Application Notes for Configuring SIP Trunking between CenturyLink SIP Trunk (Legacy Qwest) Service and Avaya IP Office R8.0 (16) – Issue 1.0

Abstract

These Application Notes describe the steps required to configure Session Initiation Protocol (SIP) trunking between a CenturyLink SIP Trunk (Legacy Qwest) Service and an Avaya IP Office R8.0 (16) telephony solution.

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.

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1. Introduction

These Application Notes describe the steps required to configure Session Initiation Protocol (SIP) trunking between CenturyLink and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office R8.0 (16) with Avaya H.323, SIP, Digital and analog endpoints. These Application Notes correspond to the CenturyLink SIP Trunk Service offered using a Sonus Network Border Switch (NBS) in the network, otherwise known as Legacy Qwest SIP Trunk Service.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 is the primary specification governing this protocol. SIP manages the establishment and termination of connections, and the transfer of related information such as the desired codec, calling party identity, etc. Within these Application Notes, SIP is used as the signaling protocol between Avaya IP Office and the CenturyLink SIP Trunk Service solution.

Customers using this Avaya IP Office telephony solution with the CenturyLink SIP Trunk Service are able to place and receive PSTN calls via a dedicated broadband Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI. The text and coverage diagram (**Figure 1**) summarizes the CenturyLink SIP Trunk Service at the time of writing these Application Notes. Please consult CenturyLink for the most current description of capabilities.

1.1. CenturyLink SIP Trunk

CenturyLink SIP Trunk Service enables the origination and termination of local and long-distance calls, as well as domestic, international and toll-free traffic across a single broadband connection. It is designed to work in conjunction with CenturyLink's Networking service, which includes a secure, managed, and scalable suite of wide area network (WAN) services.

2. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify SIP trunk interoperability between the CenturyLink SIP Trunk Service and an Avaya IP office Telephony Solution.

A simulated enterprise site using an Avaya IP Office telephony solution was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available SIP Trunk Service provided by CenturyLink.

The compliance test included the following:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by CenturyLink. Incoming PSTN calls were made to Avaya H.323, SIP, digital and analog telephones at the enterprise.
- Outgoing calls from the enterprise site were completed via CenturyLink to PSTN destinations. Outgoing calls from the enterprise to the PSTN were made from Avaya H.323, SIP, digital and analog telephones.

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- Various call types were tested including: local, long distance, international, inbound/outbound toll-free, emergency, T.38 fax, operator and directory assistance.
- Calls were placed using both the G.729A and G.711MU codecs.
- DTMF transmission using RFC 2833 for inbound and outbound calls.
- User features such as hold and resume, forwarding, transfer and conference.
- Caller ID Presentation and Caller ID Restriction.
- Mobility Features: Mobile Twinning to a mobile phone was tested and verified.

2.1. Interoperability Compliance Testing

A simulated enterprise site consisting of an Avaya IP Office telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the generally available SIP trunking solution provided by CenturyLink. This allowed the enterprise site to use SIP trunking for calls to and from the PSTN.

2.2. Test Results

The test objectives were verified. Interoperability testing of the sample configuration was completed with successful results for the CenturyLink SIP Trunk Service, except as noted below.

Items not supported or not tested included the following:

- The use of the SIP REFER method was not fully supported at the time of this compliance test. While it was tested, there were a few issues that were actively being worked at the time of writing these Application Notes.

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 SIP Trunk Service, contact CenturyLink at http://www.CenturyLink.com/.

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3. Reference Configuration

Figure 1 illustrates a sample Avaya IP Office telephony solution connected to CenturyLink SIP Trunk that was utilized for compliance testing. Since public IP addresses were used during compliance testing, those IP address are not shown in the figure below and are masked (at least partially) throughout the document.

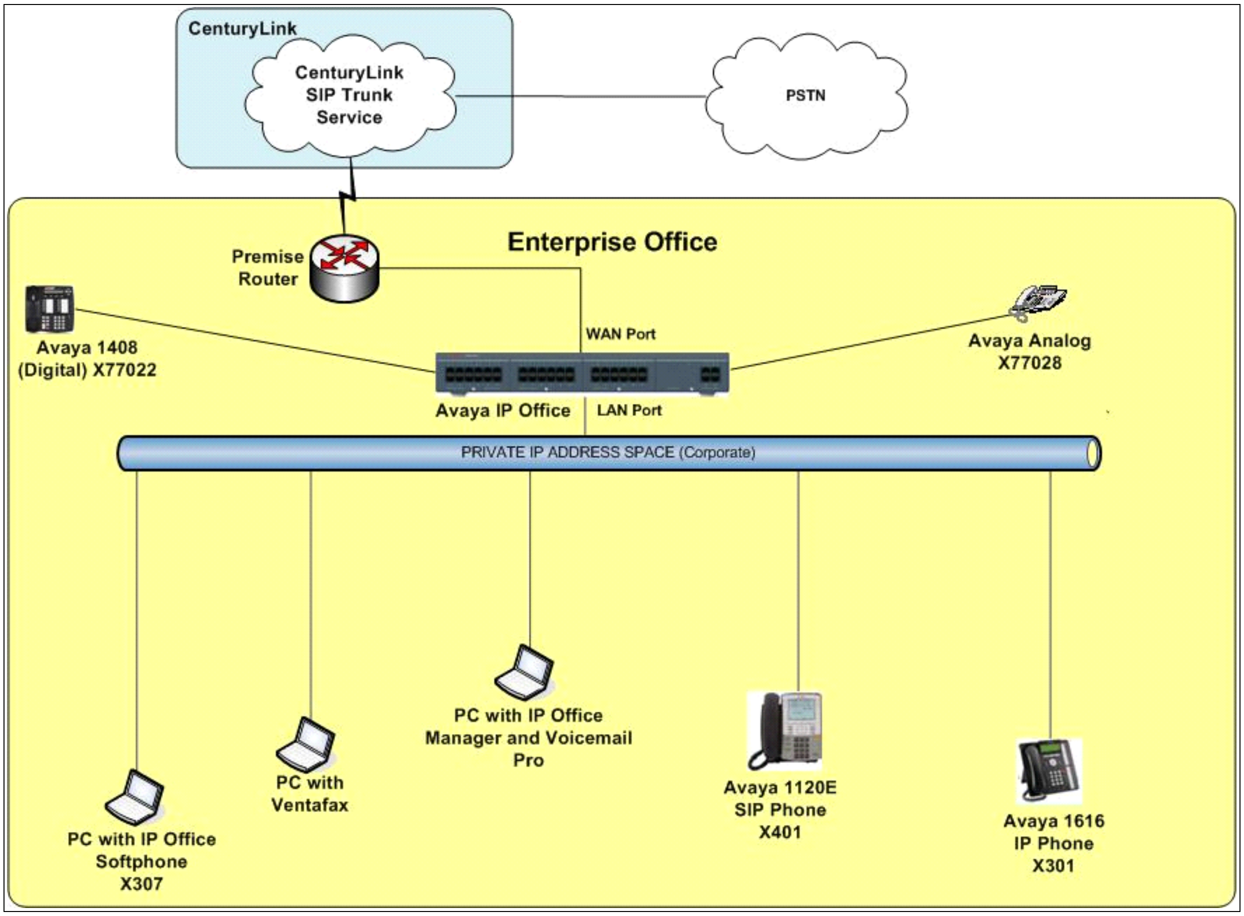

Figure 1: Avaya IP Office telephony system connected to CenturyLink SIP Trunk Service

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4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Equipment	**Software/Firmware**
Avaya IP Office 500	Avaya IP Office 500v2 R8.0.(16)
Avaya IP Office Manager (Windows PC)	Avaya IP Office Manager 10.0.(16)
Avaya IP Office Voicemail Pro	Avaya IP Office Voicemail Pro 7.0(17)
Avaya 1120E IP Telephone (SIP)	SIP1120 Load Version 04.01.13.00
Avaya 1616-I IP Telephone (H.323)	ha1616ua1_3000.bin
Avaya 1408 Digital Telephone	-
Analog Telephone and Fax	-
CenturyLink Solution Components	
Equipment	**Software/Firmware**
CenturyLink Network Border Switch (NBS)	7.3.5R6

