



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.

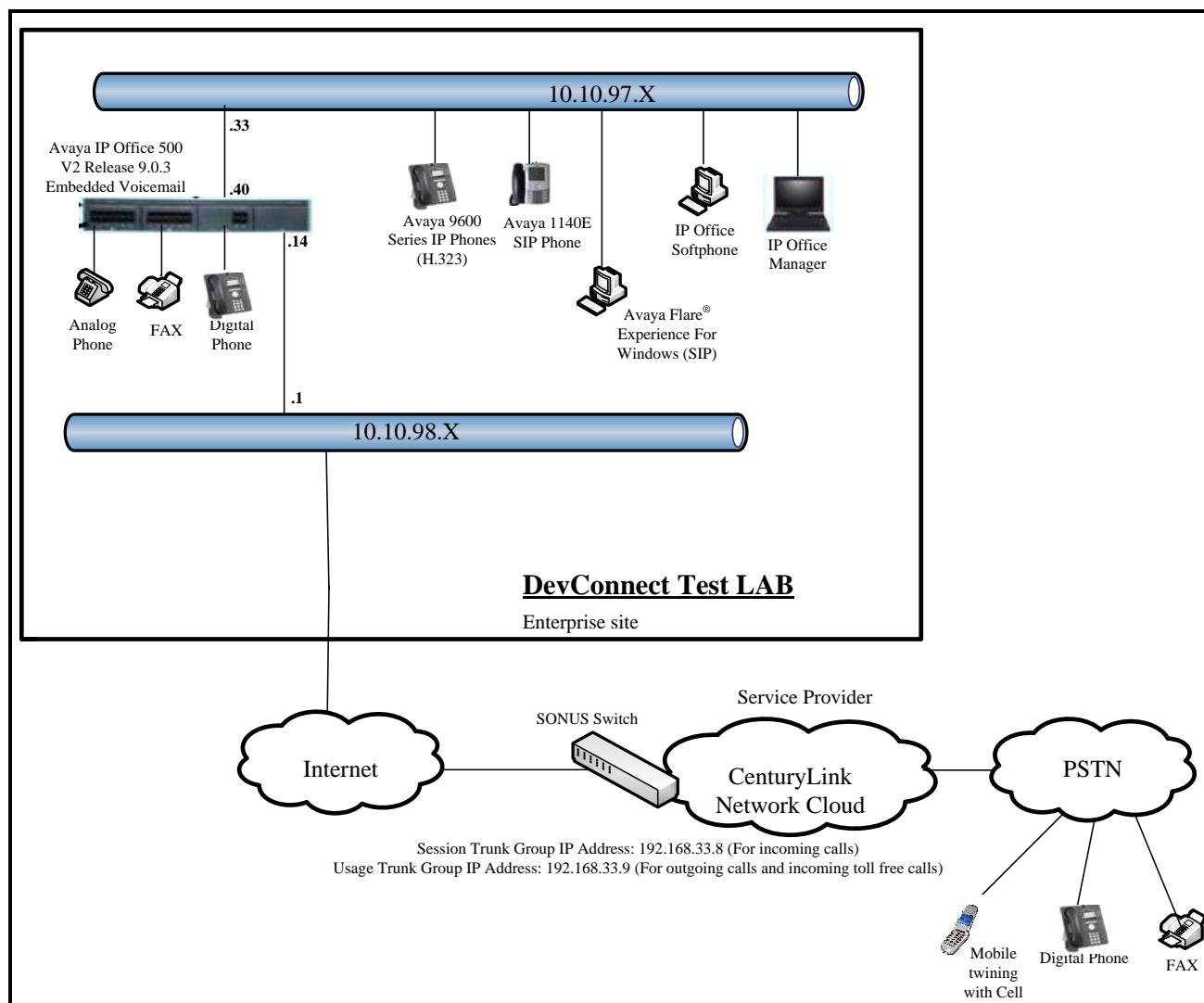


Figure 1: Test Configuration for Avaya IP Office with CenturyLink SIP Trunk Service

