



## **Application Notes for Configuring Avaya IP Office Release 9.0 to support EarthLink SIP Trunking Services – Issue 1.0**

### **Abstract**

These Application Notes describe the steps necessary for configuring Session Initiation Protocol (SIP) Trunk Service for an enterprise solution using Avaya IP Office Release 9.0 to interoperate with EarthLink SIP Trunking Services.

The test was performed to verify SIP trunk features including basic calls, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls were placed to and from the PSTN with various Avaya endpoints.

EarthLink SIP Trunking Services provides PSTN access via a SIP trunk between the enterprise and EarthLink's 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 steps necessary for configuring Session Initiation Protocol (SIP) Trunk Service for an enterprise solution using Avaya IP Office Release 9.0 to interoperate with EarthLink SIP Trunking Services.

In the sample configuration, the Avaya IP Office solution consists of Avaya IP Office (hereafter referred to as IP Office) 500v2 Release 9.0 and various Avaya endpoints, including Avaya IP Office Video Softphone, Avaya Flare® Experience for Windows and Avaya deskphones, including SIP, H.323, digital, and analog.

The EarthLink SIP Trunking Services referenced within these Application Notes is designed for business customers. Customers using this service with the IP Office solution are able to place and receive PSTN calls via a broadband WAN connection using the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI trunks. This approach generally results in lower cost for the enterprise.

The terms “Service Provider” and “EarthLink” will be used interchangeably throughout these Application Notes.

## 2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to EarthLink’s network via the public Internet and exercise the features and functionalities listed in **Section 2.1**. The simulated enterprise site was comprised of IP Office Release 9.0 and various Avaya endpoints, listed in **Section 4**.

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

To verify SIP trunk interoperability, the following features and functionalities were covered during the interoperability compliance test:

- SIP OPTIONS queries and responses.
- Incoming PSTN calls to various Avaya endpoints, including SIP, H.323, digital and analog at the enterprise. All incoming calls from the PSTN were routed to the enterprise across the SIP trunk from the service provider networks.

- Outgoing PSTN calls from Avaya endpoints including SIP, H.323, digital and analog telephone at the enterprise. All outgoing calls to the PSTN were routed from the enterprise across the SIP trunk to the service provider network.
- Incoming and outgoing PSTN calls to/from Avaya IP Office Video Softphone.
- Incoming and outgoing PSTN calls to/from Avaya Flare® Experience for Windows.
- Dialing plans including local calls, international, outbound toll-free, etc.
- Caller ID presentation.
- Proper disconnect when the caller abandons the call before the call is answered.
- Proper disconnect via normal call termination by the caller or the called parties.
- Proper disconnect by the network for calls that are not answered (with voicemail off).
- Proper response to busy endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Proper codec negotiation and two way speech-path. (Testing was performed with codecs: G.729A and G.711U, which is the EarthLink preferred codec order).
- No matching codecs.
- G.711 fax pass-through.
- Proper early media transmissions.
- Voicemail and DTMF tone support (leaving and retrieving voice mail messages from PSTN phones).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response systems).
- Calling number blocking (Privacy).
- Call Hold/Resume (long and short duration).
- Call Forward (unconditional, busy, no answer).
- Blind Call Transfers.
- Consultative Call Transfers.
- Station Conference.
- Mobility twinning of incoming calls to mobile phones.

Items not supported or not tested included the following:

- EarthLink does not support T.38 fax; only G.711 fax pass-through is currently supported.
- Operator (0) and Operator-Assisted (0 + 10-digits) calls were not tested.
- Inbound toll-free calls were not tested.
- 911 Emergency calls were not tested.

## 2.2. Test Results

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

- **SIP REFER method:** When using REFER to blind transfer an incoming call back to the PSTN, EarthLink responded to REFER from the enterprise with “202 Accepted” followed by "491 Request Pending" embedded in the Notify message, instead of “200 OK”. User experience was not negatively affected (i.e., the call was transferred successfully), however both calls remain connected to IP Office for the duration of the call. Consultative transfer of similar call to PSTN worked properly (EarthLink sends a Notify with "200 OK"). The compliance test was done with REFER Support enabled (see **Section 5.4.2**)
- **Blind Call Transfer using SIP phones do not complete until after the transferee answers the call:** When Blind Transfers are executed from Avaya 1100 Series SIP phones the transfer does not complete until after the end user (transferee) answers the call. **Scenario:** A PSTN user calls an IP Office SIP extension (Avaya 1100 Series SIP phone), the call is answered. The IP Office user then proceeds to do a blind transfer to another endpoint. **Result:** The expected behavior on the Avaya 1100 Series SIP phone is to display “transfer completed” after answering “No” to the question “Consultative transfer with party?” which implies a blind transfer. Instead, the Avaya 1100 Series SIP phone continues to display “transferring” until the transferee answers the call. This is a known issue with the Avaya 1100 Series phone, the work around is to hang up the SIP phone. There is no user impact, the transfer completes successfully. This issue is only seen with the Avaya 1100 Series phones.
- **Call from IP Office to a PSTN number that is Busy:** EarthLink does not respond with “486 Busy Here” to calls from IP Office endpoints to a PSTN number that is busy. There is no user impact, the user hears busy tone, it’s listed here simply as an observation.
- **Call Display on Transferred Calls to the PSTN:** The Caller ID display is not updated on PSTN phones involved with call transfers from IP Office to the PSTN. After the call transfer is completed, the PSTN phone does not display the actual connected party but instead shows the number of the host extension that initiated the call transfer (transferor). The PSTN phone display is ultimately controlled by the PSTN provider, thus this behavior is not necessarily indicative of a limitation of the combined Avaya/EarthLink solution. It is listed here simply as an observation.
- **No matching codec on outbound calls:** If an unsupported audio codec is received by EarthLink on the SIP trunk (e.g., 723.1), EarthLink will respond with “503 Service Unavailable” instead of “488 Not Acceptable Here”, the user will hears re-order tone.

## 2.3. Support

For support on EarthLink systems visit the corporate Web page at <http://www.earthlinkbusiness.com/>

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

### 3. Reference Configuration

**Figure 1** illustrates the test configuration used for the DevConnect compliance testing. The test configuration simulates an enterprise site with an Avaya SIP-enabled enterprise solution connected to the EarthLink SIP Trunking Services through the public Internet.

The Avaya components used to create the simulated enterprise customer site includes:

- Avaya IP Office 500v2.
- Avaya Voicemail Pro for IP Office.
- Avaya 96x0 Series H.323 IP Deskphones.
- Avaya 96x1 Series H.323 IP Deskphones.
- Avaya 1100 Series SIP IP Deskphones.
- Avaya IP Office Video Softphone.
- Avaya Flare® Experience for Windows.
- Avaya 1408 Digital Deskphones.
- Avaya 9508 Digital Deskphones.
- Analog Deskphone.
- Fax Machines.

The enterprise site contains the Avaya IP Office 500v2 with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. The LAN1 port of IP Office is connected to the enterprise LAN (private IP network) while the LAN2 port is connected to the public IP network. Endpoints include Avaya 96x0 and 96x1 Series IP Deskphones (with H.323 firmware), Avaya 1100 Series IP Deskphones (with SIP firmware), Avaya 1408 and 9508 Digital Deskphones, Analog Deskphones, Fax Machines, PC running Avaya IP Office Softphone and Avaya Flare® Experience for Windows. The site also has a Windows PC running Avaya IP Office Manager to configure and administer the IP Office system, and Avaya Voicemail Pro providing voice messaging service to the IP Office users. Mobile Twinning is configured for some of the IP Office users so that calls to these user's extensions will also ring and can be answered at the configured mobile phones.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to EarthLink. The short code 9 was stripped off by IP Office but the remaining "N" digits were sent unaltered to the network. Refer to **Section 5.5**.

In an actual customer configuration, the enterprise site may include additional network components between the service provider and the IP Office system, 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 all SIP and RTP traffic between the service provider and the IP Office system must be allowed to pass through these devices.

For confidentiality and privacy purposes, public IP addresses, user names, passwords and routable DID numbers used during the compliance testing have been masked.

