Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office R8.0 with Vodafone NL SIP Trunking Service – 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.

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.

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 Vodafone NL SIP Trunking Service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality 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

Avaya IP Office was connected to the Vodafone NL SIP Trunking Service. To verify SIP trunking 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
- 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 (1802)
- Codecs G.711A and G.711Mu
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer and conference
- Off-net call forwarding and twinning
- T.38 fax

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Vodafone NL SIP Trunking Service with the following observations:

- No inbound toll free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested
- As an Emergency Services test was not booked, no call was made to the Operator
- Transmission of incoming multi-page T.38 Fax was unreliable during test and failed often before all pages were received

2.3. Support

For technical support on Vodafone Netherlands SIP trunking services, contact Vodafone Netherlands support 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 two Avaya 1600 Series IP Telephones (with H.323 firmware), an Avaya 2420 Digital Telephone, an Avaya Analogue Telephone and a fax machine. The site also has a Windows XP 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 obscured and all phone numbers have been obscured beyond the city code.

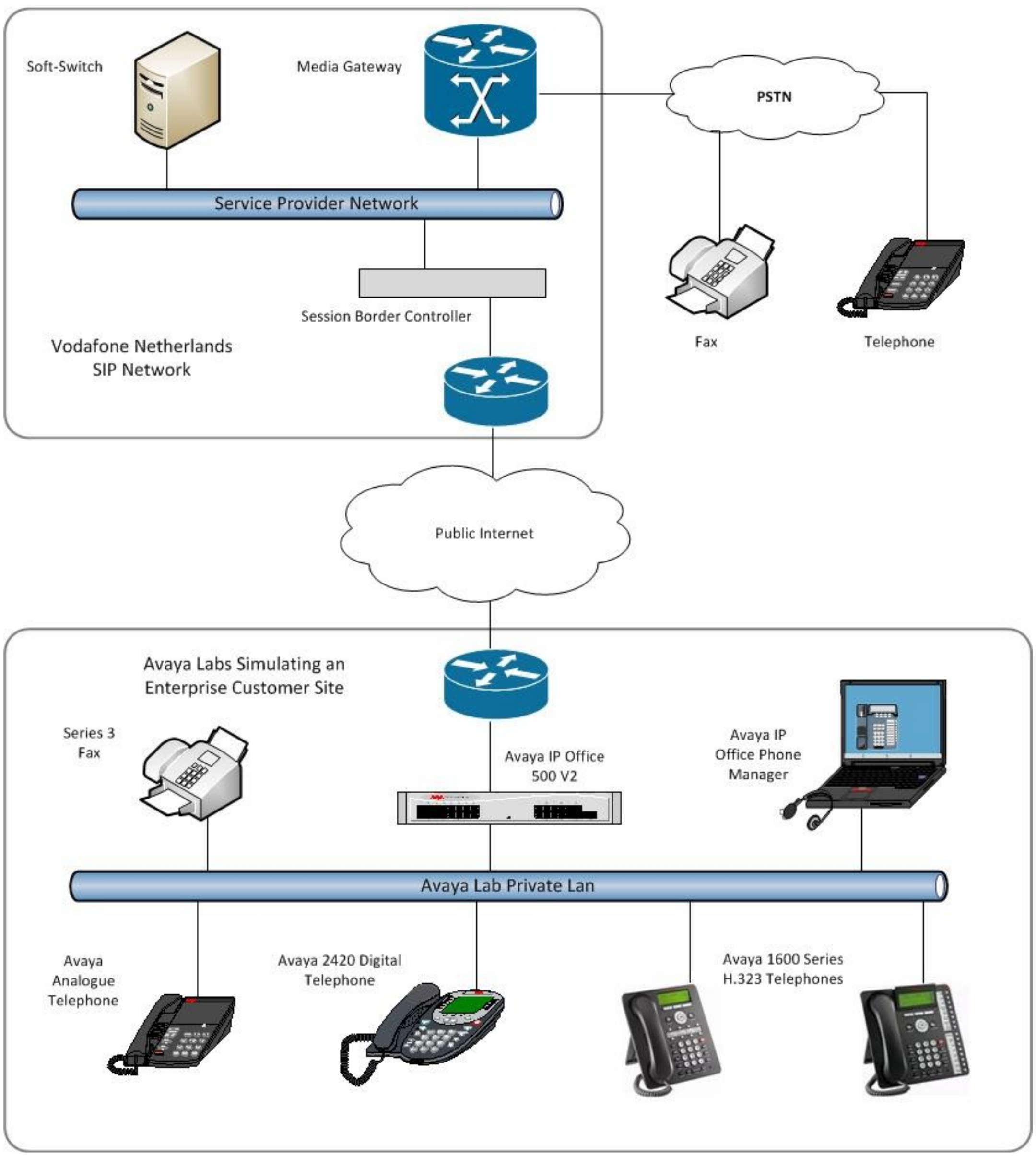


Figure 1: Vodafone NL SIP Trunking Service Solution to Avaya IP Office Topology

Avaya IP Office was configured to connect to a static IP address at the Service Provider. For the purpose of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Vodafone NL. The short code of 9 is stripped off by Avaya IP Office 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 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. Vodafone NL sends SIP signalling 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 Vodafone NL to determine the proper IP addresses and ports that require access to their network.

