**Avaya Solution & Interoperability Test Lab**

# Application Notes for Configuring Avaya IP Office 8.0 with TELUS SIP Trunk Service - Issue 1.0

## Abstract


These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider TELUS and Avaya IP Office 8.0.

During the interoperability testing, Avaya IP Office was able to interoperate with the TELUS Communication NSN HiQ switch via SIP trunking. This test was performed to verify SIP trunk features including basic call, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls were placed in both directions with various set types.

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

TELUS is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

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

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 8.0, Avaya Voicemail Pro, Avaya IP Office SIP and H.323 soft clients, Avaya 96xx series (H.323) phones, Avaya 11xx series (SIP) phones, Avaya 9508 Digital phones, Avaya 1408 Digital phones, analog phones and a fax machine.

The TELUS SIP Trunk Service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

# 2. General Test Approach and Test Results

The approach used for the tests was to connect a simulated enterprise site to the TELUS Communication NSN HiQ switch via SIP trunking and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of an Avaya IP Office and various Avaya endpoints. The testing was conducted remotely 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 SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test.:

- Static IP Authentication.
- SIP OPTIONS messages.
- Incoming calls to Avaya IP Office from the PSTN were routed to the DID numbers assigned by TELUS. Incoming PSTN calls were terminated to the following end points: Avaya IP Office Phone Manager (H.323) and Avaya IP Office Video Softphone (SIP), Avaya 96xx series (H.323) phones, Avaya 11xx series (SIP) phones, Avaya 9508 Digital phones, Avaya 1408 Digital phones, analog phones and a fax machine.
- Outgoing calls from Avaya IP Office were routed via the Nokia-Siemens HiQ switch (Release: 14). The HiQ switch connects to various CS2K Gateways (Release: CVM12).
- Proper disconnect when the caller or the callee abandoned the call before the call was answered.
- Proper disconnect with normal active call termination by the caller or the callee.
- Proper disconnect by the network for calls that were not answered (with voice mail off).

- Proper response to busy end points.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Codec G.711u.
- Proper response/error treatment with no matching codecs between the network and the enterprise.
- Voice mail and DTMF tone support (DTMF transmission using RFC 2833).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response) systems.
- Outbound/Inbound local calls.
- International calls.
- Operator services (0, 0 + 10 digits).
- Local Directory Assistance Calls (411).
- Calling number blocking from Avaya IP Office and from PSTN.
- Call Hold/Resume (long and short duration).
- Call Forward (unconditional, busy, no answer).
- Blind Call Transfers.
- Consultative Call Transfers.
- Station Conference.
- T.38 faxing support (inbound and outbound).
- Avaya IP Office Mobility Twinning.
- Simultaneous active calls.
- Long duration calls (> one hour).
- Proper response/error treatment to all trunks busy.
- Proper response/error treatment when disabling SIP connection.

A TELUS specific Test Plan with test cases for Mobility and DV endpoints was also executed.

Items not supported or not tested included the following:

- Emergency calls are supported but were not tested as part of the compliance test.
- Inbound Toll Free call.
- SIP REFER

## 2.2. Test Results

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

- **SIP REFER** – Problems were encountered with SIP REFER enabled in Avaya IP Office. The use of SIP REFER is not recommended with this solution.
- **No matching codec** – Two different behaviors were observed when Avaya IP Office was configured with codecs not supported by TELUS, as follows:
    - **Outbound calls to TELUS**: With codec G.722 configured in Avaya IP Office, TELUS responds with "488 Not acceptable here" as expected, but the call reaches the PSTN number dialed, the PSTN phone rings once. The user hears silence. The expected behavior is that the call should not complete when the response is "488 Not acceptable here".

- **Outbound calls to TELUS**: With G.723.1 configured in Avaya IP Office, TELUS responds with "487 Request Terminated". The users hears ring-back tone, the call does not complete (never reaches the PSTN number dialed). The expected behavior is a response of "488 Not acceptable here".

TELUS preliminary investigation based on captured traces indicated the problem is in their Nokia Siemens HiQ switch. Further investigation will be done by TELUS.

- **PSTN Voice Mail retrieval** – When calling a PSTN voice mail system from Avaya IP Office the first two words of the greeting message are not heard (are clipped off). No early Media (183) is received.
- **Call Displays on transferred calls to PSTN** – Caller ID display is not properly updated on PSTN phones involved with call transfers from Avaya IP Office to the PSTN. On Call Transfers from Avaya 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 DID of the extension that initiated the call transfer.
- **Blind Transfer of Outbound PSTN calls to internal extension** – Caller ID for blind transfer of outbound PSTN calls to internal extensions using H.323 phones shows the correct number (PSTN) when the call first arrives at the internal extension, but after the transfer is completed the Caller ID is updated with "External". For SIP phones (1140) the Caller ID is displayed as "External" when the call first arrives at the internal extension, but after the transfer is completed the Caller ID is updated with the correct number (PSTN).
- **Blind Transfer of Outbound calls with Avaya 1140 SIP phones** – When a call is made from the Avaya 1140 SIP Phones to the PSTN and is then blind transferred back out to the PSTN the call fails to complete. The 1140 phone sends the PSTN caller number in the FROM of the second INVITE message instead of the valid DID number doing the transfer. TELUS responds with a "500 Internal Server Error". The 1140 phone should send the DID known to TELUS in the FROM of the second invite instead of the PSTN number. This issue has been reported to Avaya development.
- **Outbound fax using T.38 interworking with G.711u** – T.38 Fallback mode was used in Avaya IP Office. In this fax mode, outgoing fax calls will attempt to use T.38 transport first. When the outbound fax call is made to TELUS, the first INVITE message from Avaya IP Office to TELUS contains codec G.711u, then a re-INVITE to switch over to T.38 is expected from TELUS, the re-INVITE is never received. Since the re-INVITE is not received by Avaya IP Office, Avaya IP Office sends an INVITE to TELUS with T.38 as the transport, TELUS responds with "488 Not Acceptable here". When the T.38 transport fails to be established, the fax transport method falls back to G.711 pass-through. The fax is then successfully sent via G.711u. In order to use T.38 transport in the TELUS network, calls have to be routed through gateways that support T.38. T.38 fax transport was successfully tested to a fax number provided by TELUS.
- **Outbound calls to TELUS Mobility End Points** – Outbound calls to TELUS Mobility end points are disconnected by Avaya IP Office when the call is answered at the Mobility end point. Avaya IP Office is sending a "bye" to TELUS after the 200 ok with SDP is received from TELUS. Avaya IP Office is ignoring the SDP being sent by TELUS in the 200 ok message due to the SDP session id/Session Version setting, both values are set to

zero (0 0). This issue has been reported to the Avaya IP Office Development group, which has confirmed it to be a limitation in the current software release of Avaya IP Office. A fix provided by the Avaya IP Office Development team was successfully tested. Avaya will include the fix in the next release of the Avaya IP Office software.

### 2.3. Support

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

For information on the TELUS services visit http://telus.com/regionselect.html

## 3. Reference Configuration

**Figure 1** below illustrates the configuration used during the compliance test.

Located at the enterprise site is an Avaya IP Office 500v2 with a Digital Expansion Module. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public network. Endpoints include Avaya IP Office Phone Manager (H.323) and Avaya IP Office Video Softphone (SIP), Avaya 96xx series (H.323) phones, Avaya 11xx series (SIP) phones, Avaya 9508 Digital phones, Avaya 1408 Digital phones, analog phones and a fax machine.

The site also has a Windows XP PC running Avaya IP Office Manager to configure and administer the Avaya IP Office system, and Avaya Voicemail Pro for providing voice messaging service to the Avaya IP Office users.

For security reasons, any actual public IP addresses used in the configuration have been masked. Similarly, any references to real routable PSTN numbers have also been masked to numbers that cannot be routed by the PSTN.

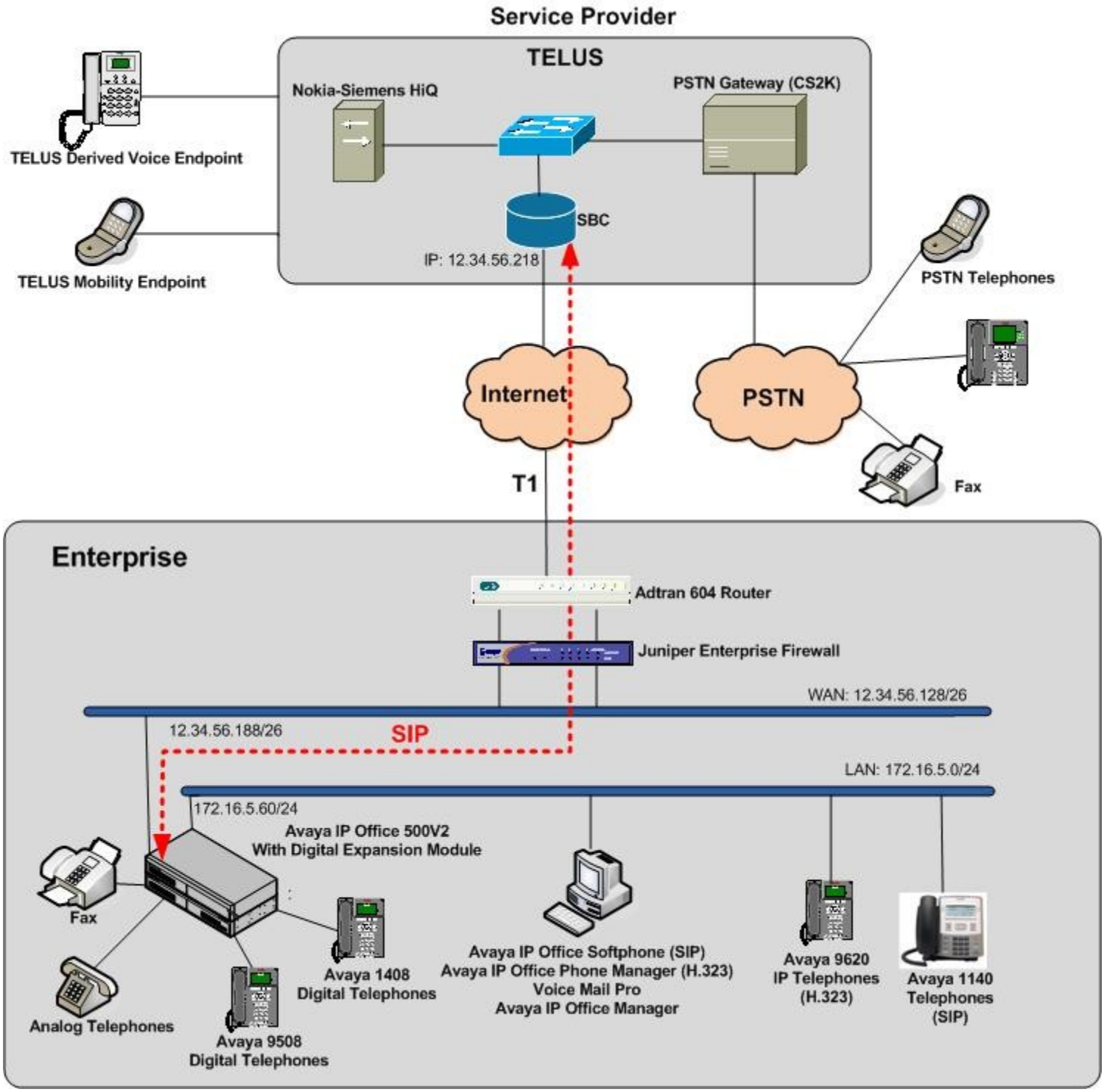



**Figure 1: Test Configuration**

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to TELUS. The short code 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to TELUS. For local calls to land lines the user dialed 9

followed by seven digits. Other calls, like mobile phones, Toll Free, long distance, international, etc. use different number lengths, and should be accordingly provisioned in the Avaya IP Office, with entries on the Short Codes or ARS forms. ARS was implemented and tested during the compliance testing, but its configuration is beyond the scope of these Application Notes. Short Codes are discussed in **Section 5.8**. For inbound calls, TELUS sent 123 plus 7 digits in the Request URI and the To headers of inbound SIP INVITE messages.