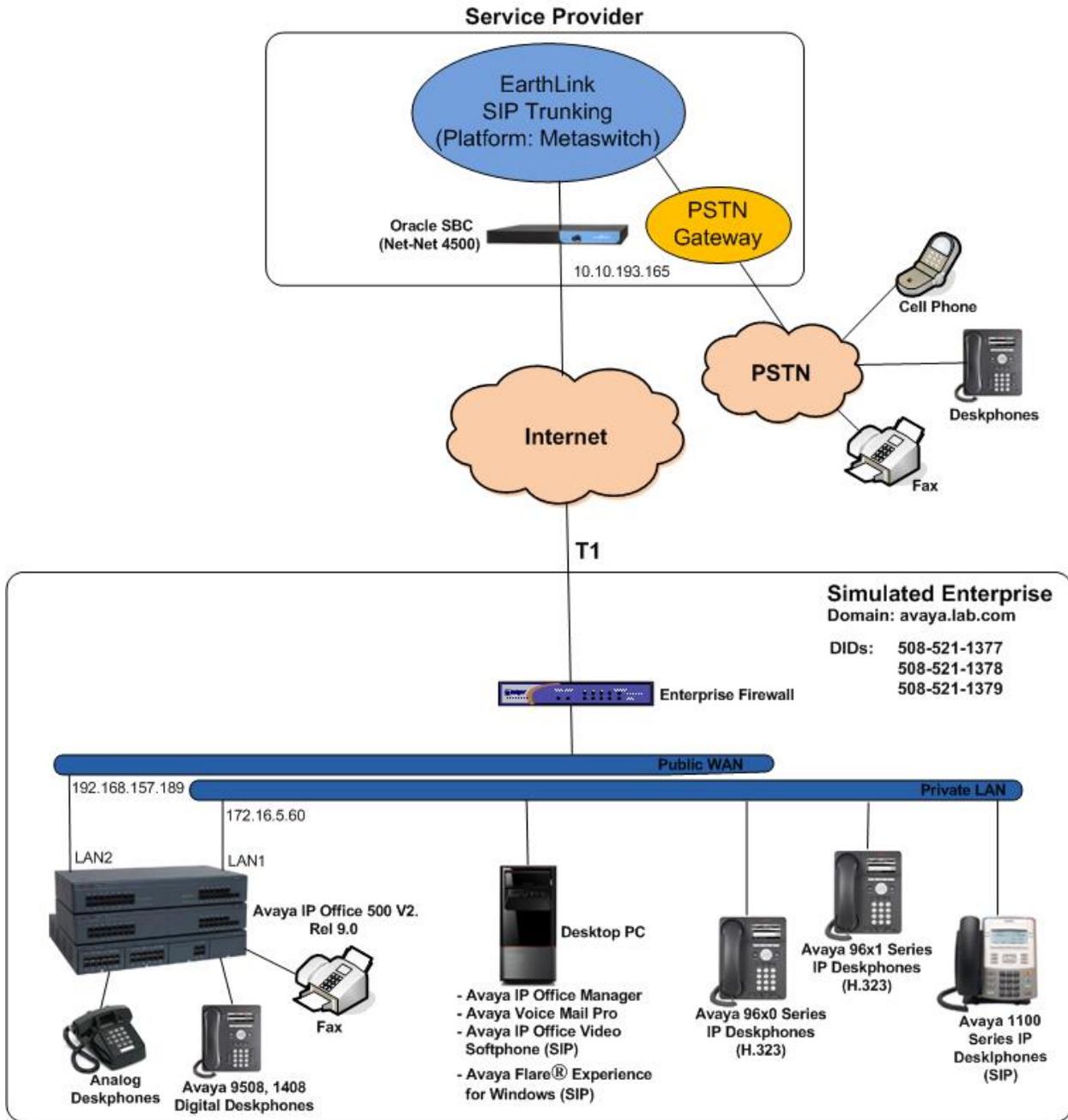


Figure 1: Avaya Interoperability Test Lab Configuration

## 4. Equipment and Software Validated

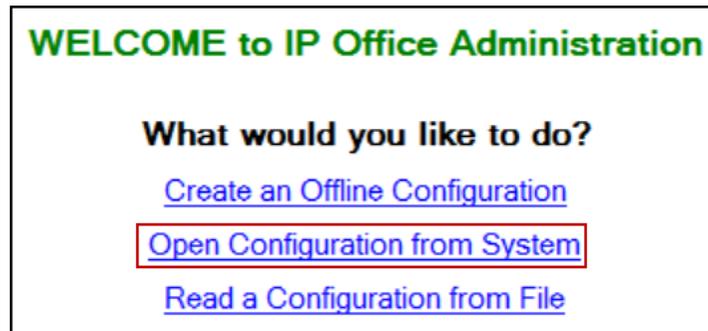
The following equipment and software were used for the sample configuration provided:

| Equipment/Software                      | Release/Version                             |
|---|---|
| <b>Avaya</b>                            |   |
| Avaya IP Office 500v2                   | 9.0.4.0 Build 965                           |
| Avaya IP Office DIG DCPx16 V2           | 9.0.4.0 Build 965                           |
| Avaya IP Office Manager                 | 9.0.4.0 Build 965                           |
| Avaya Voicemail Pro Client              | 9.0.4.0 Build 18                            |
| Avaya 96x0 Series IP Deskphones (H.323) | Avaya one-X® Deskphone Edition S3.220A      |
| Avaya 96x1 Series IP Deskphones (H.323) | Avaya one-X® Deskphone H.323 Version 6.4014 |
| Avaya 1120E IP Deskphones (SIP)         | SIP1120e Ver. 04.04.14.00                   |
| Avaya IP Office Video Softphone         | 3.2.3.49 68975                              |
| Avaya Flare® Experience for Windows     | 1.1.4.23                                    |
| Avaya Digital Deskphones 1408           | 38.0  |
| Avaya Digital Deskphones 9508           | 0.55  |
| Lucent Analog Phone                     | --  |
| <b>EarthLink</b>                        |   |
| Metaswitch                              | 7.4   |
| Oracle SBC – Net-Net 4500               | SCX6.3.0 MR-5 Patch 1                       |

Testing was performed with Avaya IP Office 500v2, but this testing also applies to Avaya IP Office Server Edition running the same software release. Note that Avaya IP Office Server Edition requires an Expansion IP Office 500v2 R9 to support analog or digital endpoints or trunks.

## 5. Configure Avaya IP Office

IP Office is configured through the Avaya IP Office Manager application. From the PC running Avaya IP Office Manager, select **Start → Programs → IP Office → Manager** to launch the application. A screen that includes the following may be displayed.



Select **Open Configuration from System**. If the above screen does not appear, the configuration may be alternatively opened by navigating to **File → Open Configuration** at the top of the Avaya IP Office Manager window. Select the proper IP Office system from the pop-up window and log in with the appropriate credentials.

The appearance of the Avaya IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and show the Details pane on the right side. Since the Group pane has been omitted, its content is shown as submenus in the Navigation pane. These panes (Navigation, Group and Details) will be referenced throughout the IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and Avaya IP Office Video Softphone support) is assumed to already be in place.

In the sample configuration, the MAC address **00E00706530F** was used as the system name. All navigation described in the following sections (e.g., **License → SIP Trunk Channels**) appears as submenus underneath the system name **00E00706530F** in the Navigation Pane.

## 5.1. Licensing and Physical Hardware

The configuration and features described in these Application Notes require IP Office 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; click **License** in the Navigation pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details pane. Note that the full **License Key** in the screen below is not shown for security purposes.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'License (74)' selected. The main area shows the 'License' configuration page for a 'Remote Server'. The 'License Mode' is set to 'License Normal' and the 'PLDS Host ID' is 111309. Below this is a table listing various licenses.

| Feature                        | License Key     | Instances | Status   | Expiry Date | Source    |
|--------------------------------|-----------------|-----------|----------|-------------|-----------|
| Small Office Edition VCM (c... | 2KO78F6LVX4u... | 255       | Obsolete | Never       | ADI Nodal |
| Small Office Edition WiFi      | eAWwB35jVOJ...  | 255       | Obsolete | Never       | ADI Nodal |
| IPSec Tunnelling               | MIKcnXtjMKys... | 255       | Valid    | Never       | ADI Nodal |
| Proactive Reporting            | ttDp8nbs9N@...  | 255       | Valid    | Never       | ADI Nodal |
| Report Viewer                  | Tvct73mdgdGt... | 255       | Valid    | Never       | ADI Nodal |
| Mobility Features              | 0ICluRgHvKOL... | 255       | Obsolete | Never       | ADI Nodal |
| Advanced Small Commun...       | DaQJl7Ve5vUU... | 255       | Obsolete | Never       | ADI Nodal |
| IP500 Voice Networking Ch...   | T39BkqBXvd6a... | 255       | Valid    | Never       | ADI Nodal |
| IP500 Upgrade Standard to...   | QaHgn76v9j6C... | 255       | Obsolete | Never       | ADI Nodal |
| IP500 Voice Networking Ch...   | JaHLHAVFXjD...  | 4         | Valid    | Never       | ADI Nodal |
| SIP Trunk Channels             | I3CQzGBYDUsc... | 255       | Valid    | Never       | ADI Nodal |
| VPN IP Extensions              | @qm3fOoR5S_...  | 255       | Obsolete | Never       | ADI Nodal |

To view the physical hardware comprising IP Office, expand the components under the **Control Unit** in the Navigation pane. In the sample configuration, the Avaya IP Office 500v2 is equipped with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. An IP Office hardware configuration with a VCM component is necessary to support SIP Trunking Services.

To view the details of the component, select the component in the Navigation pane. The following screen shows the details of the **Avaya IP 500 V2**.

The screenshot displays the Avaya IP Office configuration interface. The left pane, titled 'IP Offices', shows a hierarchical tree of components. The 'Control Unit (4)' is expanded, and '1 IP 500 V2' is selected and highlighted with a red box. The right pane, titled 'IP 500 V2', shows the configuration details for the selected unit.

| IP 500 V2           |              |
|---------------------|--------------|
| Unit                |              |
| Device Number       | 1            |
| Unit Type           | IP 500 V2    |
| Version             | 9.0.300.941  |
| Serial Number       | 172.16.5.60  |
| Unit IP Address     | 172.16.5.60  |
| Interconnect Number | 0            |
| Module Number       | Control Unit |

## 5.2. System

Configure the necessary system settings. In an IP Office the LAN2 tab settings correspond to the IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side).

