



Avaya Solution & Interoperability Test Lab

Application Notes for Charter Business SIP Trunking Service with Avaya IP Office Release 9.1 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking on an enterprise solution consisting of Avaya IP Office 9.1, to interoperate with Charter Business SIP Trunking Service.

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.

Charter Business SIP Trunking Service provides PSTN access via a SIP Trunk between the enterprise and Charter's network as an alternative to legacy analog or ISDN-PRI 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.

Charter 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) Trunking service between Charter and an Avaya SIP-enabled enterprise solution.

In the configuration used during the testing, the Avaya SIP-enabled enterprise solution consisted of Avaya IP Office 500v2 Release 9.1 (hereafter referred to as IP Office), Avaya Communicator for Windows and Avaya Deskphones, including SIP, H.323, digital, and analog.

As a required component of the Charter Business SIP Trunking service offering, Charter will install a Modular Access Router at the customer premises (enterprise site). Charter will perform the initial configuration and maintenance as required. The Modular Access Router will be considered Customer Premises Equipment (CPE).

The Charter Business SIP Trunking Service referenced within these Application Notes is designed for business customers. Customers using this service with the Avaya 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” or “Charter” will be used interchangeably throughout these Application Notes.

2. General Test Approach and Test Results

The general test approach was to simulate an enterprise site in the Solution & Interoperability Test Lab by connecting IP Office to the Charter Business SIP Trunking service via the public Internet, as depicted in **Figure 1**.

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 the Charter Business SIP Trunking service offering with Avaya IP Office, the following features and functionalities were exercised during the compliance testing:

- Response to SIP OPTIONS queries.
- 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 networks.
- Incoming and outgoing PSTN calls to/from Avaya Communicator for Windows.
- Dialing plans including local, long distance, outbound toll-free and international.
- Caller ID presentation and Caller ID restriction.
- 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 coverage to voicemail off).
- Proper response to busy endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Codec G.711MU (Charter supported audio codec).
- Proper response to no matching codecs.
- G.711 Fax Pass-through.
- Proper early media transmissions.
- Voicemail and DTMF tone support using RFC 2833 (leaving and retrieving voice mail messages, etc.).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response systems).
- 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:

- The use of the SIP REFER method for network call redirection is not currently supported by Charter.
- Inbound toll-free calls and 911 emergency calls are supported but were not tested as part of the compliance test.
- T.38 fax is not supported by Charter; therefore, T.38 fax was not tested. G.711 Fax Pass-through was tested successfully and it's recommended instead.

2.2 Test Results

Interoperability testing with Charter Business SIP Trunking service was successfully completed with the exception of observations/limitations described below:

- **No matching codec on outbound calls:** If an unsupported audio codec is received by Charter on the SIP Trunk (e.g., 722), Charter will respond with “480 Temporarily Unavailable” instead of “488 Not Acceptable Here”, the user will hear re-order tone. This issue does not have any user impact, and should not be seen since the codecs will be matched during the installation, it is listed here simply as an observation.
- **Inbound calls to an unassigned enterprise extension:** IP Office sends a “404 Not Found” message to Charter when it receives calls to an unassigned extension, the user hears re-order tone instead of the common announcement informing the user that he/she has reached a non-working number, to please check the number and to try again. This issue is considered non-service affecting and it’s being investigated by Charter, in order to apply the correct announcement to the user.

2.3 Support

For support on Charter Business SIP Trunking service visit the corporate Web page at: <https://www.charterbusiness.com/> or call 800-314-7195

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. The test configuration simulates an enterprise site with an Avaya SIP-enabled enterprise solution connected to the Charter Business SIP Trunking service through the public Internet.

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

- Avaya IP Office 500v2.
- Avaya IP Office Voicemail Pro.
- Avaya 96x0 Series H.323 IP Deskphones.
- Avaya 96x1 Series H.323 IP Deskphones.
- Avaya 1100 Series SIP IP Deskphones.
- Avaya Communicator for Windows.
- Avaya 1408 Digital Telephones.
- Avaya 9508 Digital Telephones.

In the reference configuration, a Modular Access Router was required at the simulated enterprise site, acting as a SIP interface between the Avaya enterprise (IP Office) and Charter’s network. Charter will install the Modular Access Router at the customer premises (enterprise site). Charter will perform the initial configuration and maintenance as required. The Modular Access Router will be considered Customer Premises Equipment (CPE).

Also located at the simulated enterprise site is Avaya IP Office 500v2 with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. The IP Office **LAN1** port connects to the inside interface (or private side) of Charter's Modular Access Router.

The transport protocol between IP Office and Charter's Modular Access Router, across the enterprise private IP network, is SIP over UDP.

For inbound calls, the calls flowed from Charter's network, across the public Internet, to Charter's Modular Access Router, then to IP Office.

Outbound calls to the PSTN were first processed by IP Office. Once IP Office selected the proper SIP trunk; the call was routed to Charter's Modular Access Router, across the public Internet, then to Charter's network.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to Charter (refer to **Section 5.8**). The short code 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to the network. Since Charter is a U.S. based company, a country member of the North American Numbering Plan (NANP), the users dialed 7 or 10 digits for local calls, and 11 (1 + 10) digits for other calls between the NANP.

In an actual customer configuration, the enterprise site may also include additional network components between Charter and the enterprise. 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 enterprise must be allowed to pass through these devices.

For confidentiality and privacy purposes, actual public IP addresses and DID numbers used during the compliance test have been replaced with fictitious IP addresses and DID numbers throughout the Application Notes.

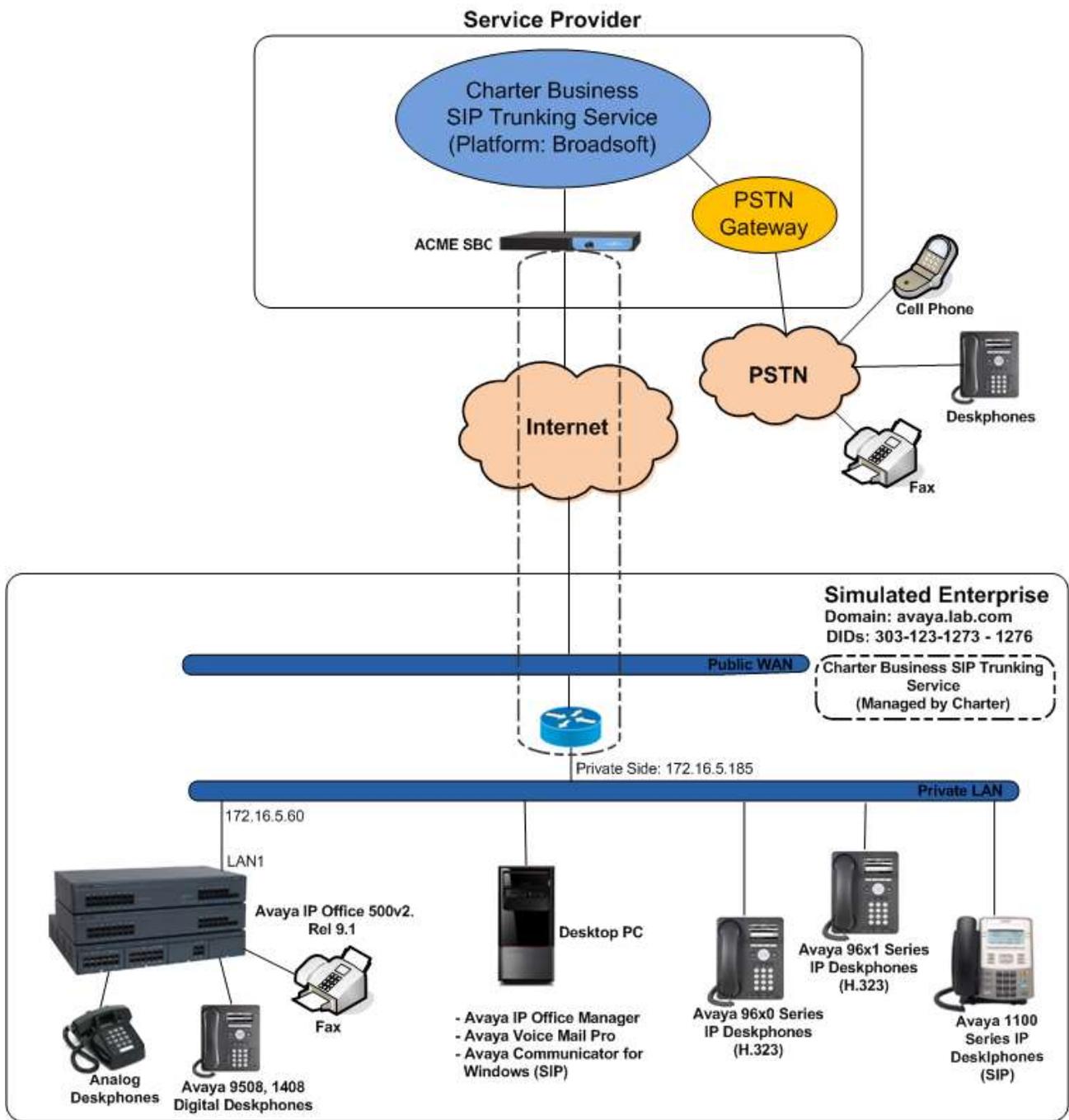


Figure 1: Avaya Interoperability Test Lab Configuration.

4. Equipment and Software Validated

The following equipment and software/firmware were used for the compliance testing.

Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500v2	9.1.4.0 Build 137
Avaya IP Office DIG DCPx16 V2	9.1.4.0 Build 137
Avaya IP Office Manager	9.1.4.0 Build 137
Avaya Voicemail Pro Client	9.1.4.0 Build 7
Avaya 96x0 IP Deskphones (H.323)	Avaya one-X® Deskphone Edition S3.230A
Avaya 96x1 Series IP Deskphones (H.323)	Avaya one-X® Deskphone H.323 Version 6.6029
Avaya 1120E IP Deskphones (SIP)	SIP1120e Ver. 04.04.18.00
Avaya Communicator for Windows	2.0.3.33
Avaya Digital Deskphones 1408	40.0
Avaya Digital Deskphones 9508	0.55
Lucent Analog Phone	--
Charter	
Broadworks Broadsoft Application Server	AS_Rel_17.sp4_1.197
ACME Packet 4500 Series SBC	SCX6.2.0 MR-9 GA (Build 1014)
Adtran NetVanta 3430 Modular Access Router	R10.3.0.V

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500v2 and also when deployed with all configurations of IP Office Server Edition without T.38 Fax Service.

5. Configure IP Office

This section describes the IP Office configuration required to interwork with Charter Business SIP Trunking service. IP Office is configured through Avaya IP Office Manager (IP Office Manager) which is a PC application. On the PC, select **Start → Programs → IP Office → Manager** to launch IP Office Manager. Navigate to **File → Open Configuration**, select the proper IP Office from the pop-up window, and log in with the appropriate credentials. A management window will appear as shown in the next sections. The appearance of IP Office Manager can be customized using the **View** menu (not shown). In the screenshots presented in this section, the **View** menu was configured to show the **Navigation Pane** on the left side and the **Details Pane** on the right side. These panes will be referenced throughout these Application Notes.