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

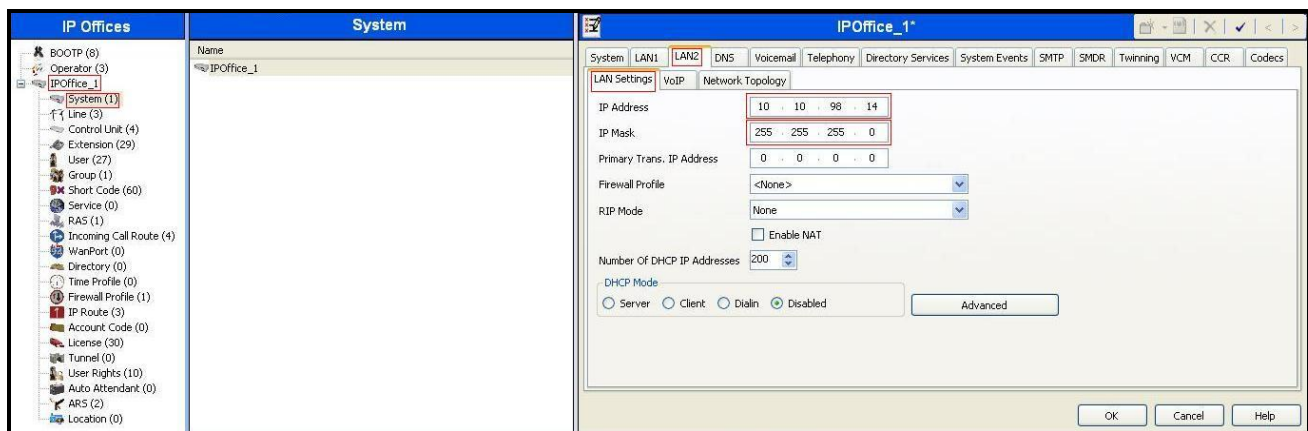
Note: Testing was performed with IP Office 500 v2 R9.0.3, but it also applies to IP Office Server Edition R9.0.3. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0.3 to support analog or digital endpoints or trunks. IP Office Server Edition does not support TAPI Wave or Group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

The screenshot shows the IOffice_1* configuration window with the VoIP tab selected. The following settings are visible and highlighted with red boxes:

- H323 Gatekeeper Enable** (checked)
- SIP Trunks Enable** (checked)
- SIP Registrar Enable** (checked)
- Auto-create Extn/User** (checked)
- Domain Name**: 10.10.98.14
- Layer 4 Protocol**:
 - UDP** (checked): UDP Port 5060, Remote UDP Port 5060
 - TCP** (checked): TCP Port 5060, Remote TCP Port 5060
 - TLS** (unchecked): TLS Port 5061, Remote TLS Port 5061
- Challenge Expiry Time (secs)**: 10
- RTP Port Number Range**:
 - Minimum: 49152, Maximum: 53246
 - Port Number Range (NAT): Minimum: 49152, Maximum: 53246
- Enable RTCP Monitoring on Port 5005** (checked)
- Keepalives**:
 - Scope: RTP
 - Periodic timeout: 30
 - Initial keepalives: Enabled
- DiffServ Settings**:
 - DSCP (Hex): B8
 - Video DSCP (Hex): B8
 - FC: FC
 - DSCP Mask (Hex): 88
 - SIG DSCP (Hex): 88

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).

The screenshot shows the Avaya IP Office configuration window titled "IPOffice_1*". The "Network Topology" tab is selected. The "Network Topology Discovery" section contains the following fields:

- STUN Server Address: 192.168.10.13
- Firewall/NAT Type: Open Internet (dropdown menu)
- Binding Refresh Time (seconds): 60 (spin box)
- Public IP Address: 10 . 10 . 98 . 14 (IP address field)
- STUN Port: 3478 (spin box)
- Public Port: 5060 (spin box, with UDP selected)
- TCP: 0 (spin box)
- TLS: 0 (spin box)

Buttons for "Run STUN" and "Cancel" are visible. A checkbox for "Run STUN on startup" is at the bottom left.

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.

