Avaya Solution & Interoperability Test Lab

Application Notes for Configuring CenturyLink SIP Trunk Service using Sonus with Avaya IP Office 9.0.3 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider CenturyLink using Sonus switch and Avaya IP Office 9.0.3

CenturyLink SIP Trunk Service (CenturyLink) 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. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between CenturyLink and Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 9.0.3, Avaya embedded Voicemail, Avaya IP Office Softphone, Avaya Flare® Experience for Windows, Avaya H.323, Avaya SIP, digital and analog endpoints.

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

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office connecting to CenturyLink.
This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**. Note: NAT devices added between Avaya IP Office and the CenturyLink network should be transparent to the SIP signaling.

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

A simulated enterprise site with Avaya IP Office was connected to CenturyLink. To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls from/to the Avaya IP Office Softphone/ Avaya Flare® Experience for Windows (SIP).
- Inbound and outbound long hold time call stability.
- Various call types including: local, long distance, international call, inbound toll-free, outbound toll-free, operator assisted call, 411 and 911 services.
- Codec G.729A and G.711U.
- Caller number/ID presentation.

- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- Telephony features such as hold and resume, transfer, and conference.
- FAX using T.38 and G.711 pass-through.
- Off-net call forwarding (CenturyLink supports Diversion Header).
- Twinning to mobile phones on inbound calls.

2.2. Test Results

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

- **CenturyLink does not send OPTIONS but response to IP Office OPTIONS with 200OK**. CenturyLink will send OPTIONS if customer request/order it.
- **Blind Call Transfer to PSTN using Avaya 1140E SIP phone does not complete until transferee picks up the call** – Call scenario is when PSTN phone calls to Avaya 1140E SIP phone, Avaya 1140E SIP phone answers the call and performs blind transfer to another PSTN endpoint. The expected behavior of Avaya 1140E SIP phone is after transfer, the phone should display "transfer completed". But in this case, user presses "transfer" button, answers question of "Consultative transfer with party ?" with "No", which implies the blind transfer, as the transferee PSTN phone is ringing, the Avaya 1140E SIP phone should be released and display "transfer successfully". Instead, the Avaya 1140E SIP phone still displays "transferring" and not released until the transferee PSTN phone answers the call. The work around is to hang up the Avaya 1140E SIP phone. This is very minor known limitation on Avaya 1140E SIP phone. There is no user impact. Transfer is still completed with 2 ways audio.

2.3. Support

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

For technical support on the CenturyLink system, please visit: http://www.centurylink.com

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to CenturyLink through the public IP network. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

Located at the enterprise site is an Avaya IP Office 500v2 with the MOD DGTL STA16 expansion module which provides connections for 16 digital stations to the PSTN, and the extension PHONE 8 card which provides connections for 8 analog stations to the PSTN as well as 64-channel VCM (Voice Compression Module) for supporting VoIP codecs. The voicemail service is embedded on Avaya IP Office. The LAN2 port of Avaya IP Office is connected to the public IP network.

Endpoints include Avaya 9600 Series IP Telephone (with H.323 firmware), Avaya 1100 Series IP Telephone (with SIP firmware), Avaya 1408D Digital Telephones, Avaya Analog Telephone, Avaya Flare® Experience for Windows, and Avaya IP Office Softphone H.323. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer Avaya IP Office.

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

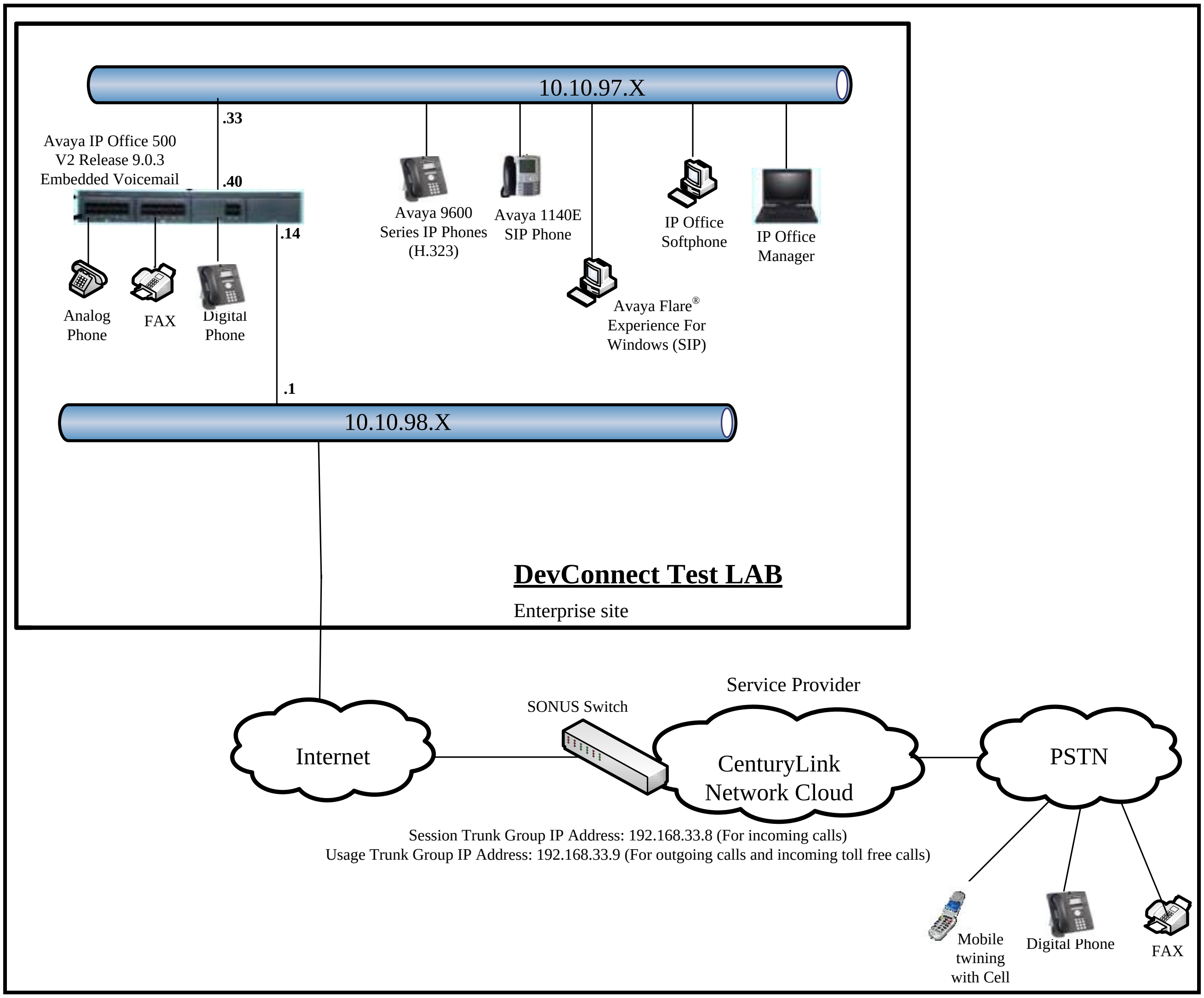


Figure 1: Test Configuration for Avaya IP Office with CenturyLink SIP Trunk Service

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 5 + N digits to send digits across the SIP trunk to CenturyLink. The short code of 5 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to CenturyLink. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office would send 11 digits in the Request URI and the To field of an outbound SIP INVITE message. It was configured to send 10 digits in the From field. For inbound calls, CenturyLink sent 10 digits in the Request URI and the To field 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 Avaya IP Office such as a session border controller or data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the service provider and the Avaya IP Office must be allowed to pass through these devices.

4. Equipment and Software Validated

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

Avaya Telephony Components	
Equipment	**Release**
Avaya IP Office 500v2	9.0.3 (Build 941)
Avaya IP Office DIG DCP*16 V2	9.0.3 (Build 941)
Avaya IP Office Ext Card Phone 8	9.0.3 (Build 941)
Avaya IP Office Manager	9.0.3 (Build 941)
Avaya 1140E IP Telephone (SIP)	04.04.12.00
Avaya IP 9640G	S3.2
Avaya IP 9630	S3.2
Avaya IP Office Softphone	3.2.3.49 68975
Avaya Flare® Experience for Windows (SIP)	1.1.4.23 SP4
Avaya Digital Telephone (1408D)	N/A
Avaya Symphony 2000 Analog Telephone	N/A
HP Officejet 4500 (fax)	N/A
CenturyLink Components	
Equipment	**Release**
Sonus NBS	7.3.7

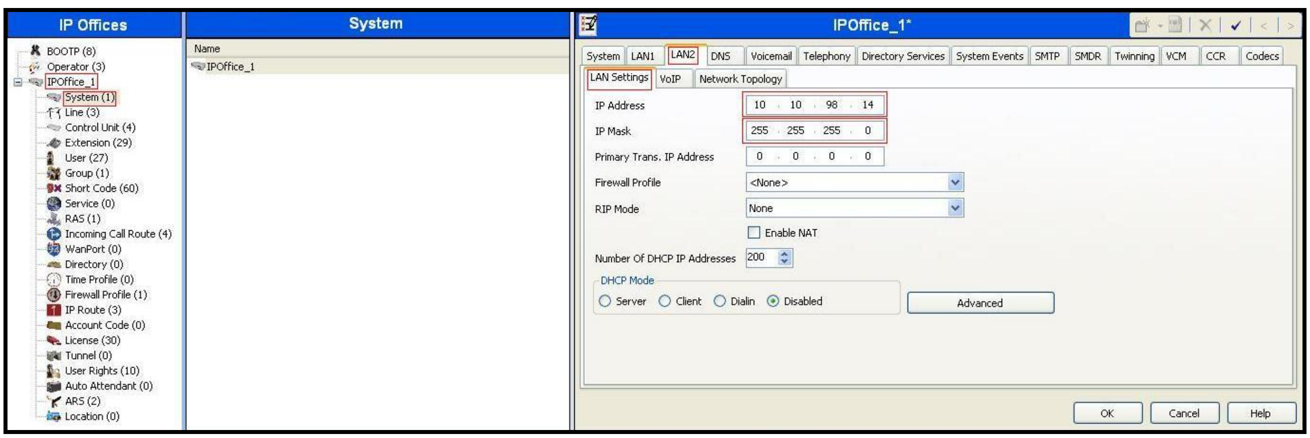
Note: Testing was performed with IP Office 500 v2 R9.0.3, but it also applies to IP Office Server Edition R9.0.3. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0.3 to support analog or digital endpoints or trunks. IP Office Server Edition does not support TAPI Wave or Group Voicemail.

