



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring CenturyLink SIP Trunking with Avaya IP Office 6.1 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider CenturyLink and Avaya IP Office 6.1.

CenturyLink SIP Trunking 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 service provider CenturyLink and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 6.1 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

The CenturyLink SIP Trunking service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks directly 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 to connect to CenturyLink SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**.

2.1. Interoperability Compliance Testing

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

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various phone types. Phone types included H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from the Avaya IP Office Softphone.
- Inbound and outbound long holding time call stability.
- Various call types including: local, long distance, international, outbound toll-free, operator service and directory assistance.
- Codec G.711MU and G.729A.
- Caller 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.
- Use of SIP REFER for call transfer to PSTN
- Off-net call forwarding.
- Twinning to mobile phones on inbound calls.

2.2. Test Results

CenturyLink SIP Trunking passed compliance testing.

Items not supported or not tested included the following:

- Inbound toll-free and emergency calls (911) are supported but were not tested as part of the compliance test.
- Faxing between the enterprise site and PSTN were not tested as part of the compliance test since CenturyLink currently does not support T.38 FoIP (Fax over IP) on SIP Trunking.

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

- **OPTIONS from IP Office** – If the periodicity of sending OPTIONS to monitor the SIP trunk connectivity is configured on Avaya IP Office to be larger than the periodicity of OPTIONS from Centurylink, IP Office would effectively cease to send OPTIONS to Century Link. This is expected behavior on Avaya IP Office since it resets the timer for sending OPTIONS starting from the most recently received OPTIONS from the network.
- **Codec Selection** – On outbound calls from the enterprise, CenturyLink would select its codec preference (if supported by the enterprise) as part of the codec negotiation. Thus, it should be noted the enterprise had little control over codec selection for these calls if multiple codecs are configured. Alternatively, CenturyLink could enable configuration for honoring remote codec precedence. Both codec negotiations were tested in the compliance test.
- **Call Display on PSTN Phone** – Call display was not properly updated on PSTN phone involved in a call transfer. After the call transfer was completed, the PSTN phone did not display the actual connected party but instead showed the party that initiated the transfer. SIP signaling trace showed that the enterprise IP Office did not send an UPDATE message to the network to update the call display of the PSTN phone in call transfer. This problem was reported to Avaya IP Office for further investigation.
- **Trunks Busy Condition** – On an inbound call, Avaya IP Office returned a 404 Not Found SIP status message and the PSTN phone would hear the recording "... the number is not connected or no longer in service ...". This is not the best treatment for this situation. For outbound calls, the Avaya IP Office phones would hear busy tones and no SIP signaling messages were sent out to the network.
- **Outbound Operator Assisted Calls** (e.g. 0+10 digits in US) – these calls failed in the compliance test because there was no access to this service from the CenturyLink lab though CenturyLink SIP Trunking production service does support Operator Assisted Calls.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit

<http://support.avaya.com>.

For technical support on CenturyLink SIP Trunking, contact CenturyLink using the Support link at

<http://www.CenturyLink.com>.

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to CenturyLink SIP Trunking service through the public IP network.

Located at the enterprise site is an Avaya IP Office 500v2 with the COMBO6210/ATM4 expansion card which provides connections for 6 digital stations, 2 analog stations, 4 analog trunks to the PSTN as well as 10-channel VCM (Voice Compression Module) for supporting VoIP codecs. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 9600 Series IP Telephone (with H.323 firmware), an Avaya 5410 and an Avaya 6408D Digital Telephones, an Avaya 6210 Analog Telephone and an Avaya IP Office Softphone. The site also has a Windows 2003 Server running Avaya Voicemail Pro for providing voice messaging service to the Avaya IP Office users. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer the Avaya IP Office.

