Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the CenturyLink IQ® SIP Trunk Service with Avaya IP Office Release 9.1 – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking for an enterprise solution using Avaya IP Office Release 9.1 to interoperate with the CenturyLink IQ® SIP Trunk Service.

The CenturyLink IQ® SIP Trunk Service provides PSTN access via a SIP trunk between the enterprise and the CenturyLink network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise. CenturyLink is a member of the Avaya DevConnect Service Provider program.

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 steps to configure Session Initiation Protocol (SIP) Trunking for an enterprise solution using Avaya IP Office Release 9.1 to interoperate the CenturyLink IQ® SIP Trunk Service.

The CenturyLink IQ® SIP Trunk Service will enable delivery of origination and termination of local, long-distance and toll-free traffic across a single broadband connection. A SIP signaling interface will be enabled to the Customer Premises Equipment (CPE).

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of Avaya IP Office and various Avaya endpoints listed in **Section 4**.

The CenturyLink IQ® SIP Trunk Service passed compliance testing with any observations or limitations described in **Section 2.2**.

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 and registration of the SIP trunk
- Sending/receiving SIP OPTIONS queries to/from the service provider
- Incoming PSTN calls (via the CenturyLink SIP trunk) to SIP and H.323 telephones at the enterprise
- Outgoing PSTN calls (via the CenturyLink SIP trunk) from SIP and H.323 telephones at the enterprise
- Inbound and outbound PSTN calls to/from Avaya Communicator for Windows
- Various call types including: local (10 digits), long distance (1 + 10 digits), outbound toll-free, international (011 + country code + number) and local directory assistance (411)
- Codecs G.711MU and G.729A
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Response to incomplete call attempts and trunk errors
- Voicemail navigation using DTMF input for inbound and outbound calls
- Voicemail message waiting indicator (MWI)

- User features such as hold and resume, internal call forwarding, transfer, and conference
- Off-net call forwarding and twinning
- T.38 and G.711 fax

Emergency calls (911) and inbound toll-free calls are supported but were not tested as part of the compliance test.

2.2. Test Results

Interoperability testing of the CenturyLink IQ® SIP Trunk Service was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **CenturyLink does not support REFER for call forward**: The call scenario is an inbound call from the PSTN to Avaya IP Office which is then forwarded to another PSTN endpoint. In this scenario, if REFER is enabled (**Section 5.4.2**), CenturyLink returns a failure to the REFER message. The call is still successful because on receipt of the failure message, Avaya IP Office stays in the middle of the call. However, the Avaya IP Office trunk will not be released until the call is terminated.
- **Call forwarding of a privacy call fails:** The call scenario is the same as in the previous bullet item except the inbound call has calling party number block (aka privacy) enabled. CenturyLink sends a 403 Forbidden or a 606 Not Acceptable response on the forwarded call leg and the call is terminated. This occurs with REFER enabled or disabled. However, since the completion of this testing, CenturyLink has reported that this issue has been corrected.
- **Disable Error Correction Mode (ECM) for T.38 fax:** CenturyLink does not support ECM for T.38; however, CenturyLink sets the ECM bit in the facsimile control field describing its capabilities in the T.30 signaling. Thus for interoperability, ECM should be disabled on Avaya IP Office so the resulting call will negotiate to not use ECM (**Section 5.4.7**).
- **SIP Line should be taken out of service before making changes:** Certain changes to the Avaya IP Office SIP Line configuration will cause the Avaya IP Office to unregister and reregister the SIP Line in rapid succession. Depending on the arrival of the messaging at the far-end, the trunk may end-up unregistered. As a workaround, the Avaya IP Office SIP Line should be taken out of service prior to making changes on the SIP Line and then placed back in service once the changes are complete. This is unrelated to CenturyLink interoperability.
- **SIP endpoints may indicate that a transfer failed even when it is successful:** Occasionally on performing a transfer operation, Avaya IP Office SIP endpoints (Avaya 1100 Series Deskphone and Avaya Communicator for Windows) may indicate on the local call display that the transfer failed even though it was successful. The frequency of this behavior can be reduced by enabling “Emulate Notify for REFER” on the IPO SIP Line (**Section 5.4.8**).

2.3. Support

For technical support on the CenturyLink IQ® SIP Trunk Service, please contact CenturyLink via the following:

- Web: http://www.centurylink.com
- Enterprise Business Support: 1-888-638-6771

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 CenturyLink IQ ® SIP Trunk Service.

The enterprise site contains an Avaya IP Office 500 V2 with various endpoints and a Windows 7 PC running Avaya IP Office Manager to configure Avaya IP Office. This PC also runs Avaya VoiceMail Pro for voicemail. For the compliance test, Avaya IP Office is deployed in a 2-port configuration. The Avaya IP Office LAN1 port is connected to the private enterprise LAN and the WAN (LAN2) port is connected to the public network.

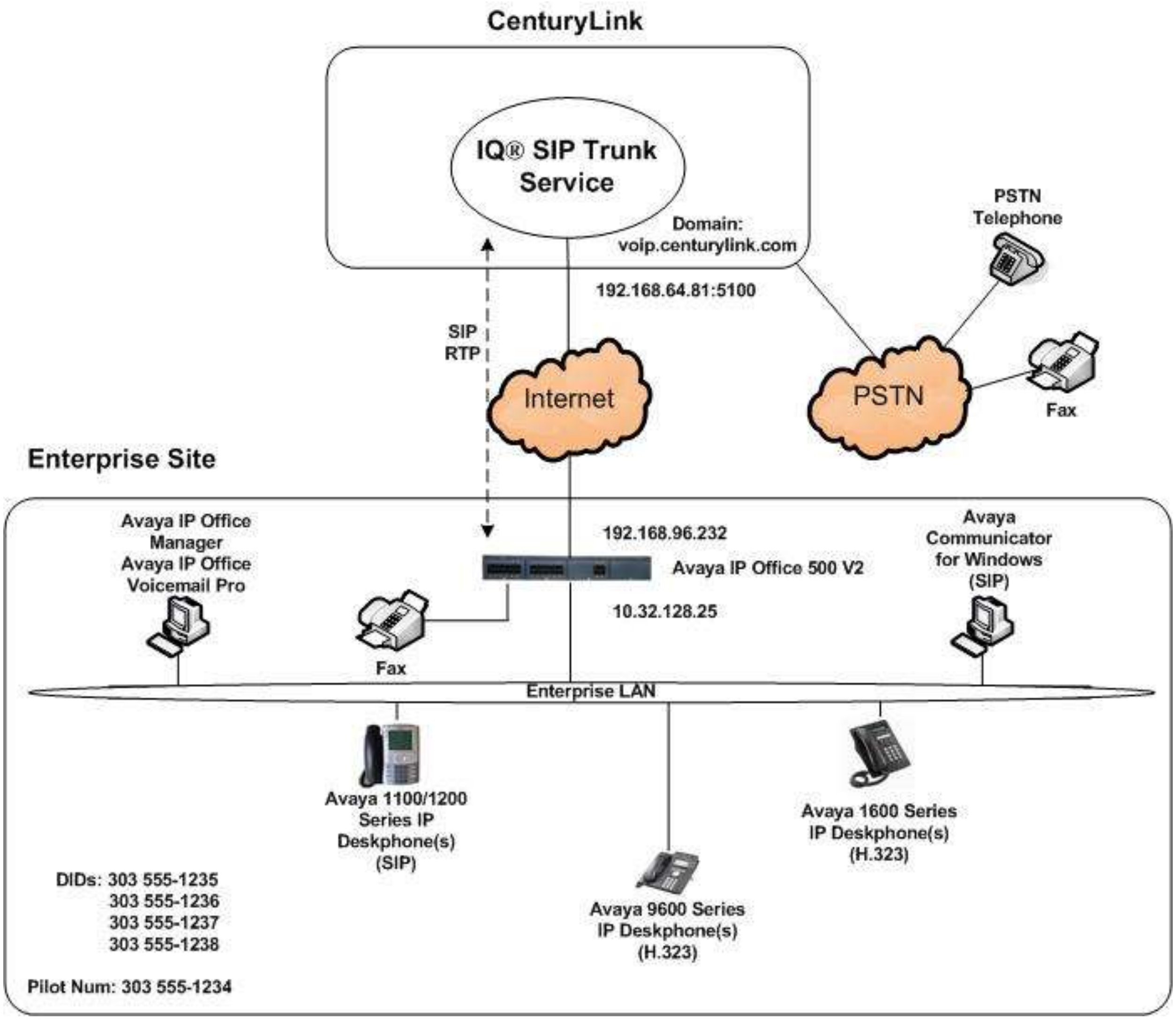


Figure 1: Avaya Interoperability Test Lab Configuration

For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with numbers that cannot be routed over the PSTN.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to CenturyLink. The short code of 9 is stripped off by Avaya IP Office and the remaining digits were sent unaltered to CenturyLink. CenturyLink accepts 11 digits in the Request-URI header for long distance and local calls. For inbound calls, CenturyLink sends 10 digits in the Request-URI.

4. Equipment and Software Validated

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

Avaya Telephony Components	
Equipment	**Software**
Avaya IP Office 500 v2	9.1 SP2 (9.1.2.0.91)
Avaya IP Office Manager	9.1 SP2 (9.1.2.0.91)
Avaya IP Office VoiceMail Pro	9.1 SP2 (9.1.2.0.61)
Avaya 1140E IP Deskphone (SIP)	4.4 SP2 (4.04.18)
Avaya 1616 IP Deskphone (H.323) running Avaya one-X® Deskphone Value Edition	1.3.6 (1.3.60A)
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

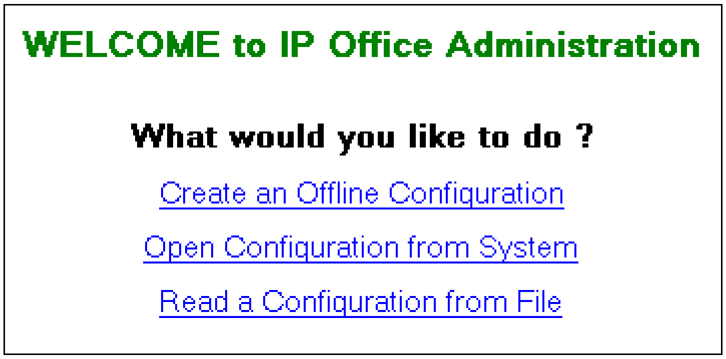
CenturyLink Components	
Equipment	**Software**
BroadSoft BroadWorks	20.0_1.606
Oracle Net-Net 6300 Session Border Controller	SCZ7.1.2 MR-3 GA (Build 359)

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

Avaya IP Office Server Edition requires an Expansion IP Office 500 V2 R9 to support analog/digital endpoints or analog/digital trunks.

5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the Avaya IP Office Manager PC, select **Start → All Programs → IP Office → Manager** to launch the application. A screen that includes the following 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 of the Avaya IP Office Manager window. Select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials.