IPOffice_1*

System LAN1 LAN2 DNS Voicemail **Telephony** Directory Services System Events SMTP SMDR Twinning WCM CCR Codecs

Telephony Park & Page Tones & Music Ring Tones SM Call Log TUI

Analogue Extensions

Default Outside Call Sequence Normal

Default Inside Call Sequence Ring Type 1

Default Ring Back Sequence Ring Type 2

Restrict Analogue Extension Ringer Voltage ☐

Dial Delay Time (secs) 4

Dial Delay Count 0

Default No Answer Time (secs) 15

Hold Timeout (secs) 600

Park Timeout (secs) 300

Ring Delay (secs) 5

Call Priority Promotion Time (secs) Disabled

Default Currency USD

Default Name Priority Favor Trunk

Media Connection Preservation Disabled

Companding Law

Switch

☒ **U-Law**

☐ A-Law

Line

☒ **U-Law Line**

☐ A-Law Line

☐ DSS Status

☒ Auto Hold

☒ Dial By Name

☒ Show Account Code

☐ **Inhibit Off-Switch Forward/Transfer**

☐ Restrict Network Interconnect

☐ Drop External Only Impromptu Conference

☐ Visually Differentiate External Call

☐ Unsupervised Analog Trunk Disconnect Handling

☒ High Quality Conferencing

☐ Strict SIP5

☒ Digital/Analogue Auto Create User

OK Cancel Help

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.



The screenshot shows the Avaya IP Office configuration window titled "IPOffice_1". The "Twinning" tab is selected and highlighted with a red box. Below the tab, there is a checkbox labeled "Send original calling party information for Mobile Twinning" which is unchecked and also highlighted with a red box. Below this checkbox is a text input field labeled "Calling party information for Mobile Twinning".

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:

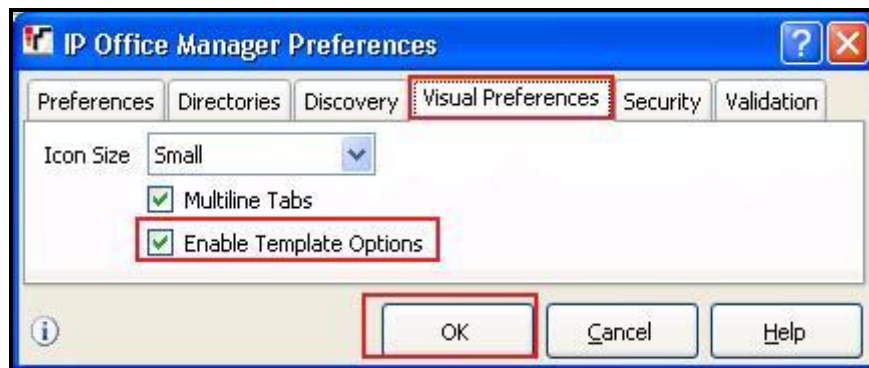
- SIP Line – Originator number for forwarded and twinning calls.
- Transport – Second Explicit DNS Server.
- SIP Credentials – Registration Required.

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..

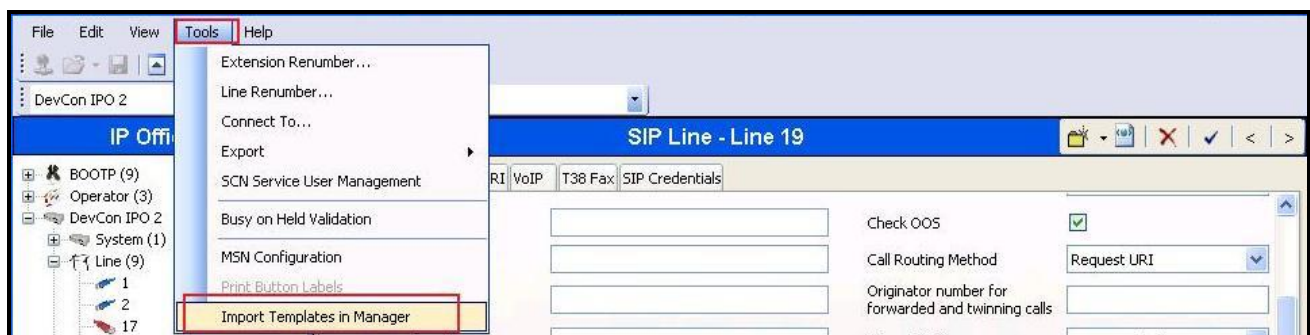
For the compliance test, SIP Line 17 was used as trunks for incoming calls and SIP Line 18 was used as trunks for outgoing and incoming toll-free calls.

