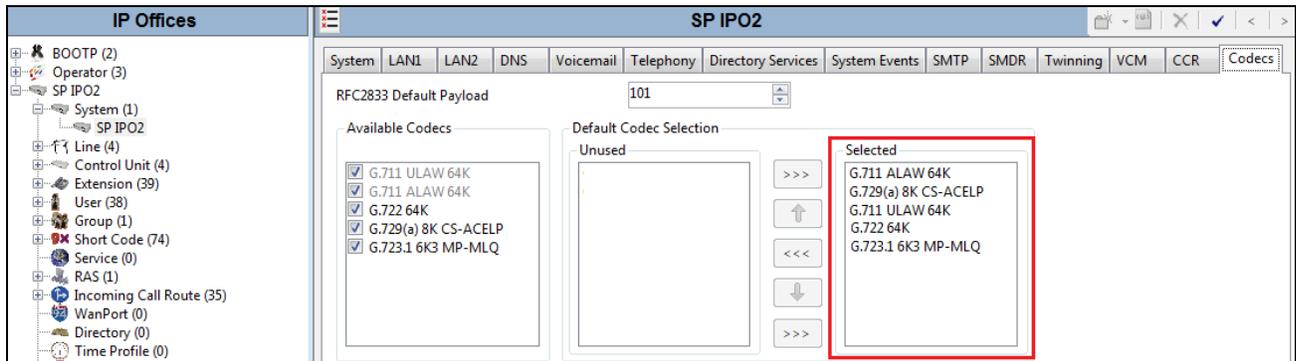
- On calls from an internal extension to a twinned phone, IP Office sends Calling Party Number of the originating extension.
- On calls from the PSTN to a twinned phone, IP Office sends Calling Party Number of the originating twinned phone.



5.5. System Codec Settings

Navigate to the Codecs tab. The Available Codecs box indicates all audio codecs available on the system. Highlight codecs in the Unused box that are to be used by the service and click on the right

arrows to move them to the Selected box. Highlight codecs in the Selected box that are not to be used and click on the left arrows to move them to the Unused box. Highlight codecs in the Selected box and use the up and down arrows to change the priority. On completion, click the OK button (not shown).



Note: During the testing, all available codecs were selected to test the codec negotiation between IP Office and the Vodafone NL network.

5.6. Administer SIP Lines

SIP lines are needed to establish the SIP connections between IP Office and the Vodafone NL SIP Trunking service. Two SIP lines are required, one is for the Vodafone Office Voice (VoV) service, and the other is for their Corporate Net over IP (CNoIP) service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.6.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses.
- SIP Credentials (if applicable).
- SIP URI entries.
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 5.6.2**.

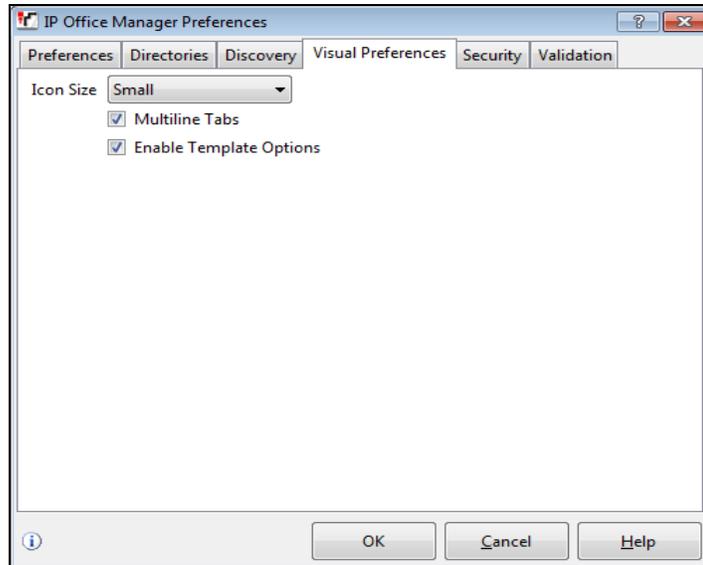
Also, the following SIP Line settings are not supported on Basic Edition:

- SIP Line – Originator number for forwarded and twinning calls.
- Transport – Second Explicit DNS Server.
- SIP Credentials – Registration Required.

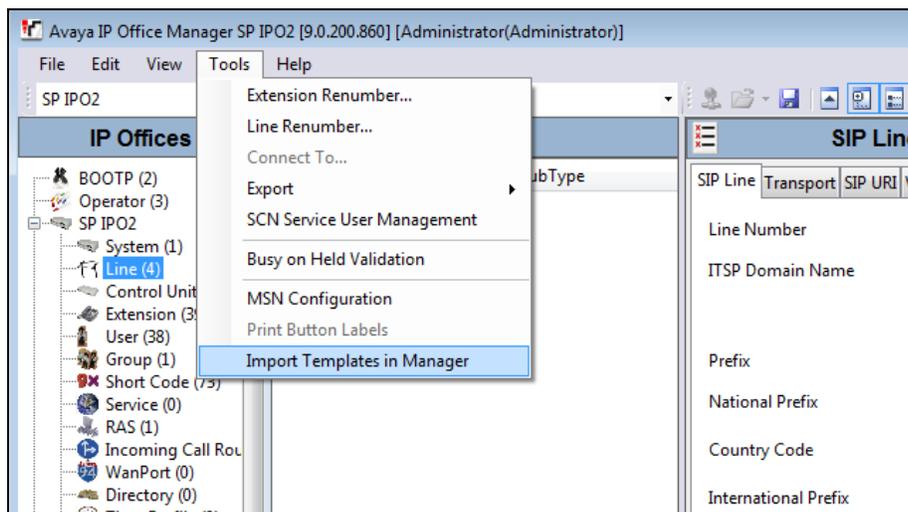
Alternatively, a SIP Line can be created manually, in order to do so right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.6.2**.

5.6.1. Create SIP Line from Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **NL_Vodafone_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.
2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the Visual Preferences tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