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

| DID Number | Avaya IP Office Extension | Client Type |
|---|---|---|
| 123-699-9464 | 3040 | Avaya 9620 IP Telephone (H323) |
| 123-699-9465 | 3041 | Avaya 9620 IP Telephone (H323) |
| 123-699-9466 | 3042 | Avaya 9620 IP Telephone (H323) |
| 123-699-9467 | 3043 | Avaya 9508 Digital Telephone |
| 123-699-9468 | 3044 | Avaya 9508 Digital Telephone |
| 123-699-9469 | 3047 | Avaya IP Office Softphone (SIP) |
| 123-699-9470 | 3048 | Avaya IP Office Phone Manager PC Softphone (H.323) |
| 123-699-9471 | 3049 | Analog Line |
| 123-699-9472 | 3050 | Avaya 1140 (SIP) |

**Table 1 – DID and Avaya IP Office Client Types Used for Testing**

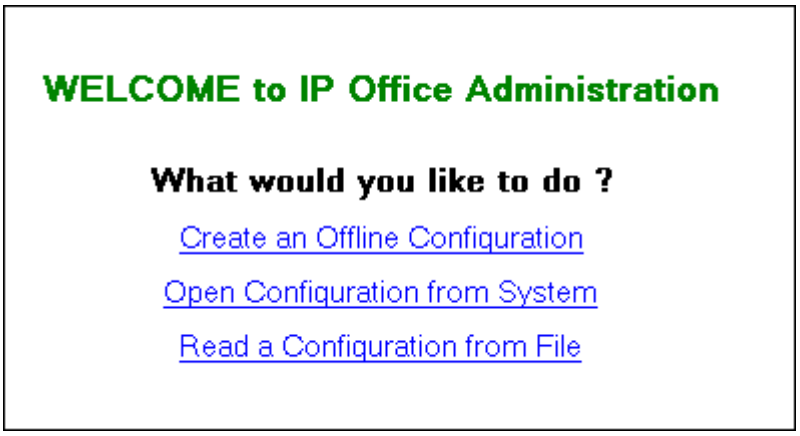
# 4. Equipment and Software Validated

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

| Component | Version |
|---|---|
| **Avaya** | |
| Avaya IP Office 500v2 | 8.0 (16) |
| Avaya IP Office Digital Expansion Module | 8.0 (16) |
| Avaya IP Office Manager | 10.0.16 |
| Avaya IP Office Voicemail Pro | 8.0.8.29 |
| Avaya 9620 IP Telephone (H.323) | Avaya one-X Deskphone Edition 3.1 |
| Avaya 1140 Telephone (SIP) | SIP1140 Load Ver: 04.03.09.00 |
| Avaya 9508 Digital Telephones | -- |
| Avaya 1408 Digital Telephones | -- |
| Avaya IP Office Softphone | 3.2.3.15_64595 |
| Avaya IP Office Phone Manager | 4.2.39 |
| **Service Provider** | |
| Nokia-Siemens HiQ switch | Rel. 14 |
| CS2K Gateways | CVM12 |

# 5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the TELUS SIP Trunk Service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running the Avaya IP Office Manager application, select **Start →Programs →IP Office →Manager** to launch the application. Navigate to **File →Open Configuration from System**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials.



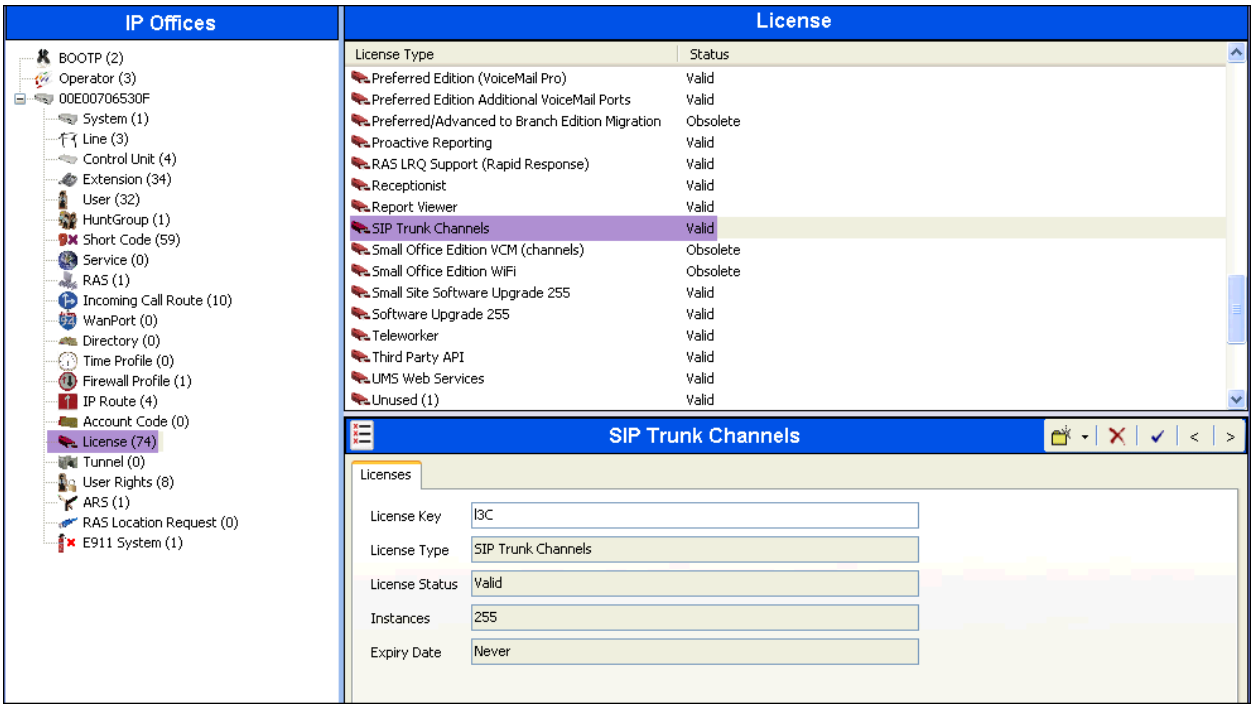

A management window will appear similar to the one shown in the next section.

The appearance of the Avaya IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interfacing with the service provider (such as LAN interface to the enterprise site, Twinning and Avaya IP Office Softphone support) is assumed to be already in place, and they are not part of these Application Notes.

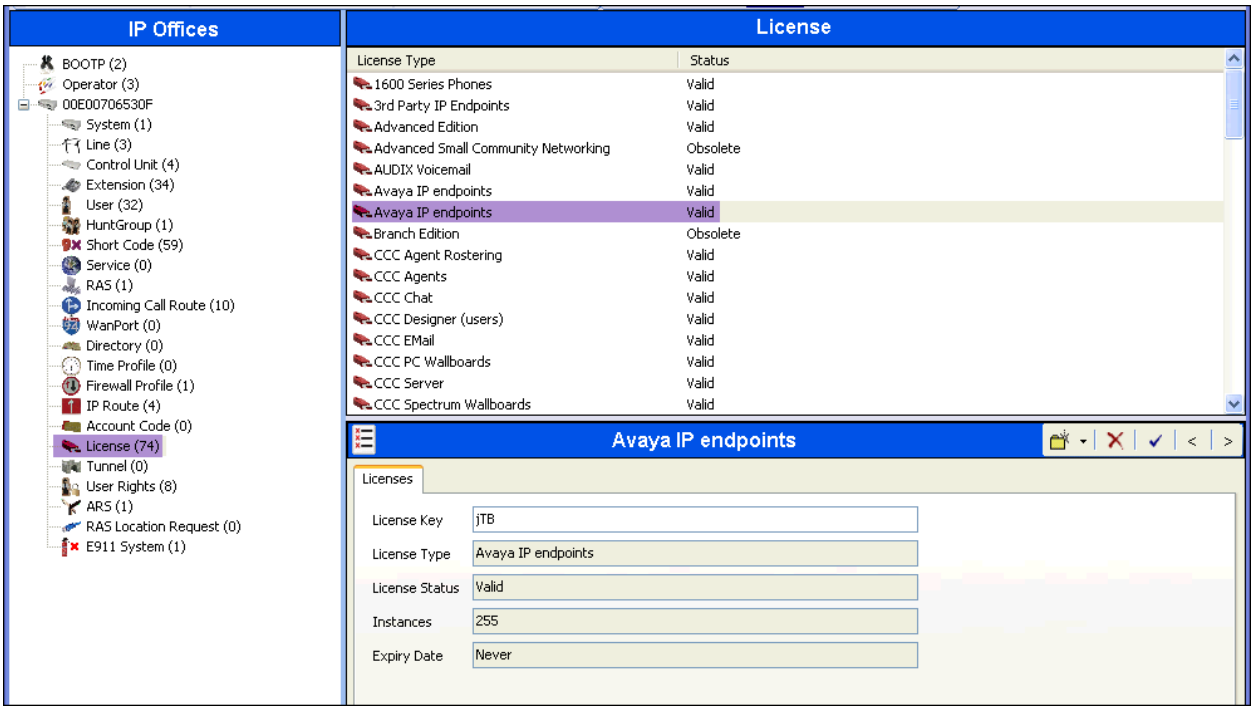
## 5.1. Licensing

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

To verify that there is a SIP Trunk Channels License with sufficient capacity; 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.



If Avaya IP Telephones will be used, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient "Instances" in the Details pane.