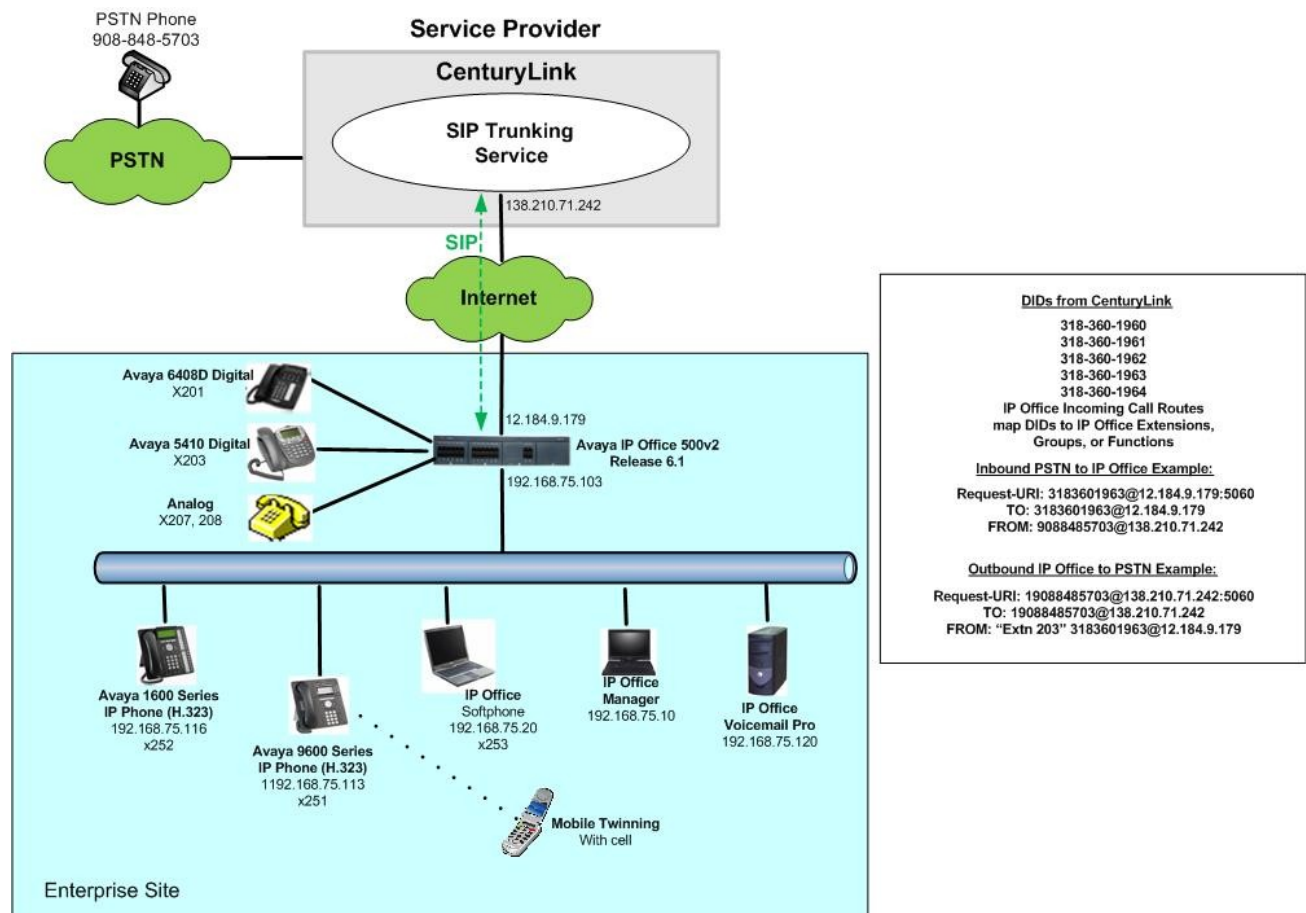


Figure 1: Test Configuration for Avaya IP Office with CenturyLink SIP Trunking Service

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 9 + N digits to send digits across the SIP trunk to CenturyLink. The short code of 9 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 SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

CenturyLink uses the phone number in the From header of a SIP INVITE message to authenticate the calling party. Thus, a call will be rejected by the network unless the From header contains a number known to CenturyLink. This is especially important for calls inbound from the PSTN which are redirected back to the PSTN by call forwarding or twinning. For call forwarding, Avaya IP Office always sends the number of the forwarding phone in the From header. This is a number known to CenturyLink. As a result, the call display on the destination phone shows the forwarding party not the original caller. For twinning, this behavior can be slightly altered through configuration. See **Sections 5.3** and **5.4** for details.

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	6.1 (5)
Avaya IP Office COMBO6210/ATM4 Module	6.1 (5)
Avaya Voicemail Pro	6.1.16
Avaya IP Office Manager	8.1 (5)
Avaya 1616 IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.2.2
Avaya 9620 IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.1
Avaya Digital Telephones (5410)	N/A
Avaya Digital Telephones (6408D)	N/A
Avaya IP Office Softphone	3.1.2.17 59616
Avaya 6210 Analog Telephone	N/A

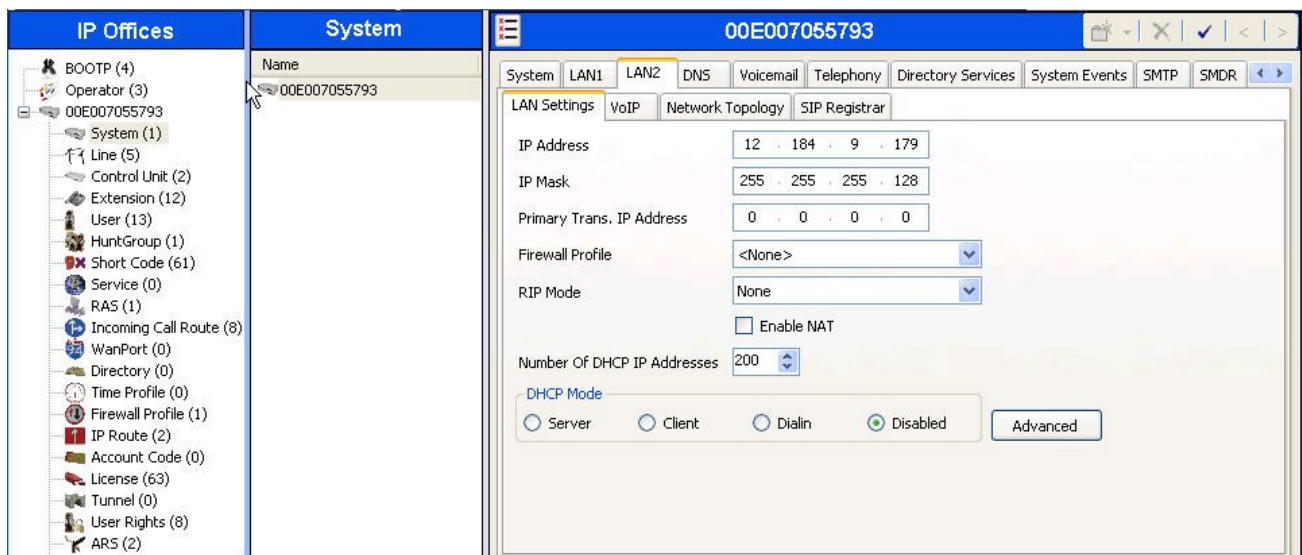
CenturyLink Components	
Equipment	Release
Acme Packet Net-Net 4250 SBC	R6.1
Broadsoft SoftSwitch	R16 sp2
Media Gateway SONUS GSX9000	V07.02.05R000