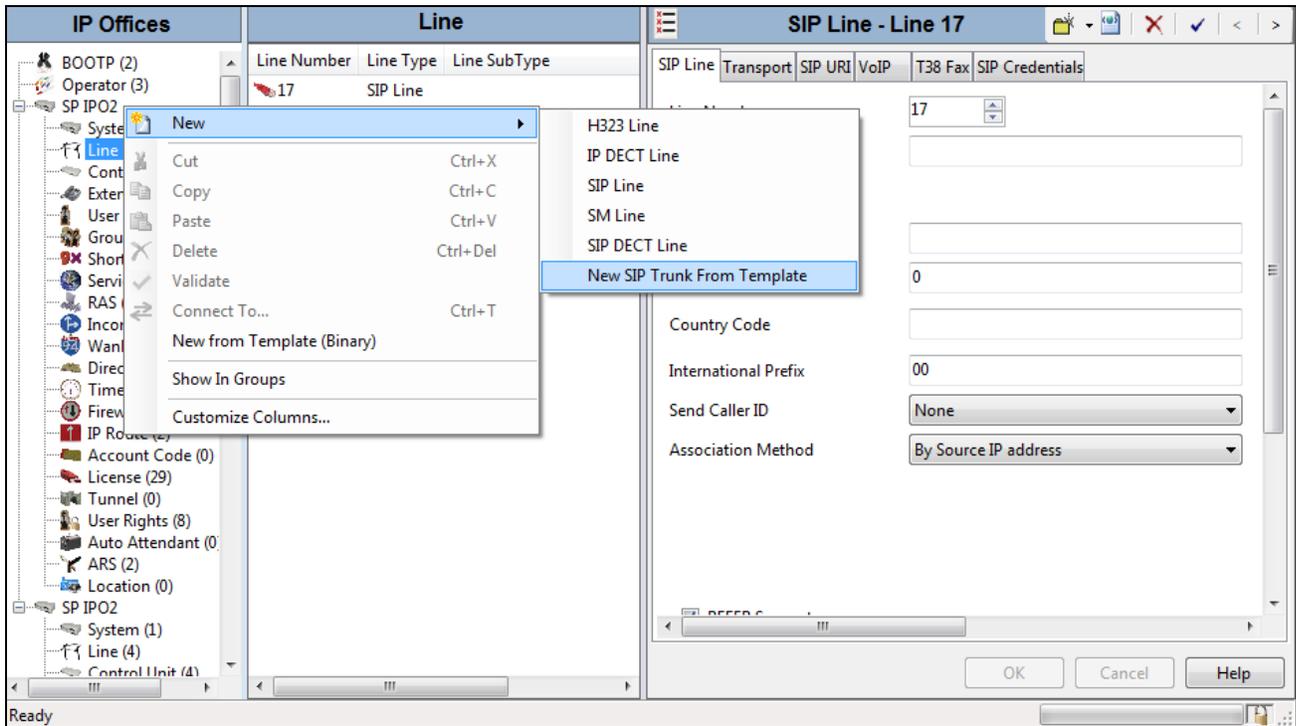


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office Manager template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

- To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New SIP Trunk From Template**.



- In the subsequent **Template Type Selection** pop-up window, select Netherlands from the **Country** pull-down menu and select Vodafone from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**NL_Vodafone_SIPTrunk.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



5.6.2. Create SIP Line Manually

To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New** → **SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- **ITSP Domain Name** field should remain blank as Vodafone NL SIP Trunking have not provided a Domain Name.
- Set **Send Caller ID** to **None** as it is only required if the box labeled **Send original calling party information for Mobile Twinning** is unchecked in **Section 5.4**.
- Ensure the **In Service** box is checked.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Default values may be used for all other parameters.

The area of the screen entitled **REFER Support** is used to enable/disable SIP REFER for call transfers. The default values of “Auto” for **Incoming** and **Outgoing** effectively disable use of SIP REFER and use Re-Invite for the call transfer. In the compliance test, only Re-Invite method was successfully tested to transfer a call between a PSTN phone and an enterprise phone to a second PSTN phone.

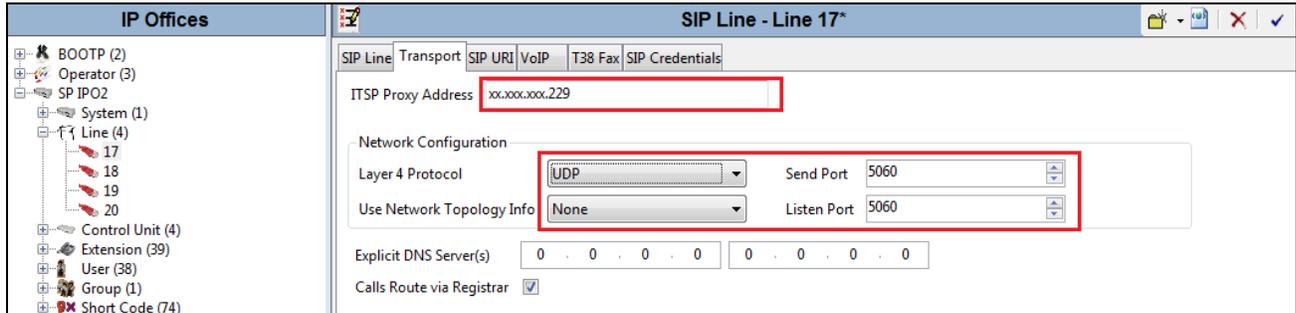
On completion, click the **OK** button (not shown).

Field	Value
Line Number	17
ITSP Domain Name	
Prefix	
National Prefix	0
Country Code	
International Prefix	00
Send Caller ID	None
Association Method	By Source IP address
In Service	<input checked="" type="checkbox"/>
URI Type	SIP
Check OOS	<input checked="" type="checkbox"/>
Call Routing Method	Request URI
Originator number for forwarded and twinning calls	
Name Priority	System Default
Caller ID from From header	<input checked="" type="checkbox"/>
Send From In Clear	<input type="checkbox"/>
User-Agent and Server Headers	
Service Busy Response	486 - Busy Here
Action on CAC Location Limit	Allow Voicemail
REFER Support	<input checked="" type="checkbox"/>
Incoming	Auto
Outgoing	Auto

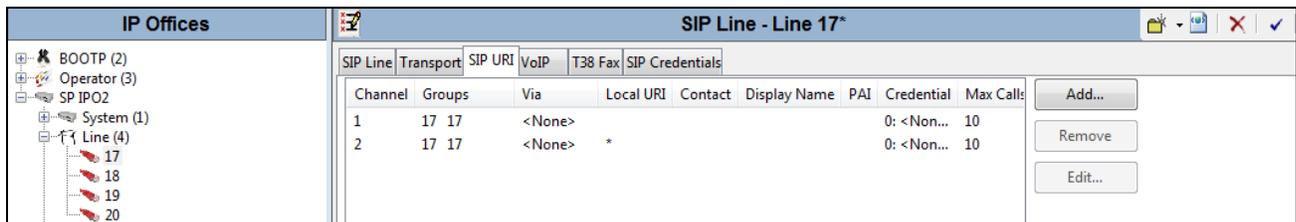
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the IP address of the VoV service on the Vodafone NL SIP proxy.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Send Port** and **Listen Port** to **5060**.
- Set **Use Network Topology Info** to **None**.

On completion, click the **OK** button (not shown).



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 and the **New Channel** area will appear at the bottom of the pane.



For the compliance test, two SIP URI entries were created for the VoV SIP line. One entry that matched any number assigned to an IP Office user. The entry was created with the parameters shown below.

- Set **Local URI** to *, This setting allows all calls with numbers defined in **Incoming Call Routing** as shown in **Section 5.9**.
- For **Registration**, select **0: <None>** from the pull-down menu since this configuration does not use SIP registration.
- 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 **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.

On completion, click the **OK** button.

Edit Channel

Via	<None>
Local URI	*
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	Use Internal Data
Registration	0: <None>
Incoming Group	17
Outgoing Group	17
Max Calls per Channel	10

OK
Cancel

The second SIP URI entry was used for outgoing calls and used to send caller ID configured in **SIP Name** in the **User → SIP** section of **Section 5.8**. On completion, click the **OK** button.

Edit Channel

Via	<None>
Local URI	Use Internal Data
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	Use Internal Data
Registration	0: <None>
Incoming Group	17
Outgoing Group	17
Max Calls per Channel	10

OK
Cancel

Repeat the process to define a SIP line for Corporate Net over IP (CNoIP). Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the IP address of the CNoIP service on the Vodafone NL SIP proxy.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Send Port** and **Listen Port** to **5060**.