These Application Notes assume the basic installation and configuration of IP Office have already been completed and are not discussed here. For further information on IP Office, please consult References in **Section 9**.

5.1 Licensing

The configuration and features described in these Application Notes require the 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; click **License** in 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. Note that the full License Keys in the screen below is not shown for security purposes.

The screenshot displays the Avaya IP Office Manager interface. On the left is the 'IP Offices' navigation pane with a tree view. The 'License' option is highlighted in blue. On the right is the 'Details Pane' showing the configuration for the selected license. The 'License Mode' is 'License Normal' and the 'PLDS Host ID' is '111309813681'. Below this is a table of features and their license details.

Feature	License Key	Instances	Status	Expiry Date	Source
Report Viewer	Tvct73mdgdG...	255	Valid	Never	ADI Nodal
Mobility Features	0ICluRgHvKOx...	255	Obsolete	Never	ADI Nodal
Advanced Small Community Netw...	DaQJl7VeSvUU...	255	Obsolete	Never	ADI Nodal
IP500 Voice Networking Channels	T39BkqBXvd6...	255	Valid	Never	ADI Nodal
IP500 Upgrade Standard to Profess...	QaHgn76v9j6...	255	Obsolete	Never	ADI Nodal
IP500 Voice Networking Channels	JaHLHAVFXjD...	4	Valid	Never	ADI Nodal
SIP Trunk Channels	IBCQzGBYDU...	255	Valid	Never	ADI Nodal
VPN IP Extensions	@qm3fOoR5S...	255	Obsolete	Never	ADI Nodal
IP500 Universal PRI (Additional cha...	2TXC@OoNQ...	255	Valid	Never	ADI Nodal
RAS LRQ Support (Rapid Response)	hXIRxBVCEKN...	255	Valid	Never	ADI Nodal
IP Office Dealer Support - Standar...	4AOGBV5D9D...	255	Valid	Never	ADI Nodal
IP Office Dealer Support - Professi...	dlyY_Dba5Uq7...	255	Valid	Never	ADI Nodal
IP Office Distributor Support - Stan...	dv956B9iXS_N...	255	Valid	Never	ADI Nodal
IP Office Distributor Support - Prof...	LlHFzqB6XleQ...	255	Valid	Never	ADI Nodal
UMS Web Services	pGcSuPdLASj...	255	Valid	Never	ADI Nodal
Customer Service Agent	jD0xhEoAADH0...	255	Valid	Never	ADI Nodal

5.2 System

Configure the necessary system settings. In an Avaya IP Office, the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side). For the compliance test, the **LAN1** interface was used to connect Avaya IP Office to the enterprise private network (LAN), **LAN2** was not used.

5.2.1 System - LAN1 Tab

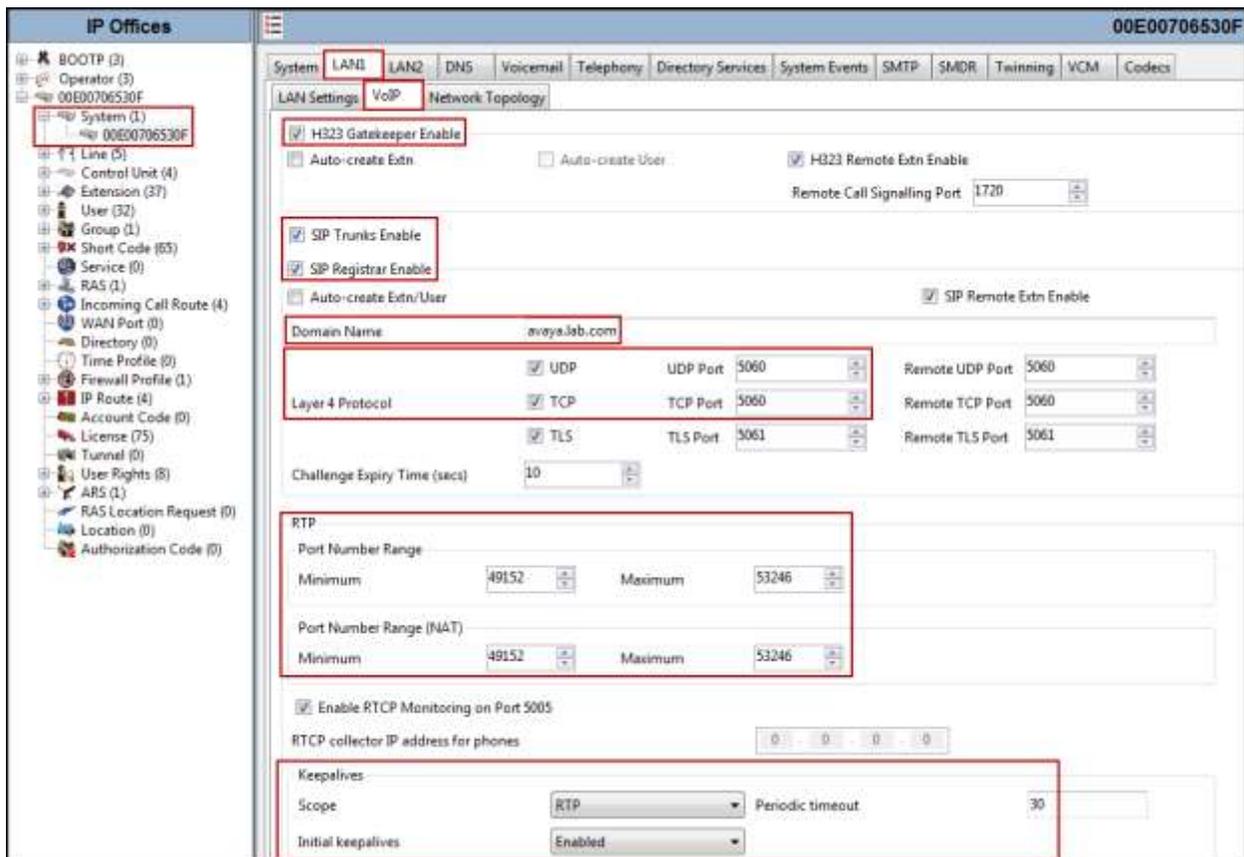
In the sample configuration, the MAC address **00E00706530F** was used as the system name. The **LAN** port connects to Charter's Modular Access Router, across the enterprise LAN (private) network. The **LAN1** settings correspond to the **LAN** port in IP Office. To access the **LAN1** settings, navigate to **System (1) → 00E00706530F** in the Navigation Pane then in the Details Pane navigate to the **LAN1 → LAN Settings** tab. The **LAN1** settings for the compliance testing were configured with following parameters:

- Set the **IP Address** field to the LAN IP address, e.g. **172.16.5.60**.
- Set the **IP Mask** field to the subnet mask of the public network, e.g. **255.255.255.0**.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit (not shown).

The screenshot displays the Avaya IP Office configuration interface. On the left is the 'IP Offices' navigation pane, showing a tree structure of system components. The 'System (1)' folder is expanded, and the system '00E00706530F' is selected. On the right is the configuration details pane, with the 'LAN1' tab selected. The 'LAN Settings' sub-tab is active. The 'IP Address' field is set to '172 . 16 . 5 . 60' and the 'IP Mask' field is set to '255 . 255 . 255 . 0'. Other settings include 'Primary Trans. IP Address' (0 . 0 . 0 . 0), 'RIP Mode' (None), 'Enable NAT' (unchecked), 'Number Of DHCP IP Addresses' (200), and 'DHCP Mode' (Disabled). An 'Advanced' button is visible at the bottom right of the settings pane.

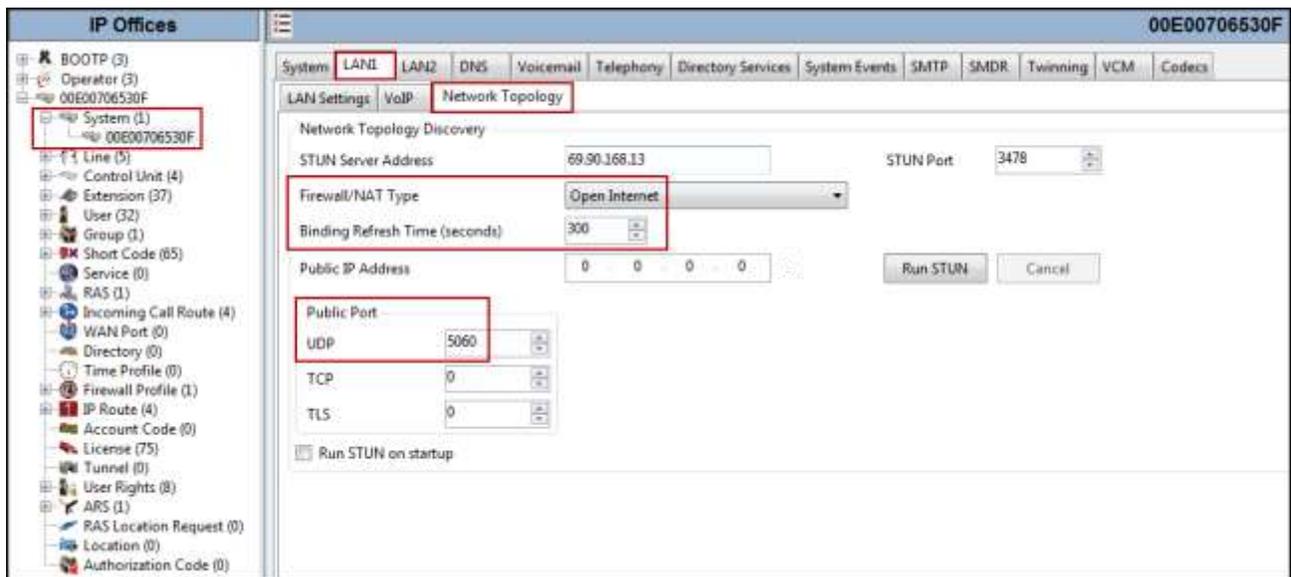
The **VoIP** tab as shown in the screenshot below was configured with following settings:

- Check the **H323 Gatekeeper Enable** to allow Avaya IP Telephones/Softphone using the H.323 protocol to register.
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to Charter.
- Check the **SIP Registrar Enable** to allow Avaya IP Telephones/Softphone to register using the SIP protocol.
- Enter the Domain Name of the enterprise under **Domain Name**.
- Verify the **UDP Port** and **TCP Port** numbers under **Layer 4 Protocol** are set to **5060**.
- Verify the **RTP Port Number Range** settings for a specific range for the RTP traffic. The **Port Range (Minimum)** and **Port Range (Maximum)** values were kept as default.
- In the **Keepalives** section at the bottom of the page, set the **Scope** field to **RTP**, **Periodic Timeout to 30**, and **Initial keepalives to Enabled**. This will cause the IP Office to send RTP keepalive packets at the beginning of the calls and every 30 seconds thereafter if no other RTP traffic is present.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit (not shown).