### 5.2.1. System – LAN2 Tab

In the sample configuration, the IP Office WAN port was used to connect to EarthLink. The LAN2 settings correspond to the WAN port on the Avaya IP Office 500v2. To access the LAN2 settings, first navigate to **System** → <Name>, where <Name> is the system name assigned to IP Office. In the case of the compliance test, the system name is the MAC address **00E00706530F**. Next, navigate to the **LAN2** → **LAN Settings** tab in the Details Pane. Set the **IP Address** field to the public IP address assigned to the IP Office WAN port. Set the **IP Mask** field to the mask used with the public IP address. All other parameters should be set to default or according to customer requirements. Click **Ok** to commit (not shown).

The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view under 'IP Offices' shows the system '00E00706530F' selected. The main pane shows the 'LAN2' tab for 'LAN Settings'. The 'IP Address' is set to '192 . 168 . 157 . 189' and the 'IP Mask' is '255 . 255 . 255 . 192'. Other settings include 'Primary Trans. IP Address' (0 . 0 . 0 . 0), 'Firewall Profile' (<None>), 'RIP Mode' (None), and 'Number Of DHCP IP Addresses' (1). The 'DHCP Mode' is set to 'Disabled'. A red box highlights the IP Address and IP Mask fields.

| Field                       | Value                         |
|-----------------------------|-------------------------------|
| IP Address                  | 192 . 168 . 157 . 189         |
| IP Mask                     | 255 . 255 . 255 . 192         |
| Primary Trans. IP Address   | 0 . 0 . 0 . 0                 |
| Firewall Profile            | <None>                        |
| RIP Mode                    | None                          |
| Number Of DHCP IP Addresses | 1                             |
| DHCP Mode                   | Server Client Dialin Disabled |

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

- Check the **SIP Trunks Enable** box to enable the configuration of SIP trunks.
- The **RTP Port Number Range** can be customized to a specific range of receive ports for RTP media. Based on this setting, IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2.

The screenshot displays the Avaya IP Office configuration interface. On the left is a tree view of the system hierarchy, with 'System (1)' selected and highlighted with a red box. The main pane shows the 'LAN2' configuration tab, with the 'VoIP' sub-tab active. The 'SIP Trunks Enable' checkbox is checked and highlighted with a red box. Below it, the 'RTP' section is also highlighted with a red box, showing the 'Port Number Range' and 'Port Number Range (NAT)' settings. Both sections have 'Minimum' and 'Maximum' values set to 49152 and 53246, respectively. Other visible settings include 'SIP Registrar Enable' (unchecked), 'Auto-create Extn/User' (checked), and various port configurations for UDP, TCP, and TLS.

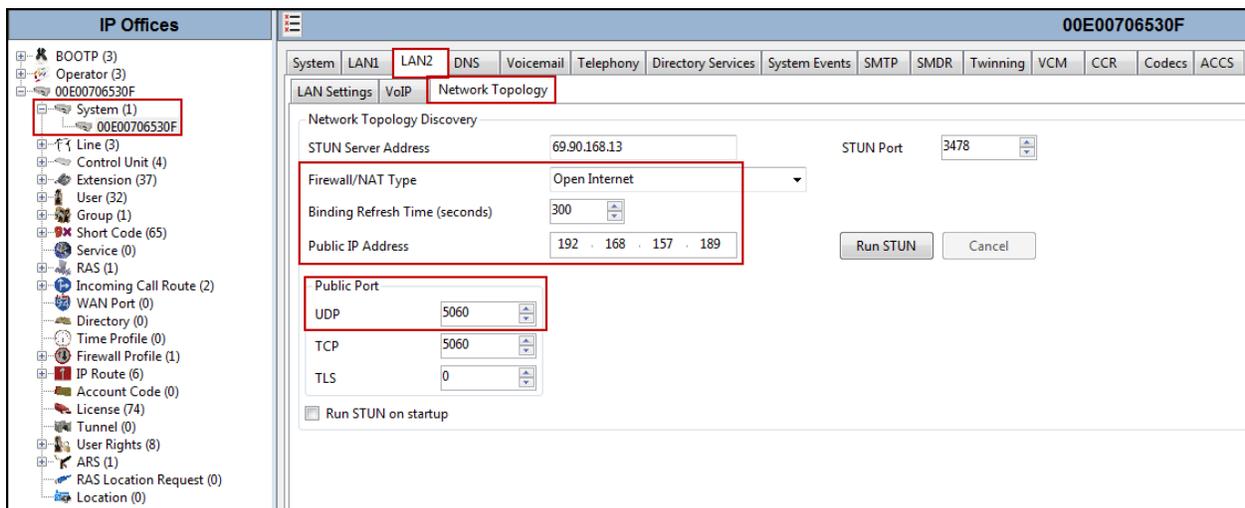
Scroll down the page.

- In the **RTP Keepalives** section, set the **Scope** to **RTP**. Set the **Periodic timeout** to **30** and the **Initial keepalives** parameter to **Enabled**. These settings will cause IP Office to send a RTP keepalive packet starting at the time of initial connection and every 30 seconds thereafter if no other RTP traffic is present. This facilitates the flow of media in cases where each end of the connection is waiting to see media from the other, as well as helping to keep firewall ports open for the duration of the call.
- In the **DiffServ Settings** section, 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 and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values will be provided by the customer.
- All other parameters should be set to default or according to customer requirements.
- Click **Ok** to commit (not shown).

The screenshot displays the Avaya IP Office Manager configuration interface. The title bar reads "Avaya IP Office Manager 00E00706530F [9.0.400.965] [Administrator/Administrator]". The main window is titled "IP Offices" and shows a tree view on the left with "System (1)" selected. The right pane is divided into tabs: "System", "LAN1", "LAN2", "DNS", "Voicemail", "Telephony", "Directory Services", "System Events", "SMTP", "SMDR", "Twinning", "VCM", "CCR", "Codecs", and "ACCS". The "LAN2" tab is active, and the "VoIP" sub-tab is selected. The "RTP" section is expanded, showing "Port Number Range" (Minimum: 49152, Maximum: 53246) and "Port Number Range (NAT)" (Minimum: 49152, Maximum: 53246). A red box highlights the "Enable RTCP Monitoring on Port 5005" checkbox (checked), the "Scope" dropdown (set to "RTP"), the "Periodic timeout" field (set to "30"), and the "Initial keepalives" dropdown (set to "Enabled"). The "DiffServ Settings" section is also visible, showing DSCP values: 88 (DSCP), 88 (Video DSCP), FC (DSCP Mask), 88 (SIG DSCP), 46 (DSCP), 46 (Video DSCP), 63 (DSCP Mask), and 34 (SIG DSCP).

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. Since no firewall or network address translation (NAT) device was used between IP Office and the EarthLink, the parameter was set to **Open Internet**.
- Set the **Binding Refresh Time (seconds)** to a desired value, the value of **300 (or every 5 minutes)** was used during the compliance testing. This value is used to determine the frequency that IP Office will send OPTIONS heartbeat to the service provider.
- Set **Public IP Address** to the IP address of the IP Office WAN port.
- In the **Public Port** section, next to the transport protocol **UDP**, select the UDP port on which IP Office will listen.
- All other parameters should be set to default or according to customer requirements.
- Click **Ok** to commit (not shown).



**Note:** In the compliance test, the LAN1 interface was used to connect IP Office to the enterprise site IP network (private network). The LAN1 interface configuration is not directly relevant to the interface with the EarthLink SIP Trunking Service, and therefore is not described in these Application Notes.

## 5.2.2. System - Telephony Tab

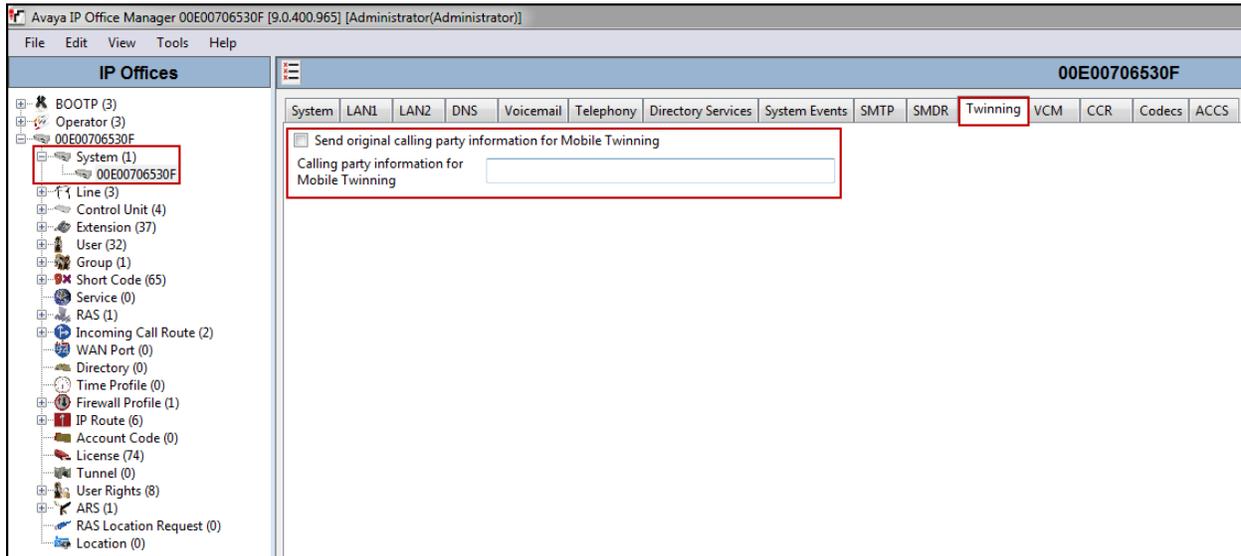
To access the System Telephony settings, navigate to the **Telephony** → **Telephony** tab in the Details Pane. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN. If for security reasons incoming calls should not be allowed to transfer back to the PSTN then leave this setting checked. All other parameters should be set to default or according to customer requirements. Click **OK** to commit (not shown).