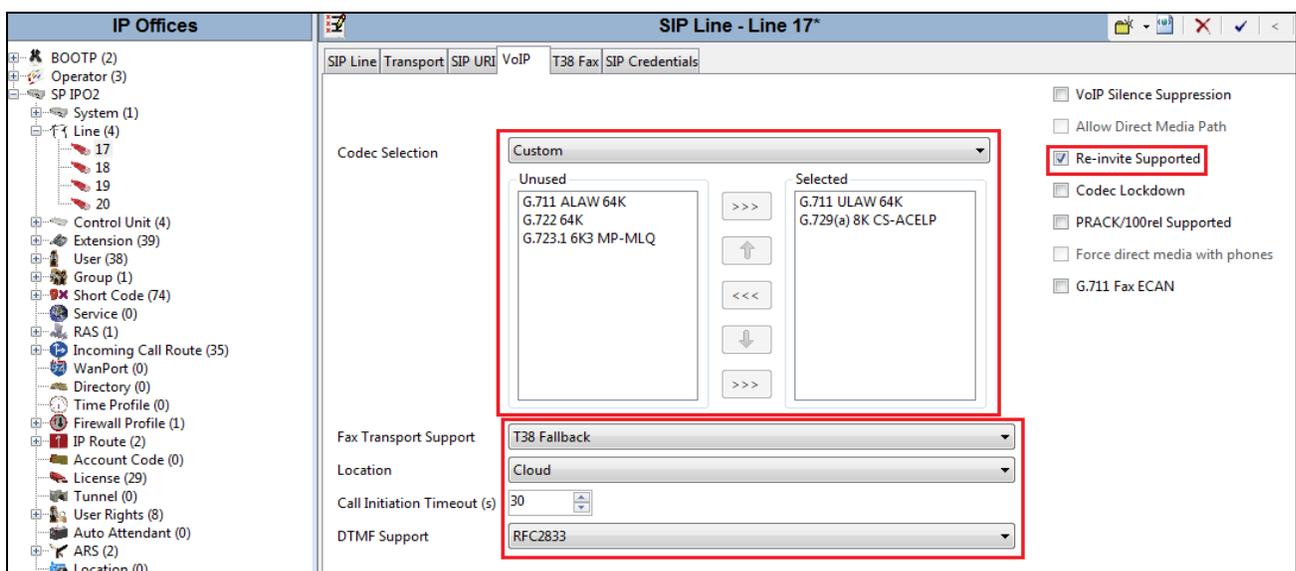
On completion, click the **OK** button (screenshots not shown). **Note:** In the testing, line 17 was used for VoV and line 18 was used for CNoIP.

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

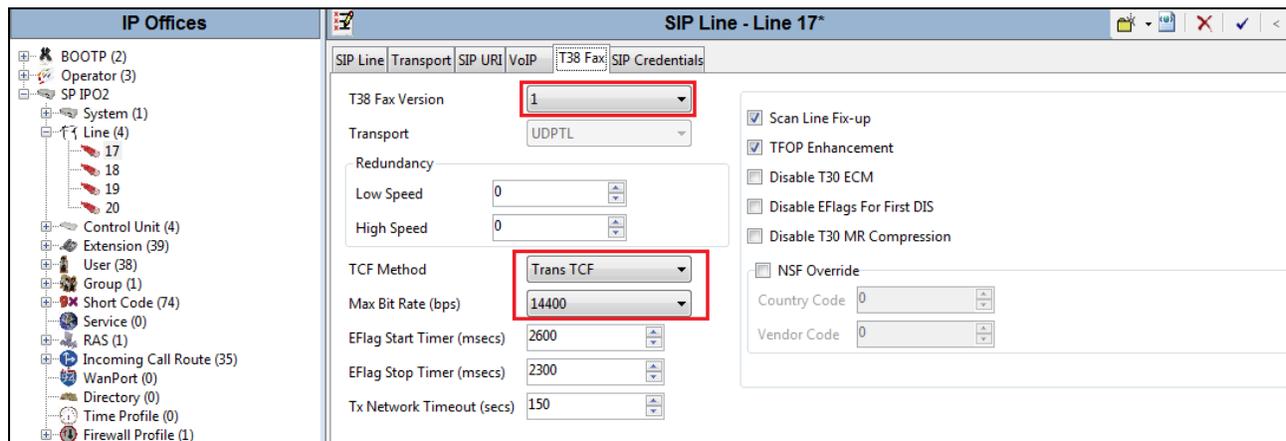
- Select **Custom** in the **Codec Selection** drop down menu to specify the preferred codecs.
- Highlight codecs in the **Unused** box that are to be used on this line and click on the right arrows to move them to the **Selected** box.
- Highlight codecs in the **Selected** box that are not to be used and click on the left arrows to move them to the **Unused** box.
- Highlight codecs in the **Selected** box and use the up and down arrows to change the priority order of the offered codecs, for testing with Vodafone NL this was **G.711 ALAW 64K** followed by **G.729(a) 8K CS-ACELP**.
- Select **T38 Fallback** in the **Fax Transport Support** drop down menu to allow T.38 fax operation.
- Select **RFC2833** in the **DTMF Support** drop down menu. This directs IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** 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.

On completion, click the **OK** button (not shown).



Select the **T.38 Fax** tab to set the T.38 parameters for the line. Un-check the **Use Default Values** box and select **1** from the **T38 Fax Version** drop down menu. Set the **Max Bit Rate (bps)** to 14400. All other field may retain their default values. On completion, click the **OK** button (not shown).

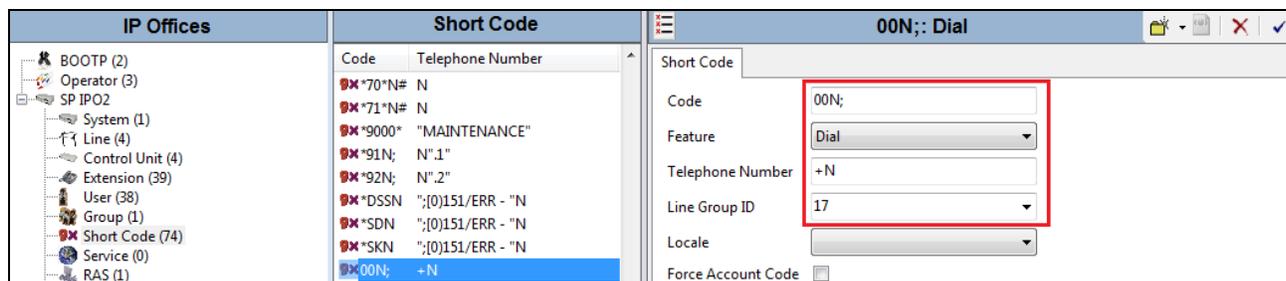


5.7. Short Code

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.