5. Configure IP Office

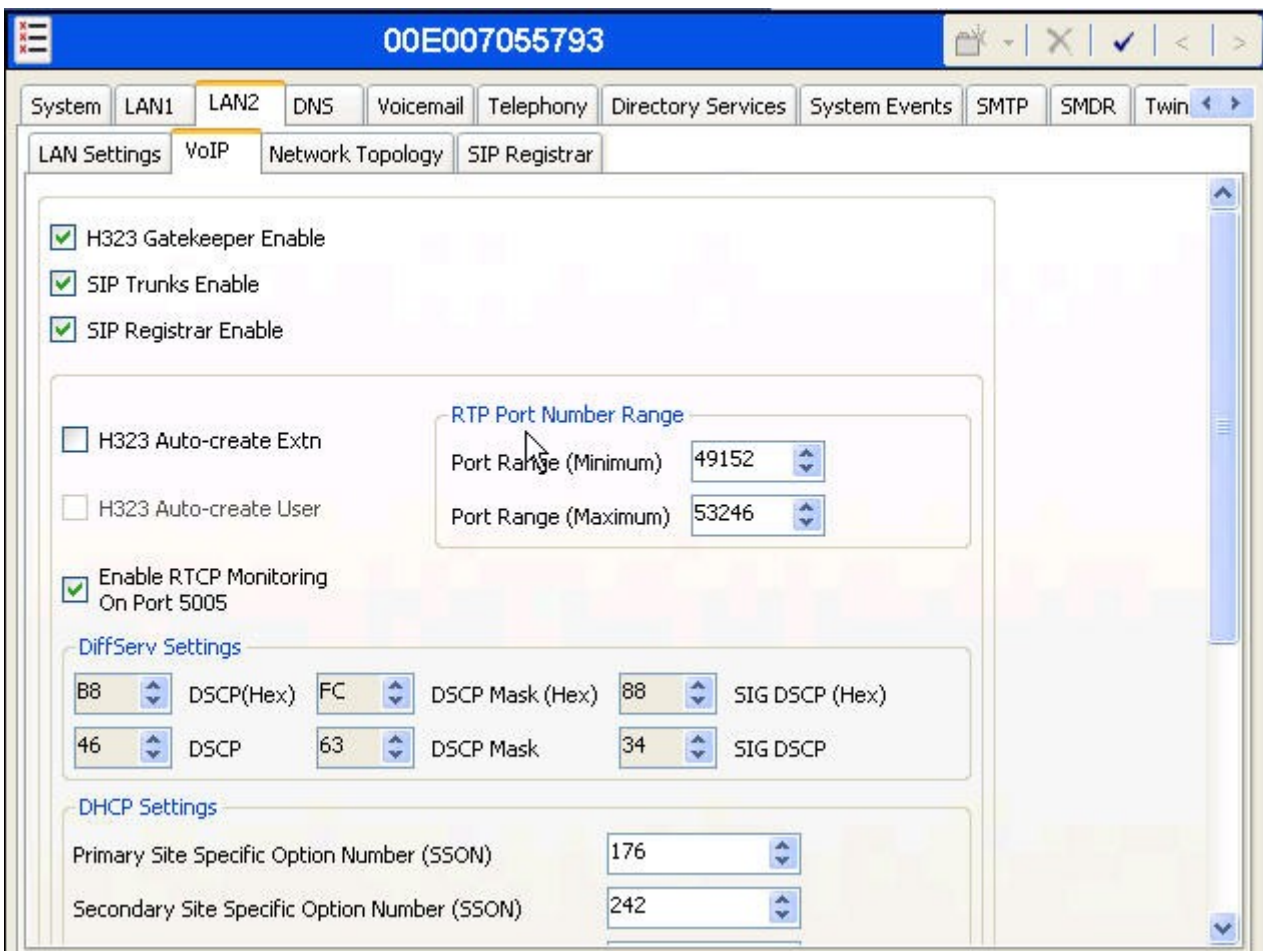
This section describes the Avaya IP Office configuration to support connectivity to CenturyLink SIP Trunking service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** 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 LAN interface to the enterprise site and IP Office Softphone support) is assumed to already be in place.

5.1. LAN2 Settings

In the sample configuration, the MAC address **00E007055793** was used as the system name and the WAN port was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office. To access the LAN2 settings, first navigate to **System → 00E007055793** in the Navigation and Group Panes and then navigate to the **LAN2 → LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.

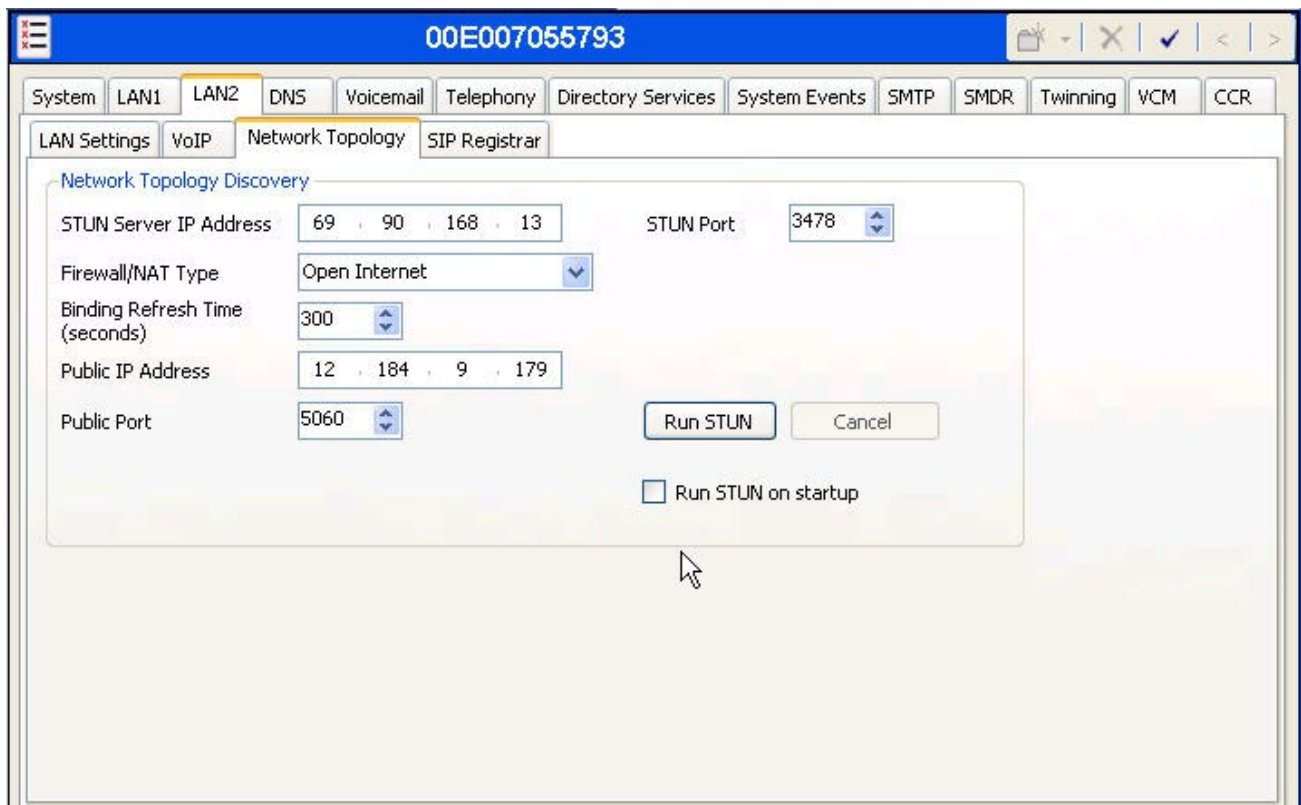


Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to CenturyLink. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series and the 9600-Series IP Telephones used in the sample configuration. The **SIP Registrar Enable** box is checked to allow Avaya IP Office Softphone usage. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements.



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

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. 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 **Binding Refresh Time (seconds)** to **300**. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. See **Section 5.10** for complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- All other parameters should be set according to customer requirements.