4. Equipment and Software Validated

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

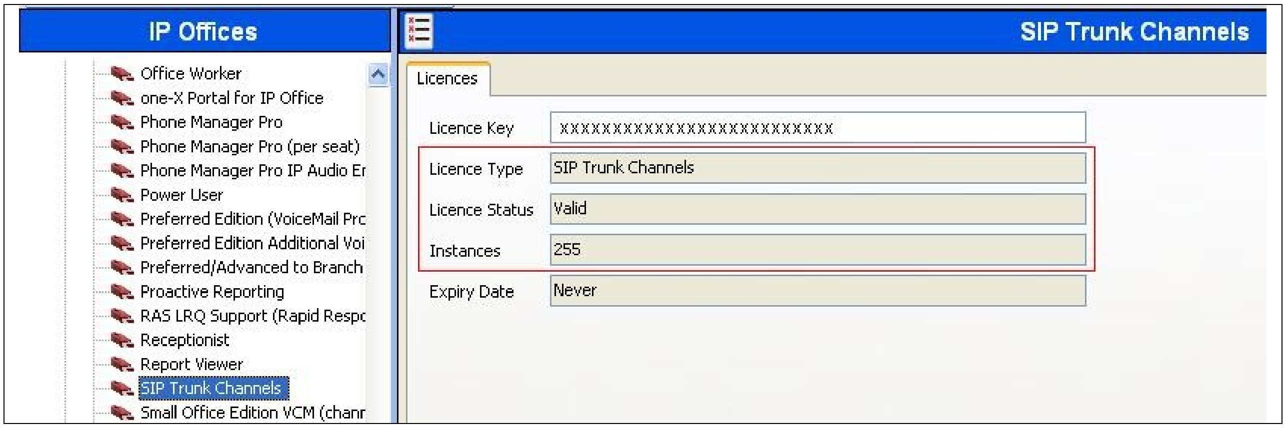
Equipment	**Software**
Avaya IP Office 500 V2	R8.0(16)
Avaya 1603 Phone (H.323)	1.3
Avaya 1608 Phone (H.323)	1.3
Avaya 2420 Digital Phone	N/A
Avaya 98390 Analogue Phone	N/A
Vodafone NL equipment	**Software**
Vodafone Office Voice	1.0
Vodafone OneVoice Corporate	1.0
ACME Net-Net 4500 Firmware	SCX6.2.0 MR-6 Patch 2 (build 876)

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the Vodafone NL SIP Trunking 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) is assumed to already be in place.

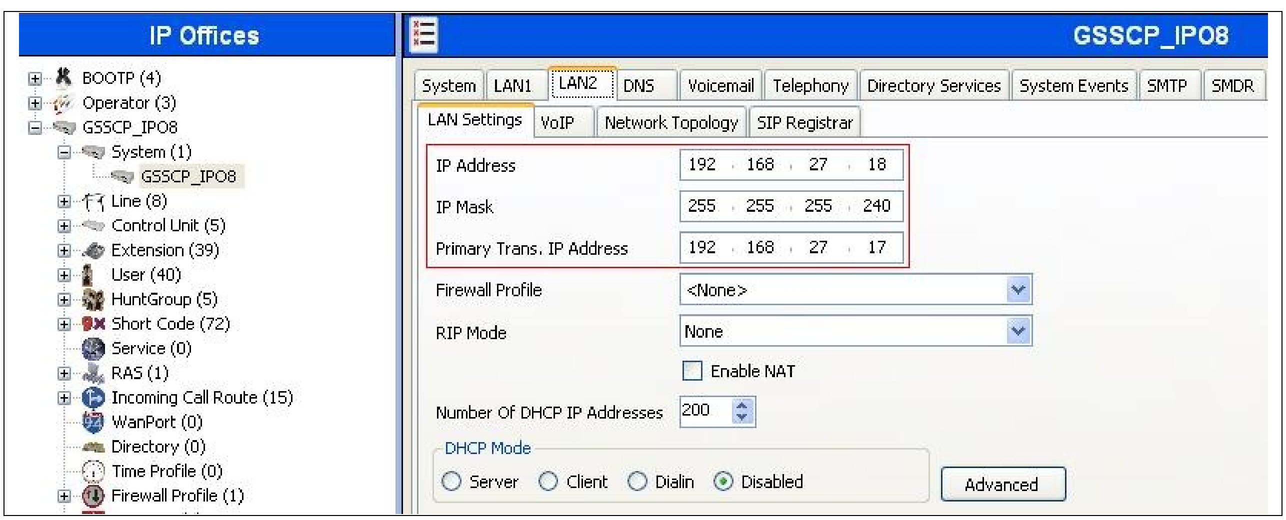
5.1. Verify System Capacity

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.