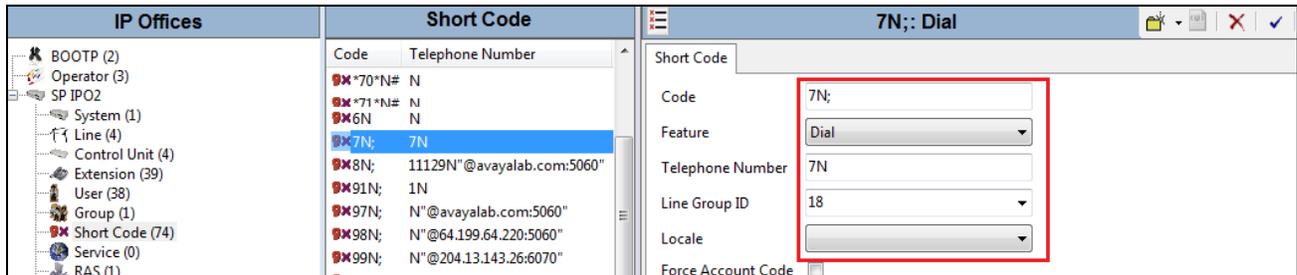
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon.
- The example shows **00N**; which will be used for international call in Netherlands and invoked when the user dials 00 followed by an international number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **+N** which will insert the E.164 number prefixed with a + in the Request URI and To headers in the outgoing SIP INVITE message.
- Set the **Line Group ID** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in [Section 5.6.2](#).

On completion, click the **OK** button (not shown).



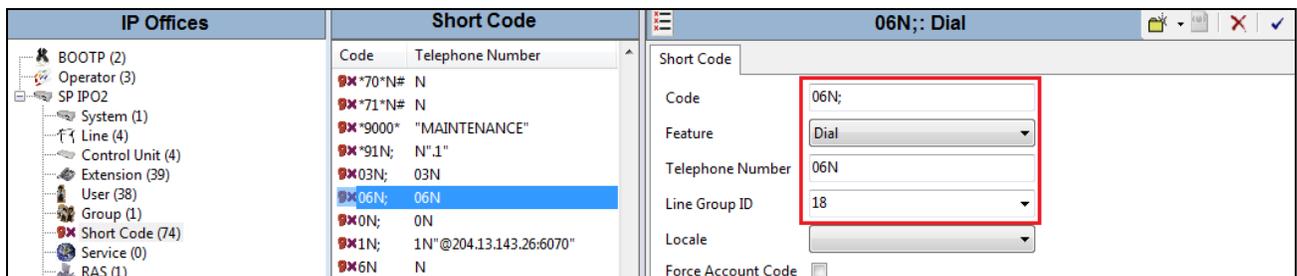
Short codes are also used for routing of national fixed and mobile calls, as well as VPN calls for CNoIP. National fixed line calls use the SIP line established for the Vodafone Office Voice (VoV) service. National mobile and VPN calls use the SIP line established for the Corporate Net over IP (CNoIP). An example for VPN calls is shown below.

- The example of a VPN call shows **7N**; which will be invoked when the user dials a four digit VPN number.
- Set **Telephone Number** to **7N** which leaves the number unchanged.
- Set the **Line Group ID** to the outgoing line group number for CNoIP defined on the **SIP URI** tab on the **SIP Line** in **Section 5.6.2**.
- Set other parameters as shown in the previous example.



An example for mobile calls is shown below.

- The example of a mobile call shows **06N**; which will be invoked when the user dials mobile numbers starting with 06.
- Set **Telephone Number** to **06N**.
- Set the **Line Group ID** to the outgoing line group number for CNoIP defined on the **SIP URI** tab on the **SIP Line** in **Section 5.6.2**.
- Set other parameters as shown in the previous examples.



5.8. User

Configure the SIP parameters for each User that will be placing and receiving calls via the SIP lines defined in **Section 5.6**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **User** tab if any changes are required. Changes are not normally required where only the newly established SIP line is to be used for an existing User. In the example below, the User is configured to use IP Office Softphone.

- Change the **Name** of the User if required, this will be used for login to the IP Office Softphone.
- Select **Teleworker User** from the **Profile** drop down menu.
- Check the **Enable Softphone** box to support IP Office Softphone.

Name	Extension
Ext210	210
Ext211	211
Ext212	212
Ext213	213
Ext214	214
Ext215	215
Ext216	216
Ext217	217
Ext218	218
Ext219	219
Ext220	220
Ext221	221
Ext222	222
Ext223	223
Ext224	224
Ext225	225
Ext226	226
Ext227	227
Ext228	228
Ext229	229
Ext230	230
Ext231	231
Ext232	232
Ext233	233
Ext234	234
Ext235	235
Ext236	236

Select the **SIP** (not shown) tab in the Details Pane. To reach the **SIP** tab, click the right arrow on the right hand side of the Details Pane until the **SIP** tab appears. 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. These allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.6**). As such, these fields should be set to one of the DDI numbers assigned to the enterprise from Vodafone NL.

In the test, the digits received in the SIP URI were in national format. The received digits were provisioned for the User, these have been obscured in the screenshot below. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. On completion, click the **OK** button (not shown). **Note:** The **Contact** field must be in E.164 format for the caller ID on the called phone to display properly.

Name	Extension
Ext210	210
Ext211	211
Ext212	212
Ext213	213
Ext214	214
Ext215	215
Ext216	216

5.9. Incoming Call Route

An incoming call route maps an inbound DDI or VPN number on a specific line to an internal extension. The line is dependent on whether the call is Vodafone Office Voice (VoV) or Corporate Net Over IP (CNoIP). To create an incoming call route for VoV, 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 for VoV defined in **Section 5.6**.
- 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.

Line Group ID	Incoming Number	Destination
0	19	230 Extn230
0		200 Main
0		DialIn
17	03835	235 Extn235
17	03836	236 Extn236
17	03837	FNE33
17	03832	232 Extn232
17	03831	201 Extn201
17	03830	230 Extn230
17	03833	233 Extn233
17	03834	234 Extn234
18	6472	231 Extn231
18	6480	VoiceMail
18	6478	FNE00
18	6477	230 Extn230
18	6476	229 Extn229

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 17 are routed to extension **235**.