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**.

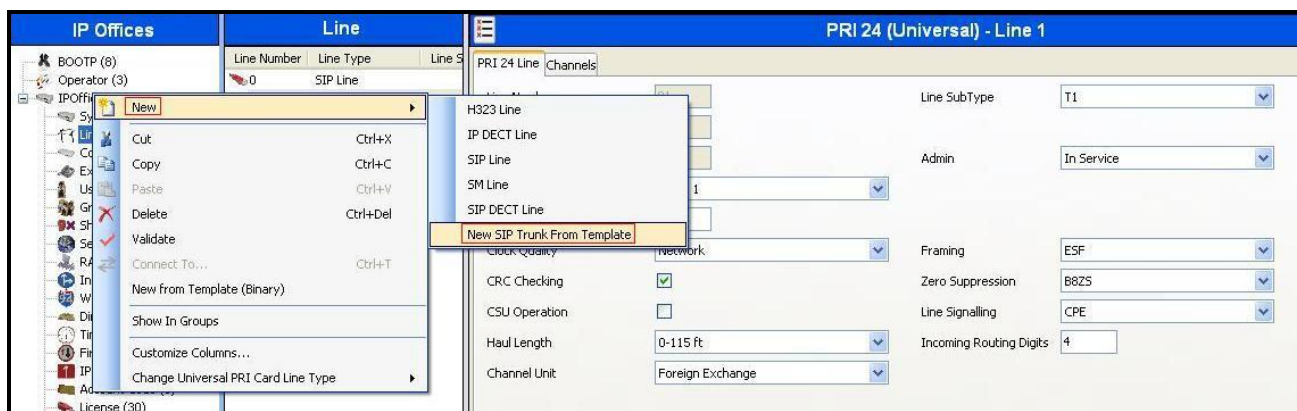


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools** → **Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default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

The screenshot displays the Avaya IP Office configuration window, specifically the 'SIP Line - Line 17' configuration pane. The left sidebar shows the 'IP Offices' tree with 'Line 17' selected. The main configuration area is divided into two panes: 'Line' and 'SIP Line'. The 'SIP Line' pane is active, showing various configuration fields. The 'Line' pane shows a table with columns 'Line Number' and 'Line Type', listing lines 1, 2, 17, and 18. Line 17 is highlighted as a 'SIP Line'.

Line Number	Line Type
1	PRI 24 (Universal)
2	PRI 24 (Universal)
17	SIP Line
18	SIP Line

The 'SIP Line' configuration fields are as follows:

- Line Number:** 17
- ITSP Domain Name:** 192.168.33.8
- Prefix:** (empty)
- National Prefix:** (empty)
- Country Code:** (empty)
- International Prefix:** (empty)
- Send Caller ID:** None
- Association Method:** By Source IP address
- REFER Support:** ☒ (checked)
- Incoming:** Auto
- Outgoing:** Auto
- Method for Session Refresh:** Reinvite
- Session Timer (seconds):** 1200
- Media Connection Preservation:** Disabled
- In Service:** ☒ (checked)
- URI Type:** SIP
- Check OOS:** ☐ (unchecked)
- Call Routing Method:** Request URI
- Originator number for forwarded and twinning calls:** (empty)
- Name Priority:** System Default
- Caller ID from From header:** ☒ (checked)
- Send From In Clear:** ☐ (unchecked)
- User-Agent and Server Headers:** (empty)
- Service Busy Response:** 503 - Service Unavailable
- Action on CAC Location Limit:** Allow Voicemail

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.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '192.168.33.8'. In the 'Network Configuration' section, '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'. The 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. The 'Calls Route via Registrar' checkbox is unchecked. The 'Separate Registrar' field is empty.

SIP Line - Line 17	
SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials	
ITSP Proxy Address: 192.168.33.8	
Network Configuration	
Layer 4 Protocol: UDP	Send Port: 5060
Use Network Topology Info: LAN 2	Listen Port: 5060
Explicit DNS Server(s): 0 . 0 . 0 . 0 0 . 0 . 0 . 0	
Calls Route via Registrar: <input type="checkbox"/>	
Separate Registrar:	

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.

