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Application Notes for Configuring Avaya IP Office 7.0 with AT&T Mobility SIP Trunk Services in Puerto Rico - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider AT&T Mobility in Puerto Rico and Avaya IP Office 7.0.

AT&T Mobility SIP Trunk Services in Puerto Rico provides PSTN access via a SIP trunk between the enterprise and the AT&T network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

AT&T Mobility in Puerto Rico 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.

MA; Reviewed:
SPOC 8/24/2011

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1 of 27
ATTPR-SipTrIPO7

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider AT&T Mobility in Puerto Rico and an Avaya IP Office solution.

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 7.0, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

The AT&T Mobility SIP Trunk Service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

During the next pages and for brevity in these Application Notes, the service provider's name "AT&T Mobility in Puerto Rico" will be abbreviated and referred as "AT&T Mobility" or just as "AT&T".

2. General Test Approach and Test Results

The test approach was to configure a simulated enterprise site in the Test Lab, using Avaya IP Office, connecting to the AT&T SIP Trunk Service in Puerto Rico, by means of a SIP trunk over the Internet. This scenario may differ from a real customer environment in Puerto Rico, in which a private network connection could be provided by AT&T to the customer site.

The configuration shown in **Figure 1** was used to exercise the features and functionality tests listed in **Section 2.1**.

AT&T Mobility SIP Trunk Services in Puerto Rico passed compliance testing.

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 H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.

MA; Reviewed:
SPOC 8/24/2011
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2 of 27
ATTPR-SipTrIPO7

- Inbound and outbound PSTN calls to/from soft clients. Avaya IP Office Softphone was used in the tests as a SIP client, and Avaya IP Office Phone Manager as the soft client for H.323
- Various call types including: local, long distance, international, outbound toll-free, emergency (911) and directory assistance (411 and 611).
- Codecs G.729A 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
- Use of the SIP REFER method as an alternative approach to performing call forwarding and call transfer of inbound calls back to the PSTN.
- T-38 Faxing.

Items not supported or not tested included the following:
- Operator services such a dialing 0 or 0 + 10 digits are not supported in this offer by AT&T in Puerto Rico.

2.2. Test Results

Interoperability testing with AT&T was completed with successful results for all test cases with the exception of the observations/limitations described below:

- **Call Display on PSTN transferred calls** – Caller ID display was not properly updated on a PSTN phone involved in a call transferred from the IP Office. After the call transfer was completed, the PSTN phone did not display the actual connected party but instead showed the ID of the host extension that initiated the transfer.
- **Call Display on Twinned Calls.** Caller ID display on the mobile phone associated with a "twinned" extension in the IP Office will display the ID of the host extension and not the one of the actual connected party.

2.3. Support

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

For technical support on the AT&T Mobility SIP Trunk Services offer, call the AT&T Mobility Network Operations Center at 787-717-9900.

MA; Reviewed:
SPOC 8/24/2011

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3 of 27
ATTPR-SipTrIPO7

3. Reference Configuration

Figure 1 below illustrates the test configuration. It shows an enterprise site connected to AT&T SIP Trunk Service through the public IP network.