5. Configure Avaya IP Office

This section describes the steps required for configuring a SIP trunk on Avaya IP Office. The procedures include the following areas:

- Verify SIP Trunk Channels License
- Configure WAN interface
- Configure SIP OPTIONS timer for “keep alive” function with CenturyLink
- Enable SIP Trunk
- Create the SIP line to CenturyLink
- Configure SIP URI parameters for the SIP Line
- Configure T38 Fax parameters for the SIP Line
- Configure a short code to route calls to CenturyLink
- Create an Incoming Call Route for the Inbound SIP calls
- Configure Users’ SIP names

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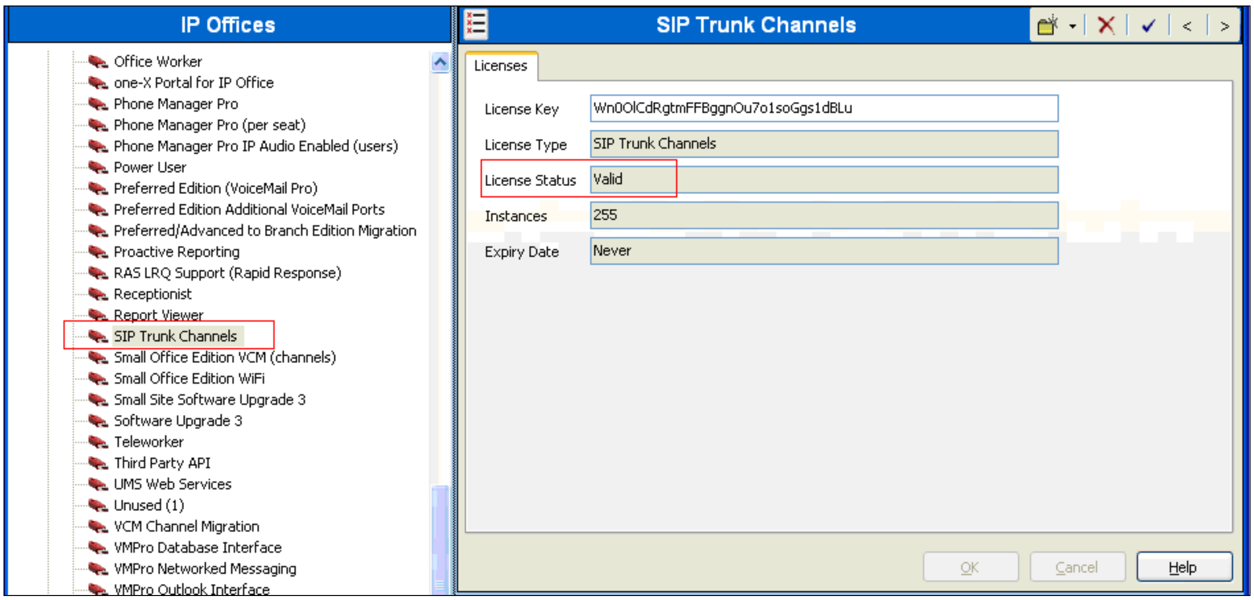
5.1. Verify SIP Trunk Channels License

Avaya IP Office is configured via the IP Office Manager application. Log into the PC running the Avaya IP Office Manager application and select **Start → All Programs → IP Office → Manager** to launch the Manager application. Select the proper Avaya IP Office system if there is more than one Avaya IP Office system listed, and log in with the appropriate credentials. From the configuration tree in the left pane, select **License → SIP Trunk Channels** to display the SIP Trunk Channels screen in the right pane. Verify that the **License Status** field is set to "Valid" and that there are enough **Instances** available.

If a required feature is not enabled or there is insufficient capacity (**Instances)**, contact an authorized Avaya sales representative to make the appropriate changes.



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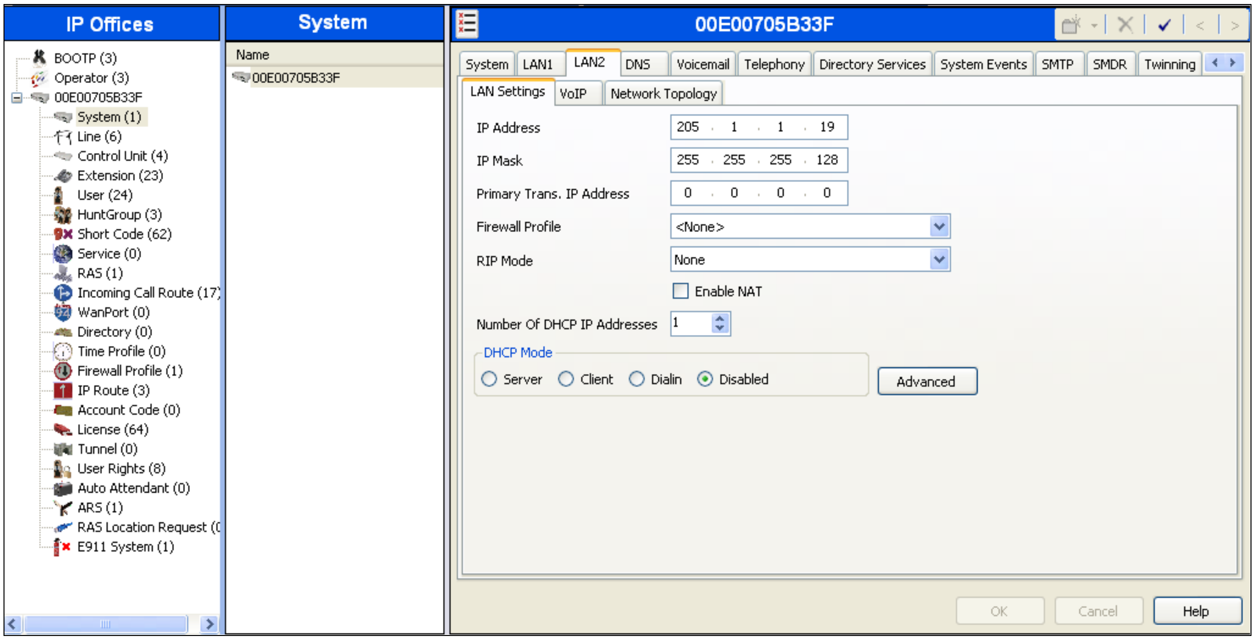
5.2. Configure WAN interface

From the configuration tree in the left pane, select **System** to display the System screen in the right pane. Click the **LAN2** tab. Under the **LAN2** tab, select the **LAN Settings** sub-tab and provide an **IP Address** and **IP Mask**.



