



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring ThinkTel SIP Trunking Service with Avaya IP Office 9.1 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the ThinkTel SIP Trunking Service and an enterprise solution using Avaya IP Office Release 9.1.

The ThinkTel SIP Trunking Service provides the enterprise with PSTN access via a SIP trunk between the enterprise and the ThinkTel network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 an enterprise solution using Avaya IP Office Release 9.1 to interoperate with the ThinkTel SIP Trunking Service.

The ThinkTel SIP Trunking Service referenced within these Application Notes is positioned for customers who have an IP-PBX or IP-based network equipment with SIP functionality, but need a network service to access the PSTN from the enterprise using IP transport to complete their solution.

The ThinkTel SIP Trunking Service will enable origination and termination of local, long-distance, toll-free, international, and other types of calls across a single broadband IP connection. A SIP signaling interface will be enabled to the Customer Premises Equipment (CPE).

For brevity, the remainder of this document sometimes uses ThinkTel to refer to the ThinkTel SIP Trunking Service.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the ThinkTel SIP Trunking Service via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site comprised of Avaya IP Office 500 V2 running Release 9.1 software, Avaya Voicemail Pro messaging application, Avaya H.323 and SIP hard phones, and SIP-based Avaya softphones. The Avaya IP Office connects directly to the ThinkTel network via the LAN2 port.

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

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

- Establishment of the SIP Trunk
- SIP OPTIONS queries and responses
- Incoming PSTN calls (via the ThinkTel SIP trunk) to SIP and H.323 telephones at the enterprise
- Outgoing PSTN calls (via the ThinkTel SIP trunk) to SIP and H.323 telephones at the enterprise
- Inbound and outbound PSTN calls to/from Avaya Communicator for Windows
- Various call types including: local, long distance, outbound toll-free, international (011 + country code + number) and local directory assistance (411)
- G.711u and G.729a codecs
- Caller ID presentation and Caller ID restriction

- DTMF transmission using RFC 2833
- Voicemail access and navigation for inbound and outbound calls
- Voicemail message waiting indicator (MWI)
- Telephony supplementary features such as hold and resume, call forward, transfer, and conference
- Twinning on inbound calls to PSTN mobile phones
- Use of the SIP REFER method for call redirection to the PSTN
- Inbound and outbound long-duration call stability
- Inbound and outbound long hold time call stability
- Response to incomplete call attempts and trunk busy or error conditions
- T.38 fax

Items not supported or not tested include the following:

- Inbound toll-free and emergency calls (911) were not tested as part of the compliance test.
- ThinkTel does not support Operator (0) and Operator-Assisted (0 + 10-digits) calls.
- ThinkTel does not initiate a SIP session timer refresh. In the compliance test, session refresh for active calls was initiated from the Avaya IP Office.

2.2. Test Results

Interoperability compliance testing of the ThinkTel SIP Trunking Service was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **OPTIONS Response** – On the test circuit used for the compliance test, ThinkTel responded to OPTIONS from Avaya IP Office with "401 Unauthorized". For ThinkTel to respond to OPTIONS with "200 OK", it requires that the OPTIONS message contain an Authorization header, but Avaya IP Office cannot be configured to satisfy this requirement. However, Avaya IP Office treats any response to OPTIONS as an indication that the far end is active and responding; therefore it will not take the SIP connection to the far end out of service. This OPTIONS response is listed here simply as an observation since it had no impact on the status of the SIP connection between Avaya IP Office and ThinkTel.
- **Codec Lockdown** – When the SDP of an outbound call INVITE contained the codec list of G.729a and G.711u in that preference order, ThinkTel's call connect "200 OK" contained the same codecs in the same order in the SDP instead of the single preferred codec (G.729a). However, when the SDP of an outbound INVITE contained the codec list in the order of G.711u and G.729a, ThinkTel's call connect "200 OK" contained the single preferred codec (G.711u). This difference in codec lockdown behavior is listed here simply as an observation since there was no user impact.
- **Unsupported Codec** – When an outbound call was configured to use a codec unsupported by ThinkTel, ThinkTel would return a "183 Session In Progress" message whose SDP contained G.711u. Avaya IP Office would then terminate the call by issuing the CANCEL message. While this behavior was acceptable, it would be more desirable for ThinkTel to return an explicit status message, like "488 Not Acceptable Here" or "415 Media Type Missing" in response to the outbound INVITE.

- **Use of REFER and Call Disconnect (BYE)** – When the SIP REFER method was used for off-net call re-direction (e.g., call forward and call transfer), after accepting the REFER from the enterprise and issuing a BYE, ThinkTel would send an INVITE towards the Avaya IP Office caller. This INVITE would elicit an Avaya IP Office response of “481 Dialog/Transaction Does Not Exist” since the call had already been terminated by the previous BYE. This behavior had no user impact.

2.3. Support

For technical support on ThinkTel SIP Trunking Service, contact ThinkTel at:

- Phone: 1 (866) 928-4465
- Email: support@thinktel.ca
- Website: <http://support.thinktel.ca/>

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

Figure 1 illustrates the sample configuration used for the DevConnect compliance testing. The sample configuration shows an enterprise site connected to the ThinkTel SIP Trunking Service.

Located at the enterprise is an Avaya IP Office 500 V2 running Release 9.1 software. Endpoints include various Avaya IP Telephones (with H.323 and SIP firmware) and a SIP-based Avaya softphone (Avaya Communicator for Windows). The site also has a Windows PC running Avaya Voicemail Pro for providing voice messaging to the Avaya IP Office users, and Avaya IP Office Manager for administering the Avaya IP Office.

Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user phones will also ring and can be answered at the configured mobile phones.

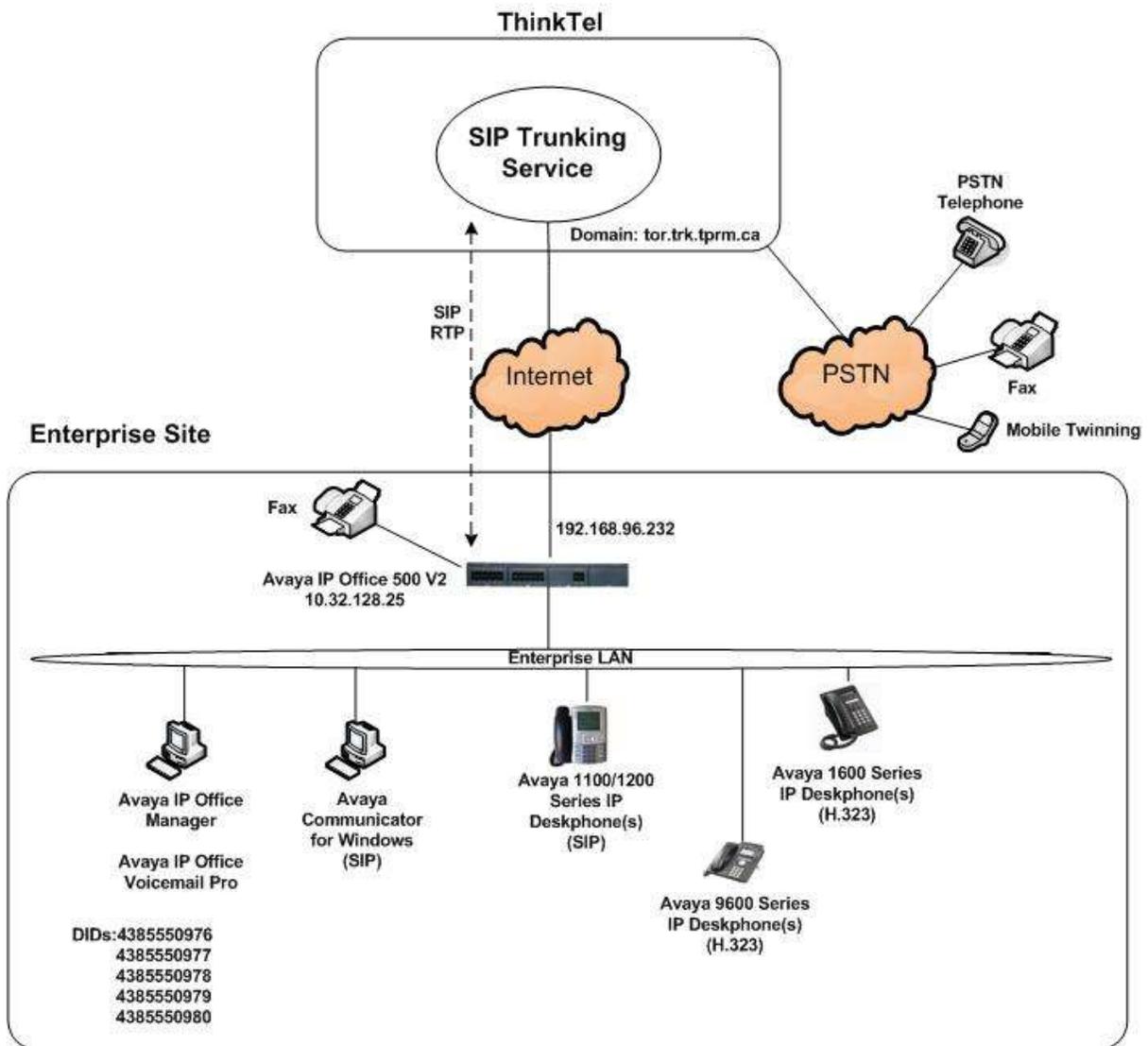


Figure 1: Test Configuration

For security purposes, any actual public IP addresses used in the compliance test were changed to 192.168.x.x throughout these Application Notes.