In the **Network Topology** tab, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu to the option that matches the network configuration. In the compliance testing, it was set to **Open Internet**. With this configuration, even though the default STUN settings are populated, they will not be used.
- 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.
- Verify the **Public IP Address** is set to **0.0.0.0**.
- Set the **Public Port** to **5060 for UDP**.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit (not shown).



5.2.2 System - Telephony Tab

Navigate to the **Telephony** → **Telephony** Tab in the Details Pane, configure the following parameters:

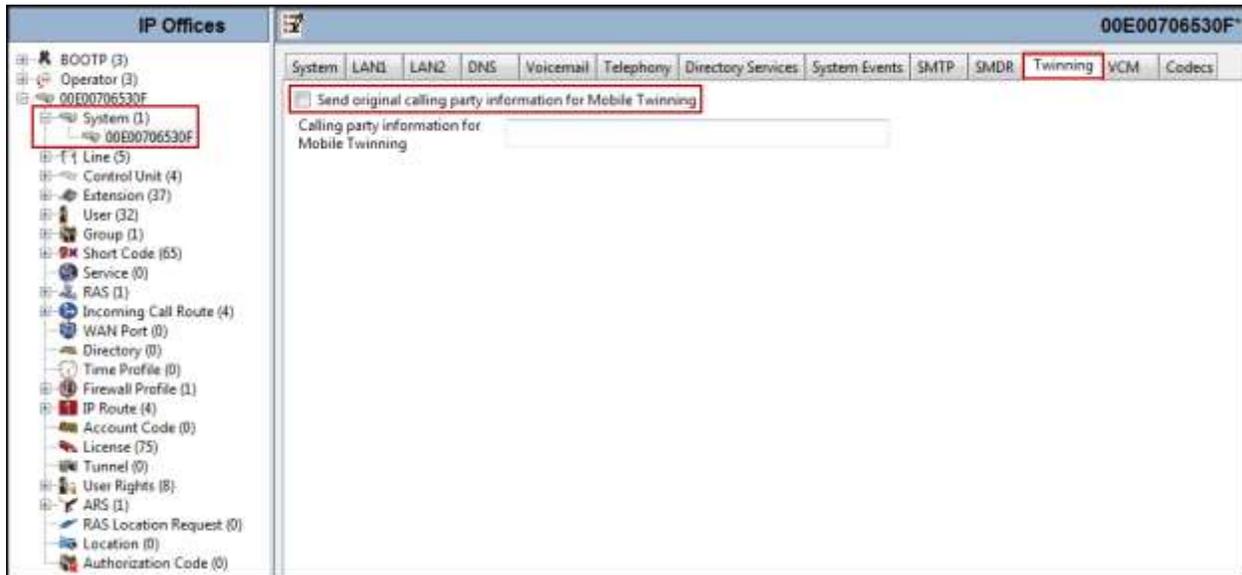
- Choose the **Companding Law** typical for the enterprise location, **U-Law** was used.
- Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the SIP trunk to the service provider.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit (not shown).

The screenshot displays the Avaya IP Office configuration interface for system 00E00706530F. The 'Telephony' tab is active, and the 'Companding Law' section is highlighted with a red box. In this section, the 'Switch' and 'Line' options are both set to 'U-Law'. The 'Inhibit Off-Switch Forward/Transfer' checkbox is also highlighted with a red box and is currently unchecked. Other visible settings include 'Default Outside Call Sequence' set to 'Normal', 'Default Inside Call Sequence' set to 'Ring Type 1', and 'Default Ring Back Sequence' set to 'Ring Type 2'. The 'Dial Delay Time (secs)' is set to 3, 'Dial Delay Count' is 0, 'Default No Answer Time (secs)' is 20, 'Hold Timeout (secs)' is 0, 'Park Timeout (secs)' is 300, and 'Ring Delay (secs)' is 5. The 'Call Priority Promotion Time (secs)' is set to 'Disabled', 'Default Currency' is 'USD', 'Default Name Priority' is 'Favor Trunk', and 'Phone Failback' is 'Manual'. The 'Login Code Complexity' section shows 'Enforcement' checked, 'Minimum length' set to 4, and 'Complexity' unchecked. The 'DSS Status' is unchecked, 'Auto Hold' is checked, 'Dial By Name' is checked, 'Show Account Code' is checked, 'Restrict Network Interconnect' is unchecked, 'Include location specific information' is unchecked, 'Drop External Only Impromptu Conference' is unchecked, 'Visually Differentiate External Call' is unchecked, 'Unsupervised Analog Trunk Disconnect Handling' is unchecked, 'High Quality Conferencing' is checked, 'Digital/Analogue Auto Create User' is checked, and 'Directory Overrides Barring' is unchecked.

5.2.3 System - Twinning Tab

Navigate to the **Twinning** tab on the Details Pane, configure the following parameters:

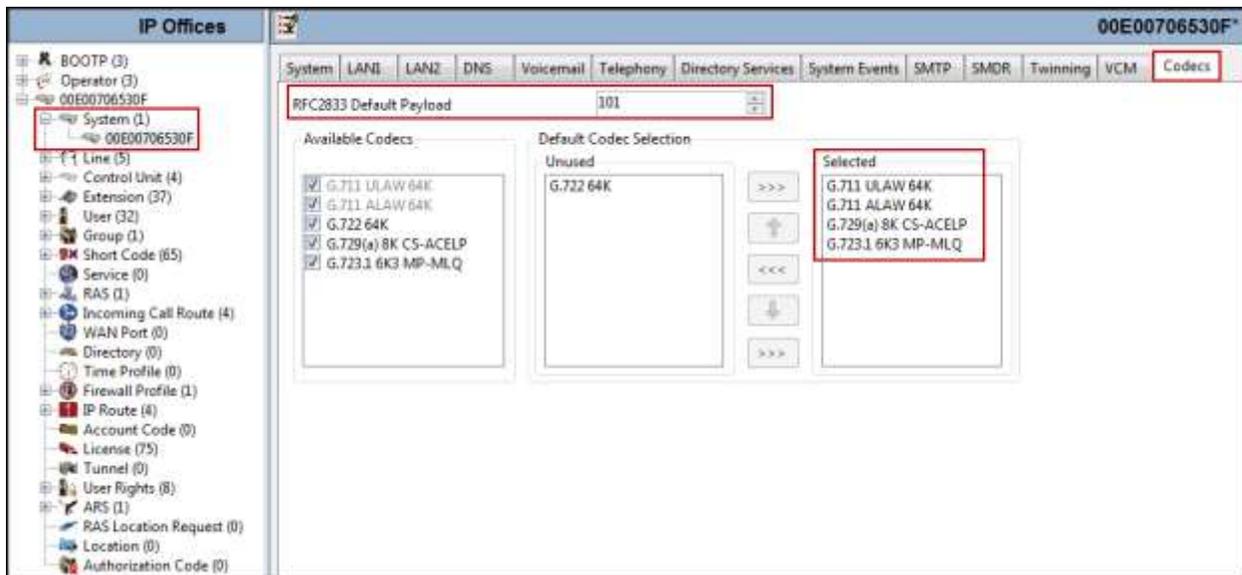
- Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be controlled by the setting on the SIP Line (**Section 5.4**). This setting also impacts the Caller ID for call forwarding.
- Click **OK** to commit (not shown).



5.2.4 System - Codecs Tab

For Codec's settings, navigate to the **System (1) → 00E00706530F** in the Navigation Pane, select the **Codecs** tab and configure the following parameters:

- The **RFC2833 Default Payload** field is new in IP Office release 9.1. It allows the manual configuration of the payload type used on SIP calls that are initiated by the IP Office. The default value **101** was used.
- 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 lines and phones (SIP and H.323) will use the system default codec selection shown here, unless configured otherwise for a specific line or extension. The example below shows the codecs used for IP phones (SIP and H.323), the system's default codecs and order was used.
- Click **OK** to commit (not shown).



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 5.4.6** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

5.3 IP Route

In the reference configuration, the IP Office LAN1 interface and the private interface of the Charter’s Modular Access Router resided on the same IP subnet, so an IP route was not necessary. In an actual customer configuration, these two interfaces may be in different IP subnets, and in that case an IP route would have to be created to specify the IP address of the gateway or router where the IP Office needs to send the packets, in order to reach the IP subnet where the Charter Modular Access Router resides.

To create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets in order to reach the IP subnet where the Charter Modular Access Router resides (if located in different subnets), on the left navigation pane, right-click on **IP Route** and select **New**.

- Set the **IP Address** and **IP Mask** of the subnet of the private side of Charter’s Modular Access Router, or enter **0.0.0.0** to make this the default route.
- Set **Gateway IP Address** to the IP Address of the default router in the IP Office subnet.
- Set **Destination** to **LAN1** from the pull-down menu.
- Click **OK** to commit (not shown).

The screenshot shows the IP Office configuration interface. On the left is a tree view of configuration objects under the heading 'IP Offices'. The 'IP Route (4)' object is selected and highlighted with a red box, showing a sub-tree with '0.0.0.0' and three other routes: '172.31.20.0', '192.168.188.0', and '192.168.99.0'. On the right, the configuration details for the selected '0.0.0.0' IP Route are displayed. The 'IP Address' and 'IP Mask' fields are both set to '0 . 0 . 0 . 0'. The 'Gateway IP Address' field is set to '172 . 16 . 5 . 254'. The 'Destination' field is set to 'LAN1'. The 'Metric' field is set to '0'. There is a checkbox for 'Proxy ARP' which is currently unchecked.

0.0.0.0	
IP Address	0 . 0 . 0 . 0
IP Mask	0 . 0 . 0 . 0
Gateway IP Address	172 . 16 . 5 . 254
Destination	LAN1
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 Charter SIP Trunking Service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by Avaya IP Office Manager to create a SIP Line. Follow the steps in **Sections 5.4.1** and **5.4.2** 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 **Section 5.4.3** to **5.4.7**.