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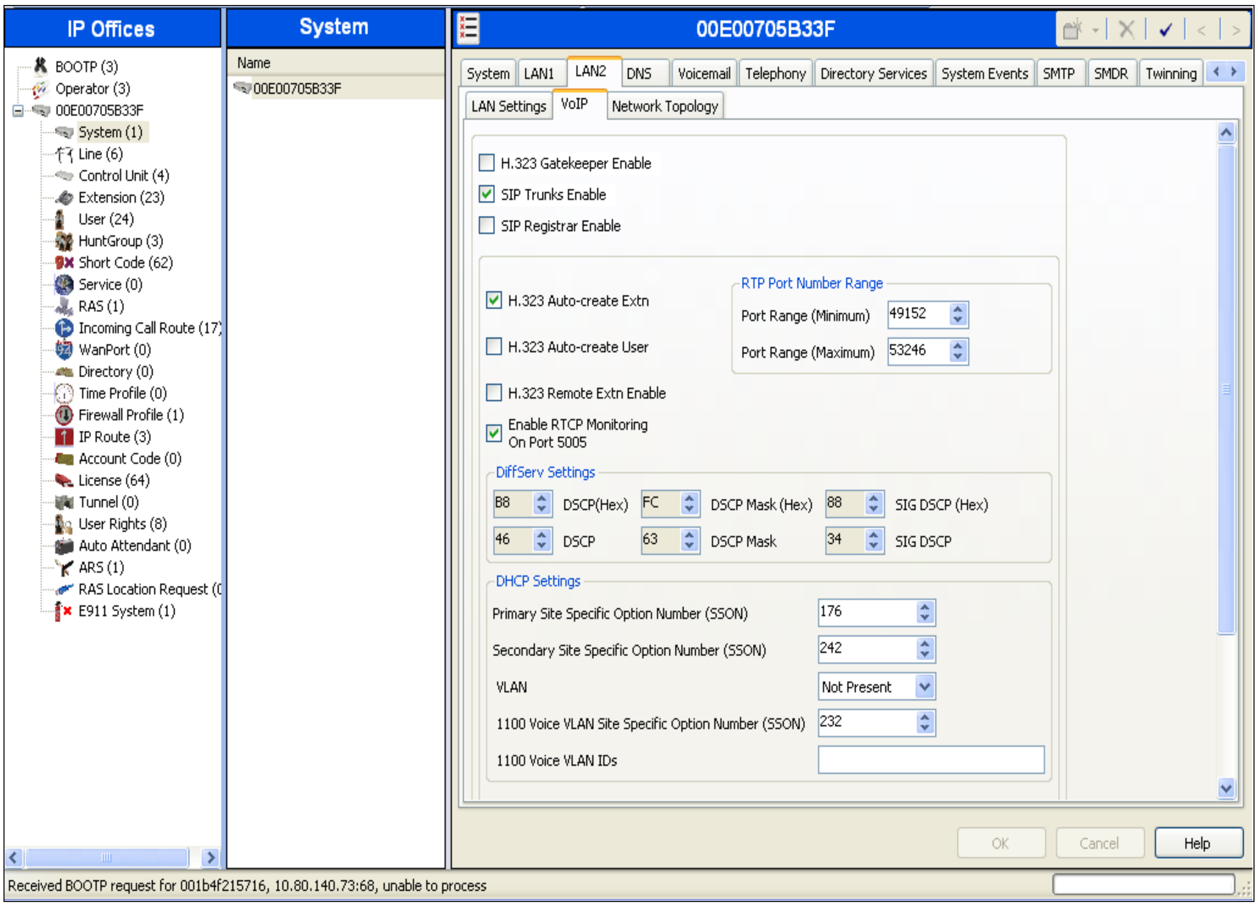
5.3. Enable SIP Trunk

Under the **LAN2** tab, select the **VoIP** sub-tab, and check the **SIP Trunks Enable** box. Click the **OK** button.

Note: During the initial configuration of Avaya IP Office, **LAN1** was configured as a private network (LAN) and **LAN2** was configured as a public network (WAN). Avaya IP Office can support SIP extensions on the **LAN1** and/or **LAN2** interfaces. In the compliance test, the **LAN2** interface was used for the SIP trunk termination but not for remote SIP endpoints. On the public network, the **SIP Registrar Enable** check box was left unchecked for **LAN2** but was enabled for **LAN1** to support SIP phones on the private network.



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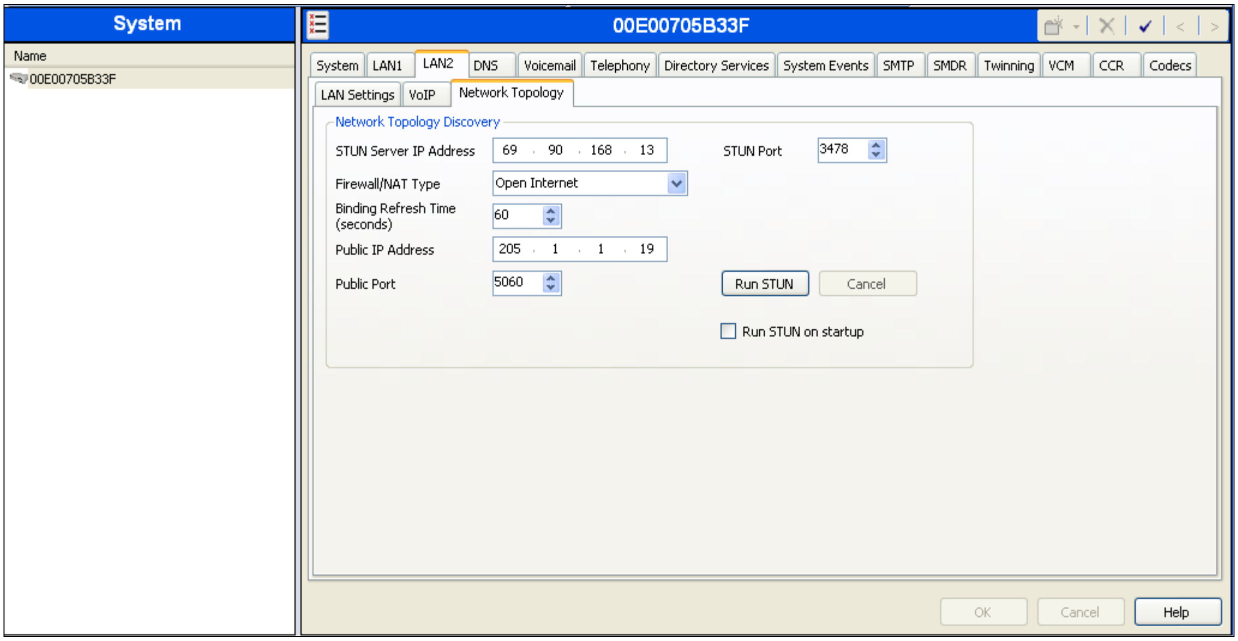
5.4. Configure SIP OPTIONS Timer for "keep alive" Function with CenturyLink

Under the **LAN2** tab, select the **Network Topology** sub-tab. Set the **Binding Refresh Time** to the desired interval that OPTIONS messages will be sent to CenturyLink. For this compliance test, it is set to "60". For **Public IP Address**, enter the Avaya IP Office system public IP address (WAN), confirm that **Public Port** is set to "5060" and accept the default values for all other fields. Click the **OK** button.



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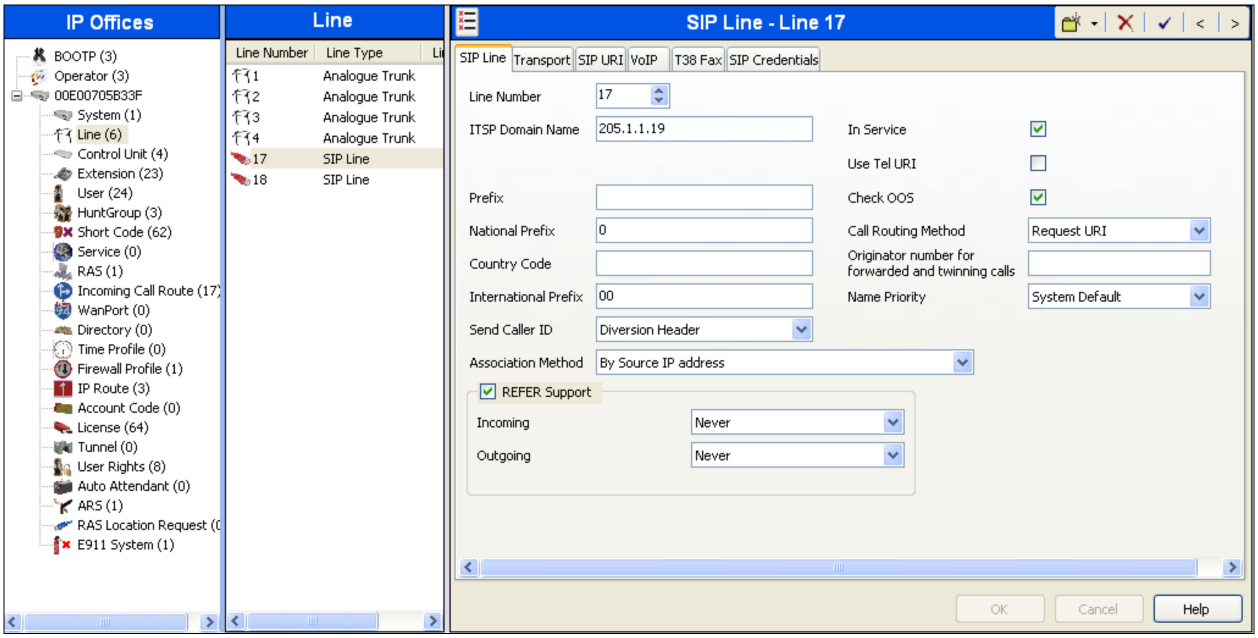
5.5. Create SIP Line to CenturyLink