5. Configure IP Office

This section describes the Avaya IP Office configuration to support connectivity to CenturyLink. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one shown in the next section. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interface with the service provider (such as the LAN interface to the enterprise site and IP Office Softphone support) is assumed to be already in place.

5.1. LAN2 Settings

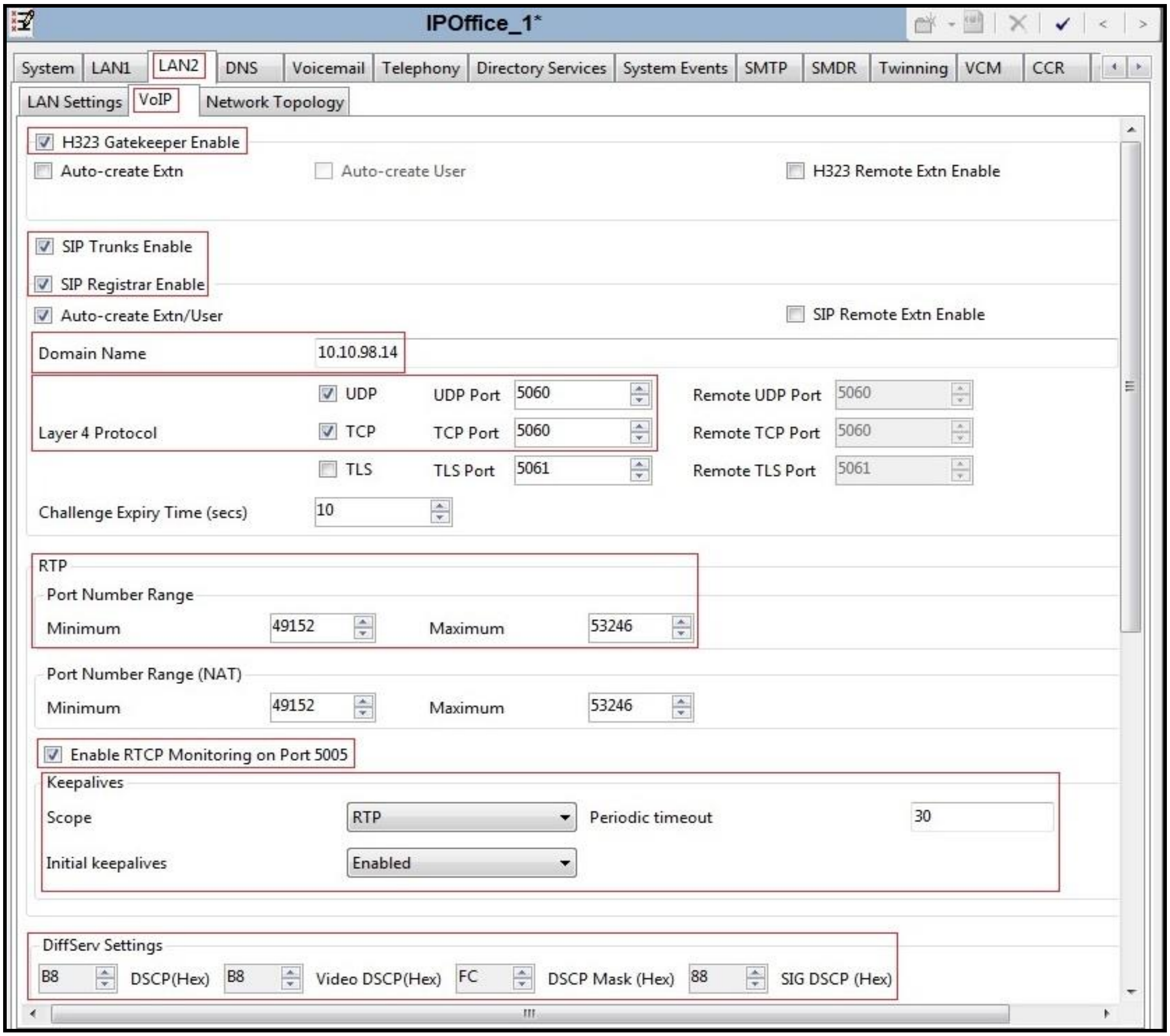
In the sample configuration, the **IPOffice_1** was used as the system name and the LAN2 port was used to connect to CenturyLink. To access the LAN2 settings, first navigate to **System (1) → IPOffice_1** in the Navigation and Group Panes and then navigate to the **LAN2→ LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office LAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements. Click **OK** to submit the change.



The **VoIP** tab as shown in the screenshot below was configured with following settings.

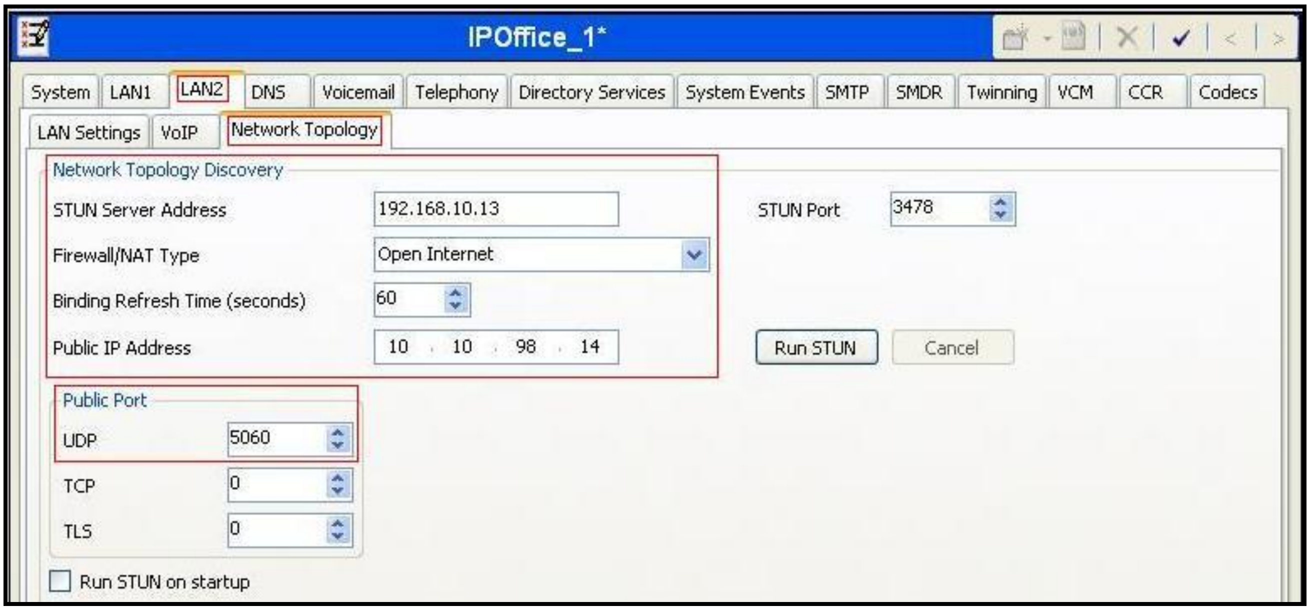
- Check the **H323 Gatekeeper Enable** to allow Avaya IP Telephones/Softphones using the H.323 protocol to register.
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to CenturyLink.
- Check the **SIP Registrar Enable** to allow Avaya IP Telephones/Softphones to register using the SIP protocol.
- Input **Domain Name** as **10.10.98.14**.
- The **Layer 4 Protocol** use **UDP** with port **5060** and **TCP** with port **5060**.

- Verify the **RTP Port Number Range** settings for a specific range for the RTP traffic. The **Port Range (Minimum)** and **Port Range (Maximum)** values were kept as default.
- Check **Enable RTCP Monitoring on Port 5005**.
- Verify **RTP Keepalives** settings were enabled with **Scope** as **RTP**, **Periodic timeout** in **30** seconds, and **Initial keepalives** as **Enabled**. This allows IP Office to send IP packets to keep the active RTP session alive in every 30 seconds if there is no audio detected on the SIP Trunk.
- Verify the **DiffServ Settings** were kept as default for the Differentiated Services Code Point (DSCP) parameters in the IP packet headers 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.
- All other parameters should be set according to customer requirements.
- Click OK (not shown).