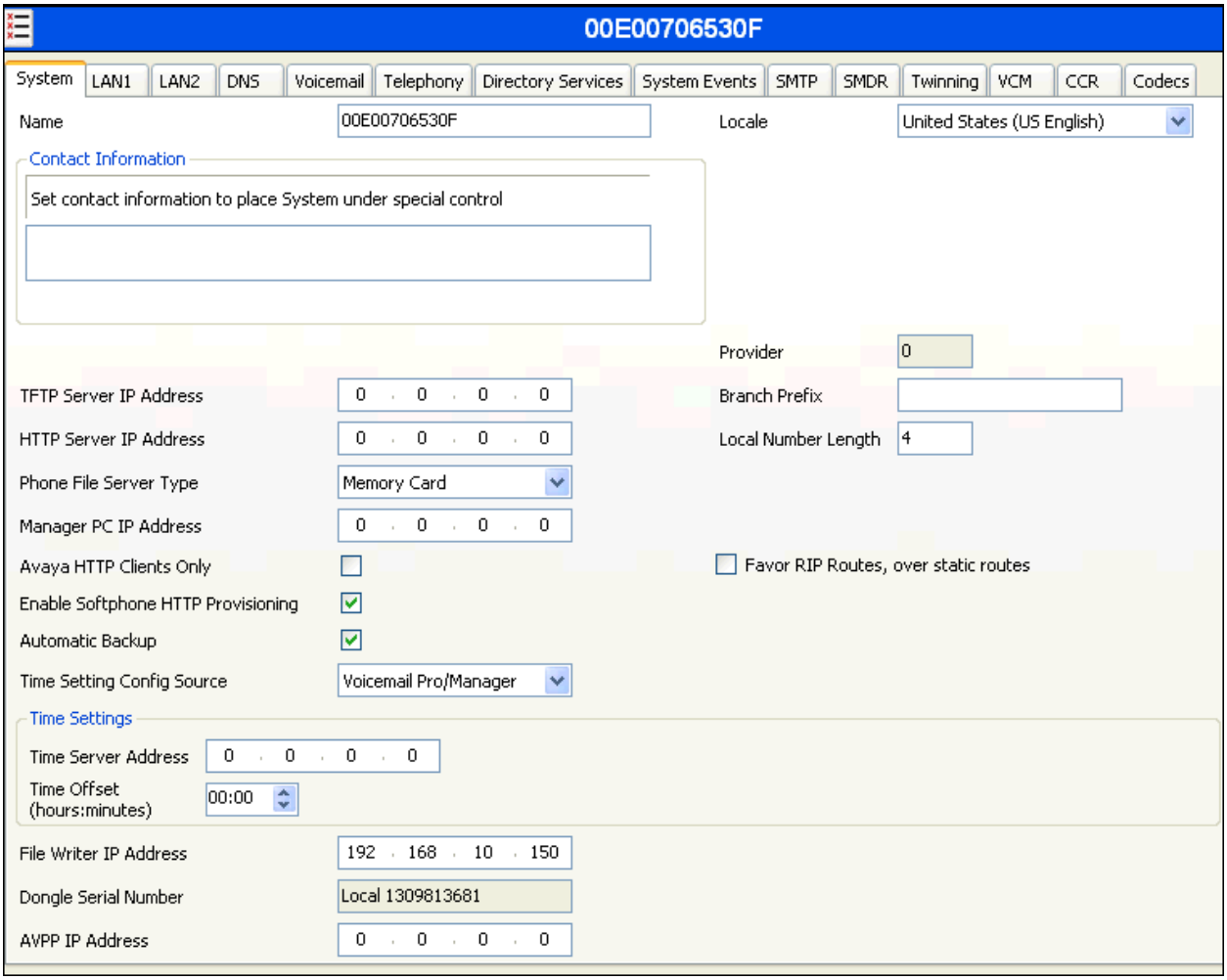

## 5.2. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane, and select the proper system name in the Group pane. Similar screens as shown in the following tabs will be presented. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings relevant to these Application Notes. Note that the **Codecs** tab on the far right is new in Avaya IP Office Release 8.

In the sample configuration, **00E00706530F** was used as the system name, **LAN1** was used to connect Avaya IP Office to the enterprise, the **WAN** port or **LAN2** was used to connect the Avaya IP Office to the public network.

### 5.2.1. System Tab

As shown in the following screen, the **Name** field can be used for a descriptive name of the system. In this case, the MAC address is used as the name. The **Enable SoftPhone HTTP Provisioning** box is checked to facilitate Avaya IP Office Softphone usage.

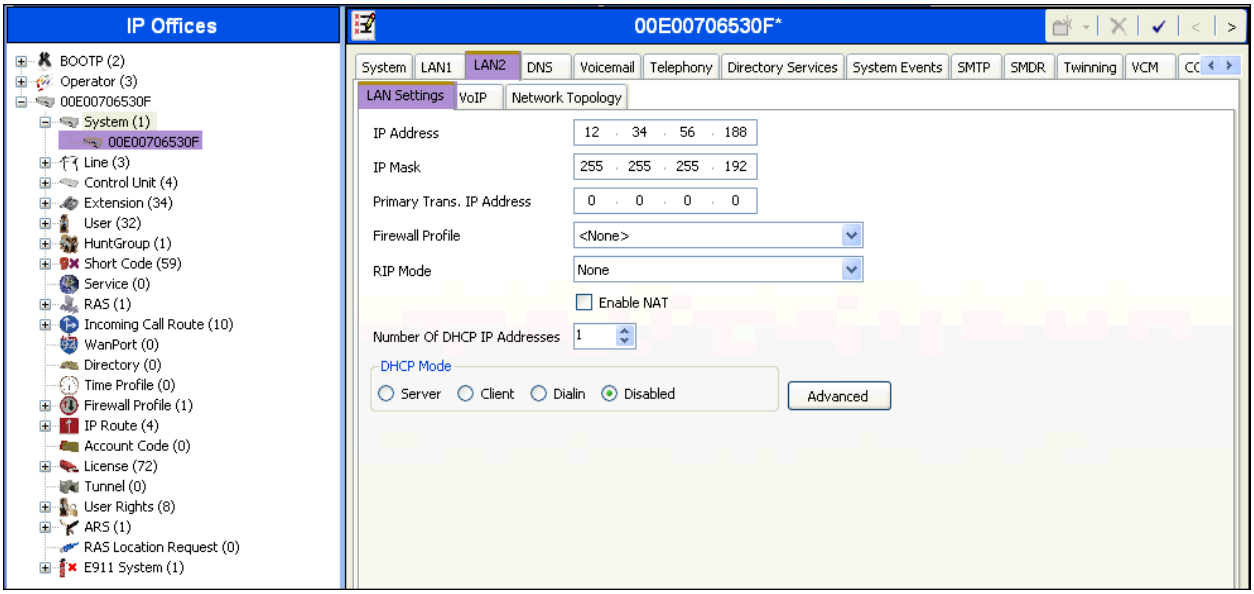



### 5.2.2. LAN1 Tab

In the sample configuration, **LAN1** (not shown) was used to connect the Avaya IP Office to the enterprise network. Other LAN choices (e.g., LAN2) may also be used. The **LAN1** interface configuration is not directly relevant to the interface with the TELUS SIP Trunk Service, and therefore is not described in these Application Notes.

### 5.2.3. LAN2 Tab

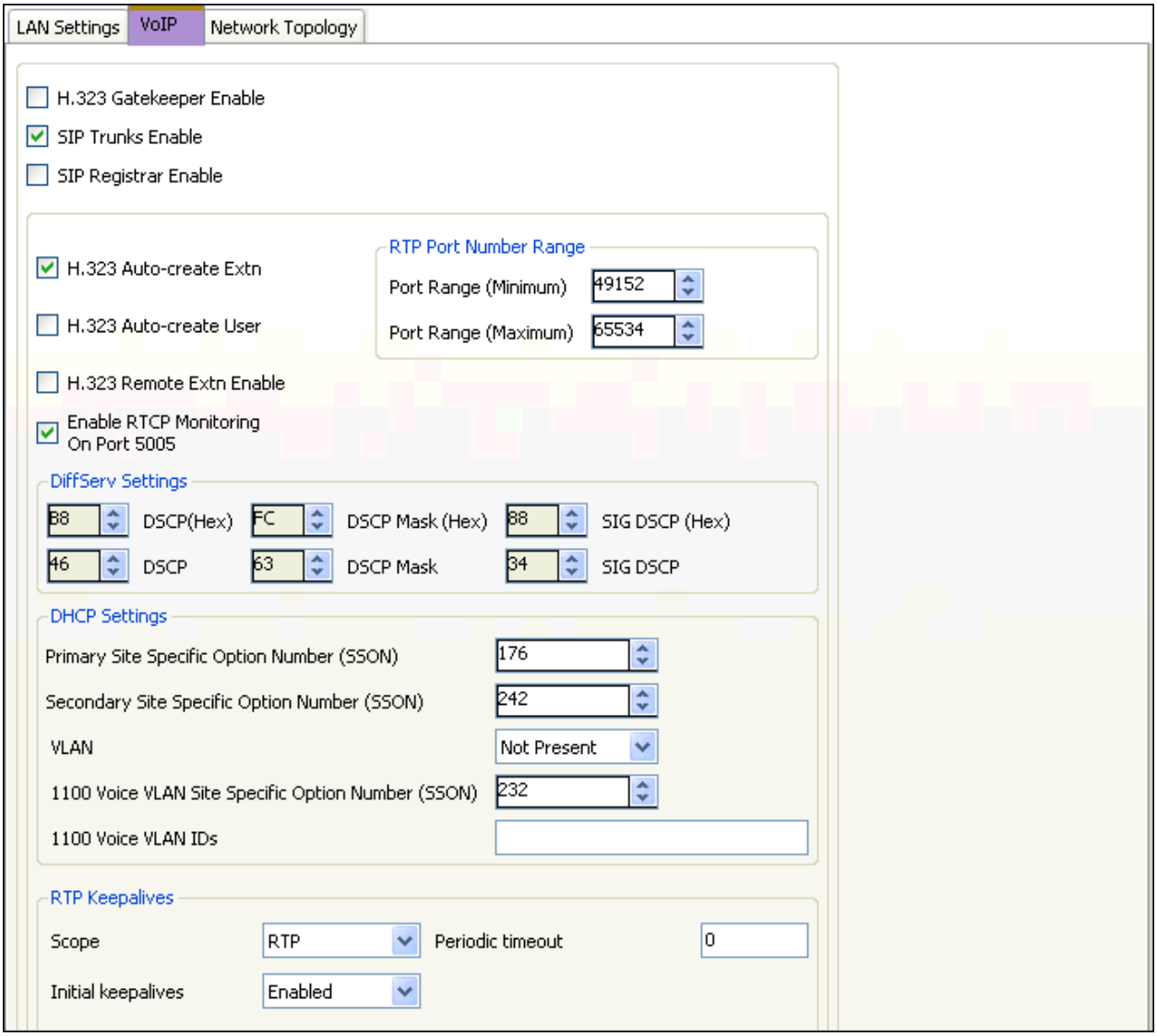
The **LAN2** settings correspond to the **WAN** port on the Avaya IP Office. To access the **LAN2** settings, first navigate to **System (1) →00E00706530F** in the Navigation and Group panes and then navigate to the **LAN2→LAN Settings** tab in the details pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office **WAN** port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.




On the **VoIP** tab in the Details pane (Navigation Pane not shown), 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 the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using **LAN2**. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below.