SIP Line - Line 17

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	17 0	1...					0: <Non...	20

Edit Channel

Via: 10.10.98.14

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 0: <None>

Incoming Group: 17

Outgoing Group: 0

Max Calls per Channel: 20

Buttons: Add..., Remove, Edit..., OK, Cancel

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.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'VoIP' tab selected. The window has a tabbed interface with 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'VoIP' tab contains the following settings:

- Codec Selection:** A pull-down menu set to 'Custom'. Below it are two lists: 'Unused' (G.711 ALAW 64K, G.723.1 6K3 MP-MLQ) and 'Selected' (G.711 ULAW 64K, G.729(a) 8K CS-ACELP). Arrows allow moving items between the lists.
- VoIP Silence Suppression:** ☐ (unchecked)
- Allow Direct Media Path:** ☐ (unchecked)
- Re-invite Supported:** ☒ (checked)
- Codec Lockdown:** ☒ (checked)
- PRACK/100rel Supported:** ☐ (unchecked)
- Force direct media with phones:** ☐ (unchecked)
- G.711 Fax ECAN:** ☐ (unchecked)
- Fax Transport Support:** A pull-down menu set to 'G.711'.
- Location:** A pull-down menu set to 'Cloud'.
- Call Initiation Timeout (s):** A numeric field set to '4'.
- DTMF Support:** A pull-down menu set to 'RFC2833'.

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.

The screenshot displays the IP Office configuration window, specifically the 'SIP Line - Line 18' configuration page. The left pane shows a tree view of the configuration hierarchy, with 'Line 18' selected under 'Line'. The main pane is divided into two sections: 'SIP Line' and 'Transport'. The 'SIP Line' section contains the following fields and settings:

- Line Number:** 18
- ITSP Domain Name:** 192.168.33.9
- Prefix:** (empty)
- National Prefix:** (empty)
- Country Code:** (empty)
- International Prefix:** (empty)
- Send Caller ID:** Diversion Header
- Association Method:** By Source IP address
- REFER Support:** (checked)
- Incoming:** Auto
- Outgoing:** Auto
- Method for Session Refresh:** Reinvite
- Session Timer (seconds):** 1200
- Media Connection Preservation:** Disabled

The 'Transport' section contains the following fields and settings:

- In Service:** (checked)
- URI Type:** SIP
- Check OOS:** (checked)
- Call Routing Method:** Request URI
- Originator number for forwarded and twinning calls:** (empty)
- Name Priority:** System Default
- Caller ID from From header:** (checked)
- Send From In Clear:** (unchecked)
- User-Agent and Server Headers:** (empty)
- Service Busy Response:** 503 - Service Unavailable
- Action on CAC Location Limit:** Allow Voicemail

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.

The screenshot shows the 'SIP Line - Line 18' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '192.168.33.9'. In the 'Network Configuration' section, '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'. The 'Explicit DNS Server(s)' field shows two sets of IP addresses, all set to '0 . 0 . 0 . 0'. The 'Calls Route via Registrar' checkbox is unchecked. There is a 'Separate Registrar' field at the bottom.

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.

SIP Line - Line 18

SIP Line | Transport | **SIP URI** | VoIP | T38 Fax | SIP Credentials

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	18 18	1...				N...	0: <Non...	20

Add...
Remove
Edit...

Edit Channel

Via: 10.10.98.14

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 0: <None>

Incoming Group: 18

Outgoing Group: 18

Max Calls per Channel: 20

OK
Cancel

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.

SIP Line - Line 18

SIP Line | Transport | SIP URI | **VoIP** | T38 Fax | SIP Credentials

Codec Selection: Custom

Unused:

- G.711 ALAW 64K
- G.723.1 6K3 MP-MLQ

Selected:

- G.711 ULAW 64K
- G.729(a) 8K CS-ACELP

Fax Transport Support: G.711

Location: Cloud

Call Initiation Timeout (s): 4

DTMF Support: RFC2833

☐ VoIP Silence Suppression
☐ Allow Direct Media Path
☒ Re-invite Supported
☒ Codec Lockdown
☐ PRACK/100rel Supported
☐ Force direct media with phones
☐ G.711 Fax ECAN

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 screenshot displays the Avaya SIP Office configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'Short Code (60)' highlighted. The main area is divided into two panes. The left pane, titled 'Short Code', contains a table with the following data:

Code	Telephone Number	Feature
*20*N#	N	Set Hunt Group
5N;	N"@192.168.33.9"	Dial
FNE00	00	FNE Service

The right pane, titled '5N;; Dial', shows the configuration details for the selected short code. The fields are as follows:

- Code:** 5N;
- Feature:** Dial (selected from a dropdown menu)
- Telephone Number:** N"@192.168.33.9"
- Line Group ID:** 18 (selected from a dropdown menu)
- Locale:** United States (US English) (selected from a dropdown menu)
- Force Account Code:** ☐

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.