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

5.2. System Telephony Settings

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

The screenshot displays the 'Telephony' configuration page for extension 00E007055793. The interface includes a top navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony (selected), Directory Services, System Events, SMTP, SMDR, Twinning, and VCM. Below this, there are sub-tabs for Telephony, Tones & Music, and Call Log. The main configuration area is divided into several sections:

- Analogue Extensions:** Includes dropdown menus for Default Outside Call Sequence (Normal), Default Inside Call Sequence (Ring Type 1), and Default Ring Back Sequence (Ring Type 2). A checkbox for Restrict Analogue Extension Ringer Voltage is unchecked.
- Companding Law:** Features two columns: 'Switch' and 'Line'. Both have radio buttons for ULAW (selected) and ALAW. There are also checkboxes for DSS Status, Auto Hold (checked), Dial By Name (checked), Show Account Code (checked), Inhibit Off-Switch Forward/Transfer (unchecked), Restrict Network Interconnect (unchecked), Drop External Only Impromptu Conference (unchecked), and Visually Differentiate External Call (unchecked).
- Call Handling Parameters:** A list of numeric settings with up/down arrows: Dial Delay Time (secs) at 4, Dial Delay Count at 0, Default No Answer Time (secs) at 15, Hold Timeout (secs) at 120, Park Timeout (secs) at 300, Ring Delay (secs) at 5, Call Priority Promotion Time (secs) set to Disabled, Default Currency set to USD, and Automatic Codec Preference set to G.711 ULAW 64K.

5.3. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affects 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 **System→Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.4**).

If **Send original calling party information for Mobile Twinning** on the **System→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 in the first paragraph.

For the compliance test, the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab was checked which overrides any setting of the **Send Caller ID** parameter on the **SIP Line** form.



5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and CenturyLink SIP Trunking service. 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 LAN2 IP address (12.184.9.179) so that IP Office uses 12.184.9.179 as the host portion of SIP headers such as the From header and Diversion header.
- Set **Send Caller ID** to *None*. For the compliance test, this parameter was ignored since the **Send original calling party information for Mobile Twinning** option is checked in **Section 5.3**.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Default values may be used for all other parameters.

The area of the screen entitled **REFER Support** is new for Avaya IP Office Release 6.1. In the following screen, the default values of “Auto” are shown for **Incoming** and **Outgoing**, which is sufficient to effectively disable use of SIP REFER for call transfers. To enable SIP REFER, select “Always” from the drop-down menu for **Incoming** and **Outgoing**. In the compliance test, both configurations were successfully tested to transfer a call between a PSTN phone and an enterprise phone to a second PSTN phone.

The screenshot displays the Avaya IP Office configuration interface for a SIP Line. The interface is divided into three main panes:

- IP Offices:** A tree view on the left showing the hierarchy of the system, including BOOTP (4), Operator (3), System (1), Line (5), Control Unit (2), Extension (12), User (13), HuntGroup (1), Short Code (61), Service (0), RAS (1), Incoming Call Route (8), WanPort (0), Directory (0), Time Profile (0), Firewall Profile (1), IP Route (2), Account Code (0), License (63), and Tunnel (0).
- Line:** A table listing the configured lines. Line 17 is selected and highlighted as a SIP Line.
- SIP Line - Line 17:** The configuration details for the selected line, including:
 - Line Number:** 17
 - ITSP Domain Name:** 12.184.9.179
 - In Service:**
 - Use Tel URI:**
 - Check OOS:**
 - Call Routing Method:** Request URI
 - Originator number for forwarded and twinning calls:** (empty field)
 - Prefix:** (empty field)
 - National Prefix:** 0
 - Country Code:** (empty field)
 - International Prefix:** 00
 - Send Caller ID:** None
 - REFER Support:** (checked), with **Incoming** and **Outgoing** both set to Auto.

Select the **Transport** tab. The **ITSP Proxy Address** is set to the CenturyLink IP Address provided by CenturyLink. As shown in **Figure 1**, this IP Address is **138.210.71.242**. In the **Network Configuration** area, **UDP** is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by CenturyLink. The **Use Network Topology Info** parameter is set to **LAN 2**. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab. Other parameters retain default values in the screen below.

The screenshot shows the configuration window for 'SIP Line - Line 17'. The 'Transport' tab is selected. The 'ITSP Proxy Address' is set to '138.210.71.242'. In the 'Network Configuration' section, 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', and 'Use Network Topology Info' is set to 'LAN 2'. 'Listen Port' is also '5060'. 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty.

Field	Value
ITSP Proxy Address	138.210.71.242
Layer 4 Protocol	UDP
Send Port	5060
Use Network Topology Info	LAN 2
Listen Port	5060
Explicit DNS Server(s)	0 . 0 . 0 . 0
Calls Route via Registrar	<input checked="" type="checkbox"/>
Separate Registrar	

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

- Set **Local URI**, **Contact** and **Display Name** to **Internal Data**. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.6**.
- Set **PAI** to **Internal Data**. With this setting IP Office will populate the SIP P-Asserted-Identity header on outgoing calls with the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.6**.
- The **Registration** parameter is left at the default setting **0: <None>** since CenturyLink SIP Trunking service does not require registration.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **17** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

The screenshot shows the 'SIP Line - Line 17' configuration window. The 'SIP URI' tab is selected, displaying a table with one entry:

Channel	Groups	Via	Local URI	Contact	Display Name	PAI
1	17 17	12.184.9.179				

Below the table is the 'Edit Channel' dialog box with the following fields:

- Via: 12.184.9.179
- 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: 17
- Max Calls per Channel: 10

Buttons for 'Add...', 'Remove', 'OK', and 'Cancel' are visible.

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

- The **Compression Mode** was configured using the **Advanced** button, allowing an explicit ordered list of codecs to be specified. Check marks next to the **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** codecs cause Avaya IP Office to include both codes in the Session Description Protocol (SDP) offer, in that order.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box. By unchecking the **VoIP Silence Suppression** box, calls can be established with the G.729 codec but without silence suppression.
- Uncheck the **Fax Transport Support** box. This box should be unchecked since T.38 faxing is not supported by CenturyLink.
- Check the **Re-invite Supported** box.
- Default values may be used for all other parameters.

