Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 8.1 with Wind Telecom SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider Wind Telecom and Avaya IP Office 8.1.

During the interoperability testing, Avaya IP Office was able to interoperate with the Wind Telecom Softswitch via SIP trunking. This test was performed to verify SIP trunk features including basic call, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls were placed in both directions using various endpoint types.

Wind Telecom 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.

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1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider Wind Telecom and an Avaya IP Office solution.

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 8.1, Avaya Voicemail Pro, Avaya IP Office soft clients and Avaya desk phones, including SIP, H.323, digital, and analog endpoints.

The Wind Telecom SIP Trunk Service referenced within these Application Notes is designed for business customers in the Dominican Republic. Customers using this service with the Avaya IP Office solution are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI trunks. This approach generally results in lower cost for the enterprise.

The Avaya IP Office system can be configured to authenticate with the SIP service provider using either SIP Trunk Registration or Static IP Authentication. These Application Notes cover the configuration of IP Office using Static IP Authentication with the service provider.

2. General Test Approach and Test Results

A simulated enterprise site was configured in the test lab using Avaya IP Office, connected to the Wind Telecom SIP Trunk Service via a SIP trunk over the public Internet. This scenario may differ from a real customer environment, in which a dedicated private network connection could be provided by Wind Telecom to the actual customer site.

The 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.

Testing was performed using IP Office 500v2 R8.1, but results also apply to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R8.1 to support analog or digital endpoints and trunks.

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various phone types. Phone types included SIP, 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 SIP, H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from soft clients.
- Various call types including: local, long distance, international, outbound toll-free, etc.
- Codecs G729A and G.711MU.
- G.711 Fax.
- 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 with Wind Telecom was completed with successful results with the exception of the observations/limitations described below:

- **OPTIONS** - Wind Telecom was not configured to send OPTIONS messages to the IP Office during the compliance test. IP Office still sent periodic OPTIONS messages to monitor the status of the SIP trunk, to which Wind Telecom responded sending back "200 OK" messages.
- **SIP REFER -** On PSTN calls to or from the IP Office that are transferred back to the PSTN on the SIP trunk, Wind Telecom responds with a "202 Accepted" to the REFER message sent from the IP Office, but the call between the two PSTN endpoints drops. REFER needs to be disabled in the IP Office for the Call Transfer to complete, otherwise the call transfer fails. The implication is that IP Office is not released after the call transfer is completed, and 2 trunks remain busy for the duration of the call.
- **T.38 Fax** - On inbound calls from the PSTN, the SDP on the initial INVITE from Wind Telecom contains simultaneous voice codec offers (G729A, G711U) and fax (T.38). With **Fax Transport Support** set to **T.38** or **T.38 Fallback** on the **SIP Line/VoIP** tab, IP Office responds to the INVITE by connecting at the preferred voice codec (G729A), but ignoring the DTMF Payload Type 101 offered in the SDP, effectively switching to Inband DTMF support. This creates problems with DTMF recognition on inbound voice (non-fax) calls. As a workaround, T.38 fax was disabled in the IP Office, and G.711 fax was used instead. Fax calls with **Fax Transport Support** set to **G.711** were successfully completed during the compliance test. JIRA IPOFFICE-38671 was created with IP Office Development for investigation.

2.3. Support

For technical support on the Wind Telecom SIP Trunk Service offer, visit
www.windtelecom.com.do