Located at the enterprise site is an Avaya IP Office 500v2 with analog and digital extensions expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network. Endpoints include Avaya 9600 Series IP Telephones (with H.323 firmware), Avaya 1408 and 9508D Digital Telephones, an Avaya 6210 Analog Telephone, an Officejet 4500 fax machine and Avaya IP Office Softphones in H.323 and SIP modes. The site also has a a Windows XP PC running Avaya IP Office Manager to configure and administer the Avaya IP Office, and Avaya Voicemail Pro for 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.
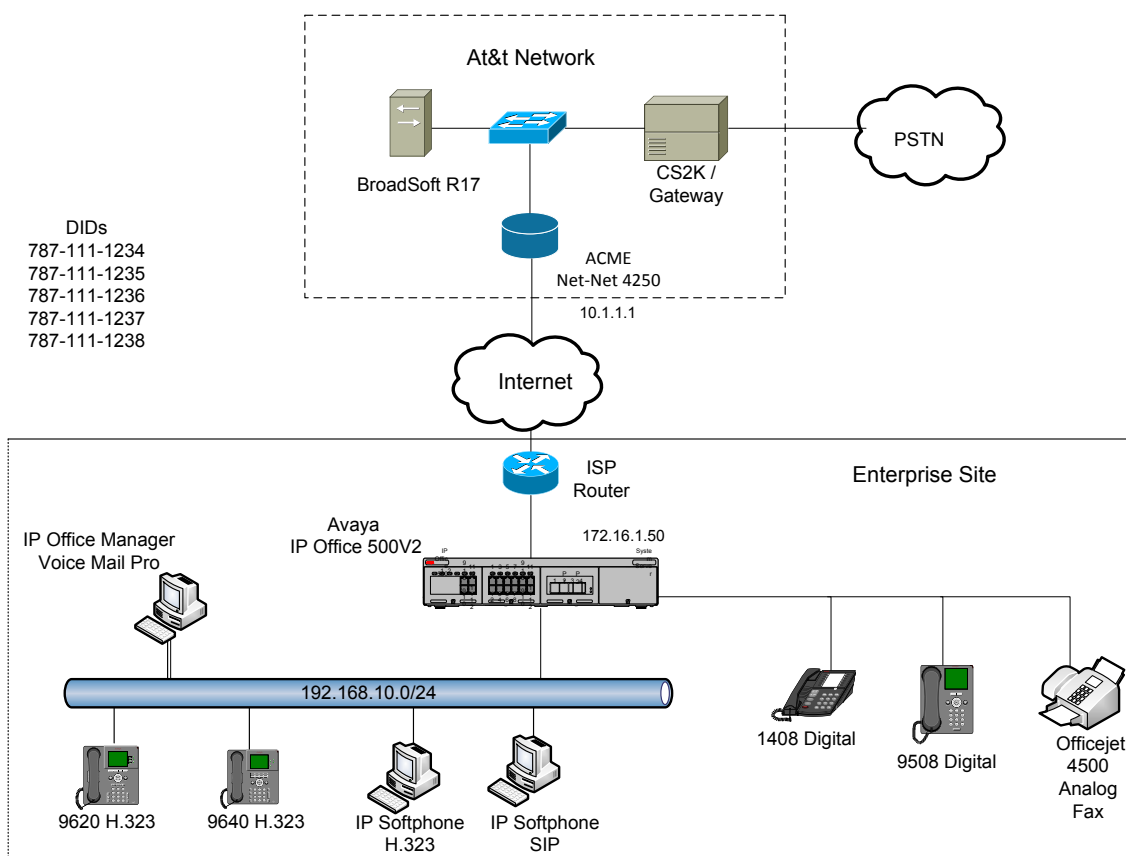
Figure 1: Test Configuration

For security purposes, private addresses are shown in these Application Notes for the IP Office WAN and the ITSP network interfaces, instead of the real public IP addresses used during the

tests. Also PSTN routable phone numbers used in the compliance test have been changed to non-routable ones.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to AT&T. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to AT&T. Since Puerto Rico is a country member of the North American Numbering Plan (NANP), the user dialed 10 digits for local calls, and 11 (1 + 10) digits for other calls between the NANP. Thus for these NANP calls, Avaya IP Office sent 10 or 11 digits in the Request URI and the To header of an outbound SIP INVITE message. For inbound calls, AT&T sent 10 digits in the Request URI and the To header of inbound SIP INVITE messages.

IP Office can be configured to use either SIP Trunk Registration or Static IP Authentication with the service provider. These Application Notes shows the IP Office using the Static IP Authentication method with AT&T.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the 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 the Avaya IP Office 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	7.0 (5)
Avaya IP Office Digital Expansion Module DCPx16	9.0 (5)
Avaya IP Office Manager	9.0 (5)
Avaya IP Office Voicemail Pro	7.0 .17
Avaya 9620 IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.1
Avaya 9640 IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.1
Avaya Digital Telephone 1408	N/A
Avaya Digital Phone 9508	N/A
Avaya IP Office Phone Manager	4.2.39
Avaya IP Office Softphone	3.1.2.17_59616
AT&T Mobility SIP Trunk Service	
Acme-Packet Net-Net 4250 SBC	Firmware SC6.1.0 MR-9 GA (Build 938)
BroadWorks Soft Switch	R17
Nortel CS2K PSTN Gateway	CVM11

5. Configure IP Office

This section describes the Avaya IP Office configuration to support connectivity to the AT&T Mobility 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**, 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. Proper licensing as well as 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) is assumed to be already in place, and they are not part of this Application Notes.

5.1. LAN2 Settings

In the sample configuration, IP500V2 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** 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.

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.

On the **RTP Keepalives** section at the bottom of the page, set the **Scope** field to **RTP**, and **Initial keepalives** to **Enabled**. In a scenario like the one used for the compliance test, where no network address translation (NAT) was used at the IP Office site, this will help compensate for the NAT late binding algorithm used in the service provider's network, and avoid problems of media deadlock with certain types of transferred calls that are routed from the IP Office back to the network, over the same SIP trunk.

All other parameters should be set according to customer requirements.

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. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***. 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 as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. See **Section 5.9** for complete details.
- Set **Public IP Address** to the IP address of 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 AT&T SIP Trunk Services, and therefore is not described in these Application Notes.