The screenshot shows the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree is expanded to 'IPOffice_1'. The 'Short Code' table in the center lists the following entries:

Code	Telephone Number	Feature
*20*N#	N	Set Hunt Group
*SN;	N"@192.168.33.9"	Dial
FNE00	00	FNE Service

The 'FNE00: FNE Service' configuration pane on the right shows the following settings:

- Short Code: FNE00
- Feature: FNE Service
- Telephone Number: 00
- Line Group ID: 0
- Locale: United States (US English)
- Force Account Code: ☐

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.

The screenshot shows the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree is expanded to 'IPOffice_1'. The 'User' table in the center lists the following entries:

Name	Extension
4188	4188
4189	4189
4190	4190
FA/4191	4191
Digital 4192	4192
4197	4197
NoUser	
RemoteManager	

The '4188: 4188' configuration pane on the right shows the following settings:

- SIP Name: 7203624188
- SIP Display Name (Alias): 7203624188
- Contact: 7203624188
- Anonymous: ☐

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.

4188: 4188

Voice Recording Button Programming Menu Programming **Mobility** Group Membership Announcements SIP Personal Directory

☐ Internal Twinning

Twinned Handset <None>

Maximum Number of Calls 1

☐ Twin Bridge Appearances

☐ Twin Coverage Appearances

☐ Twin Line Appearances

☒ **Mobility Features**

☒ **Mobile Twinning**

Twinned Mobile Number (including dial access code) 516139675206

Twinning Time Profile <None>

Mobile Dial Delay (secs) 2

Mobile Answer Guard (secs) 0

☐ Hunt group calls eligible for mobile twinning

☐ Forwarded calls eligible for mobile twinning

☐ Twin When Logged Out

☐ one-X Mobile Client

☒ **Mobile Call Control**

☒ Mobile Callback

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.

Incoming Call Route		
Line Group ID	Incoming Number	Destination
17	7203624192	FNE00
17	7203624191	FAX4191
18	7203624193	4193
17	7203624189	4189 4189
17	7203624190	4190 4190
17	7203624188	4188 4188
17	7203624197	VoiceMail

17 7203624188	
Standard	
Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	7203624188
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

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**.

17 7203624188		
Standard		
TimeProfile	Destination	Fallback Extension
Default Value	4188 4188	

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

The screenshot shows a configuration window for DID number 17 7203624192. The window has a title bar with the number and standard window controls. Below the title bar are three tabs: Standard, Voice Recording, and Destinations. The Destinations tab is active. It contains a table with three columns: TimeProfile, Destination, and Fallback Extension. The first row has a right-pointing arrow in the TimeProfile column, 'Default Value' in the Destination column, and an empty dropdown in the Fallback Extension column. The 'Destination' cell is highlighted with a red rectangle, and 'FNE00' is visible in the dropdown menu.

TimeProfile	Destination	Fallback Extension
▶ Default Value	FNE00	

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

The screenshot shows a configuration window for DID number 17 7203624197. The window has a title bar with the number and standard window controls. Below the title bar are three tabs: Standard, Voice Recording, and Destinations. The Destinations tab is active. It contains a table with three columns: TimeProfile, Destination, and Fallback Extension. The first row has a right-pointing arrow in the TimeProfile column, 'Default Value' in the Destination column, and an empty dropdown in the Fallback Extension column. The 'Destination' cell is highlighted with a red rectangle, and 'VoiceMail' is visible in the dropdown menu.