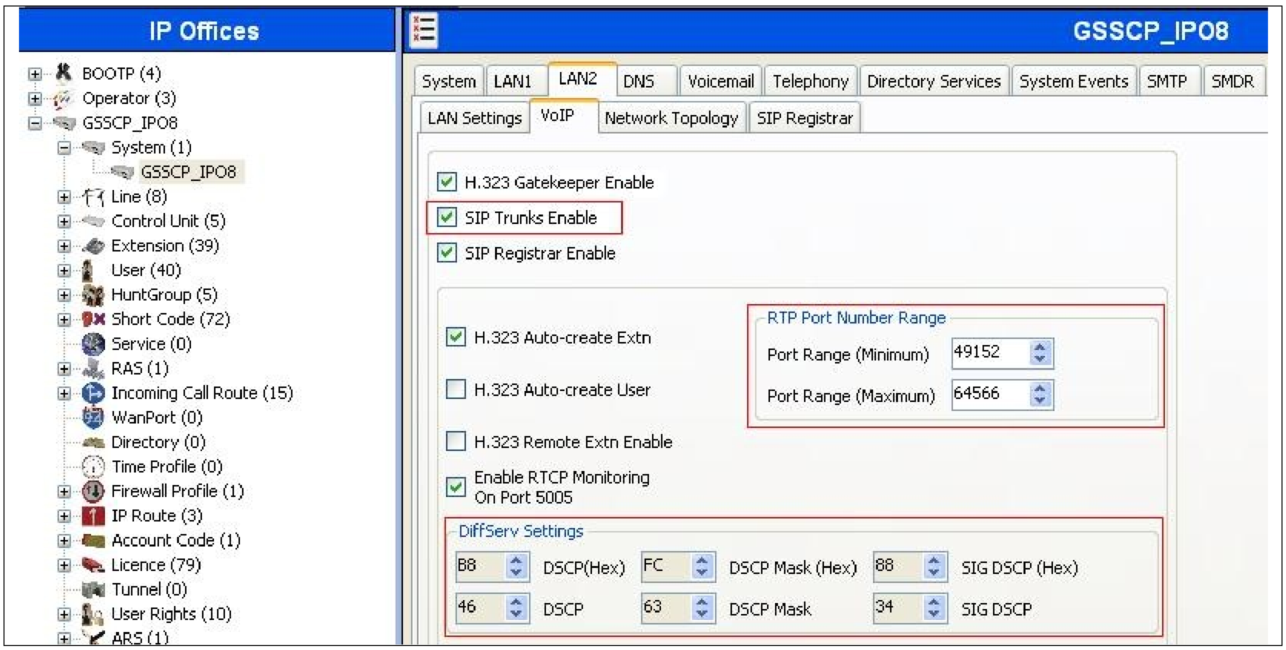


5.2. LAN2 Settings

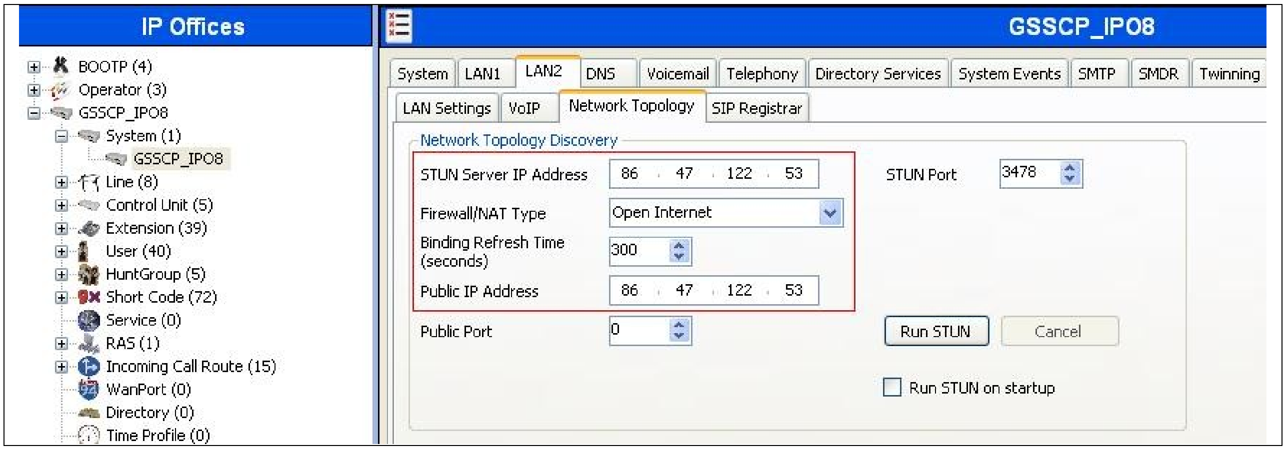
In the sample configuration, the **LAN2** port was used to connect Avaya IP Office to the external internet. To access the **LAN2** settings, first navigate to **System → GSSCP_IPO8** in the Navigation Pane where GSSCP_IPO8 is the name of the Avaya IP Office. Navigate to the **LAN2 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the public interface of Avaya 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, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both 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, enter the **Public IP Address** for Avaya IP Office. The same **Public IP Address** is used in the **STUN Server IP Address** field, even if not running STUN. It is important that the **Binding Refresh Time** is set to the correct value. 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, see **Section 5.9** for more details. Below is a sample configuration. On completion, click the **OK** button (not shown).

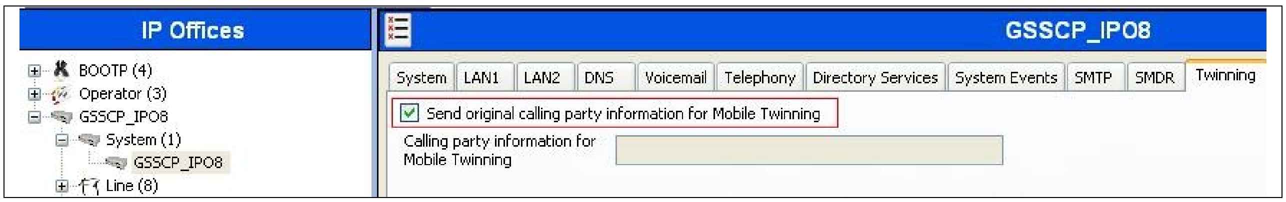


5.3. System Telephony Settings

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

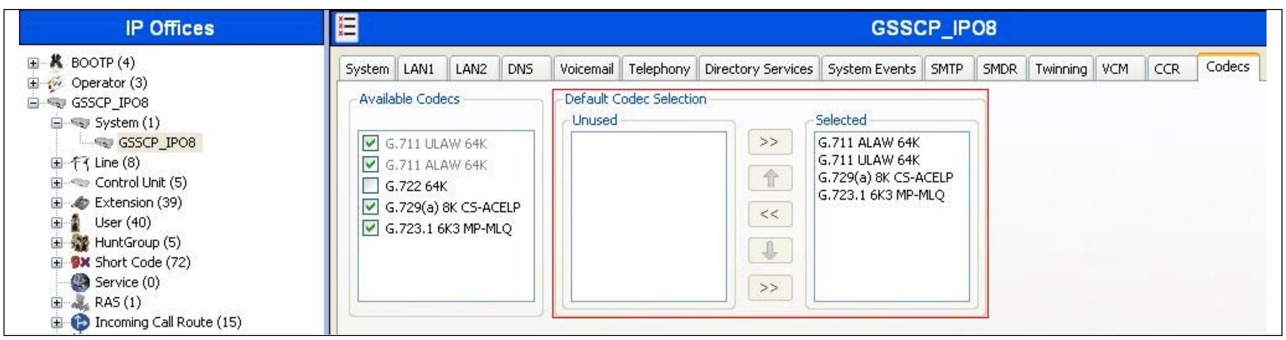
5.4. System Twinning Settings

Navigate to the **Twinning** tab, and check the box labeled **Send original calling party information for Mobile Twinning**. With this setting, Avaya IP Office will send the original calling party number to the twinned phone in the SIP From header (not the associated desk phone number) for calls that originate from an internal extension. For inbound PSTN calls to a twinned enabled phone, Avaya 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**). On completion, click the **OK** button (not shown).



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).