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

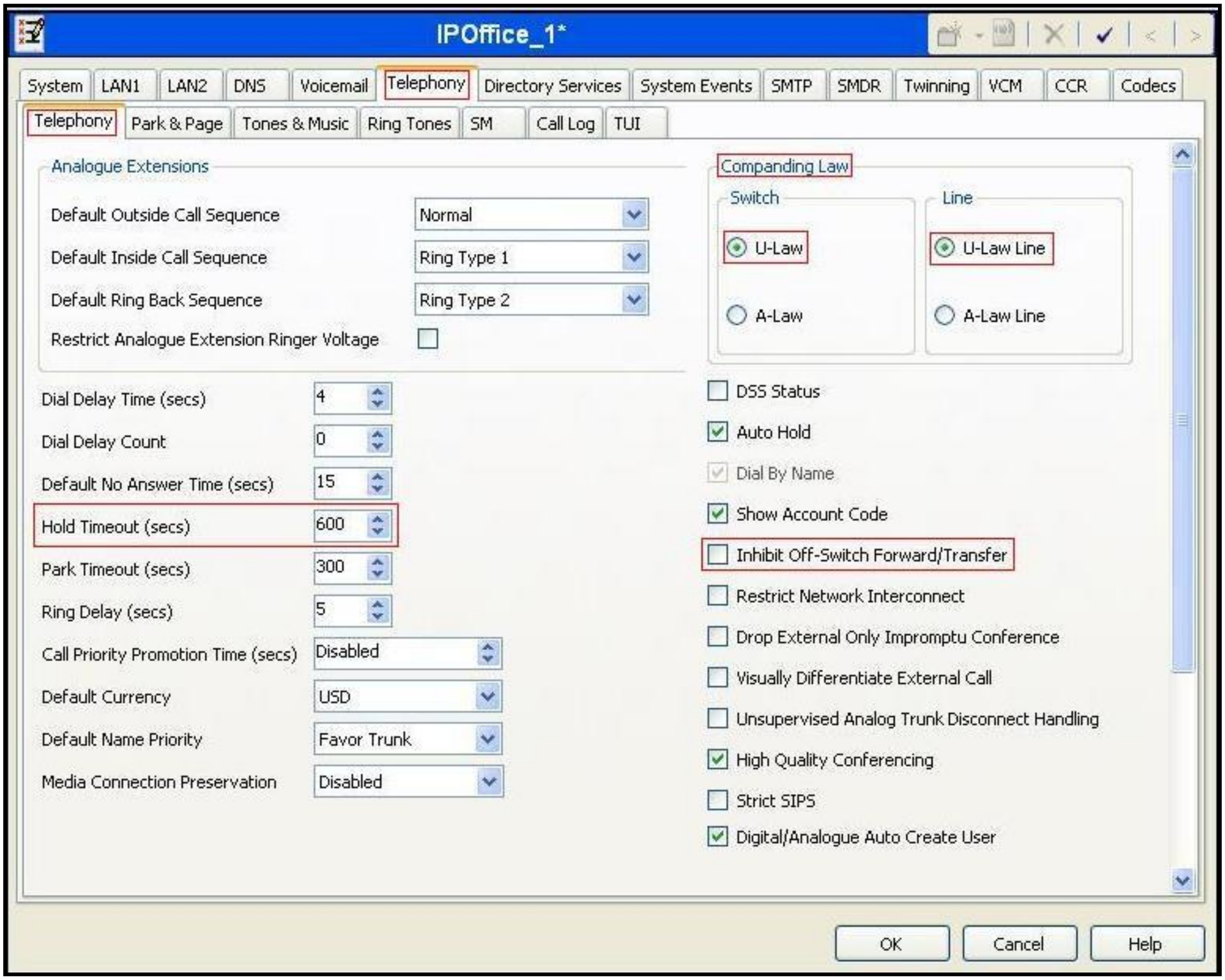
- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**. With this configuration, STUN will not be used.
- Set the **Binding Refresh Time (seconds)** to **60**. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the IP address of the Avaya IP Office LAN2 port.
- Set **Public Port** for **UDP** as **5060**.
- All other parameters should be set according to customer requirements.
- Click OK (not shown).



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

5.2. System Telephony Settings

Navigate to **System (1) → IPOffice_1** in the Navigation and Group Panes and then navigate to the **Telephony → Telephony** tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. For North America, **U-Law** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the service provider across the SIP trunk. Set **Hold Timeout (secs)** to **600**. Click **OK** to submit the change.



5.3. System Codec Settings

Navigate to **System (1)** → **IPOffice_1** in the Navigation and Group Panes and then navigate to the **Codecs** tab in the Details Pane. Choose the **RFC2833 Default Payload** as IP Office default of **101**. Select codecs **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** that CenturyLink supports. Click **OK** to submit the change.

5.4. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affect twinning and do not impact the messaging or operation of other redirected calls such as forwarded calls. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **Twinning** tab, as shown below. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section** Error! Reference source not found.).

If **Send original calling party information for Mobile Twinning** on the **Twinning** tab is optioned, the setting of the second parameter is ignored and Avaya IP Office will send the following in the SIP From Header:

- On calls from an internal extension to a twinned phone, Avaya IP Office will send the calling party number of the originating extension.
- On calls from the PSTN to a twinned phone, Avaya IP Office will send the calling party number of the host phone associated with the twinned destination (instead of the number of the originating caller).

If this option is unchecked, the value sent in the SIP From header is determined by the setting of the second parameter mentioned above.

- For the compliance test, the **Send original calling party information for Mobile Twinning** box in the **System→Twinning** tab was unchecked. The value sent in the SIP From header is determined by the setting of the **Send Caller ID** parameter on the **SIP Line** form**.**

5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and CenturyLink 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 5.5.1** 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 **Section** Error! Reference source not found.**.**

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

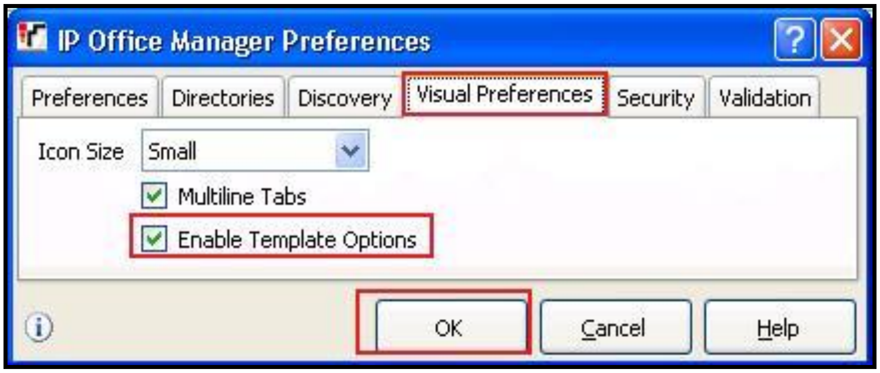
- SIP Line – Originator number for forwarded and twinning calls.
- Transport – Second Explicit DNS Server.
- SIP Credentials – Registration Required.

Alternatively, a SIP Line can be created manually. To do so right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section** Error! Reference source not found.**.**
For the compliance test, SIP Line 17 was used as trunks for incoming calls and SIP Line 18 was used as trunks for outgoing and incoming toll-free calls.

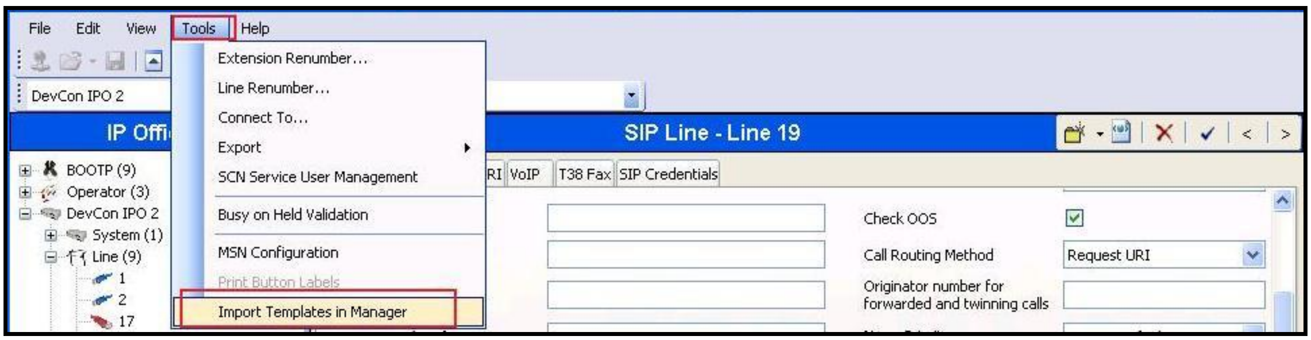
5.5.1. Create SIP Line from Template.

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **CenturyLink_17.xml** (for SIP Line 17) and **CenturyLink_18.xml** (for SIP Line 18). 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 box is checked next to **Enable Template Options**. Click **OK**.

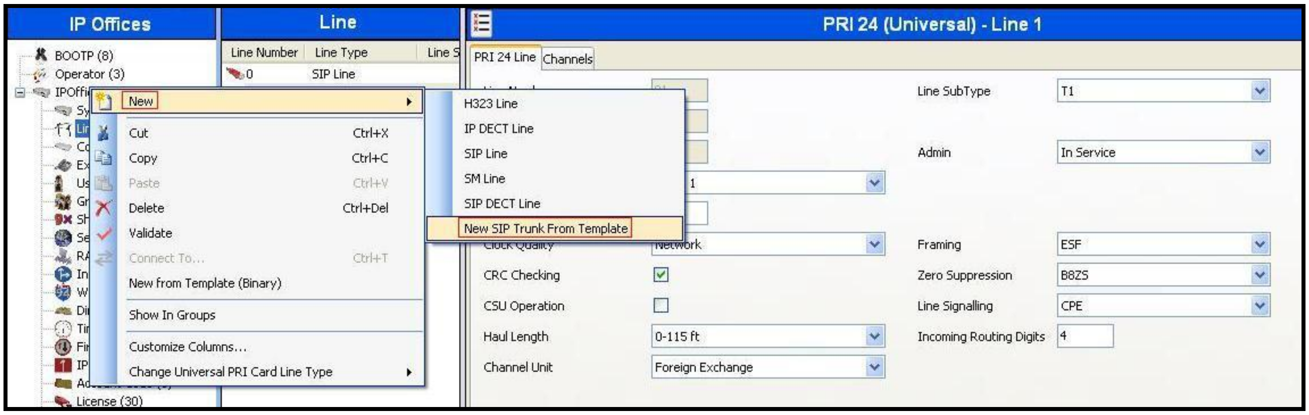


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template 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. Then 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.

4. 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**.



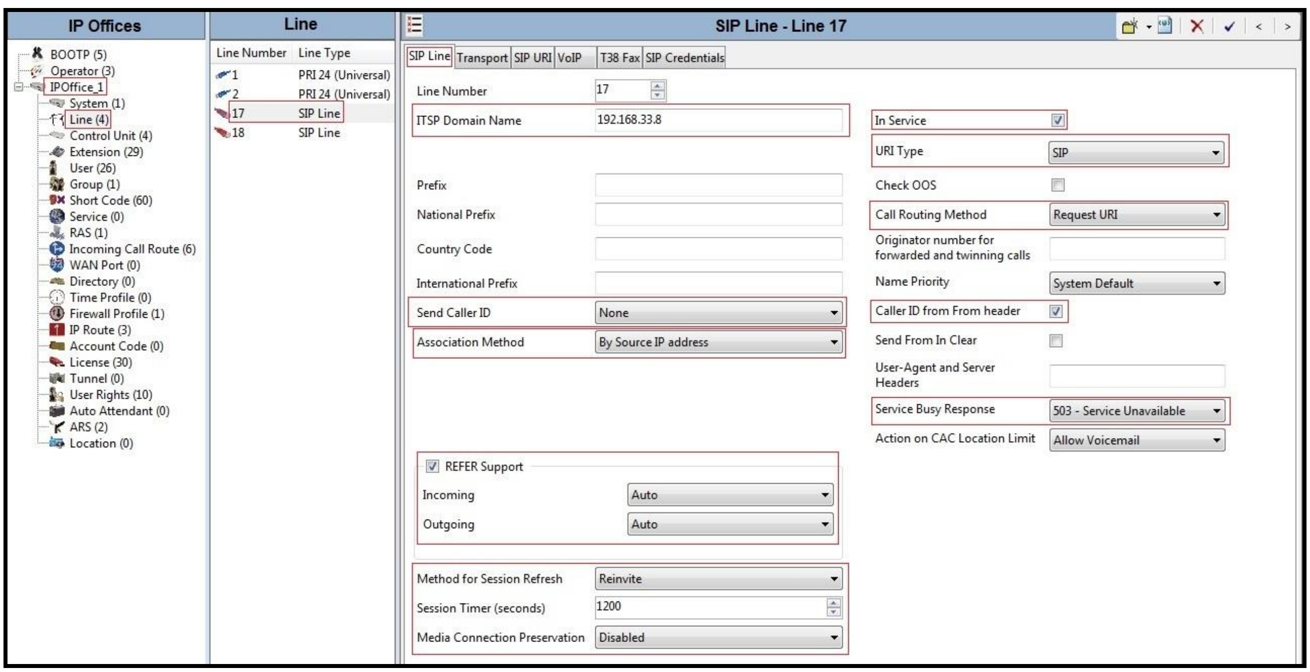
5. In the subsequent Template Type Selection pop-up window, select **United States** from the **Country** pull-down menu and select **CenturyLink_17 or CenturyLink_18** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name **(CenturyLink_17.xml** and **CenturyLink_18.xml)** created in **Step 1**. 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 **Section 5.5.2.**