5.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** Tab in the Details Pane. Set the **Automatic Codec Preference** for the default codec to be used for intra-enterprise traffic. Choose the **Companding Law** typical for the enterprise location. For North America, *ULAW* is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk.
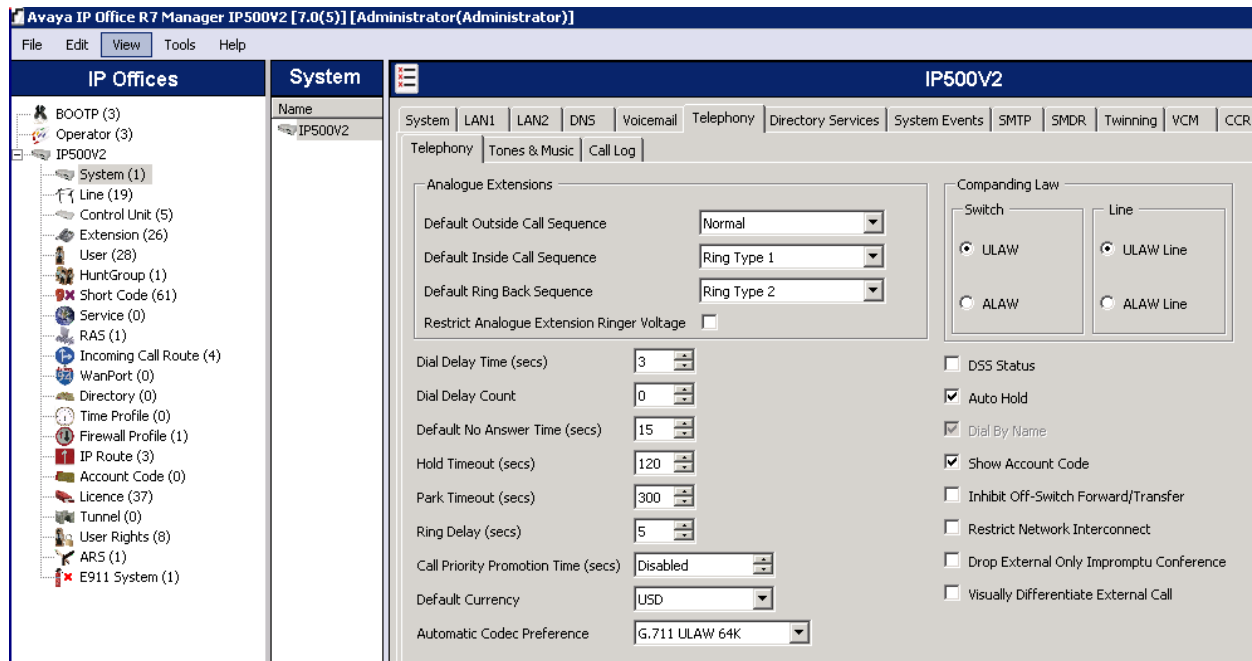
5.3. Twinning Calling Party Settings

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 5.4**). In Avaya IP Office 7.0, this setting also impacts the Caller ID for call forwarding.
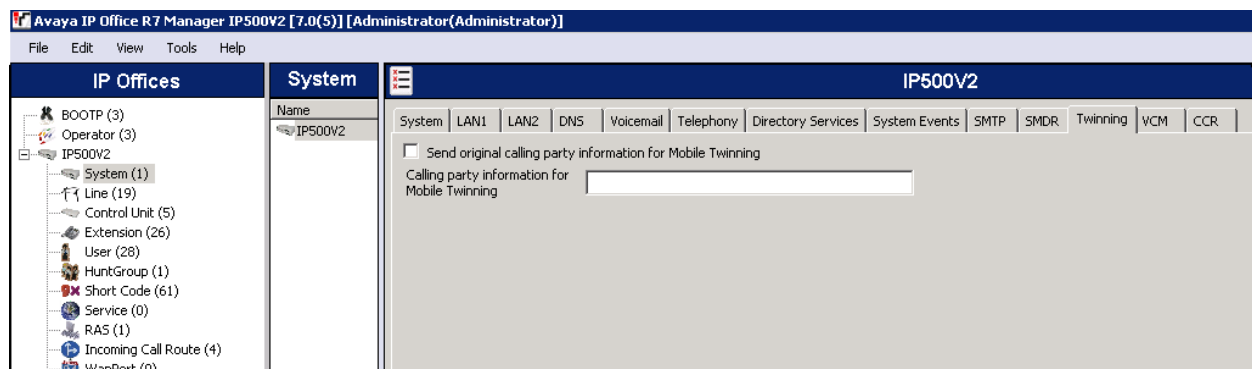
5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the AT&T SIP Trunk Service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New→ SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set the **ITSP Domain Name** to the IP address of the IP Office, LAN2. With this setting IP Office will use this IP address as the host portion of the SIP URI in SIP headers, such as From headers.
- 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 **To Header**. On inbound calls to the IP Office, AT&T will include the customer's "lead" DID number in the Request URI in the INVITE message of every call, and the actual DID number of the called party in the To header. Selecting the **Call Routing Method** to *To Header*, IP Office will use the To header to direct the call to the intended destination, ignoring the Request URI in this decision.
- Set **Send Caller ID** to *Diversion Header*. This field is only used if the **Send original calling party information for Mobile Twinning** box is unchecked in **Section 5.3.** For twinning and call forwarding off-net calls, Avaya IP Office will include the Diversion header in the outbound SIP INVITE message. The Diversion header will contain the number of the re-directing party in IP Office, and the From header will contain the original caller number. The number in the Diversion Header is known to AT&T but the number in the From header is not. At the time of the compliance test, AT&T allowed these calls to complete, based on the known number sent by IP Office on the Diversion header, but the original caller-ID number present in the From header is not being propagated to the called PSTN party. This is a limitation in the feature implementation by AT&T. At the time of writing these Application Notes, this issue is under review by AT&T engineers for a resolution.

