Avaya Solution & Interoperability Test Lab

Application Notes for Configuring EarthLink SIP Trunk Service with Avaya IP Office using UDP/RTP - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider EarthLink and Avaya IP Office Release 10.0.

EarthLink SIP Trunk Service (EarthLink) provides PSTN access via a SIP trunk between the enterprise and the EarthLink network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

EarthLink 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 EarthLink and Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists Avaya IP Office 500 V2 Release 10.0, Avaya embedded Voicemail, Avaya IP Office Application Server (with WebRTC and one-X Portal services enabled), Avaya Communicator for Windows (SIP mode), Avaya Communicator for Web, Avaya H.323, Avaya SIP, digital and analog endpoints. The enterprise solution connects to the EarthLink network.

The EarthLink 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 EarthLink.

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 EarthLink 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 EarthLink. To verify SIP trunking interoperability, following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types. Phone types included Avaya H.323, Avaya 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 Avaya H.323, Avaya 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 Communicator for Windows (SIP mode)
- Inbound and outbound PSTN calls from/to the Avaya Communicator for Web with basic telephony transfer feature

- Inbound and outbound long hold time call stability
- Various call types including: local; long distance; international call; inbound toll-free; outbound to assisted operator, 411 and 911 services during the compliance testing
- SIP transport using UDP as supported
- Codec G.711MU, G.729A
- 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 G.711 pass-through mode
- Use Diversion header in off-net call forwarding
- Use SIP re-Invite or SIP Refer in off-net call transfer
- Twinning to mobile phones on inbound calls

Item not supported by EarthLink Lab Environment include the following:

- Outbound toll free call
- Fax T.38

2.2. Test Results

Interoperability testing of EarthLink was completed with successful results for all test cases with the exception of the limitation described below:

- **Blind Call Transfer using Avaya 1140E SIP phone did not complete until transferee picked up the call** - The expected behavior of the SIP phone was that after transferring, the phone should display “Transfer successful”. In this case, the user pressed “Trnsfr” button, answered “No” to the question of “Consult with party?” which implied the blind transfer, the transferee phone was ringing and the SIP phone should be released and displaying “Transfer successful”. Instead, the SIP phone was still displaying “Transferring” and did not released until the transferee phone answered the call. This is very minor known limitation on Avaya 1140E SIP phone. There was no user impact. Transfer was still completed with 2-way 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 EarthLink SIP Trunking, contact EarthLink at https://www.earthlink.com

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to EarthLink 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.

The Avaya components used to create the simulated customer site included:

- Avaya IP Office 500 V2
- Avaya embedded Voicemail for IP Office
- Avaya Application Server (Enabled WebRTC and one-X Portal services)
- Avaya 9600 Series IP Deskphones (H.323)
- Avaya 11x0 Series IP Deskphones (SIP)
- Avaya 1408 Digital phone
- Avaya Analog phone
- Avaya Communicator for Windows (SIP)
- Avaya Communicator for Web

Located at the enterprise site is an Avaya IP Office 500 V2 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 Telephone, Avaya Analog Telephone, and Avaya Communicator for Windows.

A separate Windows 10 Enterprise PC runs Avaya IP Office Manager to configure and administer Avaya IP Office system.

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 configured mobile phones.

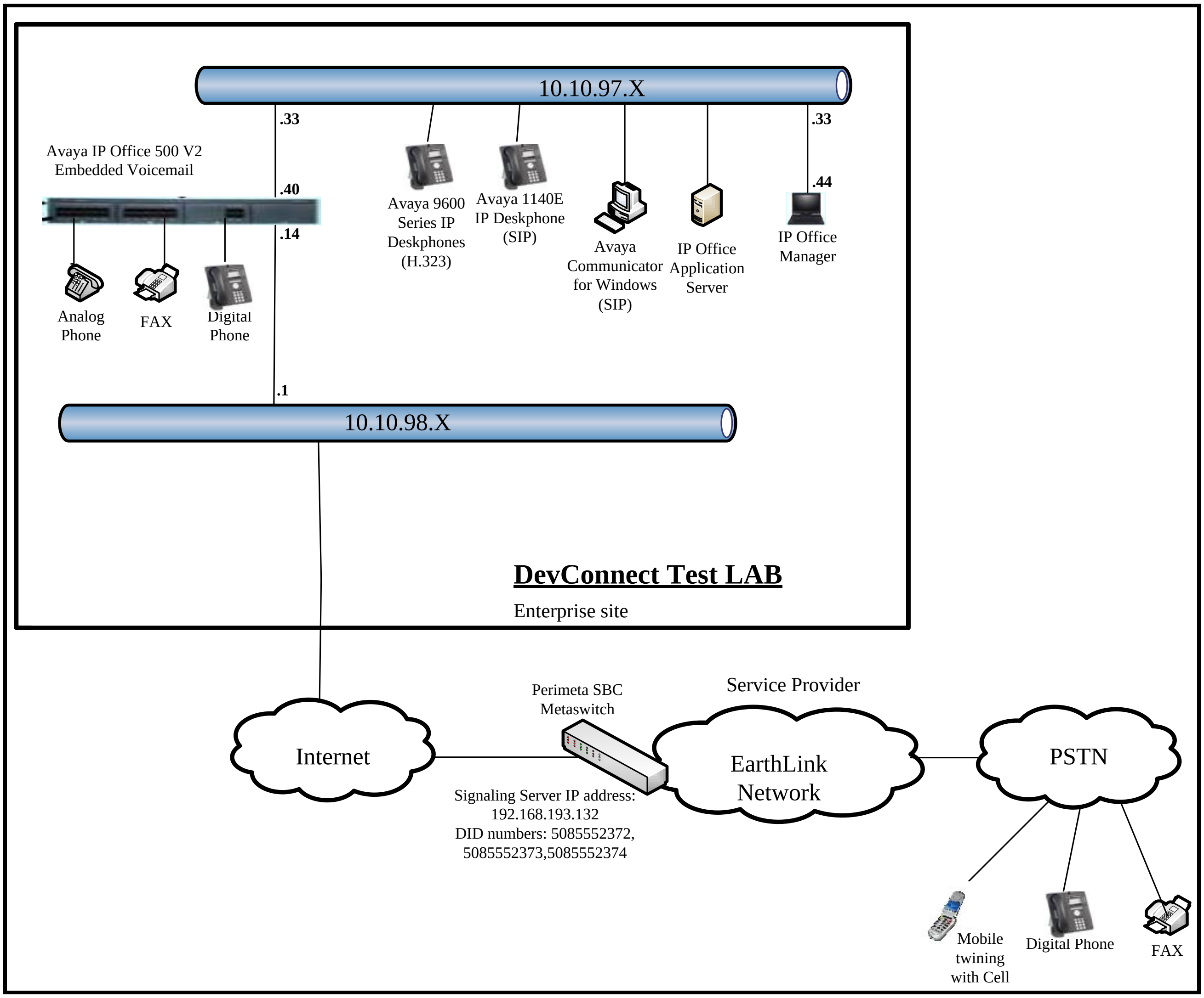

Figure 1 - Test Configuration for Avaya IP Office with EarthLink SIP Trunk Service

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 EarthLink. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to EarthLink. 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, EarthLink 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 solution ▪ Avaya IP Office 500 V2 ▪ Embedded Voicemail ▪ Avaya Web RTC Gateway ▪ Avaya one-X Portal ▪ Avaya IP Office Manager ▪ Avaya IP Office Analogue PHONE 8 ▪ Avaya IP Office VCM64/PRID U ▪ Avaya IP Office DIG DCPx16 V2	 10.0.0.2.0 build 10 10.0.0.2.0 build 10 10.0.0.3.0 build 7 10.0.0.2.0 build 13 10.0.0.2.0 build 10 10.0.0.2.0 build 10 10.0.0.2.0 build 10 10.0.0.2.0 build 10
Avaya 1140E IP Deskphone (SIP)	04.04.23
Avaya 9641G IP Deskphone	6.6302
Avaya 9621G IP Deskphone	6.6302
Avaya Communicator for Windows (SIP)	2.1.3.237
Avaya Communicator for Web	1.0.16.1718
Avaya 1408D Digital Deskphone	R46
Avaya Analog Deskphone	N/A
HP Officejet 4500 (fax)	N/A
EarthLink Components	
Equipment	**Release**
Perimeta SBC - Metaswitch	V4.0.20

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office Solution

This section describes the Avaya IP Office solution configuration necessary to support connectivity to the EarthLink. It is assumed that the initial installation and provisioning of the Avaya IP Office 500 V2 has been previously completed and therefore is not covered in these Application Notes. For information on these installation tasks refer to Additional References **Section 9**.

This section describes the Avaya IP Office configuration to support connectivity to EarthLink system. 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 click **OK** button. Log in using appropriate credentials.