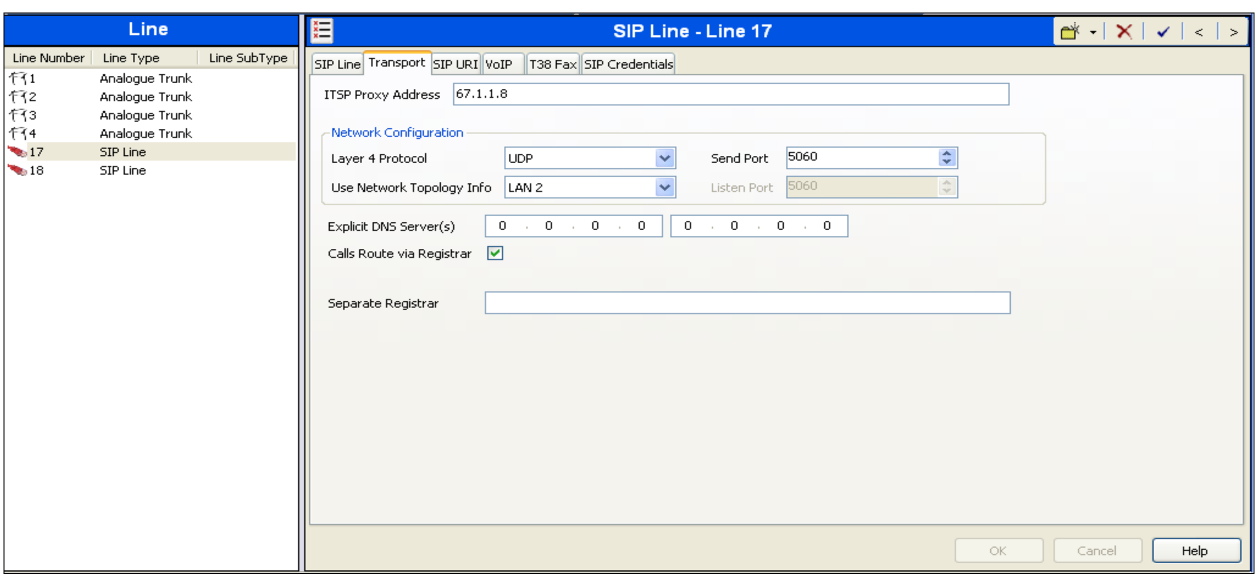
Select **Line** in the left pane. Right-click and select **New → SIP Line** to create a new **SIP Line**. During the compliance test SIP line "17" was configured. Click the **OK** button.

Note: REFER Support was set to "Never" due to issues being worked regarding the use of the SIP REFER method at the time of writing these Application Notes, as described in **Section 2.2**.



Select the **Transport** sub-tab, and provide the following information:

- **ITSP Proxy Address** – Enter the IP address of the CenturyLink SIP termination point.
- **Layer 4 Protocol** – Select **UDP**.
- **Use Network Topology Info** – Select **LAN2** (WAN interface).
- Click the **OK** button.



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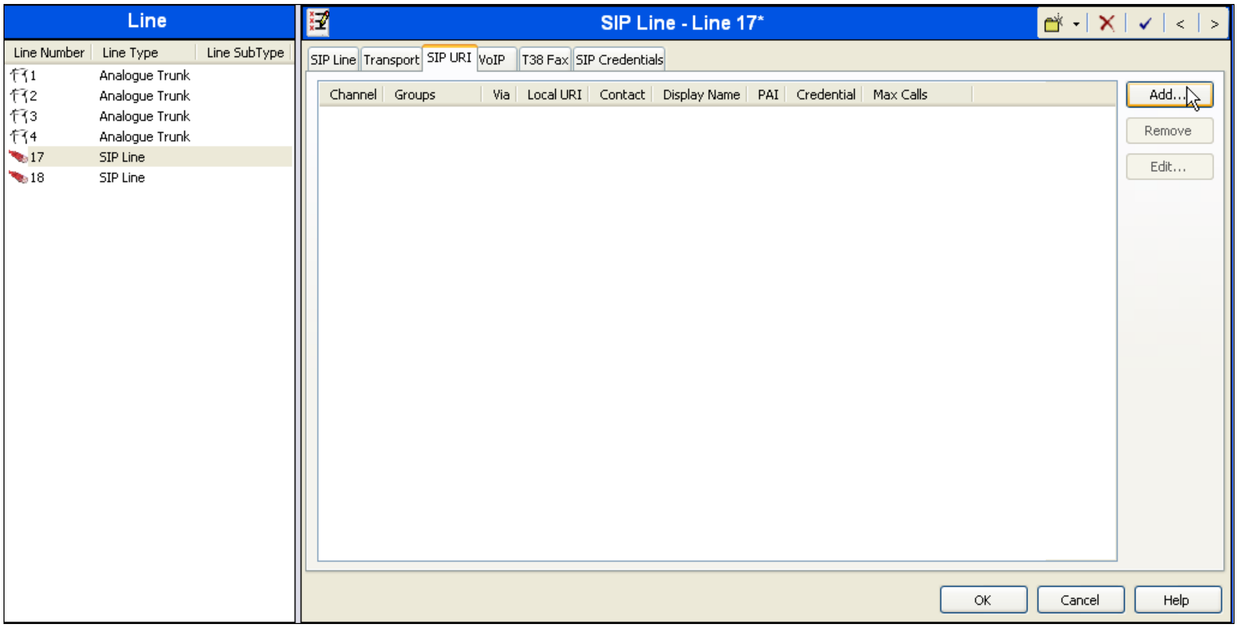
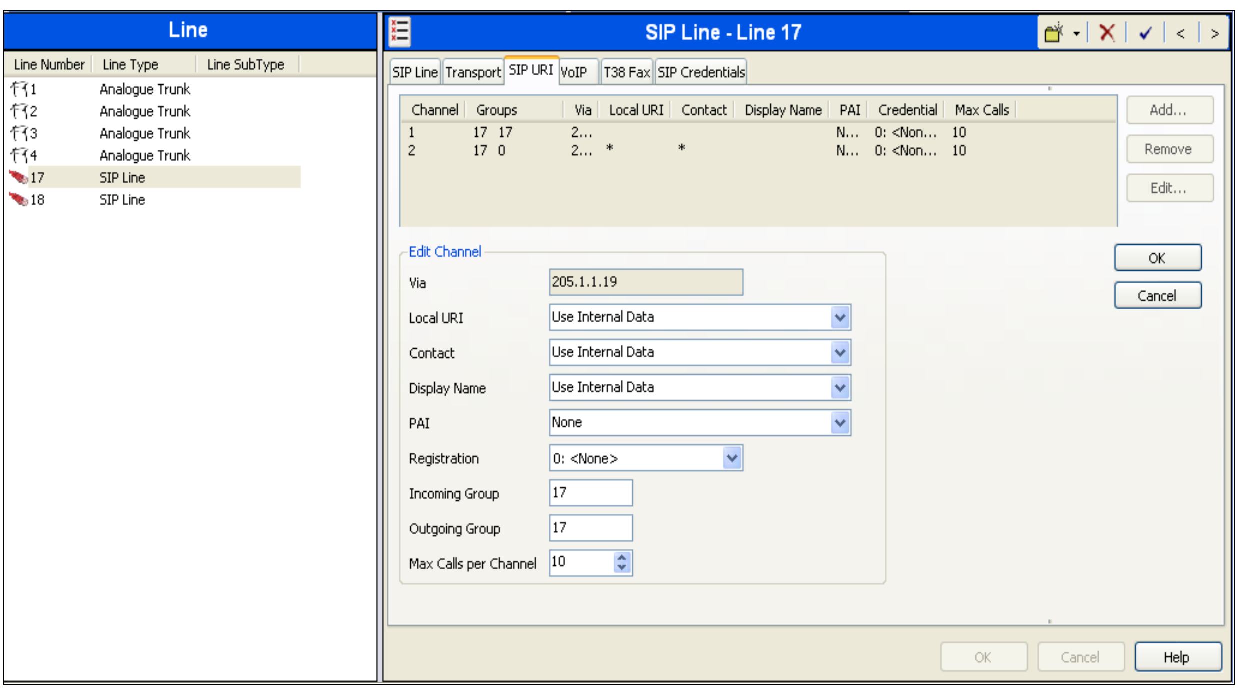
5.6. Configure SIP URI Parameters for the SIP Line

Select the **SIP URI** tab to configure SIP URI parameters for the SIP Line. Click on the **Add** button.