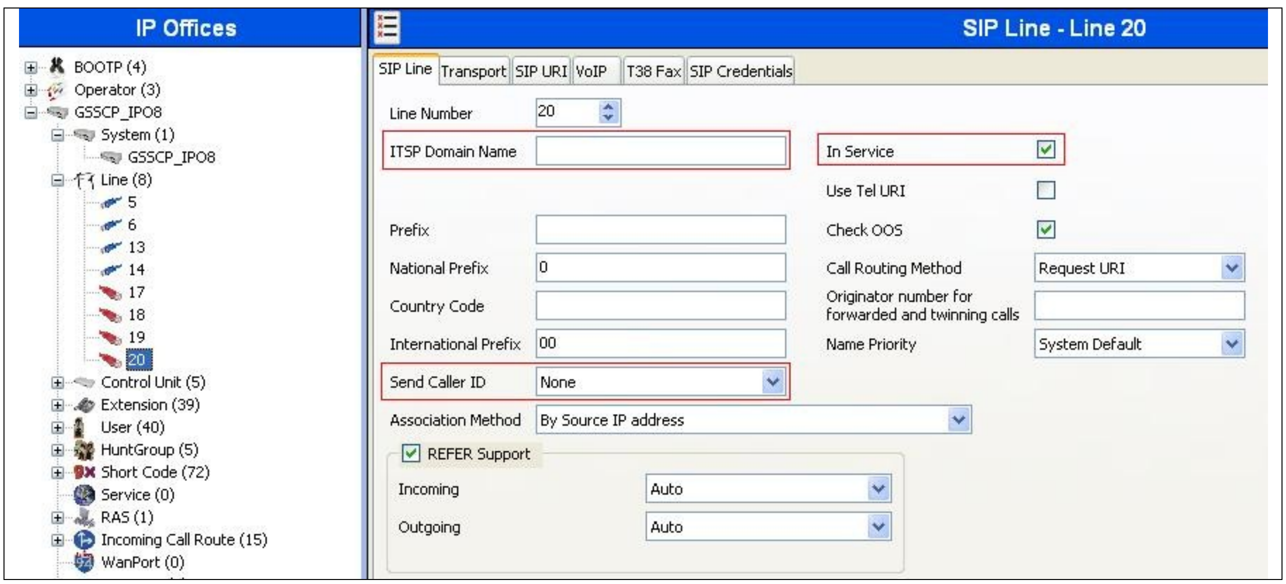
Note: During the test, all available codecs were selected to test the codec negotiation between Avaya IP Office and the Vodafone NL network.

5.6. Administer SIP Lines

SIP lines are needed to establish the SIP connections between Avaya 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 the Corporate Net over IP (CNoIP). To create a SIP line for VoV, 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.

- **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
- Default values may be used for all other parameters

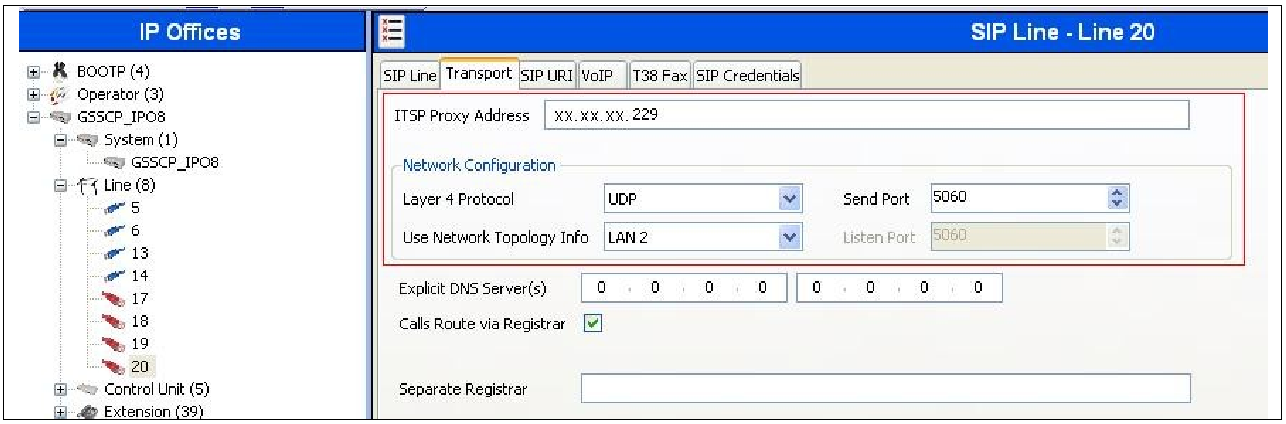
On completion, click the **OK** button (not shown).



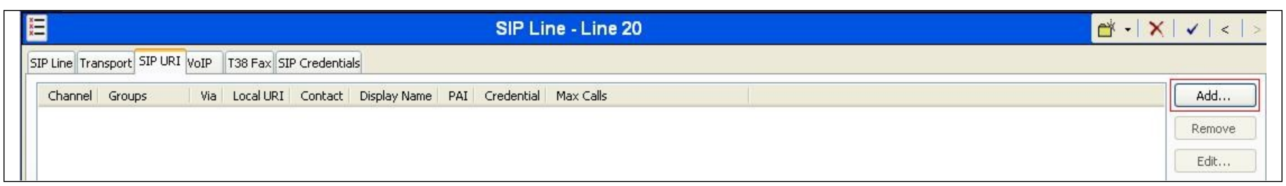
Select the **Transport** tab and set the following:

- 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**

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



After the SIP line parameters 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 for the VoV SIP line that matched any number assigned to an Avaya 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 **20** was defined that was associated to a single line (line 20)
- 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.

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**

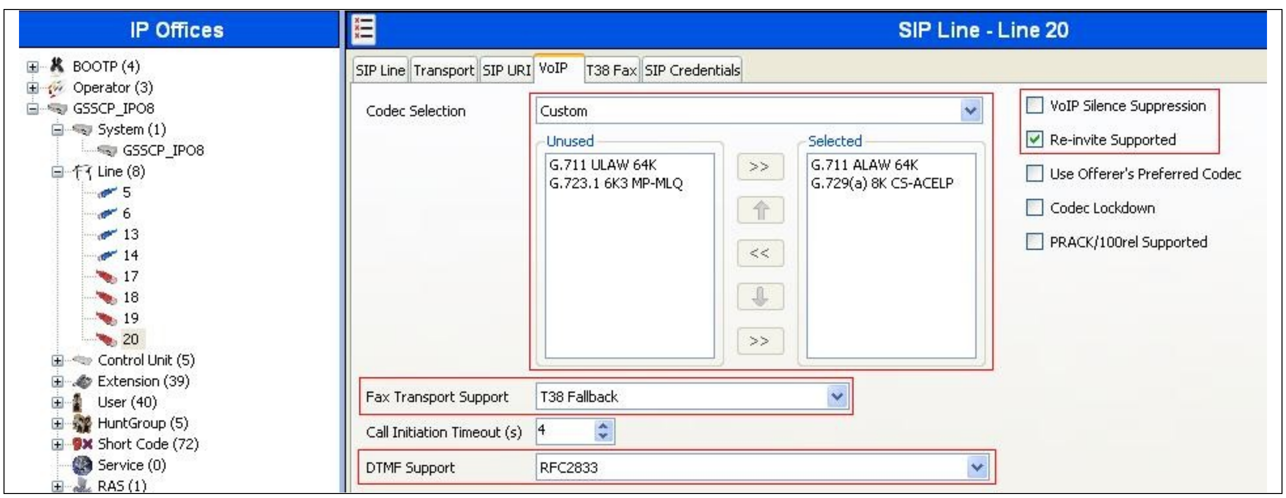
On completion, click the **OK** button (screenshots not shown).