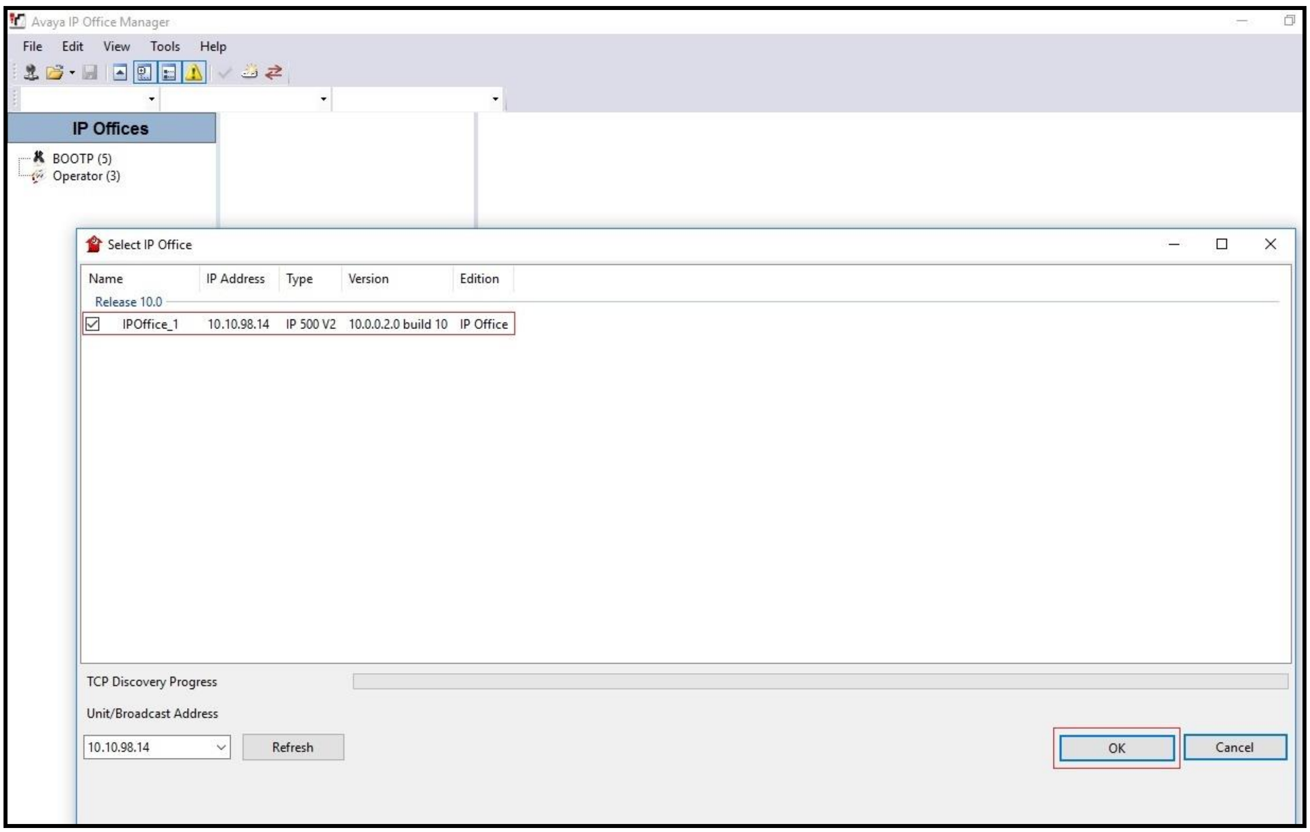

Figure 2 – Avaya IP Office Selection

5.1. Licensing

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

To verify that there is a SIP Trunk Channels license with sufficient capacity, select **IPOffice_1 → License** on the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm that there is a valid license with sufficient "Instances" (trunk channels) in the Details pane.

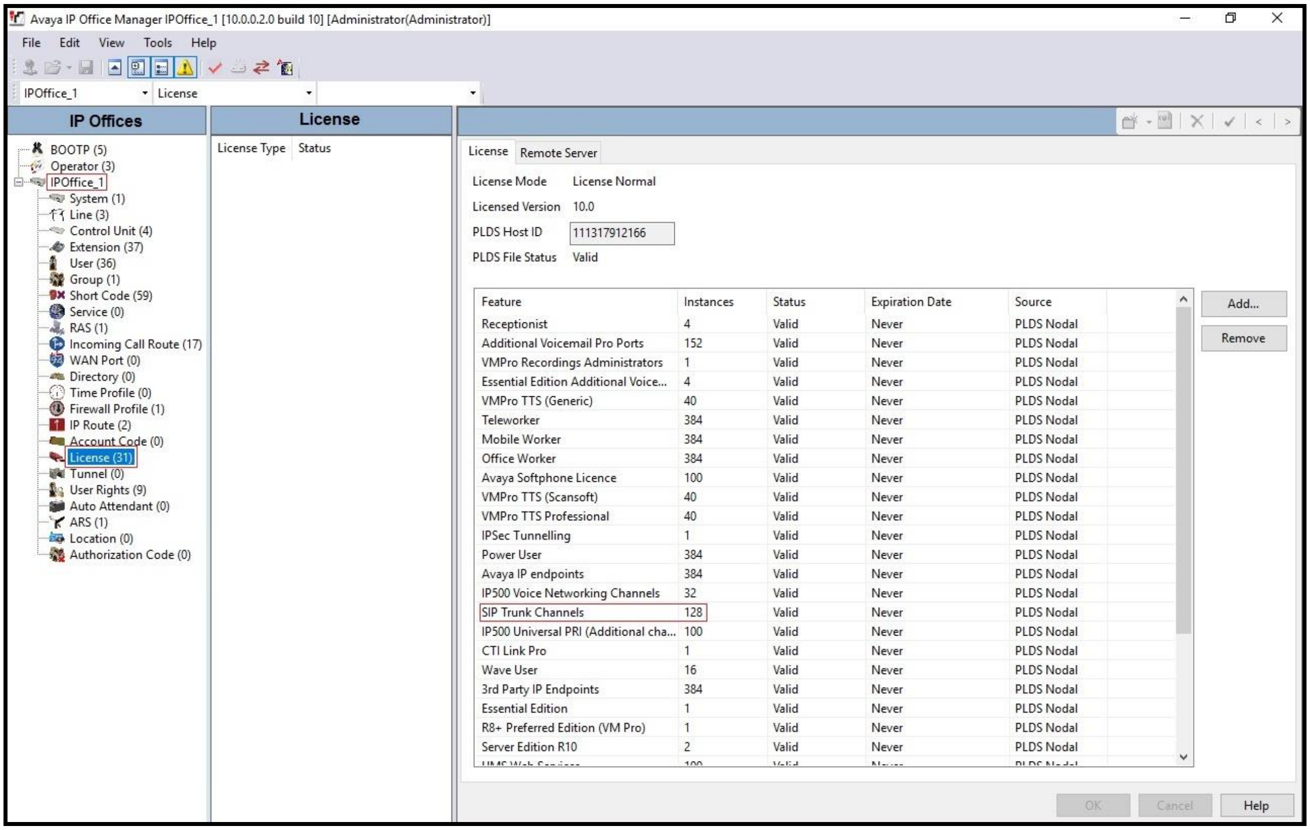

Figure 3 – Avaya IP Office License

5.2. System Tab

Navigate to **System (1)** under the **IPOffice_1** on the left pane and select the **System** tab in the Details pane. The Name field can be used to enter a descriptive name for the system. In the reference configuration, **IPOffice_1** was used as the name in IP Office.

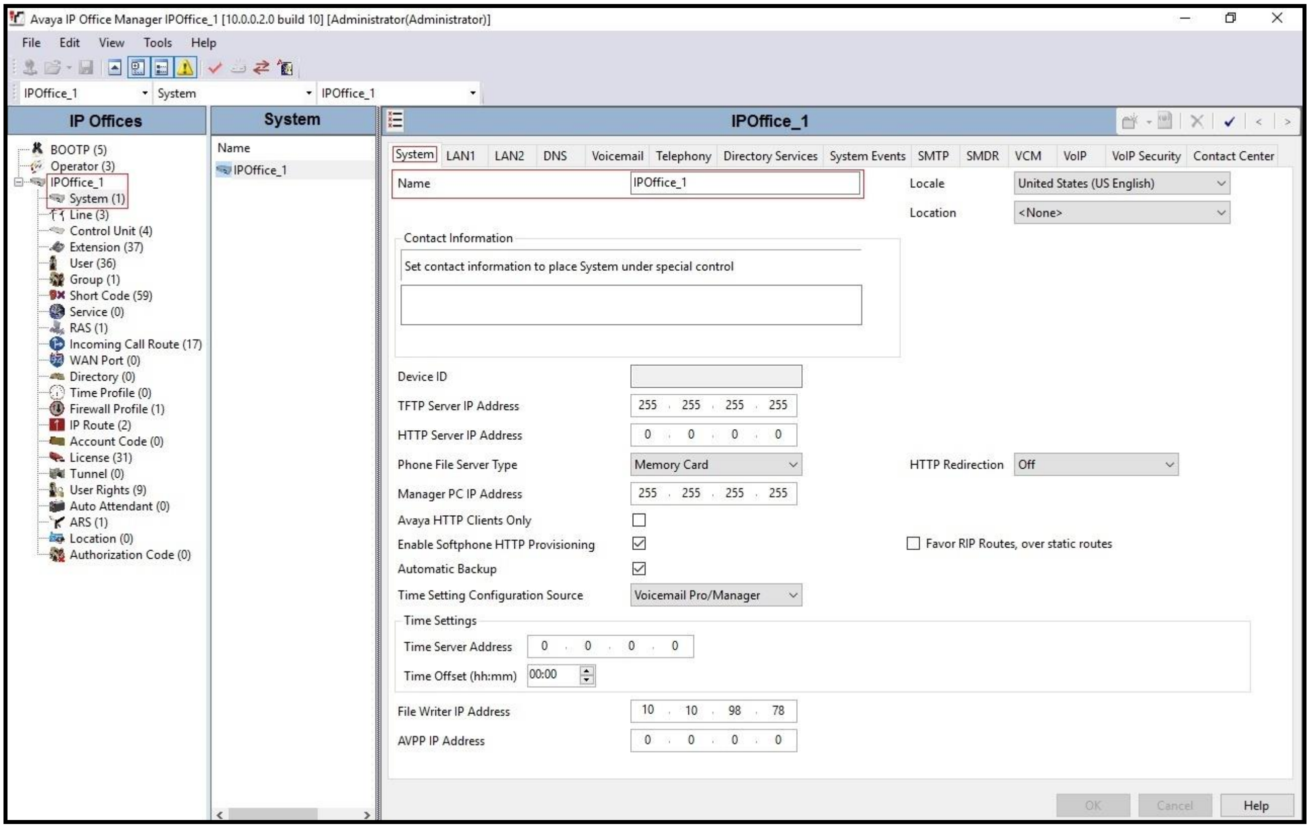

Figure 4 - Avaya IP Office System Configuration

5.3. LAN2 Settings

In the sample configuration, LAN2 is used to connect the enterprise network to EarthLink network.

To configure the LAN2 settings on the IP Office, complete the following steps. Navigate to **IPOffice_1 → System (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 LAN2 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.