Set the **Local URI**, **Contact** and **Display Name** fields to "Use Internal Data" so that outbound calls will use what is populated in the **SIP** tab for each user, for Caller ID. Be sure to set the **Incoming Group** and **Outgoing Group** fields to the appropriate SIP line number, in this case "17".



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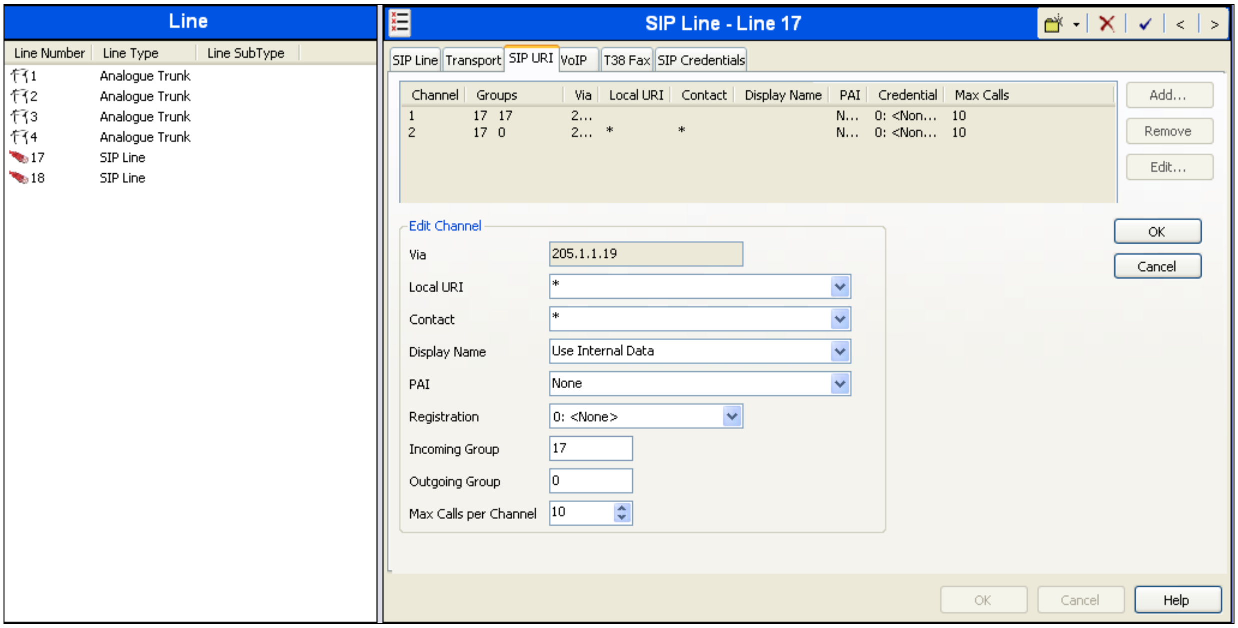
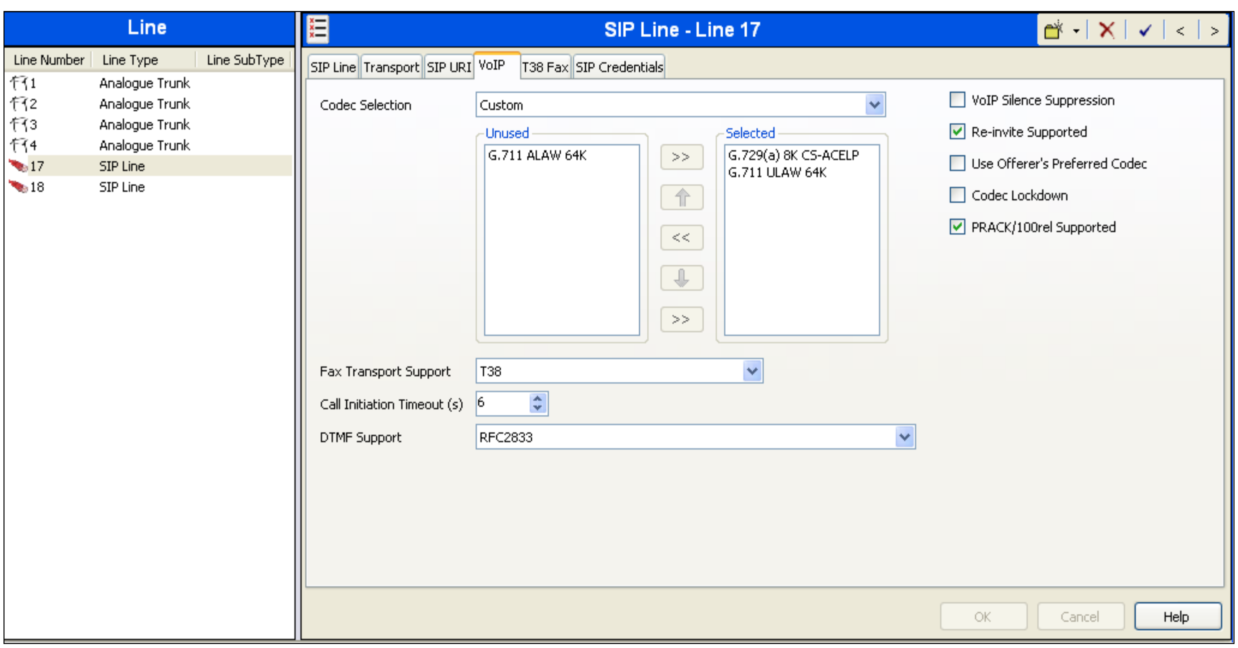
Click **Add** again under the **SIP URI** tab to add another entry for inbound calls. In this case the **Local URI** and **Contact** fields simply contain the wildcard character "*" to accept inbound calls from any number. There is no need to populate the **Outgoing Group** field as this is for inbound calls only. Click **OK** when finished.



Select the **VoIP** tab to configure the appropriate codecs and fax settings. In the **Codec Selection** field, change the dropdown menu from "System Default" to "Custom" (not shown) and add the appropriate codecs to the **Selected** box. Make sure that the **Re-Invite Supported** check box is checked. In the **Fax Transport Support** field, select "T38" using the drop down list. Default values may be used for all other fields.