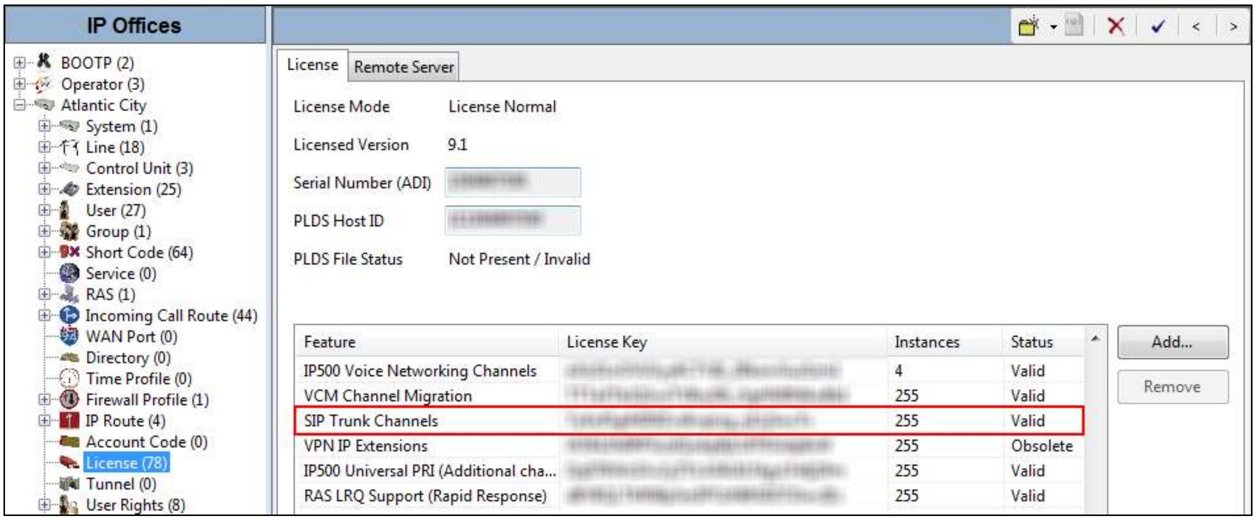
The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and show the Details pane on the right side. Since the Group pane has been omitted, its content is shown as submenus in the Navigation pane. The Navigation and Details panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and Avaya Communicator 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.

5.1. Licensing and Physical Hardware

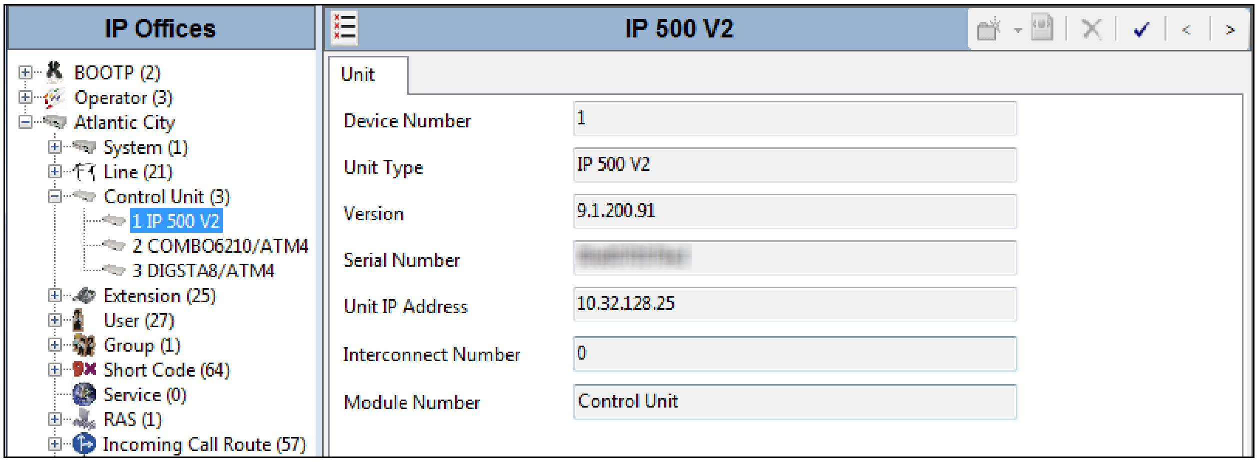
The configuration and features described in these Application Notes require Avaya IP Office to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

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) appears in the Details pane.



To view the physical hardware comprising Avaya IP Office, 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 stations 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 following screen shows the details of the **IP 500 V2**.

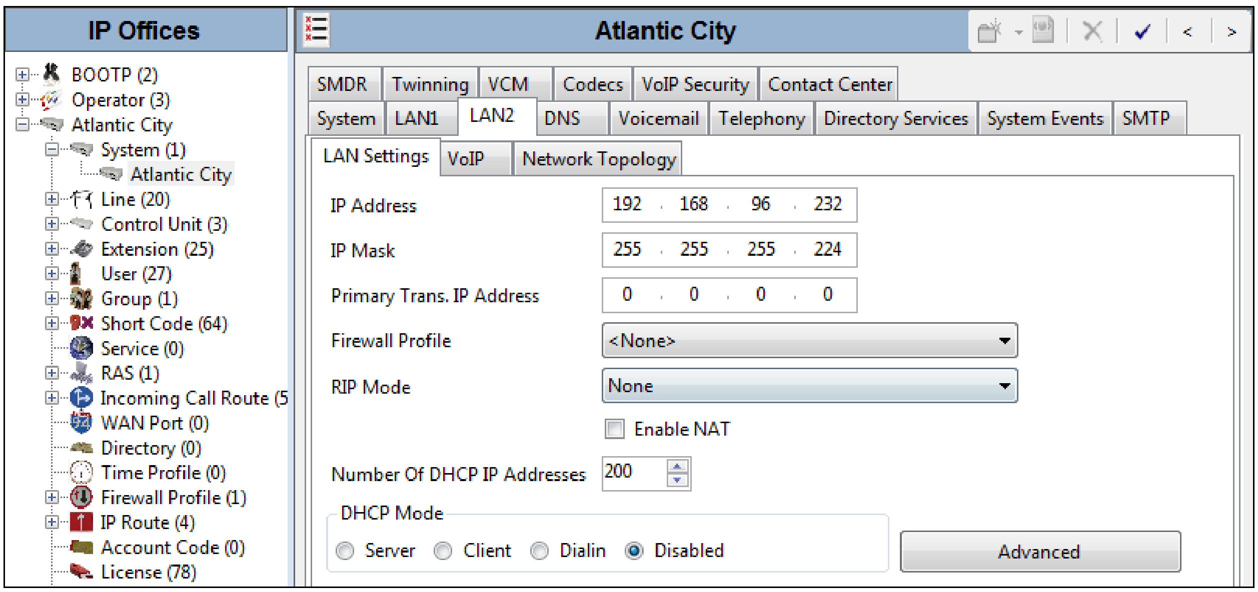


5.2. System

Configure the necessary system settings.

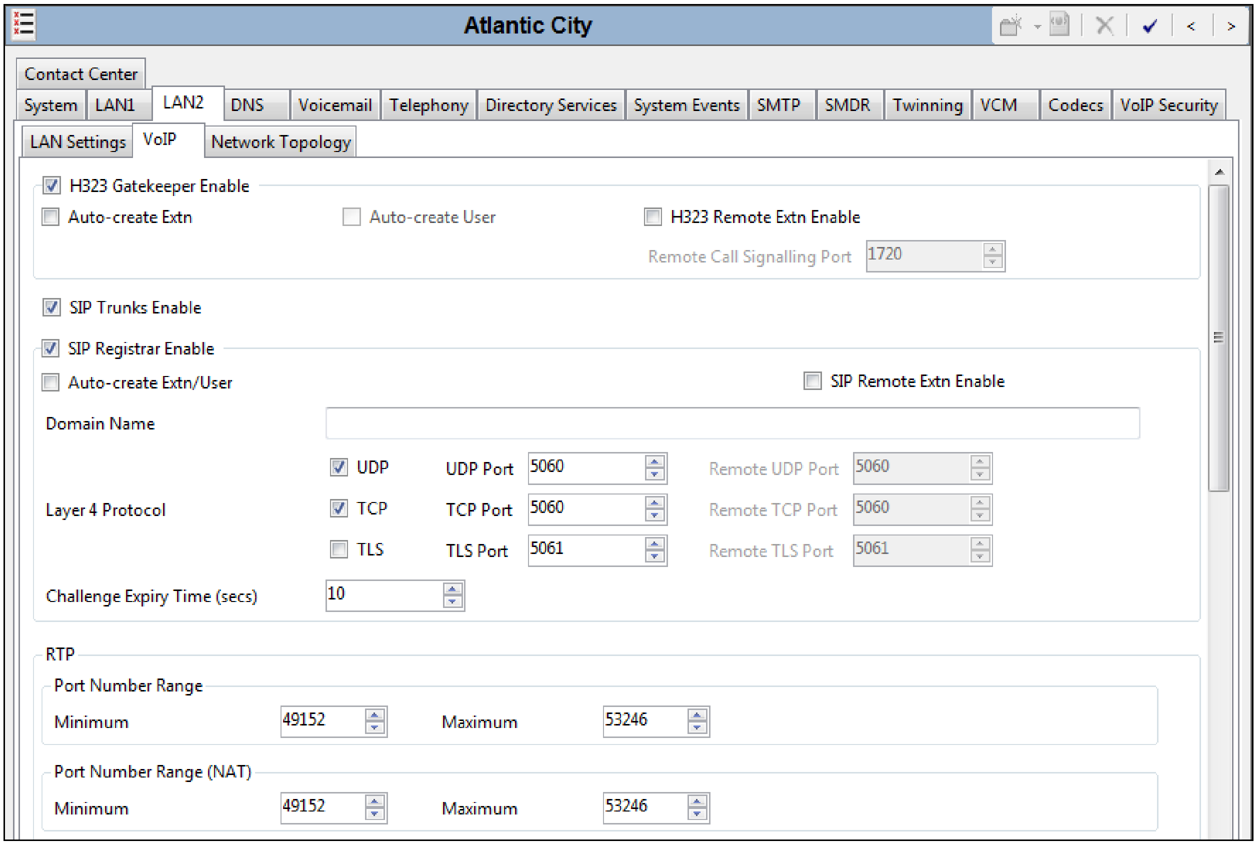
5.2.1. System – LAN2 Tab

In the sample configuration, the Avaya IP Office WAN 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 on the public network. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.



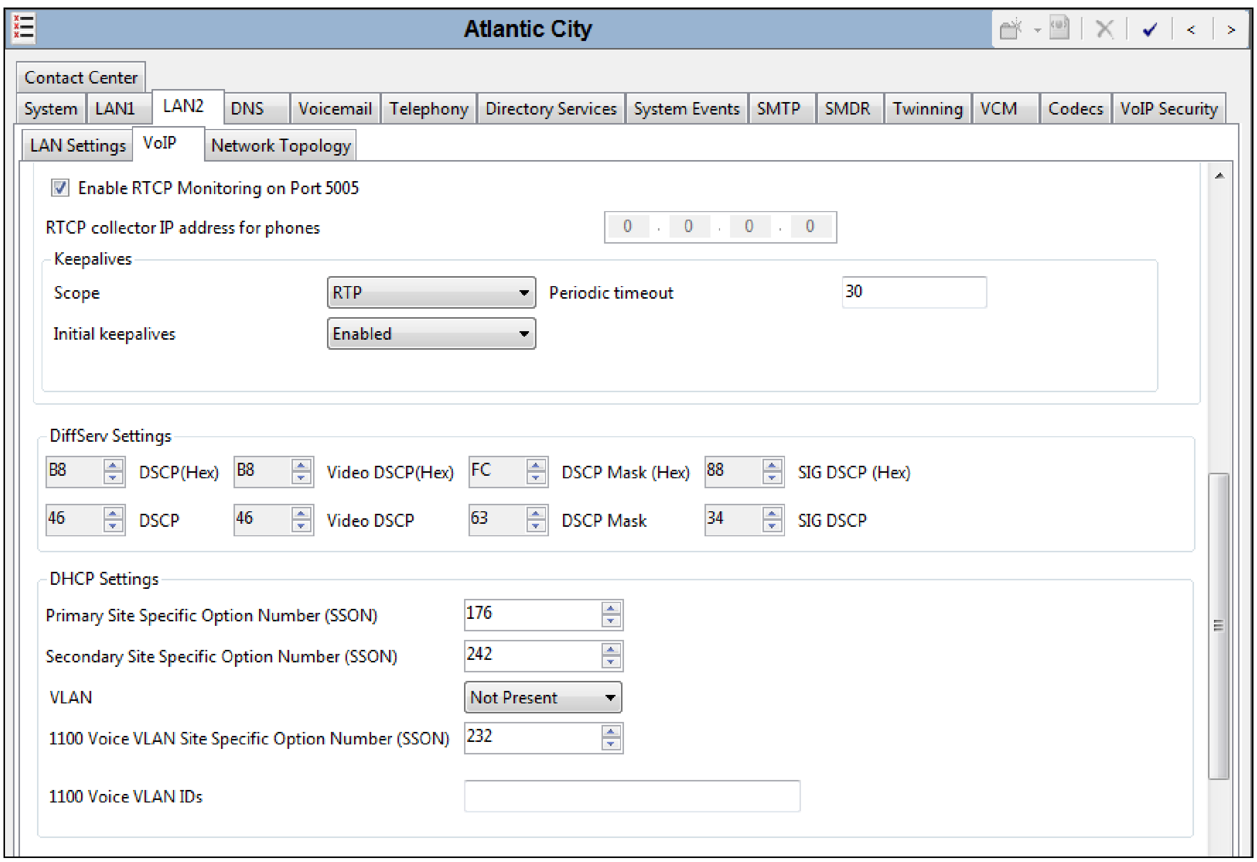
On the **VoIP** tab in the Details Pane configure the following parameters:

- 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.