Note: In the test, line 20 was used for VoV and line 19 was used for CNoIP.

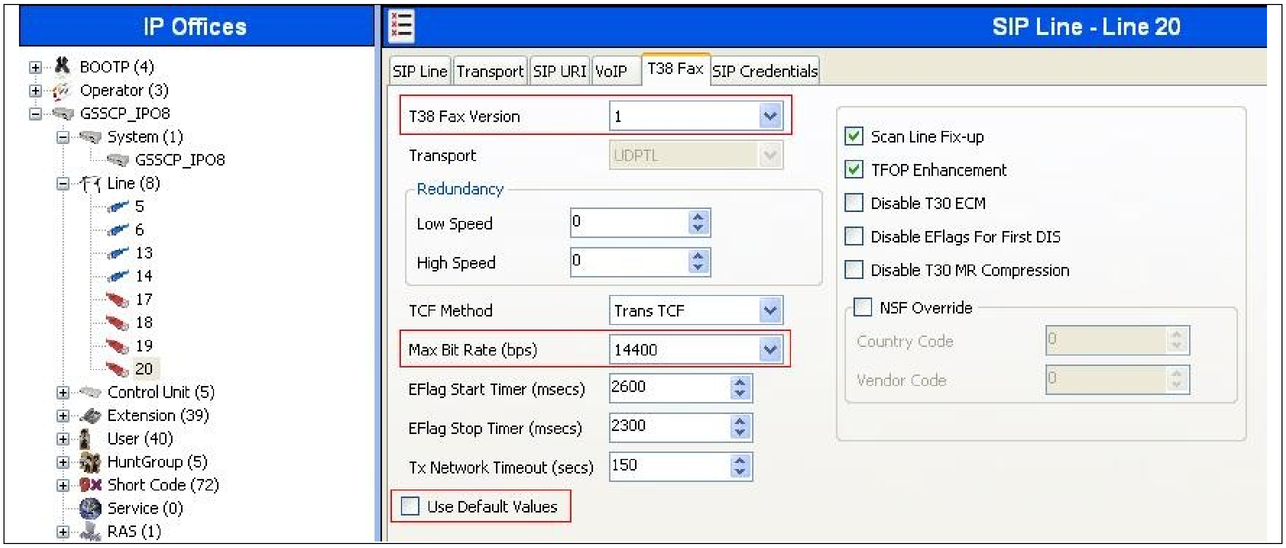
Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- 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 Avaya 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).



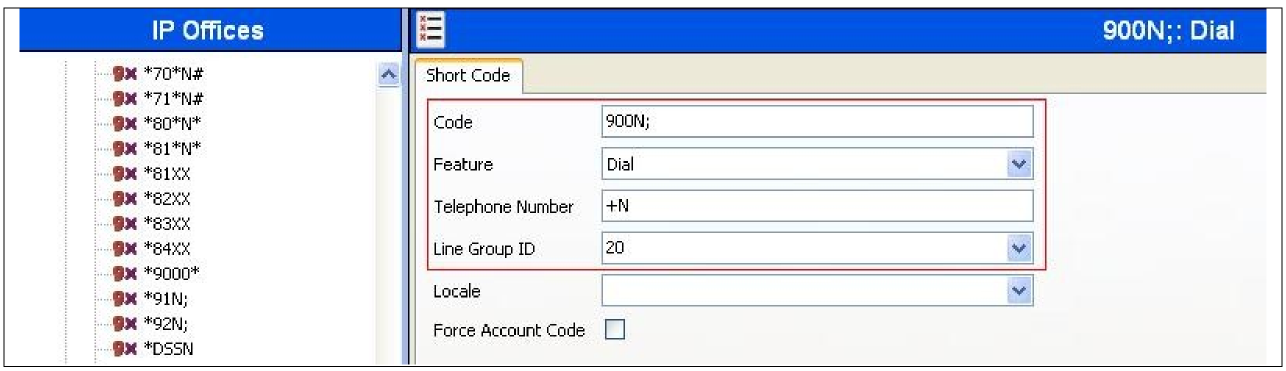
Note: It is advisable at this stage to save the configuration as described in **Section 5.11** to make the Line Group ID available in **Section 5.6.**

5.7. 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.

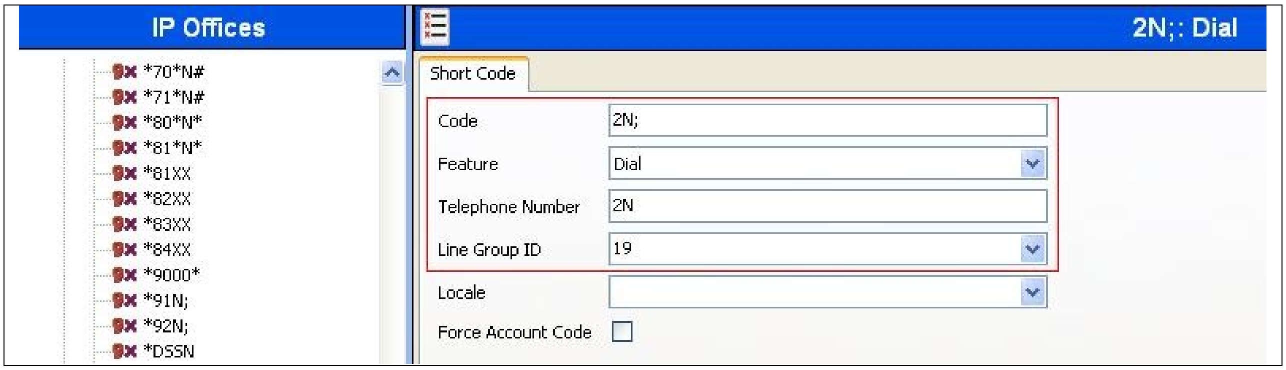
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon
- The example shows **900N**; which will be invoked when the user dials 9 followed by an international number
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- 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**