3. Reference Configuration

Figure 1 below illustrates the test configuration. It shows the enterprise site connected to the
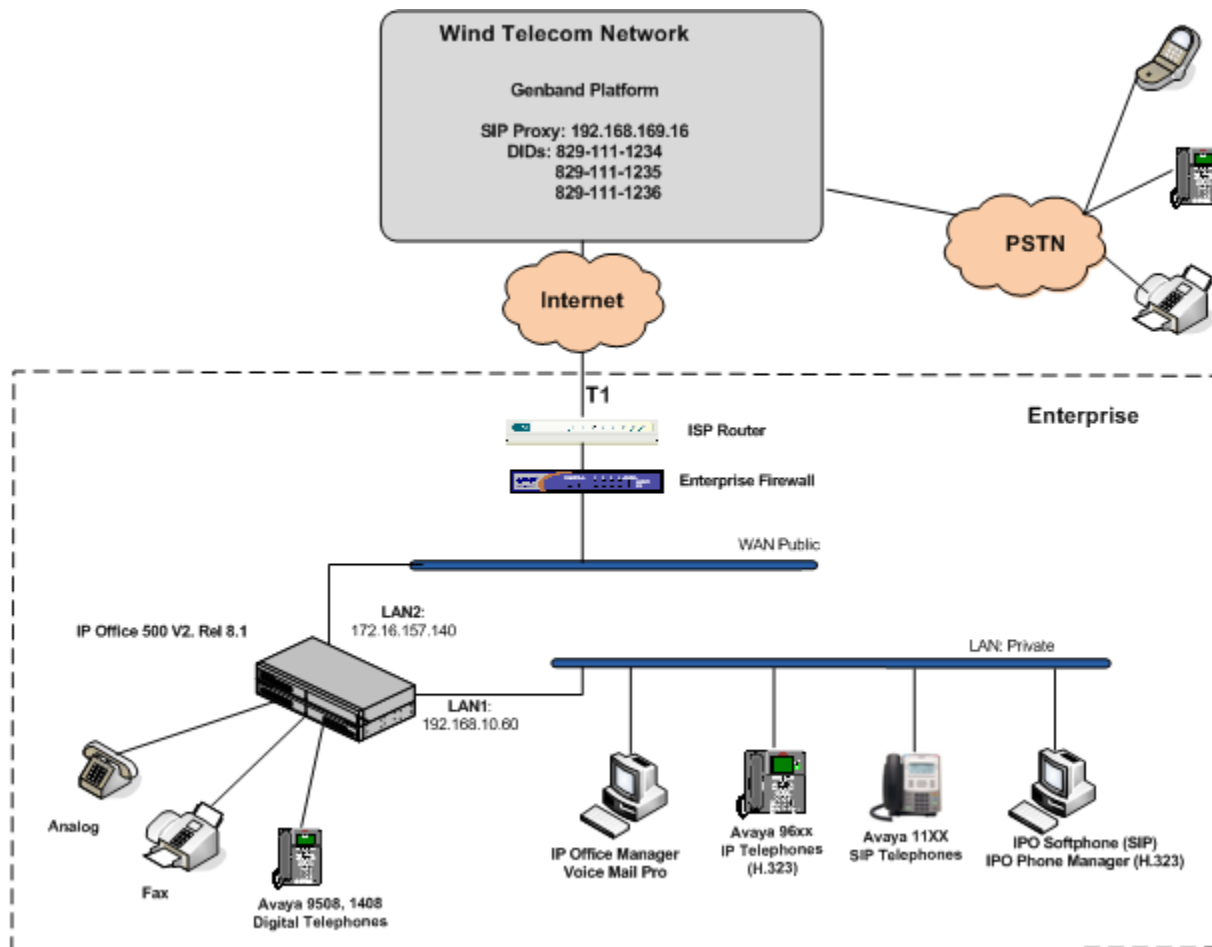Wind Telecom SIP Trunk Service through the public IP network.

Figure 1: Test Configuration

Note that for security purposes, all public IP addresses and phone numbers shown throughout
these Application Notes have been edited so that the actual IP addresses of the network elements
and public PSTN numbers are not revealed.

The enterprise site contains the Avaya IP Office 500v2 with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. The LAN1 port of Avaya IP Office is connected to the enterprise LAN while the LAN2 port is connected to the public IP network. Endpoints include Avaya 9600 Series IP Telephones (with H.323 firmware), Avaya 1140E IP Telephones (with SIP firmware), Avaya 1408 and 9508D Digital Telephones, analog telephones, Officejet 4500 fax machines and PCs running Avaya IP Office Softphone and Avaya IP Office Phone Manager as SIP and H.323 soft clients, respectively. The site also has a Windows XP PC running Avaya IP Office Manager to configure and administer the Avaya IP Office system, and Avaya Voicemail Pro providing voice messaging service to the Avaya IP Office users. Mobile Twinning is configured for some of the Avaya IP Office users so that calls to these users' extensions will also ring and can be answered at the configured mobile phones.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to Wind Telecom. The short code 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to the network. Since the Dominican Republic is a country member of the North American Numbering Plan (NANP), the users dialed 10 digits for local calls, including the area code, and 11 (1 + 10) digits for other calls between the NANP.

In an actual customer configuration, the enterprise site may include additional network components between the service provider and the Avaya IP Office system, 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 all SIP and RTP traffic between the service provider and the Avaya IP Office system must be allowed to pass through these devices.

4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya IP Office 500v2	8.1 (56)
Avaya IP Office Digital Expansion Module DCPx16	10.1 (56)
Avaya IP Office Manager	10.1 (56)
Avaya IP Office Voicemail Pro	8.1.2004.0
Avaya 9620 IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.105S
Avaya 9640 IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.105S
Avaya 1140E IP Telephone (SIP)	04.03.12.00
Avaya Digital Telephone 1408	16
Avaya Digital Phone 9508	0.44
Avaya IP Office Softphone	3.2.3.48.67009
Avaya IP Office Phone Manager	4.2.42
Wind Telecom SIP Trunk Service	
Genband Softswitch	C20 CVM 14

5. Configure IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the Wind Telecom SIP Trunk Service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration** (not shown), 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 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 Avaya IP Office configuration.