The screenshot displays the Avaya System Manager interface for system 00E00706530F. The left-hand pane shows a tree view of system components, with 'System (1)' and its sub-item '00E00706530F' highlighted. The main pane shows the 'Telephony' configuration page, which 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). There is also a checkbox for 'Restrict Analogue Extension Ringer Voltage'.
- Companing Law:** Contains two columns: 'Switch' and 'Line'. Each column has radio buttons for 'U-Law' and 'A-Law'. 'U-Law' is selected in both.
- Checkboxes:** A list of settings including 'DSS Status', 'Auto Hold', 'Dial By Name', 'Show Account Code', 'Inhibit Off-Switch Forward/Transfer' (highlighted with a red box), 'Restrict Network Interconnect', 'Drop External Only Impromptu Conference', 'Visually Differentiate External Call', 'Unsupervised Analog Trunk Disconnect Handling', 'High Quality Conferencing', 'Strict SIPs', and 'Digital/Analogue Auto Create User'.
- Fields:** Several numeric fields with up/down arrows: 'Dial Delay Time (secs)' (3), 'Dial Delay Count' (0), 'Default No Answer Time (secs)' (15), 'Hold Timeout (secs)' (0), 'Park Timeout (secs)' (300), and 'Ring Delay (secs)' (5). There are also dropdown menus for 'Call Priority Promotion Time (secs)' (Disabled), 'Default Currency' (USD), 'Default Name Priority' (Favor Trunk), and 'Media Connection Preservation' (Disabled).

### 5.2.3. System - Twinning Tab

To view or change the System Twinning settings, navigate to the **Twinning** tab in the Details Pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. Click **OK** to commit (not shown).



## 5.2.4. System – Codecs Tab

- In the **Codecs** tab of the Details Pane, select or enter **101** for **RFC2833 Default Payload**. This setting was recommended by EarthLink for use with out-band DTMF tone transmissions.
- For codec selection, select the codecs and codec order of preference on the right, under the **Selected** column. The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis. The buttons between the two lists can be used to move codecs between the **Unused** and **Selected** lists, and to change the order of the codecs in the **Selected** codecs list. By default, all IP phones (SIP and H.323) will use the system default codec selection shown here, unless configured otherwise for a specific extension. The example below shows the codecs used for IP phones (SIP and H.323).

The screenshot displays the 'Codecs' configuration page for a system. The left sidebar shows a tree view with 'System (1)' selected. The main content area has tabs for 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', 'SMTP', 'SMDR', 'Twinning', 'VCM', 'CCR', 'Codecs', and 'ACCS'. The 'Codecs' tab is active, showing 'RFC2833 Default Payload' set to '101'. Below this, there are two columns: 'Available Codecs' and 'Default Codec Selection'. The 'Available Codecs' list includes G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ. The 'Default Codec Selection' area has an 'Unused' list with G.711 ALAW 64K, G.722 64K, and G.723.1 6K3 MP-MLQ. The 'Selected' list contains G.711 ULAW 64K and G.729(a) 8K CS-ACELP. Navigation buttons (right arrow, up arrow, down arrow, left arrow) are located between the 'Unused' and 'Selected' lists.

**Note:** The codec selections defined under this section (System – Codecs Tab) are the codecs selected for the IP phones/extensions. The codec selections defined under **Section 5.4.5** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

### 5.3. IP Route

Create an IP route to specify the IP address of the gateway or router where IP Office needs to send the packets in order to route calls to EarthLink's network.

Navigate to **IP Route** → **0.0.0.0** in the left Navigation Pane if a default route already exists, otherwise, to create the default route, right-click on **IP Route** and select **New**. Create/verify a default route with the following parameters:

- Set **IP Address** and **IP Mask** to **0.0.0.0**.
- Set **Gateway IP Address** to the IP address of the default router for the public network where IP Office is connected.
- Set **Destination** to **LAN2** from the drop-down list.
- Click the **OK** to commit (not shown).

The screenshot displays the IP Office configuration interface. On the left is the 'IP Offices' navigation pane, which is expanded to show 'IP Route (5)'. The '0.0.0.0' route is selected and highlighted with a red box. The main configuration area on the right is titled '0.0.0.0\*' and contains the following fields:

|                    |                                    |
|--------------------|------------------------------------|
| IP Address         | 0 . 0 . 0 . 0                      |
| IP Mask            | 0 . 0 . 0 . 0                      |
| Gateway IP Address | 192 . 168 . 157 . 129              |
| Destination        | LAN2                               |
| Metric             | 0                                  |
|                    | <input type="checkbox"/> Proxy ARP |

## 5.4. SIP Line

A SIP Line is needed to establish the SIP connection between IP Office and EarthLink SIP Trunking Services. 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 5.4.1** 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 trunk Registration Credentials.
- 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 **Sections 5.4.2 – 5.4.5**.

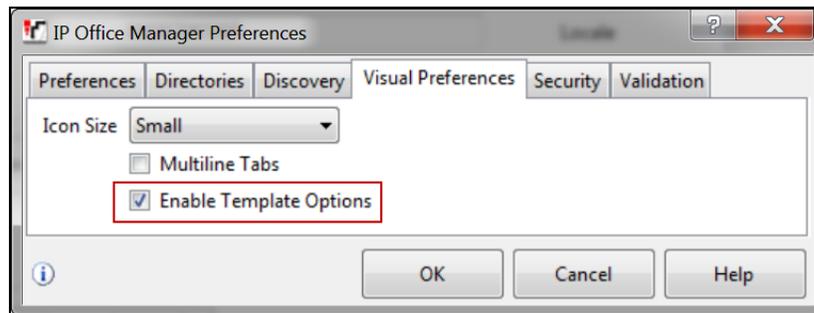
Also, the following SIP Line settings are not supported on Basic Edition:

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

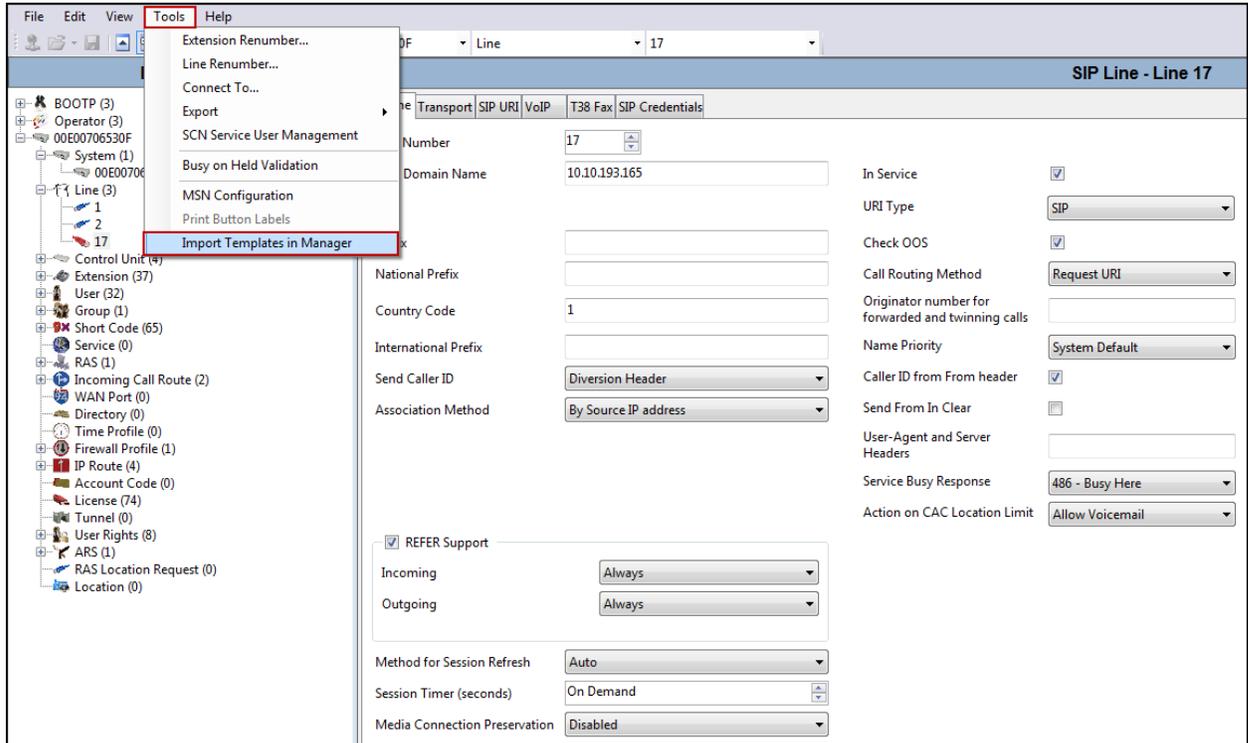
To create a SIP Line manually, right-click **Line** in the Navigation Pane and select **New → SIP Line**; then, follow the steps outlined in **Sections 5.4.2 – 5.4.5**.