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



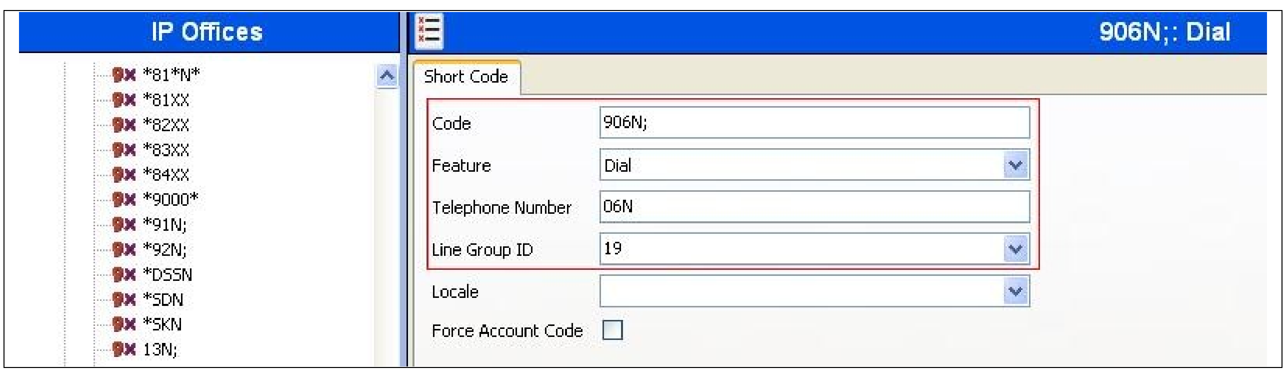
Short codes are also used for routing of national fixed and mobile calls, as well as VPN calls for CNoIP. National fixed line calls use the SIP line established for the Vodafone Office Voice (VoV) service. National mobile and VPN calls use the SIP line established for the Corporate Net over IP (CNoIP).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

- The example of a VPN call shows **2N;** which will be invoked when the user dials a four digit VPN number
- Set **Telephone Number** to **2N** 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**
- Set other parameters as shown in the previous example



An example for mobile calls is shown below.

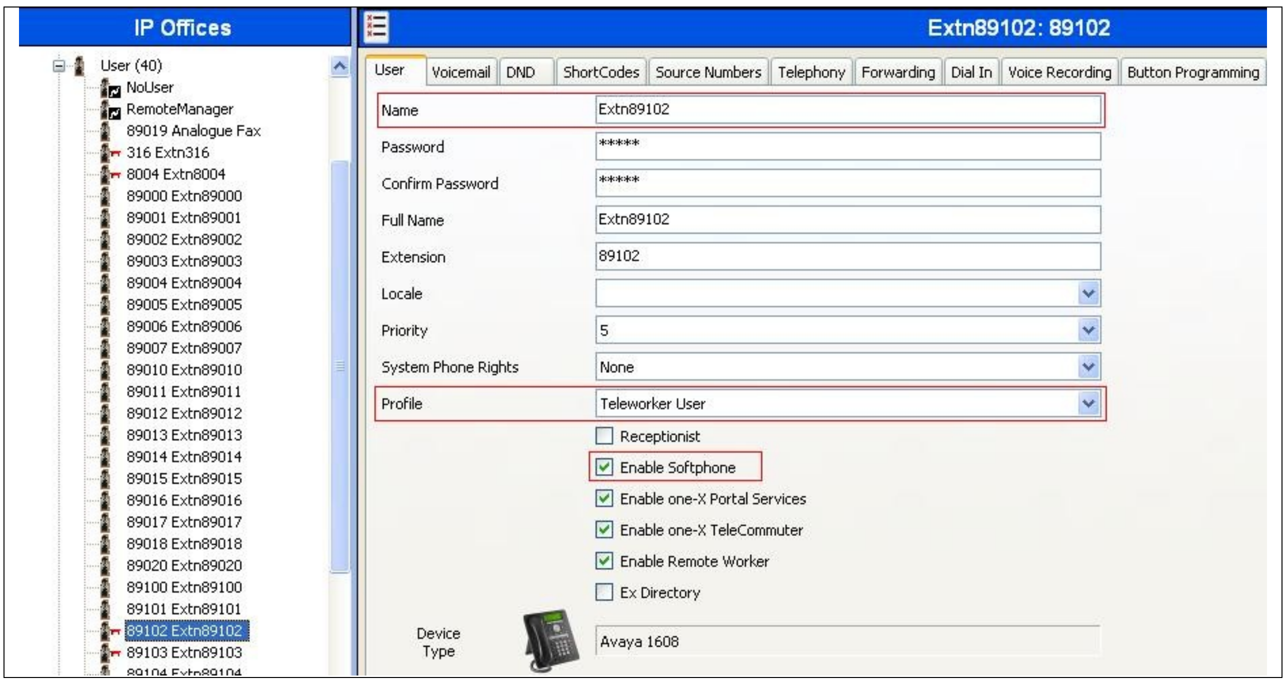
- The example of a mobile call shows **906N;** which will be invoked when the user dials **9** followed by a mobile number
- Set **Telephone Number** to **06N** which removes the digit **9**
- Set the **Line Group ID** to the outgoing line group number for CNoIP defined on the **SIP URI** tab on the **SIP Line** in **Section 5.6**
- 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, which replaced Phone Manager at Avaya IP Office R8.0.

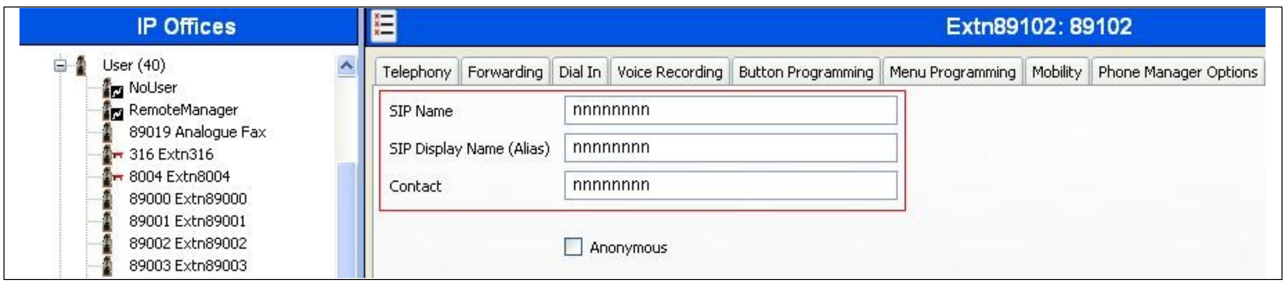
- Change the **Name** of the User if required, this will be used for login to the Avaya IP Office Softphone
- Select **Teleworker User** from the **Profile** drop down menu
- Check the **Enable Softphone** box



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).