Standard feature configurations that are not directly related to the interfacing with the service provider (such as LAN interface to the enterprise site, Twinning and IP Office Softphone support) are assumed to be already in place, and they are not part of these Application Notes.

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5.1. Licensing

The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm that there is a valid license with sufficient "Instances" (trunk channels) in the Details pane. Note that the actual License Key in the screen below was edited for security purposes.

5.2. LAN2 Settings

In the sample configuration, *IP500V2 Main* was used as the system name, and the WAN port was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office. To access the LAN2 settings, first navigate to **System (1)** → **IP500V2 Main** in the Navigation and Group panes and then navigate to the **LAN2** → **LAN Settings** tab in the Details pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.

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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 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.

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

- Select the **Firewall/NAT Type** from the pull-down menu to the option that matches the network configuration. Since no network address translation (NAT) was used in the compliance test, the parameter was set to *Open Internet*. With this configuration, the **STUN Server IP Address** and **STUN Port** are not used.
- Set **Binding Refresh Time (seconds)** to *180.* This value is used to determine the frequency at which Avaya IP Office will send SIP OPTION messages to the service provider.
- Set **Public IP Address** to the IP address that was set for LAN2.
- Set **Public Port** to *5060*.
- All other parameters should be set according to customer requirements.

In the compliance test, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with the Wind Telecom SIP Trunk Service, and therefore is not described in these Application Notes.

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5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** Tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. *U-Law* was used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the SIP trunk to the service provider.

5.4. Twinning Calling Party Settings

Navigate to the **Twinning** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be controlled by the setting on the SIP Line (**Section 5.6**). This setting also impacts the Caller ID for call forwarding.

5.5. IP Route

Create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets, in order to reach the subnet where the SIP proxy is located on the Wind Telecom network. On the left navigation pane, right-click on **IP Route** and select **New**.

- Set the **IP Address** and **IP Mask** of the remote Wind Telecom SIP Proxy subnet.
- Set **Gateway IP Address** to the IP Address of the router used to reach the external network. For the test configuration, this was the IP address of the local ISP router.
- Set **Destination** to *LAN2* from the pull-down menu.

5.6. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Wind Telecom SIP Trunk Service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New→ SIP Line**.

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5.6.1. SIP Line Tab

On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set the **ITSP Domain Name** to the IP address of the IP Office LAN2 interface. IP Office will use this address as the host portion of the SIP URI in the From headers of messages sent to the network.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Set **Call Routing Method** to *Request URI*.
- Set **Send Caller ID** to *Remote Party ID.*
- Uncheck the **REFER support** box. IP Office will not send REFER headers for calls that are transferred back to the PSTN. See **Section 2.2** for more information.
- Set **UPDATE Supported** to *Auto*.
- Default values may be used for all other parameters.

5.6.2. Transport Tab

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

- Set the **ITSP Proxy Address** to the IP address of the Wind Telecom proxy server.
- Set the **Layer 4 Protocol** to *UDP*.
- Set **Use Network Topology Info** to *LAN2* as configured in **Section 5.2**.
- Set the **Send Port** to *5060*.
- Default values may be used for all other parameters.

5.6.3. SIP URI Tab

A SIP URI entry needs to be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to *Use Internal Data*. This setting allows calls on this line whose SIP URI match the number set in the **SIP** tab of any User as shown in **Section 5.8**.

- 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 only contains this line (line 17).

- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

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5.6.4. VoIP Tab

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

- In the sample configuration, the **Codec Selection** was configured using the *Custom* option, allowing an explicit ordered list of codecs to be specified. The buttons allow setting the specific order of preference for the codecs to be used on the line, as shown.
- Set **Fax Transport Support** to *G.711*. This is necessary as a workaround to the problem observed with the DTMF Payload Type when using *T.38* or *T.38 Fallback* modes, as described in **Section 2.2**.
- Set the **DTMF Support f**ield to *RFC2833.* This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check the **PRACK/100rel Supported** box, to advertise the support for provisional responses and Early Media to Wind Telecom.
- Default values may be used for all other parameters.

5.7. Fax Extensions

During the compliance test, it was observed that in order for outbound G.711 fax calls to complete successfully, analog extensions used as fax machines in the IP Office needed to be classified as such. By doing this, IP Office will offer codec G.711U directly on outbound fax calls, avoiding the initial voice setup using G.729A.

Select **Extensions** on the left Navigation pane. On the center Group pane, select the extension to be used as a fax machine. On the Details pane, select the **Analogue** tab. On the **Equipment Classification** section, check the **FAX Machine** option. The screen below shows the configuration for extension 1503, used as a fax machine in the sample configuration.