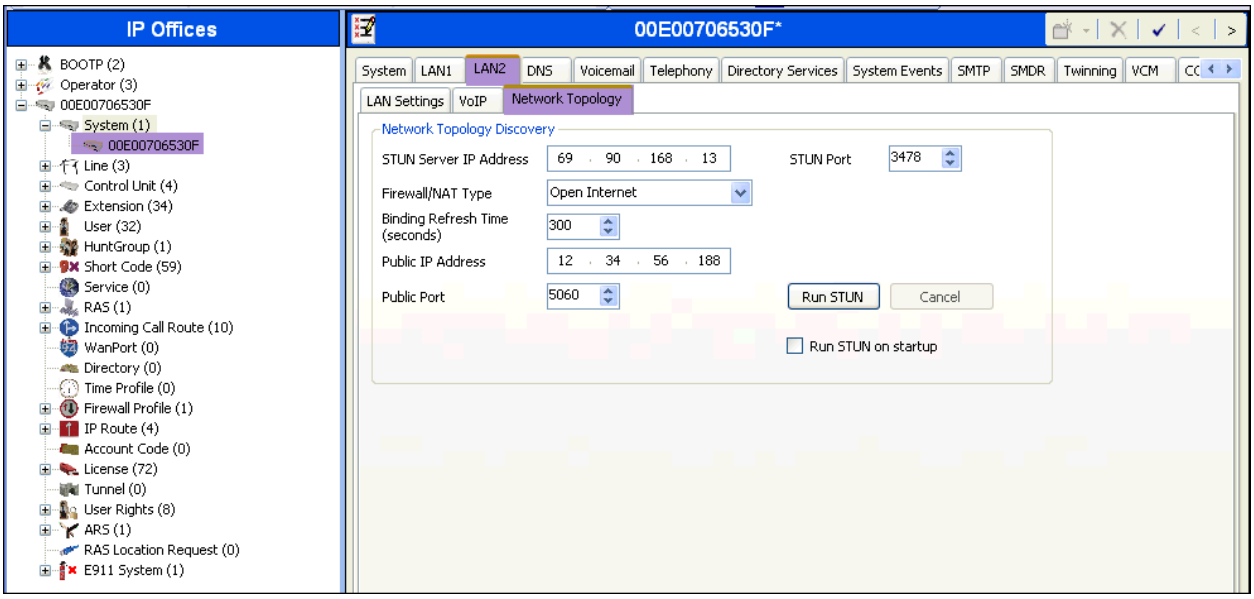
In the **RTP Keepalives** section at the bottom of the page, set the **Scope** field to **RTP**, and **Initial keepalives** to **Enabled**. This will cause the Avaya IP Office to send RTP keepalive packets at the beginning of the calls, to avoid problems of media deadlock that can occur with certain types of forwarded calls that are routed from the Avaya IP Office back to the network, over the same SIP trunk.

All other parameters should be set according to customer requirements.



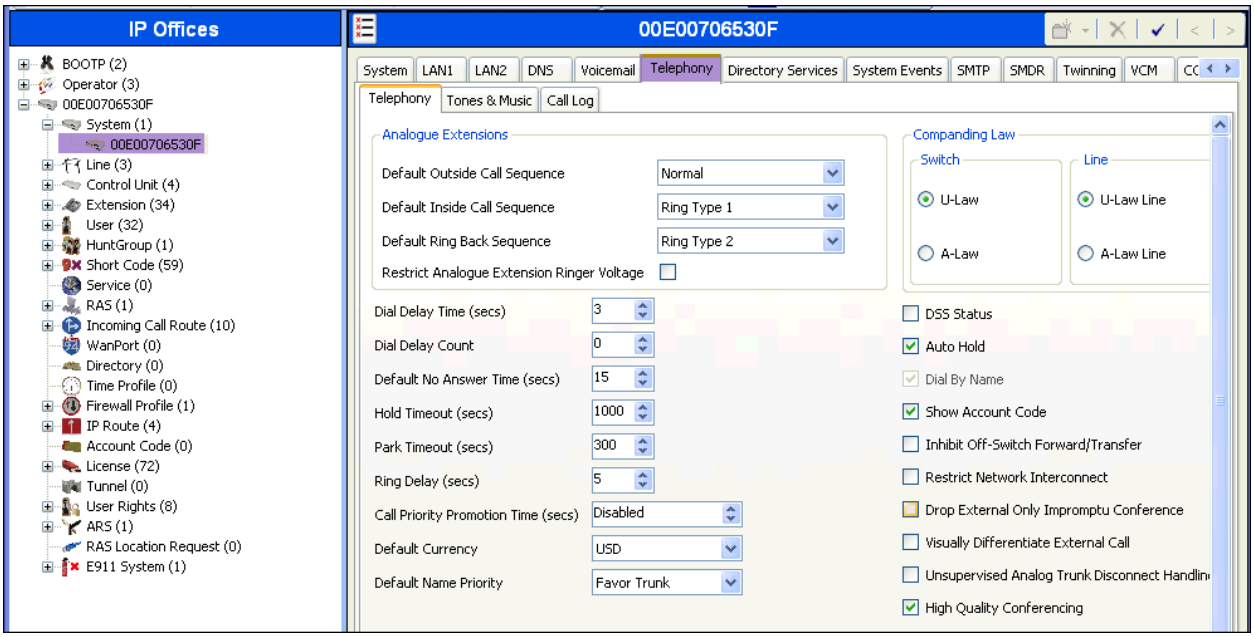

On the **Network Topology** tab in the Details pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu to the option that matches the network configuration. No network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**. With this configuration, the **STUN Server IP Address** and **STUN Port** are not used.
- Set **Binding Refresh Time (seconds)** to **300.** This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTION messages to the service provider.
- Set **Public IP Address** to the IP address that was set for LAN2.
- Set **Public Port** to **5060**.
- All other parameters should be set according to customer requirements




## 5.2.4. Telephony Tab

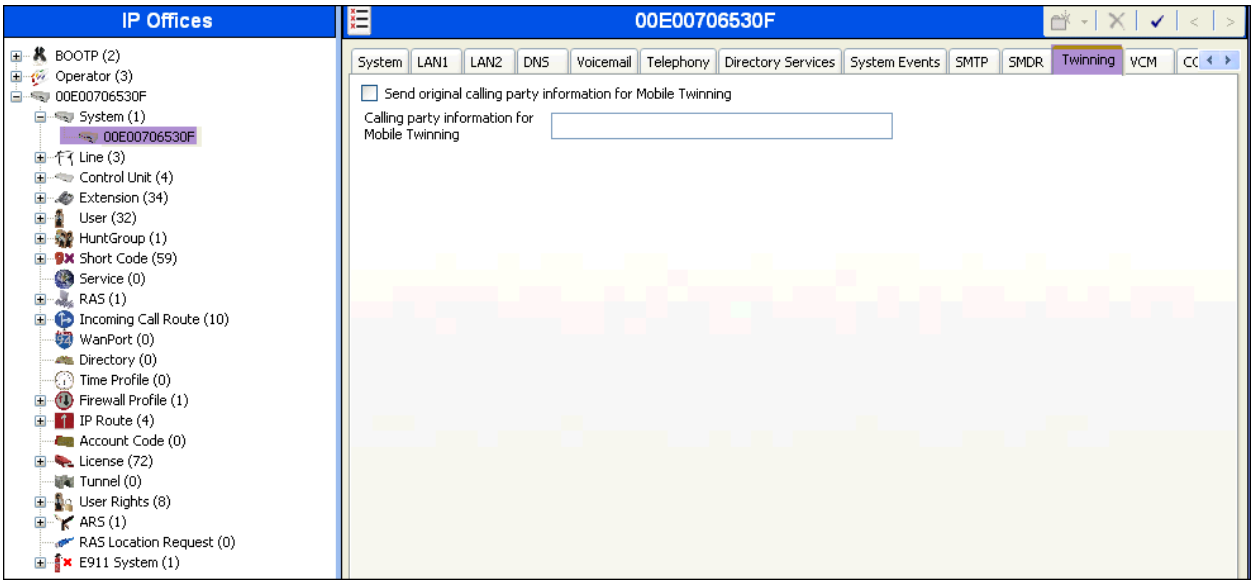
Navigate to the **Telephony →Telephony** Tab in the Details Pane. Set the **Automatic Codec Preference** for the default codec to be used for intra-enterprise traffic. Choose the **Companding Law** typical for the enterprise location. In North America, **U-LAW** is used. Uncheck the **Inhibit Off-Switch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




## 5.2.5. Twinning Tab

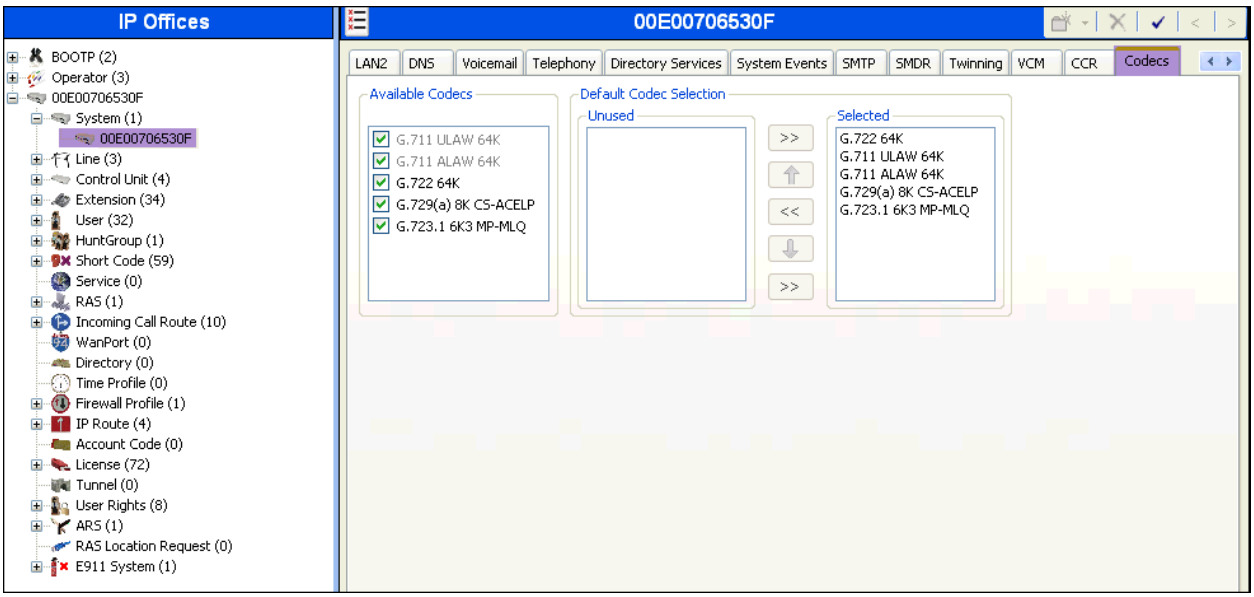
Navigate to the **Twinning** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be

controlled by the setting on the SIP Line (**Section 5.4**). This setting also impacts the Caller ID for call forwarding.