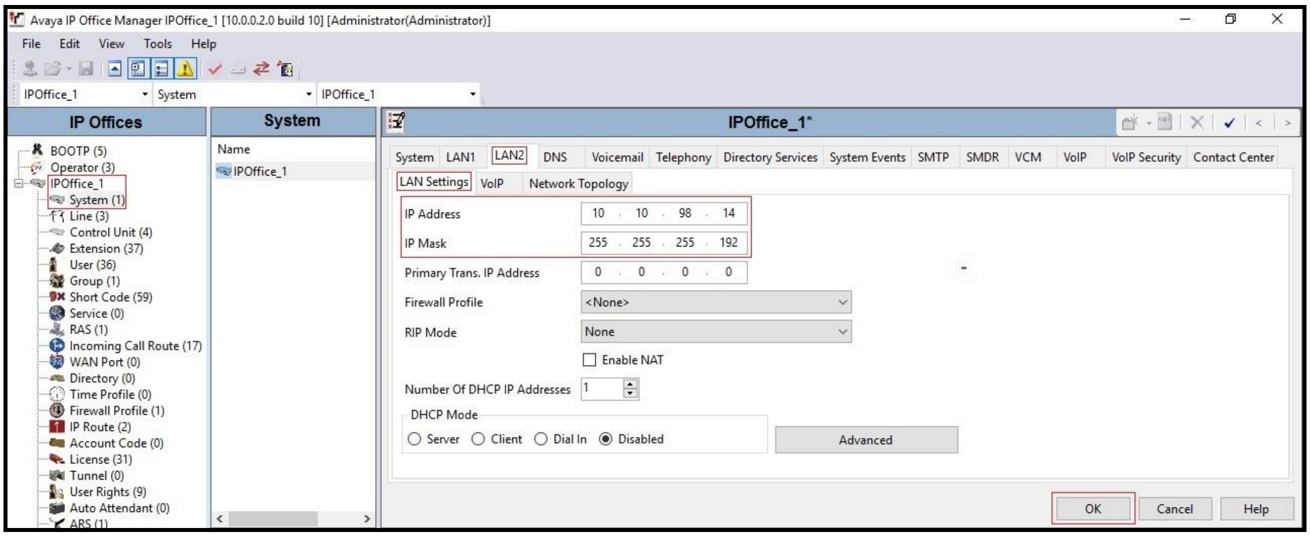

Figure 5 - Avaya IP Office LAN2 Settings

The **VoIP** tab as shown in the screenshot on the next page was configured with following settings:

- Check the **H323 Gatekeeper Enable** to allow Avaya IP Deskphones/Softphones using the H.323 protocol to register
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to EarthLink system
- Check the **SIP Registrar Enable** to allow Avaya IP Deskphones/Softphones to register using the SIP protocol.
- Input **SIP Domain Name** as **10.10.98.14**
- The **Layer 4 Protocol** uses **UDP** with **UDP Port** as **5060**
- Verify **Keepalives** to select **Scope** as **RTP-RTCP** with **Periodic timeout 60** and select **Initial keepalives** as **Enabled**
- All other parameters should be set according to customer requirements
- Click **OK** to submit the changes

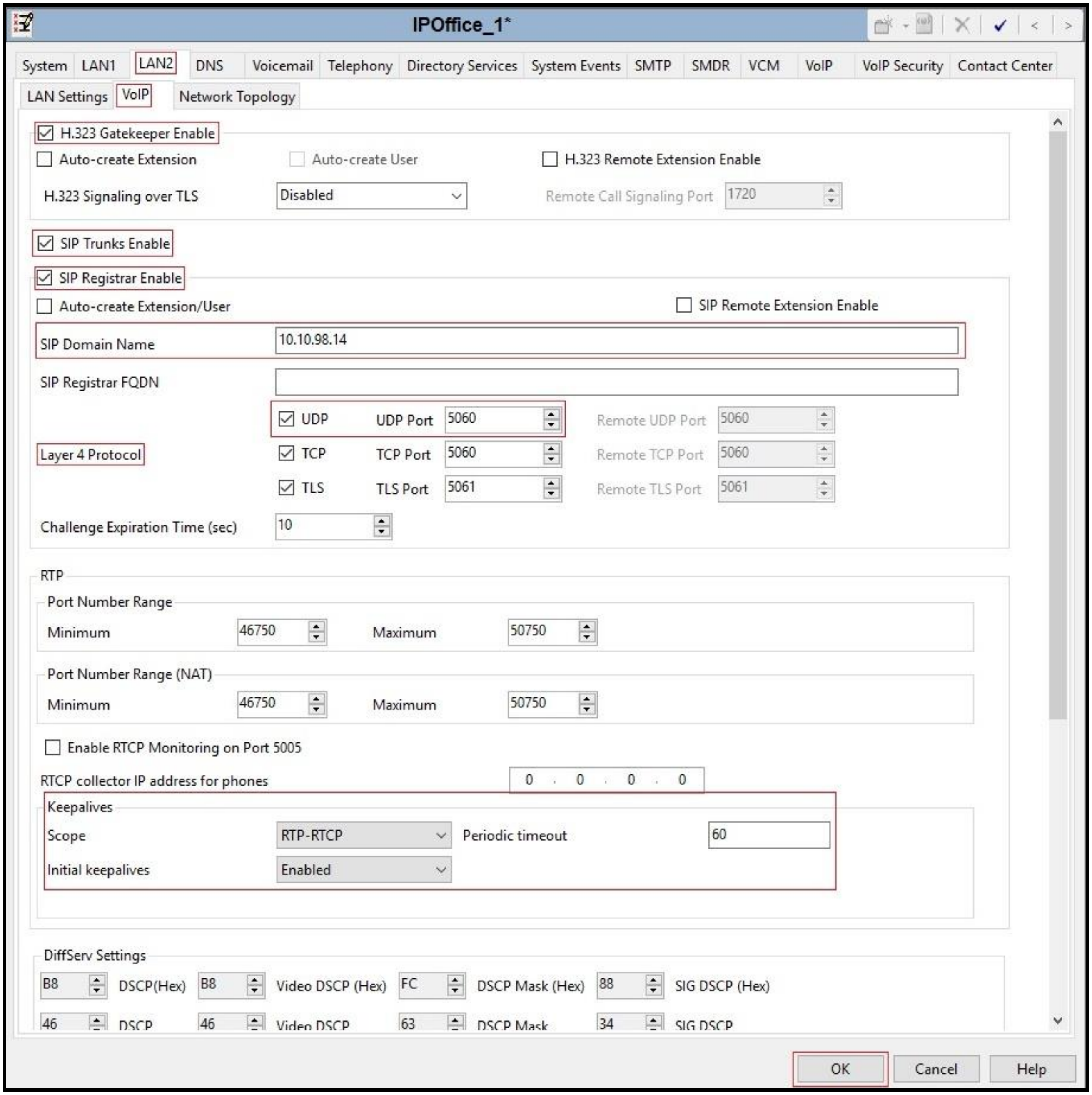

Figure 6 - Avaya IP Office LAN2 VoIP

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 Server will not be used
- Set the **Binding Refresh Time (sec)** to **60**. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTION messages to service provider to keep the connection alive
- Set **Public IP Address** to the IP address of the Avaya IP Office LAN2 port
- Set **Public Port** for **UDP** as **5060** (For SIP Trunk connecting to EarthLink)
- All other parameters should be set according to customer requirements
- Click **OK** to submit the changes

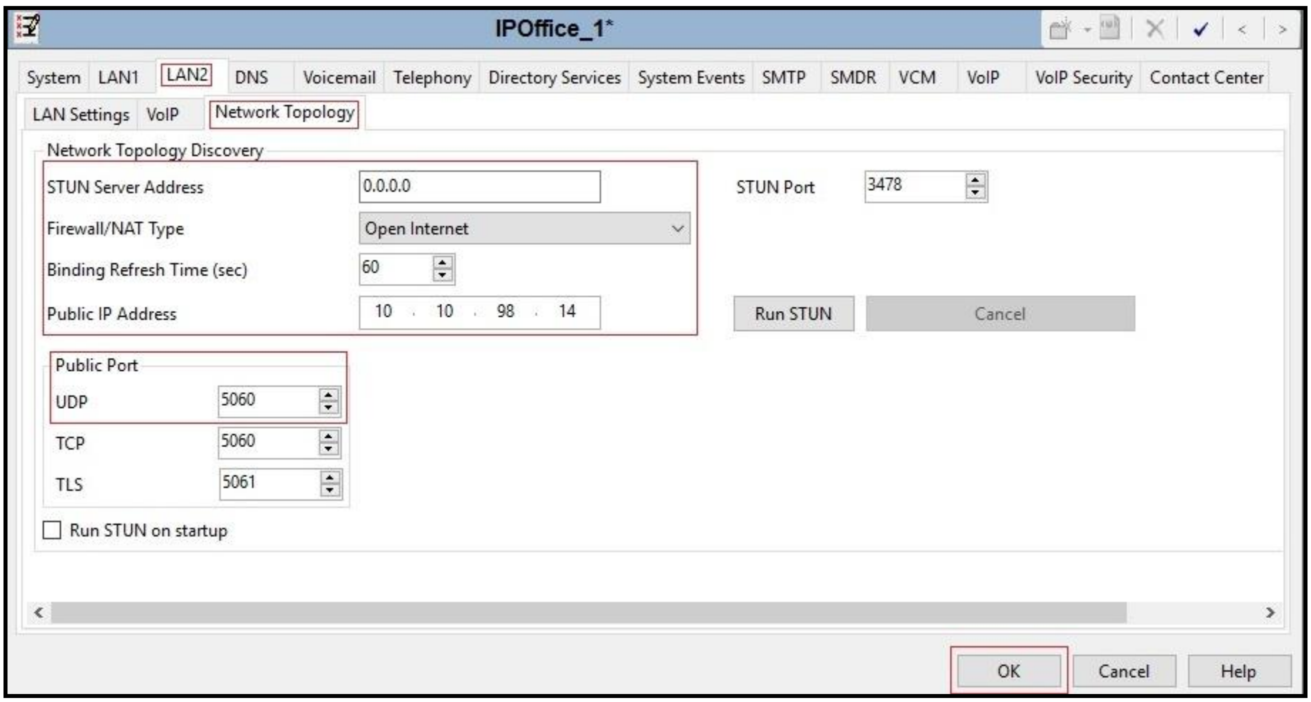

Figure 7 - Avaya IP Office LAN2 Network Topology

5.4. System Telephony Settings

Navigate to **IPOffice_1 → System (1)** in the Navigation and Group Panes (not shown) 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 (sec)** to a valid number. Set **Default Name Priority** to **Favor Trunk**. Defaults were used for all other settings. Click **OK** to submit the changes.