TimeProfile	Destination	Fallback Extension
Default Value	235 Extn235	

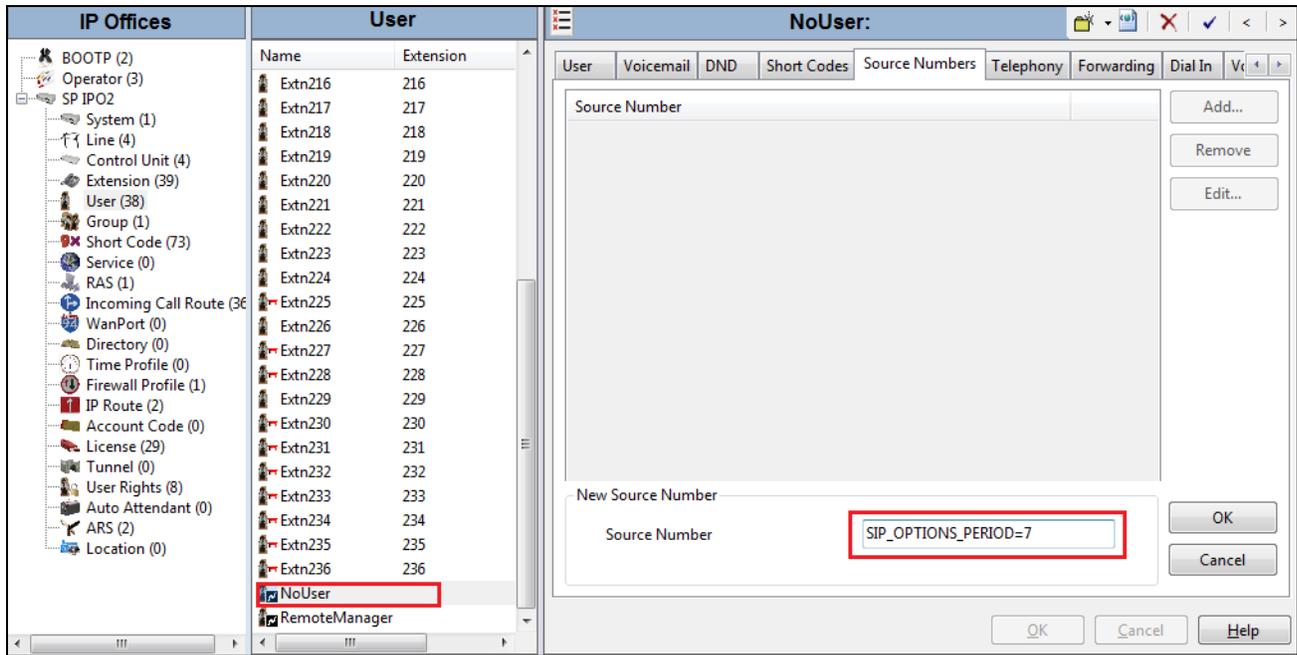
5.10. SIP Options

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** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **noUser** user. The OPTIONS period is determined in the following manner:

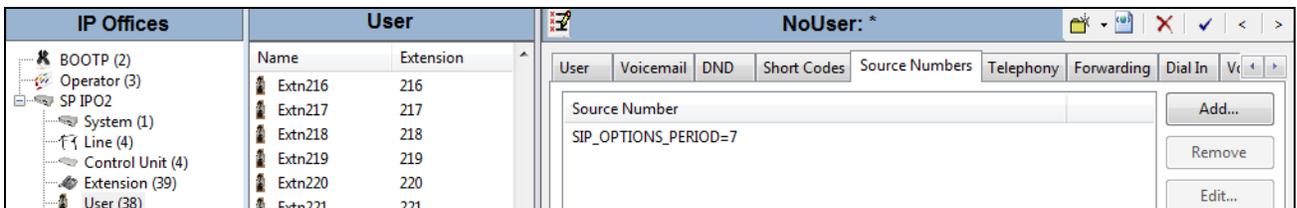
- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 300 seconds is used.
- To establish a period less than 300 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds.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

- To establish a period greater than 300 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 seconds. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User** → **noUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button (not shown). 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. Note that for the compliance test, an OPTIONS period of 1 minute was desired. The **Binding Refresh Time** was set to **60** seconds (1 minute) in **Section 5.2**. The **SIP_OPTIONS_PERIOD** was not created and used at all. This section was just mentioned for those who wish to configure the IP Office sending out the OPTIONS greater than 300 seconds.



5.11. Save Configuration

Navigate to **File** → **Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

6. Vodafone NL SIP Trunking Configuration

Vodafone NL is responsible for the configuration of Vodafone SIP Trunking service. The customer will need to provide the IP address used to reach IP Office at the enterprise. Vodafone will provide the customer the necessary information to configure IP Office SIP connection to Vodafone. The provided information from Vodafone includes:

- IP address of the Vodafone SIP proxy.
- Supported codecs.
- DDI numbers.
- IP addresses and port numbers used for signaling or media through any security devices.

7. Verification Steps

The following steps may be used to verify the configuration:

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

The screenshot shows the AVAYA IP Office System Status application. The left sidebar contains a navigation tree with 'System' expanded and 'Line: 17' selected. The main window displays the 'SIP Trunk Summary' for the selected line. The summary includes fields for Peer Domain Name, Resolved Address, Line Number, Number of Administered Channels, Number of Channels in Use, Administered Compression, Silence Suppression, Layer 4 Protocol, SIP Trunk Channel Licenses, and SIP Trunk Channel Licenses in Use. A green progress indicator shows 0% utilization. Below the summary is a table with columns for Channel Ref, Call Ref, Current State, Time in State, Remote Media, Co... Conn..., Caller ID or..., Other Party on Call, Direc..., Round Trip, Recei..., Rece..., Tran..., and Tran... The table shows 8 channels, all in an 'Idle' state with a time in state of '00:0...'. The table is scrollable, with a scrollbar on the right side.

Chan...	U...	Call	Curr...	Time in	Remote	Co...	Conn...	Caller	Other Party	Direc...	Round	Recei...	Rece...	Tran...	Tran...
Ref		Ref	State	State	Media ...			ID or...	on Call		Trip ...				
1			Idle	00:0...											
2			Idle	00:0...											
3			Idle	00:0...											
4			Idle	00:0...											
5			Idle	00:0...											
6			Idle	00:0...											
7			Idle	00:0...											
8			Idle	00:0...											

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

The screenshot shows the AVAYA IP Office System Status interface. The 'Alarms' tab is selected, displaying a table for 'Alarms for Line: 17 SIP sip://62. [redacted] 229'. The table has three columns: 'Last Date Of Error', 'Occurrences', and 'Error Description'. The table is currently empty, indicating no active alarms.

- Verify that a phone connected to PSTN can successfully place a call to the IP Office with two-way audio.
- Verify that a phone connected to IP Office can successfully place a call to the PSTN with two-way audio.
- Using a network sniffing tool e.g. Wireshark to monitor the SIP signalling between the enterprise and Vodafone. The sniffer traces are captured at the public interface of IP Office.

Following screenshots show an example incoming call from Vodafone to the enterprise.

- Incoming INVITE request from Vodafone.

```
INVITE sip:038xxxxx@10.0.1.10:5060 SIP/2.0
Via: SIP/2.0/UDP 61.xxx.xxx.229:5060;branch=z9hG4bK28BB3811FA
From: <sip:001613xxxxx58@61.xxx.xxx.229>;tag=BEDDA55C-83F
To: <sip:038xxxxx@10.0.1.10>
Date: Wed, 23 Apr 2014 18:04:20 GMT
Call-ID: 87661A76-CA4811E3-A7C4F60A-331B0F48@61.xxx.xxx.229
Supported: timer,resource-priority,sdp-anat
Min-SE: 1800
User-Agent: Vodafone-NL-SIP-Gateway-V1.1
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, REGISTER
CSeq: 101 INVITE
Timestamp: 1398276260
Contact: <sip:001613xxxxx58@61.xxx.xxx.229:5060>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 67
P-Preferred-Identity: <sip:001613xxxxx58@61.xxx.xxx.229>
Session-Expires: 1800
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 287

v=0
o=CiscoSystemsSIP-GW-UserAgent 4448 748 IN IP4 61.xxx.xxx.229
s=SIP Call
c=IN IP4 61.xxx.xxx.229
t=0 0
m=audio 23834 RTP/AVP 8 18 101
c=IN IP4 61.xxx.xxx.229
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
```

```
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

- **Outgoing 200OK response from the enterprise.**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 61.xxx.xxx.229:5060;branch=z9hG4bK28BB3811FA
From: <sip:001613xxxxx58@61.xxx.xxx.229>;tag=BEDDA55C-83F
Call-ID: 87661A76-CA4811E3-A7C4F60A-331B0F48@61.xxx.xxx.229
CSeq: 101 INVITE
Contact: "Extn234" <sip:038xxxxx@10.0.1.10:5060;transport=udp>
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,NOTIFY
Supported: timer
Server: IP Office 9.0.2.0 build 860
Min-SE: 1800
Require: timer
Session-Expires: 1800;refresher=uac
To: <sip:038xxxxx@10.0.1.10>;tag=bd8f6b8208b0e3d0
Content-Type: application/sdp
Content-Length: 198

v=0
o=UserA 1819019355 3393600283 IN IP4 10.0.1.10
s=Session SDP
c=IN IP4 10.0.1.10
t=0 0
m=audio 49154 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Following screenshots show an example outgoing call from the enterprise to Vodafone.

- **Outgoing INVITE request from the enterprise.**