### 5.4.1. SIP Line From Template

1. Copy the template file to the computer where Avaya IP Office Manager is installed. Rename the template file to **US\_EarthLink\_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.
2. Verify that template options are enabled in Avaya IP Office Manager. In Avaya IP Office Manager, navigate to **File → Preferences**. In the **Avaya IP Office Manager Preferences** window that appears, select the **Visual Preferences** tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

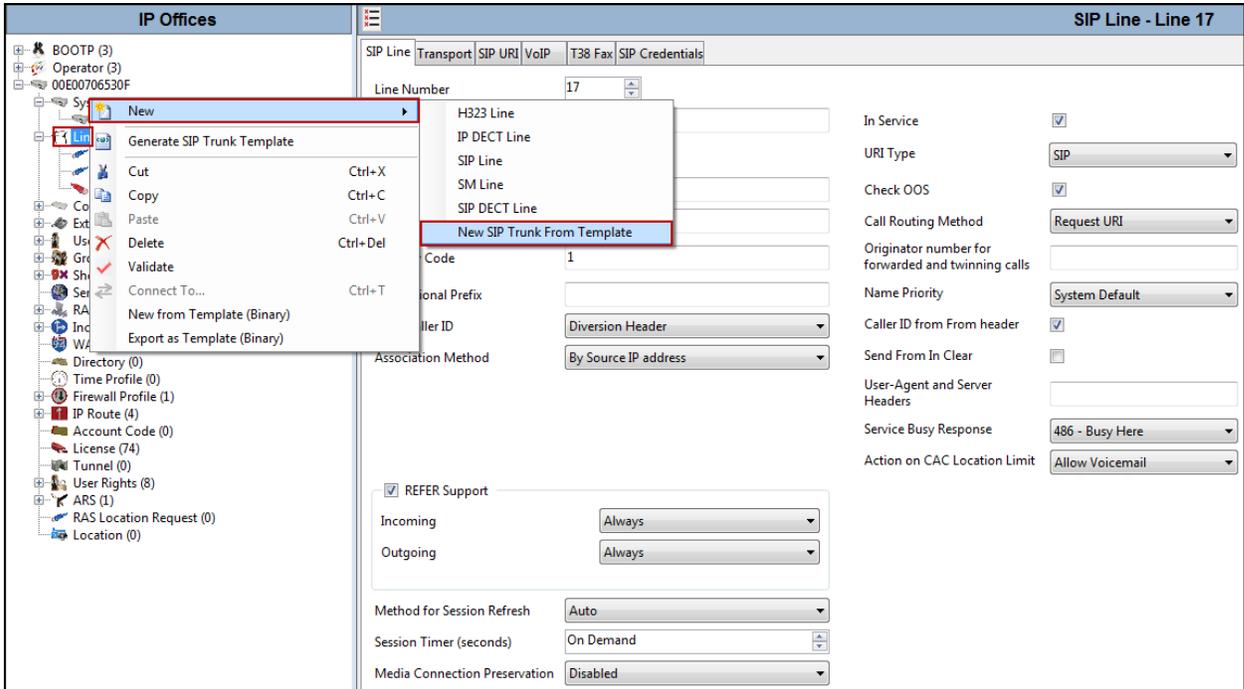


3. Import the template into Avaya IP Office Manager. From Avaya IP Office Manager, select **Tools** → **Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the Avaya IP Office Manager pull-down menus in **Step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



In the pop-up window that appears (not shown), select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure, click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

- To create the SIP trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New** → **New SIP Trunk From Template**.



5. In the subsequent **Template Type Selection** pop-up window, select **United States** from the **Country** pull-down menu and select **EarthLink** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**US\_EarthLink\_SIPTrunk.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



6. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2 – 5.4.5**.

## 5.4.2. SIP Line – SIP Line Tab

On the **SIP Line** tab in the Details Pane, configure or verify the parameters as shown below.

- Set **ITSP Domain Name** to **10.10.193.165**, the IP address of the Service Provider's SIP Proxy (Same IP address used in **Section 5.4.3** under ITSP Proxy Address).
- Set **Send Caller ID** to **Diversion Header**.
- Verify that **REFER Support** is checked (see **Section 2.2**). Set **Incoming** and **Outgoing** under REFER Support to **Always**.
- Verify that **In Service** box is checked, which is the default value. This makes the trunk available to incoming and outgoing calls.
- Verify that **Check OOS** box is checked, the default value. IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the Binding Refresh Time for LAN2, as shown in **Section 5.2.1**.
- Verify that **Call Routing Method** is set to **Request URI**, which is the default value.
- Verify that **Caller ID from From header** box is checked.

All other parameters should be set to default or according to customer requirements. Click **OK** to commit (not shown).

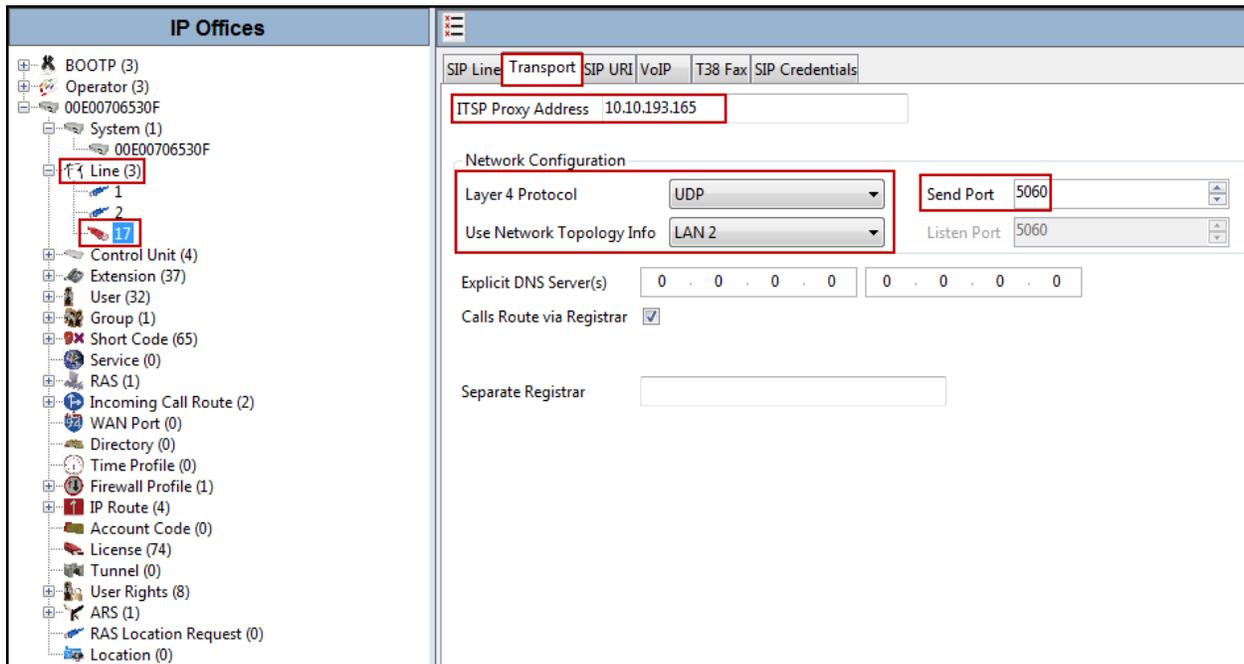
The screenshot displays the configuration for SIP Line - Line 17. The left pane shows a tree view of IP Offices, with 'Line (3)' selected. The right pane shows the configuration details for the selected line. The 'SIP Line' tab is active, and the 'Line Number' is 17. The configuration fields are as follows:

| Field  | Value                               |
|--|-------------------------------------|
| Line Number  | 17                                  |
| ITSP Domain Name                                   | 10.10.193.165                       |
| Prefix   |                                     |
| National Prefix                                    |                                     |
| Country Code                                       | 1                                   |
| International Prefix                               |                                     |
| Send Caller ID                                     | Diversion Header                    |
| Association Method                                 | By Source IP address                |
| In Service   | <input checked="" type="checkbox"/> |
| URI Type   | SIP                                 |
| Check OOS  | <input checked="" type="checkbox"/> |
| Call Routing Method                                | Request URI                         |
| Originator number for forwarded and twinning calls |                                     |
| Name Priority                                      | System Default                      |
| Caller ID from From header                         | <input checked="" type="checkbox"/> |
| Send From In Clear                                 | <input type="checkbox"/>            |
| User-Agent and Server Headers                      |                                     |
| Service Busy Response                              | 486 - Busy Here                     |
| Action on CAC Location Limit                       | Allow Voicemail                     |
| REFER Support                                      | <input checked="" type="checkbox"/> |
| Incoming   | Always                              |
| Outgoing   | Always                              |
| Method for Session Refresh                         | Auto                                |
| Session Timer (seconds)                            | On Demand                           |
| Media Connection Preservation                      | Disabled                            |

### 5.4.3. SIP Line - Transport Tab

Select the **Transport** tab. Set or verify the parameters as shown below.