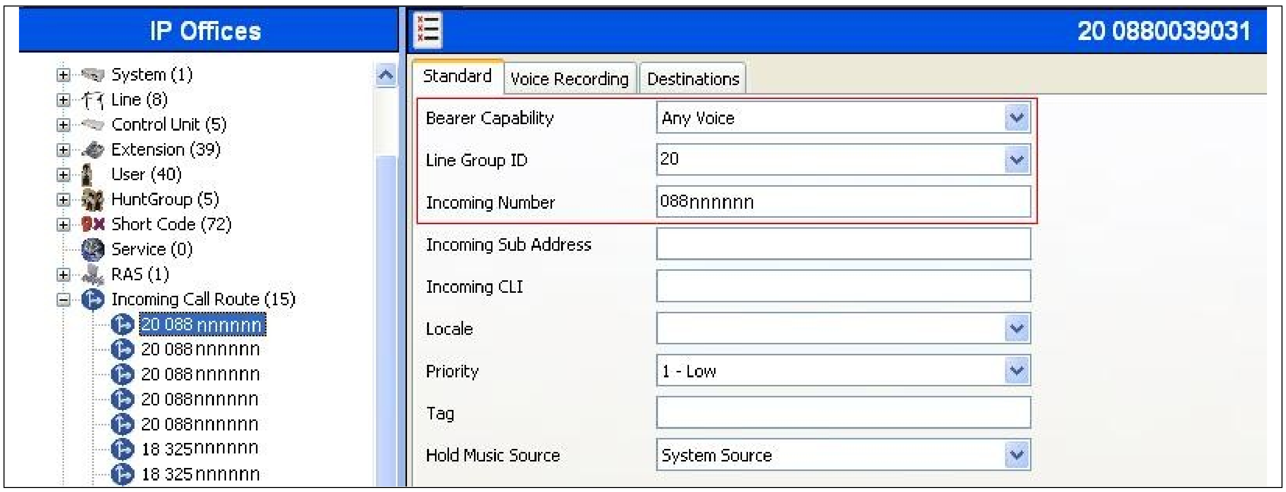
Note: The **Contact** field must be in E.164 format for the caller ID on the called phone to display properly.



5.9. Incoming Call Routing

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:

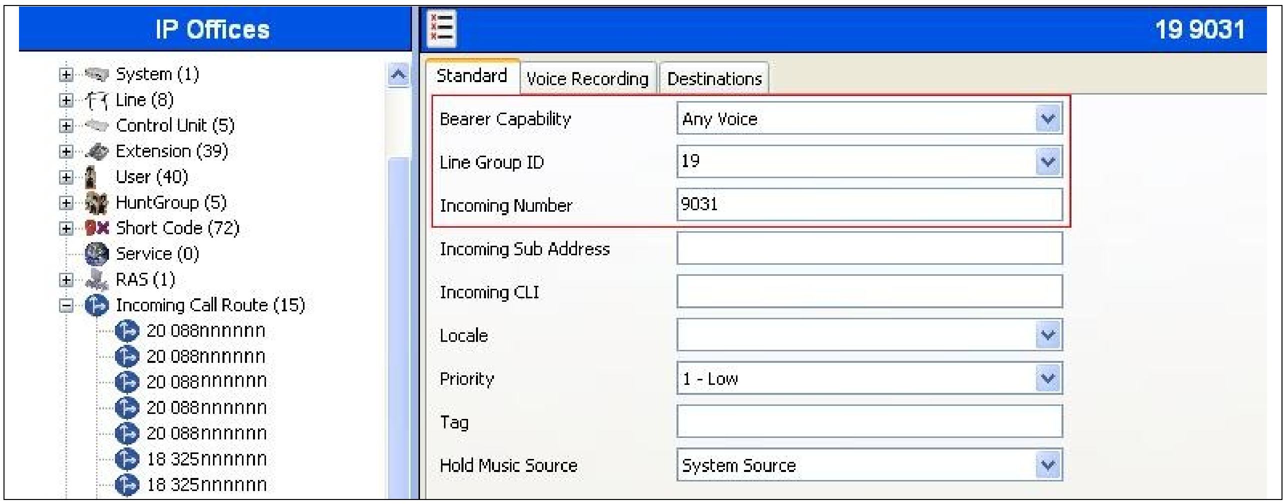
- Set the **Bearer Capacity** to **Any Voice**
- Set the **Line Group ID** to the incoming line group of the SIP line for VoV defined in **Section 5.6**
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left
- Default values can be used for all other fields



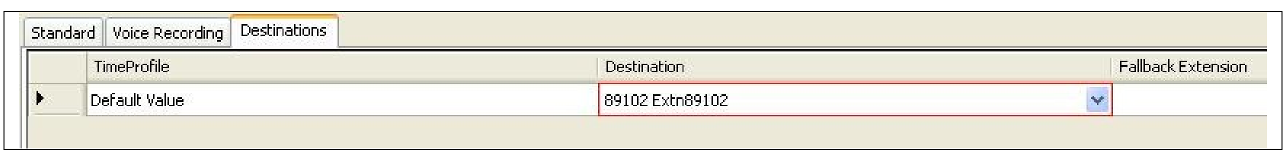
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 20 are routed to extension 89012.

To create an incoming call route for CNoIP, 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 CNoIP 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



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 VPN number on line 19 are routed to extension 89012.



Note: The above example shows how both VoV DDI numbers and CNoIP numbers can both be routed to the same extension.