5.5.2. Create SIP Line Manually for Incoming Calls

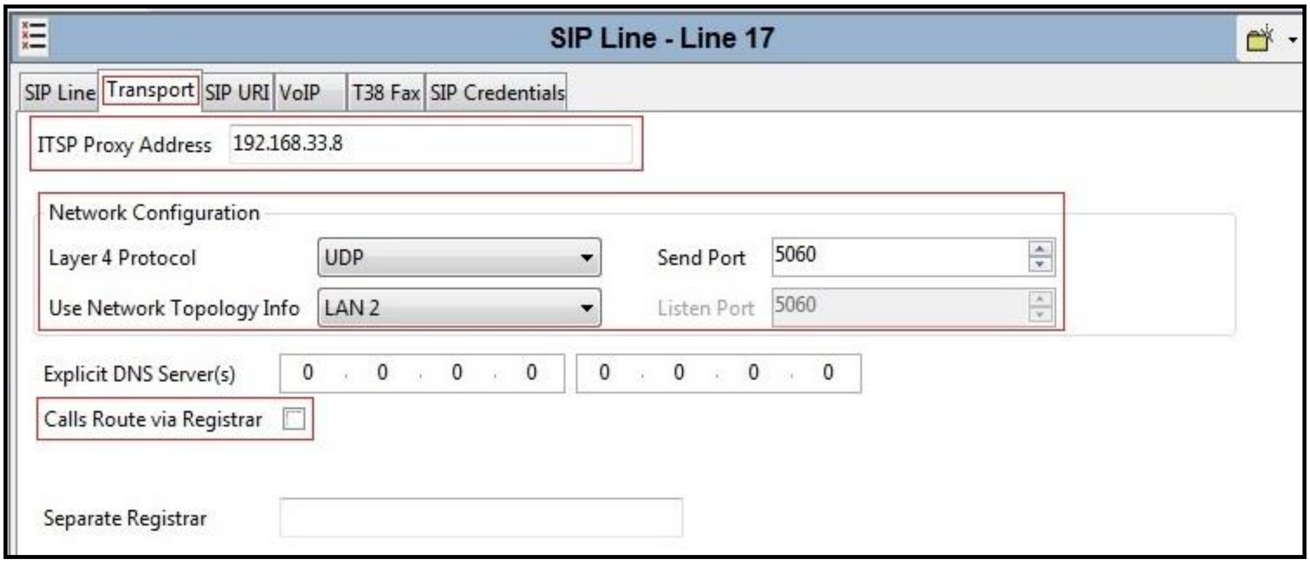
To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line**.
On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the IP address of CenturyLink signaling server (Session Trunk Group) **192.168.33.8**.
- Check the **In Service** box.
- Set **URI Type** to **SIP**.
- Set **Call Routing Method** to **Request URI**.
- Check **Caller ID from From header** box.
- Set **Send Caller ID** to **None**.
- Set **Association Method** to **By Source IP address**.
- Set **Service Busy Response** as **503 – Service Unavailable**.
- Check **REFER Support** to enable SIP REFER for call transfers. Select the default values of "**Auto**" for **Incoming** and **Outgoing** to use of SIP REFER effectively.
- Set **Method for Session Refresh** to **Reinvite** with **Session Timer (seconds)**: **1200**. If the Method for Session Refresh set to the default value "On Demand", then IP Office does not initiate Session Timer and only supports it if it is initiated by the other side.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown) then press Ctrl + S to save.



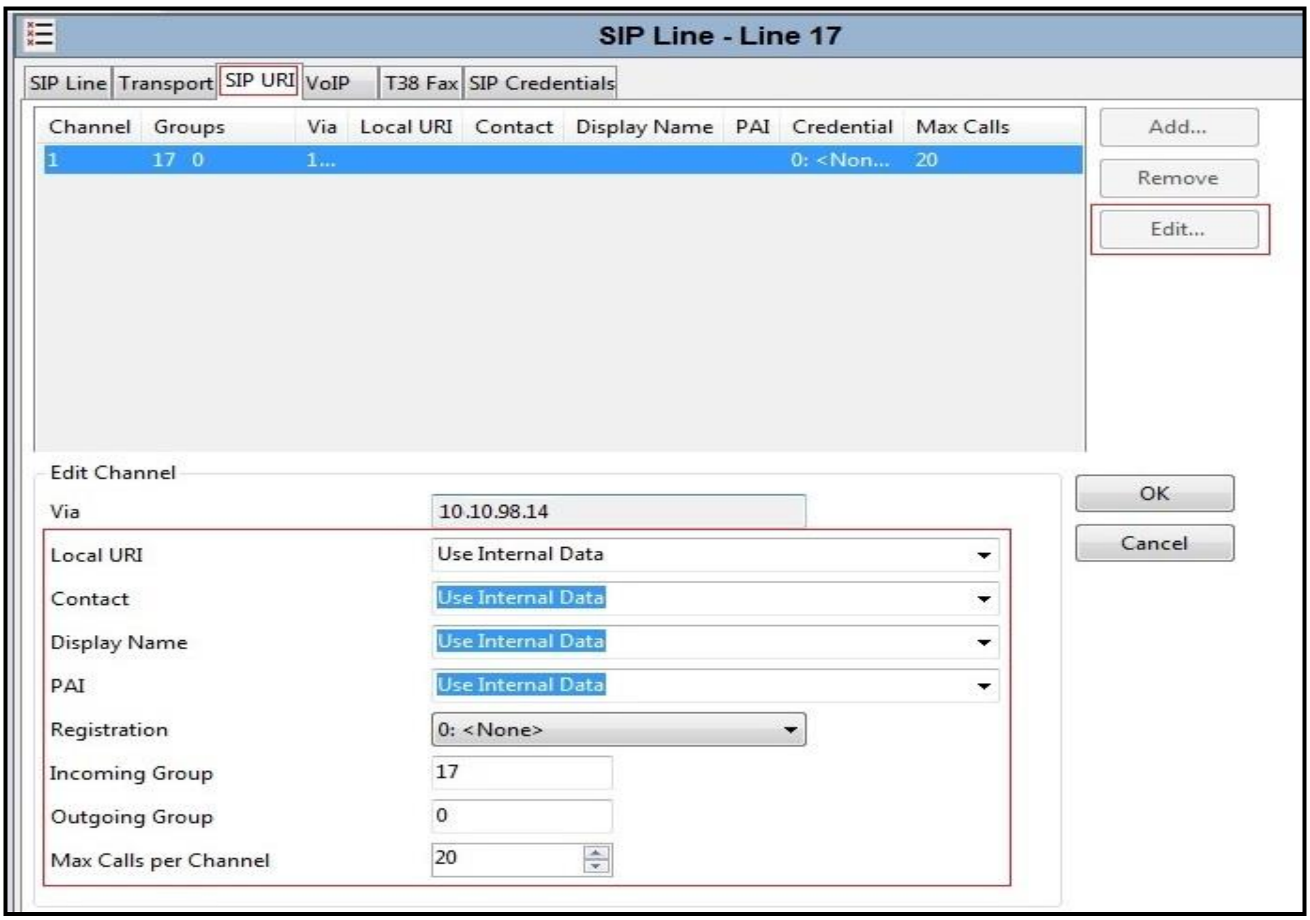
On the **Transport** tab in the Details Pane, configure the parameters as shown below:

- The **ITSP Proxy Address** was set to the IP Address of CenturyLink signaling server (Session Trunk Group) **192.168.33.8** as shown in **Figure 1**.
- In the **Network Configuration** area, **UDP** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **5060** which is the port number provided by CenturyLink.
- The **Use Network Topology Info** parameter was set to **LAN 2**. This associates the SIP Line 17 with the parameters in the **System → LAN2 → Network Topology** tab.
- The **Calls Route via Registrar** was unchecked. In this certification testing, CenturyLink did not support the dynamic Registration on the SIP Trunk.
- Other parameters retain default values.
- Click **OK** to commit (not shown) then press Ctrl + S to save.