- Set **ITSP Proxy Address** to **10.10.193.165**, the IP address of the Service Provider's SIP Proxy.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN2**, the network port used by the SIP Line to access the far-end and configured in **Section 5.2.1**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.
- Click **Ok** to commit (not shown).



#### 5.4.4. SIP Line - SIP URI Tab

A SIP URI entry needs to be created to match each incoming number that IP Office will accept on this line. Select the **SIP URI** tab, and then 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 example screen below, a previously configured entry was edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to **Use Internal Data**.
- Set **PAI** to **None**.
- Associate this line with an incoming line group by entering the 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.
- Click **OK**.
- Click **OK** again to commit (not shown).

The screenshot displays the IP Office configuration interface. On the left is a tree view of the system hierarchy, with 'Line (3)' expanded to show line 17. The main pane shows the 'SIP URI' tab with a table of channels. The 'Edit Channel' dialog is open, showing the following configuration:

| Channel | Groups | Via  | Local URI | Contact | Display Name | PAI  | Credential | Max Calls |
|---------|--------|------|-----------|---------|--------------|------|------------|-----------|
| 1       | 17 17  | 1... |           |         |              | N... | 0: <Non... | 10        |

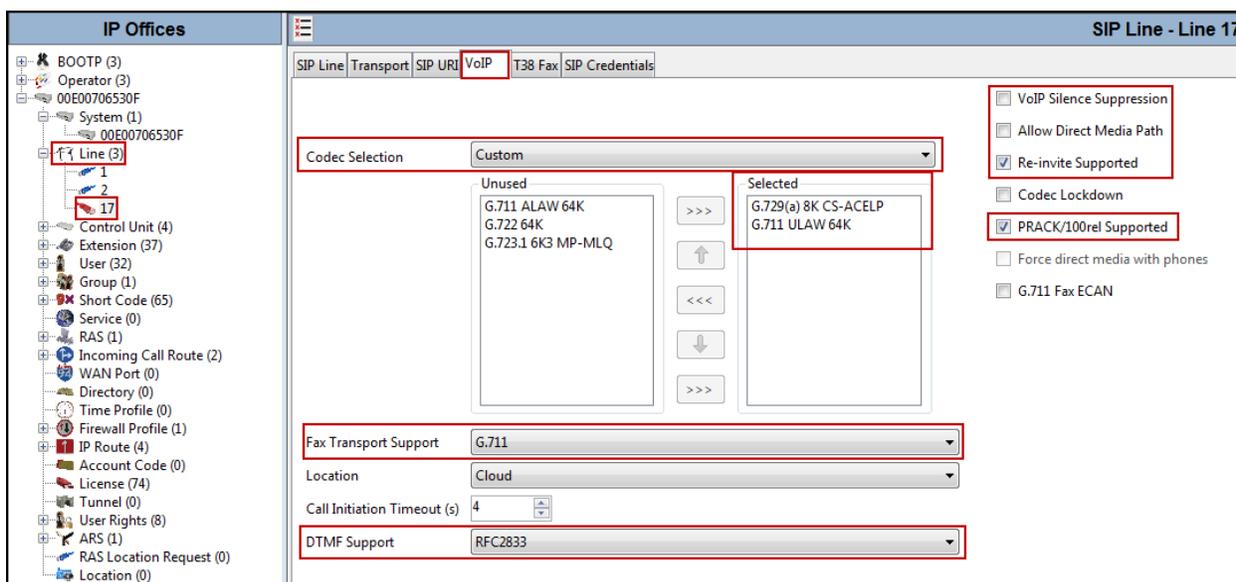
| Edit Channel          |                   |
|-----------------------|-------------------|
| Via                   | 192.168.157.189   |
| Local URI             | Use Internal Data |
| Contact               | Use Internal Data |
| Display Name          | Use Internal Data |
| PAI                   | None              |
| Registration          | 0: <None>         |
| Incoming Group        | 17                |
| Outgoing Group        | 17                |
| Max Calls per Channel | 10                |

Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user, such as a short code. These URIs are created in the same manner as shown above with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, and **Display Name** fields.

### 5.4.5. SIP Line - VoIP Tab

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

- Set the **Codec Selection** to **Custom**.
- Select **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** from the **Unused** box and move these two selections to the **Selected** box. Use the up and down arrows in the middle to order these 2 codes. The G.729A and G.711MU codecs are supported by EarthLink; codecs are shown in preferred order (top to bottom). G.729A was configured as the preferred codec for the compliance test as shown below.
- Select **G.711** for **Fax Transport Support**.
- Set the **DTMF Support** field to **RFC2833**. This directs IP Office to send DTMF tones as out-band RTP events as per RFC2833.
- Uncheck the **VoIP Silence Suppression** option box.
- Verify that **Allow Direct Media Path** is unchecked.
- Check the **Re-invite Supported** option box.
- Check the **PRACK/100rel Supported** option box. This setting enables support by IP Office for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.
- Click the **OK** to commit (not shown).



**Note:** The codec selections defined under this section (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk). The codec selections defined under **Section 5.2.4** (System – Codec tab) are the codecs selected for the IP phones/extension (H.323 and SIP).

## 5.5. Outbound Call Routing

For outbound call routing, a combination of system short codes and Automatic Route Selection (ARS) entries are used. With ARS, features like time-based routing criteria 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. While detailed coverage of ARS is beyond the scope of these Application Notes, and alternate routing was not used in the reference configuration, this section includes some basic screen illustrations of the ARS settings used during the compliance testing.

### 5.5.1. Short Codes and Automatic Route Selection

To create a short code to be used for ARS, right-click on **Short Code** on the Navigation Pane and select **New**. The screen below shows the short code **9N** created (note that the semi-colon is not used here). In this case, when the IP Office user dials 9 plus any number **N**, instead of being directed to a specific Line Group ID, the call is directed to **Line Group 50: Main**, which is configurable via ARS.

- In the **Code** field, enter the dial string which will trigger this short code. In this case, **9N** was used (note that the semi-colon is not used here).
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N**. The value **N** represents the number dialed by the user after removing the **9** prefix. This value is passed to ARS.
- Set the **Line Group ID** to **50: Main** to be directed to **Line Group 50: Main**, which is configurable via ARS.
- Click the **OK** to commit (not shown).

The screenshot displays the IP Office configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'Short Code (65)' highlighted. The main window shows the configuration for a short code named '9N: Dial'. The configuration fields are as follows:

| Field            | Value    |
|------------------|----------|
| Code             | 9N       |
| Feature          | Dial     |
| Telephone Number | N        |
| Line Group ID    | 50: Main |
| Locale           |          |

At the bottom of the configuration pane, there is a checkbox labeled 'Force Account Code' which is currently unchecked.

The following screen shows a sample ARS configuration for the route **Main**. Note the sequence of **X**'s used in the **Code** column of the entries to specify the exact number of digits to be expected, following the access code and the first set of digits on the string. This type of setting results in a much quicker response in the delivery of the call by IP Office.

To create a short code to be used for ARS, select **ARS → 50: Main** on the Navigation Pane and click **Add**.

- In the **Code** field, enter the dial string which will trigger this short code. In this case, **1** followed by **10 X**'s to represent the exact number of digits.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **1N**. The value **N** represents the additional number of digits dialed by the user after dialing **1** (The **9** will be stripped off).
- Set the **Line Group ID** to the Line Group number being used for the SIP Line, in this case **Line Group ID 17** was used.
- Set **Locale** to **United States (US English)**.
- Click **OK** to commit.

| Field              | Value                      |
|--------------------|----------------------------|
| Code               | 1XXXXXXXXXX                |
| Feature            | Dial                       |
| Telephone Number   | 1N                         |
| Line Group ID      | 17                         |
| Locale             | United States (US English) |
| Force Account Code | <input type="checkbox"/>   |

Repeat the above procedure for additional dial patterns to be used by the enterprise to dial out from IP Office.

The example highlighted below shows that for calls in the North American numbering plan, the user dialed **9**, followed by **1** and **10** digits (represented by **10 X**'s). The **9** is stripped off, the remaining digits, including the **1**, are included in the SIP INVITE message IP Office sends to EarthLink.