Alternatively, a SIP Line can be created manually. To do so, right-click on **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.3** to **5.4.7**

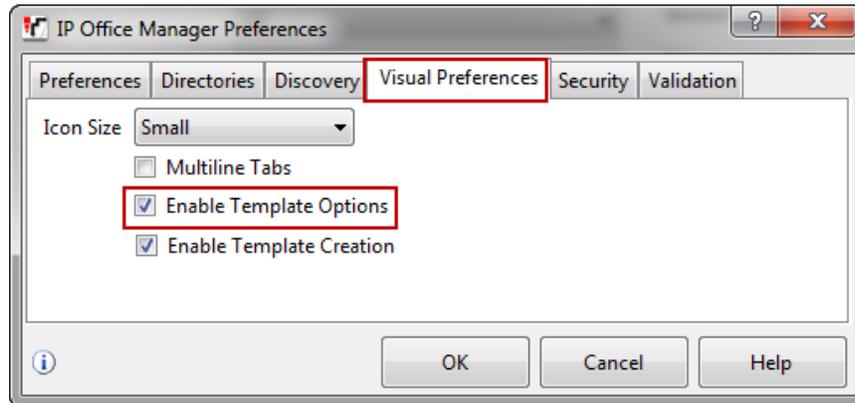
5.4.1 Importing a SIP Line Template

Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500v2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

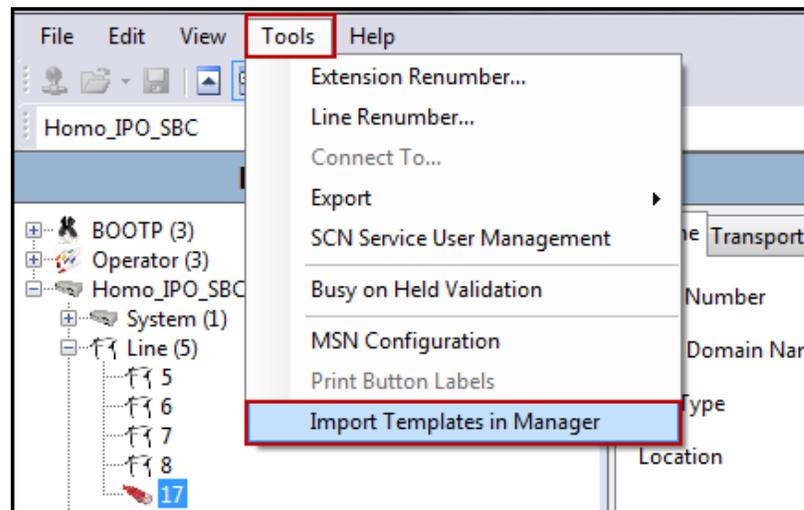
1. Copy a previously created template file to a location (e.g., *C:\Temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_<user supplied text>_SIPTrunk.xml**, where the *<user supplied text>* portion is entered during template file creation.

Note – If necessary, the *<user supplied text>* portion of the template file name may be modified, however the **AF_<user supplied text>_SIPTrunk.xml** format of the file name must be maintained. For example, an original template file **AF_TEST _SIPTrunk.xml** could be changed to **AF_Test1_SIPTrunk.xml**. The template file name is selected in **Section 5.4.2, step 2**, to create a new SIP Line.

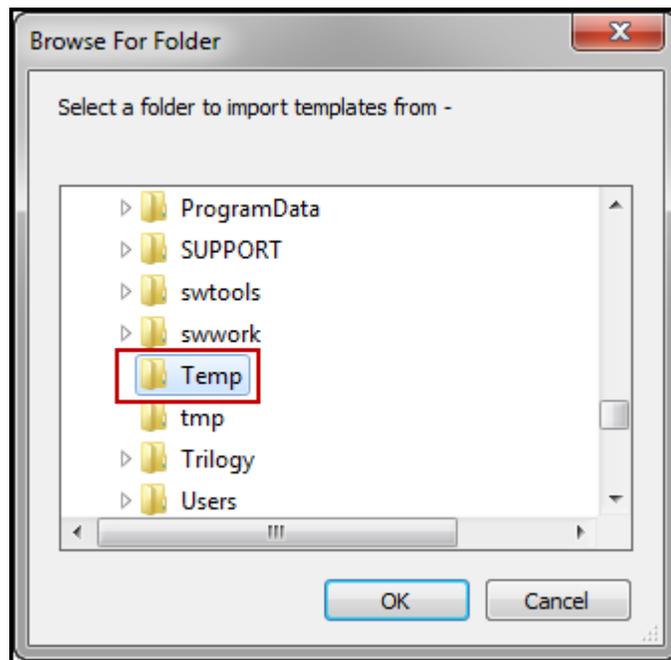
2. Verify that Template Options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Check the box next to **Enable Template Options**. Click **OK**.



3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.



4. A folder browser will open. Select the directory used in **step 1** to store the template(s) (e.g., C:\Temp).

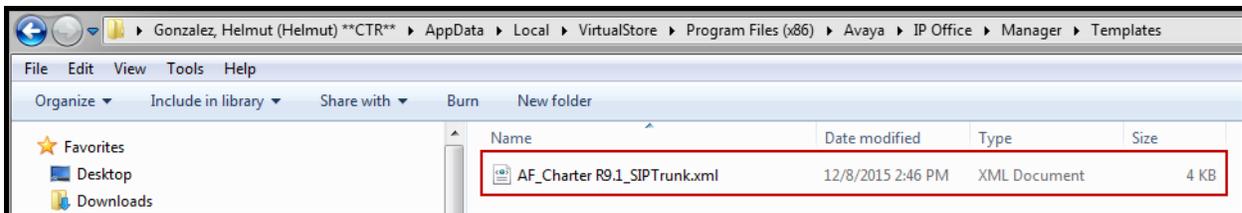
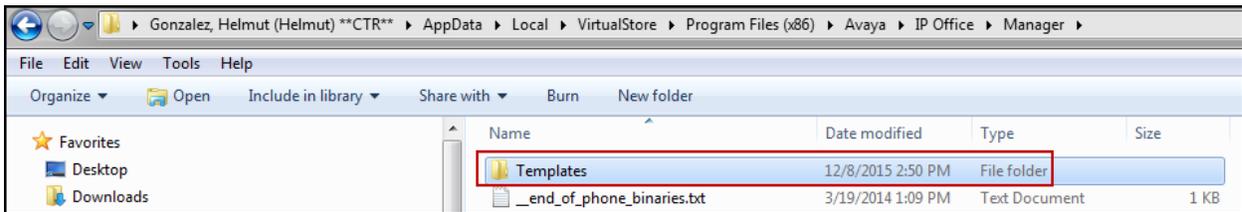
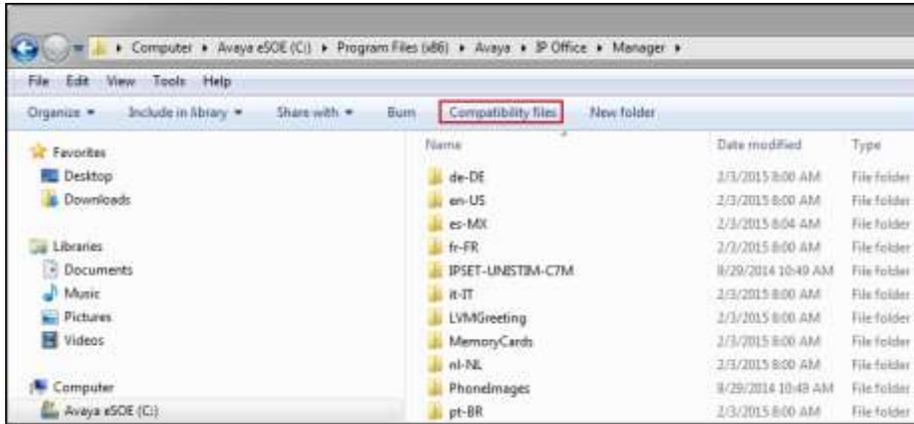


In the reference configuration, template files **AF_Charter R9.1_SIPTrunk.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

5. After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.

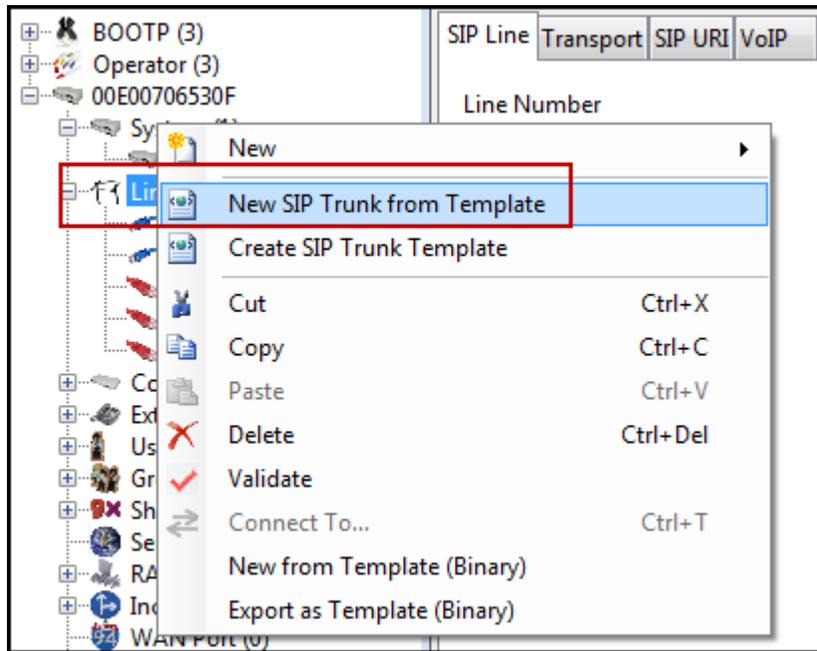


Note –Windows 7 (and later) locks the Avaya IP Office 9.1 \Templates directory, and it cannot be viewed. To enable browsing of the \Templates directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager\Templates** (or C:\Program Files (x86)\Avaya\IP Office\Manager\Templates), and then click on the **Compatibility files** option shown below. The \Templates directory and its contents can then be viewed.



5.4.2 Creating a SIP Trunk from an XML Template

1. To create the SIP Trunk from a template, right-click on **Line** in the Navigation Pane, and select **New SIP Trunk from Template**.

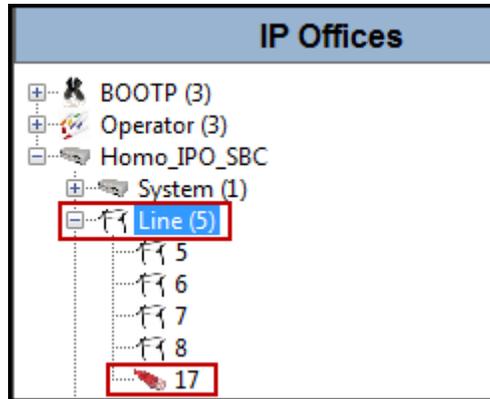


2. In the subsequent **Template Type Selection** pop-up window, from the **Service Provider** pull-down menu, select the XML template name from **Section 5.4.1**. Click **Create new SIP Trunk**.

Note – The drop down menu will display the *<user supplied text>* part of the template file name (see **Section 5.4.1**). If you check the **Display All** box, then the full template file name is displayed.



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line 17).