MA; Reviewed:
SPOC 8/24/2011
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11 of 27
ATTPR-SipTrIPO7

- Check the **REFER support** box. Select **Always** for both **Incoming** and **Outgoing** to enable the IP Office to send REFER headers for transferred and forwarded calls that are routed back to the PSTN.
- Default values may be used for all other parameters.

Select the **Transport** tab and set the following:
- Set the **ITSP Proxy Address** to the IP address of the AT&T P. R. proxy server.
- Set the **Layer 4 Protocol** to *UDP*.
- Set **Use Network Topology Info** to *LAN2* as configured in **Section 5.1**.
- Set the **Send Port** to *5060*.
- Default values may be used for all other parameters.

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. For the compliance test, a single SIP URI entry was created that matched any number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

- Set **Local URI**, **Contact, Display Name** and **PAI** to *Use Internal Data*. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.6**.

- 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.

- Default values may be used for all other parameters.

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

- The **Compression Mode** was configured using the **Advanced** button, allowing an explicit ordered list of codecs to be specified. Check marks next to *G.729(a) 8K CS-ACELP* and *G.711 ULAW 64K*, which are the codecs supported by AT&T, in that specific order of preference.
- Select *T38* for **Fax Transport Support,** which is the fax standard supported by AT&T.
- Set the **DTMF Support** field to *RFC2833*. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box.
- 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.
- Default values may be used for all other parameters.

Since default values were used for T38 fax, and AT&T Mobility SIP Trunk Service in Puerto Rico does not require Trunk Registration, the **T38 Fax** and **SIP Credentials** tabs need not be visited.

5.5. Short Code

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, 9*N;* . This short code will be invoked when the user dials 9 followed by any number.

- Set **Feature** to *Dial*. This is the action that the short code will perform.

- Set **Telephone Number** to *N"@10.1.1.1"*. 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 of the AT&T SIP proxy server follows the @ sign in the above expression.

- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4**. This short code will use this line group when placing the outbound call

- Default values may be used for all other parameters.

The simple "9N;" short code illustrated above, although effectively routes the outbound calls to the SIP trunk, does not provide the means of alternate routing if the configured SIP Line is out of service or temporarily unavailable, or other features like Secondary Dial Tone. When alternate routing options and/or more customized features and analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. ARS was implemented and tested during the compliance tests, but its configuration is beyond the scope of these Application Notes.

5.6. Users

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. To configure these settings, first navigate to **User** in the left Navigation Pane, 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 1105***. Select the **SIP** tab in the Details Pane.

The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. 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 5.4**). The example below shows the settings for user "Extn 1105". The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by AT&T. In the example, the DID number 7871111235 was used. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.
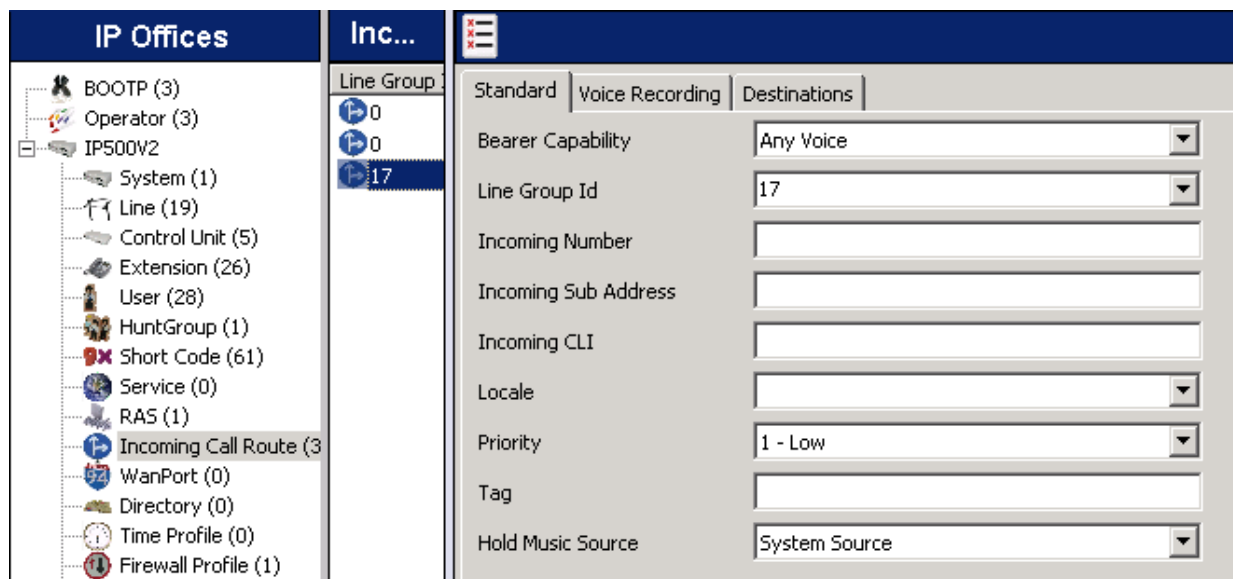
5.7. Incoming Call Route