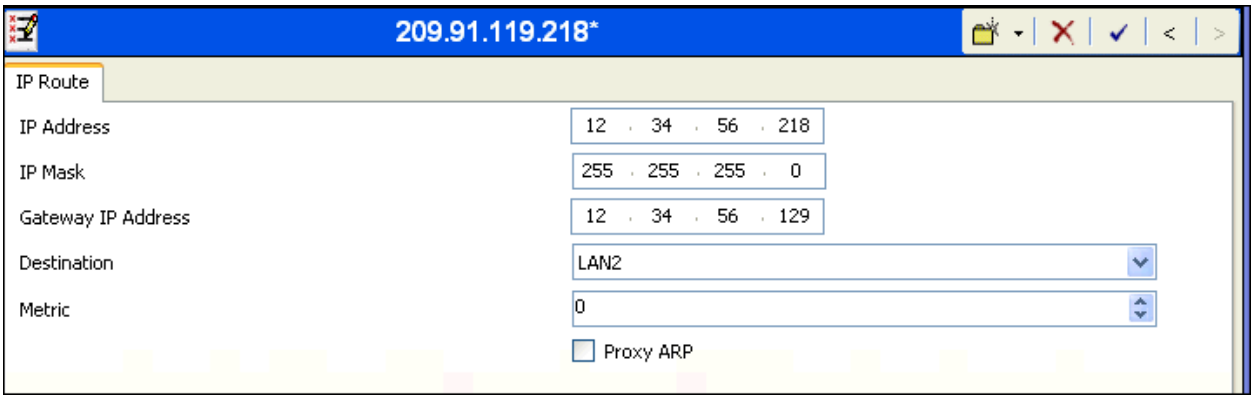

## 5.2.6. Codecs Tab

The **System →Codecs** tab is new in Avaya IP Office Release 8. The list of **Available Codecs** shows all the codecs supported by the system, and those selected as usable. 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 the **Selected** lists, and to change the order of the codecs in the **Selected** codecs list. By default, all IP (SIP and H.323) lines and extensions will use this system default codec selection, unless configured otherwise for a specific line or extension.



## 5.3. IP Route

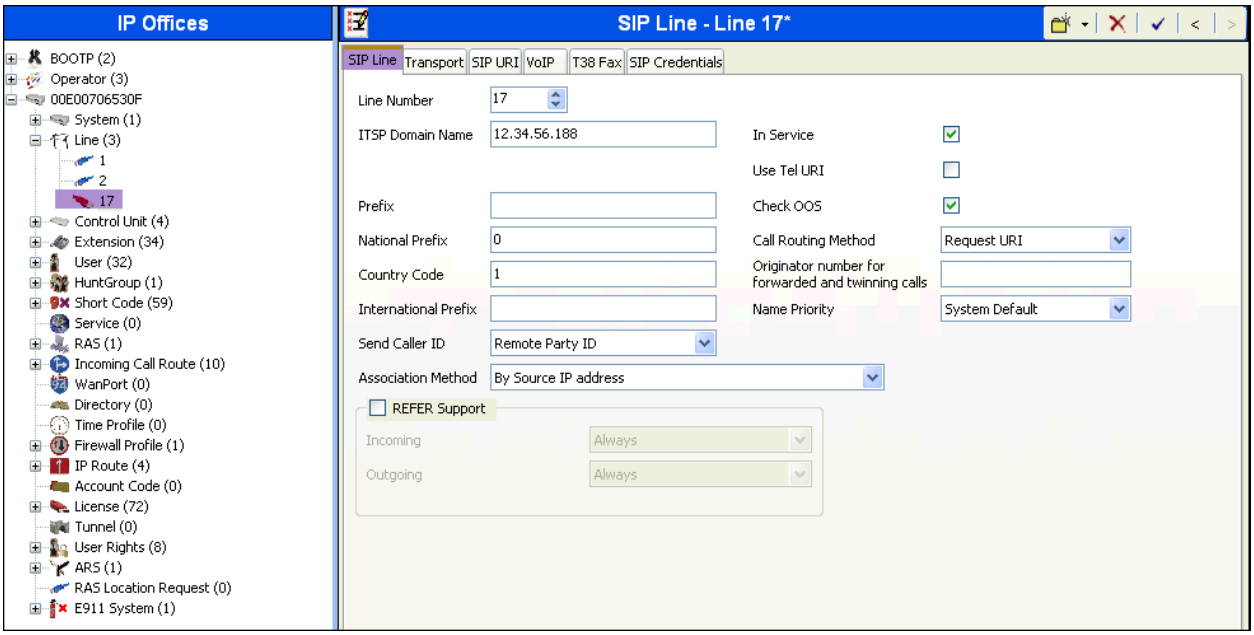
Create an IP route to specify the gateway or router address where the Avaya IP Office needs to send the packets, in order to reach the SIP proxy on the TELUS network. On the left navigation pane, right-click on **IP Route** and select **New**. The values used during the compliance test are shown on the screen below. (See **Figure 1**)




## 5.4. Administer SIP Line

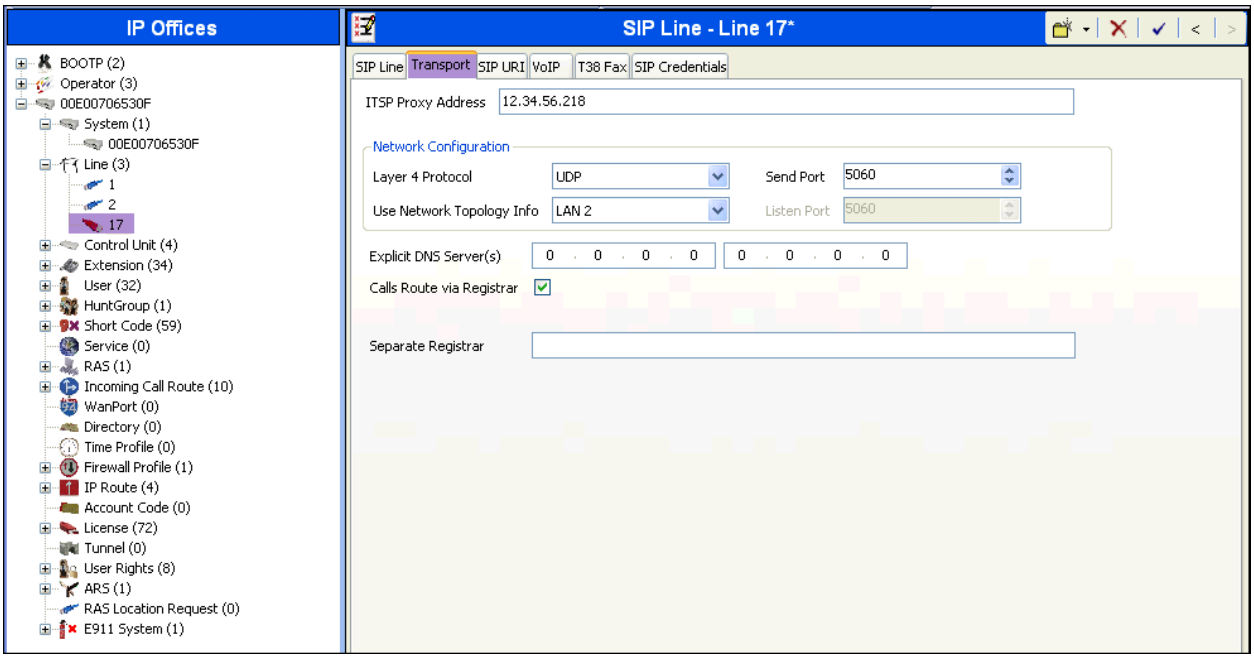
A SIP line is needed to establish the SIP connection between Avaya IP Office and the TELUS SIP Trunk Service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New →SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the **SIP IP Address** of **LAN2**, this IP Address is 12.34.56.188. Avaya IP Office uses this field on the host portion in SIP headers such as the "From" header.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, Avaya IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Set **Send Caller ID** to **Remote Party ID**.
- Uncheck the **REFER Support** box. At the time of the compliance test, for the practical purposes of network redirection, the REFER message was not supported by TELUS. If support for REFER is to be enabled, this box should be checked here.
- Default values may be used for all other parameters.



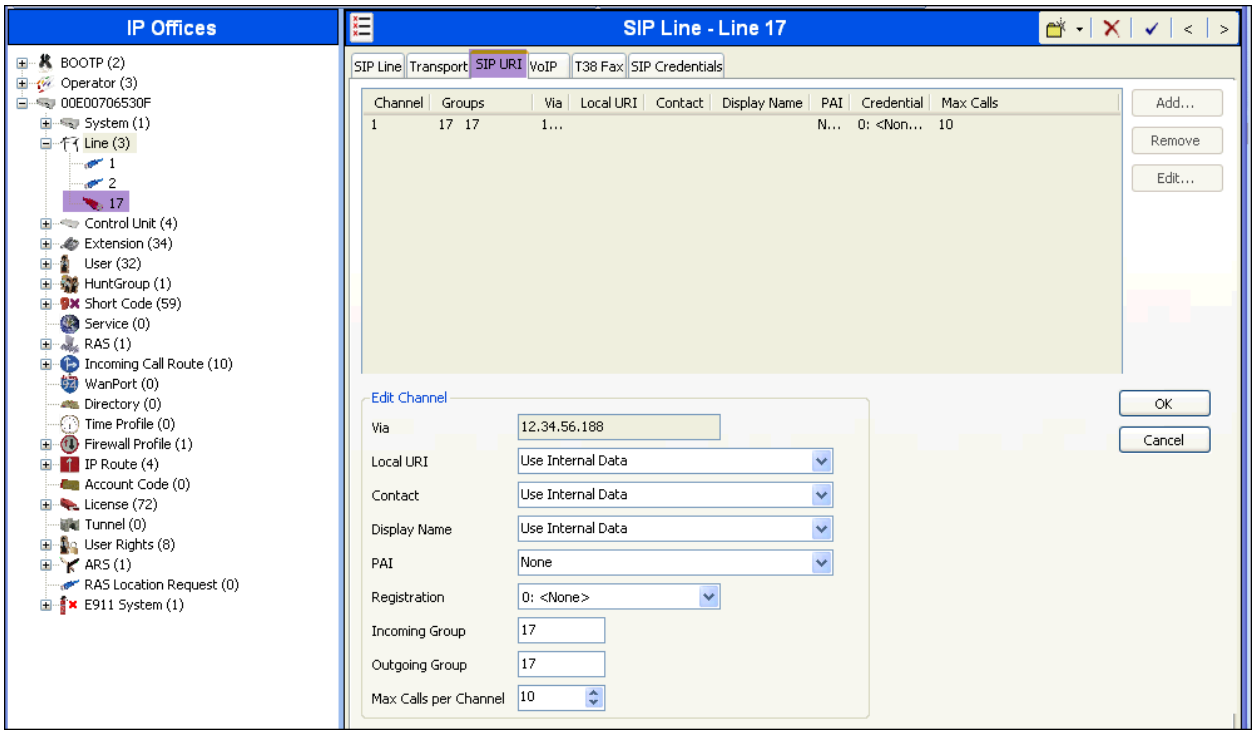

Select the **Transport** tab and set the following:

- Set the **ITSP Proxy Address** to the IP address of the TELUS proxy server.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN2** as configured in **Section 5.2.3**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.