For the purposes of the compliance test, users dialed a prefix digit 9 plus N digits to send an outbound call to the number N across the SIP trunk to ThinkTel. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent to the service provider network. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits for long distance and local calls. Thus, for these NANP calls, Avaya IP Office sent 11 digits in the Request URI and the To header of an outbound SIP INVITE message. ThinkTel sent 10 digits in the Request URI and the To header of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the enterprise network such as a 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.

The administration of the Avaya Voicemail Pro messaging service and endpoints on Avaya IP Office are standard. Since these configuration tasks are not directly related to the inter-operation with the ThinkTel SIP Trunking Service, they are not included in these Application Notes.

4. Equipment and Software Validated

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

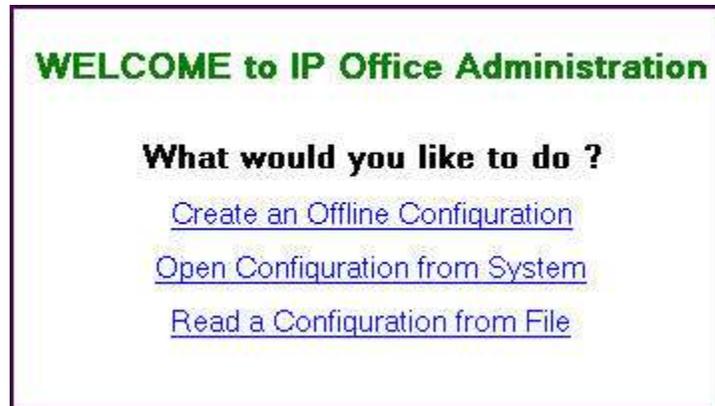
Avaya Telephony Components	
Equipment / Software	Release / Version
Avaya IP Office 500 V2	9.1.5 Build 145
Avaya IP Office COMBO6210/ATM4 Module	9.1.5 Build 145
Avaya IP Office Manager	9.1.5 Build 145
Avaya Preferred Edition (a.k.a Voicemail Pro)	9.1.5.02
Avaya 1140E IP Telephone (SIP)	4.4 SP2 (4.04.18)
Avaya 1616 IP Deskphone (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP5 (1.3.50B)
Avaya 9641G IP Deskphone (H.323) running Avaya one-X® Deskphone Edition	6.6.0 (6.6.0.29)
Avaya Communicator for Windows	2.0.3.30
ThinkTel Components	
Equipment / Software	Release / Version
Metaswitch	8.1
Opensips Session Border Controller	1.11.6 (LTS)

This compliance testing is applicable when the tested solution is deployed with a standalone Avaya IP Office 500 V2 and also when deployed with all configurations of Avaya IP Office Server Edition without T.38 Fax Service.

Avaya IP Office Server Edition requires an Expansion Avaya IP Office 500 V2 to support analog/digital endpoints or trunks.

5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running Avaya IP Office Manager, select **Start → Programs → IP Office → Manager** to launch the application. A screen that includes the following in the center may be displayed:



Select **Open Configuration from System**. If the above screen does not appear, the configuration may be alternatively opened by navigating to **File → Open Configuration** at the top menu of the Avaya IP Office Manager window. Select the proper Avaya IP Office system from the pop-up window (not shown) and log in with the appropriate credentials.

The appearance of the Avaya IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and show the Details pane on the right side. Since the Group Pane has been omitted, its content is shown as submenus in the Navigation pane. These panes (Navigation and Details) will be referenced throughout the Avaya IP Office configuration.

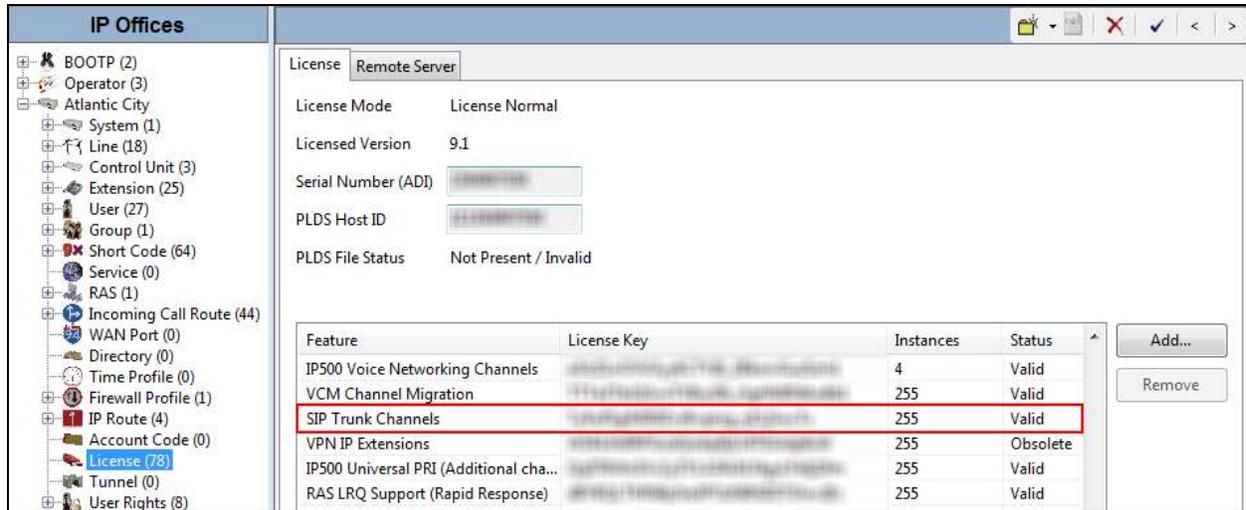
All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and Avaya Communicator for Windows support) is assumed to already be in place.

In the sample configuration, **Atlantic City** was used as the system name. All navigation described in the following sections (e.g., **License → SIP Trunk Channels**) appears as submenus underneath the system name **Atlantic City** in the Navigation Pane. The configuration screens only highlight values/settings configured for the compliance test. Defaults were used for other values and may be customized based upon requirements in the field.

5.1. Licensing and Physical Hardware

The configuration and features described in these Application Notes require Avaya IP Office 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. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details pane.



The screenshot shows the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree view including categories like BOOTP (2), Operator (3), Atlantic City, System (1), Line (18), Control Unit (3), Extension (25), User (27), Group (1), Short Code (64), Service (0), RAS (1), Incoming Call Route (44), WAN Port (0), Directory (0), Time Profile (0), Firewall Profile (1), IP Route (4), Account Code (0), License (78), Tunnel (0), and User Rights (8). The 'License' section is selected. The main pane shows the 'License Remote Server' tab with fields for License Mode (License Normal), Licensed Version (9.1), Serial Number (ADI), PLDS Host ID, and PLDS File Status (Not Present / Invalid). Below these fields is a table of license features:

Feature	License Key	Instances	Status
IP500 Voice Networking Channels	...	4	Valid
VCM Channel Migration	...	255	Valid
SIP Trunk Channels	...	255	Valid
VPN IP Extensions	...	255	Obsolete
IP500 Universal PRI (Additional cha...	...	255	Valid
RAS LRQ Support (Rapid Response)	...	255	Valid

Buttons for 'Add...' and 'Remove' are visible on the right side of the table.

To view the physical hardware comprising the Avaya IP Office system, expand the components under the **Control Unit** in the Navigation pane. In the sample configuration, the second component listed is a Combination Card. This module has 6 digital station ports, two analog extension ports, 4 analog trunk ports and 10 VCM channels. The VCM is a Voice Compression Module supporting VoIP codecs. An Avaya IP Office hardware configuration with a VCM component is necessary to support SIP trunking.

To view the details of the component, select the component in the Navigation pane. The screen below shows the details of the IP 500 V2.

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation tree, and on the right is the configuration details for the selected 'IP 500 V2' unit.

IP Offices		IP 500 V2	
<ul style="list-style-type: none"> BOOTP (2) Operator (3) Atlantic City <ul style="list-style-type: none"> System (1) Line (25) <ul style="list-style-type: none"> Control Unit (3) <ul style="list-style-type: none"> 1 IP 500 V2 2 COMBO6210/ATM4 3 DIGSTA8/ATM4 Extension (25) User (27) Group (1) Short Code (64) Service (0) RAS (1) Incoming Call Route (73) 	<ul style="list-style-type: none"> Unit Device Number: 1 Unit Type: IP 500 V2 Version: 9.1.500.145 Serial Number: [Redacted] Unit IP Address: 10.32.128.25 Interconnect Number: 0 Module Number: Control Unit 		

The screen below shows the details of the Combination Card:

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation tree, and on the right is the configuration details for the selected 'COMBO6210/ATM4' unit.

IP Offices		COMBO6210/ATM4	
<ul style="list-style-type: none"> BOOTP (2) Operator (3) Atlantic City <ul style="list-style-type: none"> System (1) Line (25) <ul style="list-style-type: none"> Control Unit (3) <ul style="list-style-type: none"> 1 IP 500 V2 2 COMBO6210/ATM4 3 DIGSTA8/ATM4 Extension (25) User (27) Group (1) Short Code (64) Service (0) RAS (1) Incoming Call Route (73) 	<ul style="list-style-type: none"> Unit Device Number: 2 Unit Type: COMBO6210/ATM4 Version: 9.1.500.145 Serial Number: [Redacted] Unit IP Address: 0.0.0.0 Interconnect Number: 0 Module Number: Control Unit 		

5.2. System

This section configures the necessary system settings.