Incoming call routes map inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, voicemail, etc. in the IP Office. Incoming call routes should be defined for each DID number assigned by the service provider, matching incoming numbers on the SIP line with defined destinations in IP Office.

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 extension in IP Office. The routing decision for the call is based on the parameters already configured for **Call Routing Method,** the **SIP URI (Section 5.4)** and the users **SIP Name** and **Contact** fields, already populated with the assigned AT&T DID numbers **(Section 5.6)**
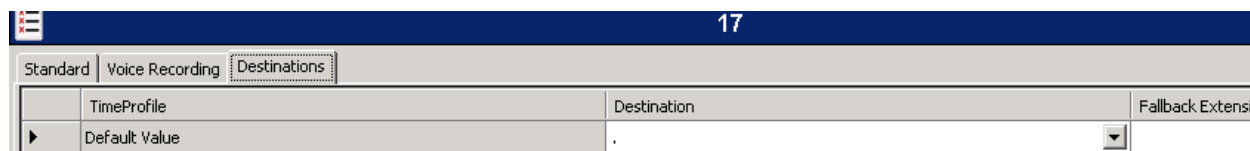
From the left Navigation Pane, right-click on **Incoming Call Route** and select **New.**
On the Details Pane, under the **Standard** tab, set the parameters as show bellow:

- Set **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4**.
- Default values may be used for all other parameters.

Under the **Destinations** tab, enter "**.**" as the Default Value. This will enable all incoming calls to be routed to any extension.