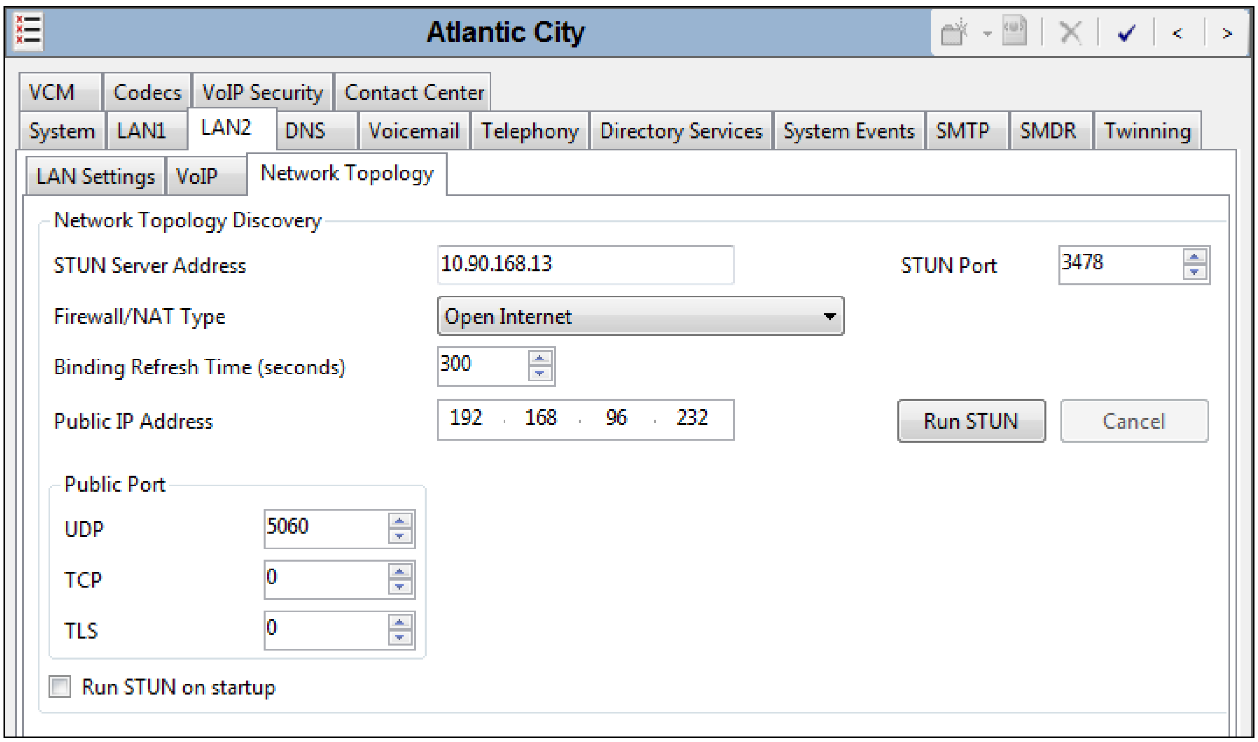
Scroll down the page.

- 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, the **DSCP** and **SIG DCSP** fields should be set to values provided by CenturyLink. CenturyLink uses a DSCP value of 40 (decimal) for media and a SIG DSCP value of 24 (decimal) for signaling.
- All other parameters should be set according to customer requirements.



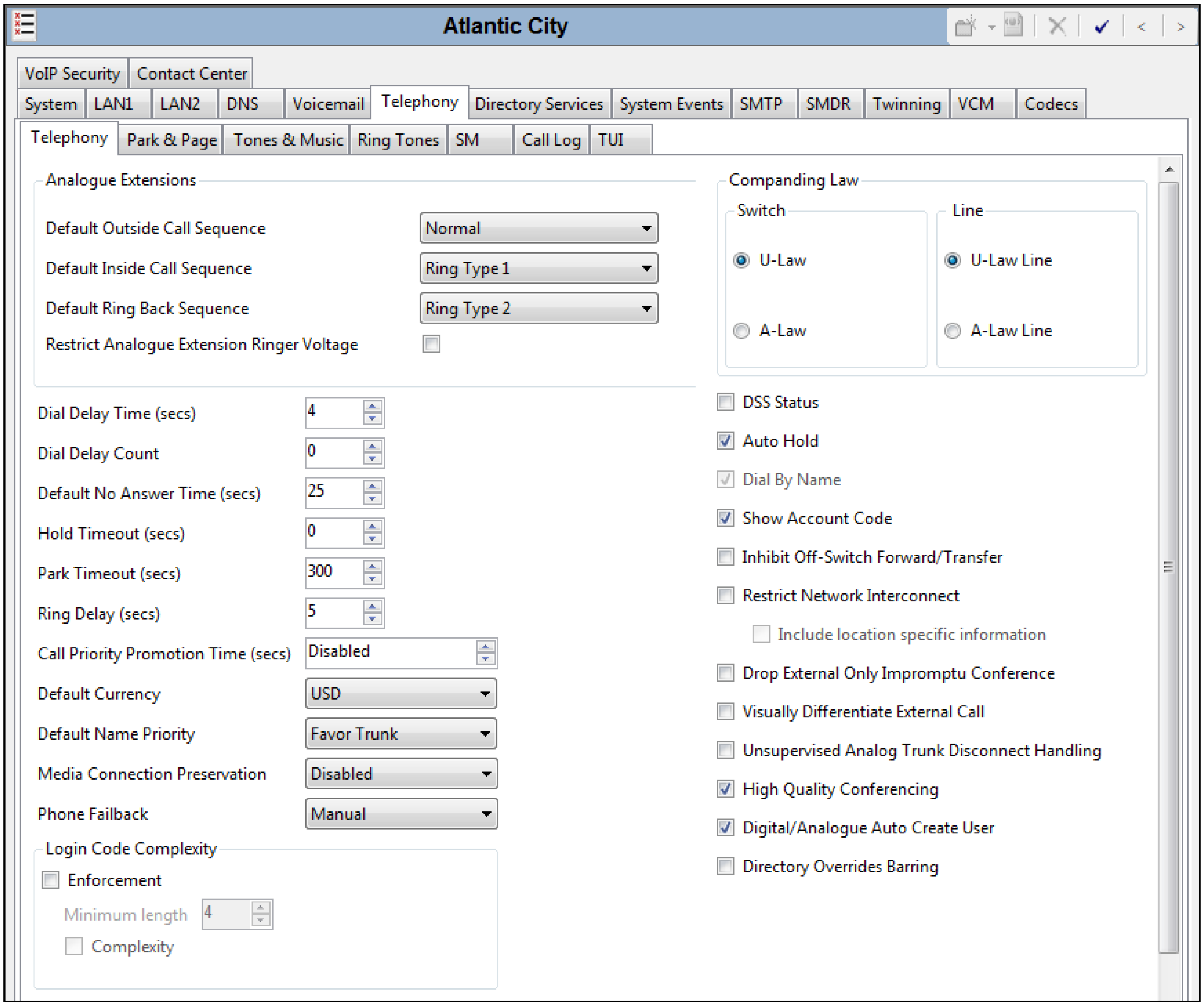
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. Since no firewall or network address translation (NAT) device was used between the Avaya IP Office and the public network, the parameter was set to **Open Internet**.
- Set **Binding Refresh Time (seconds)** to **300**. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set the **Public IP Address** to the IP address assigned to the Avaya IP Office WAN port in the **LAN2→LAN Settings** form earlier in this section.
- In the **Public Port** section, next to the transport protocol **UDP**, select the UDP port on which Avaya IP Office will listen.
- All other parameters should be set according to customer requirements.



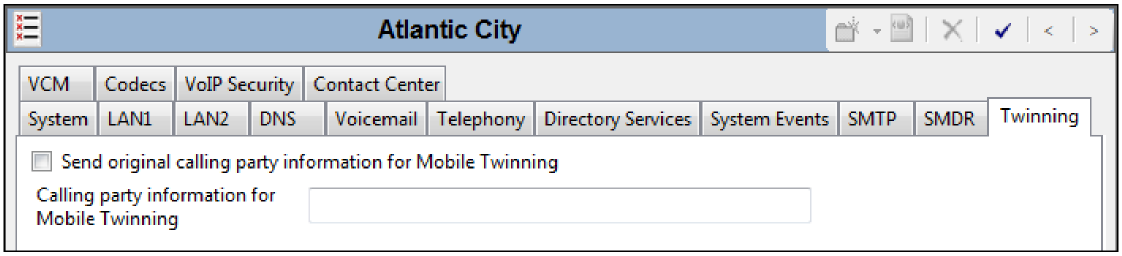
5.2.2. System - Telephony Tab

To access the System Telephony settings, navigate to the **Telephony → Telephony** tab in the Details Pane. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN. If for security reasons incoming calls should not be allowed to transfer back to the PSTN then leave this setting checked.



5.2.3. System - Twinning Tab

To view or change the System Twinning settings, navigate to the **Twinning** 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. Click the **OK** button at the bottom of the page (not shown).