```
INVITE sip:+1613xxxxx58@61.xxx.xxx.229 SIP/2.0
Via: SIP/2.0/UDP 10.0.1.10:5060;rport;branch=z9hG4bK7d313b45dcba378c779caec14abacdef
From: "Extn234" <sip:038xxxxx@61.xxx.xxx.229>;tag=4dc2d7546038c8c9
To: <sip:+1613xxxxx58@61.xxx.xxx.229>
Call-ID: e4a2f3eb5220e495e77aa42f982428af
CSeq: 990816612 INVITE
Contact: "Extn234" <sip:038xxxxx@10.0.1.10:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,NOTIFY
Content-Type: application/sdp
Supported: timer
User-Agent: IP Office 9.0.2.0 build 860
P-Asserted-Identity: "Extn234" <sip:038xxxxx@10.0.1.10:5060>
Content-Length: 245

v=0
o=UserA 3618421882 3791846658 IN IP4 10.0.1.10
s=Session SDP
c=IN IP4 10.0.1.10
t=0 0
m=audio 49154 RTP/AVP 8 18 101
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

- Incoming 200OK response from Vodafone.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.1.10:5060;rport;branch=z9hG4bK7d313b45dcba378c779caec14abacdef
From: "Extn234" <sip:038xxxxx@61.xxx.xxx.229>;tag=4dc2d7546038c8c9
To: <sip:+1613xxxxx58@61.xxx.xxx.229>;tag=BEDDC954-16A1
Date: Wed, 23 Apr 2014 18:04:29 GMT
Call-ID: e4a2f3eb5220e495e77aa42f982428af
CSeq: 990816612 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, REGISTER
Allow-Events: telephone-event
Contact: <sip:+1613xxxxx58@61.xxx.xxx.229:5060>
Supported: replaces
Supported: sdp-anat
Server: Cisco-SIPGateway/IOS-15.2.4.M3
Session-Expires: 1800;refresher=uac
Require: timer
Supported: timer
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 253