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SPOC 8/24/2011

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17 of 27
ATTPR-SipTrIPO7

5.8. Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "restricted" and "anonymous" respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. 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.
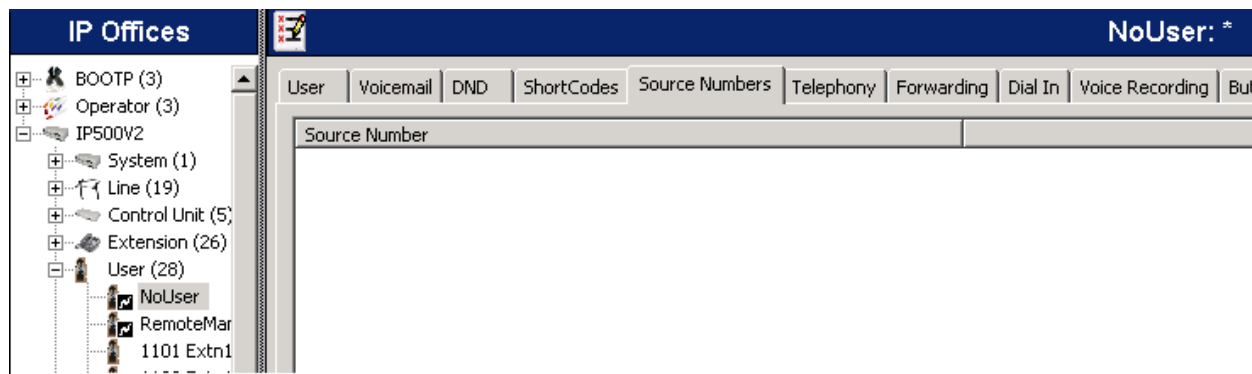
At the bottom of the Details Pane, the **Source Number** field will appear. Enter *SIP_USE_PAI_FOR_PRIVACY*. Click **OK**.
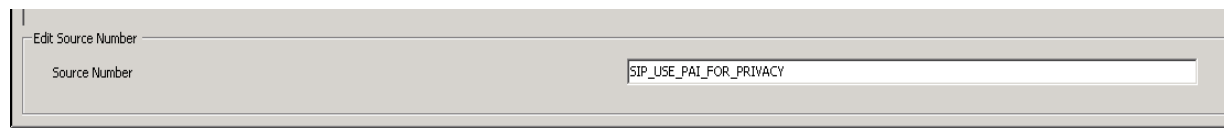
The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.
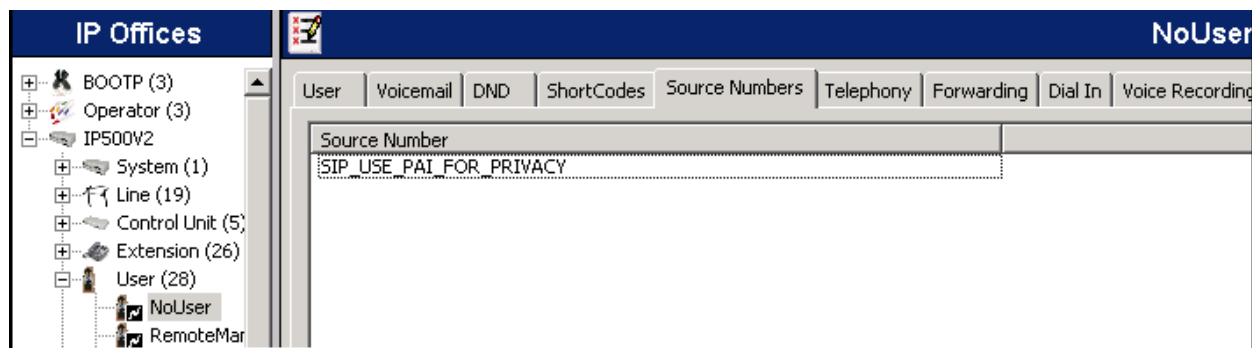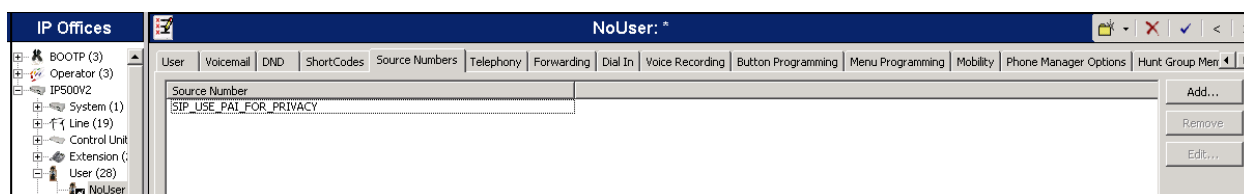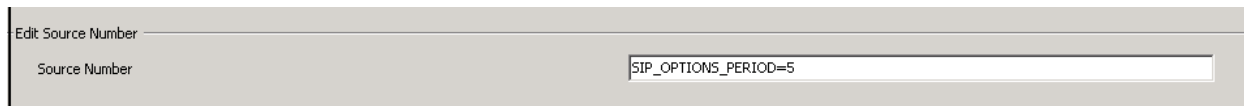
5.9. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.1** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **NoUser** user. The OPTIONS period is determined in the following manner:

- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used.

- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.

- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

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

At the bottom of the Details Pane, the **Source Number** field will appear. Enter *SIP_OPTIONS_PERIOD=X*, where *X* is the desired value in minutes. Click **OK**.

The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 3 minutes was desired. The **Binding Refresh Time** was set to *180* seconds (3 minutes) in **Section 5.1**. The **SIP_OPTIONS_PERIOD** was set to *5* minutes. Avaya IP Office chose the Binding Refresh Time of 180 seconds (3 Minutes) as the smaller of these two values. Click the **OK** button (not shown).
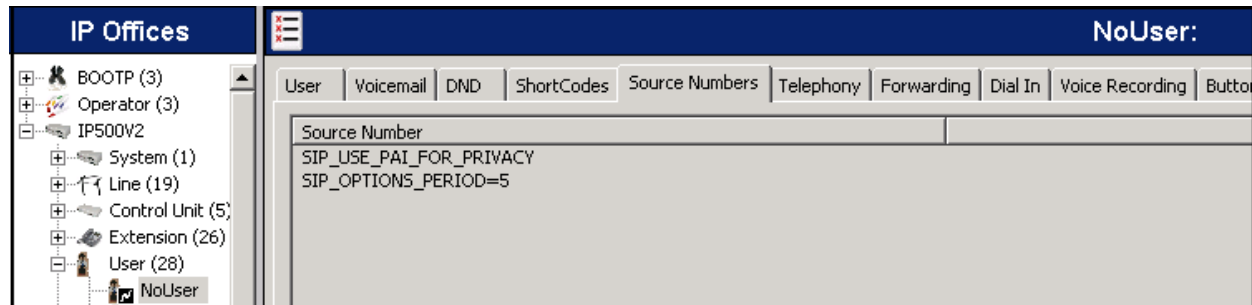
5.10. Save Configuration

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.