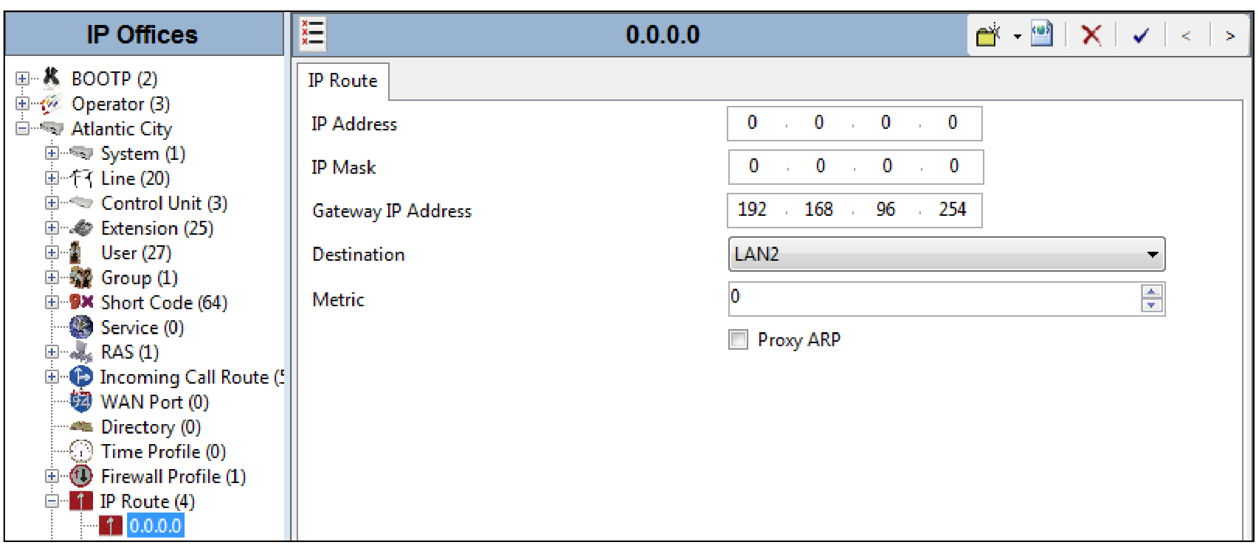


5.3. IP Route

A default route is needed so IP Office can reach other network subnets other than the one where it resides. 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 default router on the public network where Avaya IP Office is connected.
- Set **Destination** to **LAN2** from the pull-down list.

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



5.4. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the CenturyLink IQ® SIP Trunk 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 data 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:

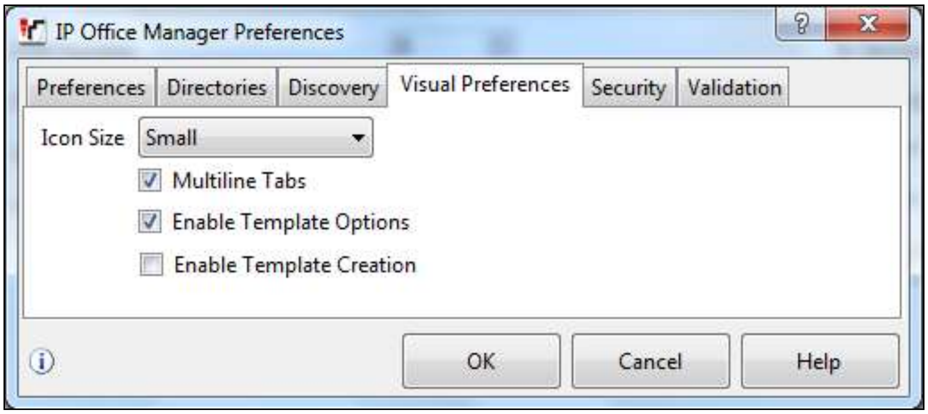
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required
- SIP Advanced
- Engineering

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**.

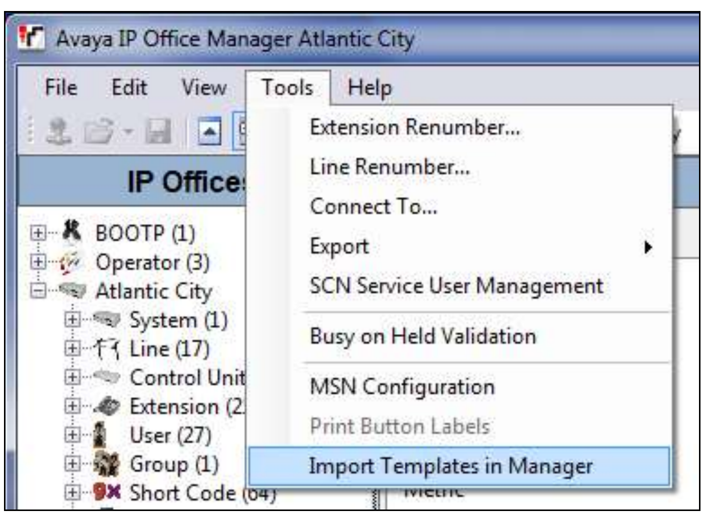
5.4.1. SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to a name with the following format **AF_<*user_supplied_text*>_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**. The example used in these Application Notes is **AF_CenturyLink-noSBCE_SIPTrunk.xml**.

2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the Visual Preferences tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

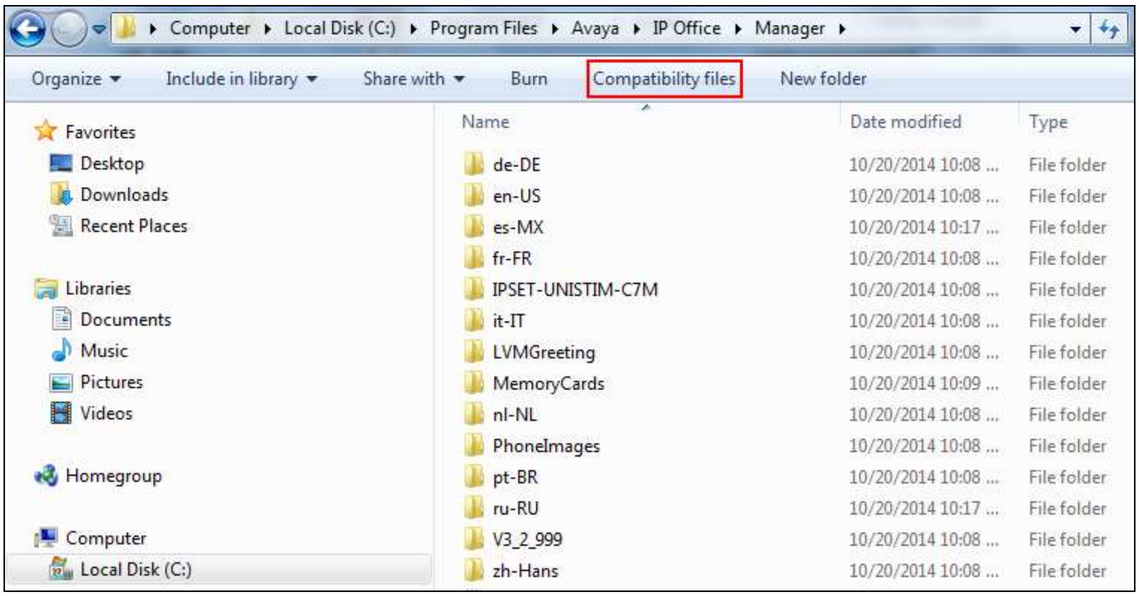


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template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

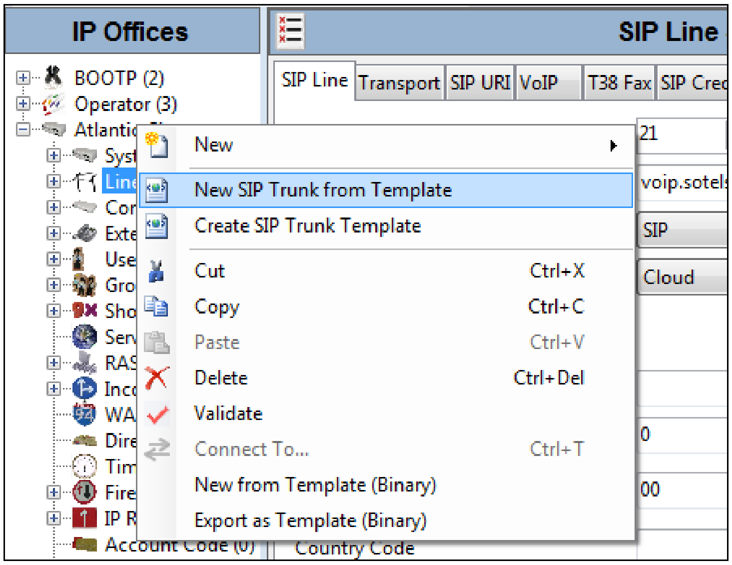


In the pop-up window that appears (not shown), select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window will appear stating success or failure (not shown). Click **OK** (not shown) to continue. If preferred, this step may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

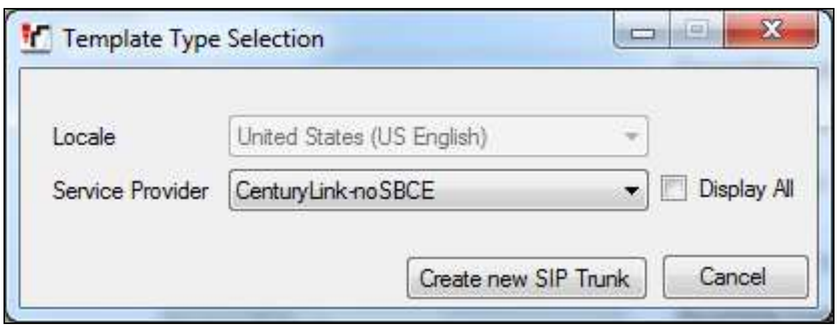
Note –Windows 7 (and later) locks the Avaya IP Office 9.1 **\Templates** directory, and it cannot be viewed. To enable browsing of the **\Templates** directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager** (or C:*Program Files (x86)*\Avaya\IP Office\Manager), and then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



4. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, and select **New SIP Trunk From Template**.



5. In the subsequent Template Type Selection pop-up window, use the **Service Provider** pull-down menu to select the XML template file created in **Step 1**. This menu will display the <*user_supplied_text*> portion of the file name of each template located in the directory. Select the entry that corresponds to the desired file name. In the case of this example, **CenturyLink-noSBCE** was selected, corresponding to a file name of **AF_CenturyLink-noSBCE_SIPTrunk.xml**. Click **Create new SIP Trunk** to finish creating the trunk.



6. 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

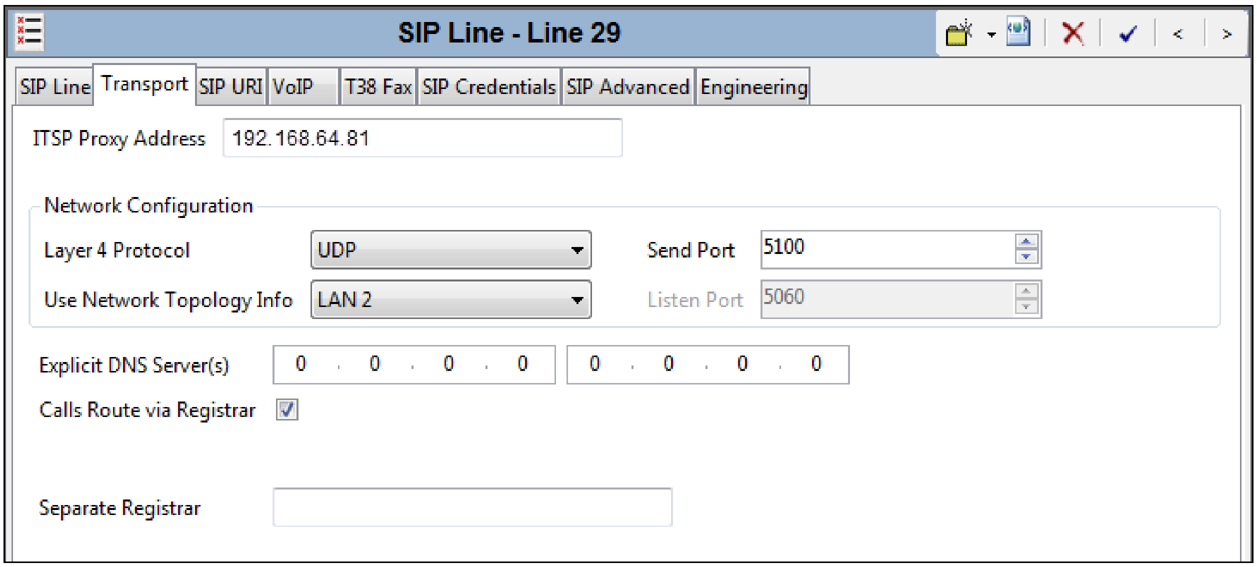
On the **SIP Line** tab in the Details Pane, configure or verify the parameters as shown below.