The screenshot displays the Avaya IP Office Manager interface for configuring an ARS (Automatic Route Selection) entry. The left-hand pane shows a tree view of the system configuration, with 'ARS (1)' and its sub-entry '50: Main' highlighted. The main configuration area for 'ARS' includes the following settings:

- ARS Route Id: 50
- Route Name: Main
- Dial Delay Time: System Default (3)
- Secondary Dial tone:  (checked)
- SystemTone: SystemTone
- Check User Call Barring:  (checked)
- In Service:  (checked)
- Out of Service Route: <None>
- Time Profile: <None>
- Out of Hours Route: <None>

The ARS entries table is as follows:

| Code          | Telephone Number | Feature        | Line Group ID |
|---------------|------------------|----------------|---------------|
| 911           | 911              | Dial Emergency | 0             |
| 001XXXXXXXXXX | 001N             | Dial           | 17            |
| 8XXXXXXXXXX   | 8N               | Dial           | 17            |
| 1XXXXXXXXXX   | 1N               | Dial           | 17            |
| 6XXXXXX       | 6N               | Dial           | 17            |
| 3XXXXXXXXXX   | 3N               | Dial           | 17            |
| 28XXXXXX      | 28N              | Dial           | 17            |

Below the table, the configuration continues with:

- Alternate Route Priority Level: 3
- Alternate Route Wait Time: 30
- Alternate Route: <None>

## 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 navigate to **User** → *Name* in the Navigation Pane where *Name* is the name of the user to be modified. In the example below, the name of the user is **Ext3040 H323**. Select the **SIP** tab in the Details Pane (tab not shown). The values entered for the **SIP Name** 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 5.4.4**). The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise 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. This can also be accomplished by activating Withhold Number on H.323 Deskphones (not shown). Click the **OK** to commit (not shown).

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane, and on the right is the configuration details pane for a user.

**IP Offices Navigation Pane:**

- BOOTP (3)
- Operator (3)
- 00E00706530F
- System (1)
- Line (3)
- Control Unit (4)
- Extension (37)
- User (32) [highlighted]

  - NoUser
  - RemoteManager
  - 3040 Ext3040 H323 [highlighted]
  - 3041 Ext3041 H323
  - 3042 Ext3042 H323

**User Configuration Details Pane:**

| User                     | Voicemail | DND | Short Codes  | Source Numbers | Telephony | Forwarding |
|--------------------------|-----------|-----|--------------|----------------|-----------|------------|
| SIP Name                 |           |     | 5085211377   |                |           |            |
| SIP Display Name (Alias) |           |     | Ext3040 H323 |                |           |            |
| Contact                  |           |     | 5085211377   |                |           |            |

Anonymous

## 5.7. Incoming Call Route

An incoming call route maps inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc., within the IP Office system.

In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP Line to reach any predefined extension in IP Office. The routing decision for the call is based on the parameters previously configured for Call Routing Method and SIP URI (**Section 5.4.4**) and the users SIP Name and Contact, already populated with the assigned EarthLink DID numbers (**Section 5.6**).

### 5.7.1. Incoming Call Route – Standard Tab

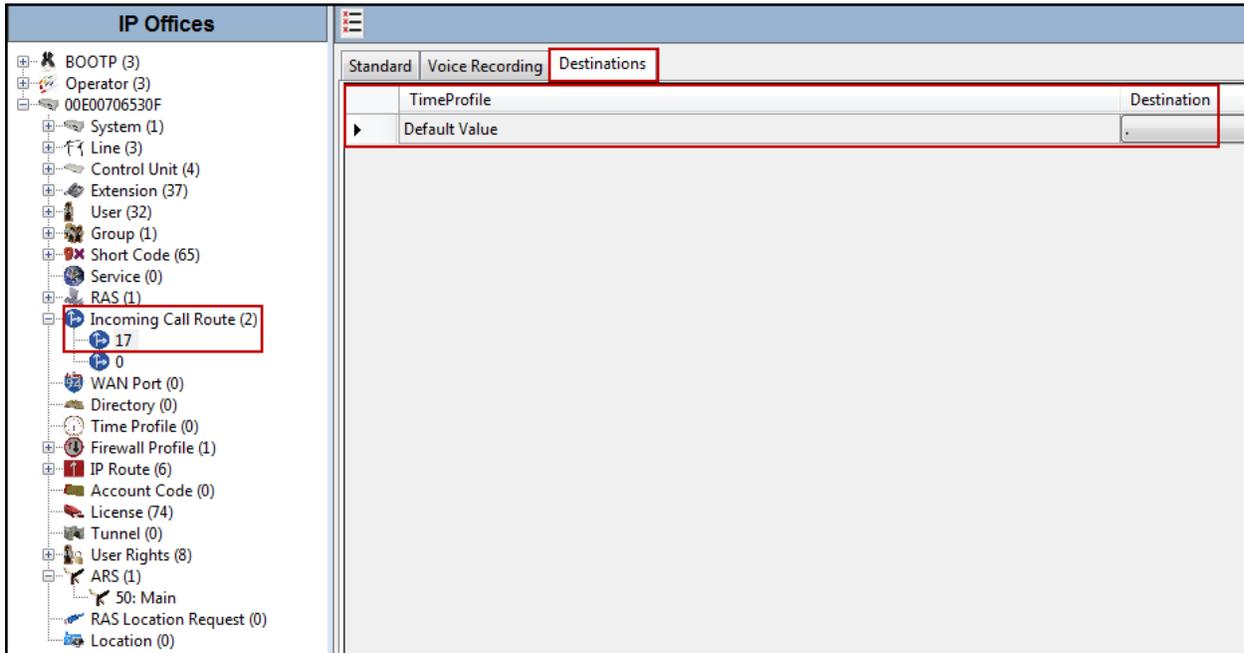
On the **Standard** tab of the Details pane, enter the parameters as shown below.

- Set **Bearer Capacity** to **Any Voice**.
- Set the **Line Group ID** to the incoming line group of the SIP Line defined in **Section 5.4**, in this case **Line Group ID 17** was used.
- Default values can be used for all other fields.
- Click **Ok** to commit (not shown).

The screenshot displays the IP Office configuration interface. On the left, a tree view under 'IP Offices' shows the hierarchy: BOOTP (3), Operator (3), 00E00706530F, System (1), Line (3), Control Unit (4), Extension (37), User (32), Group (1), Short Code (65), Service (0), RAS (1), Incoming Call Route (2), and 17. The 'Incoming Call Route (2)' folder is expanded, and the '17' sub-item is selected. On the right, the 'Standard' tab of the configuration pane is active. The 'Bearer Capacity' dropdown is set to 'Any Voice', and the 'Line Group ID' dropdown is set to '17'. Other fields include 'Incoming Number', 'Incoming Sub Address', 'Incoming CLI', 'Locale', 'Priority' (set to '1 - Low'), 'Tag', 'Hold Music Source' (set to 'System Source'), and 'Ring Tone Override' (set to 'None').

### 5.7.2. Incoming Call Route – Destinations Tab

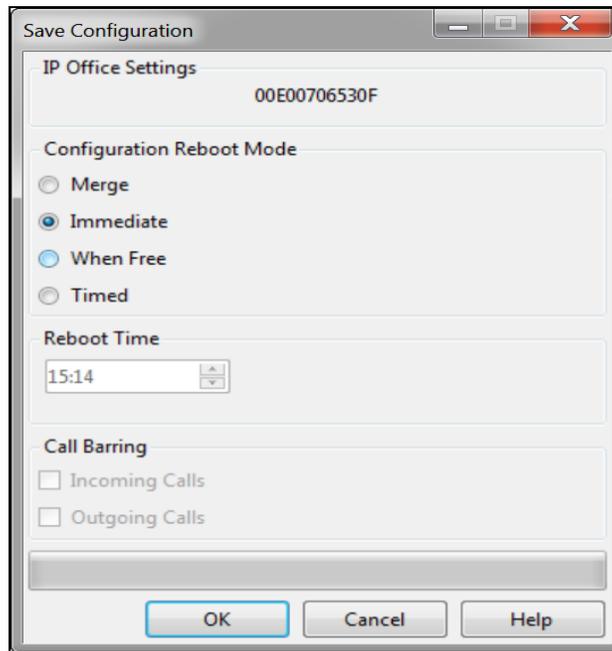
Under the **Destinations** tab, enter “.” for the **Default Value**. This setting will allow the call to be routed to any destination with a value of the **SIP Name** field, defined on the **SIP** tab of the **User**, which matches the number present on the user part of the “To” header on the incoming INVITE message received from EarthLink. Click **OK** to commit (not shown).



## 5.8. 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.

The following will appear, with either **Merge** or **Immediate** selected, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** to proceed.



## 6. EarthLink SIP Trunking Services Configuration

To use EarthLink's SIP Trunking Services, a customer must request the service from EarthLink using the established sales processes. The process can be started by contacting EarthLink via the corporate web site at: <http://www.earthlinkbusiness.com/> and requesting information.

During the signup process, EarthLink and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to EarthLink's network.

EarthLink is responsible for the configuration of EarthLink SIP Trunking Services. The customer will need to provide the public IP address used to reach IP Office at the enterprise. In the case of the compliance test, this is the public IP address of the IP Office WAN port (LAN2).

EarthLink will provide the customer the necessary information to configure IP Office following the steps discussed in the previous sections, including:

- EarthLink's SIP Proxy IP address.
- DID numbers.
- Etc.

## 7. Verification Steps

This section provides verification steps that may be performed to verify that the solution is configured properly.