5.2.1. System – LAN2 Tab

In the sample configuration, the Avaya IP Office WAN (LAN2) port was used to connect to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office 500 V2. To access the LAN2 settings, first navigate to **System** → <Name>, where <Name> is the system name assigned to the Avaya IP Office. In the case of the compliance test, the system name is **Atlantic City**. Next, 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.

The screenshot displays the Avaya IP Office configuration interface for the system named "Atlantic City". The left-hand pane shows a tree view of system components, including BOOTP (2), Operator (3), Atlantic City, System (1), Atlantic City, Line (25), Control Unit (3), Extension (25), User (27), Group (1), Short Code (64), Service (0), RAS (1), Incoming Call Route (73), WAN Port (0), Directory (0), Time Profile (0), Firewall Profile (1), IP Route (4), Account Code (0), and License (78). The main pane is titled "Atlantic City" and contains several tabs: SMDR, Twinning, VCM, Codecs, VoIP Security, Contact Center, System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, and SMTP. The "LAN2" tab is selected, and the "LAN Settings" sub-tab is active. The configuration fields are as follows:

IP Address	192 . 168 . 96 . 232
IP Mask	255 . 255 . 255 . 224
Primary Trans. IP Address	0 . 0 . 0 . 0
Firewall Profile	<None>
RIP Mode	None
Enable NAT	<input type="checkbox"/>
Number Of DHCP IP Addresses	200
DHCP Mode	<input type="radio"/> Server <input type="radio"/> Client <input type="radio"/> Dialin <input checked="" type="radio"/> Disabled

An "Advanced" button is located at the bottom right of the configuration area.

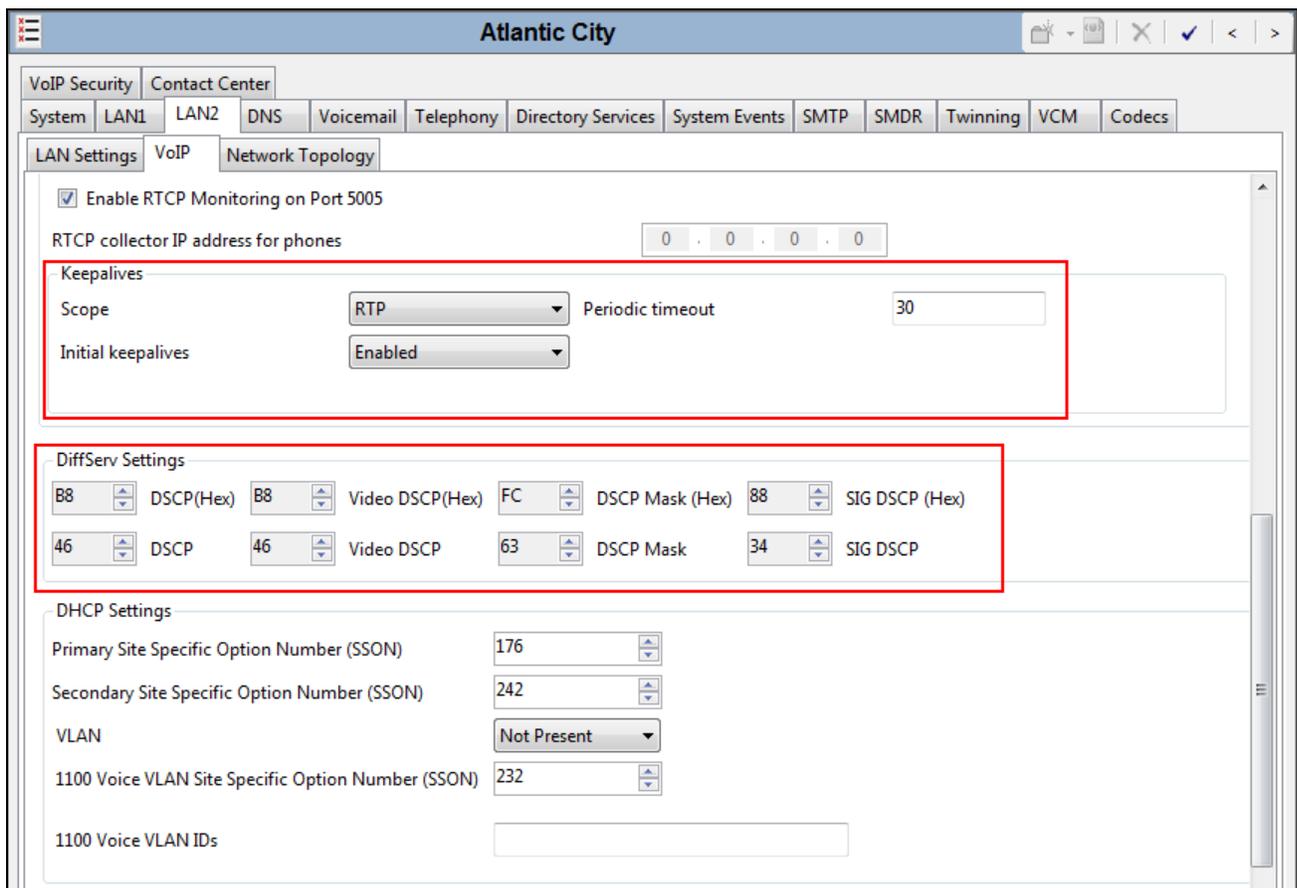
On the **VoIP** tab of LAN2 in the Details Pane, configure the following parameters:

- 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 ports that Avaya IP Office will use for RTP media. This port range will be used to select a destination port for incoming RTP and a source port for outgoing RTP for calls using LAN2. The default values were used.

The screenshot displays the configuration interface for Atlantic City, specifically the VoIP tab for LAN2. The interface includes a navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, and Codecs. The LAN2 tab is active, and the VoIP sub-tab is selected. The configuration is divided into several sections: H323 Gatekeeper, SIP Trunks, SIP Registrar, and RTP. The 'SIP Trunks Enable' checkbox is checked and highlighted with a red box. The 'RTP' section is also highlighted with a red box, showing 'Port Number Range' with Minimum 49152 and Maximum 53246, and 'Port Number Range (NAT)' with Minimum 49152 and Maximum 53246. Other visible settings include H323 Remote Extn Enable (checked), Remote Call Signalling Port (1720), SIP Registrar Enable (unchecked), Auto-create Extn/User (unchecked), SIP Remote Extn Enable (unchecked), Domain Name (empty), Layer 4 Protocol (UDP, TCP, TLS), UDP Port (5060), Remote UDP Port (5060), TCP Port (5060), Remote TCP Port (5060), TLS Port (5061), Remote TLS Port (5061), and Challenge Expiry Time (secs) (10).

Scroll down the page.

- In the **Keepalives** section, set the **Scope** to **RTP**. Set the periodic timeout to **30** and the **Initial keepalives** parameter to **Enabled**. These settings will cause Avaya IP Office to send a RTP keepalive packet starting at the time of initial connection and every 30 seconds thereafter if no other RTP traffic is present. This facilitates the flow of media in cases where each end of the connection is waiting to see media from the other, as well as helping to keep firewall ports open for the duration of the call.
- In the **DiffServ Settings** section, 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 were the Avaya IP Office default values and are shown in the screenshot below. Quality of Service (QoS) is not specifically tested as part of the compliance test. For a customer installation, if the default values are not sufficient, appropriate values should be provided by the customer.
- All other parameters should be set according to customer requirements.



On the **Network Topology** tab of LAN2 in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu 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 the **Open Internet** setting, **STUN Server Address** is not used.
- Set **Binding Refresh Time (seconds)** to the desired value. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. The compliance test used a value of **300** seconds.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set **Public Port** to **5060** for **UDP**.

The screenshot shows the configuration interface for Atlantic City, specifically the Network Topology Discovery settings for LAN2. The interface includes a navigation bar with tabs for VoIP Security, Contact Center, System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, and Codecs. The Network Topology Discovery section contains the following fields and controls:

- STUN Server Address: 10.90.168.13
- STUN Port: 3478
- Firewall/NAT Type: Open Internet
- Binding Refresh Time (seconds): 300
- Public IP Address: 192 . 168 . 96 . 232
- Public Port: UDP (5060), TCP (0), TLS (0)
- Run STUN on startup:

Buttons for Run STUN and Cancel are also visible.

5.2.2. System - Voicemail Tab

In the **Voicemail** tab of the Details Pane, configure the **SIP Settings** section. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from ThinkTel. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Uncheck the **Anonymous** box to allow the Voicemail Caller ID information to be sent to the network.

Note the selection in **Voicemail Type** and the IP address setting for **Voicemail IP Address**. These are for configuring Voicemail Pro as the voice messaging service for Avaya IP Office (part of the standard Avaya IP Office setup beyond the scope of these Application Notes).