The screenshot shows the configuration window for 'SIP Line - Line 17'. The 'VoIP' tab is selected. The 'Compression Mode' is set to 'Advanced', which has opened a list of codecs. The first two, 'G.711 ULAW 64K' and 'G.729(a) 8K CS-ACELP', are checked. Other options include 'G.711 ALAW 64K' and 'G.723.1 6K3 MP-MLQ'. To the right, there are checkboxes for 'VoIP Silence Suppression' (unchecked), 'Fax Transport Support' (unchecked), 'Re-invite Supported' (checked), and 'Use Offerer's Preferred Codec' (unchecked). The 'Call Initiation Timeout (s)' is set to 4, and 'DTMF Support' is set to RFC2833.

5.5. Short Code

Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New**. 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 “9N;” 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, **9N;**. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@138.210.71.242"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. The IP address 138.210.71.242 is the IP address of the CenturyLink SIP proxy.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4**. This short code will use this line group when placing the outbound call.

The screenshot displays the Avaya IP Office configuration interface. The left pane shows the 'IP Offices' tree with 'Short Code (61)' selected. The middle pane shows a list of short codes, with '9N;: N"@138.210.71.242"' selected. The right pane shows the configuration details for this short code:

Field	Value
Code	9N;
Feature	Dial
Telephone Number	N"@138.210.71.242"
Line Group Id	17
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>

The simple “9N;” short codes illustrated above does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code 6N is illustrated for access to ARS. When the Avaya IP Office user dials 6 plus any number N, rather than being directed to a specific **Line Group Id**, the call is directed to **Line Group ID “50: Main”**, configurable via ARS. See **Section 5.8** for example ARS route configuration for “50: Main” as well as a backup route.

The screenshot shows a configuration window titled "6N: Dial". The window has a blue header bar with the title and standard window controls (minimize, maximize, close, checkmark, back, forward). Below the header is a tab labeled "Short Code". The main area contains several fields:

- Code:** 6N
- Feature:** Dial (dropdown menu)
- Telephone Number:** N
- Line Group Id:** 50: Main (dropdown menu)
- Locale:** (dropdown menu)

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

5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. To configure these settings, first 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 “Allan”. 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 (**Section 5.4**). The example below shows the settings for user Allan. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from CenturyLink. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user’s information from the network.

The screenshot displays the Avaya user configuration interface. On the left, the 'IP Offices' pane shows a tree view with 'User (13)' selected. The center pane, titled 'User', lists users with their names and extensions: Allan (251), Extn201 (201), Extn202 (202), Extn203 (203), Extn204 (204), Extn205 (205), Extn206 (206), Extn207 (207), Extn208 (208), James (252), John (253), NoUser, and RemoteManager. The right pane, titled 'Allan: 251', shows the 'SIP' tab selected. The 'SIP' tab contains the following fields: 'SIP Name' (3183601960), 'SIP Display Name (Alias)' (Allan), and 'Contact' (3183601960). There is an unchecked checkbox labeled 'Anonymous'. At the bottom of the right pane are 'OK', 'Cancel', and 'Help' buttons.

From **Figure 1**, note that 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 Allan. 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 917328963305. Other options can be set according to customer requirements.

The screenshot shows a web-based configuration interface for a user named Allan: 251*. The interface has a blue header bar with the user name and a toolbar with icons for help, refresh, save, and navigation. Below the header is a tabbed menu with the following tabs: Dial In, Voice Recording, Button Programming, Menu Programming, Mobility (selected), Phone Manager Options, and Hunt Group Membership. The main content area is divided into two sections. The first section, 'Internal Twinning', is currently unchecked and contains a dropdown for 'Twinned Handset' set to '<None>', a dropdown for 'Maximum Number of Calls' set to '1', and three unchecked checkboxes: 'Twin Bridge Appearances', 'Twin Coverage Appearances', and 'Twin Line Appearances'. The second section, 'Mobility Features', is checked and contains a sub-section 'Mobile Twinning' which is also checked. This sub-section includes a text field for 'Twinned Mobile Number (including dial access code)' containing '917328963305', a dropdown for 'Twinning Time Profile' set to '<None>', a text field for 'Mobile Dial Delay (secs)' set to '2', and a spinner for 'Mobile Answer Guard (secs)' set to '0'. Below these are four unchecked checkboxes: 'Hunt group calls eligible for mobile twinning', 'Forwarded calls eligible for mobile twinning', 'Twin When Logged Out', and 'one-X Mobile Client'. At the bottom of the section are three more unchecked checkboxes: 'Mobile Call Control' and 'Mobile Callback'.

5.7. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to *Any Voice*.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4**.
- Set the **Incoming Number** to the incoming number on which this route should match. Matching is right to left.
- Default values can be used for all other fields.

Line Group Id	Incoming Number	Destination
0		200 Main
0		DialIn
1		203 Extn203
17	3183601964	252 James
17	3183601963	203 Extn203
17	3183601962	201 Extn201
17	3183601961	253 John
17	3183601960	251 Allan

17 3183601960

Standard | Voice Recording | Destinations

Bearer Capacity: Any Voice

Line Group Id: 17

Incoming Number: 3183601960

Incoming Sub Address:

Incoming CLI:

Locale: United States (US English)

Priority: 1 - Low

Tag:

Hold Music Source: System Source

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to 318-360-1960 on line 17 are routed to extension 251.