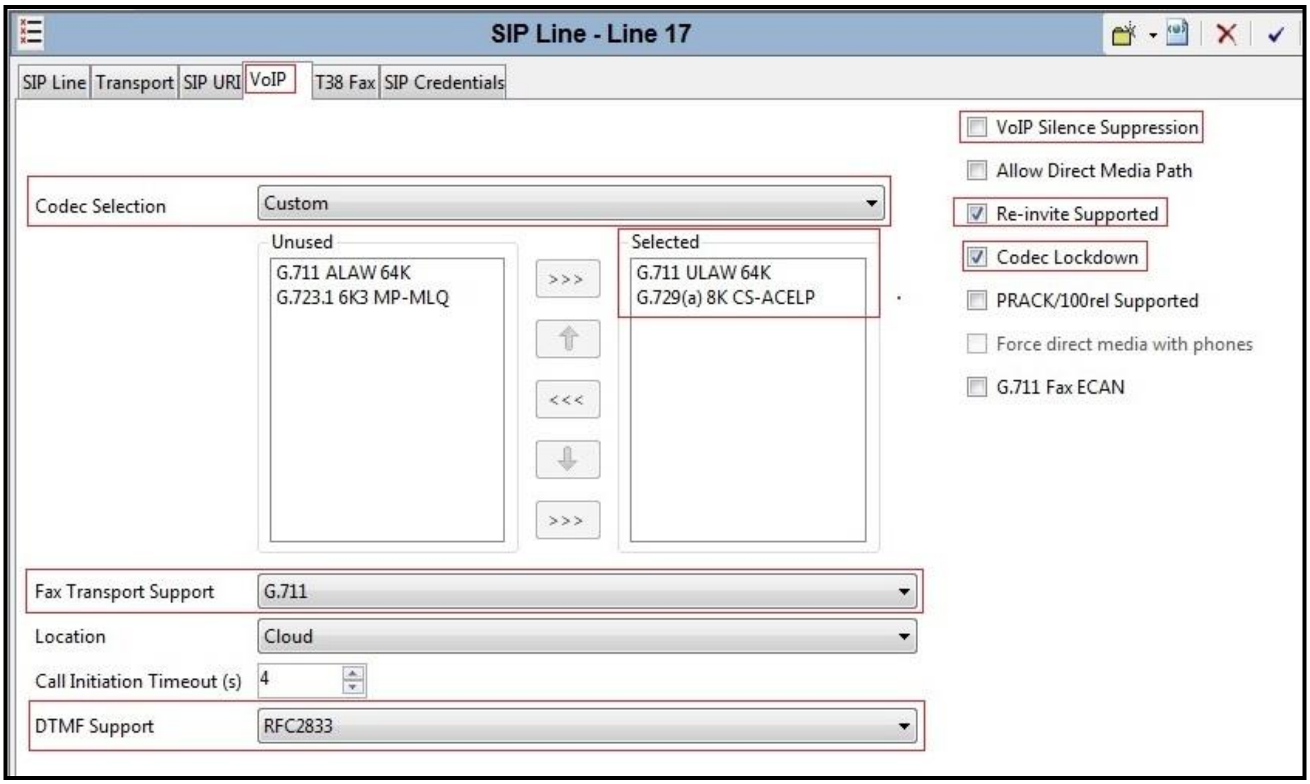
A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab; click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click **Edit…** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact**, **Display Name**, and **PAI** to **Use Internal Data**. This setting allows calls on this line which SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.7**.
- Set **Registration** to **0: <None>** as CenturyLink does not require registration.
- Associate this line with an incoming line group in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. For the compliance test, a new incoming group **17** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Click **OK** to submit the changes.



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

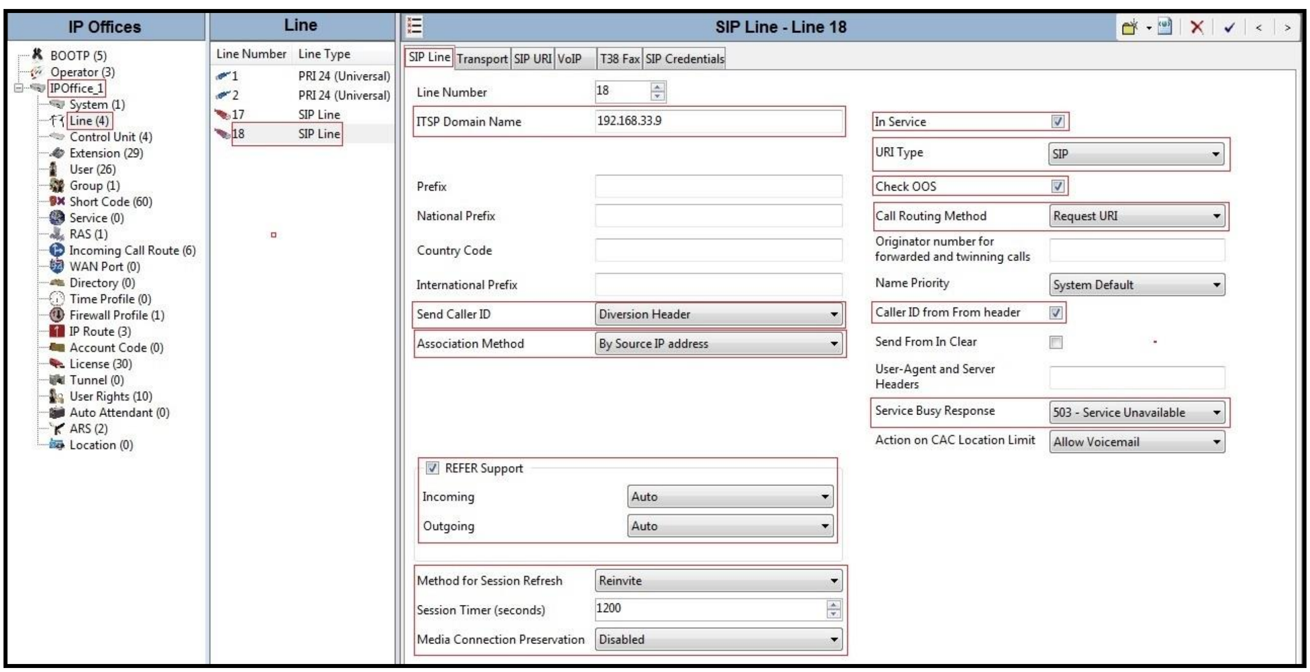
- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. Selecting **G.711 ULAW 64K**, and **G.729(a) 8K CS –ACELP** codecs causes Avaya IP Office to support these codecs, which are sent by the CenturyLink, in the Session Description Protocol (SDP) offer, in that order.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Check **Codec Lockdown**.
- Set **Fax Transport Support** to **T38** or **G.711** from the pull-down menu. CenturyLink supports both Fax T.38 and Fax G.711 pass-through modes.
- Set the **DTMF Support** to **RFC2833** from the pull-down menu. 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**.**
- Click OK (not shown) to submit the changes.



5.5.3. Create SIP Line Manually for Outgoing Calls

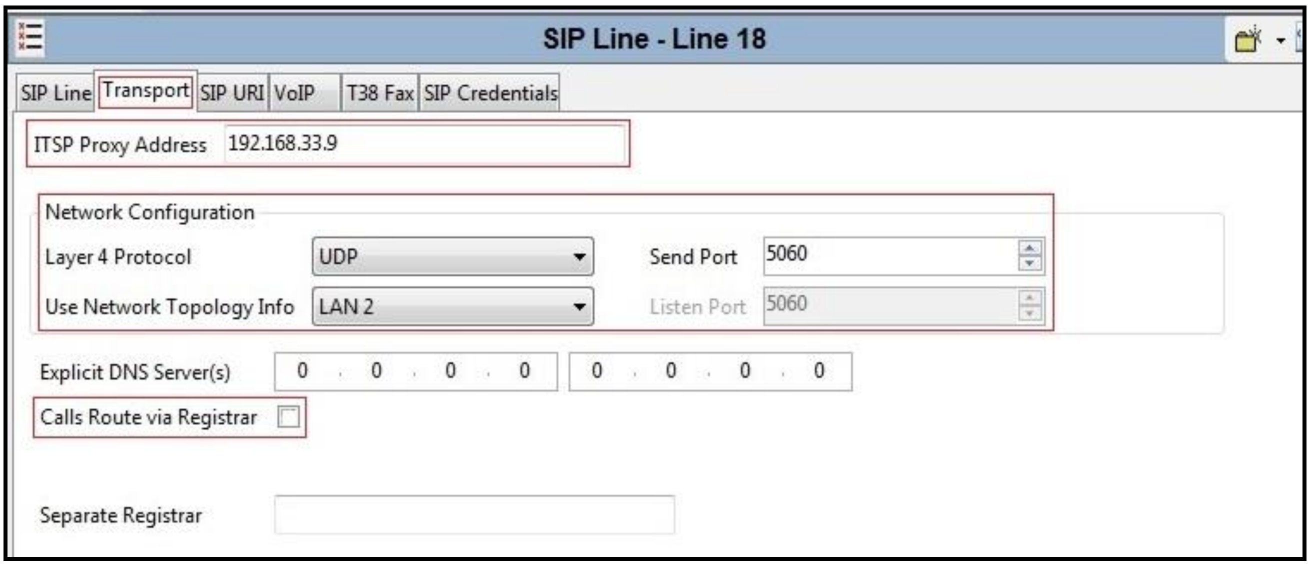
On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the IP address of CenturyLink signaling server (Usage Trunk Group) **192.168.33.9**.
- Check the **In Service** box.
- Set **URI Type** to **SIP**.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Set **Call Routing Method** to **Request URI**.
- Check **Caller ID from From header** box.
- Set **Send Caller ID** to **Diversion Header**.
- Set **Association Method** to **By Source IP address**.
- Set **Service Busy Response** as **503 – Service Unavailable**.
- Check **REFER Support** to enable SIP REFER for call transfers. Select the default values of "**Auto**" for **Incoming** and **Outgoing** to use of SIP REFER effectively.
- Set **Method for Session Refresh** to **Reinvite** with **Session Timer (seconds)**: **1200**. If the Method for Session Refresh set to the default value "On Demand", then IP Office does not initiate Session Timer and only supports it if it is initiated by the other side.
- Default values may be used for all other parameters.
- Click OK to commit (not shown) then press Ctrl + S to save.