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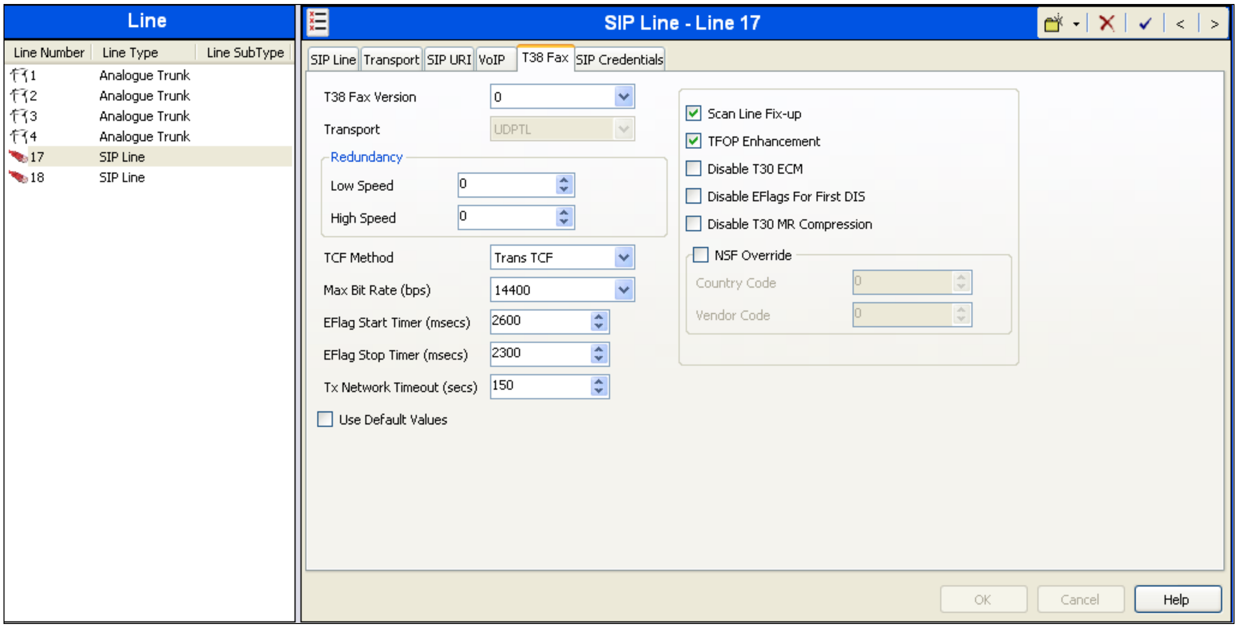
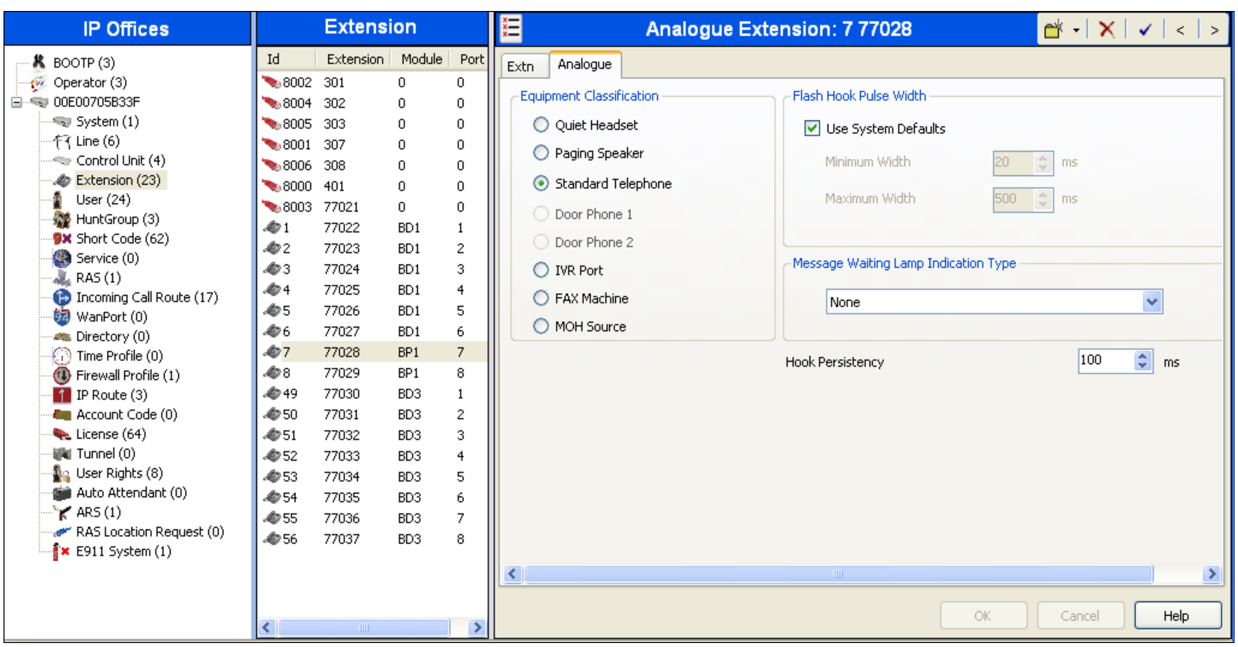
5.7. Configure T38 Fax Parameters for the SIP Line

Select the **T38 Fax** tab to configure T38 Fax parameters for the SIP Line. Uncheck the **Use Default Values** check box and change the **T38 Fax Version** field from "3" to "0" (At the time of writing these Application Notes CenturyLink only supported Fax Version 0). Default Values may be used for all other fields. Click the **OK** button.



A fax machine was connected to an analog extension. The following screen shows the analog extension that was configured as a fax line. Set the Equipment Classification field as **Standard Telephone** not **Fax Machine**. Click the **OK** button.



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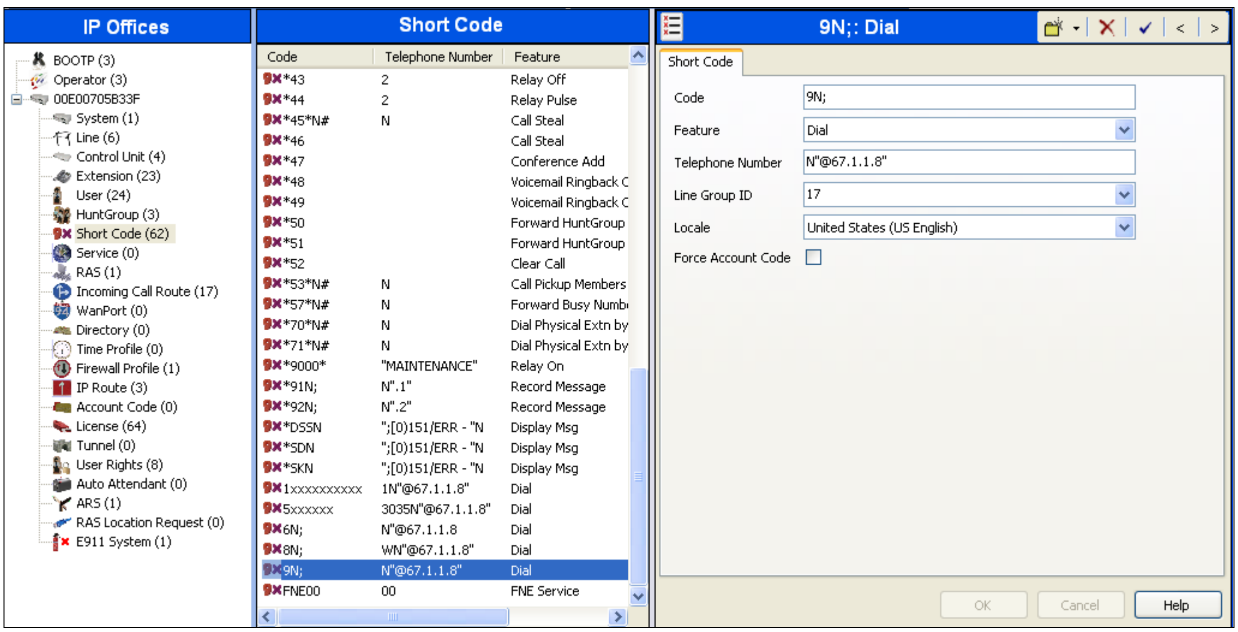
5.8. Configure a Short Code to Route Calls to CenturyLink

Select **Short Code** in the left panel. Right click and select **Add**. Enter [**x]N;**, where **[x]** is a valid number in the **Code** text box. The number "9**"** is used for **[x]** in the example below. This code requires the user to dial the digit "9" followed by the destination's telephone number, symbolized by **N,** in order to route the call out the SIP Trunk.

Note: N can be any number other than a local Avaya IP Office extension. For example, a 10-digit number, a toll-free number, directory assistance, information service, etc.

Select "Dial**"** for the **Feature**. Enter the **Outgoing Group** number created in **Section 5.5** for the **Line Group ID** field. Enter "N" representing the dialed number followed by **"**@<IP address of CenturyLink NBS>**"** for the **Telephone Number** field. The **Telephone Number** field is used to construct the SIP URI in the **To** header of the outgoing SIP INVITE message. Use default values for all other fields. Click the **OK** button.