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5.8. Users

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, first navigate to **User** in the left Navigation Pane, and then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is *Extn 1102dcp*. 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 and Contact headers for outgoing SIP trunk calls. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.6**). The example below shows the settings for user "Extn1102dcp". The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by Wind Telecom. In the example, the DID number *8291111234* was used. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.

5.9. Incoming Call Route

An incoming call route maps inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc, within the IP Office system. Incoming call routes should be defined for each DID number assigned by the service provider.

In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any predefined extension in IP Office. The routing decision for the call is based on the parameters previously configured for **Call Routing Method** and **SIP URI (Section 5.6)** and the users **SIP Name** and **Contact**, already populated with the assigned Wind Telecom DID numbers **(Section 5.8)**

From the left Navigation Pane, right-click on **Incoming Call Route** and select **New.**
On the Details Pane, under the **Standard** tab, set the parameters as show bellow:
- Set **Bearer Capacity** to *Any Voice*.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6**.
- Default values may be used for all other parameters.

- Under the **Destinations** tab, enter "**.**" for the **Default Value**. This setting will allow the call to be routed to any destination with a value on its **SIP Name** field, entered on the **SIP** tab of that **User,** which matches the number present on the user part of the incoming Request URI.

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5.10. Short Code

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.
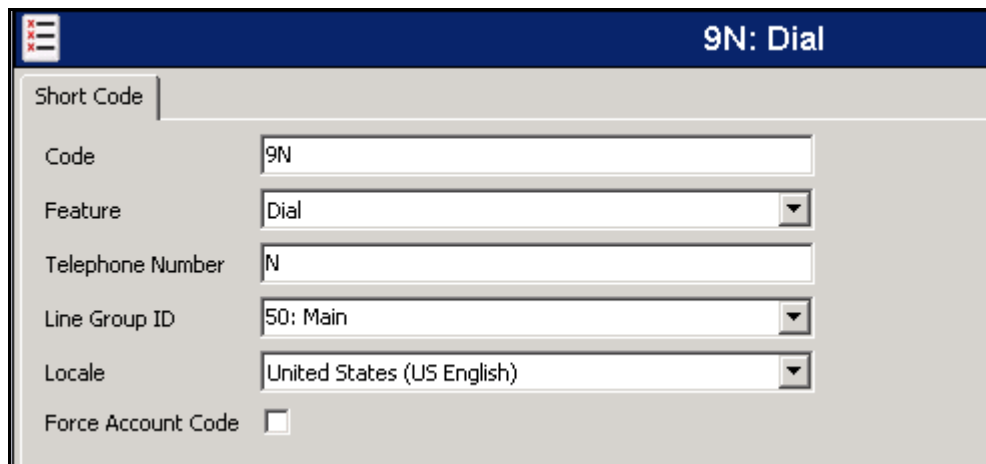
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, *8N;*. This short code will be invoked when the user dials 8 followed by any number.
- Set **Feature** to *Dial*. This is the action that the short code will perform.
- Set **Telephone Number** to *N"@192.168.169.16"*. 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. The IP address 192.168.169.16 represents the IP address of the Wind Telecom SIP proxy.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.6.3**. This short code will use this line group when placing outbound calls.
- Default values may be used for all other parameters.

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5.11. Automatic Route Selection

Optionally, Automatic Route Selection (ARS) can be used rather than the simple short code approach described above. With ARS, secondary dial tone can be provided after the access code. Other features like time-based routing criteria and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. ARS also facilitates a more granular treatment for different types of calls, and permits a more specific matching of the telephone number dialed following the access code. While detailed coverage of ARS is beyond the scope of these Application Notes, and alternate routing was not used in the reference configuration, this section includes some basic screen illustrations of the ARS settings used during the compliance test.

To create a short code to be used for ARS, right-click on **Short Code** in the Navigation Pane and select **New**. The screen below shows the short code **9N** created. Note that the semi-colon is not used here. In this case, when the Avaya IP Office user dials 9 plus any number **N**, instead of being directed to a specific Line Group ID, the call is directed to **Line Group *50: Main***, which is configurable via ARS.

The following screen shows the example ARS configuration for the route *Main*. Note the sequence of *X*s used in the **Code** column of the entries to specify the exact number of digits to be expected, following the access code and the first digit on the string. This type of setting results in a much quicker response in the delivery of the call by the IP Office. The example below shows that for local calls, the user dialed 9, then 10 digit numbers starting with an 8. For calls to other area codes in the North American Numbering Plan, the user dialed 9, followed by 11 digits, starting with a 1.

5.12. Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "restricted" and "anonymous" respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. By default, Avaya IP Office will use PPI for privacy. For the compliance test, PAI was used for the purposes of privacy.