- Set **ITSP Domain Name** to the domain name provided by CenturyLink.
- 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 the **Refresh Method** to **Auto** and the **Timer (seconds)** to **600**. This will cause Avaya IP Office to send an UPDATE message every 300 seconds (1/2 the **Timer** value) to check the state of each active session.
- Set **Send Caller ID** to **Diversion Header**. With this setting and the related configuration in **Section 5.2.3**, Avaya IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to CenturyLink. It will also include the Diversion Header for calls that are call forwarded out the SIP Line.
- CenturyLink supports REFER except for call forward. REFER may still be enabled on Avaya IP Office; however, for call forwarding, the call will succeed but the Avaya IP Office SIP trunk will not be released until the call is terminated (**Section 2.2**). To enable REFER, under **Redirect and Transfer**, set the **Incoming Supervised REFER** field and **Outgoing Supervised REFER** field to **Always**. To disable, set these fields to **Never**.
- Default values may be used for all other parameters.

5.4.3. SIP Line - Transport Tab

Select the **Transport** tab. Set or verify the parameters as shown below.

- Set the **ITSP Proxy Address** to the IP address of the CenturyLink SIP proxy provided by CenturyLink.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to the network port used by the SIP line to access the far-end and configured in **Section 5.2.1**.
- Set the **Send Port** to **5100**. This port number was provided by CenturyLink.
- Default values may be used for all other parameters.

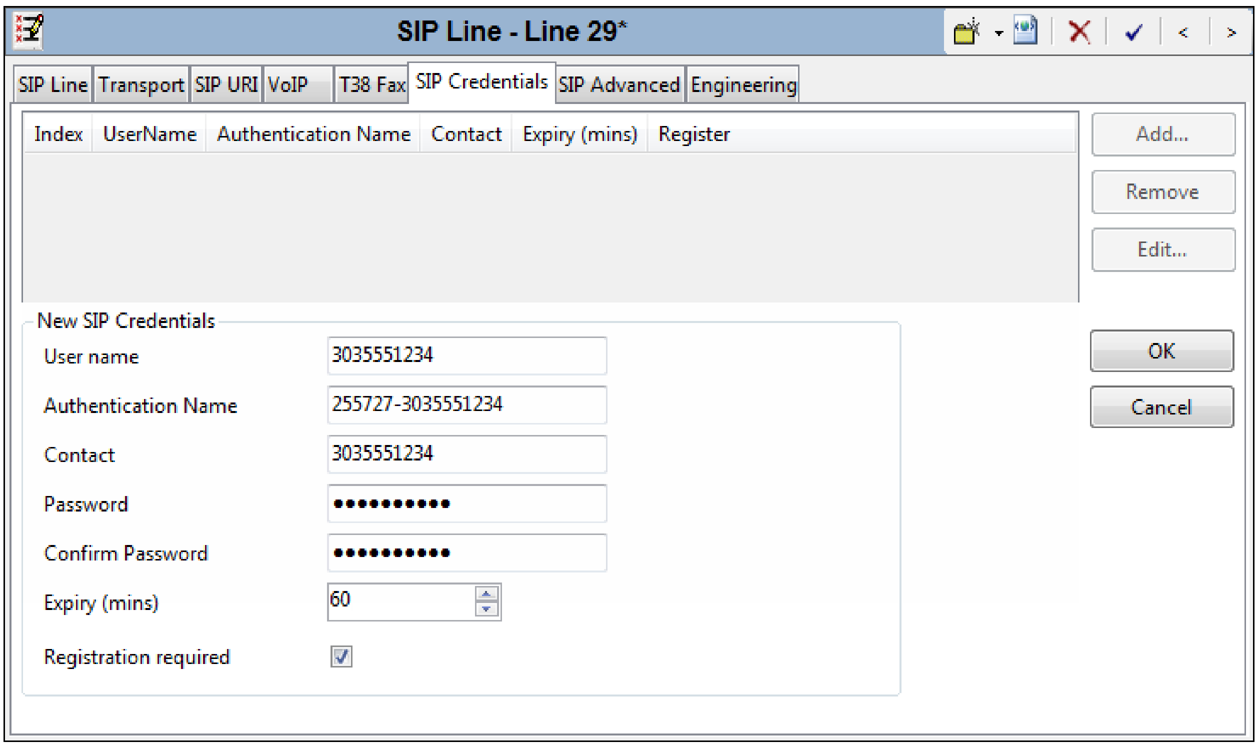


5.4.4. SIP Line – SIP Credentials

SIP Credentials are provided by CenturyLink for registration and authentication of the enterprise. 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 the **User name** and **Contact** to the pilot number provided by CenturyLink.
- Set the **Authentication Name** to the user name provided by CenturyLink.
- Set the **Password** and **Confirm Password** to the password provided by CenturyLink.
- Set **Expiry (mins)** to **60**. This is the value recommended by CenturyLink.
- Check the **Registration required** box.

Click **OK**.

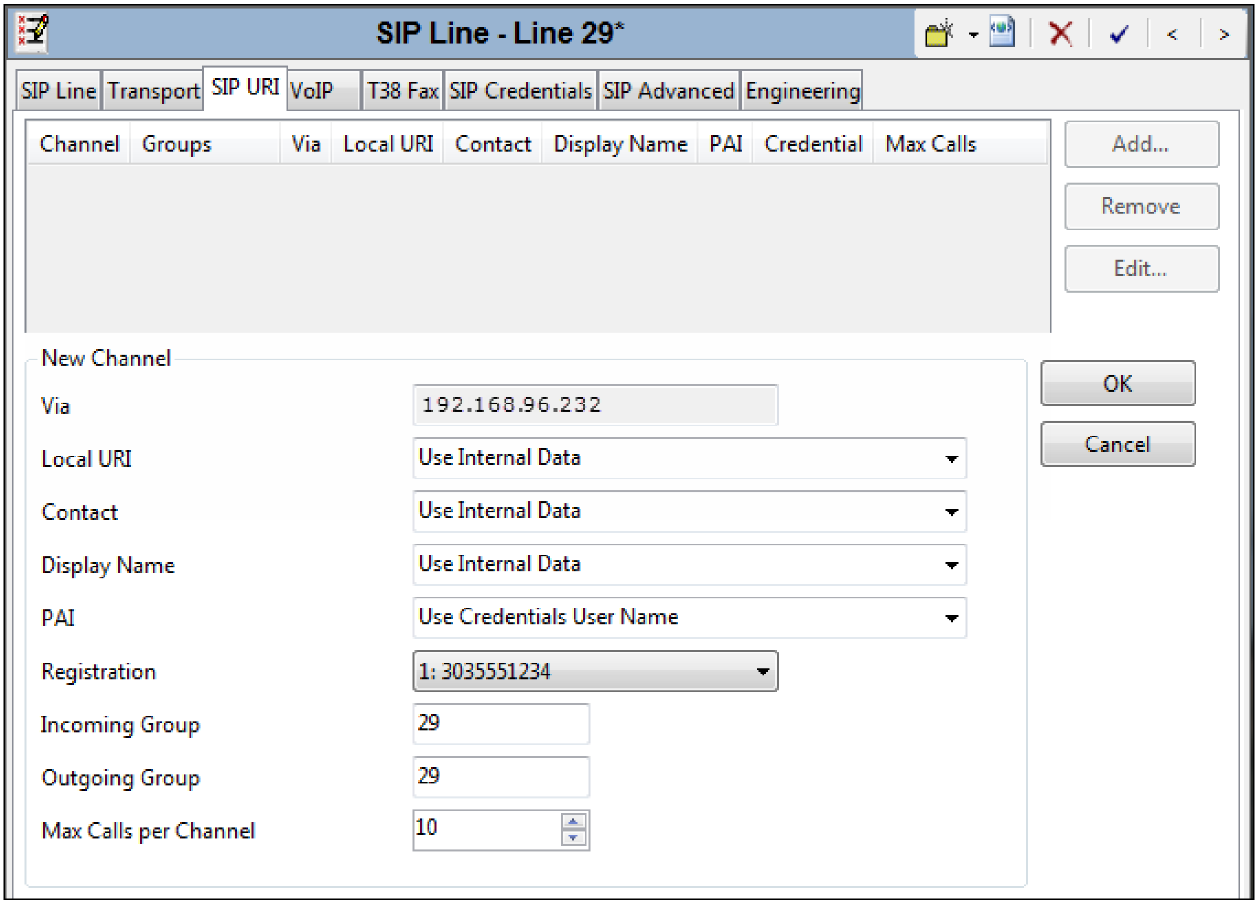


5.4.5. SIP Line - SIP URI Tab

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit** button. In the example screen below, a new entry is created. The entry was created with the parameters shown below:

- Set **Local URI** to **Use Internal Data**. The **Local URI** parameter controls which calls will be accepted by the system on this SIP URI and also defines the contents of the From header for outbound calls. The setting of **Use Internal Data** will enable the **User → SIP** tab for each Avaya IP Office user where the SIP information can be configured. The setting of **Use Internal Data** will allow inbound calls whose user portion of the incoming Request-URI matches a value configured on the system for a user (**User → SIP**), hunt group (**Hunt Group → SIP**) or voicemail (**System → Voicemail**). For outbound calls, the From header is populated with the **SIP Name** configured for the user (**Section 5.7**).
- Set **Contact**, and **Display Name** to **Use Internal Data**. This setting will populate each of these headers with the corresponding value from the **User → SIP** tab of the Avaya IP Office user involved in the call. See **Section 5.7**.
- Set **PAI** to **Use Credentials User Name**. CenturyLink requires the pilot number to appear as the user part of the URI in the PAI header. The pilot number is configured as the SIP Credentials User Name in **Section 5.4.4**.
- For the **Registration** field, select the credentials created in **Section 5.4.4** from the pull-down menu.
- 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 in **Section 5.8.1**. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining ARS entries for routing outbound traffic to this line in **Section 5.5**. For the compliance test, a new incoming and outgoing group **29** was defined that only contained this line (line 29).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

Click **OK**.