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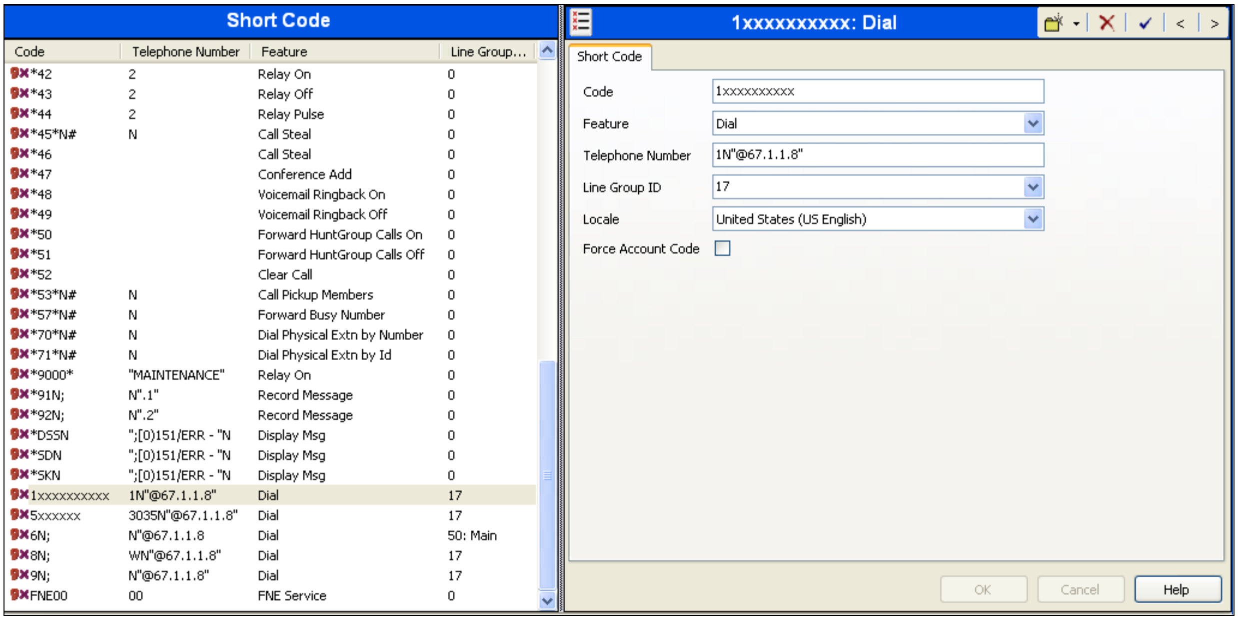
Right click and select **Add** again to add another short code. This time enter “1xxxxxxxxxx” in the **Code** field, select “Dial**”** for the **Feature**, and enter the **Outgoing Group** number created in **Section 5.5** for the **Line Group ID** field. Enter “1N” representing the dialed number followed by **“**@<IP address of CenturyLink NBS>**”** for the **Telephone Number** field. This **Short Code** allows users to dial 11 digits without the preceding 9. 10 and 7 digit dialing can be setup this way as well, if necessary. Use default values for all other fields. Click the **OK** button.



(**Note**: *The configuration of the Short Codes above is shown only as examples and is not meant to be prescriptive*)

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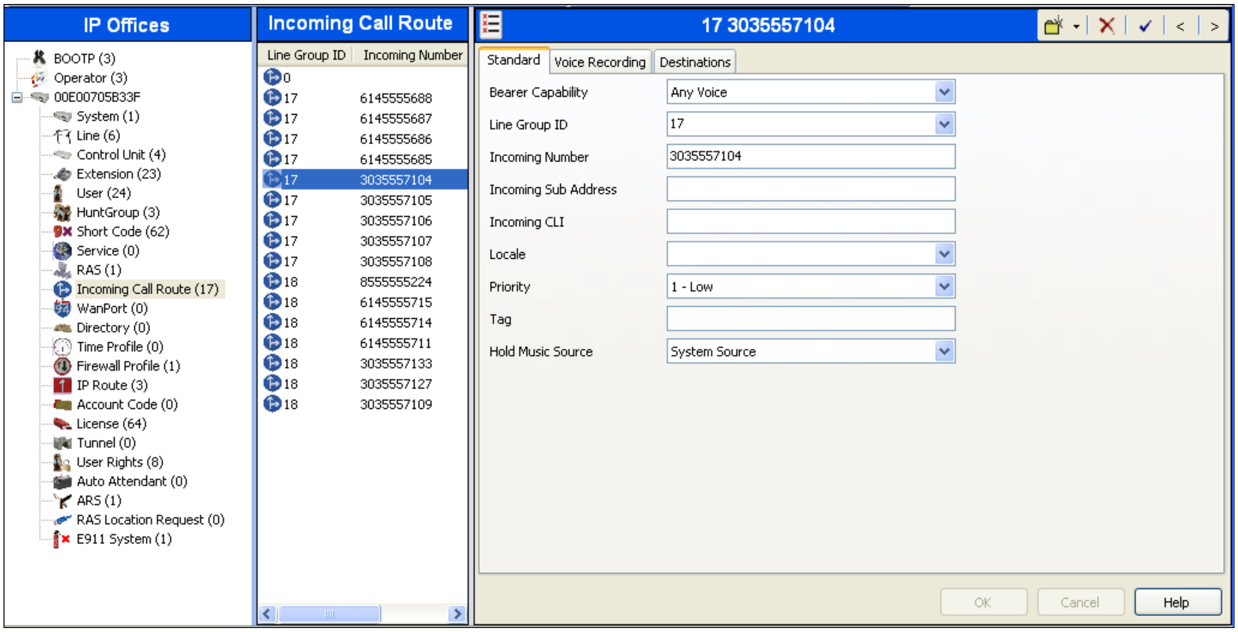
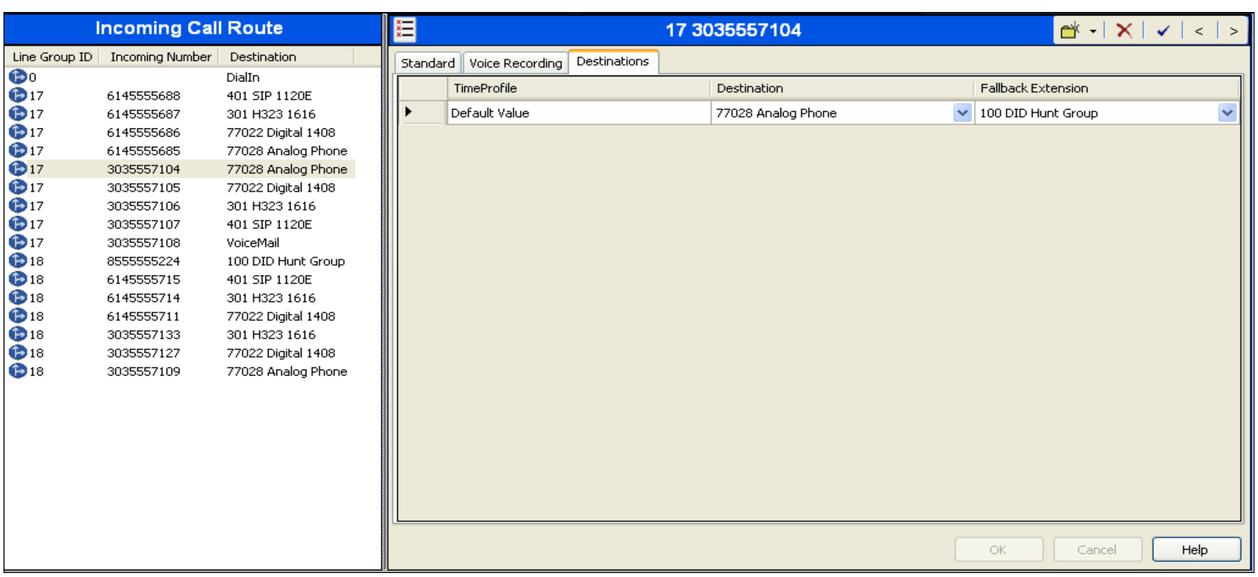
5.9. Create an Incoming Call Route for the Inbound SIP Calls

Select **Incoming Call Route** in the left pane. Right-click and select **New**. Enter the following:

- "Any Voice" for the **Bearer Capability** field.
- The **Incoming Group** number created for the URI in **Section 5.5** in the **Line Group ID** field.
- The 10-digit DID number provided by CenturyLink that is mapped back to a local Avaya IP Office extension, in the **Incoming Number** field.
- Use default values for all other fields.