TimeProfile	Destination	Fallback Extension
Default Value	251 Allan	

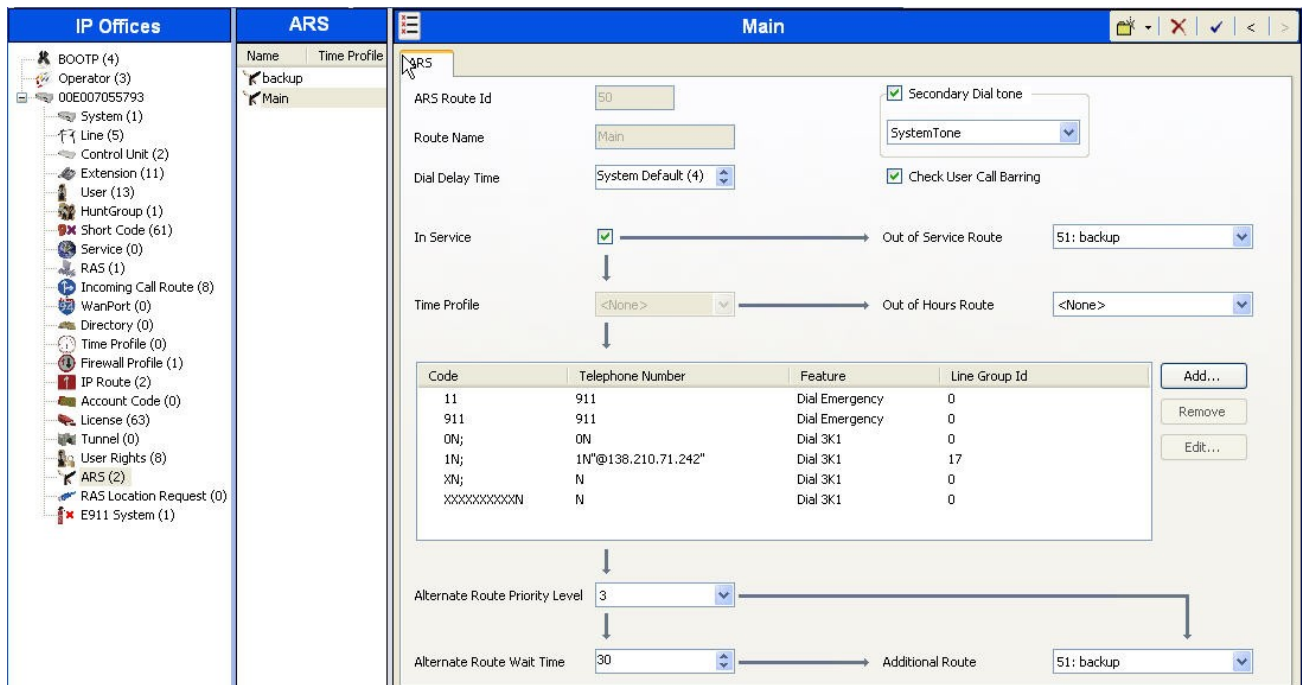
5.8. ARS and Alternate Routing

While detailed coverage of Automatic Route Selection (ARS) is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations. ARS is illustrated here mainly to illustrate alternate routing should the SIP Line be out of service or temporarily not responding.

Optionally, ARS can be used rather than the simple “9N;” short code approach documented in **Section 5.5**. With ARS, a secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named “Main”. The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter. IP Office short codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service Route, and the means to manually activate the Out of Service Route can be helpful for scheduled maintenance or other known service-affecting events for the primary route.



Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., 6N in **Section 5.5**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 6-1-908-848-5704, the call would be directed to Line Group 17, the SIP Line configured and described in these Application Notes. If Line Group 17 cannot be used, the call can automatically route to the route name configured in the **Additional Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

The following screen shows an example ARS configuration for the route named "backup" with ARS Route ID 51. Continuing the example, if the user dialed 6-1-908-848-5704, and the call could not be routed via the primary route "50: Main" described above, the call will be delivered to this "backup" route. Per the configuration shown below, the call will be delivered to Line Group 1, using an analog trunk connecting the Avaya IP Office to the PSTN as a backup connection. In this case, the originally dialed number (sans the short code 6) will be dialed as is through the analog trunk to the PSTN. Additional codes (e.g., 411, 0+10, etc.) can be added to the ARS route by pressing the **Add...** button to the right of the list of previously configured codes.

The screenshot shows the configuration for an ARS route named "backup". The fields are as follows:

- ARS Route Id: 51
- Route Name: backup
- Dial Delay Time: System Default (4)
- In Service:
- Out of Service Route: <None>
- Time Profile: <None>
- Out of Hours Route: <None>
- Alternate Route Priority Level: 3
- Alternate Route Wait Time: 30
- Additional Route: <None>

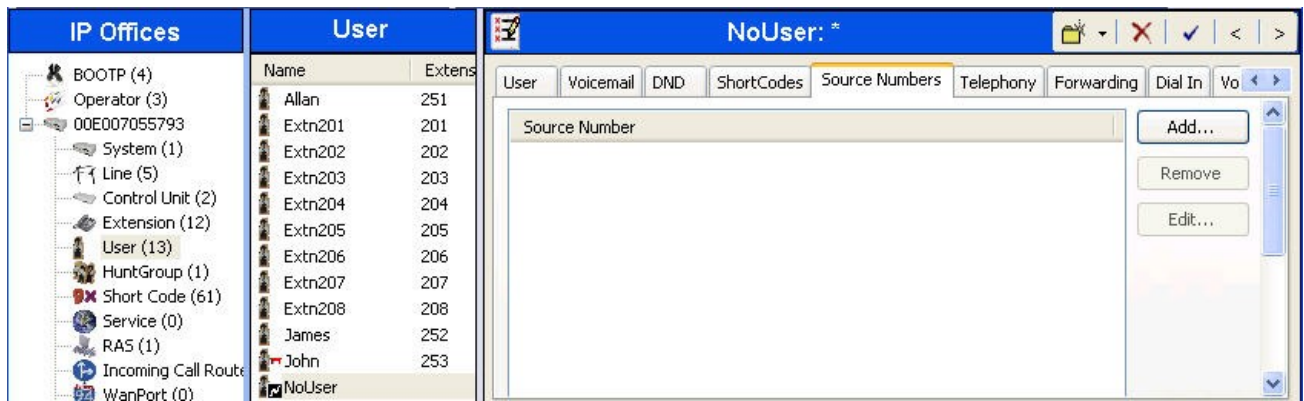
Code	Telephone Number	Feature	Line Group Id
11	911	Dial Emergency	0
911	911	Dial Emergency	0
1N;	1N	Dial	1

In the testing associated with the configuration, calls were successfully delivered to SIP Line 17 via the primary ARS route “50: Main” or to the analog trunk via the backup ARS route shown above. When the primary route experiences a network outage, Avaya IP Office successfully routed the call via the backup route.