6. AT&T Mobility SIP Trunking Configuration

AT&T Mobilty in Puerto Rico is responsible for the configuration of the AT&T SIP Trunk service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise site. AT&T will provide the customer the necessary information to configure the Avaya IP Office SIP connection to AT&T including:

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

7. Verification Steps

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).

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

MA; Reviewed:
SPOC 8/24/2011

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21 of 27
ATTPR-SipTrIPO7

- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Verify that a phone connected to PSTN can successfully place a call to the Avaya IP Office with two-way audio.

8. Conclusion

AT&T Mobility SIP Trunk Service in Puerto Rico passed compliance testing. These Application Notes describe the procedures required to configure the SIP trunk connectivity between Avaya IP Office and AT&T Mobility in Puerto Rico, as shown in **Figure 1**.

9. Additional References

[1] *IP Office 7.0 IP Office Standard Version Installation,* Document number15-601042, May 2011.
[2] *IP Office Release 7.0 Manager 9.0,* Document number15-601011, April 2011.
[3] *IP Office Release 7.0 Voicemail Pro Installation and Maintenance*, Document Number 15-601063 March 2011.
[4] *IP Office Release 6.0 System Status Application,* Document number15-601758, February 2010.
[5] *IP Office Softphone Installation*, Issue 2b, February 2011.

Product documentation for Avaya products may be found at http://support.avaya.com.
Product documentation for AT&T Mobility SIP Trunk Service is available from AT&T Mobility.

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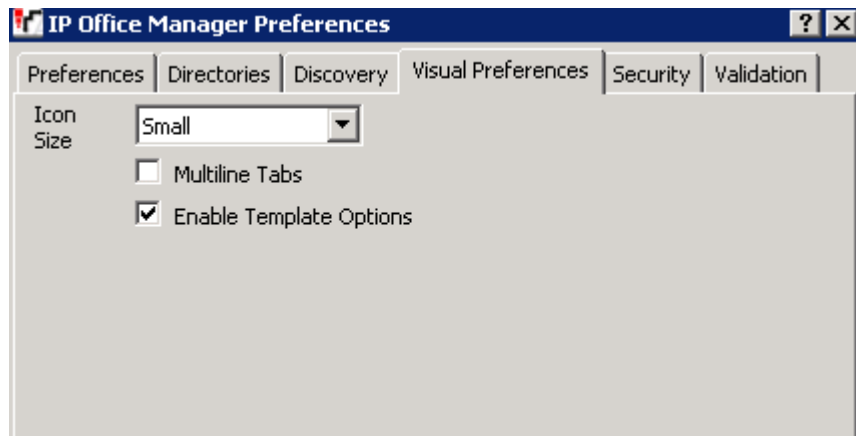
22 of 27
ATTPR-SipTrIPO7

Appendix: SIP Line Template

Avaya IP Office Release 7.0 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 5.4** 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.

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

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SPOC 8/24/2011

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23 of 27
ATTPR-SipTrIPO7

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**.
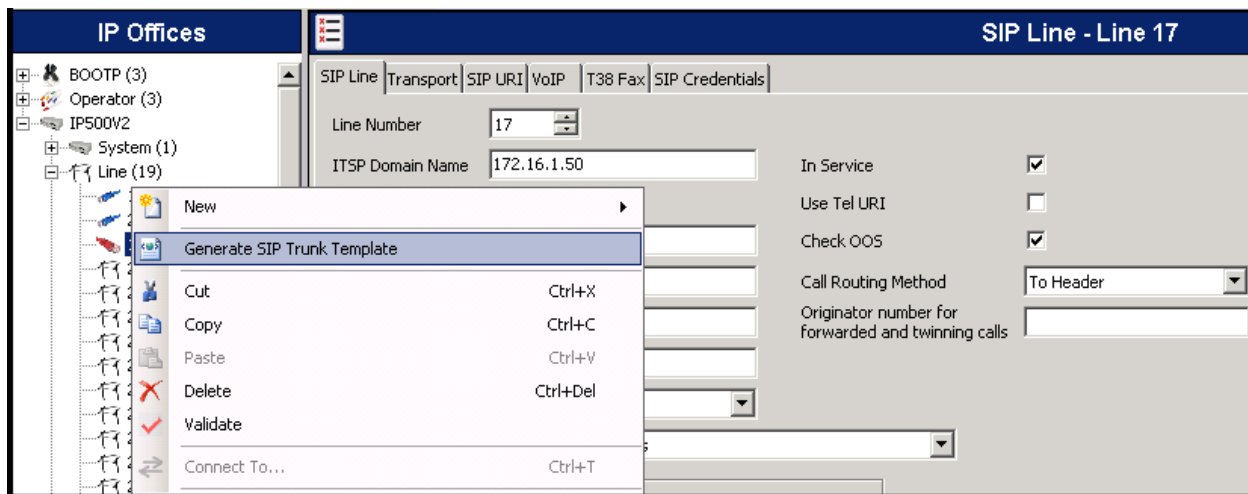
The trunks settings are displayed as configured in **Section 5.4.** Adjust the settings if required and then click on **Export** (not shown).