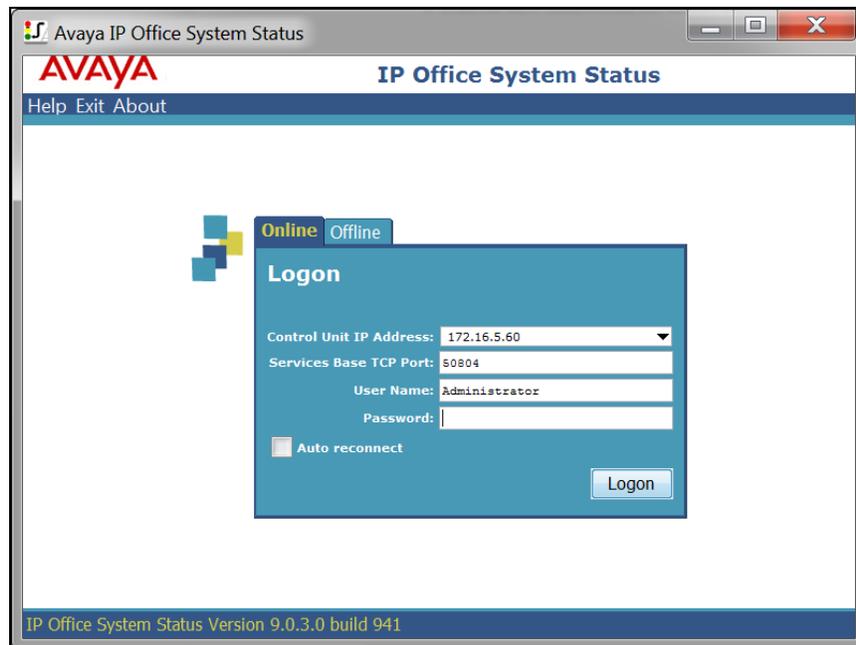
The following steps may be used to verify the configuration:

- Verify that endpoints at the enterprise site can place calls to the PSTN.
- Verify that endpoints at the enterprise site can receive calls from the PSTN.
- Verify that users at the PSTN can end active calls to endpoints at the enterprise by hanging up.
- Verify that endpoints at the enterprise can end active calls to PSTN users by hanging up.

### 7.1. Avaya IP Office System Status

The following steps can also be used to verify the configuration.

Use the Avaya IP Office System Status application to verify the state of SIP connections. Launch the application from **Start** → **Programs** → **IP Office** → **System Status** on the PC where Avaya IP Office System Status is installed, log in with the proper credentials.



Select the SIP Line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current State** is **Idle** for each channel.

The screenshot shows the Avaya IP Office System Status interface. The left navigation pane has 'Trunks (3)' selected, with 'Line: 17' highlighted. The main area has the 'Status' tab selected, displaying the 'SIP Trunk Summary' for Line 17. The summary includes details like Peer Domain Name, Resolved Address, Line Number, and a green progress indicator for SIP Trunk Channel Licenses at 0%.

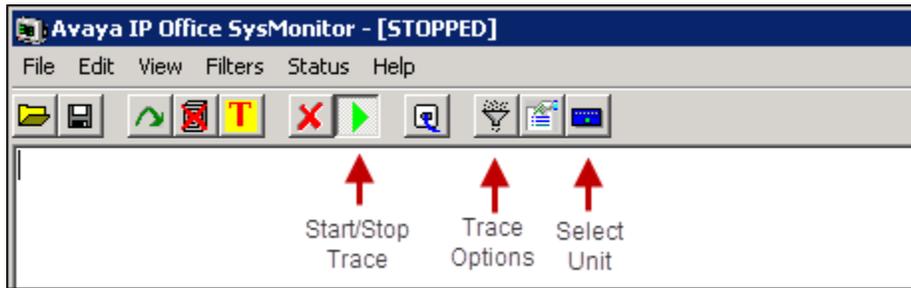
| Channel Number | URI Group | Call Ref | Current State | Time in State | Remote Media Address | Codec | Connection Type | Caller ID or Dialed Digits | Other Party on Call |
|----------------|-----------|----------|---------------|---------------|----------------------|-------|-----------------|----------------------------|---------------------|
| 1              |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |
| 2              |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |
| 3              |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |
| 4              |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |
| 5              |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |
| 6              |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |
| 7              |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |
| 8              |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |
| 9              |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |
| 10             |           |          | Idle          | 00:06:03      |                      |       |                 |                            |                     |

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

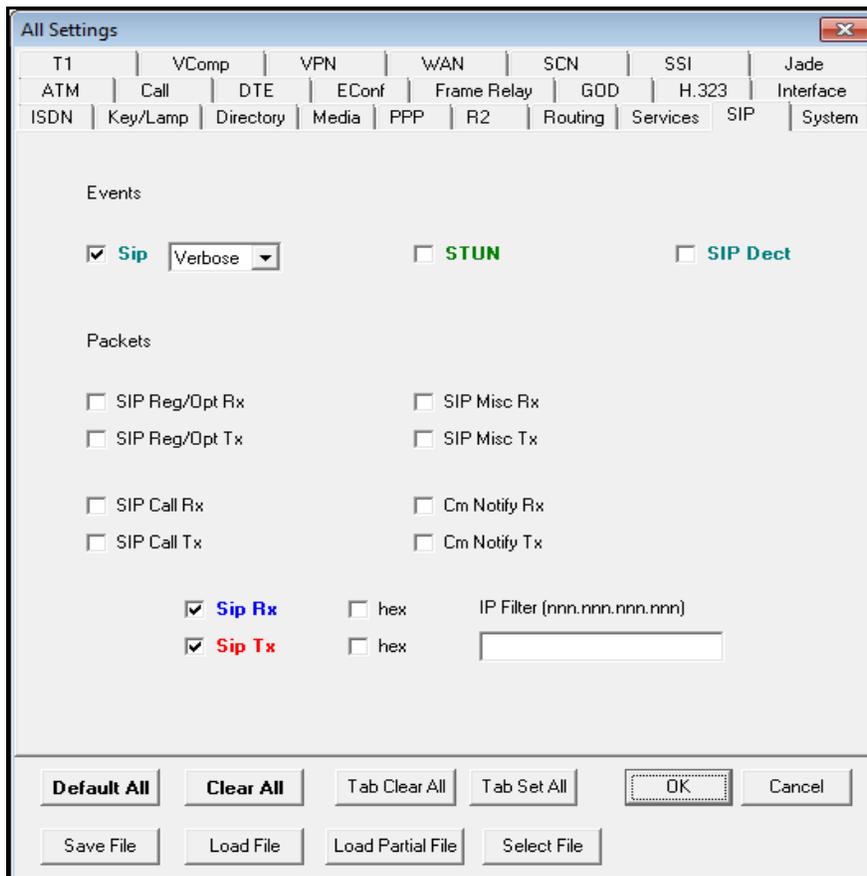
The screenshot shows the Avaya IP Office System Status interface with the 'Alarms' tab selected. The header indicates 'Alarms for Line: 17 SIP [redacted].193.165'. Below this, there are columns for 'Last Date Of Error', 'Occurrences', and 'Error Description', but no alarm entries are present.

## 7.2. Avaya IP Office Sys Monitor

The Avaya IP Office Sys Monitor application can be used to monitor and troubleshoot signaling messaging on the SIP trunk. Launch the application from **Start → Programs → IP Office → Monitor** on the PC where Avaya IP Office Sys Monitor was installed. Click the **Select Unit** icon on the taskbar and Select the IP address of the IP Office system under verification.



Clicking the **Trace Options** icon on the taskbar, selecting the **SIP** tab allows modifying the threshold used for capturing events, types of packets to be captured, filters, etc. Additionally, the color used to represent the packets in the trace can be customized by right clicking on the type of packet and selecting the desired color.



## 8. Conclusion

These Application Notes describe the configuration steps necessary for configuring Session Initiation Protocol (SIP) Trunk Service for an enterprise solution using Avaya IP Office Release 9.0 to interoperate with EarthLink SIP Trunking Services. EarthLink SIP Trunking Services is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks.

EarthLink SIP Trunking Services passed compliance testing with the observations/limitations outlined in the scope of testing in **Section 2.1** as well as under test results in **Section 2.2**

## 9. References

This section references the documentation relevant to these Application Notes. Product documentation for Avaya IP Office, including the following, is available at:

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

- [1] *IP Office 9.0.3 IP500/IP500 V2 Installation*, Document Number 15-601042, Issue 29g, October 07, 2014.
- [2] *IP Office Manager Release 9.0.3*, Document Number 15-601011, Issue 9.0.3, May 8, 2014.
- [3] *Using System Status*, Document Number 15-601758, Issue 09c, August 15, 2013.
- [4] *IP Office Release 9.0.3 Administering Voicemail Pro*, Document Number 15-601063, Issue 9.0d, July 09, 2014.
- [5] *Using IP Office System Monitor*, Document Number 15-601019, Issue 05f, October 07, 2014.

Additional Avaya IP Office documentation can be found at:

<http://marketingtools.avaya.com/knowledgebase/>

---

**©2014 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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](mailto:devconnect@avaya.com).