The screenshot displays the configuration interface for the Voicemail tab in the Atlantic City system. The interface includes a navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, Codecs, and VoIP Security. The Voicemail tab is active, showing the following settings:

- Voicemail Type:** Voicemail Lite/Pro
- Voicemail Destination:** [Empty]
- Voicemail IP Address:** 10 . 32 . 128 . 79
- Backup Voicemail IP Address:** 0 . 0 . 0 . 0
- Voicemail Channel Reservation:**
 - Unreserved Channels: 259
 - Auto-Attendant: 0
 - Voice Recording: 0
 - Mandatory Voice Recording: 0
 - Announcements: 0
 - Mailbox Access: 0
- DTMF Breakout:**
 - Reception / Breakout (DTMF 0): [Empty]
 - Breakout (DTMF 2): [Empty]
 - Breakout (DTMF 3): [Empty]
- Voicemail Code Complexity:**
 - Enforcement:
 - Minimum length: 3
 - Complexity:
- SIP Settings (highlighted with a red box):**
 - SIP Name: 4385550980
 - SIP Display Name (Alias): Voicemail
 - Contact: 4385550980
 - Anonymous:
- Call Recording:** [Empty]

5.2.3. System - Telephony Tab

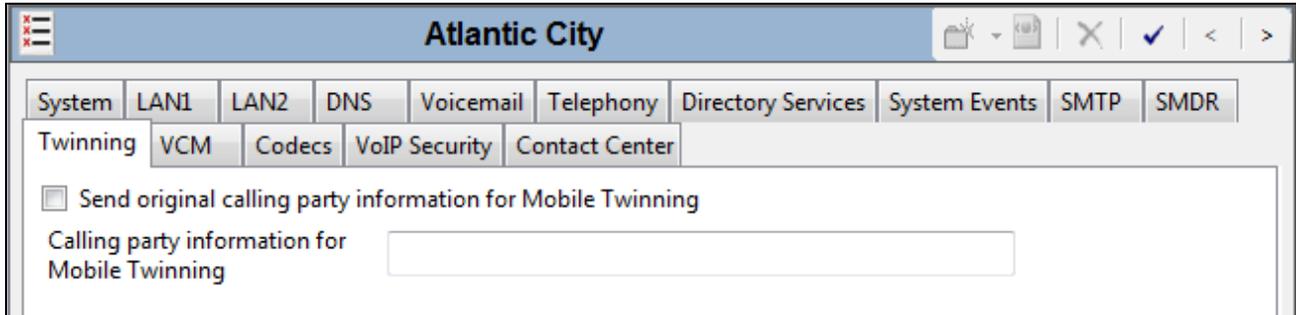
Navigate to the **Telephony** → **Telephony** tab in the Details Pane. Enter or select **0** for **Hold Timeout (secs)** so that calls on hold will not time out. Choose the **Companding Law** typical for the enterprise site. For the compliance test, **U-Law** was used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk per customer business policies. Note that this configuration might pose a security issue (Toll Fraud). Customers should exercise caution with this configuration.

The screenshot displays the configuration interface for the Atlantic City system, specifically the Telephony tab. The interface is organized into several sections:

- Analogue Extensions:** Includes dropdown menus for Default Outside Call Sequence (Normal), Default Inside Call Sequence (Ring Type 1), and Default Ring Back Sequence (Ring Type 2). A checkbox for Restrict Analogue Extension Ringer Voltage is present.
- Time and Delay Settings:** Includes Dial Delay Time (secs) set to 4, Dial Delay Count set to 0, Default No Answer Time (secs) set to 25, Hold Timeout (secs) set to 0 (highlighted with a red box), Park Timeout (secs) set to 300, Ring Delay (secs) set to 5, Call Priority Promotion Time (secs) set to Disabled, Default Currency set to USD, Default Name Priority set to Favor Trunk, Media Connection Preservation set to Disabled, and Phone Failback set to Manual.
- Companding Law:** A section highlighted with a red box containing two columns: Switch and Line. Under Switch, U-Law is selected (radio button checked) and A-Law is unselected. Under Line, U-Law Line is selected (radio button checked) and A-Law Line is unselected.
- Advanced Settings:** Includes checkboxes for DSS Status, Auto Hold (checked), Dial By Name (checked), Show Account Code (checked), Inhibit Off-Switch Forward/Transfer (unchecked and highlighted with a red box), Restrict Network Interconnect, Include location specific information, Drop External Only Impromptu Conference, Visually Differentiate External Call, Unsupervised Analog Trunk Disconnect Handling, High Quality Conferencing (checked), Digital/Analogue Auto Create User (checked), and Directory Overrides Barring.
- Login Code Complexity:** Includes checkboxes for Enforcement and Complexity, with a Minimum length field set to 4.

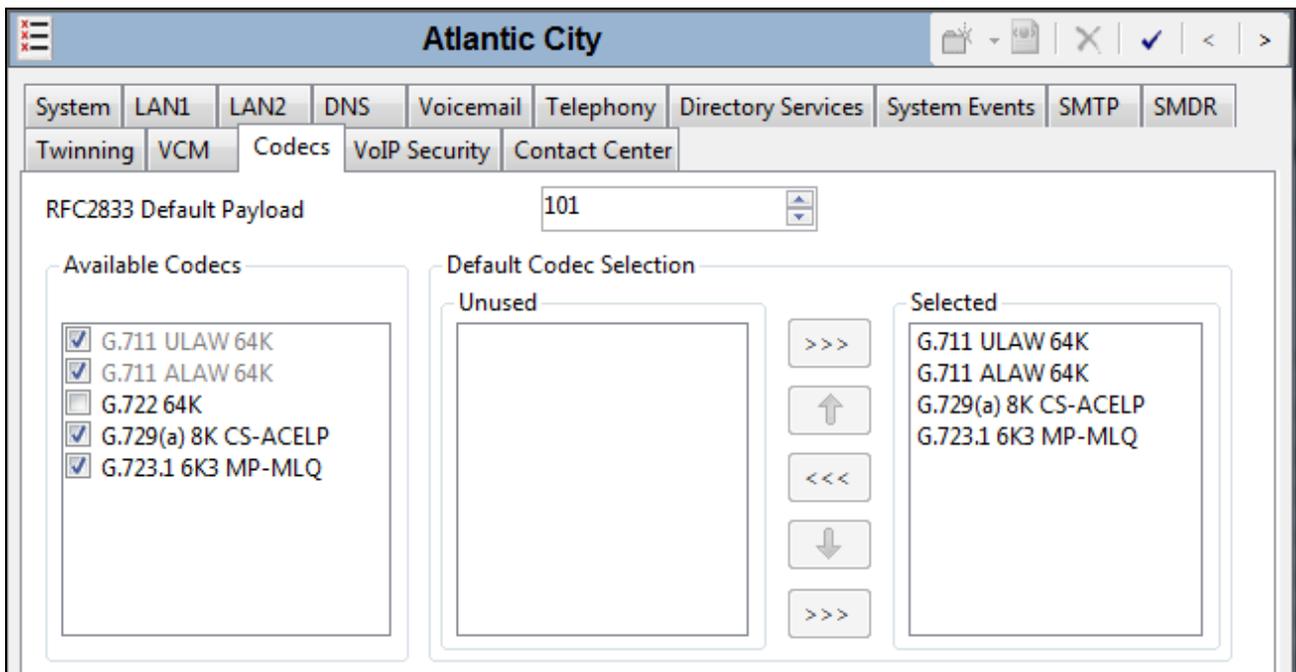
5.2.4. System - Twinning Tab

To view or change the System Twinning settings, navigate to the **Twining** tab in the Details Pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank.



5.2.5. System – Codecs Tab

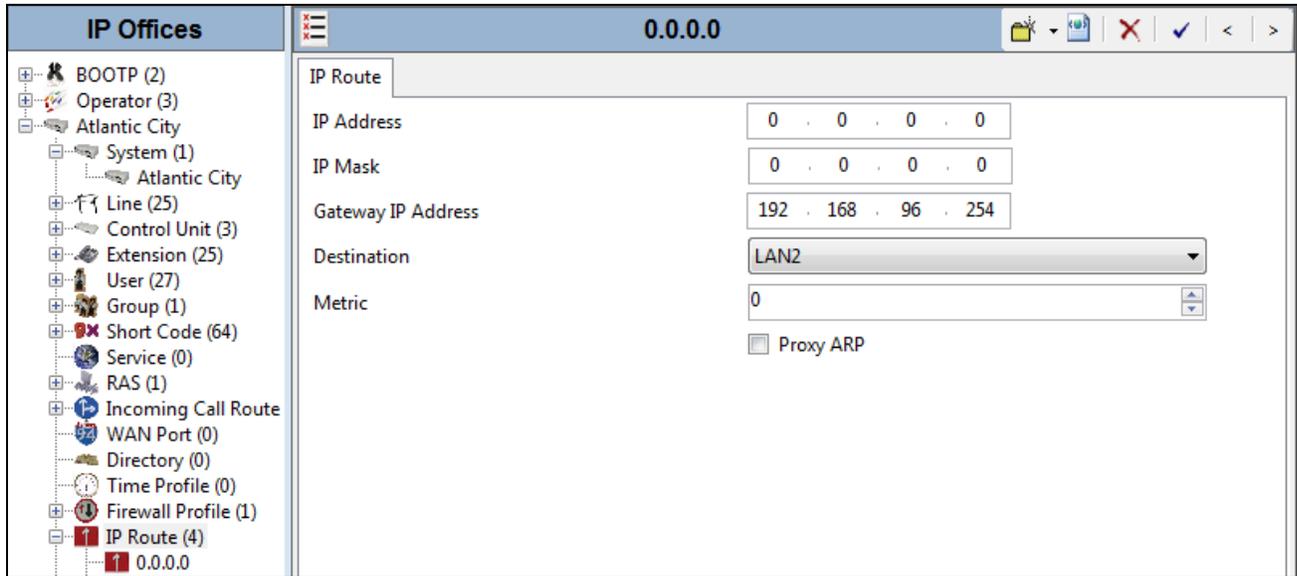
In the **Codecs** tab of the Details Pane, select or enter **101** for **RFC2833 Default Payload**. This setting matched the ThinkTel configuration for use with out-of-band DTMF tone transmissions.