TimeProfile	Destination	Fallback Extension
▶ Default Value	VoiceMail	

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

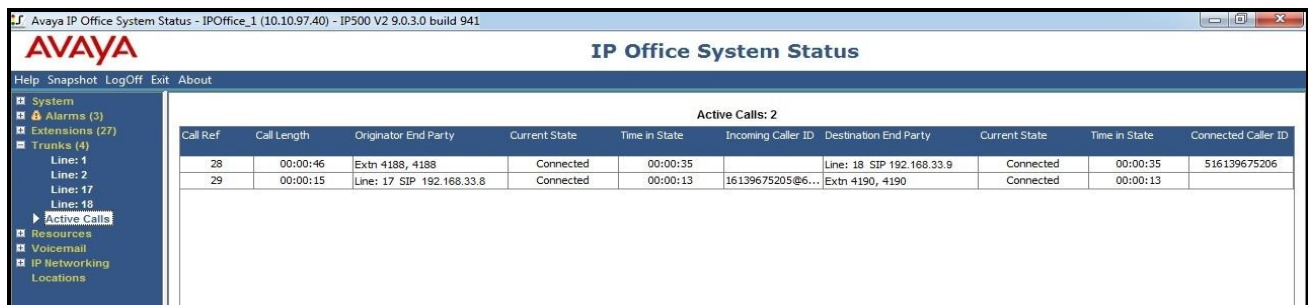
CenturyLink is responsible for the configuration of CenturyLink SIP Trunk Service. The customer must provide the IP address used to reach the Avaya IP Office at the enterprise. CenturyLink will provide the customer the necessary information to configure the SIP connection between Avaya IP Office and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

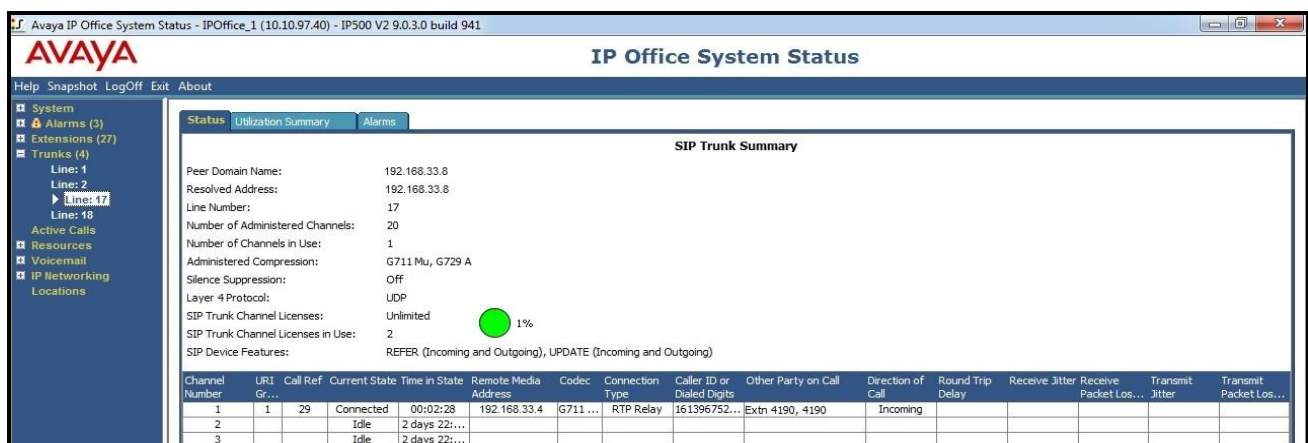
The following steps may be used to verify the configuration:

- 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).



The screenshot shows the 'Active Calls' tab in the Avaya IP Office System Status application. The left sidebar lists various system components, with 'Active Calls' selected. The main pane displays a table of active calls.