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

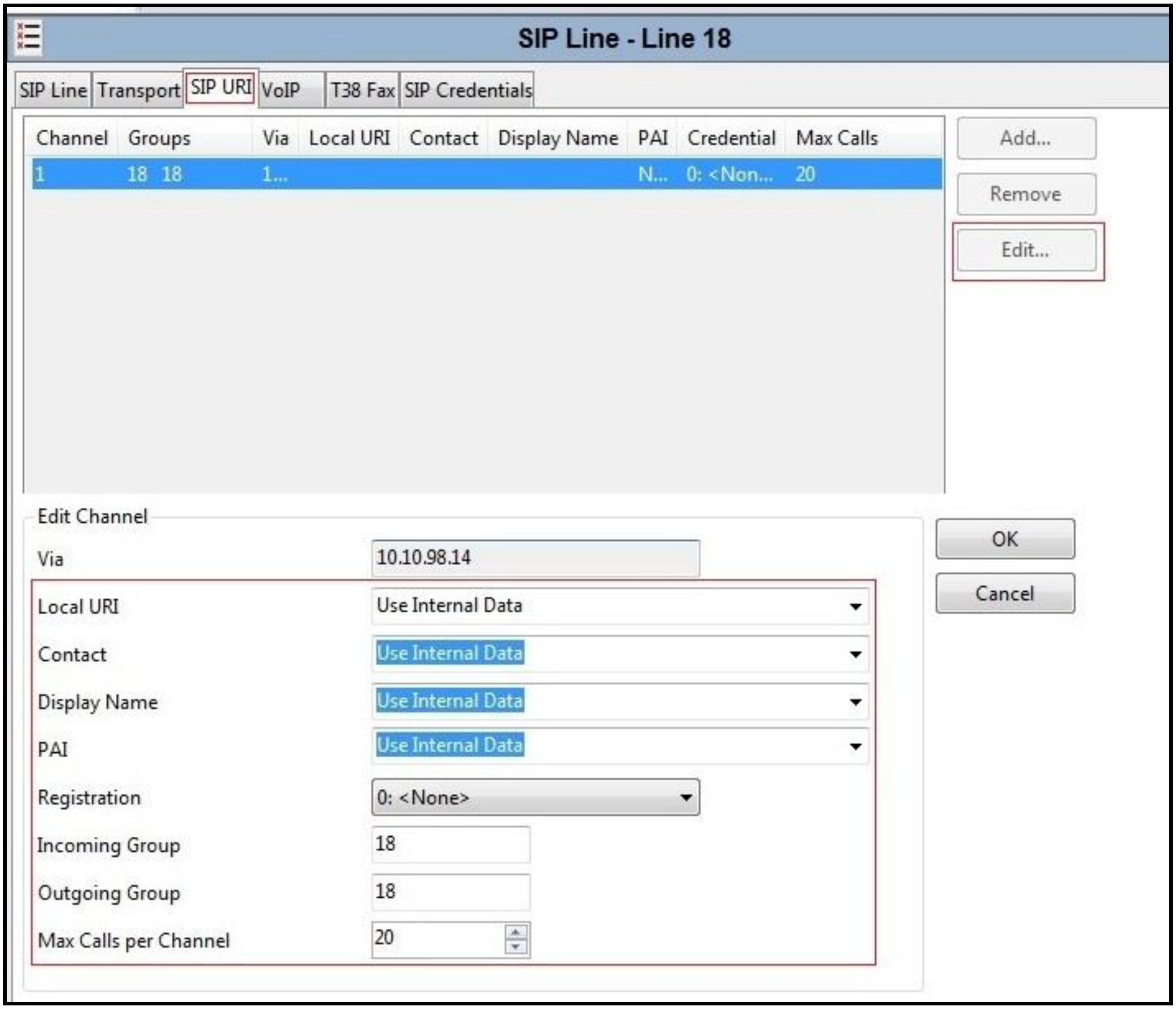
- The **ITSP Proxy Address** was set to the IP Address of CenturyLink signaling server (Usage Trunk Group) **192.168.33.9**.
- In the **Network Configuration** area, **UDP** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **5060** which is the port number provided by CenturyLink.
- The **Use Network Topology Info** parameter was set to **LAN 2**. This associates the SIP Line 18 with the parameters in the **System → LAN2 → Network Topology** tab.
- The **Calls Route via Registrar** was unchecked. In this certification testing, CenturyLink did not support the dynamic Registration on the SIP Trunk.
- Other parameters retain default values.
- Click OK to commit (not shown) then press Ctrl + S to save.



A SIP URI entry must be created to match each outgoing number that Avaya IP Office will send on this line. Select the **SIP URI** tab; click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact**, **Display Name**, and **PAI** to **Use Internal Data**. This setting allows calls on this line which SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.7**.
- Set **Registration** to **0: <None>** as CenturyLink does not require registration.
- Associate this line with an outgoing line group in the **Outgoing Group** field. This line group number will be used in defining outgoing call routes for this line. For the compliance test, a new outgoing group **18** was defined that only contains this line (line 18).
  Note: For the specific inbound toll-free test, associate this line 18 with an incoming line group in the **Incoming Group** field.

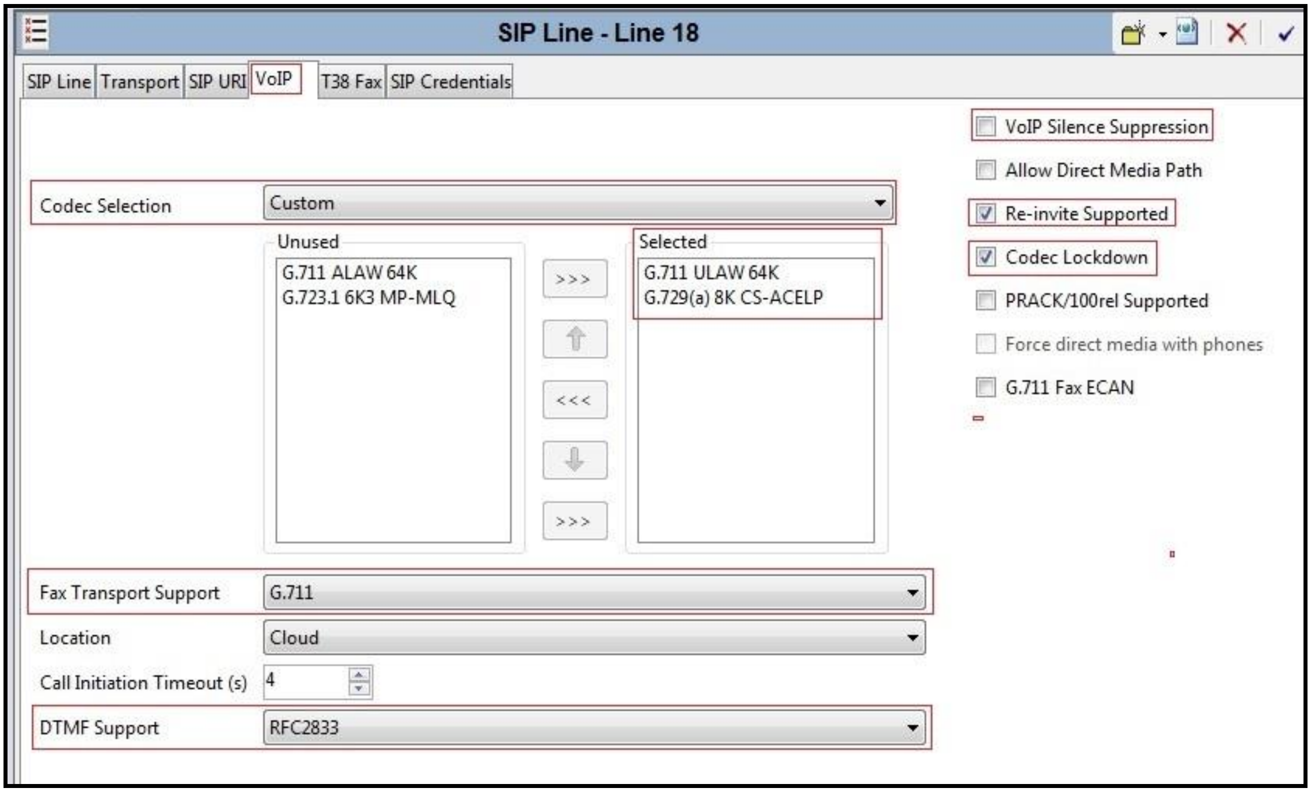
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Click **OK** to submit the changes.



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

- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. Selecting **G.711 ULAW 64K**, and **G.729(a) 8K CS –ACELP** codecs causes Avaya IP Office to include these codecs, which are supported by the CenturyLink, in the Session Description Protocol (SDP) offer, in that order.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Check **Codec Lockdown**.
- Set **Fax Transport Support** to **T38** or **G.711** from the pull-down menu. CenturyLink support both Fax T.38 and Fax G.711 pass-through modes.

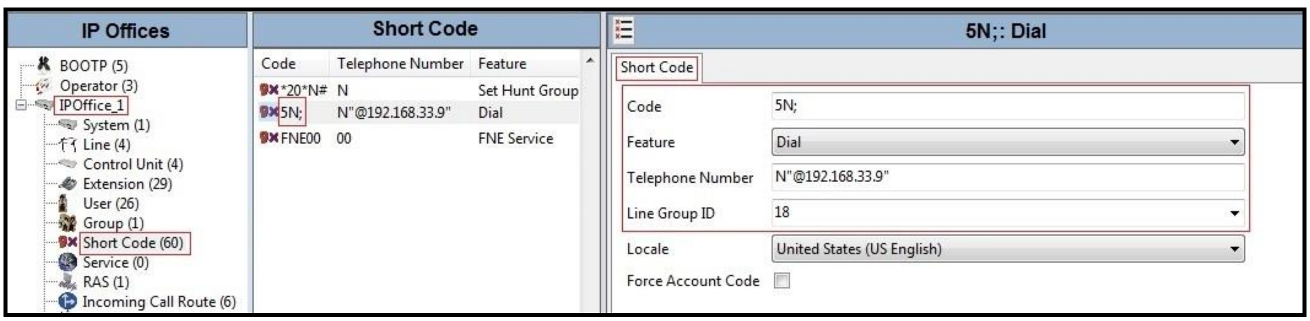
- Set the **DTMF Support** to **RFC2833** from the pull-down menu. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters**.**
- Click OK (not shown) to submit the changes.