v=0
o=CiscoSystemsSIP-GW-UserAgent 5094 4509 IN IP4 61.xxx.xxx.229
s=SIP Call
c=IN IP4 61.xxx.xxx.229
t=0 0
m=audio 23836 RTP/AVP 8 101
c=IN IP4 61.xxx.xxx.229
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
```

8. Conclusion

Vodafone NL SIP Trunking passed the compliance testing. These Application Notes describe the procedures required to configure the SIP connection between Avaya IP Office and Vodafone NL SIP Trunking service as shown in **Figure 1**.

9. Additional References

- [1] IP Office 9.0 Installation, Document number 15-601042 Issue 28, 16 April 2014
- [2] IP Office 9.0 Manager 9.0, Document number 15-601011 Issue 9.01, 16 April 2014
- [3] IP Office 9.0 Administering Voicemail Pro, Document number 15-601063 Issue 9.0 Release 1.0, April 2014
- [4] IP Office Embedded Voicemail User Guide (IP Office Mode), Document number 15-604067 Issue 9.0, 16 April 2014

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

<http://marketingtools.avaya.com/knowledgebase/>

Product documentation for Vodafone NL SIP Trunking is available from Vodafone Netherlands.

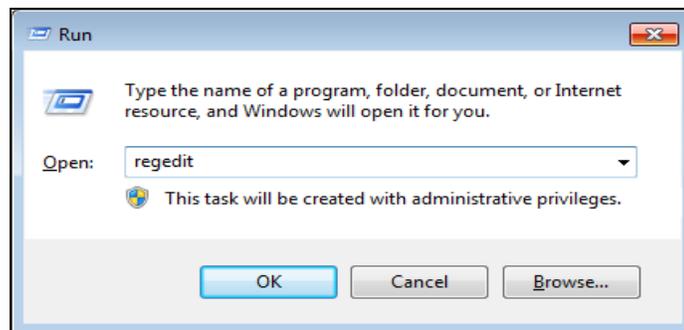
10. Appendix: SIP Line Template

Avaya IP Office Release 9.0 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

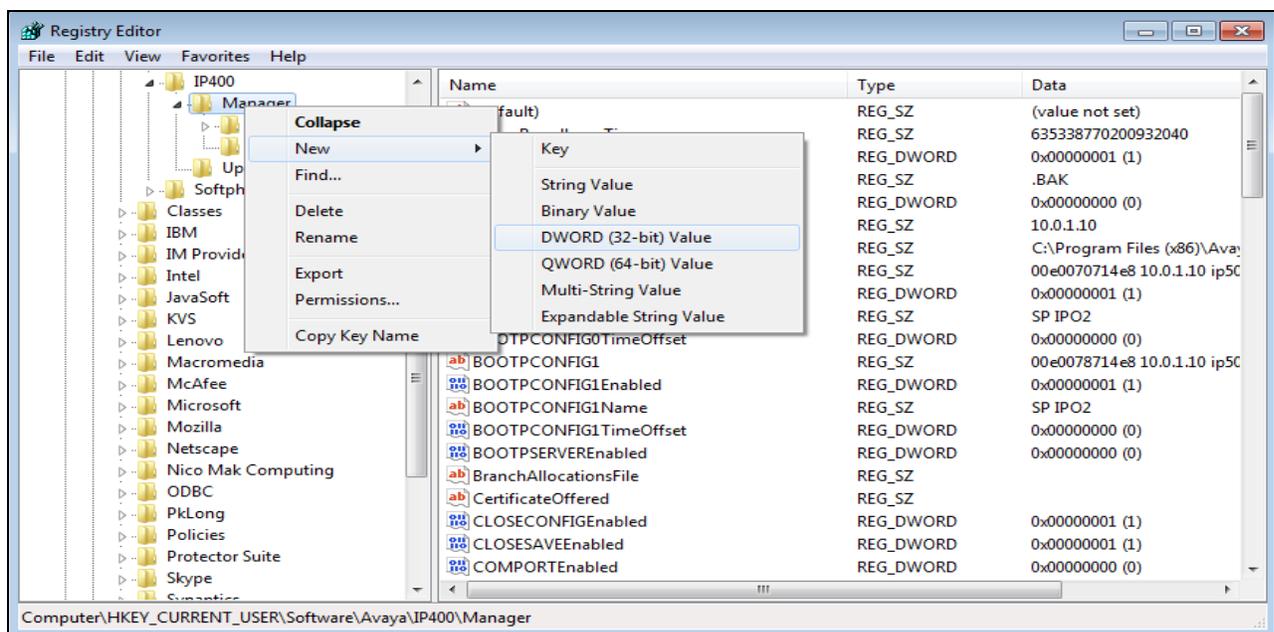
Not all of the configuration information is included in the SIP Line Template, therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported, and additional configuration be supplemented using **Section 5.6.2** in these Application Notes as a reference.

To create a SIP Line Template from the configuration described in these Application Notes, configure the parameters as described below.

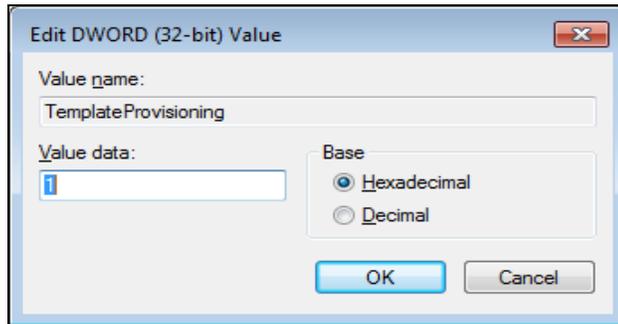
Use the Windows Registry Editor on the PC where IP Office Manager is installed to add a new *TemplateProvisioning* registry entry. Select **Start** → **Run**. Enter *regedit* in the **Open** box.



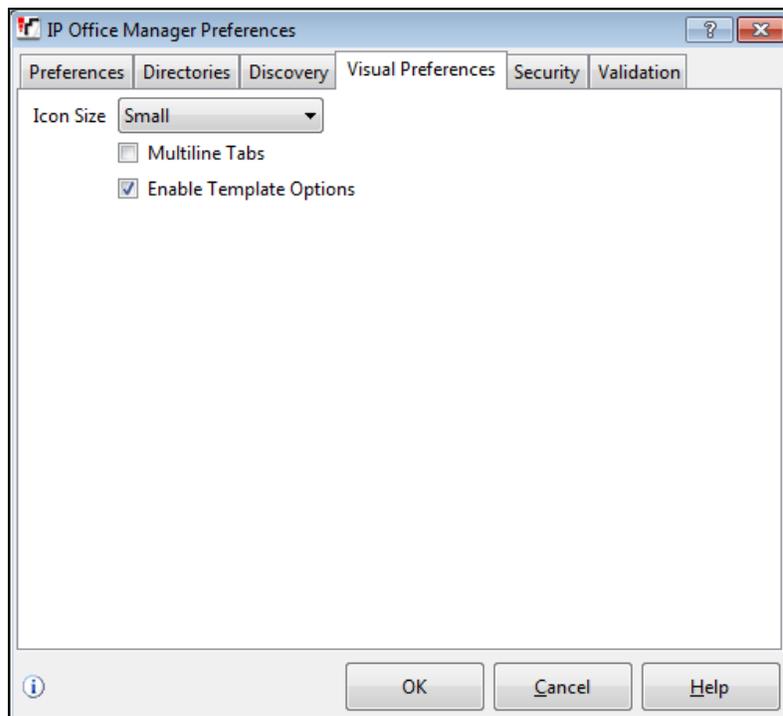
On the Registry Editor, navigate to **HKEY_CURRENT_USER** → **Software** → **Avaya** → **IP400**. Right click on **Manager** and select **New** → **DWORD Value**.



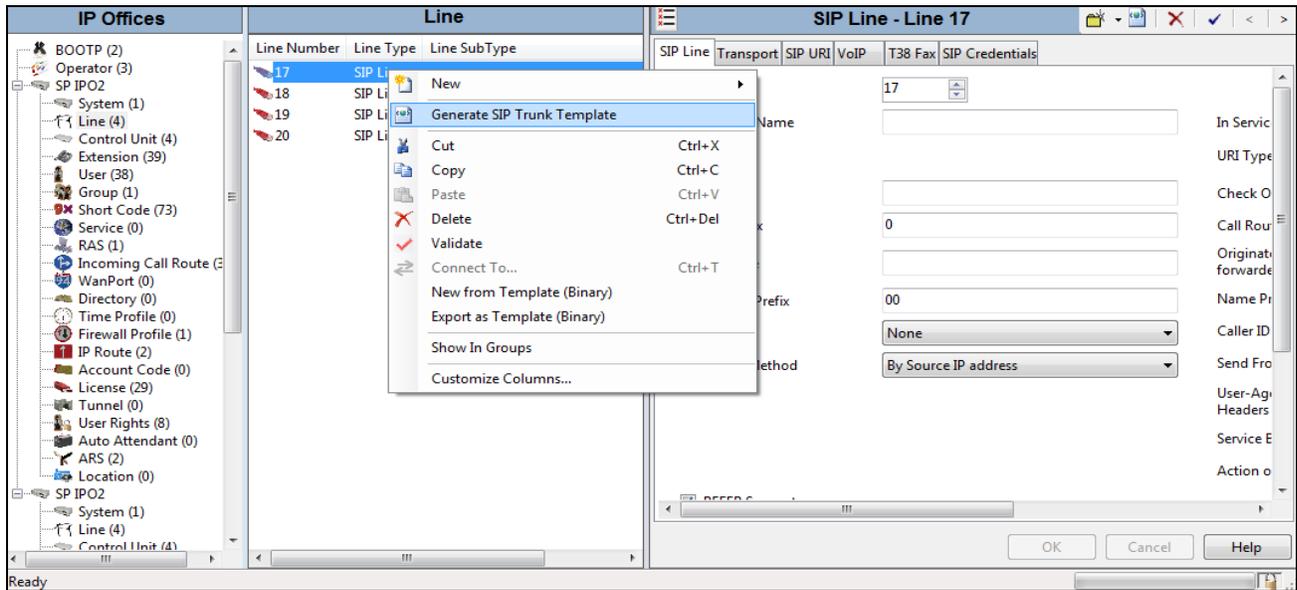
Right click the newly created entry and rename it to **TemplateProvisioning**. Double click the entry and change the value under **Value Data** from “0” to “1”. Restart the PC.



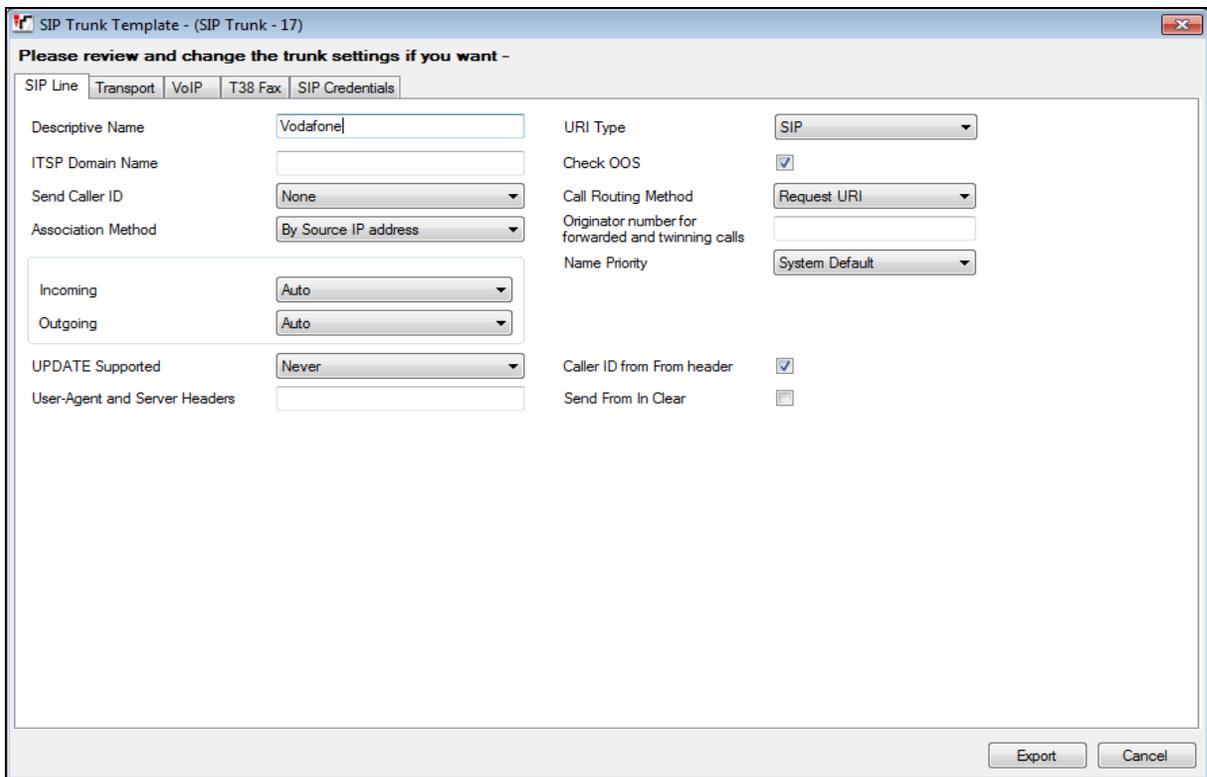
To enable template support in the IP Office Manager, select **File**, then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.



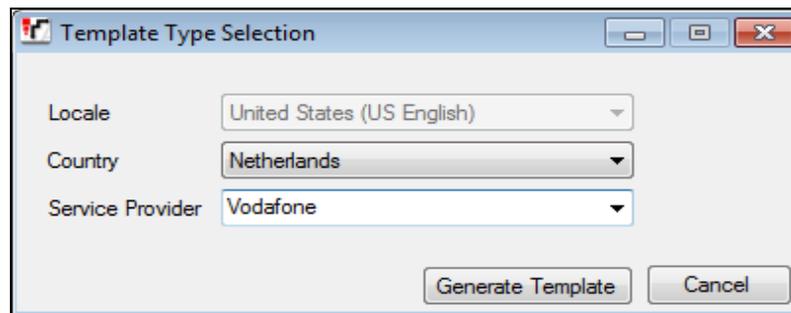
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click the Sip Line (17), and select **Generate SIP Trunk Template**. Note that the Generate SIP Trunk Template is only presented when the *TemplateProvisioning* key created in the PC where the IP office Manager installed.



The trunk's settings are displayed as configured in **Section 5.6.2** Enter a descriptive name for the template, adjust the settings if required, and then click on **Export**.



On the next screen, **Template Type Selection**, select the **Country**, enter the name for the **Service Provider**, and click **Generate Template**.



The following is the exported SIP Line Template file NL_Vodafone_SIPTrunk.xml.

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20140423</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Vodafone</DescriptiveName>
  <ITSPDomainName> </ITSPDomainName>
  <SendCallerID>CallerIDNone</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>2</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateNever</UpdateSupport>
  <URIType>SIPURI</URIType>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>true</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>62.xxx.xxx.229</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
  <AdvCodecPref>G.711 ALAW 64K,G.729(a) 8K CS-ACELP</AdvCodecPref>
  <CallInitiationTimeout>30</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_T38FB</FaxTransportSupport>
  <UseOffererPreferredCodec>false</UseOffererPreferredCodec>
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>false</Rel100Supported>
  <T38FaxVersion>1</T38FaxVersion>
  <Transport>UDPTL</Transport>
```