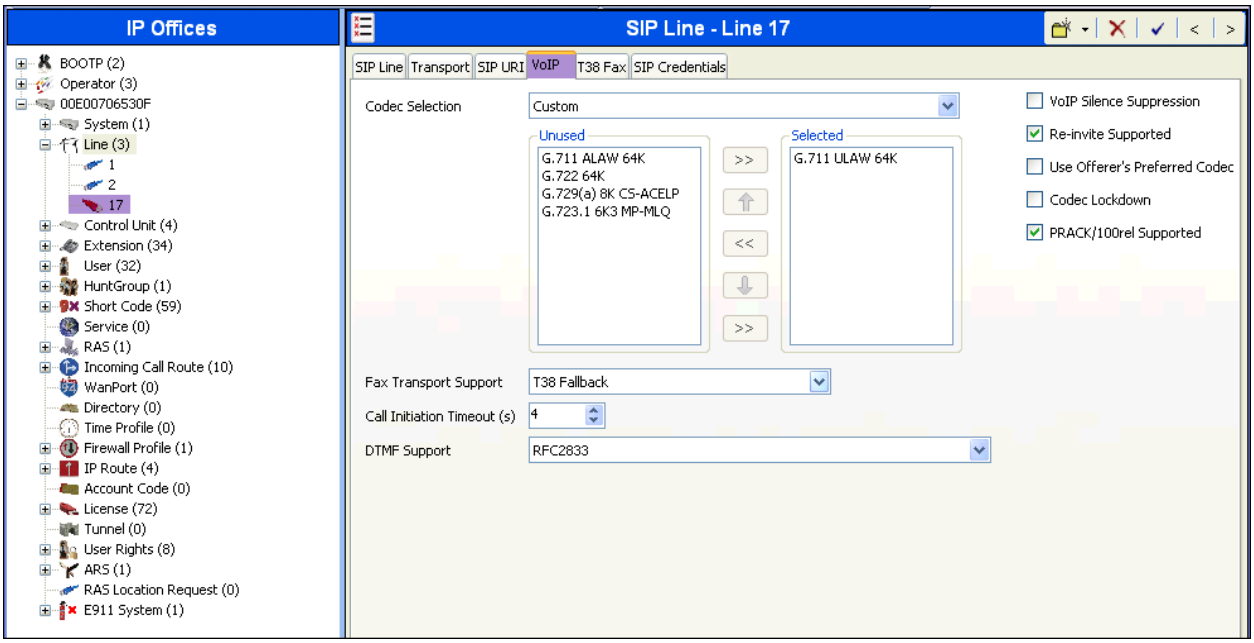
Select the **SIP URI** tab. A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Under the **SIP URI** tab, click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit** button. 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 following parameters:

- Set **Local URI**, **Contact** and **Display Name** to **Internal Data**. This setting allows calls on this line with SIP URI matching the number set in the **SIP** tab of any **User** as shown in **Section 5.6**.
- Set **PAI** to **None**. The **PAI** parameter was introduced in Avaya IP Office Release 6.1, and the value "None" is shown selected from the drop-down menu. With PAI set to "None", Avaya IP Office Release 6.1 and 7.0 will behave like Avaya IP Office Release 6.0 with respect to the SIP P-Asserted-Identity header (e.g., Avaya IP Office will not include a PAI header for an outbound call unless privacy is asserted).
- **Registration** parameter is set to the default **0: <None>** since TELUS SIP Trunk service does not require registration.
- Associate this line with an incoming line group 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. For the compliance test, a new incoming and outgoing group **17** was defined.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

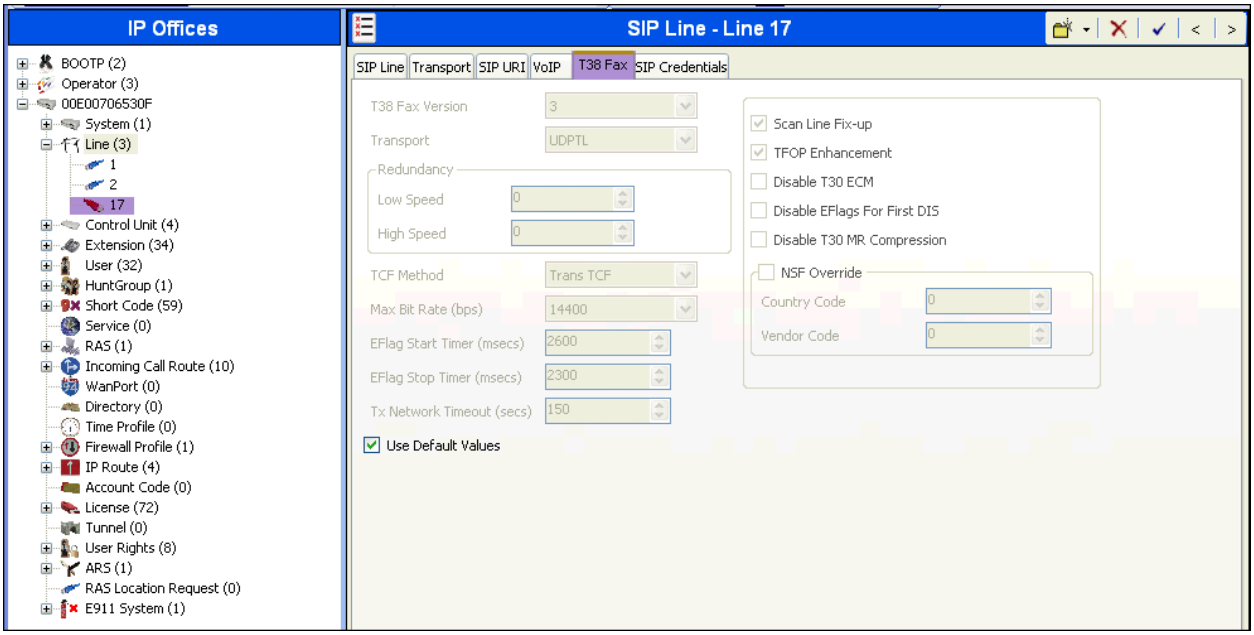


Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- In the sample configuration, the **Codec Selection** was configured using the "Custom" option, allowing an explicit ordered list of codecs to be specified, different from the system default defined in **Section 5.2**. The buttons allows configuration of an explicit list of codecs to be used on the line, in that specific order of preference. Only G.711u was used for the compliance testing
- Uncheck the **VoIP Silence Suppression** box.
- 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 **PRACK/100rel Supported**, this field is new in Avaya IP Office Release 8. It's used for early media support. With this field checked Avaya IP Office will advertise support for early media; Avaya IP Office will also acknowledge 183 messages with a PRACK response.
- Default values may be used for all other parameters.
- Under **T.38 Fax Transport Support** Select **T.38 Fallback;** this field is new in Avaya IP Office Release 8. With this setting outgoing fax calls will use T38 fax but when the called destination rejects the call with failures 488, 415 or 606, a re-invite it sent for fax transport over G.711. Incoming audio calls that detect fax tones also initiate fax transport using T38 Fallback. If there is an established G.711 call before T.38 fax is initiated, the G.711 call is reused when fallback to G.711 fax occurs.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- All other fields may retain their default values.



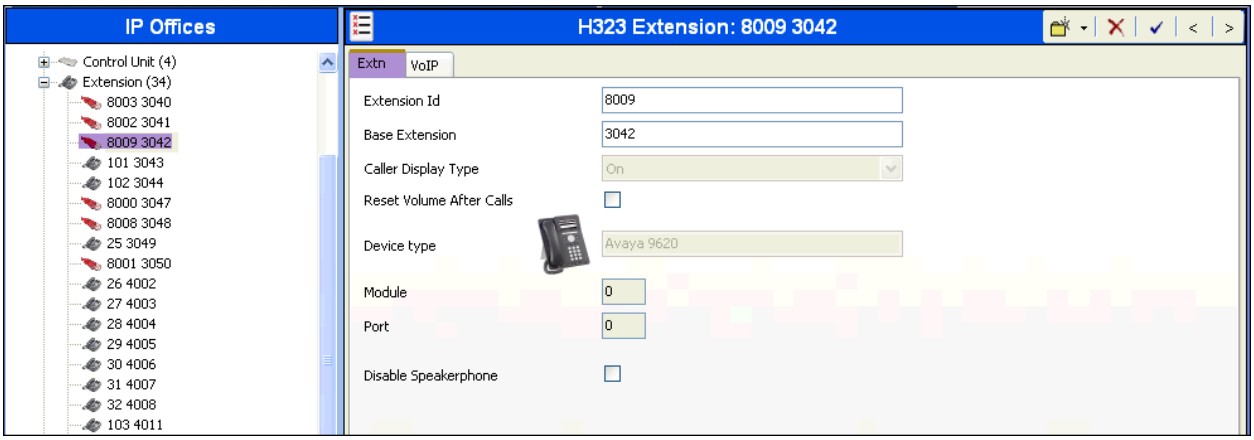
Select the **T.38 Fax** tab, check **Use Default Values.**



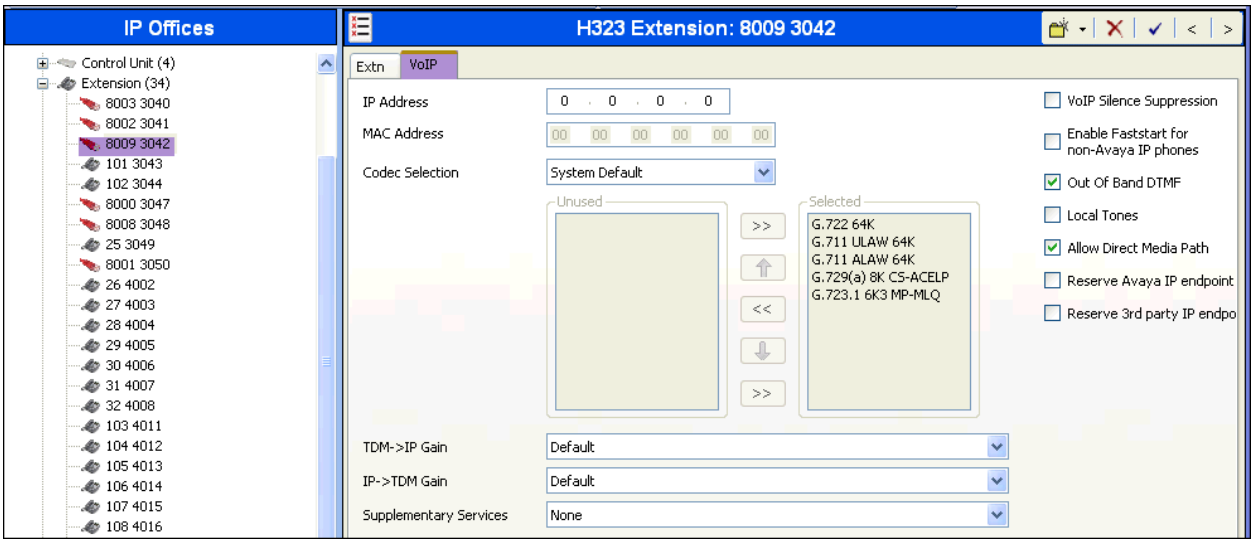

## 5.5. Extension

In this section, examples of Avaya IP Office Extensions will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. 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 3042; this extension corresponds to an H.323 extension.