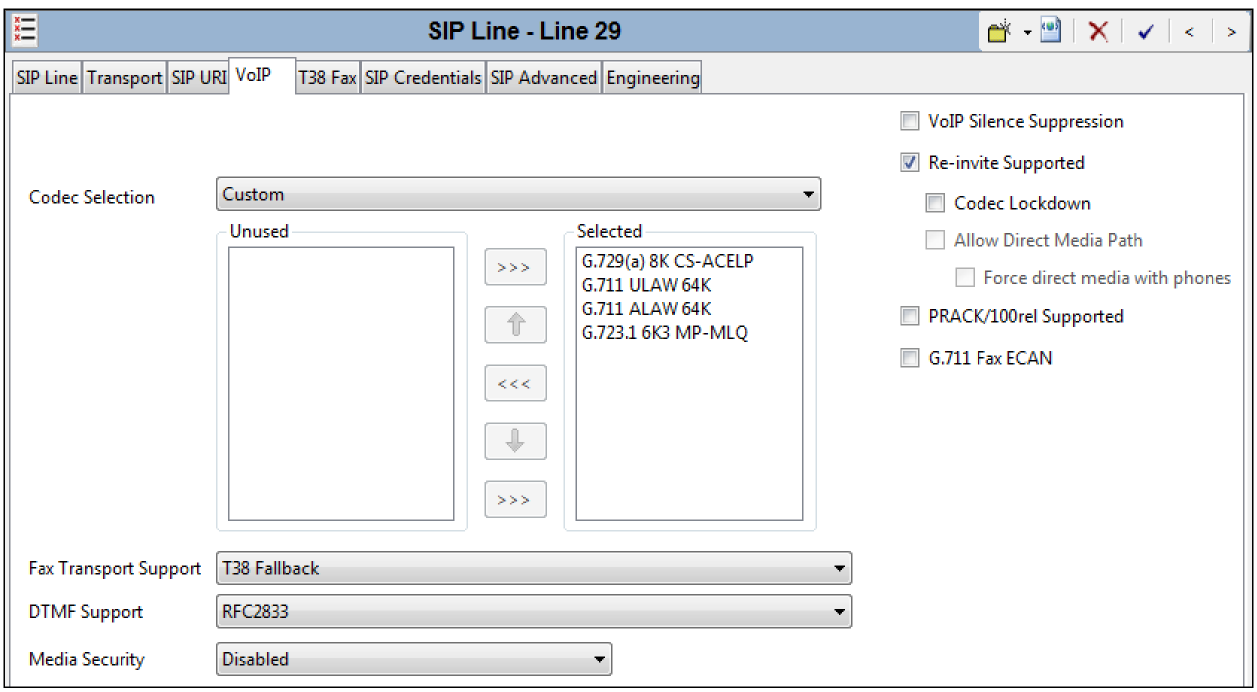


Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user, such as a short code. These URIs are created in the same manner as shown above with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, and **Display Name**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

5.4.6. SIP Line - VoIP Tab

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

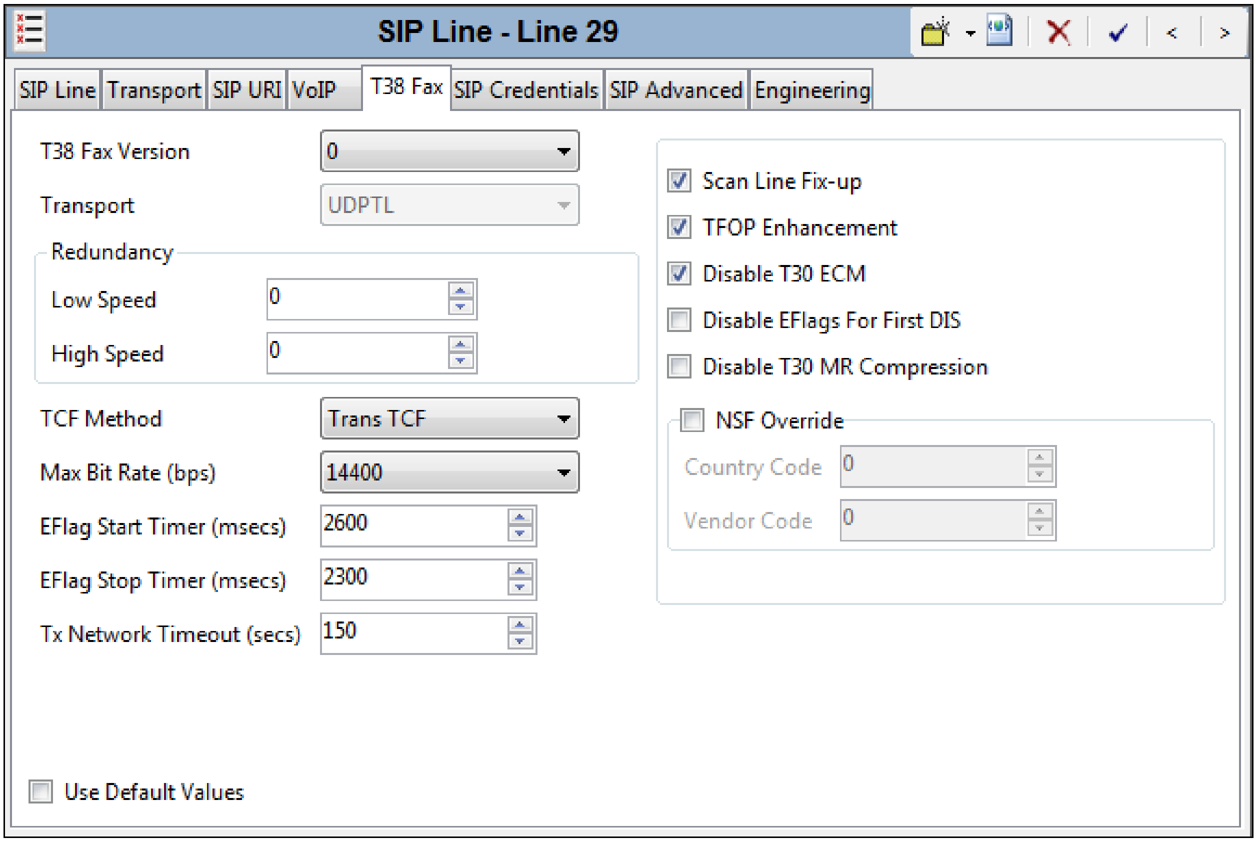
- For **Codec Selection**, select **Custom** from the pull-down menu to use a custom list of codecs. Next, move unwanted codecs from the **Selected** column to the **Unused** column if needed. Lastly, move the codecs up or down the list in the **Selected** column to achieve the desired order of preference. The example below shows the codecs used for the compliance test. This codec order was used to match the codec order used by CenturyLink as close as possible.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Set the **Fax Transport Support** to **T38 Fallback**. In general, CenturyLink supports T.38 fax but not necessarily on all media gateways in the network. Using the **T38 Fallback** setting will allow all fax calls to succeed, though some may use G.711 fax instead of T.38.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters.



5.4.7. SIP Line – T38 Fax Tab

Select the **T38 Fax** tab. Set the parameters as shown below.

- Uncheck the **Use Default Values** box at the bottom of the page.
- Set the **T38 Fax Version** to **0**.
- Check the **Disable T30 ECM** box.

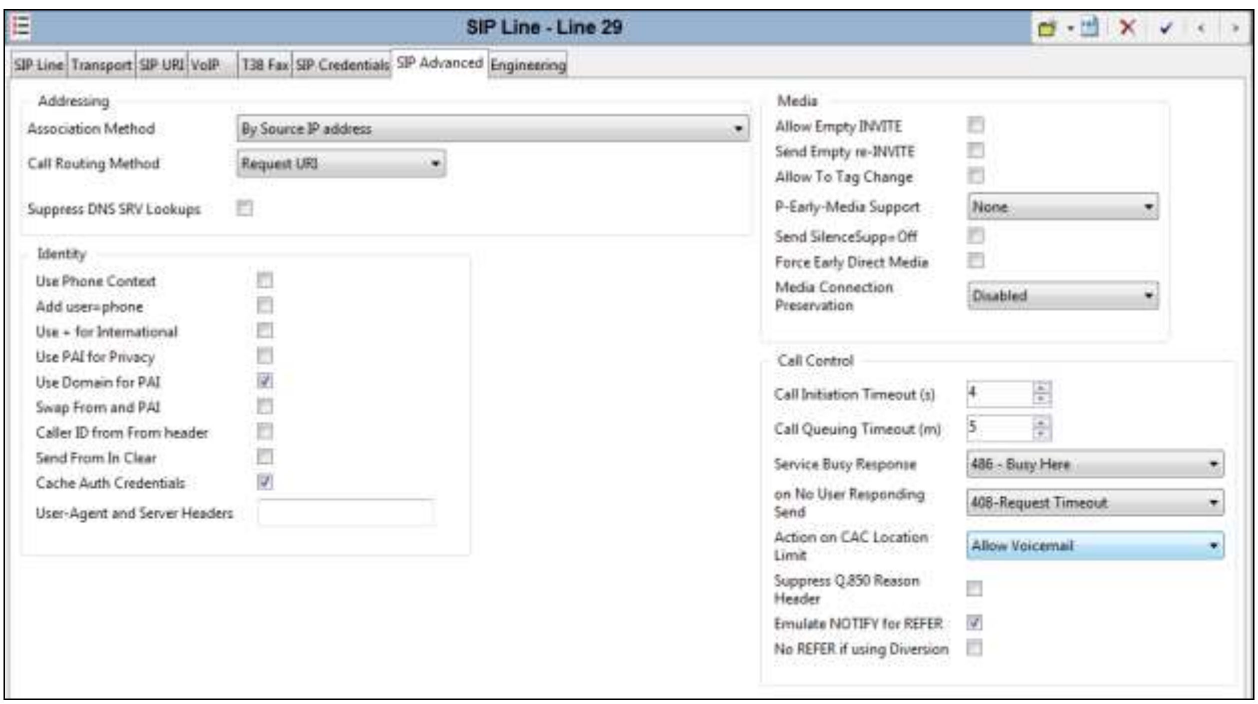


5.4.8. SIP Line – SIP Advanced

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

- Check the **Use Domain for PAI** box. CenturyLink requires the CenturyLink SIP domain to appear in URI of the PAI header. Avaya IP Office sends the public IP address of the Avaya IP Office in the PAI header even if a SIP domain in configured on the SIP Line (**Section 5.4.2**). To change this behavior, the **Use Domain For PAI** needs to be enabled. Without this setting, outbound privacy calls will fail and the calling party number for call forwarding to the PSTN and twinning will incorrectly show the pilot number instead of the original PSTN caller.
- Check the **Emulate NOTIFY for REFER** box. With REFER enabled (**Section 5.4.2**), 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).



5.5. ARS

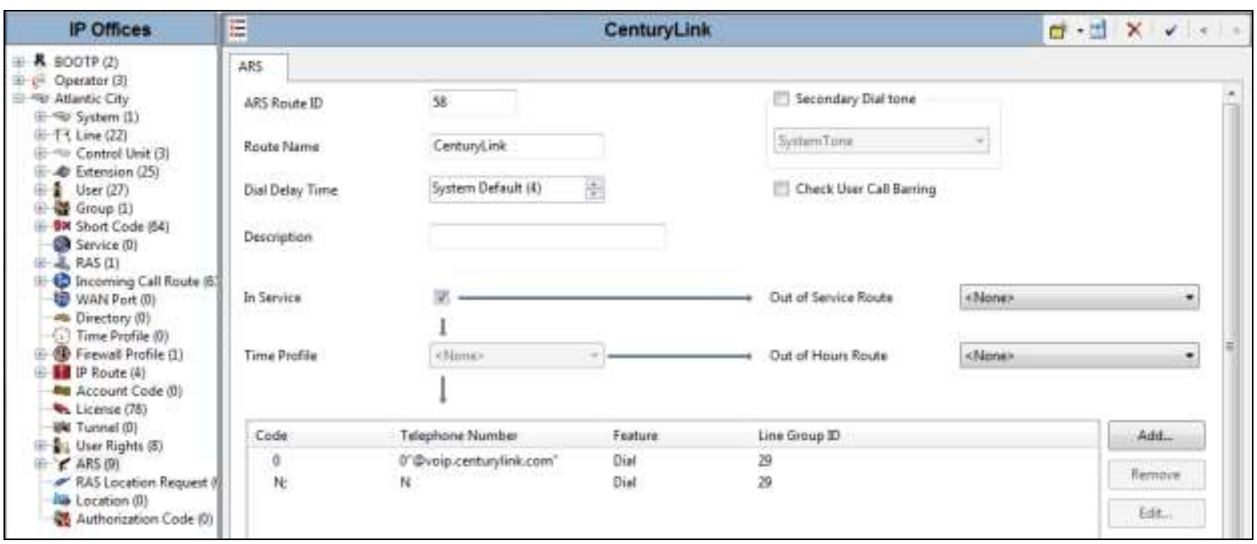
ARS is used to route outbound traffic to the SIP line. To define a new ARS route, right-click **ARS** in the Navigation pane and select **New**. In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

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 5.6** 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 SIP domain provided by CenturyLink and used to configure the trunk in **Section 5.4.2**. Adding the domain in this field was required to ensure that the correct domain appeared in the outbound Request-URI header when dialing 0. This was not required when matching on any other dialed number as shown next.
- 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).