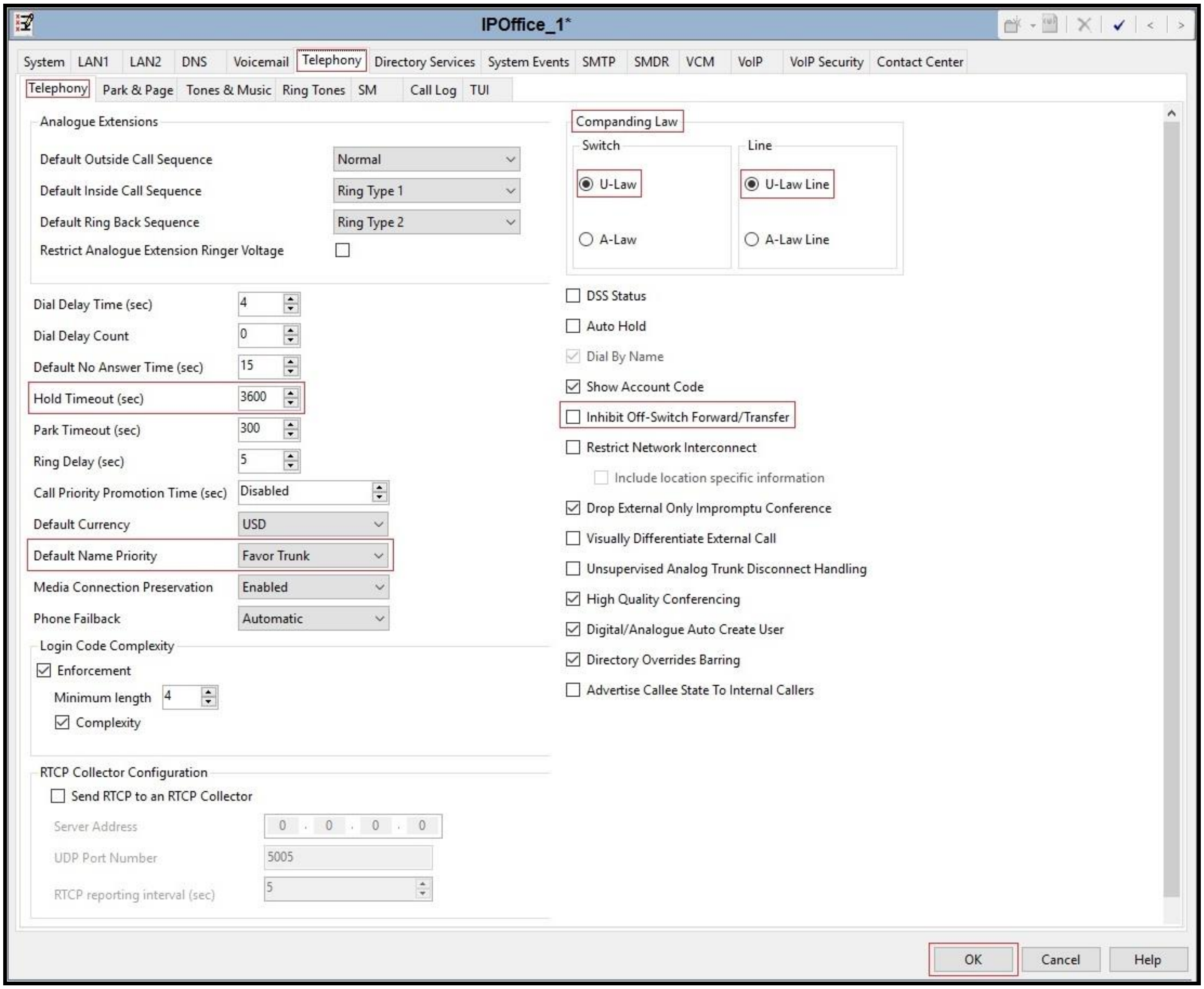

Figure 8 - Avaya IP Office Telephony

5.5. System VoIP Settings

Navigate to **IPOffice_1 → System (1)** in the Navigation and Group Panes and then navigate to the **VoIP** tab in the Details Pane. Leave the **RFC2833 Default Payload** as default of **101**. Select codec **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** which EarthLink supports. Click **OK** to submit the changes.

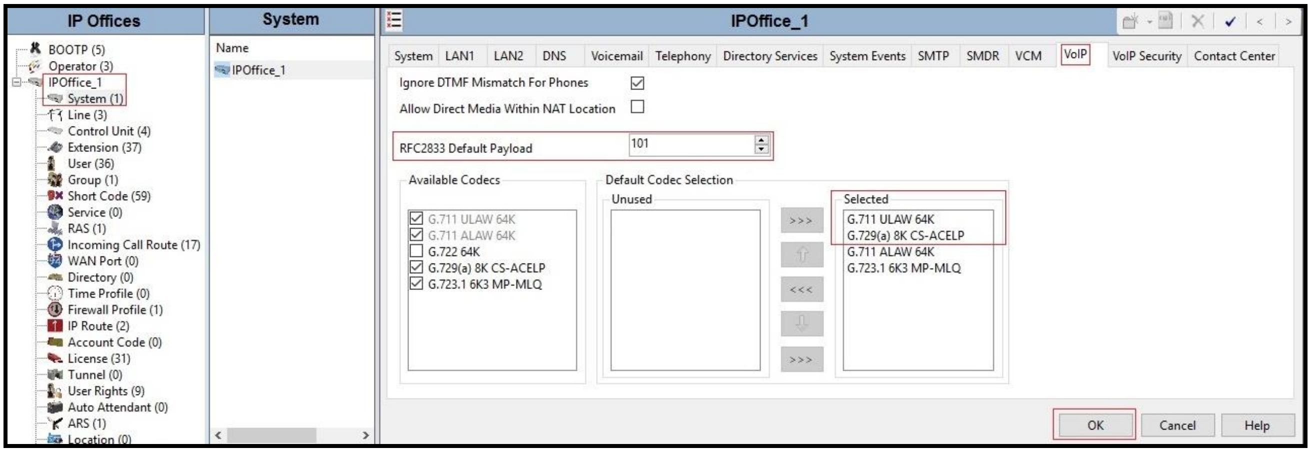

Figure 9 - Avaya IP Office VoIP

5.6. Administer SIP Line

A SIP Line is needed to establish the SIP connection between Avaya IP Office and EarthLink system. 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 Avaya IP Office Manager to create a SIP Line. Follow the steps in **Section 5.6.1** 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 5.6.2**.

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

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

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 5.6.2**. For the compliance test, SIP Line 17 was used as trunk for both outgoing and incoming calls.

5.6.1. Create SIP Line from Template

1. Create a new folder in the computer where Avaya IP Office Manager is installed (e.g., C:\EarthLink\Template). Copy the template file (**EL-IPO10.xml**) to this folder.
2. Create the SIP Trunk from the template: Right-click on **Line** in the Navigation Pane, then navigate to **New from Template → Open from file**.

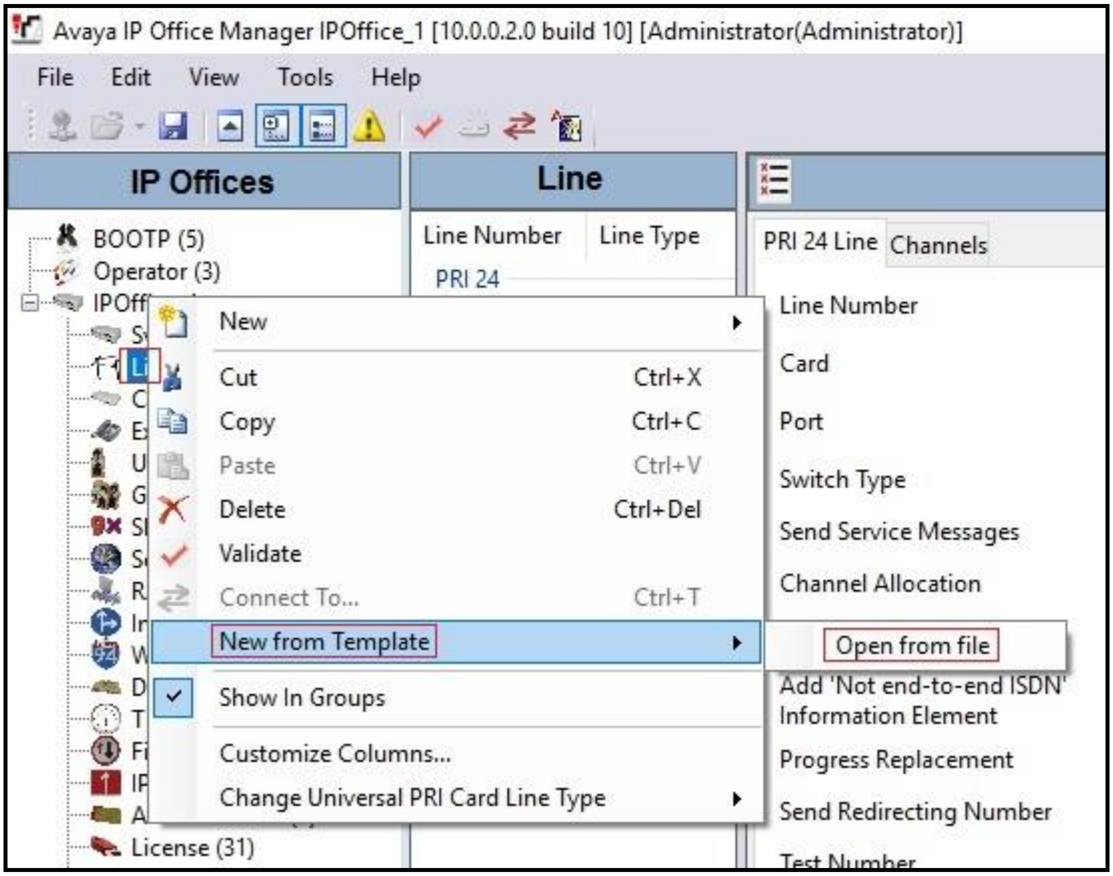

Figure 10 – Create SIP Line from Template

3. Select the **Template Files (*.xml)** and select the copied template from step 1 at IP Office template directory **C:\EarthLink\Template**. Click **Open** button to create a SIP line from template.