5.3. IP Route

Navigate to **IP Route** → **0.0.0.0** in the left Navigation Pane if a default route already exists. Otherwise, to create the default route, right-click on **IP Route** and select **New**. Create/verify a default route with the following parameters:

- Set **IP Address** and **IP Mask** to **0.0.0.0**.
- Set **Gateway IP Address** to the IP address of the enterprise LAN gateway for the subnet where the Avaya IP Office is connected.
- Set **Destination** to **LAN2** from the drop-down list.



5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the ThinkTel SIP Trunking Service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.1** to create the SIP Line from the template.

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

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

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line settings can be verified against the manual configuration shown in **Sections 5.4.2 – 5.4.8**.

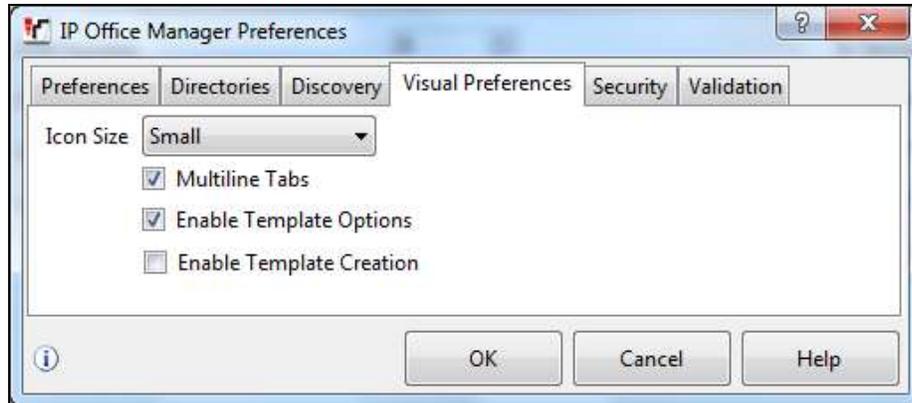
Also, the following SIP Line settings are not supported on Basic Edition:

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

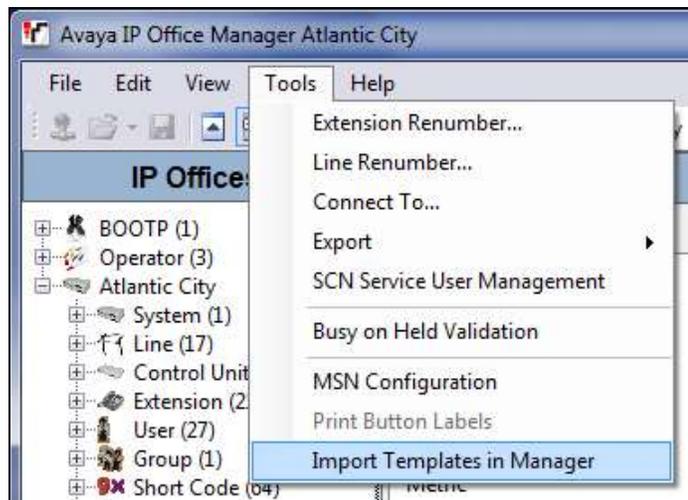
To create a SIP Line manually, right-click **Line** in the Navigation Pane and select **New → SIP Line**; then, follow the steps outlined in **Sections 5.4.2 – 5.4.8**.

5.4.1. Create SIP Line from Template

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



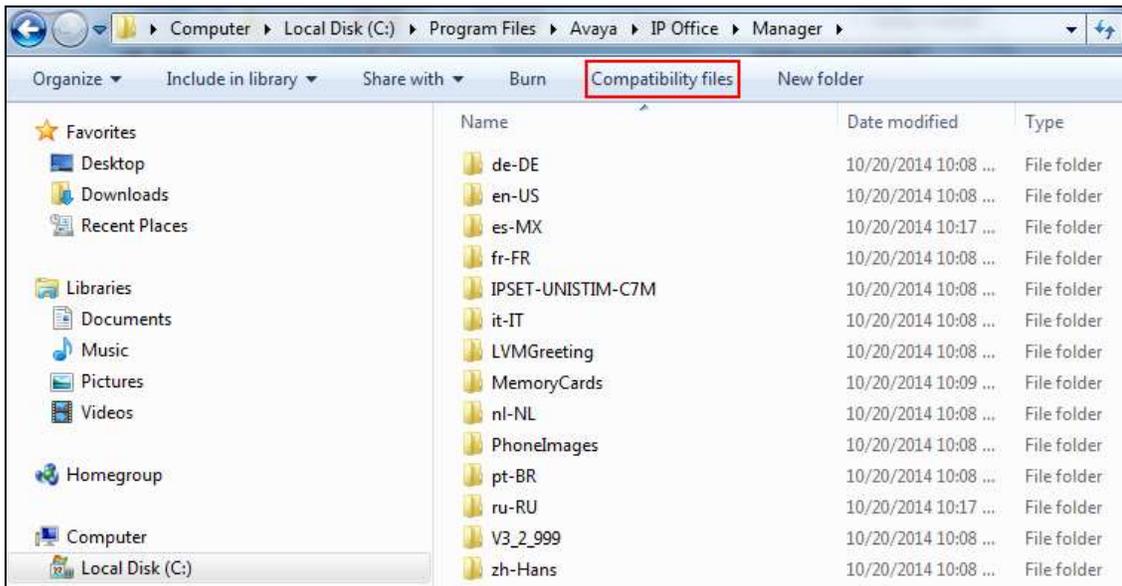
3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template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



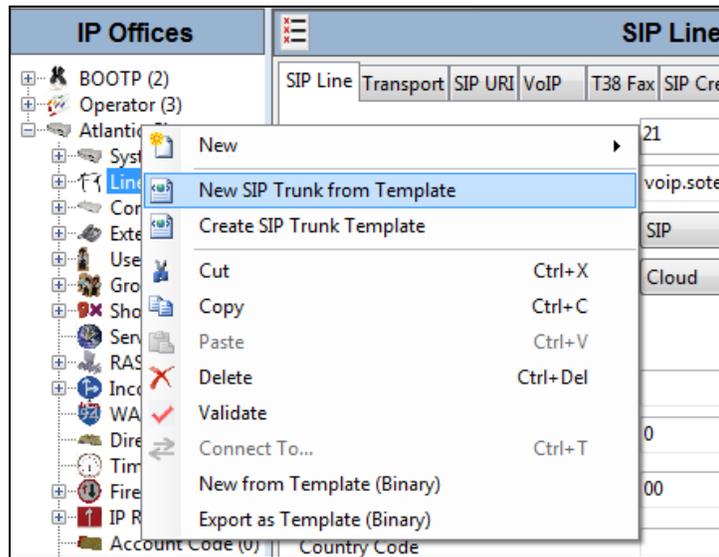
In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** to continue.

If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

Note –Windows 7 (and later) locks the Avaya IP Office 9.1 **\Templates** directory, and it cannot be viewed. To enable browsing of the **\Templates** directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager** (or *C:\Program Files (x86)\Avaya\IP Office\Manager*), and then click on the **Compatibility files** option shown below. The **\Templates** directory and its contents can then be viewed.



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



- In the subsequent **Template Type Selection** pop-up window, select **ThinkTel** from the **Service Provider** drop-down list as shown below. This value corresponds to part of the file name (**AF_ThinkTel_SIPTrunk.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



- Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2 – 5.4.8**.

5.4.2. SIP Line – SIP Line Tab

In the **SIP Line** tab of the Details Pane, configure the parameters as shown below:

- Set the **ITSP Domain Name** to service provider domain provided by ThinkTel.
- Check the **In Service** box. This makes the trunk available to incoming and outgoing calls.
- Check the **Check OOS** box. Avaya IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by Avaya IP Office will use the **Binding Refresh Time** for LAN2, as shown in **Section 5.2.1**.
- Set **Refresh Method** to *Auto*. With this setting, Avaya IP Office will send UPDATE messages for session refresh if the remote party supports UPDATE. If UPDATE is not supported, re-INVITE messages are sent.
- Set **Timer (seconds)** to a desired value. With the value as shown below, Avaya IP Office will send session refresh UPDATE or re-INVITE to the service provider every 5 minutes (half of the specified value).
- Set **Send Caller ID** to *Diversion Header*. With this setting and the related configuration in **Section 5.2.4**, Avaya IP Office will include the Diversion Header for calls that are redirected via Mobile Twinning out the SIP Line to the PSTN. It will also include the Diversion Header for calls that are forwarded out the SIP Line.
- ThinkTel supports using either REFER or re-INVITE for off-net call re-direction as in call transfer. If REFER is used, the media path will be released from the enterprise after the call is redirected. To use REFER, under **Redirect and Transfer**, set the **Incoming Supervised REFER** field and **Outgoing Supervised REFER** field to *Always*. To use reINVITE, set these fields to *Never*.