5.6. Short Code

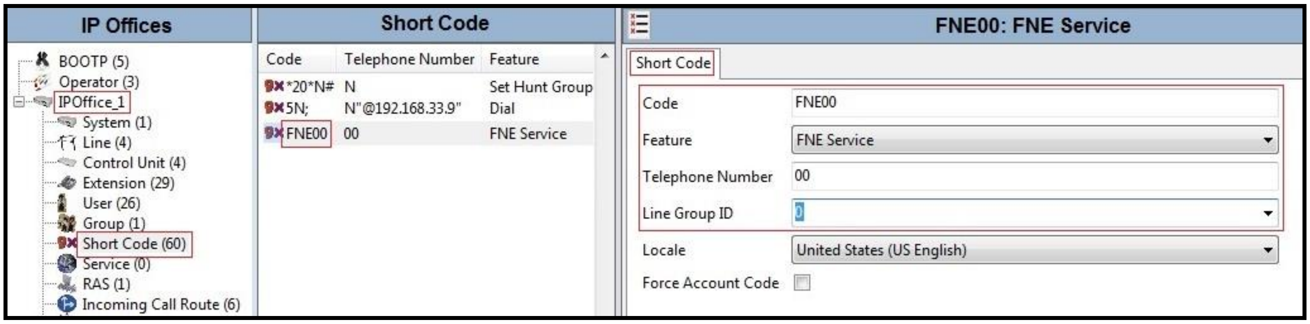
Define a short code to route outbound traffic on the SIP line to CenturyLink. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New** (Not shown). On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered "5N;" short code used in the test configuration.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **5N;**, this short code will be invoked when the user dials 5 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@192.168.33.9"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. The host part following the "@" is the IP address of CenturyLink signaling server (Usage Trunk Group).
- Set the **Line Group ID** to the **Outgoing Group 18** defined on the **SIP URI** tab on the **SIP Line** in **Section** Error! Reference source not found.. This short code will use this line group when placing the outbound call.



The feature of incoming calls from mobility extension to idle-appearance FNE (Feature Name Extension) is hosted by Avaya IP Office. The Short Code **FNE00** was configured with following parameters:

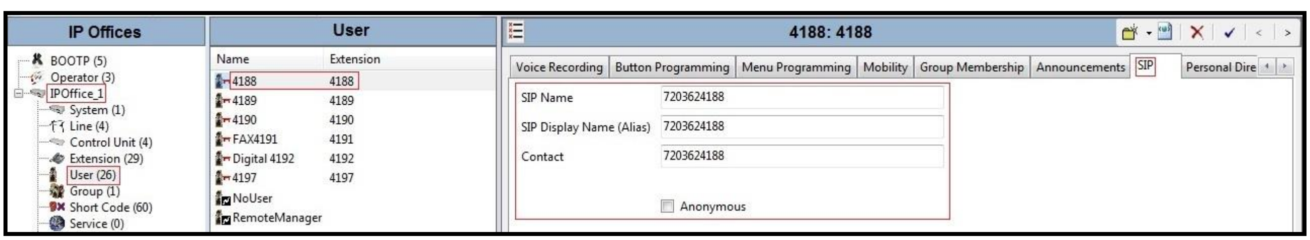
- For **Code** field, enter FNE feature code as **FNE00** for dial tone.
- Set **Feature** to **FNE Service**.
- Set **Telephone Number** to **00**.
- Set **Line Group ID** to **0**.
- Default values may be used for other parameters.



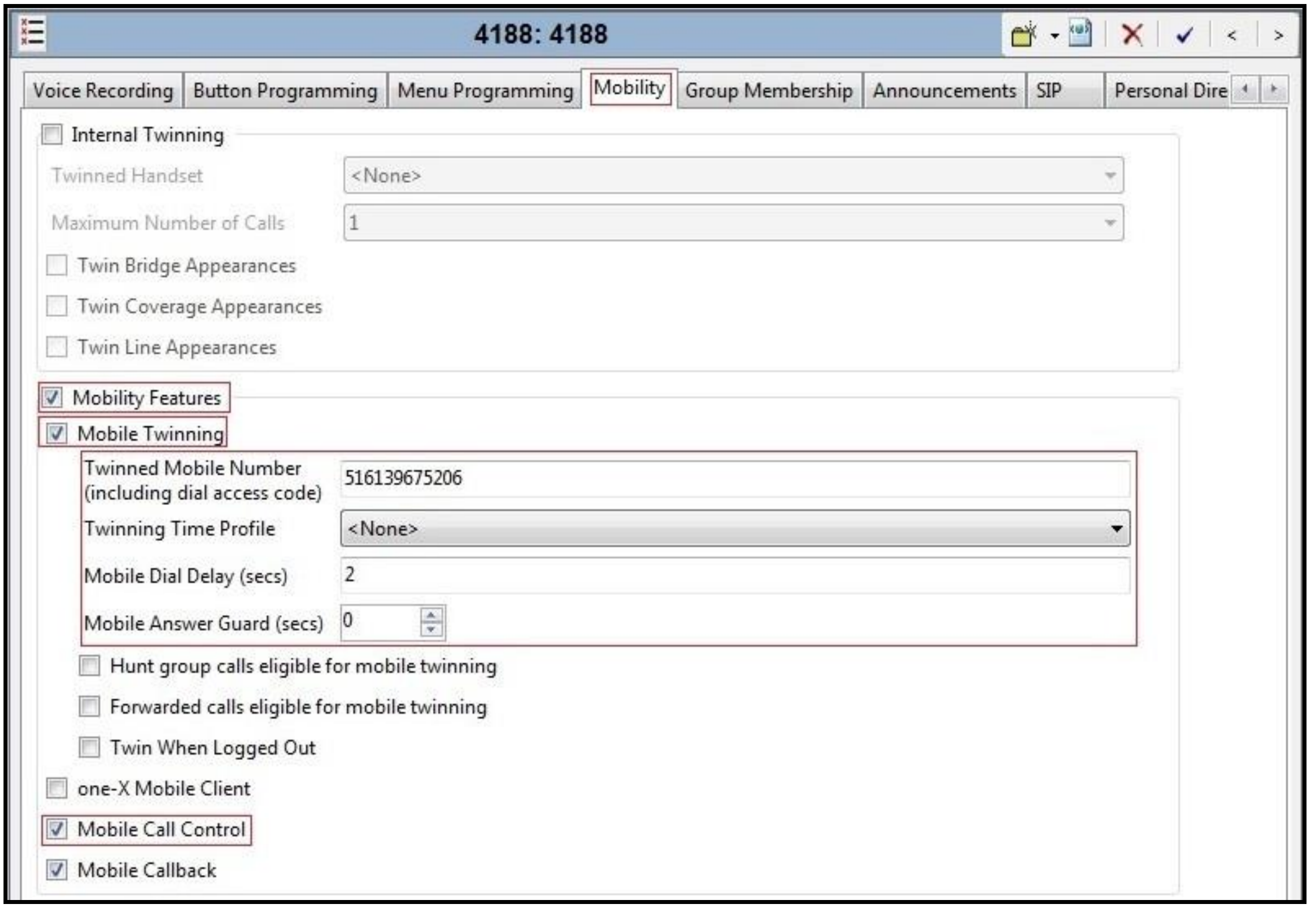
5.7. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.5**. To configure these settings, first select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **4188**. Select the **SIP** tab in the Details Pane.

The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line. The example below shows the settings for user **4188**. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise provided by 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.



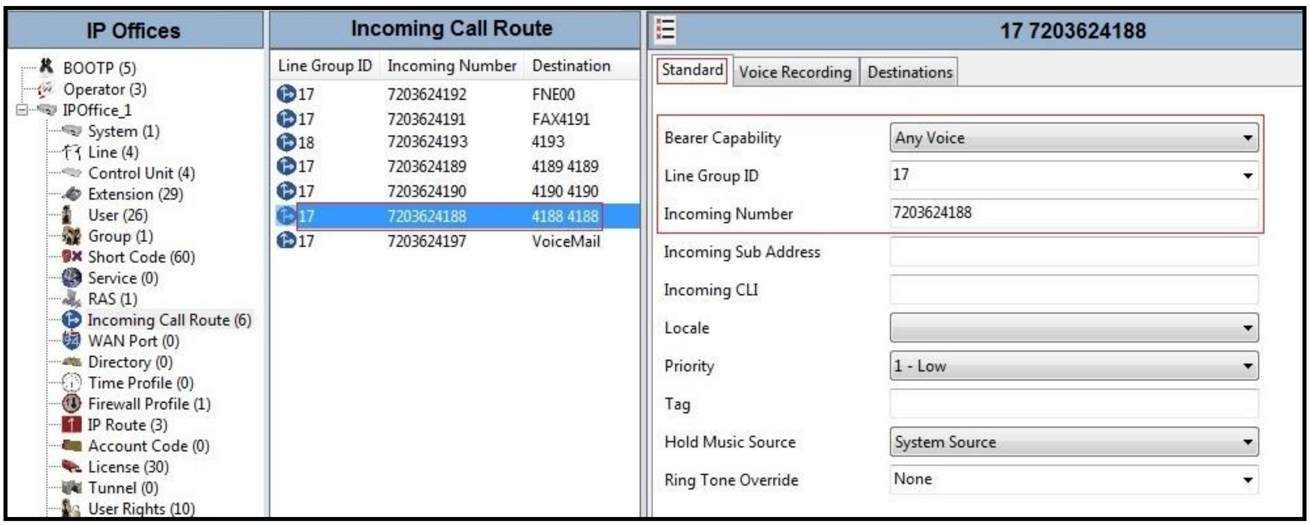
One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for **User 4188**. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **516139675206**. Check **Mobile Call Control** to allow incoming calls from mobility extension to access FNE00 (see **Section 5.6**). Other options can be set according to customer requirements.



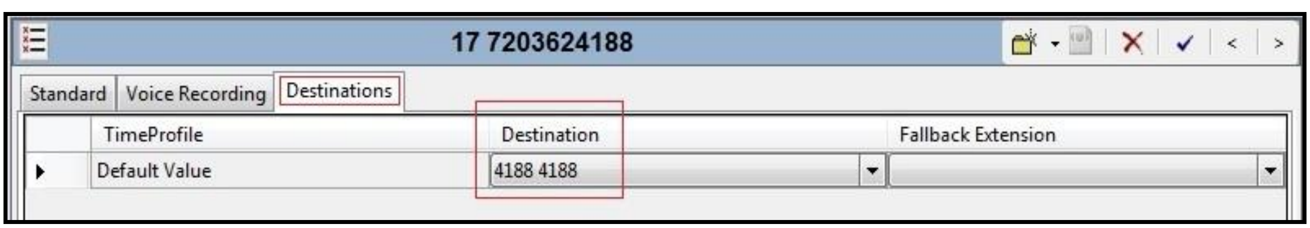
5.8. Incoming Call Route

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 service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New** (Not shown). On the **Standard** tab of the Details Pane, enter the parameters as shown below:

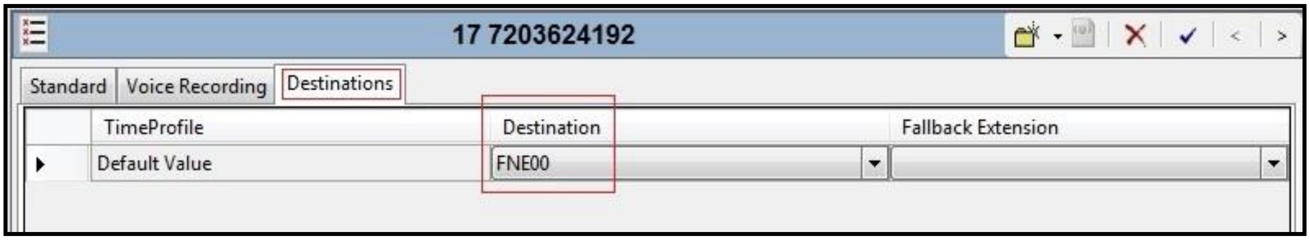
- Set the **Bearer Capability** to **Any Voice**.
- Set the **Line Group ID** to the **Incoming Group 17** defined on the **SIP URI** tab on the **SIP Line** in **Section** Error! Reference source not found..
  Note: For the specific inbound toll-free test, associate incoming DID number to an incoming Line Group ID 18.
- Set the **Incoming Number** to the incoming DID number on which this route should match.
- Default values can be used for all other fields.