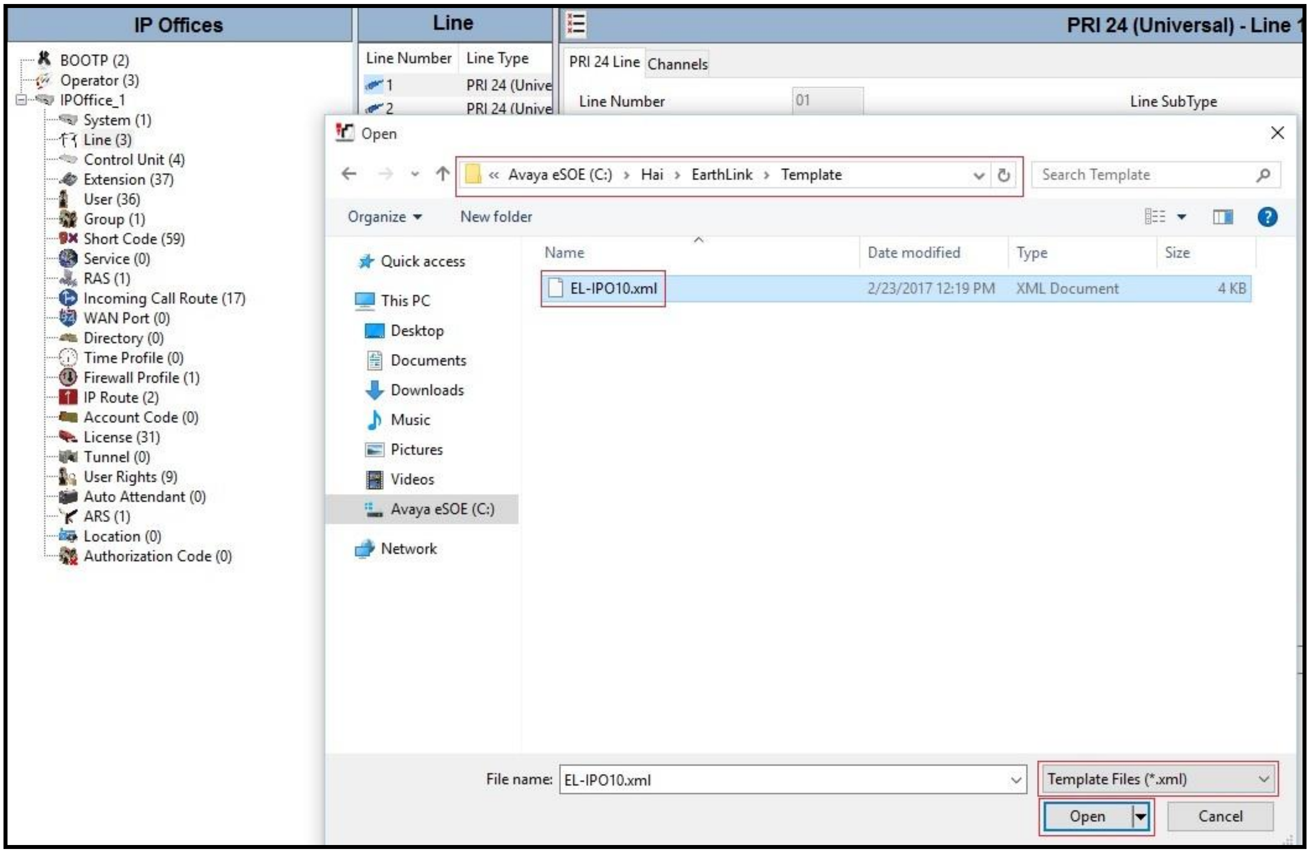

Figure 11 – Create SIP Line from IP Office Template Directory

A pop-up window below will appear stating success (or failure). Then click **OK** to continue.

Figure 12 – Create SIP Line from Template successfully

4. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.6.2**.

5.6.2. Create SIP Line Manually

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** (not shown).

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

- Select available **Line Number**, **17** in this case
- Set **ITSP Domain Name** to EarthLink SIP domain name. This field is used to specify the default host part of the SIP URI in the To and R-URI fields for outgoing calls
- Set **Local Domain Name** to IP address of Avaya IP Office LAN2 port. This field is used to specify the default host part of the SIP URI in the From field for outgoing calls
  Note: For the user making the call, the user part of the From SIP URI is determined by the settings of the SIP URI channel record being used to route the call (see SIP URI → Local URI). For the destination of the call, the user part of the To and R-URI fields are determined by dial short codes of the form 9N;/N"@stat-msblt-pmmar.voiplab.elnk.us" where N is the user part of the SIP URI and "@stat-msblt-pmmar.voiplab.elnk.us" can be used to override the host part of the To and R-URI
- Check the **In Service** and **Check OOS** boxes
- Set **URI Type** to **SIP**
- For **Session Timers**, set **Refresh Method** to **Re-invite** with **Timer (sec)** to **1200**
- Set **Name Priority** to **Favor Trunk**. As described in **Section 5.4**, the **Default Name Priority** parameter may retain the default **Favor Trunk** setting, or can be configured to **Favor Directory**. As shown below, the default **Favor Trunk** setting was used in the reference configuration
- For **Redirect and Transfer**, set **Incoming Supervised REFER** and **Outgoing Supervised REFER** to **Auto** or **Always**. Note: EarthLink supports both re-Invite and SIP Refer in off-net transfer call during the compliance testing
- Default values may be used for all other parameters
- Click **OK** to commit then press Ctrl + S to save

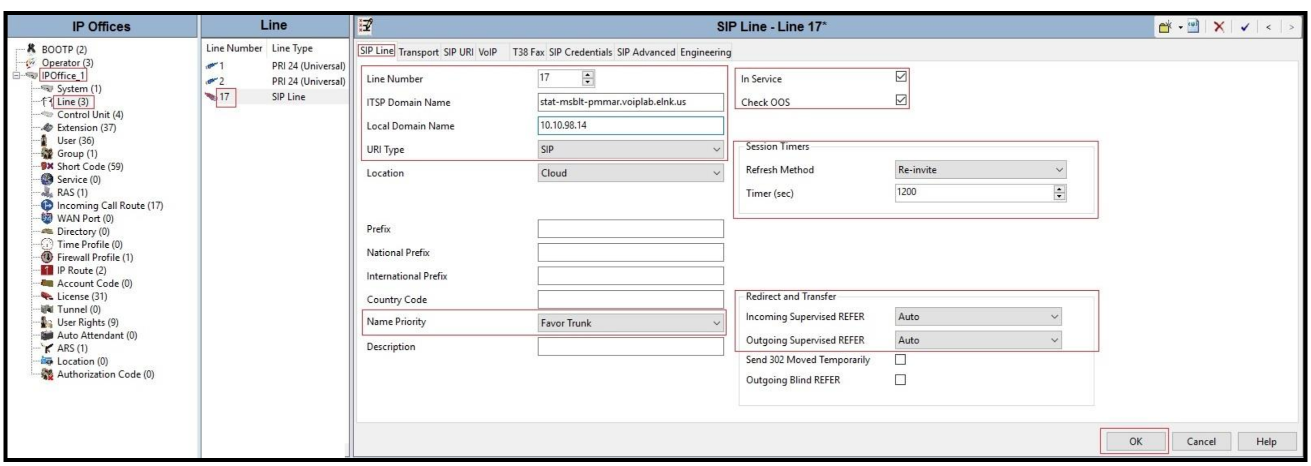

Figure 13 – SIP Line Configuration

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 EarthLink Signaling Server: **192.168.193.132**. This is the SIP Proxy IP address used for outgoing SIP calls
- In the **Network Configuration** area, **UDP** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **5060**
- The **Use Network Topology Info** parameter was set to **LAN 2**. This associates the SIP Line 17 with the parameters in the **IPOffice_1 → System (1) → LAN2 → Network Topology** tab. The **Listen Port** was set to **5060**
- The **Calls Route via Registrar** was unchecked as EarthLink did not support the dynamic Registration on the SIP Trunk
- Other parameters retain default values
- Click **OK** to commit then press Ctrl + S to save

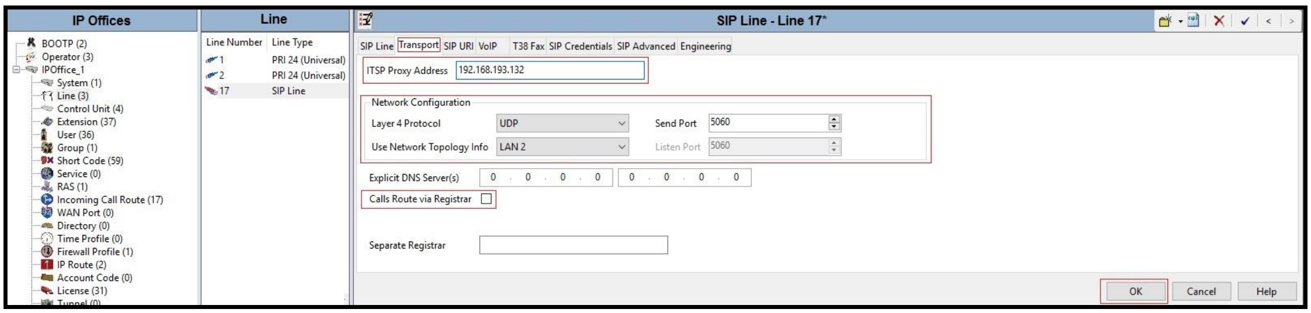

Figure 14 – SIP Line Transport Configuration

A SIP Credentials entry must be created for Digest Authentication used by EarthLink to authenticate calls from the enterprise to the PSTN. To create a SIP Credentials entry, first select the **SIP Credentials** tab. Click the **Add** button and the **New SIP Credentials** 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 SIP Credentials area will be opened. In the example screen below, a previously configured entry is edited. The entry was created with the parameters shown below:

- Set **User name** and **Authentication Name** to the value provided by the service provider
- Set **Password** and **Confirm Password** to the value provided by the service provider **Expiration (mins)** is set to **60**
- Un-check the **Registration required** option. EarthLink does not require registration for Digest Authentication