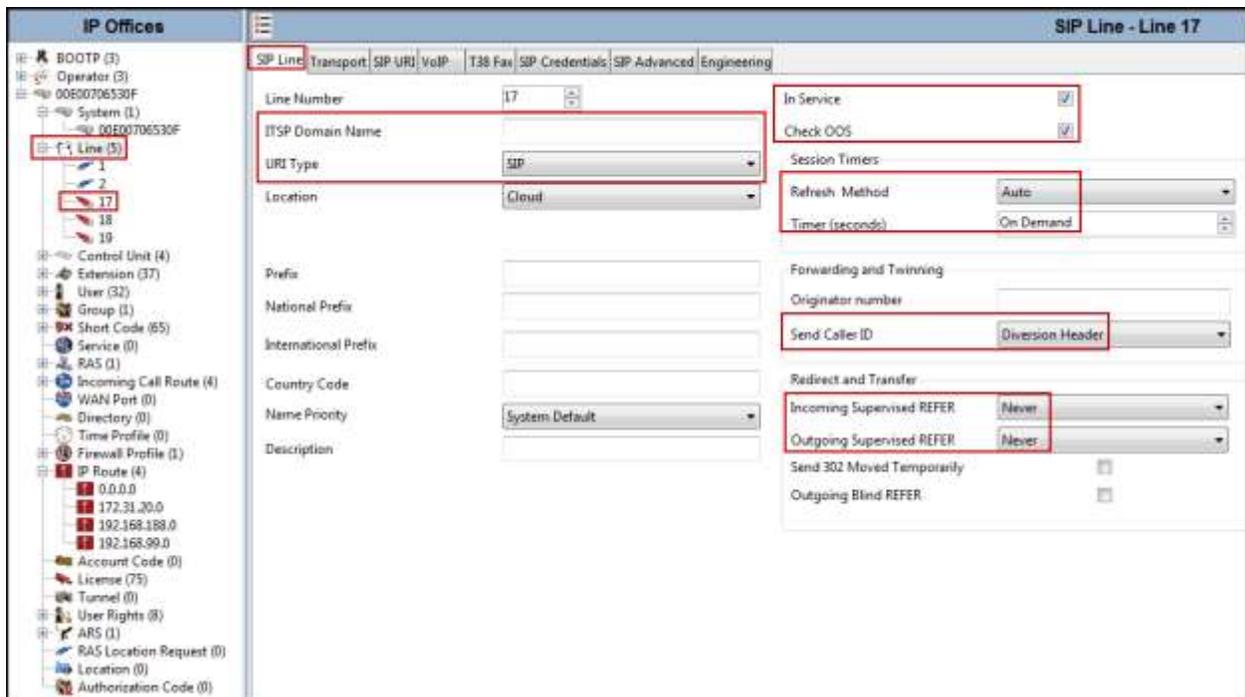


It is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.3 to 5.4.7**.

5.4.3 SIP Line - SIP Line Tab

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

- Leave the **ITSP Domain Name** blank. Note that if this field is left blank, then IP Office inserts the ITSP Proxy Address from the Transport tab as the ITSP Domain in the SIP messaging.
- Verify that **URI Type** is set to **SIP**.
- 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 LAN1, as shown in **Section 5.2.1**.
- Verify that **Refresh Method** is set to **Auto**.
- Verify that **Timer (seconds)** is set to **On Demand**.
- Set **Send Caller ID** to **Diversion Header**.
- Under **Redirect and Transfer**, set **Incoming Supervised REFER** Support and **Outgoing Supervised REFER** to **Never** (see **Sections 2.1**).
- All other parameters should be set to default or according to customer requirements.
- Click **OK** to commit (not shown).



5.4.4 SIP Line - Transport Tab

Select the **Transport** tab; configure the parameters as shown below:

- Set the **ITSP Proxy Address** to the IP address of the inside interface (or private side) assigned to Charter's Modular Access Router, as shown on **Figure 1**.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN1** as configured in **Section 5.2**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).

The screenshot displays the configuration interface for a SIP Line in the 'Transport' tab. The left sidebar shows a tree view of 'IP Offices' with 'Line (5)' selected. The main configuration area includes the following fields and settings:

- ITSP Proxy Address:** 172.16.5.185
- Network Configuration:**
 - Layer 4 Protocol:** UDP
 - Send Port:** 5060
 - Use Network Topology Info:** LAN1
 - Listen Port:** 5060
- Explicit DNS Server(s):** 0 . 0 . 0 . 0 | 0 . 0 . 0 . 0
- Calls Route via Registrar:**
- Separate Registrar:** (empty text field)

5.4.5 SIP Line - SIP URI Tab

A SIP URI entry needs to be created to match each incoming number that Avaya 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 Avaya IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact**, **Display Name** to **Use Internal Data**.
- Set **PAI** to **None**.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **17** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Click **OK**.
- Click **OK** again to commit (not shown).

The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view under 'IP Offices' shows a hierarchy including 'Line (5)', with 'Line 17' selected and highlighted with a red box. The main window shows the 'SIP URI' tab for a channel. A table lists the channel configuration:

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

Below the table is the 'Edit Channel' form. The 'SIP URI' tab is selected. The form fields are:

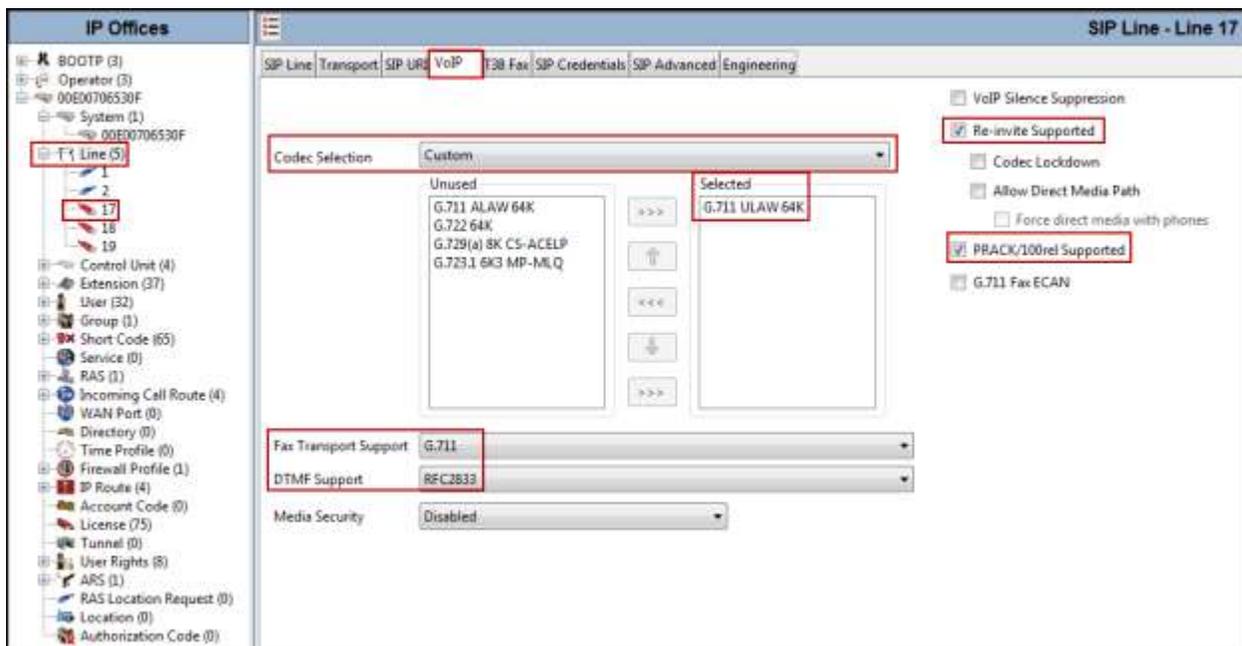
- Via: 172.16.5.60
- Local URI: Use Internal Data
- Contact: Use Internal Data
- Display Name: Use Internal Data
- PAI: None
- Registration: 0: <None>
- Incoming Group: 17
- Outgoing Group: 17
- Max Calls per Channel: 10

Buttons for 'Add...', 'Remove', 'Edit...', 'OK', and 'Cancel' are visible on the right side of the interface.

5.4.6 SIP Line - VoIP Tab

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

- In the sample configuration, the **Codec Selection** was configured using the **Custom** option, allowing an explicit order of codecs to be specified for the SIP Line. The buttons allow setting the specific order of preference for the codecs to be used on the SIP Line, as shown. Charter only supports codec G.711ULAW for audio.
- Select G.711 for **Fax Transport Support** (Refer to **Section 2.1**).
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check the **PRACK/100rel Supported** box, to advertise the support for reliable provisional responses and Early Media to Charter.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).

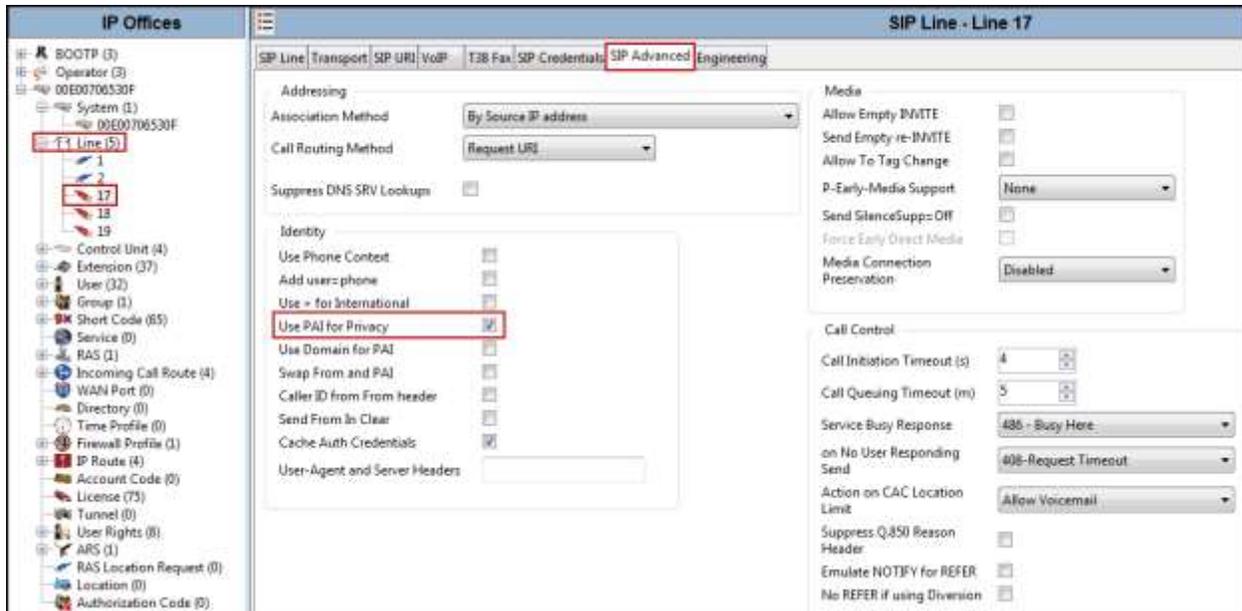


Note: The codec selections defined under this section (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk). The codec selections defined under **Section 5.2.4** (System – Codecs tab) are the codecs selected for the IP phones/extension (H.323 and SIP).

5.4.7 SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab. For outbound calls with privacy enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “anonymous”. IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing purposes. By default, IP Office will use the PPI header for privacy. To configure IP Office to use the PAI header for privacy calls:

- Check the box for **Use PAI for Privacy**.
- Default values may be used for all other parameters.
- Click OK to commit (not shown).



5.5 Extension

In this section, an example of an Avaya IP Office Extension will be illustrated. In the interest of brevity, not all users and extensions will be presented, since the configuration can be easily extrapolated to other users and extensions. To add an Extension, right click on **Extension** then select **New** → **Select H323 or SIP**.