On the next screen, **Template Type Selection**, select the **Country,** enter the name for the **Service Provider**, and click **Generate Template**.
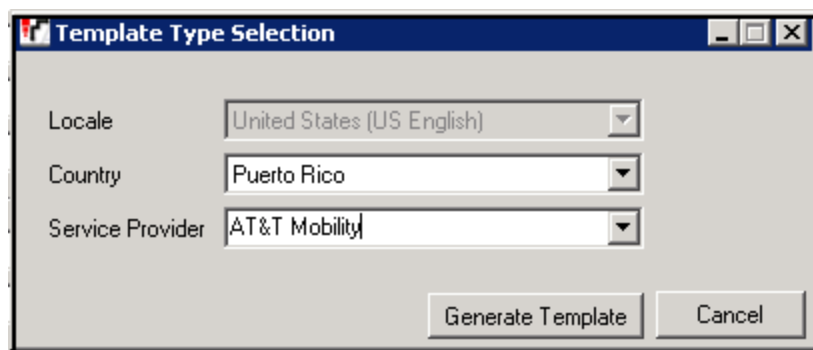
The following is the exported SIP Line Template file, PR_AT&TMobility_SIPTrunk.xml:

```
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20110823</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>SIP Trunk</DescriptiveName>
  <ITSPDomainName>172.16.1.50</ITSPDomainName>
```

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SPOC 8/24/2011
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24 of 27
ATTPR-SipTrIPO7

```xml
<SendCallerID>CallerIDDIV</SendCallerID>
<ReferSupport>true</ReferSupport>
<ReferSupportIncoming>1</ReferSupportIncoming>
<ReferSupportOutgoing>1</ReferSupportOutgoing>
<RegistrationRequired>false</RegistrationRequired>
<UseTelURI>false</UseTelURI>
<CheckOOS>true</CheckOOS>
<CallRoutingMethod>0</CallRoutingMethod>
<OriginatorNumber />
<AssociationMethod>SourceIP</AssociationMethod>
<ITSPProxy>10.1.1.1</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>
<CallInitiationTimeout>4</CallInitiationTimeout>
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<VoipSilenceSupression>false</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_T38</FaxTransportSupport>
<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown>
<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 to 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 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**:
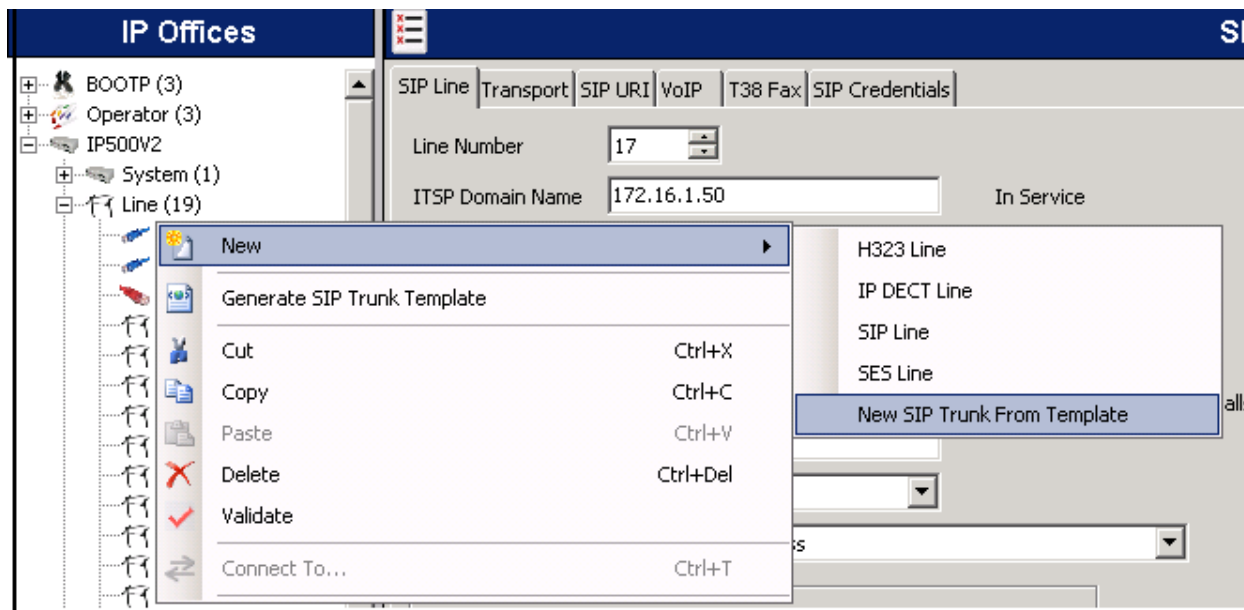
On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.