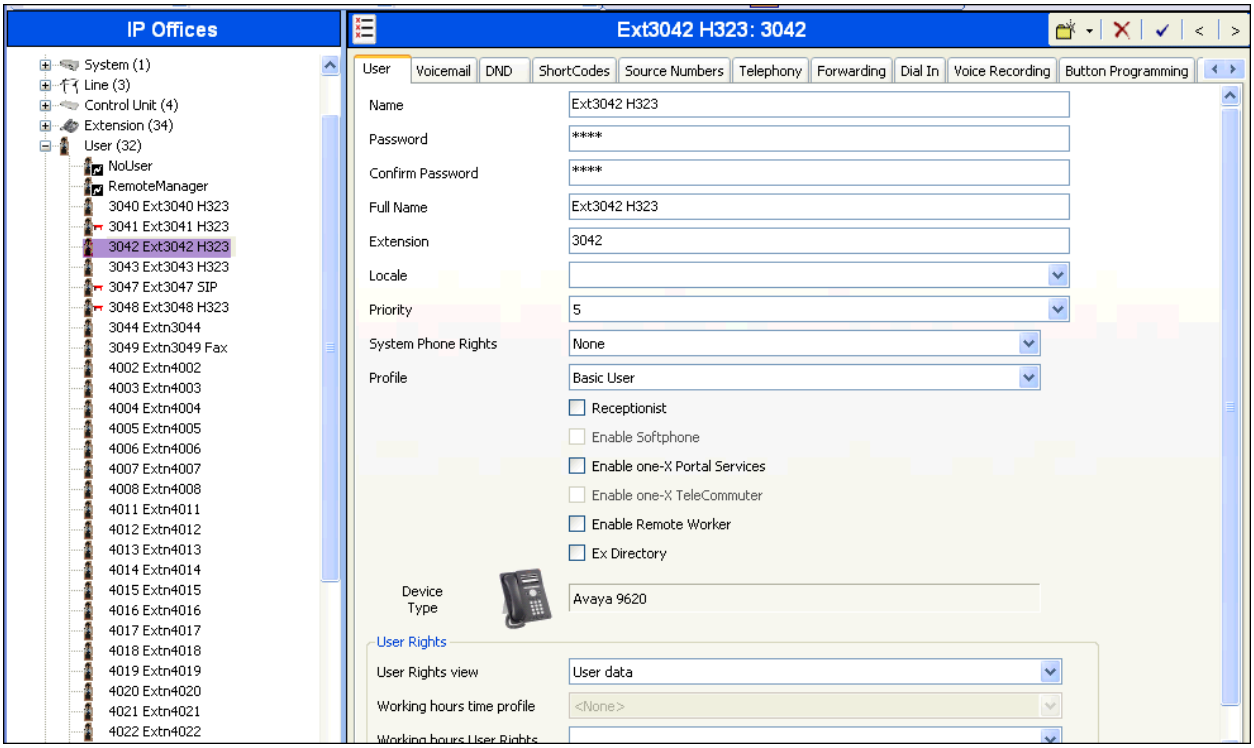



Select the **VOIP** tab. Use default values on VoIP tab. Following is an example for Extension 3042; this extension corresponds to an H.323 extension.

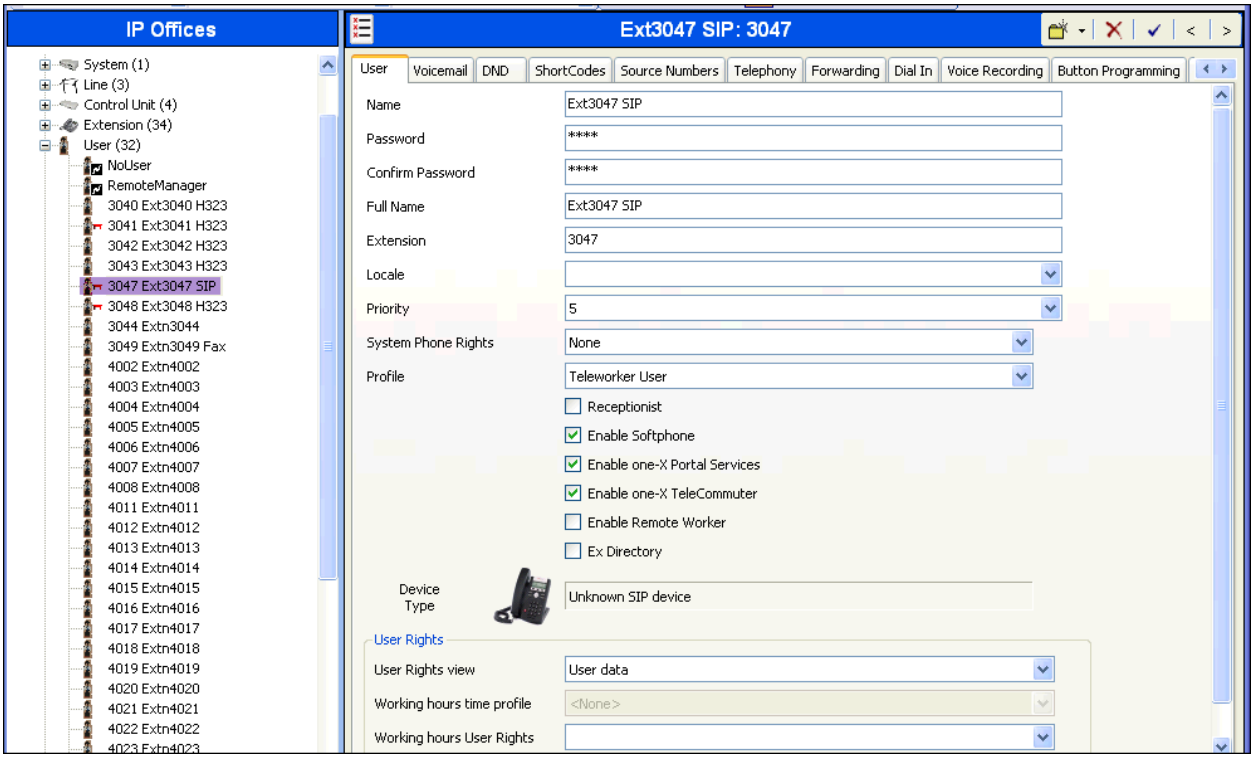



## 5.6. User

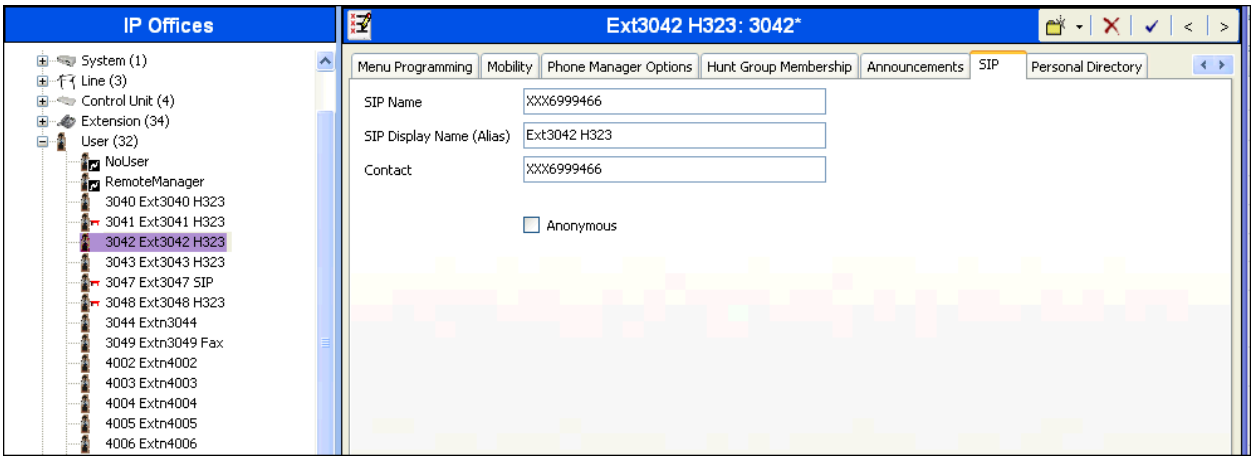
Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. To configure these settings, first expand **User** in the left Navigation Pane, and then select the name of the user to be modified. In the example below, the name of the user is "Ext3042 H323", an Avaya 9620 IP Telephone (H.323).




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

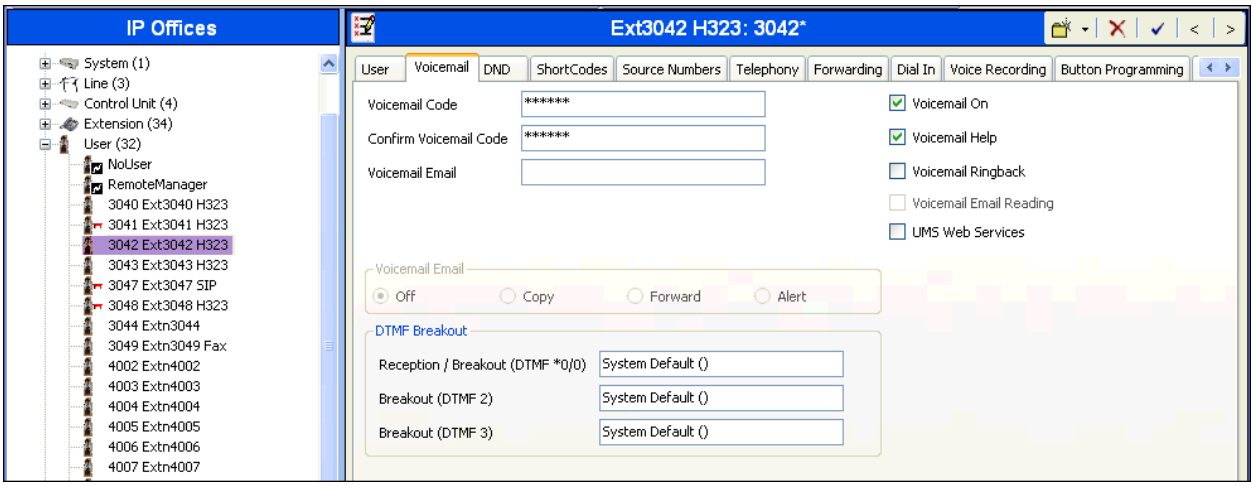



Select the **SIP** tab. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the "From" header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.4**). The example below shows the settings for user Ext3042 H323. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise. 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.