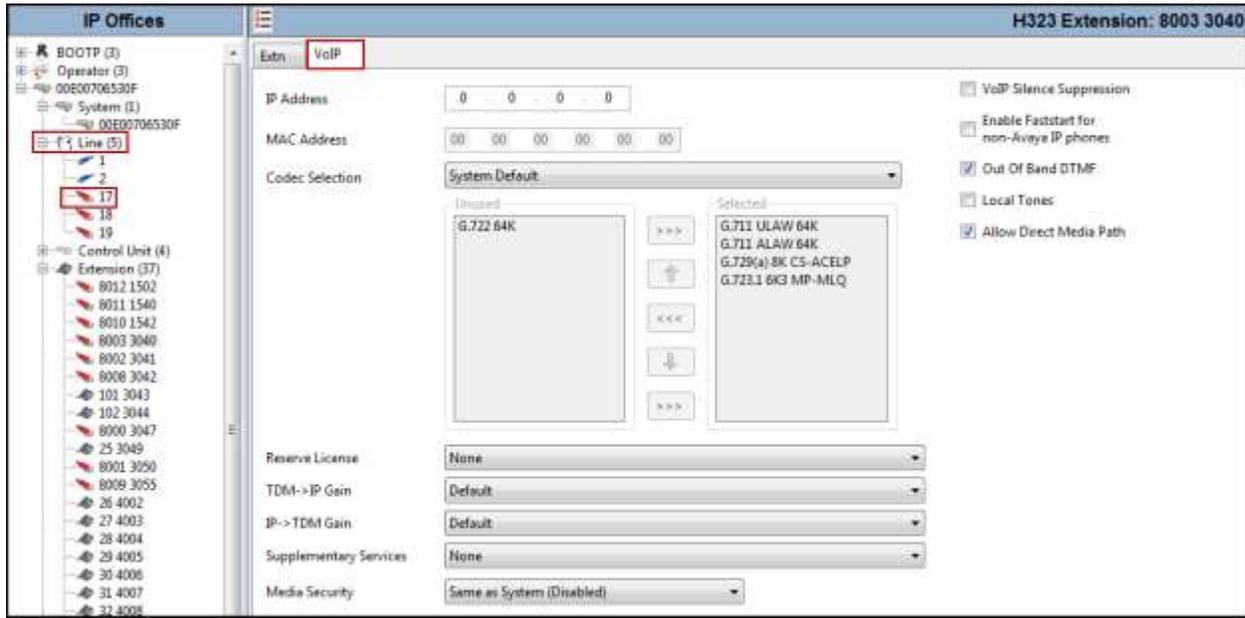
Select the **Extn** tab. Following is an example of extension 3040; this extension corresponds to an H.323 extension.

The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view under 'IP Offices' shows a hierarchy: BOOTP (5), Operator (3), System (1), Line (3), and Control Unit (4). Under 'Control Unit (4)', the 'Extension (38)' folder is selected, and the extension '8003 3040' is highlighted. The main panel on the right is titled 'H323 Extension: 8003 3040' and has two tabs: 'Extn' (selected) and 'VoIP'. The configuration fields are as follows:

Field	Value
Extension Id	8003
Base Extension	3040
Phone Password	
Caller Display Type	On
Reset Volume After Calls	<input type="checkbox"/>
Device Type	Avaya 9620
Location	Automatic
Module	0
Port	0
Disable Speakerphone	<input type="checkbox"/>

Select the **VOIP** tab. Use default values on VoIP tab. Following is an example for Extension 3040; this extension corresponds to an H.323 extension.

By default, all IP phones (SIP and H.323) will use the system default codec selection configured under the System Codecs tab (**Section 5.2.4**), unless configured otherwise for a specific extension by selecting **Custom** under **Codec Selection** on the screenshot shown below. The example below shows the codecs used for IP phones (SIP and H.323).



5.6 Users

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** in the left Navigation Pane, and 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 **Ext3040 H323**.

The screenshot displays the Avaya user configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'User (32)' expanded and '3040 Ext3040 H323' selected. The main area is titled 'Ext3040 H323: 3040' and contains a 'User' tab. The configuration fields are as follows:

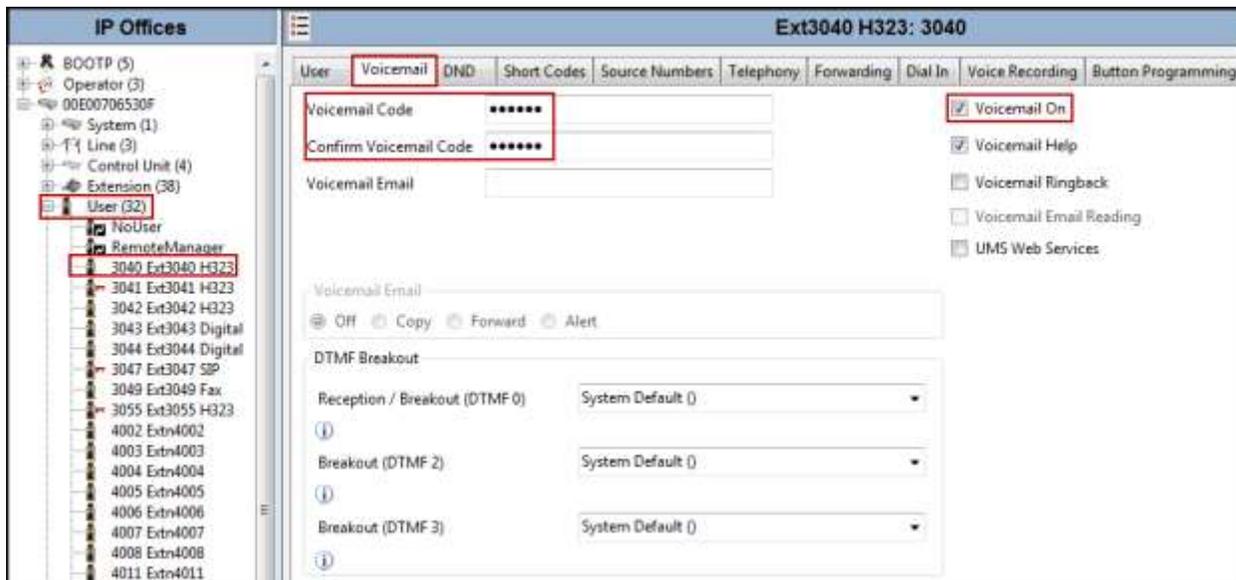
Field	Value
Name	Ext3040 H323
Password	••••
Confirm Password	••••
Account Status	Enabled
Full Name	Ext3040 H323
Extension	3040
Email Address	
Locale	
Priority	5
System Phone Rights	None
ACCS Agent Type	None
Profile	Basic User
Receptionist	<input type="checkbox"/>
Enable Softphone	<input type="checkbox"/>
Enable one-X Portal Services	<input checked="" type="checkbox"/>
Enable one-X TeleCommuter	<input type="checkbox"/>
Enable Remote Worker	<input checked="" type="checkbox"/>
Enable Flare	<input type="checkbox"/>
Enable Mobile VoIP Client	<input type="checkbox"/>
Send Mobility Email	<input type="checkbox"/>
Ex Directory	<input checked="" type="checkbox"/>
Device Type	Avaya 9620
User Rights	
User Rights view	User data
Working hours time profile	<None>
Working hours User Rights	

In the example below, the name of the user is “Ext3047 SIP”. This is an Avaya IP Office Softphone user, set the Profile to **Power User** and check **Enable Softphone**.

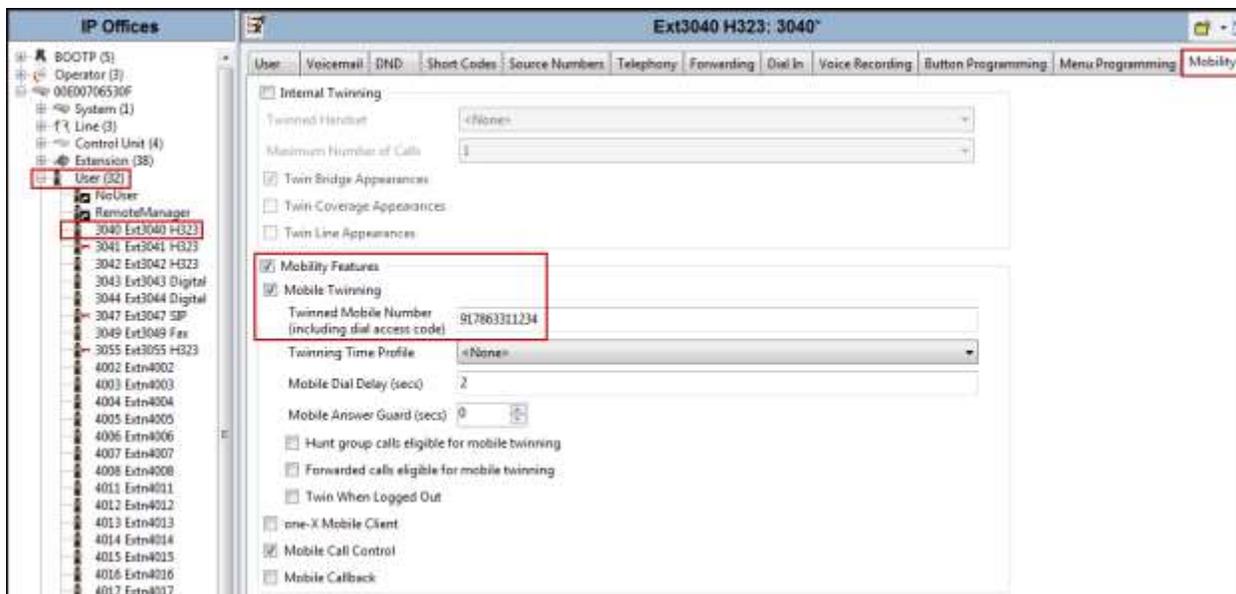
The screenshot displays the Avaya IP Office user configuration interface. On the left, a tree view shows the hierarchy: IP Offices > User (32) > 3047 Ext3047 SIP. The main configuration area is titled "Ext3047 SIP: 3047" and includes several tabs: User, Voicemail, DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. The "User" tab is active, showing the following configuration details:

- Name: Ext3047 SIP
- Password: [Redacted]
- Confirm Password: [Redacted]
- Account Status: Enabled
- Full Name: Softclient 3047
- Extension: 3047
- Email Address: [Empty]
- Locale: [Dropdown]
- Priority: 5
- System Phone Rights: None
- ACCS Agent Type: None
- Profile: Power User
- Receptionist:
- Enable Softphone:
- Enable one-X Portal Services:
- Enable one-X TeleCommuter:
- Enable Remote Worker:
- Enable Flare:
- Enable Mobile VoIP Client:
- Send Mobility Email:
- Ex Directory:
- Device Type: Unknown SIP device
- User Rights: [Dropdown]
- User Rights view: User data
- Working hours time profile: <None>

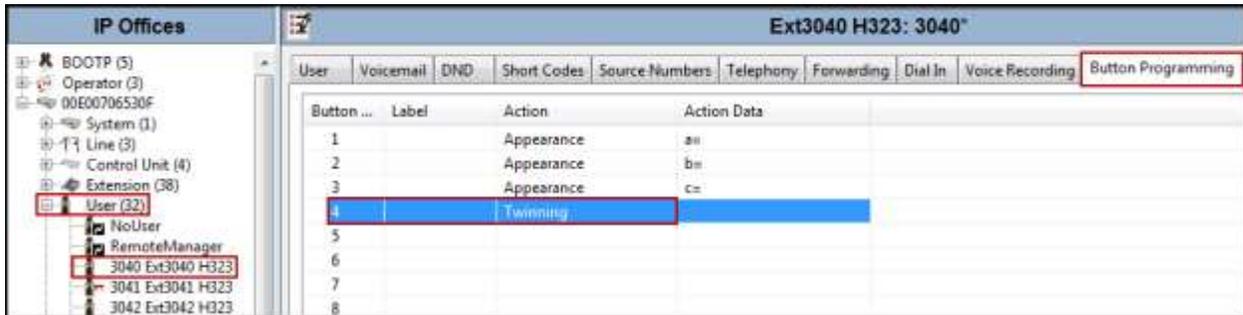
Select the **Voice Mail** tab. The following screen shows the **Voicemail** tab for the user with extension 3040. The **Voicemail On** box is checked. Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from Charter to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones to test DTMF using RFC 2833.