The screenshot displays the configuration window for 'SIP Line - Line 33'. The left sidebar shows a tree view of IP Office components, with 'Line 33' selected. The main area is divided into several sections:

- Line Information:** Line Number (33), ITSP Domain Name (tor.trk.tpms.ca), URI Type (SIP), Location (Cloud), Prefix, National Prefix (0), International Prefix (00), Country Code, Name Priority (System Default), and Description.
- Operational Settings:** In Service (checked), Check OOS (checked), Session Timers (Refresh Method: Auto, Timer: 600).
- Forwarding and Twinning:** Originator number, Send Caller ID (Diversion Header).
- Redirect and Transfer:** Incoming Supervised REFER (Always), Outgoing Supervised REFER (Always), Send 302 Moved Temporarily, and Outgoing Blind REFER.

5.4.3. SIP Line – Transport Tab

Navigate to the **Transport** tab and set the following:

- Leave the **ITSP Proxy Address** blank. Avaya IP Office will determine the proxy address by performing a DNS lookup on the domain entered in the **ITSP Domain Name** field in **Section 5.4.2**. It should be noted that the creation of the SIP trunk template forced a value to be entered in this field. For the purposes of the template, this field was set with the value entered in the **ITSP Domain Name** field in **Section 5.4.2**. Leaving the **ITSP Proxy Address** blank or setting it to the ThinkTel domain will result in the same behavior.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to the network port used by the SIP line to access the far-end as configured in **Section 5.2.1**.
- Set the **Send Port** to **5060**.

The screenshot shows the configuration window for 'SIP Line - Line 33'. The 'Transport' tab is selected. The 'ITSP Proxy Address' field is empty. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is 5060, 'Use Network Topology Info' is set to 'LAN 2', and 'Listen Port' is 5060. 'Explicit DNS Server(s)' are set to 0.0.0.0. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty.

Field	Value
ITSP Proxy Address	
Layer 4 Protocol	UDP
Send Port	5060
Use Network Topology Info	LAN 2
Listen Port	5060
Explicit DNS Server(s)	0 . 0 . 0 . 0
Calls Route via Registrar	<input checked="" type="checkbox"/>
Separate Registrar	

5.4.4. SIP Line – SIP Credentials Tab

SIP Credentials are used to register or authenticate the SIP Trunk with a service provider if required. SIP Credentials are unique per customer and therefore customers must contact the service provider to obtain the proper registration credentials for their deployment.

To enter the SIP Credentials, select the **SIP Credentials** tab and click **Add**. In the **New SIP Credentials** area that appears, enter the information as shown below.

- Set **User name**, **Authentication Name** and **Contact** to the string provided by ThinkTel. This is generally a 10-digit telephone number as shown below.
- In the **Password** and **Confirm Password** field, enter the password provided by ThinkTel.
- In the **Expiry (mins)** field, enter the time in minutes recommended by ThinkTel.
- Uncheck the **Registration required** box. ThinkTel did not require trunk registration.

Click **OK**.

The screenshot shows a software window titled "SIP Line - Line 33". At the top, there are several tabs: "SIP Line", "Transport", "SIP URI", "VoIP", "T38 Fax", "SIP Credentials" (which is selected), "SIP Advanced", and "Engineering". Below the tabs is a table with the following columns: "Index", "UserName", "Authentication Name", "Contact", "Expiry (mins)", and "Register". The table is currently empty. To the right of the table are three buttons: "Add...", "Remove", and "Edit...". Below the table is a section titled "New SIP Credentials" containing several input fields: "User name" (text box with "4385550976"), "Authentication Name" (text box with "4385550976"), "Contact" (text box with "4385550976"), "Password" (password box with 10 dots), "Confirm Password" (password box with 10 dots), "Expiry (mins)" (spin box with "60"), and "Registration required" (checkbox, which is unchecked). To the right of these fields are two buttons: "OK" and "Cancel".

5.4.5. SIP Line – SIP URI Tab

Select the **SIP URI** tab to create a SIP URI entry or edit an existing entry. A SIP URI entry matches each incoming number that Avaya IP Office will accept on this line. 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 to match any DID number assigned to Avaya IP Office users.

- Set **Local URI** to *Use Internal Data*. This setting allows calls on this line whose incoming Request-URI matches the **SIP Name** set on the **SIP** tab of any **User** as shown in **Section 5.6**. For outbound calls, the From header is populated with the **SIP Name** configured for the **User** placing the call.
- Set **Contact** and **Display Name** to *Use Internal Data*. This setting will cause the Contact and Display Name data for outbound messages to be set from the corresponding fields on the **SIP** tab of the individual **User** as shown in **Section 5.66**.
- Set **PAI** to *Use Internal Data*. This setting directs Avaya IP Office to send the PAI header (P-Asserted-Identity) when appropriate. The PAI header will be populated from the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.66**.
- Set the **Registration** value to the credentials that was configured in **Section 5.4.4**.
- Associate this line with an incoming line group by entering 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. For the compliance test, the incoming and outgoing group **33** was specified. Note that this group number can be different than the SIP Line number.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls allowed using this SIP URI pattern.

Click **OK**.

The screenshot shows the 'SIP Line - Line 33' configuration window. The 'SIP URI' tab is selected, displaying a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is a 'New Channel' form with the following fields:

- Via: 192.168.10.232
- Local URI: Use Internal Data
- Contact: Use Internal Data
- Display Name: Use Internal Data
- PAI: Use Internal Data
- Registration: 1: 4385550976
- Incoming Group: 33
- Outgoing Group: 33
- Max Calls per Channel: 10

Buttons for 'OK' and 'Cancel' are located to the right of the form.

Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user, such as a short code. These URIs are created in the same manner as shown above with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, **Display Name** and **PAI** fields.

5.4.6. SIP Line – VoIP Tab

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

- Set the **Codec Selection** to *System Default*. The default codec set includes codecs **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** which were used in the compliance test. G.711u was configured as the preferred codec. If a different codec set is needed in a specific customer deployment, set **Codec Selection** to *Custom* and use the left/right arrow buttons to move the desired codecs between the **Unused** and **Selected** columns. Use the up/down arrows to reorder the codec priority list.
- Select **T38 Fallback** for **Fax Transport Support** so that Avaya IP Office uses T.38 for sending and receiving faxes on this SIP line. If the called destination does not support T.38, the system will send a re-INVITE to change the transport method to G.711 (for falling back to G.711 pass-through fax).
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones as out-of-band RTP events as per RFC2833.
- Uncheck the **VoIP Silence Suppression** option box.
- Check the **Re-invite Supported** option box.
- Check the **PRACK/100rel Supported** option box. This setting enables support by Avaya IP Office for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.

The screenshot shows the configuration interface for a SIP Line (Line 33) in the VoIP tab. The interface includes several sections:

- Codec Selection:** A dropdown menu is set to "System Default". Below it are two columns: "Unused" (empty) and "Selected" (containing G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ). Navigation buttons (right arrow, up arrow, left arrow, down arrow, and another right arrow) are positioned between the columns.
- Fax Transport Support:** A dropdown menu is set to "T38 Fallback".
- DTMF Support:** A dropdown menu is set to "RFC2833".
- Media Security:** A dropdown menu is set to "Disabled".
- Options:** A list of checkboxes on the right side:
 - VoIP Silence Suppression
 - Re-invite Supported
 - Codec Lockdown
 - Allow Direct Media Path
 - Force direct media with phones
 - PRACK/100rel Supported
 - G.711 Fax ECAN

5.4.7. SIP Line – T.38 Fax Tab

Select the **T38 Fax** tab. Leave the T.38 settings at the default values.

The screenshot shows the 'SIP Line - Line 33' configuration window with the 'T38 Fax' tab selected. The window has a title bar with standard OS icons and a tabbed interface with the following tabs: SIP Line, Transport, SIP URI, VoIP, T38 Fax (selected), SIP Credentials, SIP Advanced, and Engineering. The 'T38 Fax' tab contains the following settings:

- T38 Fax Version: 3 (dropdown)
- Transport: UDPTL (dropdown)
- Redundancy section:
 - Low Speed: 0 (spin box)
 - High Speed: 0 (spin box)
- TCF Method: Trans TCF (dropdown)
- Max Bit Rate (bps): 14400 (dropdown)
- EFlag Start Timer (msecs): 2600 (spin box)
- EFlag Stop Timer (msecs): 2300 (spin box)
- Tx Network Timeout (secs): 150 (spin box)
- Use Default Values: (checkbox)

On the right side of the window, there is a panel with the following options:

- Scan Line Fix-up
- TFOP Enhancement
- Disable T30 ECM
- Disable EFlags For First DIS
- Disable T30 MR Compression
- NSF Override (checkbox)
 - Country Code: 0 (spin box)
 - Vendor Code: 0 (spin box)

At the bottom of the window are three buttons: OK, Cancel, and Help.

5.4.8. SIP Line – Advanced Tab

Select the **SIP Advanced** tab. Set the parameter as shown below.

- Check the **Emulate NOTIFY for REFER** box. With REFER enabled, the Avaya 1100 Series Deskphones and Avaya Communicator for Windows expects to receive a NOTIFY message to indicate that the referred (i.e., transferred) call was successful. If the NOTIFY is not received from the far-end, then the call display will indicate that the transfer failed even if the transfer was successful. If the **Emulate NOTIFY for REFER** box is checked, then Avaya IP Office will send a NOTIFY message (on behalf of the far-end) to the Avaya 1100 Series Deskphones and Avaya Communicator for Windows.

Click the **OK** button at the bottom of the page (not shown).