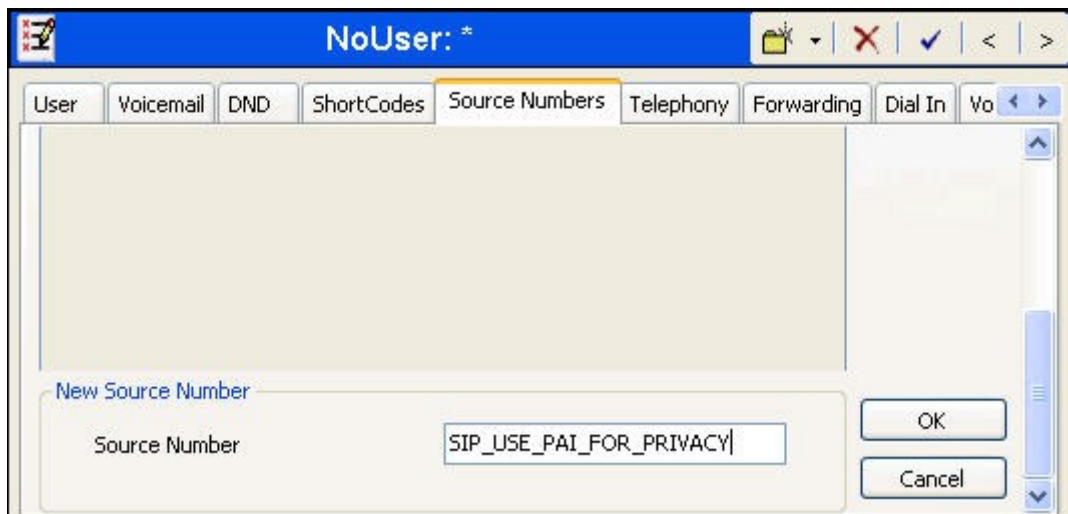
5.9. Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “restricted” and “anonymous” respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. For the compliance test, PAI was used for the purposes of privacy.

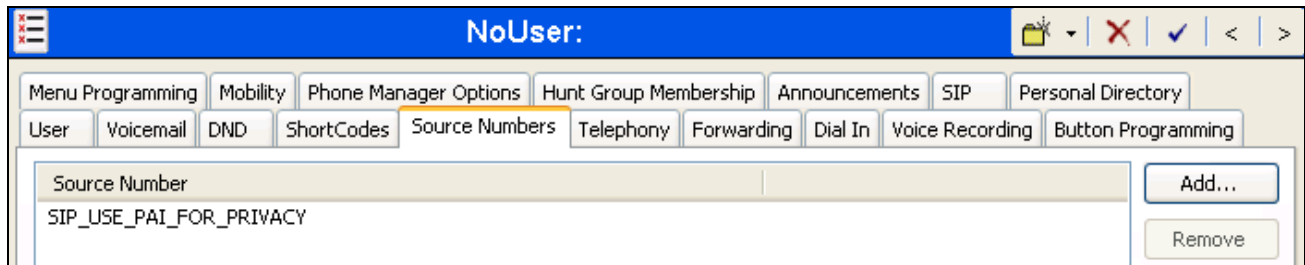
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → noUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_USE_PA_I_FOR_PRIVACY**. Click **OK**.



The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.

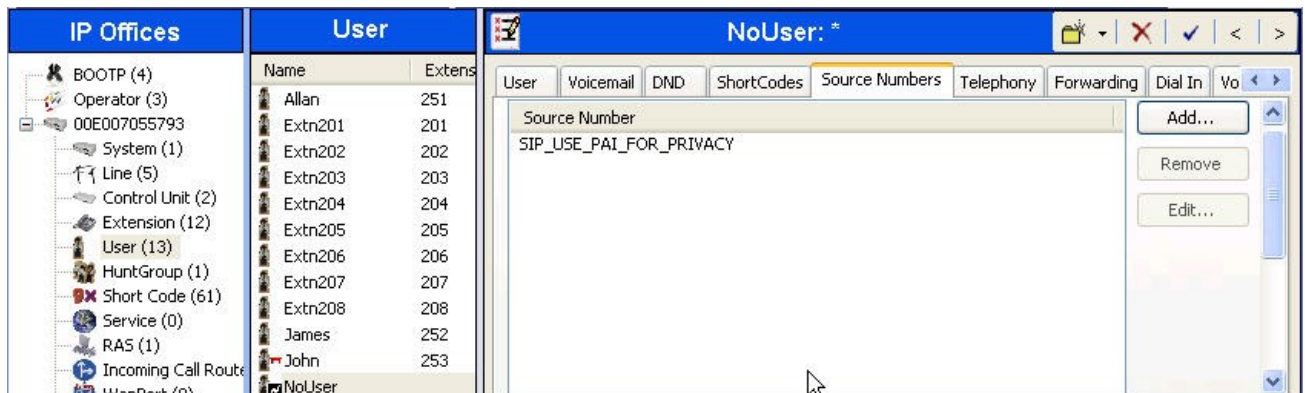


5.10. SIP Options

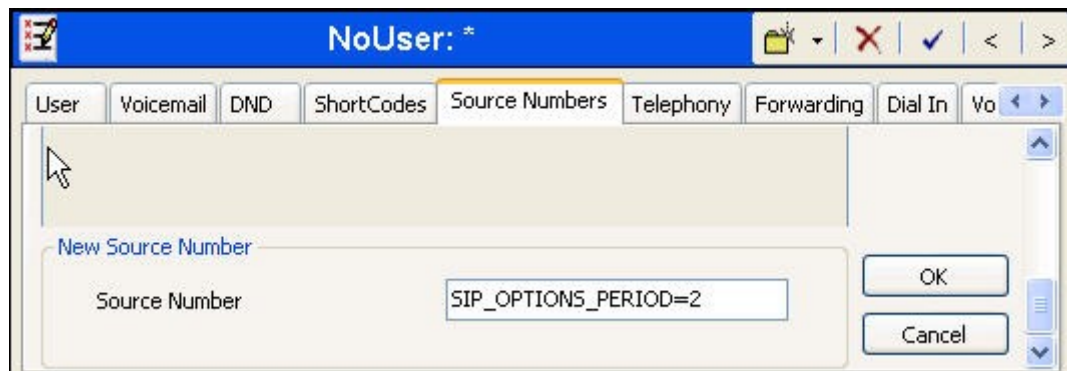
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.1** and the **SIP_OPTIONS_PERIOD**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

- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used.
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

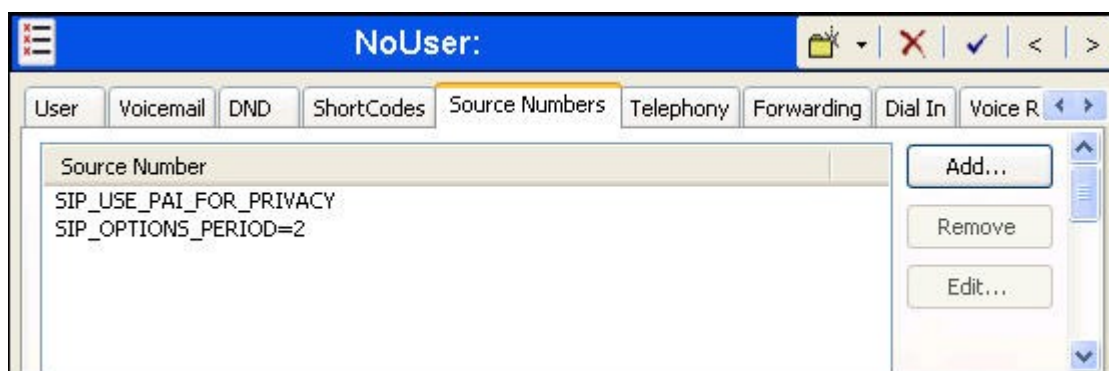
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User** → **noUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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



The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an **OPTIONS** period of 2 minutes was desired. The **Binding Refresh Time** was set to **300** seconds (5 minutes) in **Section 5.1**. The **SIP_OPTIONS_PERIOD** was set to 2 minutes. Avaya IP Office chose the **OPTIONS** period as the smaller of these two values (2 minutes). Click the **OK** button (not shown).