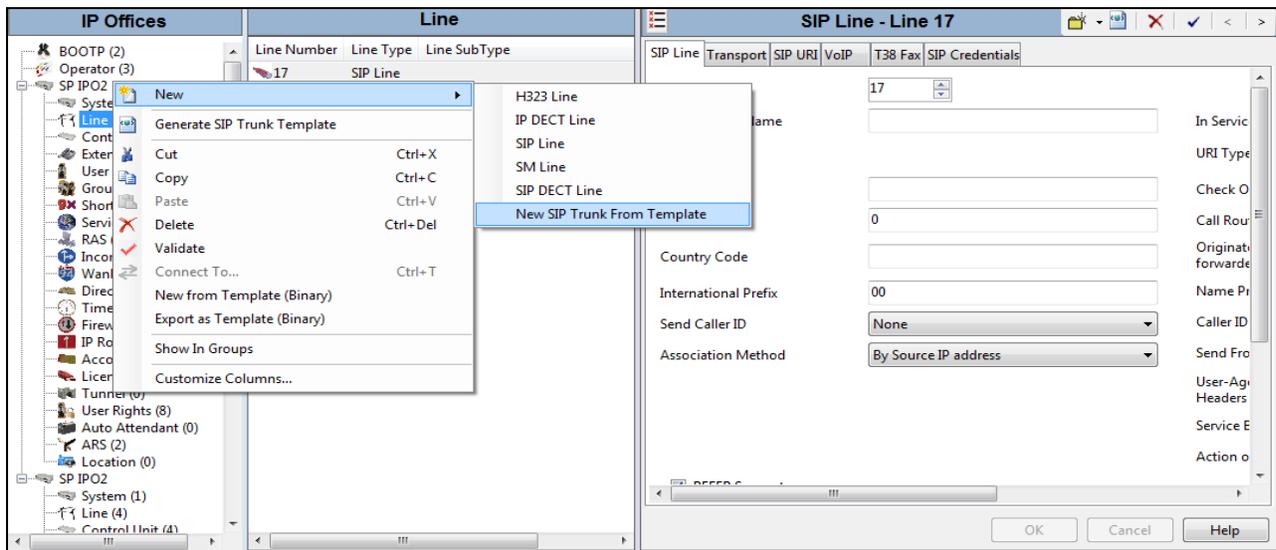
```

<LowSpeed>0</LowSpeed>
<HighSpeed>0</HighSpeed>
<TCFMethod>Trans_TCF</TCFMethod>
<MaxBitRate>FaxRate_14400</MaxBitRate>
<EflagStartTimer>2600</EflagStartTimer>
<EflagStopTimer>2300</EflagStopTimer>
<UseDefaultValues>>false</UseDefaultValues>
<ScanLineFixup>>true</ScanLineFixup>
<TFOPENenhancement>>true</TFOPENenhancement>
<DisableT30ECM>>false</DisableT30ECM>
<DisableEflagsForFirstDIS>>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>>false</DisableT30MRCompression>
<NSFOVERRIDE>>false</NSFOVERRIDE>
</Template>

```

To import the template into a new IP Office system, copy and paste the exported xml template file into the Templates directory (C:\Program Files\Avaya\IP Office\Manager\Templates) on the PC where IP Office Manager for the new system is running.

Next, import the template into the new IP Office system by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New, New SIP Trunk From Template**:



On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.



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