To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.
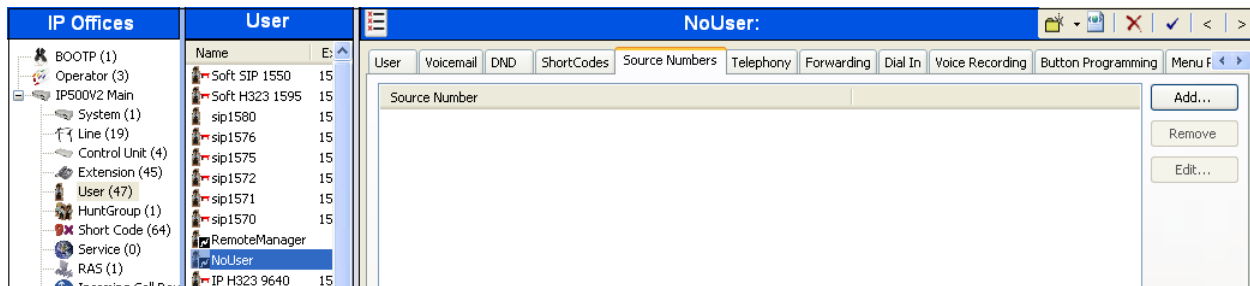
At the bottom of the Details Pane, the **Source Number** field will appear. Enter *SIP_USE_PAI_FOR_PRIVACY*. Click **OK**.
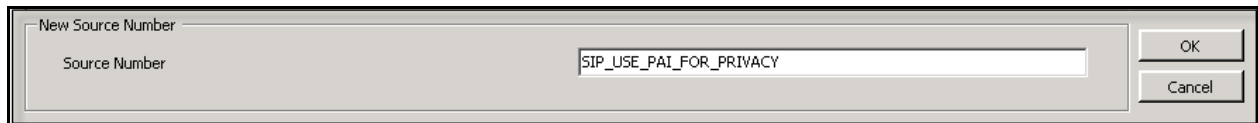
The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.
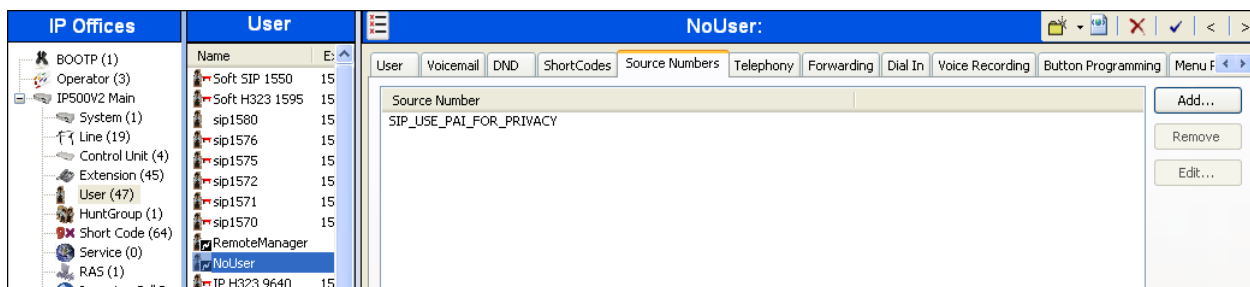
5.13. Save Configuration

Navigate to **File → Save Configuration** in the menu bar at the top left of the screen to save the configuration performed in the preceding sections (not shown).

6. Wind Telecom SIP Trunking Configuration

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

- IP address of the Wind Telecom SIP Proxy server.
- Supported codecs and order of preference.
- 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

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start** → **Programs** → **IP Office** → **System Status** on the PC where Avaya IP Office Manager was installed. Log in using the appropriate credentials and 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).

- **Select** the **Alarms** tab and verify that no alarms are active on the SIP line.
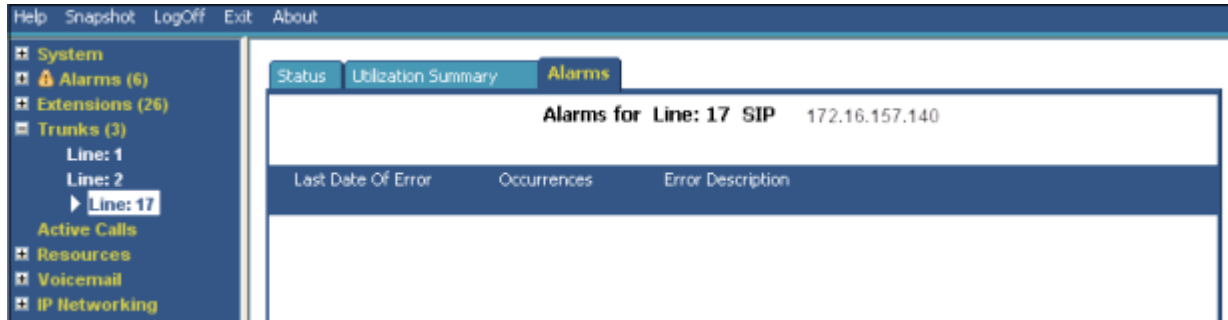
- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Verify that a phone connected to the PSTN can successfully place a call to the Avaya IP Office system with two-way audio.
- The Avaya IP Office Monitor application can also be used to monitor and troubleshoot SIP signaling messaging between Wind Telecom and the IP Office. Launch the application from **Start → Programs → IP Office → Monitor** on the PC where Avaya IP Office Manager was installed. The sample screen below shows part of the messages on an outbound call.