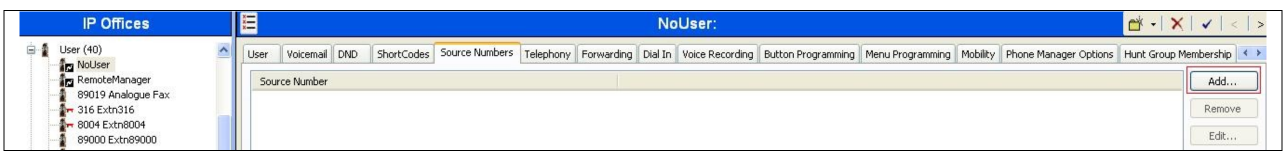
5.10. SIP Options

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 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:

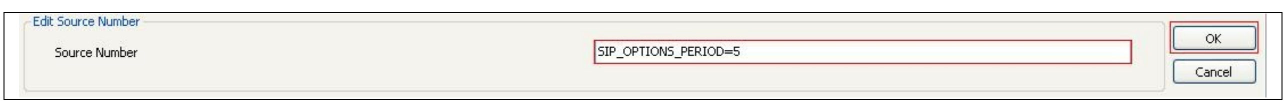
- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to the value required
- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be set to the value required

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

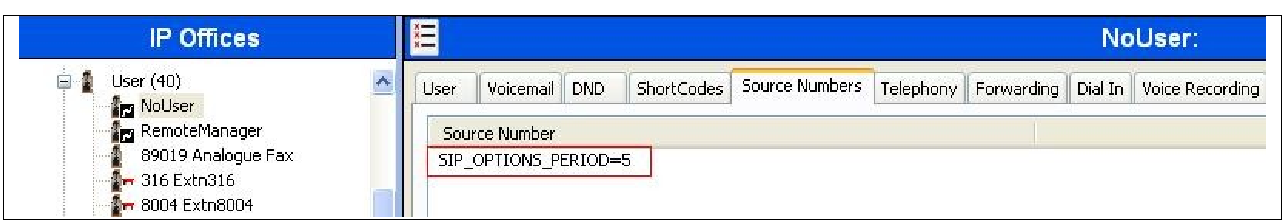
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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 2 minutes was desired. The **Binding Refresh Time** was set to **300** seconds (5 minutes) in **Section 5.2**. The **SIP_OPTIONS_PERIOD** was set to **5** minutes. Avaya IP Office chooses the OPTIONS period as the smaller of these two values. In the test, both these values were the same. Click the **OK** button (not shown).



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 the SIP Trunking service. The customer will need to provide the public IP address used to reach the Avaya IP Office at the enterprise. Vodafone NL will provide the customer the necessary information to configure the SIP connection to the SIP Trunking service including:

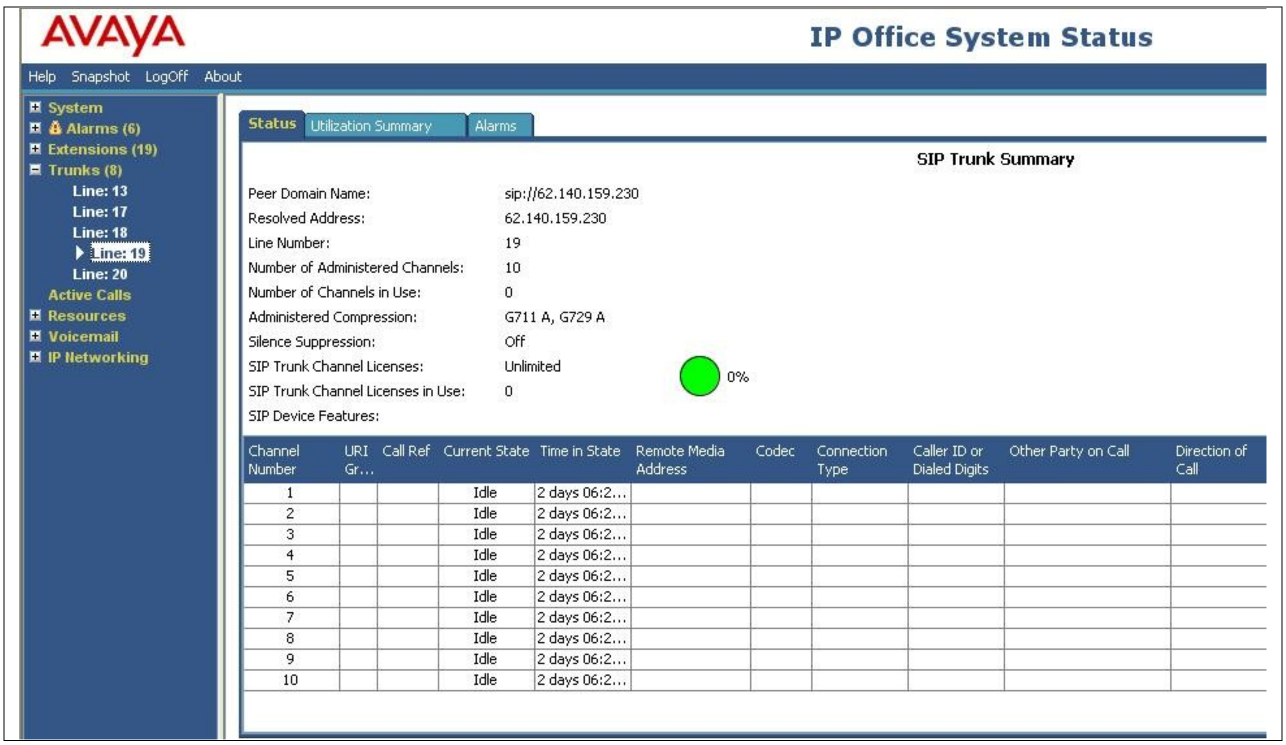
- IP address of SIP Trunking SIP proxy
- Network SIP Domain
- Supported codecs
- DDI numbers
- All IP addresses and port numbers used for signalling or media that will need access to the enterprise network through any security devices.

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 an html application found in **Program Files → Avaya → IP Office → System Status**. Launch **index.html.** From the left hand menu, expand Trunks and choose the SIP trunk (19 in this instance). Select the **Status** tab to show the status. The **Current State** column shows the state of each channel and the **Time in State** column shows the length of time that each channel has been in that state. Status should be idle if the Trunk is operational. IP address has been changed.



8. Conclusion

The Vodafone NL SIP Trunking service passed compliance testing. Interoperability testing of the sample configuration was completed with successful results for the Vodafone NL SIP Trunking Service. Transmission of incoming multi-page T.38 fax was found to be unreliable, though as this is not the preferred method of fax transmission for Vodafone Netherlands, it was not deemed to be critical for compliance testing. Refer to **Section 2.2** for test observations.

9. Additional References

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

[1] Avaya IP Office Knowledgebase 8 Documentation CD, 12th Dec 2011.

[2] Avaya IP Office Installation Manual, Document number15-601042, 20th Dec 2011.

[3] Avaya IP Office Manager Manual, Document number15-601011, 26th Jan 2012.

[4] System Status Application, Document number15-601758, 12th Nov 2011.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.