The screenshot shows the 'SIP Line - Line 33' configuration window with the 'SIP Advanced' tab selected. The window is divided into several sections:

- Addressing:** Association Method is set to 'By Source IP address' and Call Routing Method is set to 'Request URI'. Suppress DNS SRV Lookups is unchecked.
- Identity:** A list of checkboxes for various identity-related settings. 'Cache Auth Credentials' is checked. 'User-Agent and Server Headers' is empty.
- Media:** A list of checkboxes and dropdowns for media-related settings. 'Emulate NOTIFY for REFER' is checked. 'No REFER if using Diversion' is unchecked.
- Call Control:** A list of settings for call control, including timeouts and responses. 'Service Busy Response' is set to '486 - Busy Here', 'on No User Responding Send' is set to '408-Request Timeout', and 'Action on CAC Location Limit' is set to 'Allow Voicemail'.

5.5. Short Code

Define a short code to route outbound calls to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). 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 **9N;** short code, used for the compliance test, 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**. The value **N** represents the number dialed by the user after removing the **9** prefix. This value is passed to ARS (**Section 5.8**).
- Set the **Line Group ID** to the ARS route to be used (**Section 5.8**).

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

The screenshot displays the Avaya Management System interface. On the left is the 'IP Offices' navigation pane with a tree view containing: BOOTP (2), Operator (3), Atlantic City, System (1), Line (25), Control Unit (3), Extension (25), User (27), Group (1), Short Code (64), Service (0), RAS (1), Incoming Call Route (73), and WAN Port (0). The main area is titled '9N;; Dial' and contains the following configuration fields:

Short Code	
Code	9N;
Feature	Dial
Telephone Number	N
Line Group ID	51: SP SIP Route
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code ***67N;** is illustrated. This short code is similar to the **9N;** short code except that the **Telephone Number** field begins with the letter **W**, which means “withhold the outgoing calling line identification”. In the case of the compliance test, when a user dialed *67 plus the number, Avaya IP Office would include the user’s telephone number (DID number assigned to the user) in the **P-Asserted-Identity** (PAI) header and would include the **Privacy: id** header in the outbound INVITE message. Consequently, ThinkTel would prevent presentation of the caller id to the called PSTN destination.

The screenshot shows a configuration window titled '*67N;: Dial'. The window contains the following fields and options:

Short Code	
Code	*67N;
Feature	Dial
Telephone Number	WN
Line Group ID	51: SP SIP Route
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line. To configure these settings, first navigate to **User**→**Name** in the Navigation Pane, where **Name** is the name of the user to be modified. In the example below, the name of the user is **Extn243**. Select the **SIP** tab in the Details Pane. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by ThinkTel. The **SIP Display Name (Alias)** can optionally be configured with a descriptive text string. The value entered for the **Contact** field will be used in the Contact header for outgoing SIP INVITE to the service provider. The value entered for the **SIP Name** is used as the user part of the SIP URI in the From header for outgoing SIP trunk calls.

If outbound calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network (or alternatively use the ***67N**; short code as defined in **Section 5.5**).

IP Offices		Extn243: 243	
BOOTP (2)		User	Voicemail
Operator (3)		DND	Short Codes
Atlantic City		Source Numbers	Telephony
System (1)		Forwarding	Dial In
Line (25)		Voice Recording	Button Programming
Control Unit (3)		Menu Programming	Mobility
Extension (25)		Group Membership	
User (27)		Announcements	SIP
Group (1)		Personal Directory	Web Self-Administration
Short Code (64)			
Service (0)			
RAS (1)			
Incoming Call Route (73)			

SIP Name	4385550978
SIP Display Name (Alias)	Extn243
Contact	4385550978
<input type="checkbox"/> Anonymous	

5.7. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, right-click **Incoming Call Route** in the Navigation Pane and select **New** (not shown).

5.7.1. Incoming Call Route – Standard Tab

On the **Standard** tab in the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capability** to *Any Voice*.
- Set the **Line Group ID** to the **Incoming Group** of the SIP Line defined in **Section 5.4.5**.
- Set the **Incoming Number** to the incoming DID number on which this route should match.

The screenshot shows the configuration window for an Incoming Call Route. The left pane shows a tree view of system components, with 'Incoming Call Route (73)' selected. The right pane shows the 'Standard' tab with the following fields:

Bearer Capability	Any Voice
Line Group ID	33
Incoming Number	4385550978
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

5.7.2. Incoming Call Route – Destination Tab

On the **Destinations** tab, select the destination from the pull-down list of the **Destination** field. In this example, incoming calls to the DID number 4385550978 on Incoming Group 33 are to be routed to the user “Extn243” at extension 243.

The screenshot shows the 'Destinations' tab of the configuration window. It displays a table with the following data:

TimeProfile	Destination	Fallback Extension
Default Value	243 Extn243	

The screen below shows the mapping of inbound calls to IP Office Voicemail Pro for message retrieval.

The screenshot shows a software window with a title bar containing the phone number "33 4385550980" and several icons. Below the title bar are three tabs: "Standard", "Voice Recording", and "Destinations". The "Destinations" tab is active and displays a table with the following structure:

	TimeProfile	Destination	Fallback Extension
▶	Default Value	VoiceMail	

5.8. Alternate Route Selection (ARS)

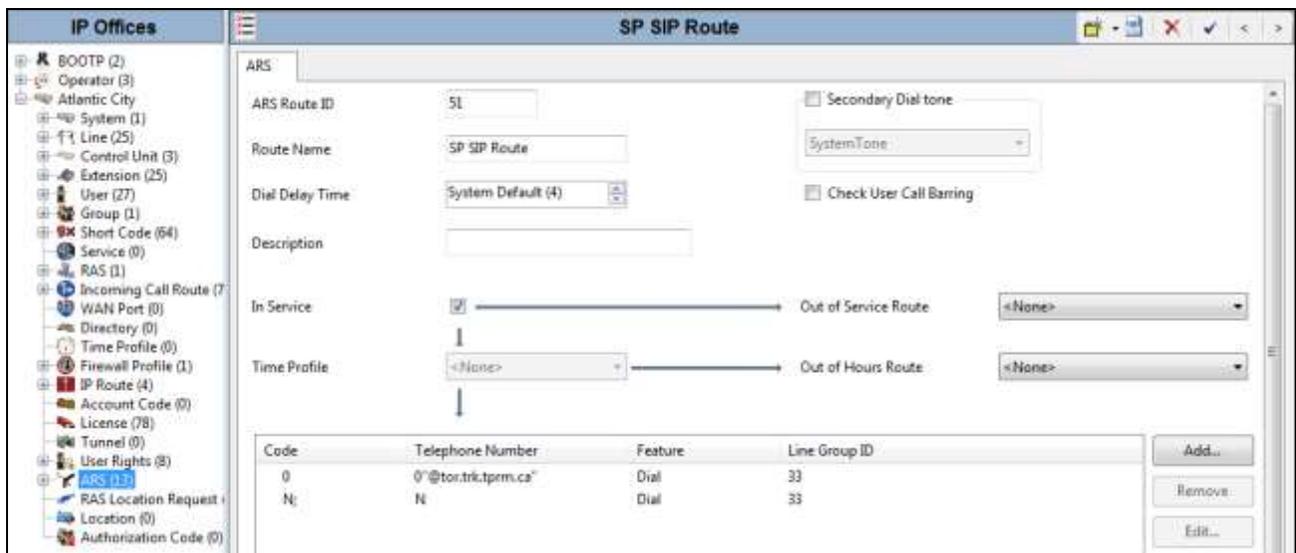
Alternate Route Selection (ARS) is used to route outbound traffic to the SIP line. To define a new ARS route, right-click **ARS** in the Navigation pane and select **New**. In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

For the compliance test, two entries were created. The first entry matches on **0** and the second entry matches on any other number **N**.

To create an entry, click the **Add** button and enter the following in the pop-up window (not shown).

- In the **Code** field, enter the pattern to match the number passed to ARS from the short code in **Section 5.5** followed by a semi-colon. The value **N** will match any number.
- Set **Feature** to **Dial**. This is the action that the entry will perform.
- For **Code 0**, set **Telephone Number** to **0"@domain"**, where *domain* is the ThinkTel domain used to configure the trunk in **Section 5.4.2**. Adding the domain in this field was required to ensure that the correct host appeared in the outbound Request-URI header when dialing 0. This was not required when matching on any other dialed number as shown next. Since ThinkTel does not support dialing an operator (0) (**Section 2.2**), the **Code 0** entry is not strictly required for interoperability. However, adding this entry will allow the Avaya IP Office configuration to support dialing 0 if ThinkTel adds this capability in the future.
- For **Code N;**, set **Telephone Number** to **N**. This field is used to construct the Request-URI and To headers in the outgoing SIP INVITE message. The value **N** represents the complete number passed to ARS.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4.5**. This entry will use this line group when placing the outbound call.