8. Conclusion

These Application Notes describe the procedures required to configure SIP trunk connectivity between Avaya IP Office 8.1 and the Wind Telecom SIP Trunk Service, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with the exception of the observations/limitations described in **Section 2.2.**

9. Additional References

[1] IP Office 8.1, IP500/IP500 V2 Installation Document Number 15-601042, November 2012, https://downloads.avaya.com/css/P8/documents/100162521

[2] IP Office Manager 10.1, Document Number 15-601011, November 2012, https://downloads.avaya.com/css/P8/documents/100162522

[3] IP Office Release 8.1 Implementing Voicemail Pro, Document Number 15-601064, December 2012, https://downloads.avaya.com/css/P8/documents/100163127

[4] IP Office System Status Application, Document Number 15-601758, November 2012 https://downloads.avaya.com/css/P8/documents/100150298

[5] Avaya IP Office Knowledgebase, http://marketingtools.avaya.com/knowledgebase

Product documentation for Avaya products may be found at http://support.avaya.com. Product documentation for the Wind Telecom SIP Trunk Service is available from Wind Telecom.

Appendix: SIP Line Template

Avaya IP Office Release 8.1 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** 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 Avaya IP Office Manager is installed to add a new *TemplateProvisioning* registry entry. Select **Start → Run**. Enter *regedit* in the **Open** box.
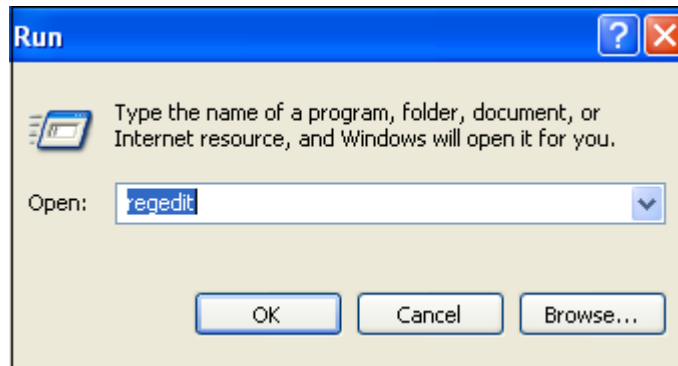
On the Registry Editor, navigate to **HKEY_CURRENT_USER** → **Software** → **Avaya** →**IP400**. Right click on **Manager** and select **New** → **DWORD Value**.
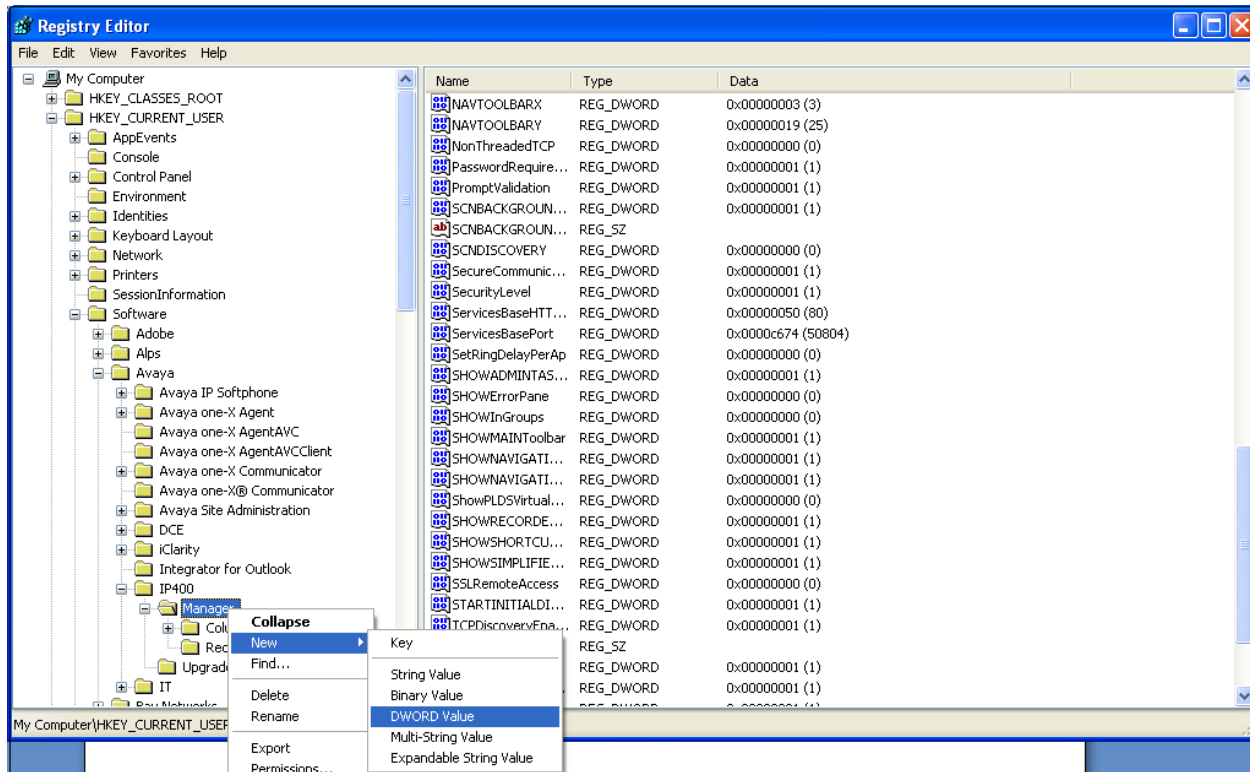
 Right click the newly created entry and rename it to *TemplateProvisioning*. Double click the entry and change the value under **Value Data** from "*0*" to "*1*". Restart the PC.
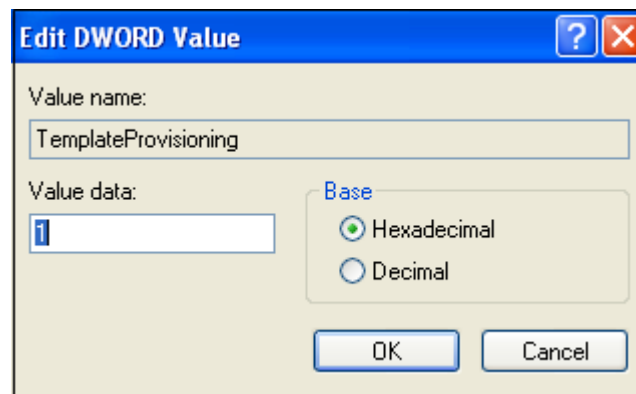
To enable template support in the IP Office Manager, select **File**, then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.
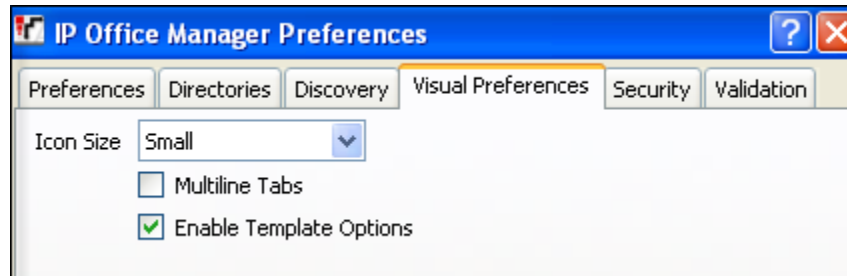
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click the Sip Line (**17**), and select **Generate SIP Trunk Template**.
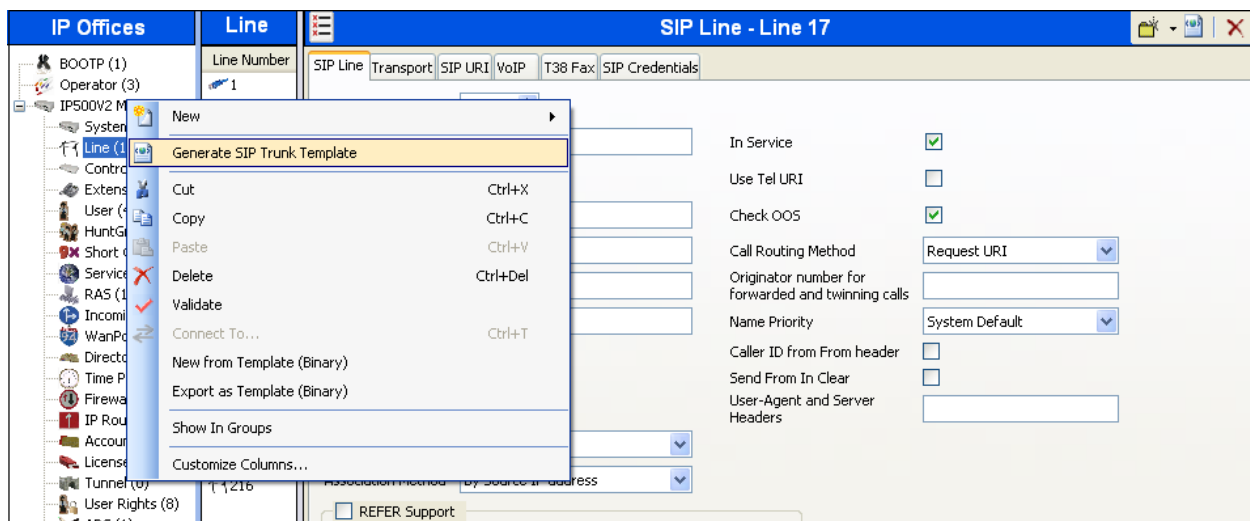
The trunk's settings are displayed as configured in **Section 5.6.** Enter a descriptive name for the template, adjust the settings if required, and then click on **Export**.