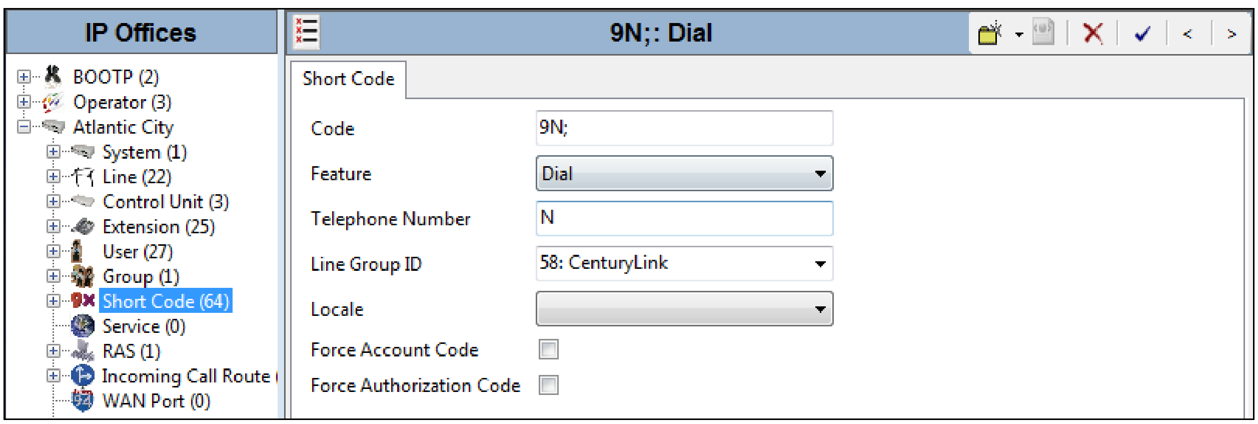


5.6. Short Codes

A short code is a dial pattern that triggers a specific function. A short code is used by the caller to route outbound traffic to ARS. 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.

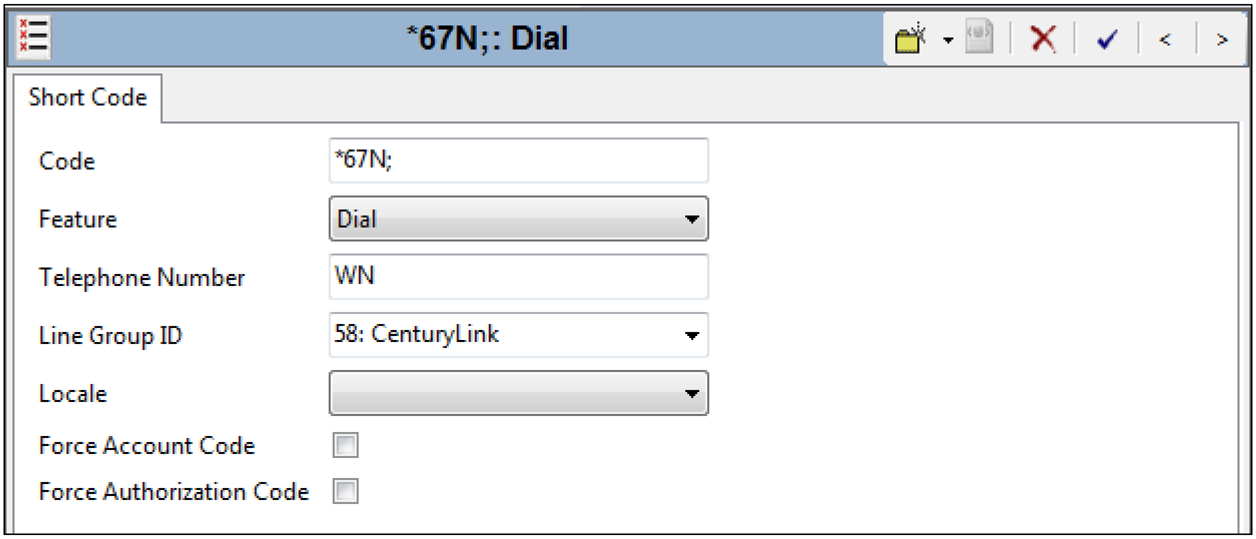
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;** was used. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to **Dial.** This is the action that the short code will perform.
- Set **Telephone Number** to **N**. The value **N** represents the number dialed by the user after removing the **9** prefix. This value is passed to ARS.
- Set the **Line Group ID** to the ARS route to be used (**Section 5.5)**.

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



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".

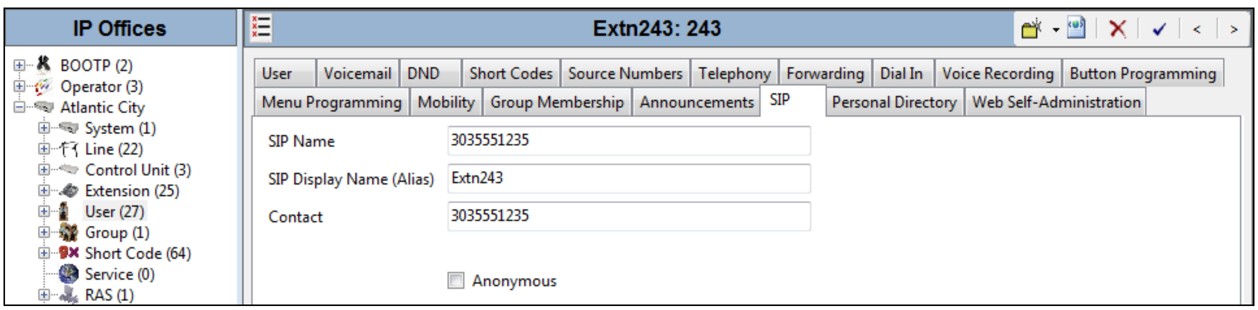
In the case of the SIP Line documented in these Application Notes, when a user dials *67 plus the number, Avaya IP Office will include the calling number in the P-Asserted-ID (PAI) header and will include the Privacy: Id header.



5.7. User

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 → *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 values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls and allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.4.5**). The example below shows the settings for user **Extn243**. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from CenturyLink. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all 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.

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



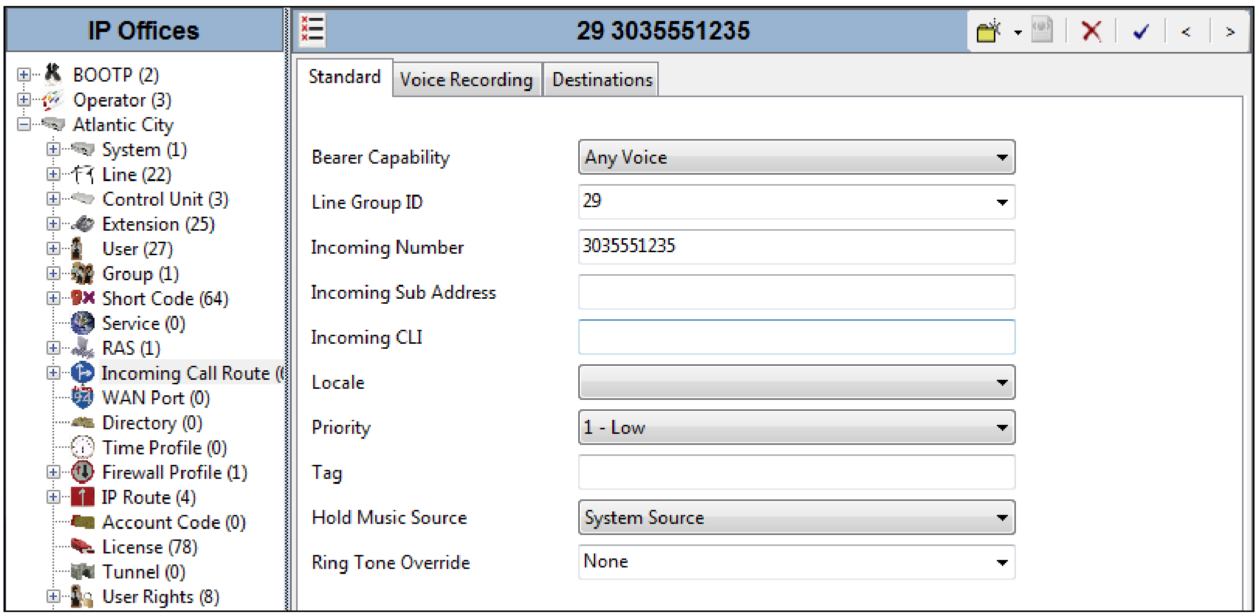
5.8. Incoming Call Route

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 Routes** in the Navigation Pane and select **New**.

5.8.1. Incoming Call Route – Standard Tab

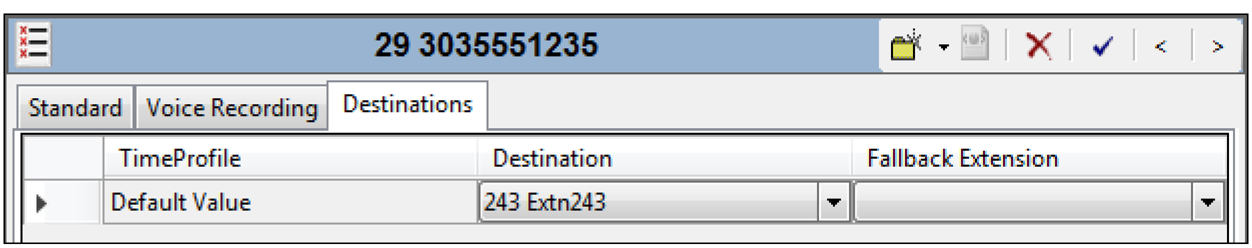
On the **Standard** tab of the Details Pane, enter the parameters as shown below.

- Set the **Bearer Capability** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4.5**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Default values can be used for all other fields.



5.8.2. Incoming Call Route – Destinations Tab

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 3035551235 on line 29 are routed to extension 243.

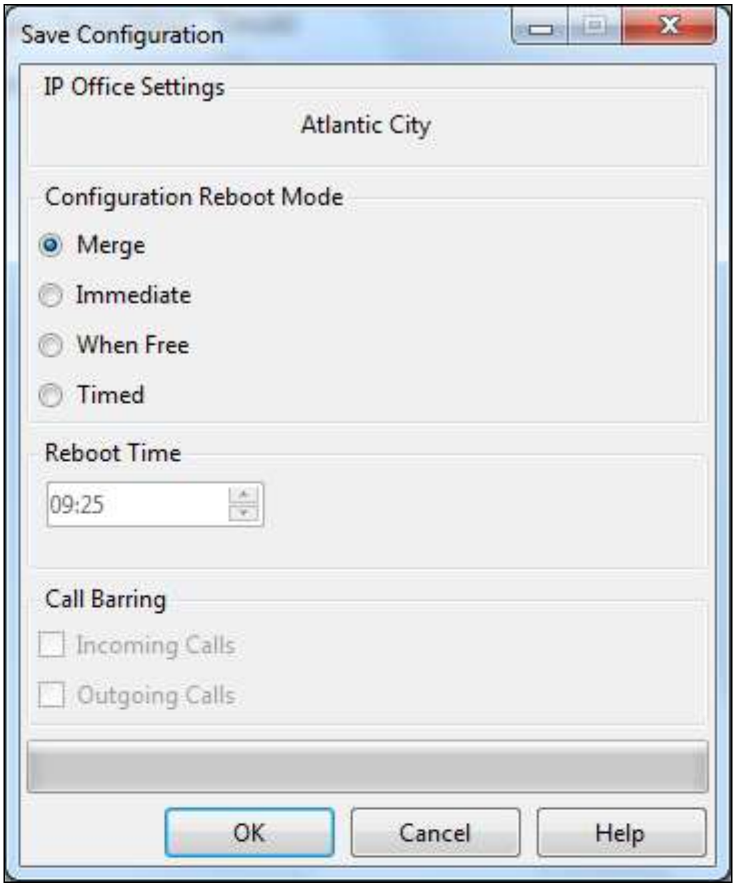


Incoming Call Routes for other direct mappings of DID numbers to Avaya IP Office users listed in **Figure 1** are omitted here, but can be configured in the same fashion.