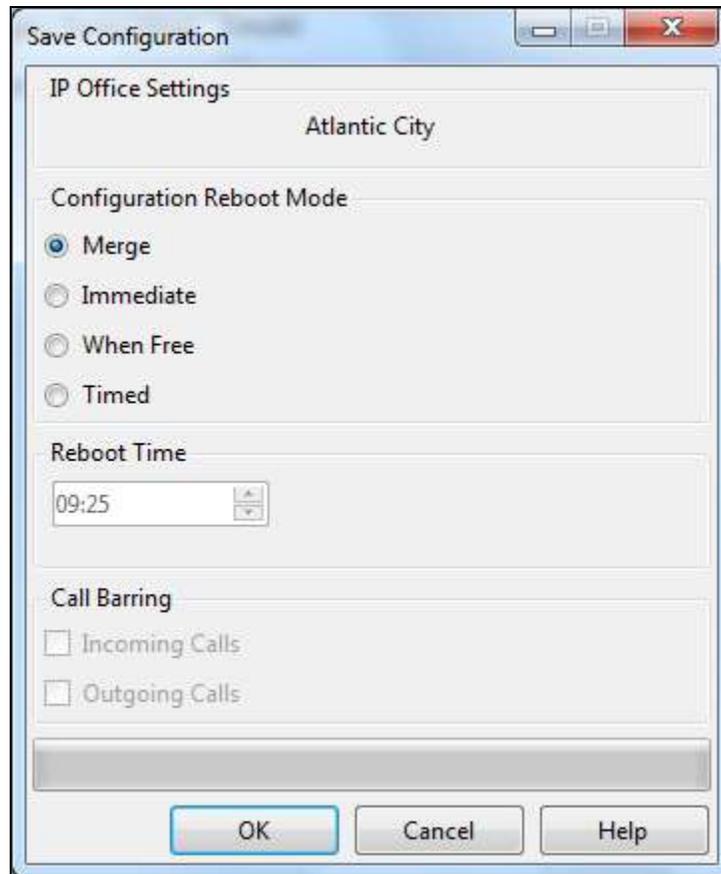
Click the **OK** button (not shown).



5.9. 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.

The following will appear, with either **Merge** or **Immediate** selected, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** to proceed.



The image shows a 'Save Configuration' dialog box with a title bar containing minimize, maximize, and close buttons. The dialog is divided into several sections:

- IP Office Settings:** A text field containing 'Atlantic City'.
- Configuration Reboot Mode:** A group box containing four radio button options: 'Merge' (selected), 'Immediate', 'When Free', and 'Timed'.
- Reboot Time:** A time selection field showing '09:25' with up and down arrow buttons.
- Call Barring:** A group box containing two unchecked checkboxes: 'Incoming Calls' and 'Outgoing Calls'.

At the bottom of the dialog are three buttons: 'OK', 'Cancel', and 'Help'.

6. ThinkTel SIP Trunking Configuration

ThinkTel is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise site (i.e., the IP address of the WAN port on the Avaya IP Office). ThinkTel will provide the customer the necessary information to configure the Avaya IP Office including:

- ThinkTel SIP Trunking Service domain.
- Transport and port for the ThinkTel SIP connection to Avaya IP Office.
- SIP Credentials
- DID numbers to assign to users at the enterprise.
- Supported codecs and their preference order.

7. Verification Steps

This section provides verification steps that may be performed in the field to verify that the solution is configured properly.

7.1. Avaya IP Office System Status

Use the Avaya IP Office System Status application to verify the SIP Line channels state and to check alarms:

- Launch the application from **Start → Programs → IP Office → System Status** on the Avaya IP Office Manager PC. Login with appropriate credentials. Select the SIP Line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current State** is **Idle** for channels where no active calls are currently in session. The state should be **Connected** for channels engaged in active calls.

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

System
Alarms (4)
Extensions (19)
Trunks (25)
Line:1 - 4
Line:5 - 8
Line:17
Line:18
Line:19
Line:20
Line:21
Line:22
Line:23
Line:24
Line:25
Line:26
Line:27
Line:28
Line:29
Line:30
Line:31
Line:32
Line:33
Active Calls
Resources
Voicemail
IP Networking
Locations

Status Utilization Summary Alarms Registration

SIP Trunk Summary

Line Service State: In Service
Peer Domain Name: tor.trk.tprm.ca
Resolved Address: 192.168.250.100
Line Number: 33
Number of Administered Channels: 10
Number of Channels in Use: 0
Administered Compression: G711 Mu, G711 A, G729 A, G7231
Enable Faststart: Off
Silence Suppression: Off
Media Stream: RTP
Layer 4 Protocol: UDP
SIP Trunk Channel Licenses: Unlimited 0%
SIP Trunk Channel Licenses in Use: 0
SIP Device Features: REFER (Incoming and Outgoing)

Channel Number	U...	Call Ref	Current State	Time in State	Remote Media ...	Co...	Conn...	Caller ID or...	Other Party on Call	Direct...	Round Trip ...	Receive Jitter	Receive Pack...	Trans...	Trans...
1			Idle	01:40...											
2			Idle	3 day...											
3			Idle	3 day...											
4			Idle	3 day...											
5			Idle	3 day...											
6			Idle	3 day...											
7			Idle	3 day...											
8			Idle	3 day...											

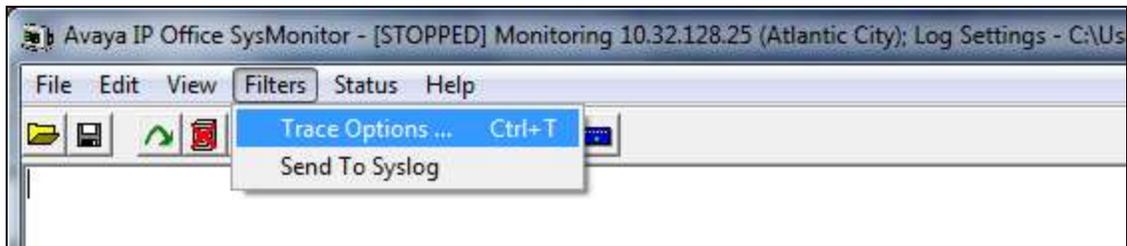
Trace Trace All Pause Ping Call Details Graceful Shutdown Force Out of Service Print...
Save As...

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

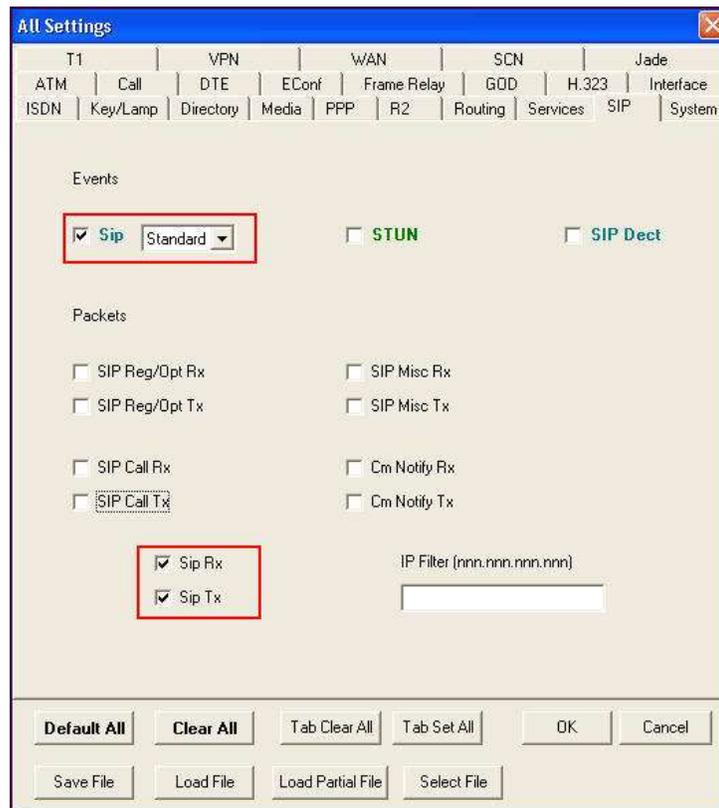
Status	Utilization Summary	Alarms	Registration
Alarms for Line: 33 SIP tor.trk.tprm.ca			
Last Date Of Error	Occurrences	Error Description	

7.2. Avaya IP Office Monitor

The Monitor application can be used to monitor and troubleshoot Avaya IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor** on the Avaya IP Office Manager PC. The application allows the monitored information to be customized. To customize, select **Filters → Trace Options ...** as shown below:



The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, **Standard** SIP Events and the **SIP Rx** and **SIP Tx** boxes are checked.



8. Conclusion

The ThinkTel SIP Trunking Service passed compliance testing with Avaya IP Office 9.1. These Application Notes describe the configuration necessary to connect Avaya IP Office 9.1 to ThinkTel as shown in **Figure 1**. Test results and observations are noted in **Section 2.2**.

9. Additional References

Avaya IP Office 9.1

- [1] *IP Office Documentation Catalog*, Release 9.1, Documentation number 16-604278 Issue 2.1, December 2014.
- [2] *IP Office 9.1 Platform Solution Description*, Issue 01.17, February 2016.
- [3] *Avaya IP Office 9.1 Deploying Avaya IP Office Platform IP500 V2*, Document Number 15-601042 Issue 30za, February 2016.
- [4] *Avaya IP Office 9.1 Administering Voicemail Pro*, Document number 15-601063 Issue 10m, February 2016.
- [5] *Administering Avaya IP Office Platform with Manager*, Issue 10.38, February 2016.
- [6] *Avaya IP Office 9.1 Using System Status*, Document Number 15-601758 Issue 10f, August 2015.
- [7] *Avaya IP Office 9.1 Using IP Office System Monitor*, Document Number 15-601019, Issue 06g, February 2016.
- [8] *Avaya IP Office 9.1 H.323 Telephone Installation Notes*, Document Number 15-601046, Issue 20h, December 2015.
- [9] *Avaya IP Office 9.1 SIP Extension Installation*, Issue 4a, May 2015.

Additional IP Office documentation can be found at
<http://marketingtools.avaya.com/knowledgebase/>.

Product documentation for the ThinkTel SIP Trunking Service is available from ThinkTel. See **Section 2.3** on how to contact ThinkTel.

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