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 **DO_Wind Telecom_SIPTrunk.xml**:

```
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
<TemplateType>SIPTrunk</TemplateType>
<Version>20130107</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>Wind Telecom IPO8.1</DescriptiveName>
<ITSPDomainName>172.16.157.140</ITSPDomainName>
<SendCallerID>CallerIDRPID</SendCallerID>
<ReferSupport>false</ReferSupport>
```

```
<ReferSupportIncoming>1</ReferSupportIncoming>
<ReferSupportOutgoing>1</ReferSupportOutgoing>
<RegistrationRequired>false</RegistrationRequired>
<UseTelURI>false</UseTelURI>
<CheckOOS>true</CheckOOS>
<CallRoutingMethod>1</CallRoutingMethod>
<OriginatorNumber />
<AssociationMethod>SourceIP</AssociationMethod>
<LineNamePriority>SystemDefault</LineNamePriority>
<UpdateSupport>UpdateAuto</UpdateSupport>
<UserAgentServerHeader />
<CallerIDfromFromheader>false</CallerIDfromFromheader>
<PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
<ITSPProxy>192.168.169.16</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.729(a) 8K CS-ACELP,G.711 ULAW 64K</AdvCodecPref>
<CallInitiationTimeout>4</CallInitiationTimeout>
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<VoipSilenceSupression>false</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_G711</FaxTransportSupport>
<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown>
<Rel100Supported>true</Rel100Supported>
<T38FaxVersion>3</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>true</UseDefaultValues>
<ScanLineFixup>true</ScanLineFixup>
<TFOPEnhancement>true</TFOPEnhancement>
<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**:
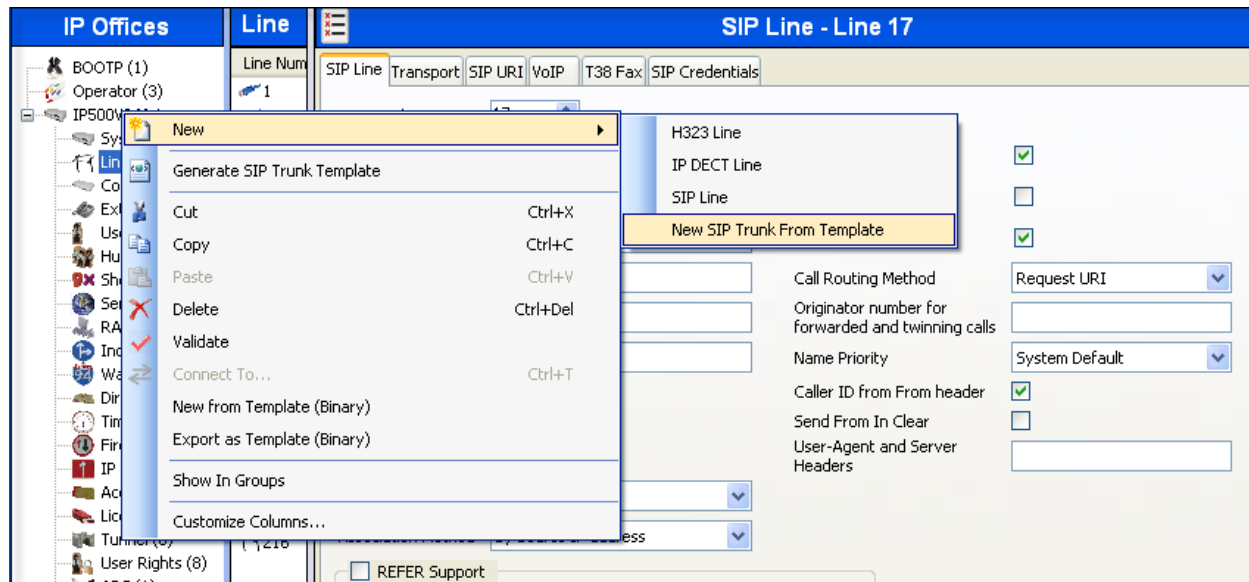
On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.

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