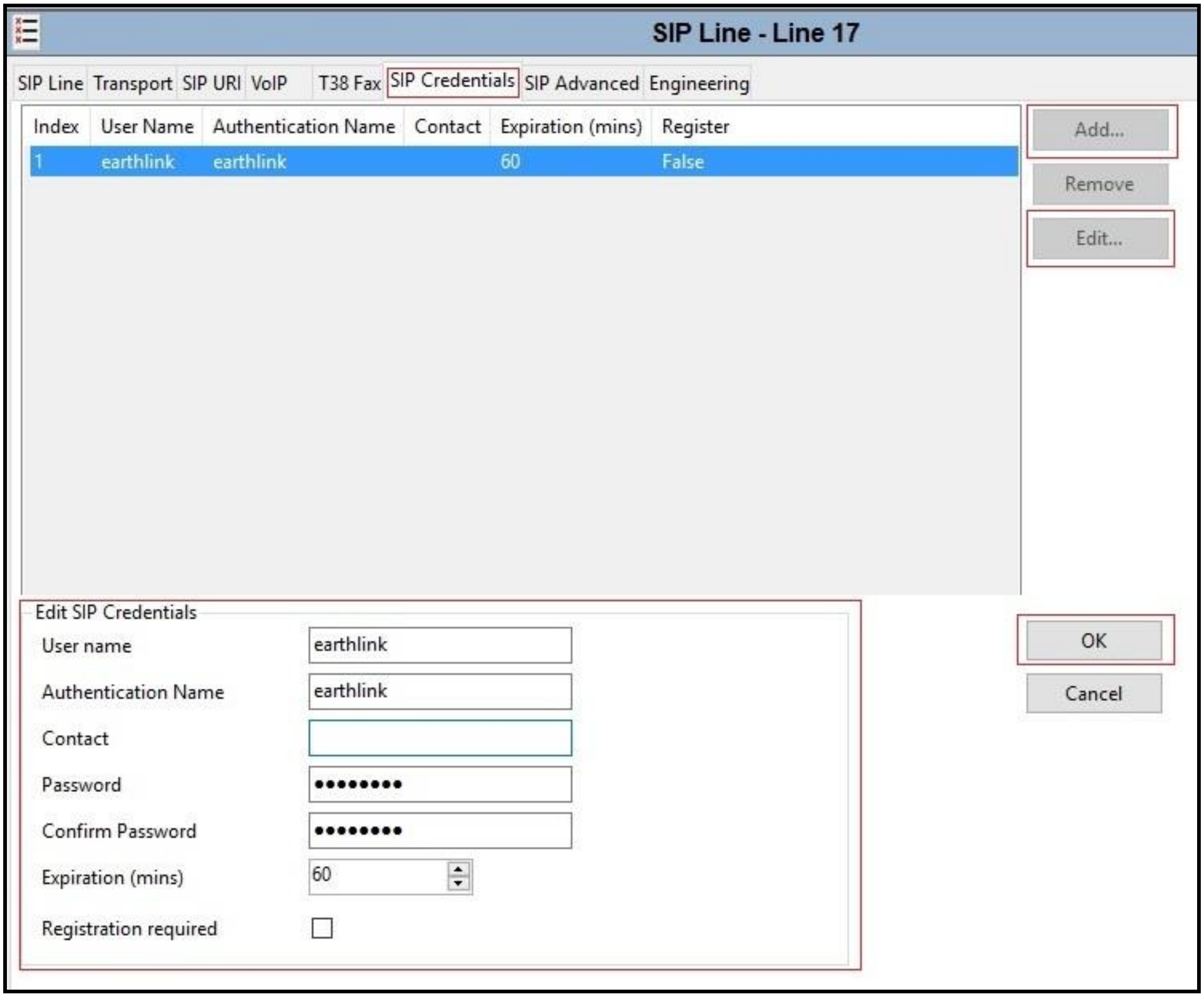

Figure 15 – SIP Line SIP Credentials Configuration

The SIP URI entry must be created to match any DID number assigned to an Avaya IP Office user and Avaya IP Office will route the calls on this SIP 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 on next page, the previously configured entries are edited.

A 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** to **Use 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 **Section5.8**
- Set **Identity** to **Auto** and **Header** to **P Asserted ID**
- For **Forwarding And Twinning**, set **Send Caller ID** to **Diversion Header**
  Note: When using twinning feature, the calling party number displayed on the twinned phone is controlled by **Send Caller ID** parameter
- Set **Diversion Header** to **None**
- Set **Registration** to **1: earthlink**
- Associate this line with an incoming line group in the **Incoming Group** field and an outgoing line group in the **Outgoing Group** field. This line group number will be used in defining incoming and outgoing call routes for this line. For the compliance test, a new line group **17** was defined that only contains this line (line 17)
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern
- Click **OK** to submit the changes

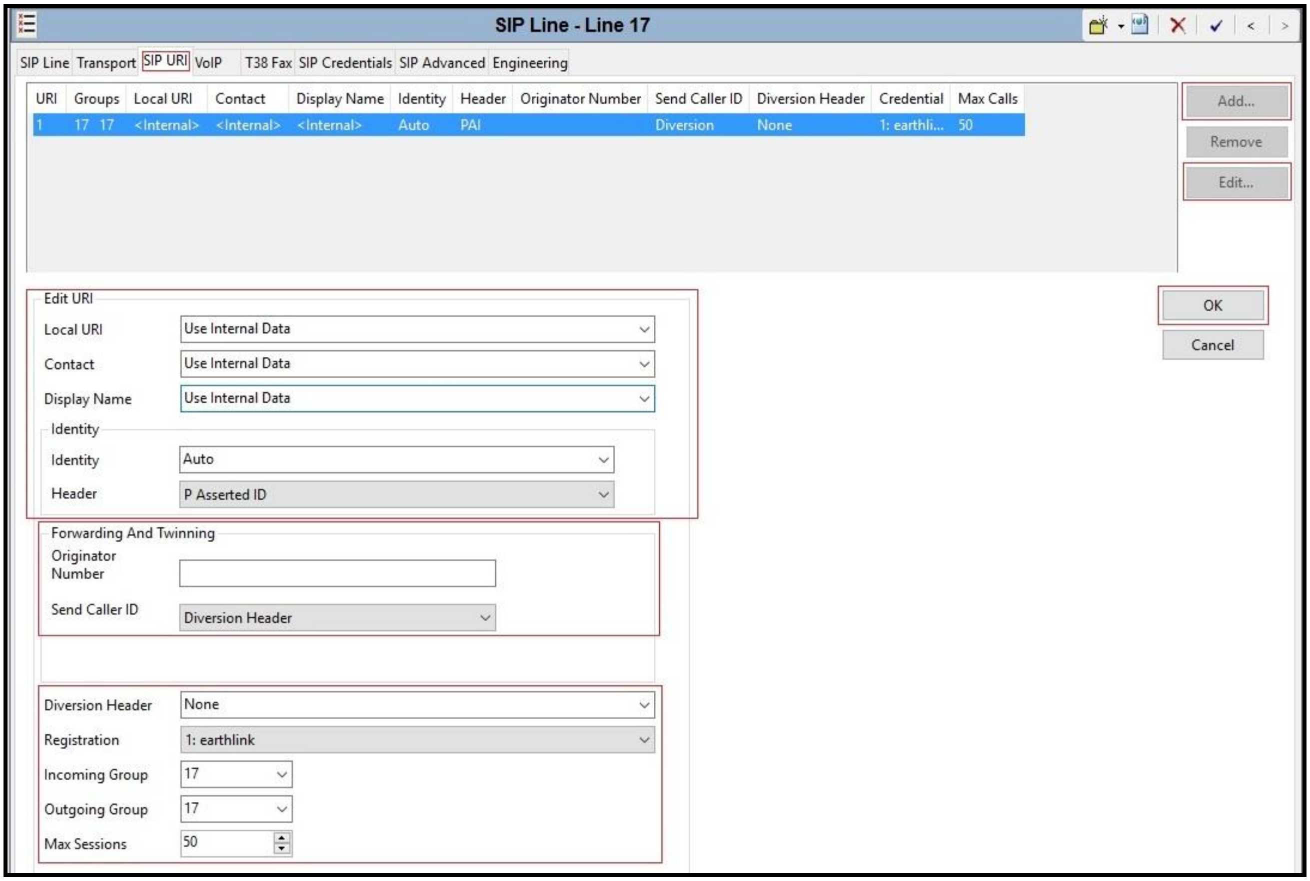

Figure 16 – SIP Line SIP URI Configuration

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. The **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** codecs are selected. Avaya IP Office supports these codecs, which are sent to the EarthLink, in the Session Description Protocol (SDP) offer
- Check the **Re-invite Supported** box
- Set **Fax Transport Support** to **G.711** from the pull-down menu. **Note**: EarthLink supports only G.711 pass-through mode during the compliance testing
- 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** to submit the changes

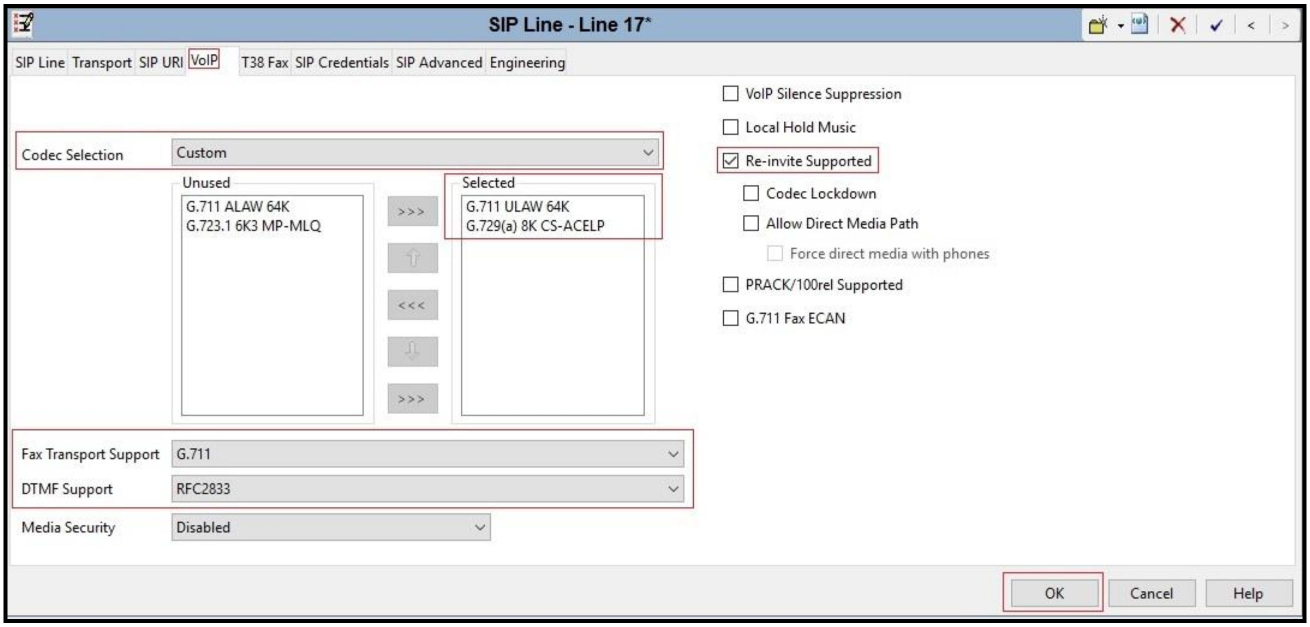

Figure 17 – SIP Line VoIP Configuration

5.7. Outgoing Call Routing – Short Code

The following section describes the Short Code for outgoing calls to EarthLink.