5.11. 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 Trunking Configuration

CenturyLink is responsible for the configuration of CenturyLink SIP Trunking service. The customer will need to 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 Avaya IP Office SIP connection to CenturyLink. The provided information from CenturyLink includes:

- IP address of the CenturyLink SIP proxy.
- Supported codecs
- DID numbers
- IP addresses and port numbers used for signaling or media through any security devices.

7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is *Idle* for each channel.

The screenshot shows the Avaya IP Office System Status application. The left sidebar contains a navigation tree with categories like System, Alarms (2), Extensions (10), Trunks (5), Active Calls, Resources, Voicemail, and IP Networking. The main window is titled "IP Office System Status" and has tabs for Status, Utilization Summary, and Alarms. The Status tab is active, displaying a "SIP Trunk Summary" section with the following details:

- Peer Domain Name: 12.184.9.179
- Resolved Address: 138.210.71.242
- Line Number: 17
- Number of Administered Channels: 10
- Number of Channels in Use: 0
- Administered Compression: Auto
- Silence Suppression: Off
- SIP Trunk Channel Licences: Unlimited
- SIP Trunk Channel Licences in Use: 0
- SIP Device Features: (indicated by a green circle and 0%)

Below the summary is a table showing the status of 10 channels:

Channel Number	URI Gro Ref	Call Ref	Current State	Time in State	Remote Address	RT Code	Connect Type	Caller ID Dialed Di	Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Loss Fra	Transmit Jitter	Tr Lc
1			Idle	23:18:34											
2			Idle	23:37:47											
3			Idle	1 day 00:57:30											
4			Idle	1 day 00:57:30											
5			Idle	1 day 00:57:30											
6			Idle	1 day 00:57:30											
7			Idle	1 day 00:57:30											
8			Idle	1 day 00:57:30											
9			Idle	1 day 00:57:30											
10			Idle	1 day 00:57:30											

At the bottom of the application, there are buttons for Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As... The system time is 4:08:20 PM and the status is Online.

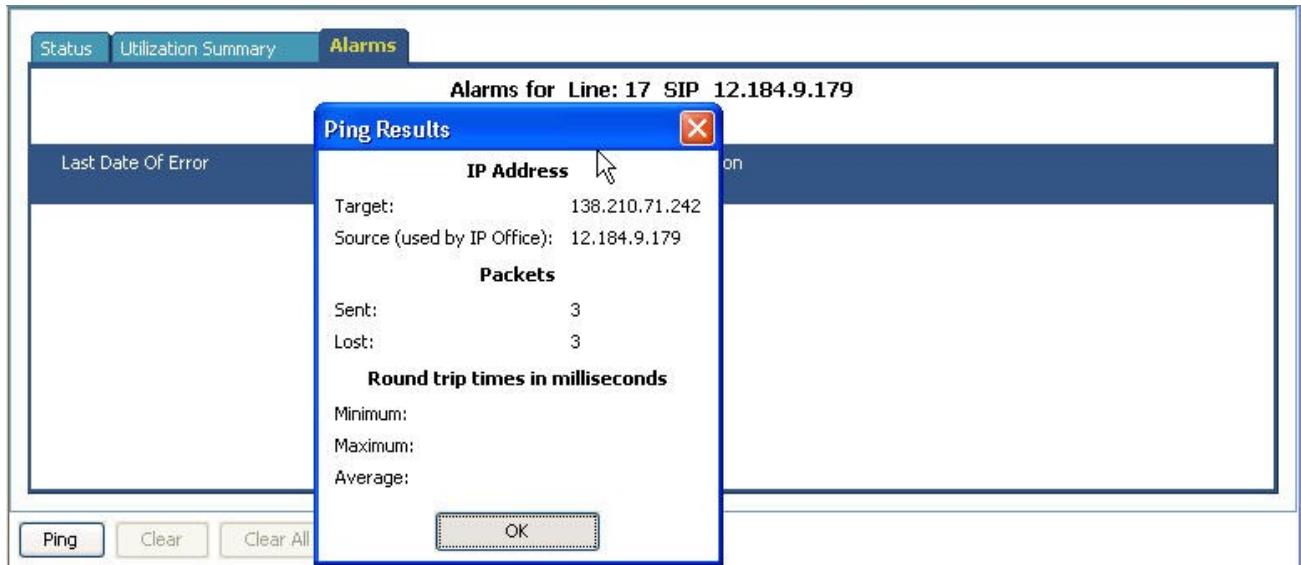
- Select the **Alarms** tab and verify that no alarms are active on the SIP line.

The screenshot shows the Avaya IP Office System Status application with the Alarms tab selected. The title is "Alarms for Line: 17 SIP 12.184.9.179". Below the title is a table with the following columns:

Last Date Of Error	Occurrences	Error Description

At the bottom of the application, there are buttons for Ping, Clear, Clear All, Print..., and Save As... The system time is 4:08:20 PM and the status is Online.

- Click the **Ping** button at the bottom, verify that Avaya IP Office can reach the CenturyLink SIP Proxy Server across the public IP network.



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

8. Conclusion

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

9. Additional References

- [1] *IP Office 6.1 Installation Manual*, Document number 15-601042, November 2010.
- [2] *IP Office Release 6.1 Manager 8.1*, Document number 15-601011, November 2010.
- [3] *IP Office Release 6.1 Voicemail Pro Installation and Maintenance*, Document Number 15-601063, November 5, 2010
- [4] *IP Office Release 6.0 System Status Application*, Document number 15-601758, February 2010.
- [5] *IP Office System Monitor*, Document Number 15-601019, November 28, 2008

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/>

Product documentation for CenturyLink SIP Trunking is available from CenturyLink.

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