Select the **Mobility** tab. In the sample configuration user 3040 was one of the users configured to test the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 3040. 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 telephone, including the dial access code “9”, in this case **917863311234**. Other options can be set according to customer requirements.

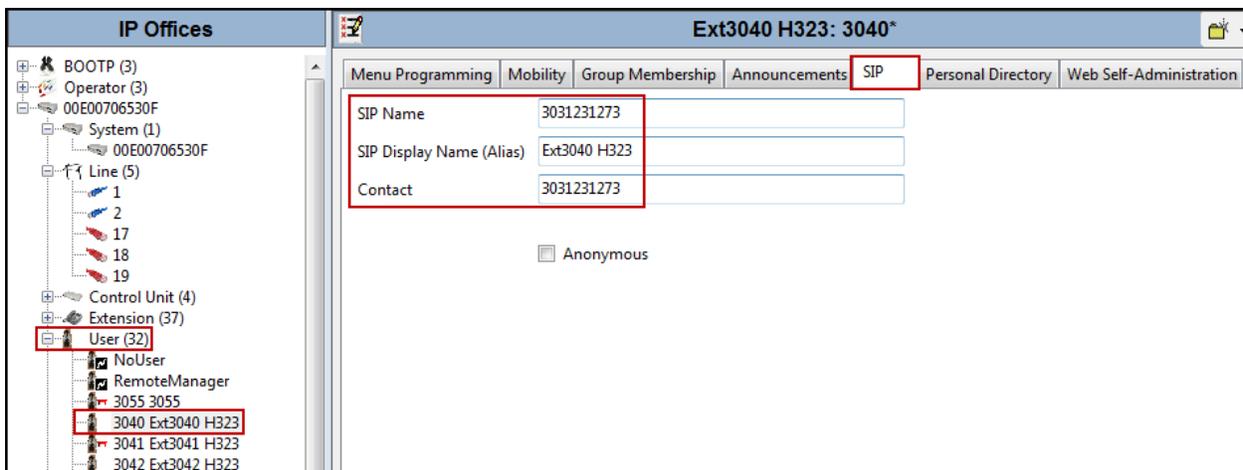


To program a key on the telephone to turn Mobil Twinning on and off, select the **Button Programming** tab on the user, then select the button to program to turn Mobil Twinning on and off, click on **Edit → Emulation → Twinning** (not shown). In the sample below, button 4 was programmed to turn Mobil Twinning on and off on user 3040.



Select the **SIP** tab, the values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the “From” and “Contact” headers for outgoing SIP trunk calls. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.4**). The example below shows the settings for user “Ext3040 H323”. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by Charter. In the example, DID number **3031231273** was used. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.

If all calls involving this user should be considered private, then the **Anonymous** box may be checked to withhold the Caller ID information from the network.



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 trunk 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 5.4.5)** and the users **SIP Name** and **Contact**, already populated with the assigned Charter DID numbers (**Section 5.6**).

From the left Navigation Pane, right-click on **Incoming Call Route** and select **New**.

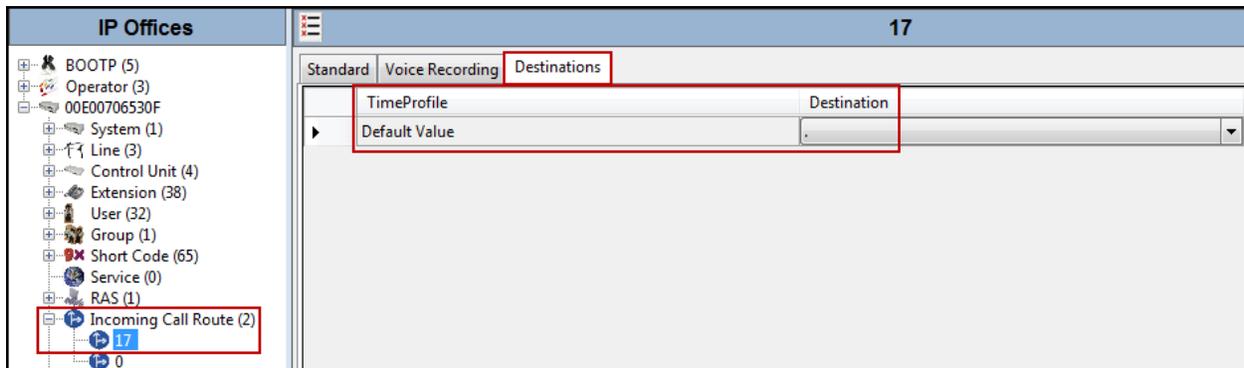
On the Details Pane (not shown), under the **Standard** tab, set the parameters as show bellow:

- 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**.
- Default values may be used for all other parameters.

The screenshot displays the IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree view. The 'Incoming Call Route (2)' item is selected and highlighted with a red box. On the right is the configuration details pane for the selected item, with the 'Standard' tab active and highlighted. The configuration fields are as follows:

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

- Under the **Destinations** tab, enter “.” for the **Default Value**. This setting will allow the call to be routed to any destination with a value on its **SIP Name** field, entered on the **SIP** tab of that **User**, which matches the number present on the user part of the incoming Request URI.
- Click **OK** to commit (not shown).



5.8 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.

5.8.1 Short Codes and Automatic Route Selection

To create the short code used for ARS, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). The screen below shows the creation of the short code **9N** used in the reference configuration. When the Avaya IP Office users dialed 9 plus any number N, calls were directed to **Line Group 50: Main**, configurable via ARS and defined next in this section.

The screenshot displays the Avaya IP Office configuration interface. On the left, a list of short codes is shown under the heading 'IP Offices'. The short code '9N' is highlighted with a red box. On the right, the configuration details for the short code '9N' are shown under the heading '9N: Dial'. The configuration includes:

- Code: 9N
- Feature: Dial
- Telephone Number: N
- Line Group ID: 50: Main
- Locale: (empty)
- Force Account Code:

The following screen shows the example ARS configuration for the route **Main**. Note the sequence of **Xs** 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 digit on the string. This type of setting results in a much quicker response in the delivery of the call by IP Office. The first example highlighted below shows that for calls to area codes in the North American Numbering Plan, the user dialed 9, followed by 11 digits, starting with a 1.

The screenshot displays the configuration for the 'Main' ARS route. The left sidebar shows the 'IP Offices' tree with 'ARS (1)' expanded to '00: Main'. The main configuration area includes fields for ARS Route ID (50), Route Name (Main), and various options like 'Secondary Dial tone' and 'Check User Call Barring'. A table lists dialing codes and their corresponding features and line group IDs.

Code	Telephone Number	Feature	Line Group ID
11	911	Dial Emergency	0
911	911	Dial Emergency	0
001XXXXXXXXX	001N	Dial	17
8XXXXXXXXX	8N	Dial	17
1XXXXXXXXX	1N	Dial	17
6XXXXXX	6N	Dial	17
3XXXXXXXXX	3N	Dial	17

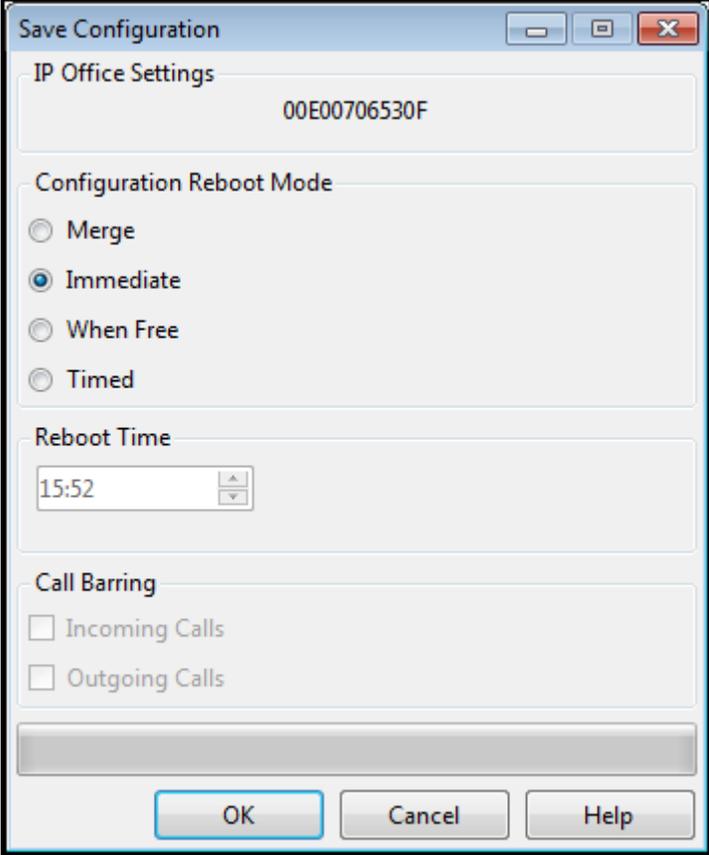
Below the table, the 'Alternate Route Priority Level' is set to 3 and the 'Alternate Route Wait Time' is set to 30. The 'Alternate Route' dropdown is set to '<None>'.

5.9 Save Configuration

When desired, send the configuration changes made in Avaya IP Office Manager to the Avaya IP Office server in order for the changes to take effect.

Navigate to **File**→**Save Configuration** in the menu bar at the top left of the screen to save the configuration performed in the preceding sections.

Once the configuration is validated, a screen similar to the following will appear, with either the **Merge** or the **Immediate** radio button chosen based on the nature of the configuration changes made since the last save. Note that clicking OK may cause a service disruption due to system reboot. Click **OK** if desired.



The image shows a 'Save Configuration' dialog box with the following fields and controls:

- IP Office Settings:** A text field containing the value '00E00706530F'.
- Configuration Reboot Mode:** A group box containing four radio buttons: 'Merge', 'Immediate' (which is selected), 'When Free', and 'Timed'.
- Reboot Time:** A time selection control showing '15:52'.
- Call Barring:** A group box containing two checkboxes: 'Incoming Calls' and 'Outgoing Calls', both of which are currently unchecked.
- Buttons:** At the bottom of the dialog are three buttons: 'OK', 'Cancel', and 'Help'.