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 "**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"@stat-msblt-pmmar.voiplab.elnk.us"**. 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 EarthLink domain name
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the **SIP URI** tab on the **SIP Line** in **Section 5.6.2**. This short code will use this line group when placing the outbound call
- Set the **Locale** to **United States (US English)**
- Default values may be used for all other parameters
- Click **OK** to submit the changes

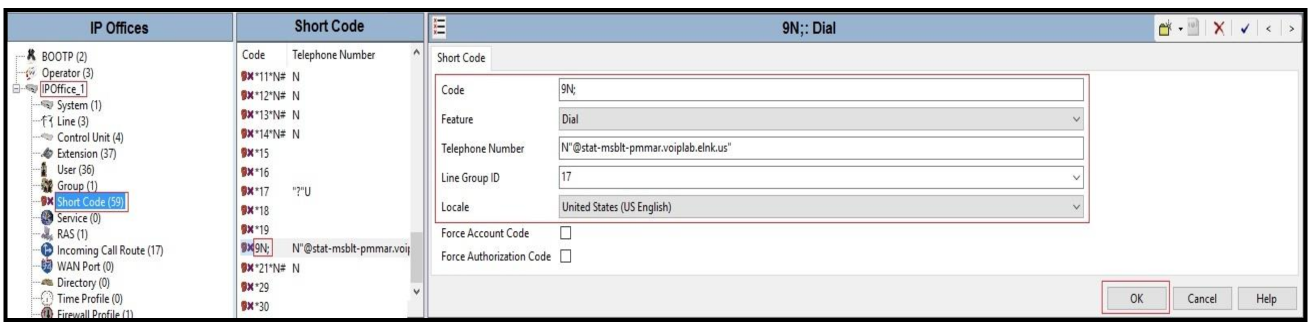

Figure 18 – Short Code 9N

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**
- Set the **Locale** to **United States (US English)**
- Default values may be used for other parameters
- Click **OK** to submit the changes

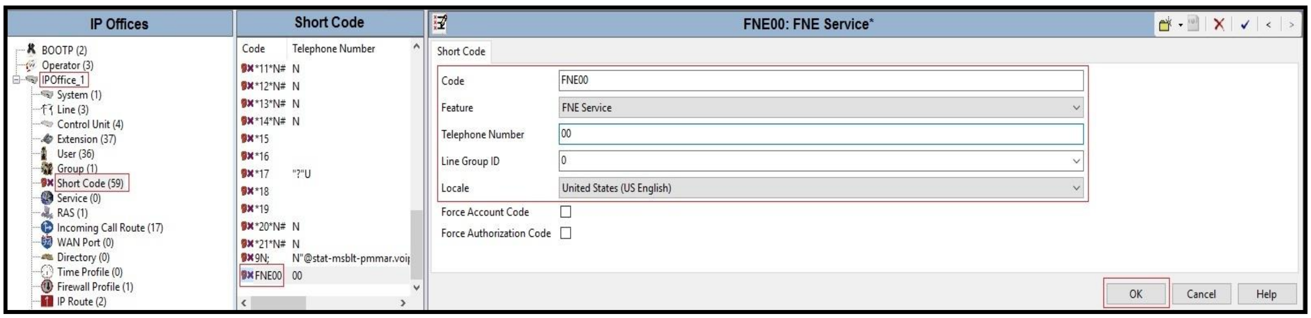

Figure 19 – Short Code FNE

5.8. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line defined in **Section 5.6**. 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 **2372**. 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 **2372**. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise provided by EarthLink. 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.

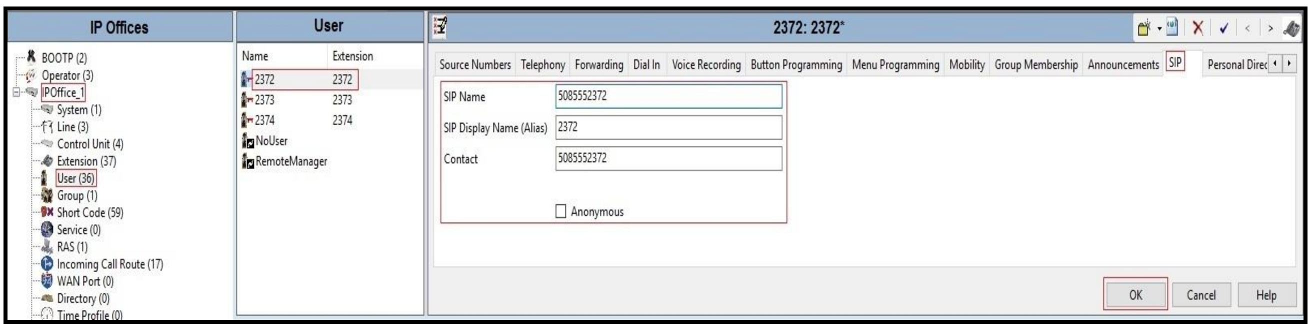

Figure 20 – User Configuration

One of the H.323 IP Deskphones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for **User 2372**. 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 **91613XXX5281**. Check **Mobile Call Control** to allow incoming calls from mobility extension to access FNE00 (defined in **Section 5.7**). Other options can be set according to customer requirements.

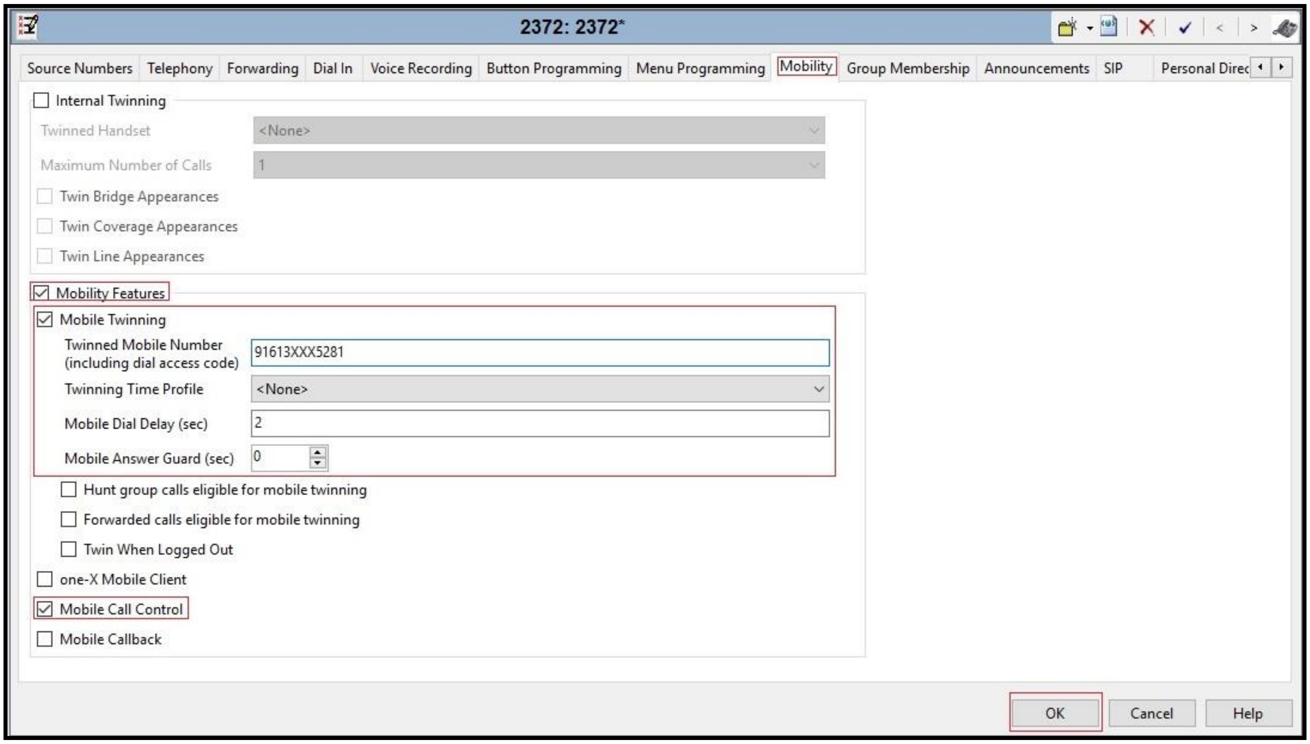

Figure 21 – Mobility Configuration for User

5.9. 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 5.6.2**
- Set the **Incoming Number** to the incoming DID number on which this route should match.
- Default values can be used for all other fields

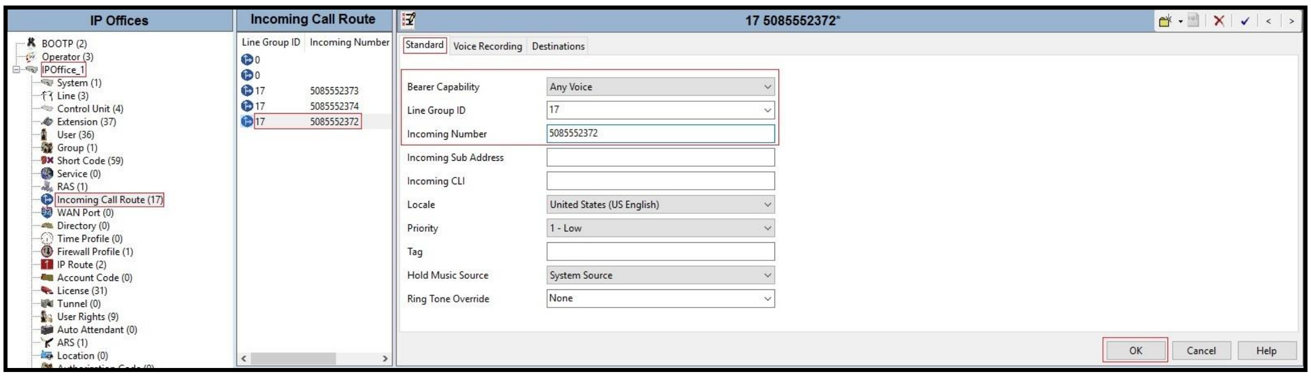

Figure 22 – Incoming Call Route Configuration

On the **Destination** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **5085552372** on line 17 are routed to **Destination 2372 2372** as below screenshot:

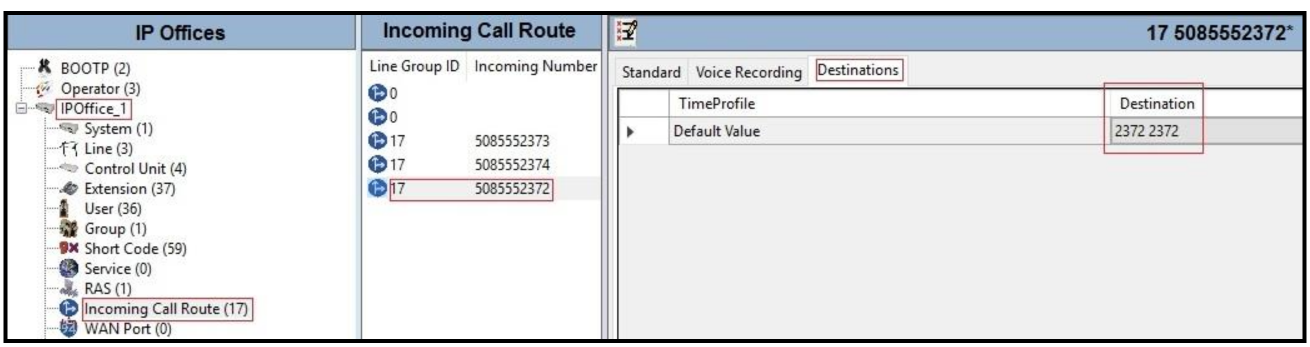

Figure 23 – Incoming Call Route for Destination 2372

For Feature Name Extension Service testing purpose, the incoming calls to DID number **5085552373** were configured to access **FNE00**. The **Destination** was appropriately defined as **FNE00** as below screenshot:

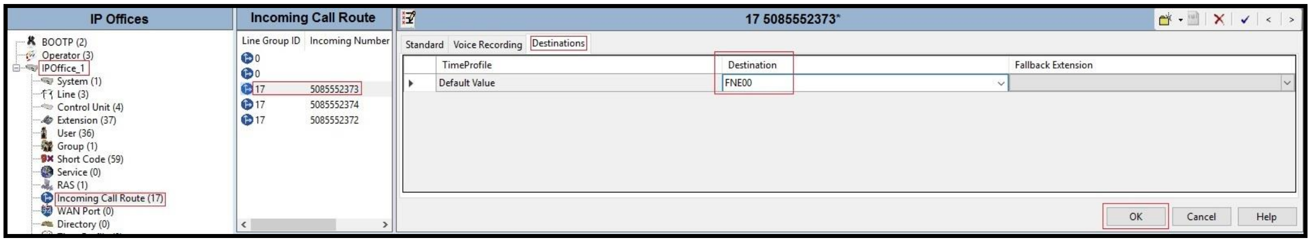

Figure 24 – Incoming Call Route for Destination FNE

For Voice Mail testing purpose, the incoming calls to DID number **5085552374** were configured to access **VoiceMail**. The **Destination** was appropriately defined as **VoiceMail** as below screenshot:

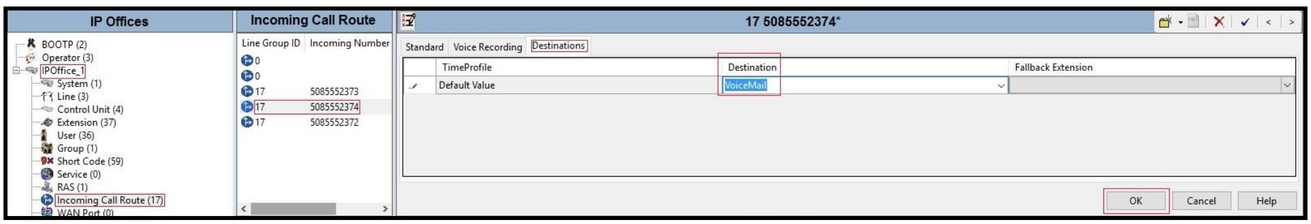

Figure 25 – Incoming Call Route for Destination Voice Mail

5.10. 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. EarthLink SIP Trunk Configuration

EarthLink is responsible for the configuration of EarthLink SIP Trunk Service. The customer must provide the IP address used to reach the Avaya IP Office LAN2 port at the enterprise. EarthLink will provide the customer necessary information to configure the SIP connection between Avaya IP Office and EarthLink. The provided information from EarthLink includes:

- IP address and port number used for signaling or media servers through any security
- DID numbers
- EarthLink SIP Trunk Specification (If applicable)

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).

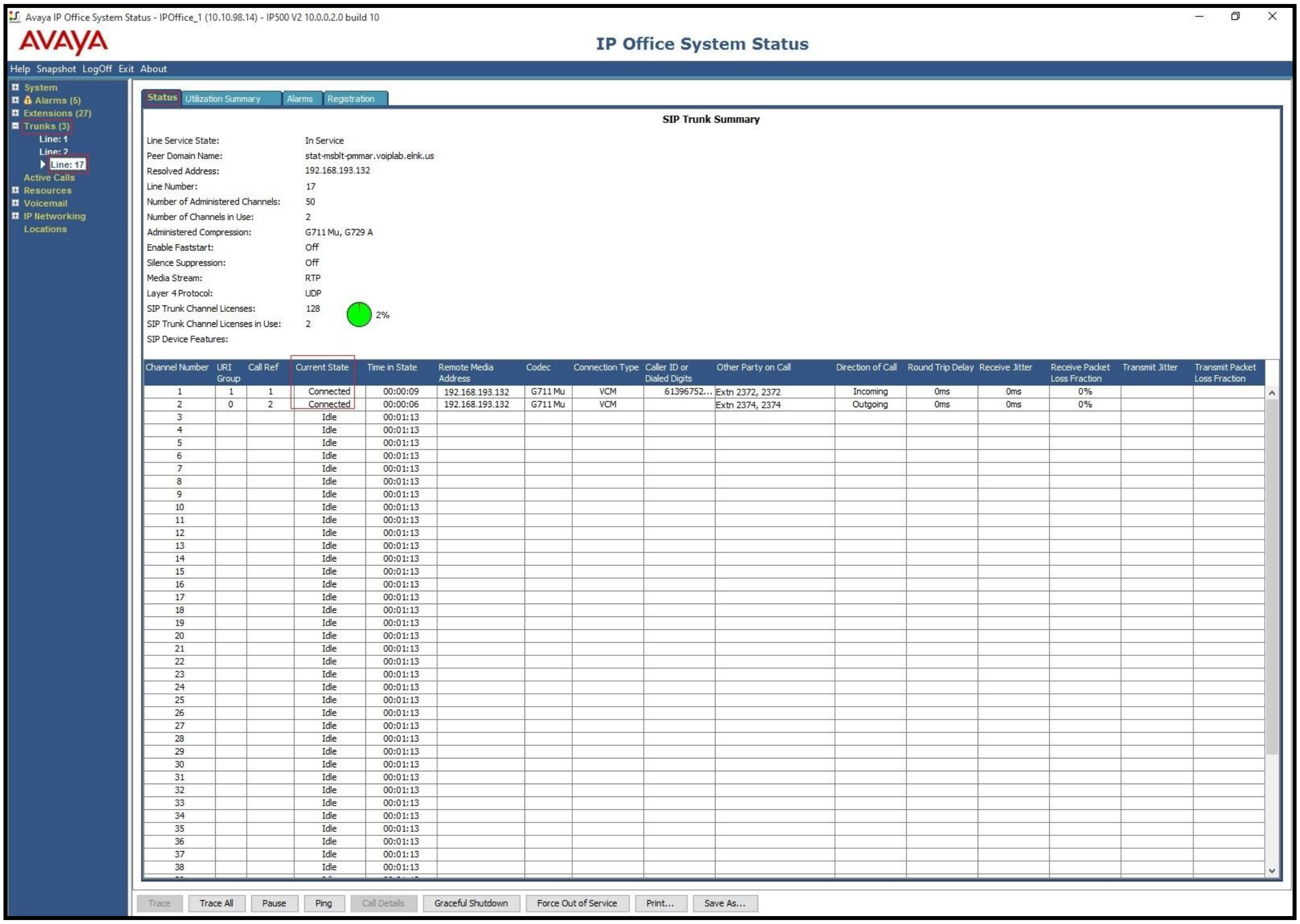

Figure 26 – SIP Trunk status

- 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.

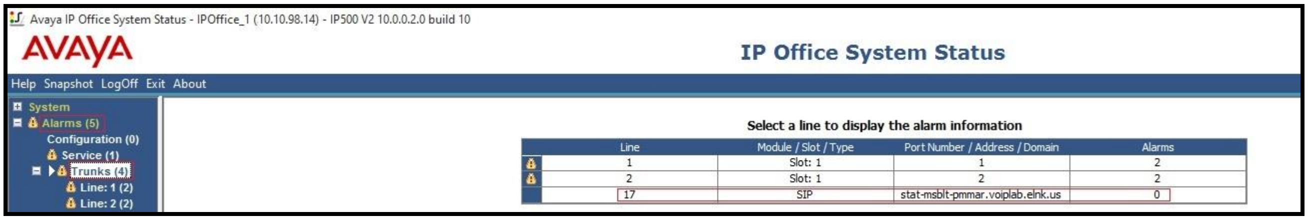

Figure 27 – SIP Trunk alarm

- 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.
- Use a network sniffing tool e.g., Wireshark to monitor the SIP signaling between the enterprise and EarthLink. The sniffer traces are captured at the LAN2 port interface of the Avaya IP Office.

8. Conclusion

EarthLink passed compliance testing excepting the limitation in **Section 2.2**. These Application Notes describe the procedures required to configure the SIP connections between Avaya IP Office and the EarthLink system as shown in **Figure 1**.

9. Additional References

[1] Administering Avaya IP Office Platform with Manager, Release 10.0, August 2016.
[2] Deploying Avaya IP Office™ Platform Solution, Release 10.0, August 2016.
[3] Using Avaya Communicator for Web, Release 1.0, Issue 1.0.6, May 2016.
[4] Using Avaya Communicator for Windows on IP Office, Release 10.0, August 2016.

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 EarthLink SIP Trunk may be found at: https://www.earthlink.com

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