On the **Destination** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **7203624188** on line 17 are routed to extension **4188**.



The incoming calls to DID number **7203624192** were configured to access **FNE00**. The **Destination** was appropriately defined as **FNE00** as below screenshot:



The incoming calls to DID number **7203624197** were configured to access **VoiceMail**. The **Destination** was appropriately defined as **VoiceMail** as below screenshot:

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.

6. CenturyLink SIP Trunk Configuration

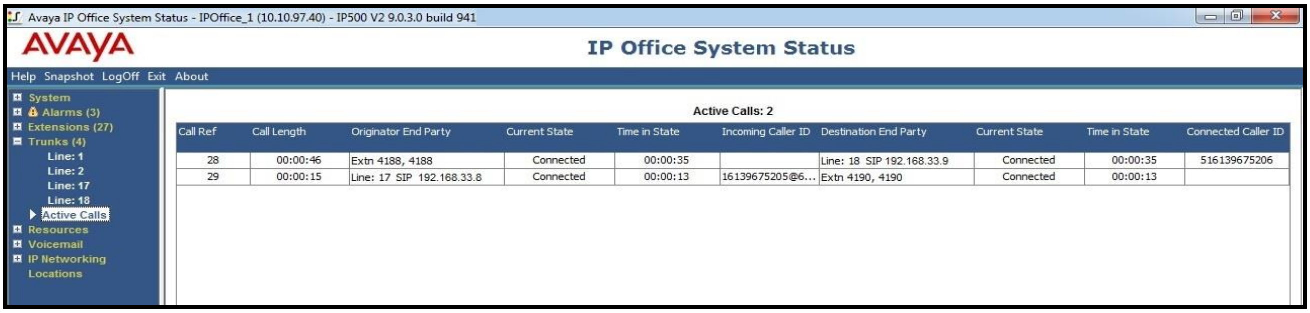
CenturyLink is responsible for the configuration of CenturyLink SIP Trunk Service. The customer must provide the IP address used to reach the Avaya IP Office at the enterprise. CenturyLink will provide the customer the necessary information to configure the SIP connection between Avaya IP Office and CenturyLink. The provided information from CenturyLink includes:

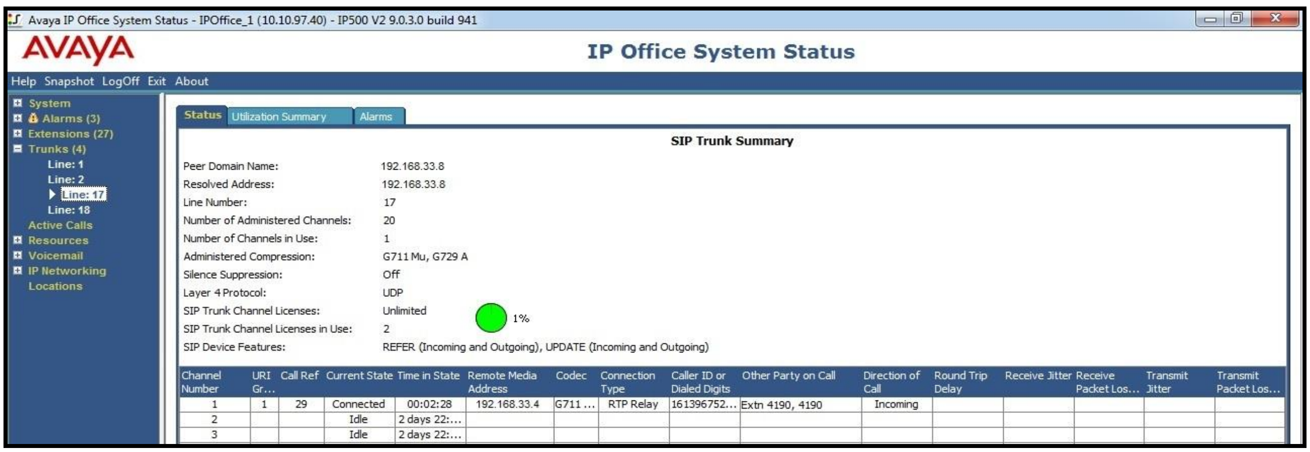
- Fully Qualified Domain Name, IP address and port number used for signaling or media through any security.
- DID numbers.
- CenturyLink SIP Trunk Specification.

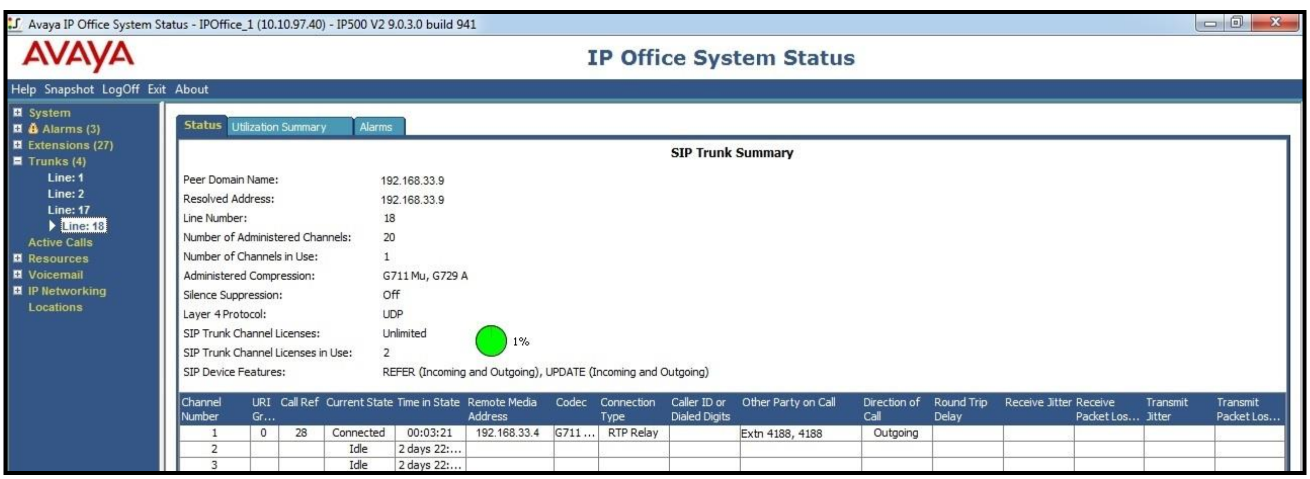
7. Verification Steps

The following steps may be used to verify the configuration:

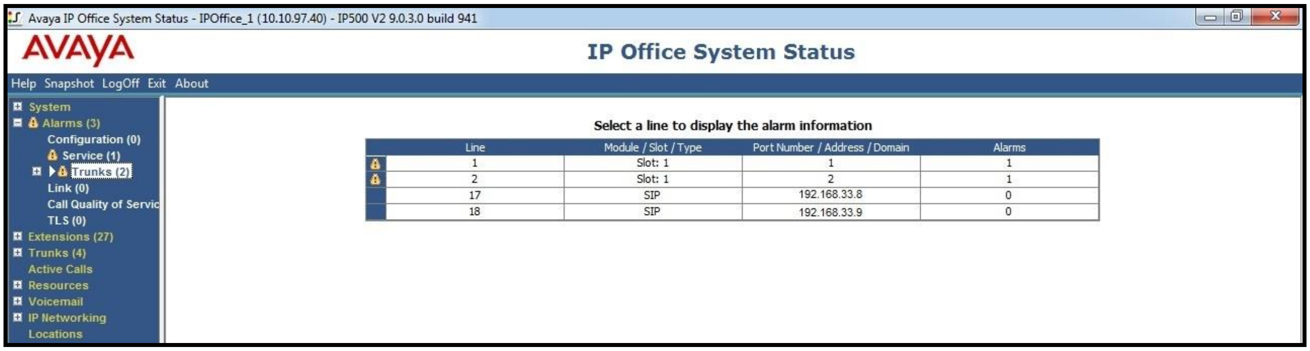
- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** for each channel (The below screen shot showed 2 active calls at present time).







- Use the Avaya IP Office System Status application to verify that no alarms are active on the SIP line. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select **Alarm → Trunks** to verify that no alarms are active on the SIP line.



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

8. Conclusion

CenturyLink passed compliance testing. These Application Notes describe the procedures required to configure the SIP connection between Avaya IP Office and the CenturyLink as shown in **Figure 1**.

9. Additional References

[1] Avaya IP Office Document Library Release 9.0, Document number 15-604278 Issue 1, September 2013

[2] Avaya IP Office 9.0.3 Installing IP500/IP500V2, Document number 15-601042 Issue 29f, 13 June 2014

[3] Avaya IP Office Manager Release 9.0, Document number 15-601011 Issue 09.03, 31 March, 2014

[4] Avaya IP Office 9.0 Softphone User Guide (Windows), Issue 07c, 24 February 2014

[5] Avaya IP Office Embedded Voicemail User Guide (IP Office Mode), Document number 15-604067 Issue 13a, 13 February 2014

Product documentation for Avaya products may be found at: http://support.avaya.com. Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/ipoffice/general/rss2html.php?XMLFILE=manuals.xml&TEMPLATE=pdf_feed_template.html

Product documentation for CenturyLink SIP Trunk may be found at: http://www.centurylink.com

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.