6. Charter Business SIP Trunk Service Configuration

To use the Charter Business SIP Trunking service offering, a customer must request the service from Charter using the established sales processes. The process can be started by contacting Charter via the corporate web site at: <https://www.charterbusiness.com/> or by calling 800-314-7195.

During the signup process, Charter and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to Charter's network. Charter will provide IP addresses, Direct Inward Dialed (DID) numbers to be assigned to the enterprise, etc. This information is used to complete the Avaya IP Office configuration discussed in the previous sections.

As previously noted, as a required component of the Charter Business SIP Trunking service offering, Charter will install a Modular Access Router at the customer premises (enterprise site). Charter will perform the initial configuration and maintenance as required. The Modular Access Router will be considered Customer Premises Equipment (CPE).

7. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting tips that can be used to troubleshoot the solution.

7.1 Verification Steps

The following steps may be used to verify the configuration:

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

7.2 Protocol Traces

The following SIP message headers are inspected using sniffer trace analysis tool:

- Request-URI: Verify the request number and SIP domain.
- From: Verify the display name and display number.
- To: Verify the display name and display number.
- P-Asserted-Identity: Verify the display name and display number.
- Privacy: Verify privacy masking with “user, id”.
- Diversion: Verify the display name and display number.

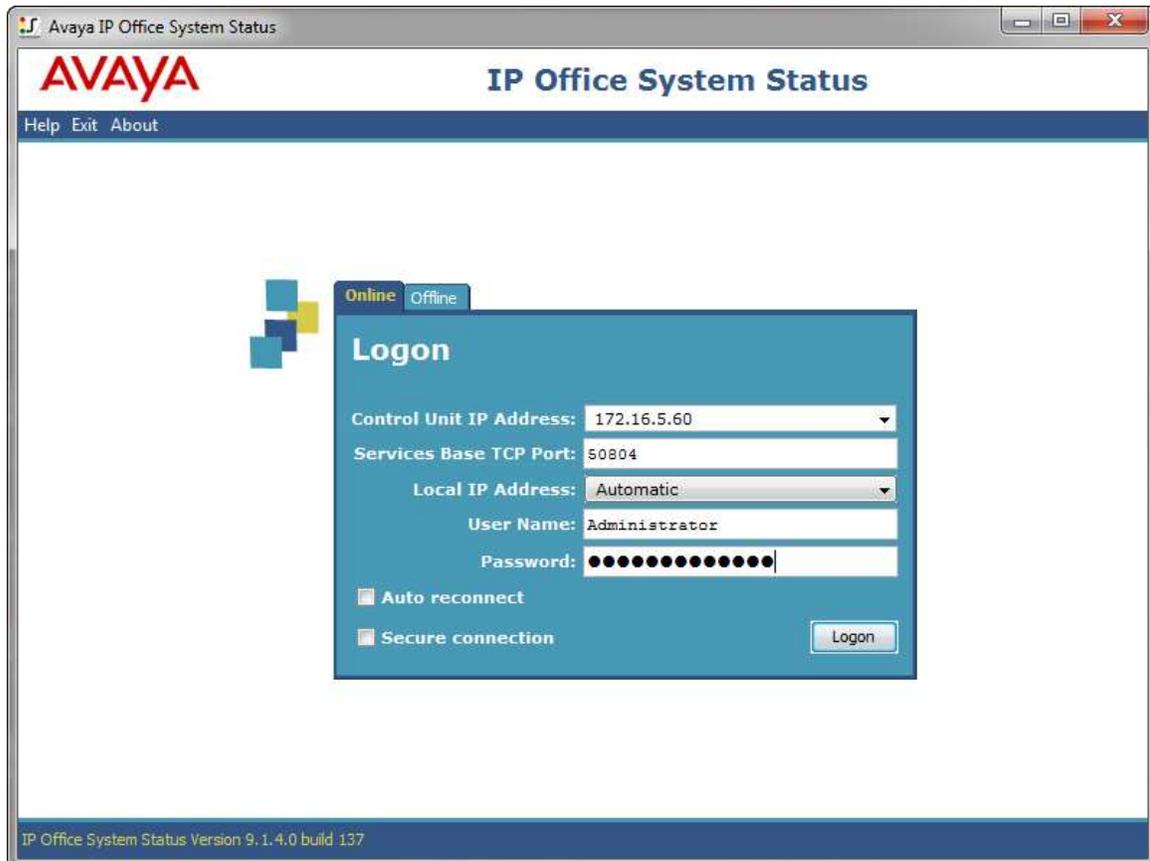
The following attributes in SIP message body are inspected using sniffer trace analysis tool:

- Connection Information (c line): Verify IP addresses of near end and far end endpoints.
- Time Description (t line): Verify session timeout value of near end and far end endpoints.
- Media Description (m line): Verify audio port, codec, DTMF event description.
- Media Attribute (a line): Verify specific audio port, codec, ptime, send/ receive ability, DTMF events.

7.3 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 IP Office Manager is installed, log in with the proper credentials.



- Select the SIP Line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is **Idle** for each channel (assuming no active calls at present time).

The screenshot shows the Avaya IP Office System Status window. The left pane shows a tree view with 'Trunks (5)' expanded to 'Line:17'. The main pane is on the 'Status' tab, displaying the 'SIP Trunk Summary' for Line 17. The summary includes fields like Line Service State (In Service), Peer Domain Name (sip://172.16.5.185), Resolved Address (172.16.5.185), Line Number (17), Number of Administered Channels (10), Number of Channels in Use (0), Administered Compression (G711 Mu), Enable Faststart (Off), Silence Suppression (Off), Media Stream (RTP), Layer 4 Protocol (UDP), SIP Trunk Channel Licenses (Unlimited), and SIP Trunk Channel Licenses in Use (0). A green progress indicator shows 0%. Below the summary is a table of channels.

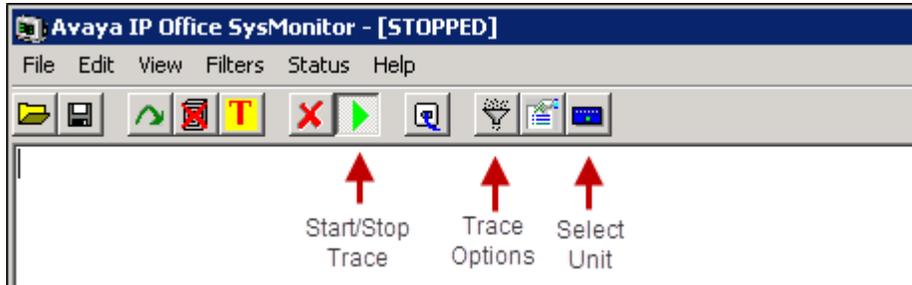
Cha...	U..	Call	Curr...	Time in	Remote	C...	Con...	Caller	Other	Dire...	Round	Rec...	Rec...	Tran...	Tran...
	Ref		State	Media...				ID o...	Party on...		Trip ...				
1			Idle	02:2...											
2			Idle	07:5...											
3			Idle	07:5...											
4			Idle	07:5...											
5			Idle	07:5...											
6			Idle	07:5...											
7			Idle	07:5...											
8			Idle	07:5...											
9			Idle	07:5...											
10			Idle	07:5...											

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

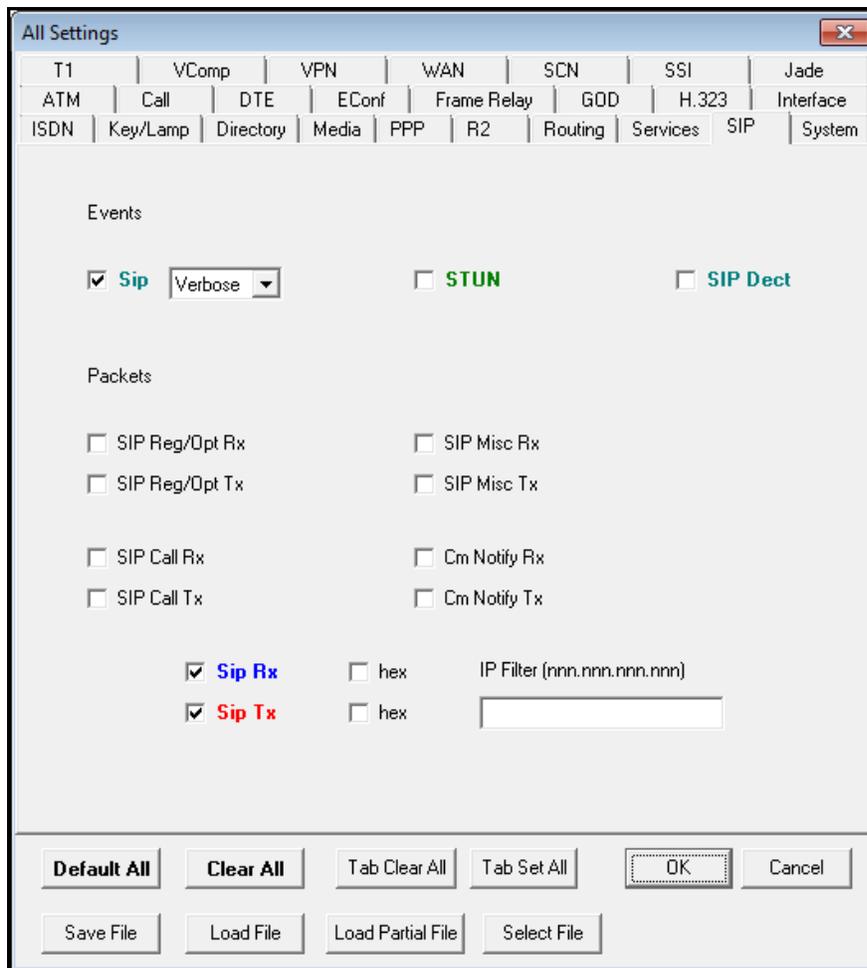
The screenshot shows the Avaya IP Office System Status window with the 'Alarms' tab selected. The left pane shows 'Trunks (3)' expanded to 'Line:17'. The main pane displays 'Alarms for Line: 17 SIP 172.16.5.185'. Below this is a table with columns for 'Last Date Of Error', 'Occurrences', and 'Error Description'. The table is currently empty.

7.4 IP Office Monitor

The Avaya IP Office 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 Manager 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 and 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 to the desired color.



8. Conclusion

These Application Notes describe the procedures required to configure SIP trunk connectivity between Avaya IP Office 9.1 and Charter Business SIP Trunking Service, as shown in **Figure 1**.

Interoperability testing was completed successfully 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] *Deploying Avaya IP Office Platform IP500 V2*, Document Number 15-601042, Issue 30u, November 06, 2015.
- [2] *Using Avaya IP Office Platform System Status*, Document Number 15-601758, Issue 10f, August 2015.
- [3] *Administering Avaya IP Office Platform Voicemail Pro*, Document Number 15-601063, Issue 10j, November 06, 2015.
- [4] *Using IP Office System Monitor*, Document Number 15-601019, Issue 06e, May 19, 2015.

Additional Avaya IP Office documentation can be found at:

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

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