Next, navigate to the **Destinations** tab and select the desired local extension number from the drop down list. This is the extension that the DID will map to when receiving an inbound call for that DID. Click the **OK** button.



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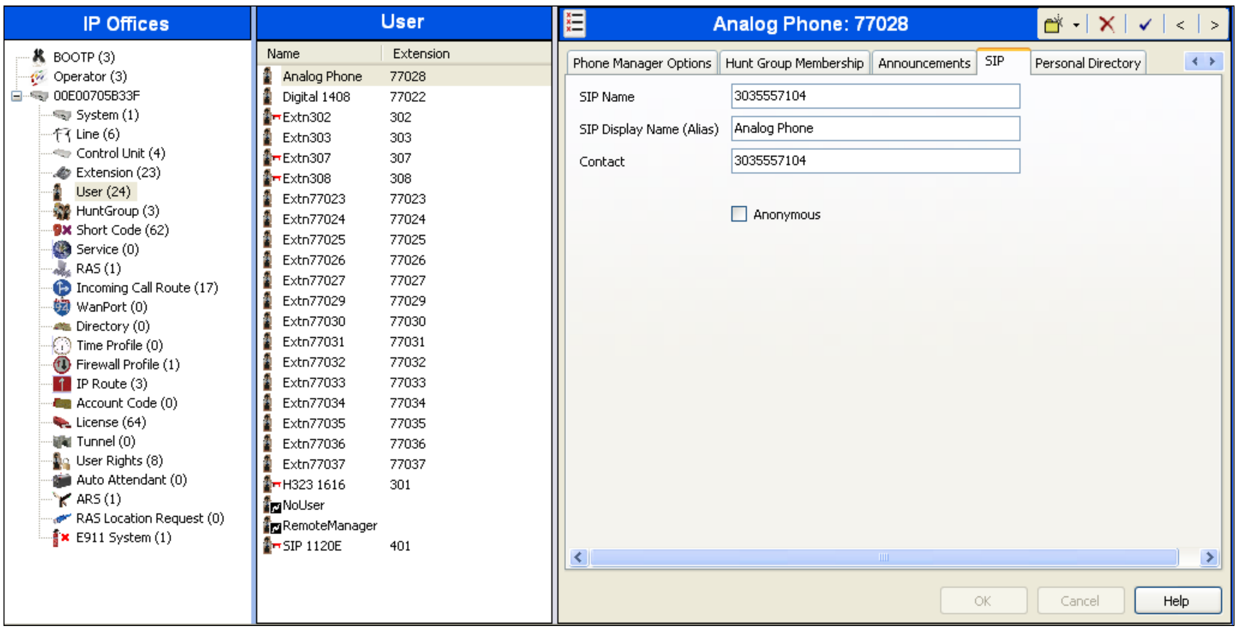
5.10. Configure SIP Parameters for Users

Select **User** in the left panel. Select the desired user by clicking on an entry in the middle panel. Select the **SIP** tab. Modify the **SIP Name** and **Contact** fields to the DID number provided by CenturyLink that is used for this particular extension. These settings instruct the system to use this DID number to construct the:

- user part of the SIP URI in the From header of an outgoing SIP INVITE message
- user part of the SIP URI in the Contact header of an outgoing SIP INVITE message

Modify the **SIP Display Name (Alias)** that will be used for the SIP Display info, to a descriptive name. Click the **OK** button.



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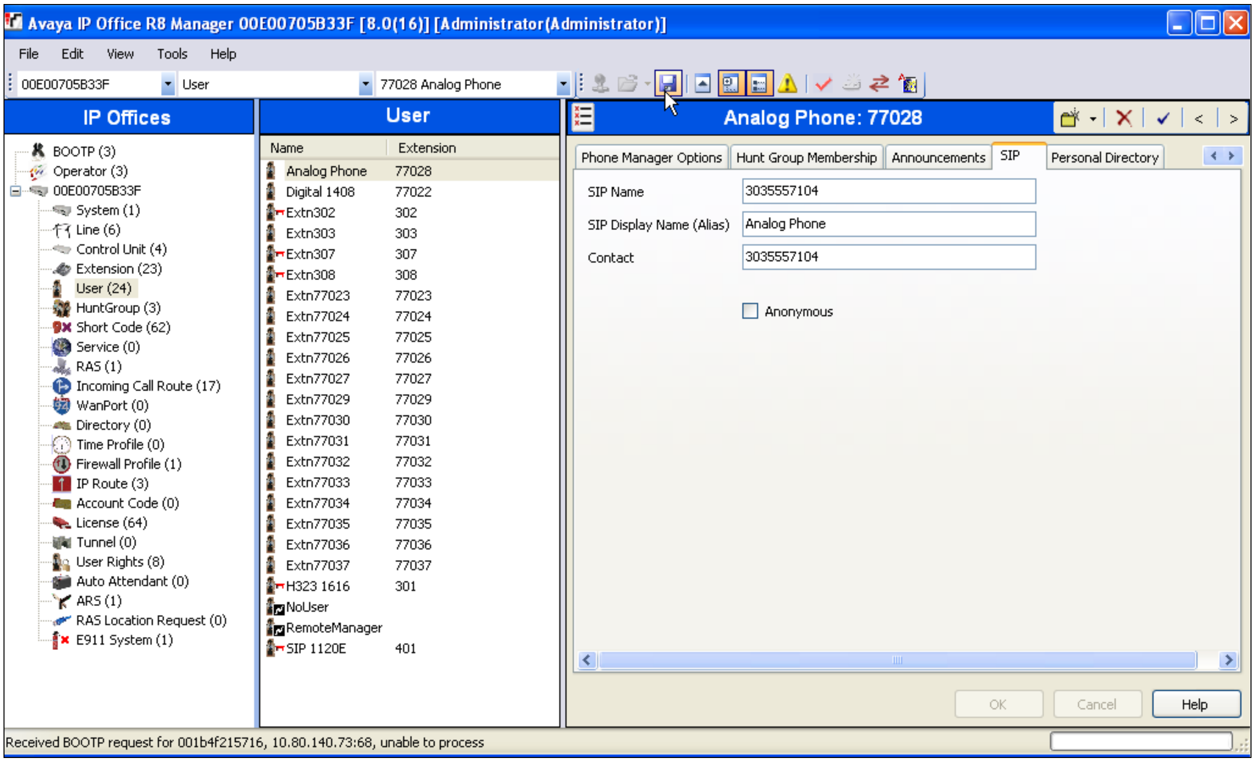
After making changes or completing the configuration, click on the floppy disk icon to push the changes to the Avaya IP Office system and have them take effect.



(**Note:** *Changes will not take effect until this step is completed. This may cause a reboot of Avaya IP Office causing service disruption.*)

6. Configure CenturyLink

To use the CenturyLink SIP Trunk Service, a customer must request service from CenturyLink using their sales processes. The process can be started by contacting CenturyLink via the corporate web site at http://www.CenturyLink.com and requesting information via the online sales links or telephone numbers.

7. Verification Steps

This section provides verification steps that may be performed to verify that the Avaya H.323, SIP, digital and analog endpoints can place outbound calls and receive inbound calls through CenturyLink's SIP Trunk Service solution.

- Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
- Verify that the user on the PSTN can terminate an active call by hanging up.
- Verify that an endpoint at the enterprise site can terminate an active call by hanging up.

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8. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya IP Office telephony system to a CenturyLink SIP Trunk Service solution. The CenturyLink SIP Trunk Service is a SIP-based Voice over IP solution. CenturyLink SIP Trunk Service passed compliance testing with the exception of items noted in **Section 2.2**.

9. Additional References

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

[1] Avaya *IP Office 8.0 Installation, Issue 24i,* December 2011, Document Number 15-601042

[2] *Avaya IP Office Release 8 Manager 8.0, Issue 04i,* January 2012, Document Number 15-601011

[3] RFC 3261 *SIP: Session Initiation Protocol* http://www.ietf.org/

[4] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals* http://www.ietf.org/

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