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.



6. CenturyLink IQ® SIP Trunk Service Configuration

CenturyLink is responsible for the configuration of the CenturyLink IQ® SIP Trunk Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. In the case of the compliance test, this is the public IP address of the Avaya IP Office WAN port. CenturyLink will provide the customer the necessary information to configure Avaya IP Office including:

- CenturyLink SIP proxy IP address
- Transport protocol and port
- SIP domain
- SIP credentials: user name and password
- Pilot number
- DID numbers

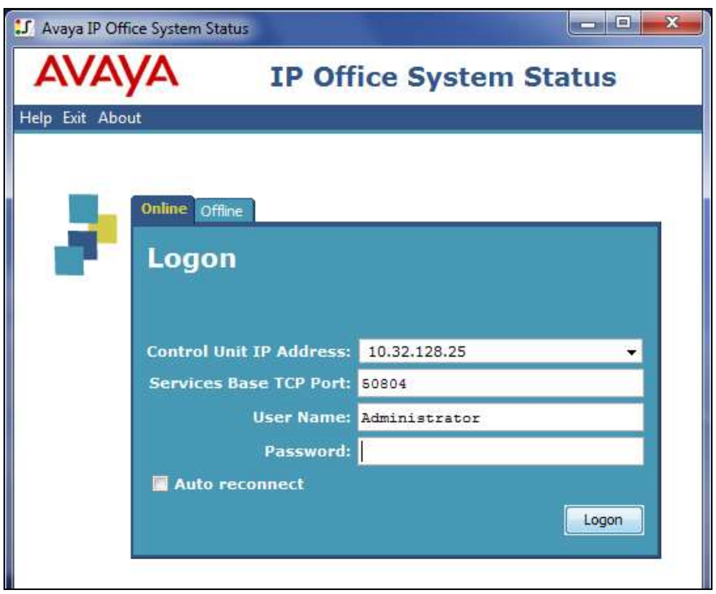
7. Verification Steps

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

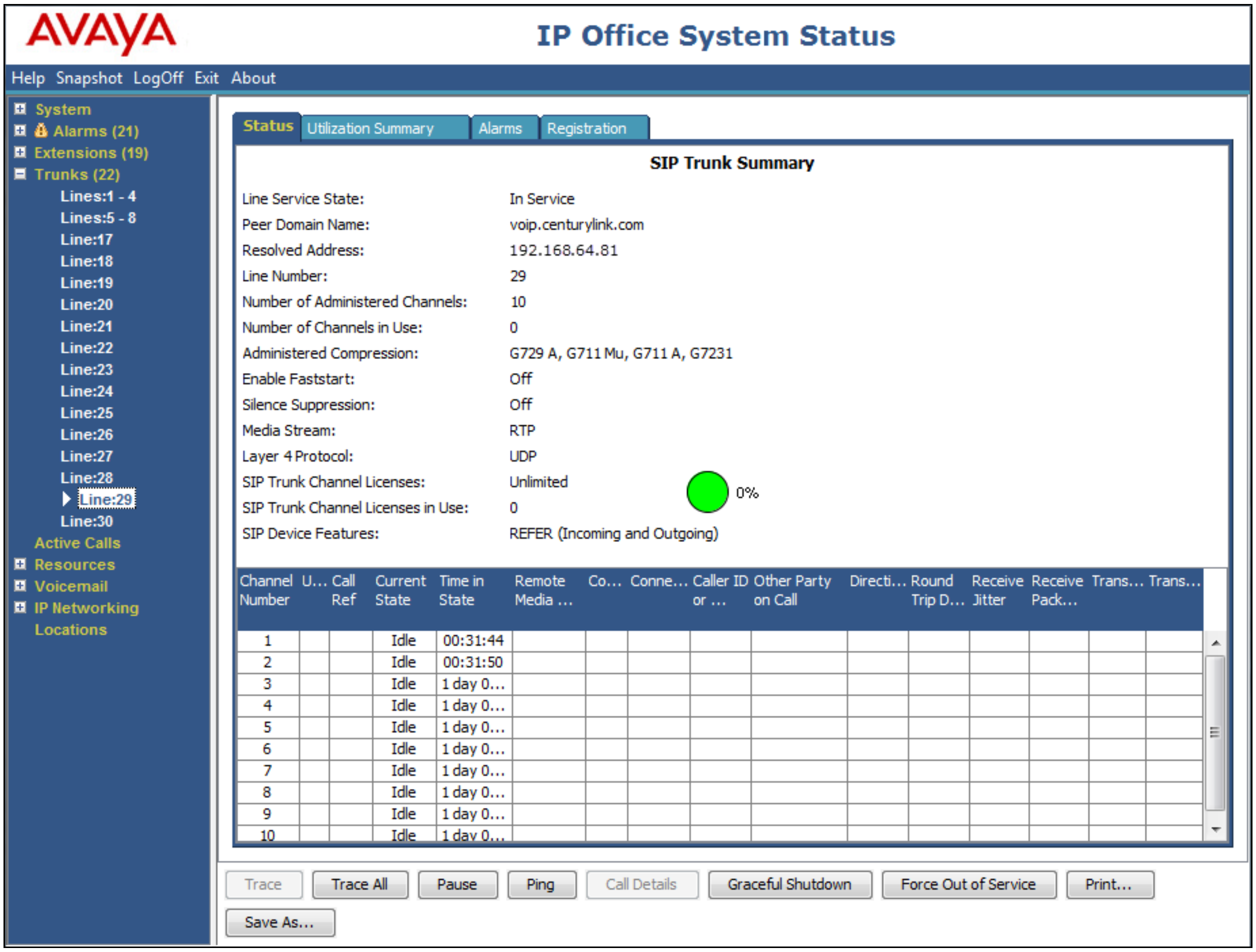
7.1. System Status

The System Status application is used to monitor and troubleshoot Avaya IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → All Programs → IP Office → System Status**.

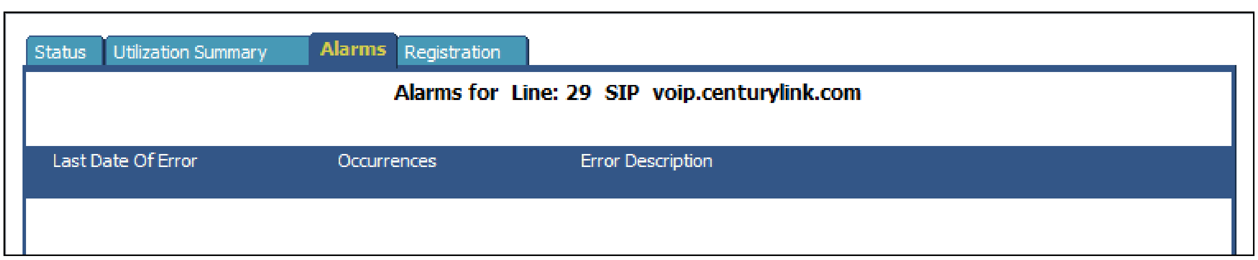
The following screen shows an example **Logon** screen. Enter the Avaya IP Office private IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current Stat**e is **Idle** for each channel.



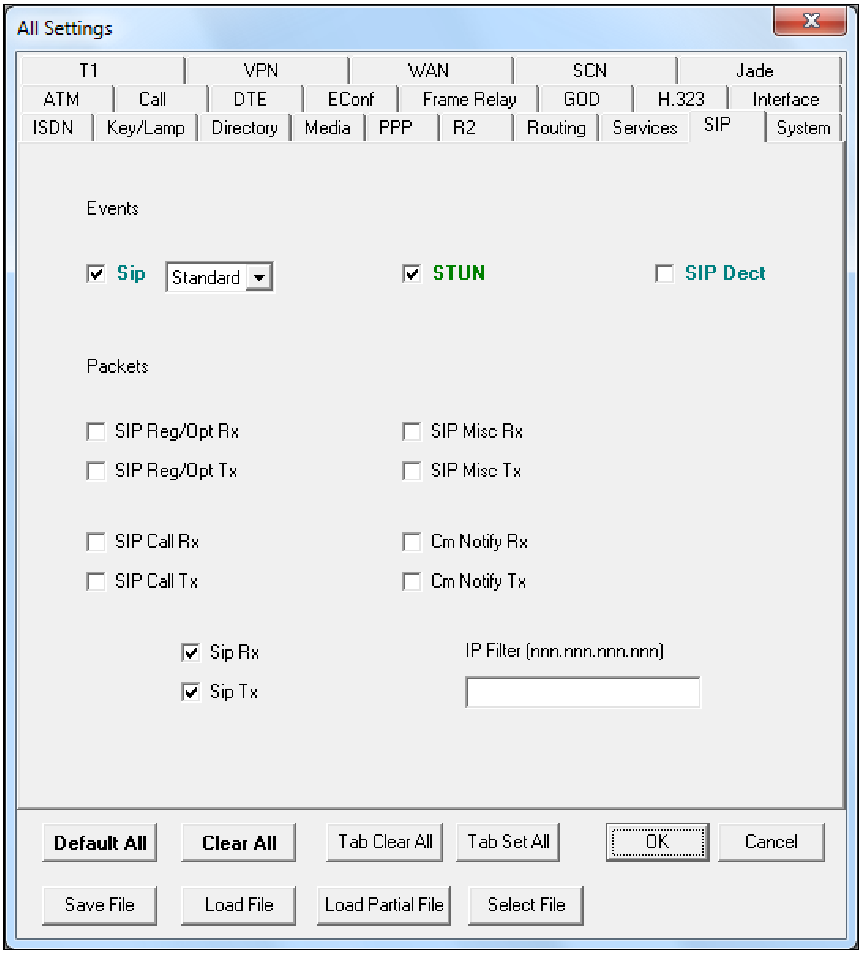
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot Avaya IP Office. Monitor can be accessed from **Start → All Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select **Filters → Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring.



8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office 9.1 to the CenturyLink IQ® SIP Trunk Service. The CenturyLink IQ® SIP Trunk Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks. The CenturyLink IQ® SIP Trunk Service passed compliance testing. Please refer to **Section 2.2** for any observations/exceptions.

9. Additional References

This section references documentation relevant to these Application Notes. In general, Avaya product documentation is available at http://support.avaya.com.

[1] *Deploying Avaya IP Office Platform IP500 V2*, Document Number 15-601042, Issue 30l, August 12, 2015.

[2] *Administering Avaya IP Office Platform with Manager*, Issue 10.21, August 2015.

[3] *Using System Status*, Document Number 15-601758, Issue 10f, August 11, 2015.

[4] *Administering Avaya IP Office Voicemail Pro*, Document Number 15-601063, Issue 10f, July 10, 2015.

[5] *Using IP Office System Monitor*, Document Number 15-601019, Issue 06e, May 19, 2015.

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

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