Call Ref	Call Length	Originator End Party	Current State	Time in State	Incoming Caller ID	Destination End Party	Current State	Time in State	Connected Caller ID
28	00:00:46	Extn 4188, 4188	Connected	00:00:35		Line: 18 SIP 192.168.33.9	Connected	00:00:35	516139675206
29	00:00:15	Line: 17 SIP 192.168.33.8	Connected	00:00:13	16139675205@6...	Extn 4190, 4190	Connected	00:00:13	

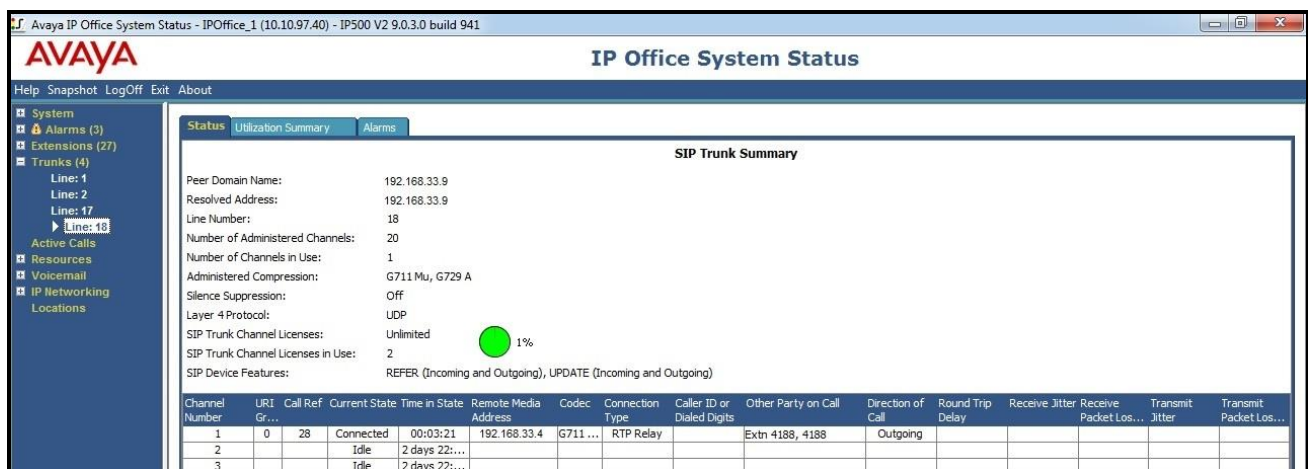


The screenshot shows the 'SIP Trunk Summary' for Line 17. The left sidebar has 'Line: 17' selected. The main pane displays the 'Status' tab with a summary of SIP trunk configuration and a table of channel states.

SIP Trunk Summary

Peer Domain Name: 192.168.33.8
Resolved Address: 192.168.33.8
Line Number: 17
Number of Administered Channels: 20
Number of Channels in Use: 1
Administered Compression: G711 Mu, G729 A
Silence Suppression: Off
Layer 4 Protocol: UDP
SIP Trunk Channel Licenses: Unlimited
SIP Trunk Channel Licenses in Use: 2
SIP Device Features: REFER (Incoming and Outgoing), UPDATE (Incoming and Outgoing)

Channel Number	URI Gr...	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Packet Los...	Transmit Jitter	Transmit Packet Los...
1		29	Connected	00:02:28	192.168.33.4	G711 ...	RTP Relay	161396752...	Extn 4190, 4190	Incoming					
2			Idle	2 days 22:...											
3			Idle	2 days 22:...											



The screenshot shows the 'SIP Trunk Summary' for Line 18. The left sidebar has 'Line: 18' selected. The main pane displays the 'Status' tab with a summary of SIP trunk configuration and a table of channel states.

SIP Trunk Summary

Peer Domain Name: 192.168.33.9
Resolved Address: 192.168.33.9
Line Number: 18
Number of Administered Channels: 20
Number of Channels in Use: 1
Administered Compression: G711 Mu, G729 A
Silence Suppression: Off
Layer 4 Protocol: UDP
SIP Trunk Channel Licenses: Unlimited
SIP Trunk Channel Licenses in Use: 2
SIP Device Features: REFER (Incoming and Outgoing), UPDATE (Incoming and Outgoing)

Channel Number	URI Gr...	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Packet Los...	Transmit Jitter	Transmit Packet Los...
1		28	Connected	00:03:21	192.168.33.4	G711 ...	RTP Relay		Extn 4188, 4188	Outgoing					
2			Idle	2 days 22:...											
3			Idle	2 days 22:...											

- 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.

Line	Module / Slot / Type	Port Number / Address / Domain	Alarms
1	Slot: 1	1	1
2	Slot: 1	2	1
17	SIP	192.168.33.8	0
18	SIP	192.168.33.9	0

- 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>

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