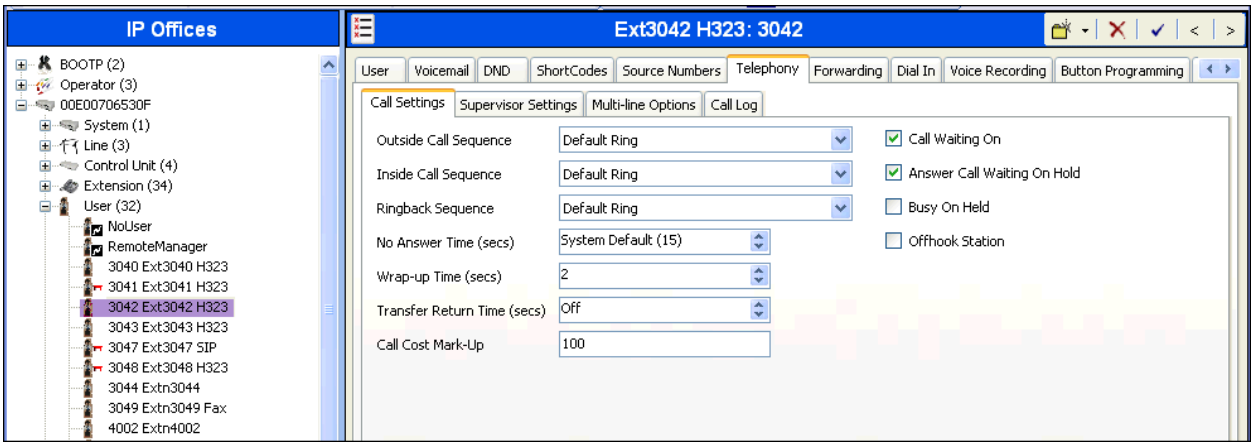



Select the **Voice Mail** tab. The following screen shows the **Voicemail** tab for the user with extension 3042. The **Voicemail On** box is checked. Voicemail password can be configured using

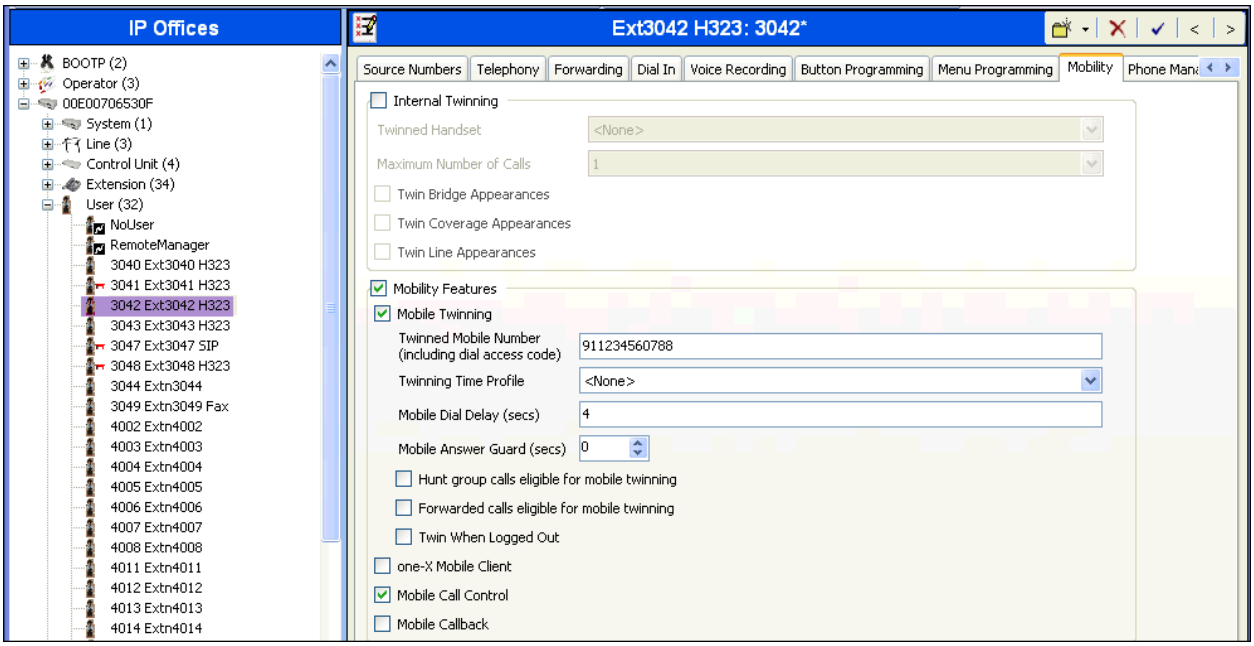
the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from TELUS SIP Trunk to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones to test DTMF using RFC 2833.



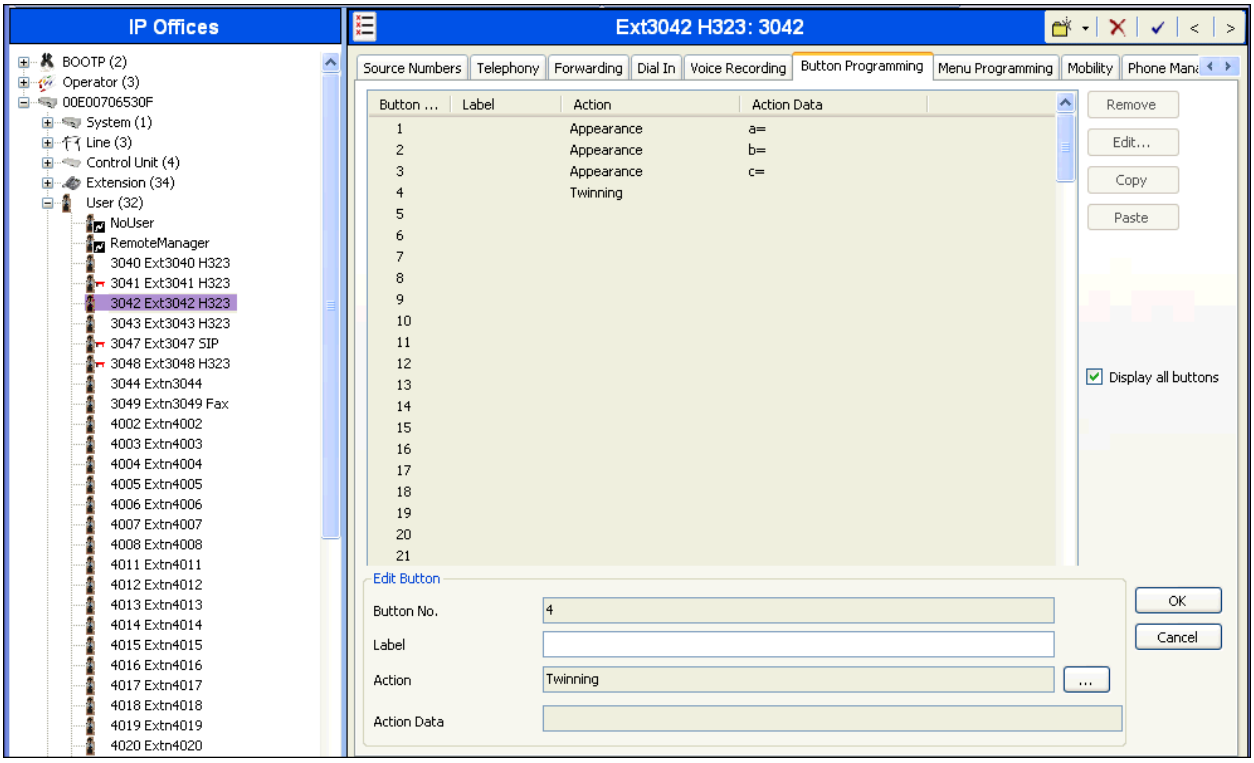

Select the **Telephony** tab, then **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an Avaya IP Office phone logged in as this extension to have multiple call appearances. Note: **Call Waiting On** is also necessary for call transfer.




Select the **Mobility** tab. In the sample configuration user 3042 was one of the users configured to test the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 3042. 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, in this case **911234560788**. Other options can be set according to customer requirements.

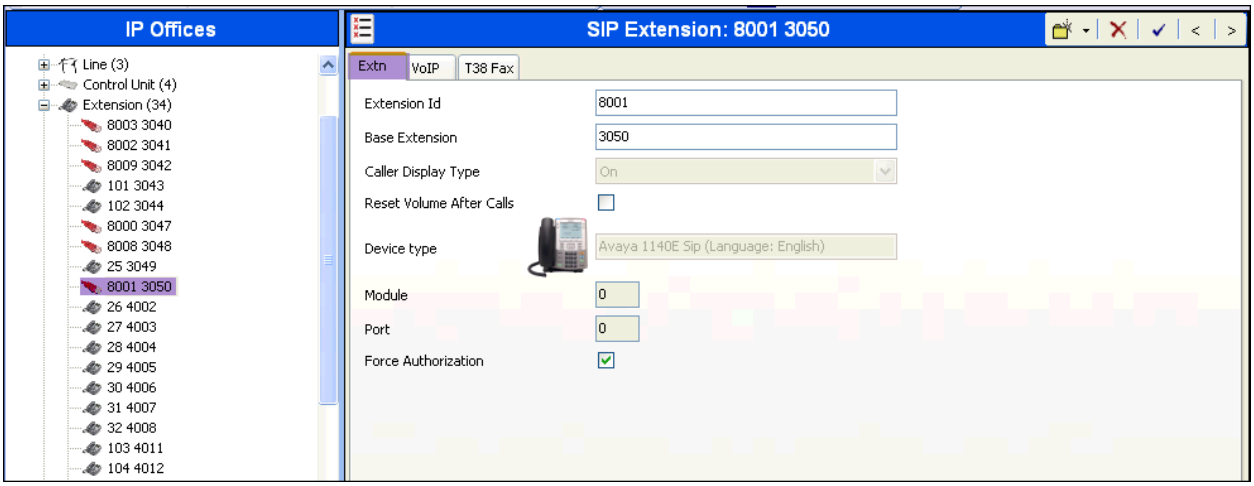



To program a key on the telephone to turn Mobil Twinning on and off, select the **Button Programming** tab on the user, then select the button to program to turn Mobil Twinning on and off, click on **Edit → Emulation → Twinning**. In the sample below, button 4 was programmed to turn Mobil Twinning on and off on user 3042.

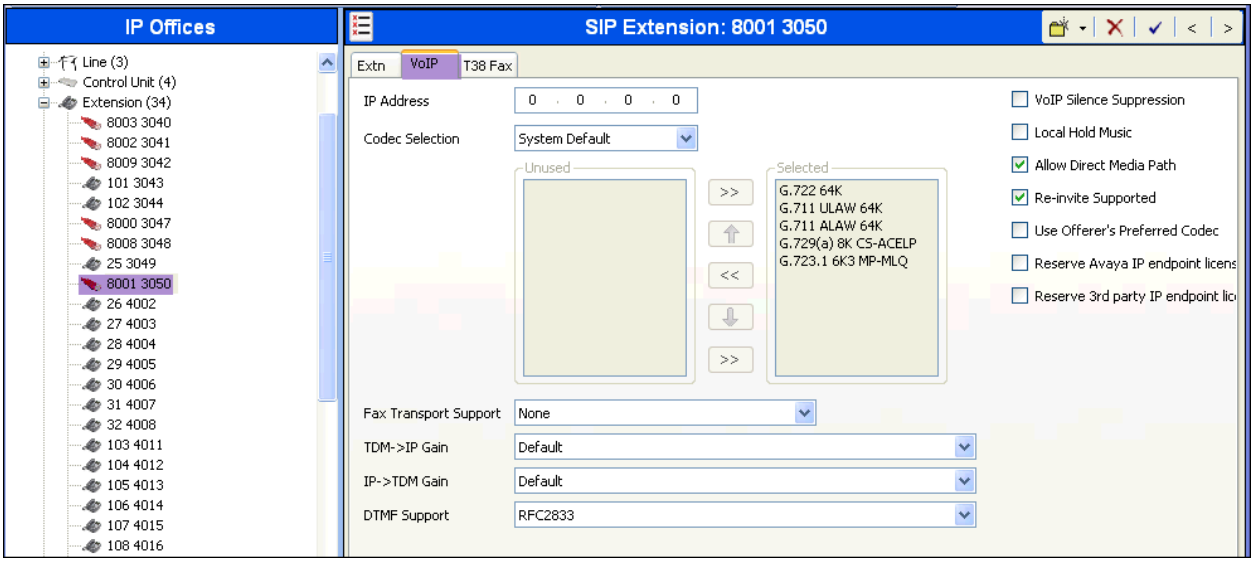


## 5.7. SIP Telephone Users (Avaya 1140E)

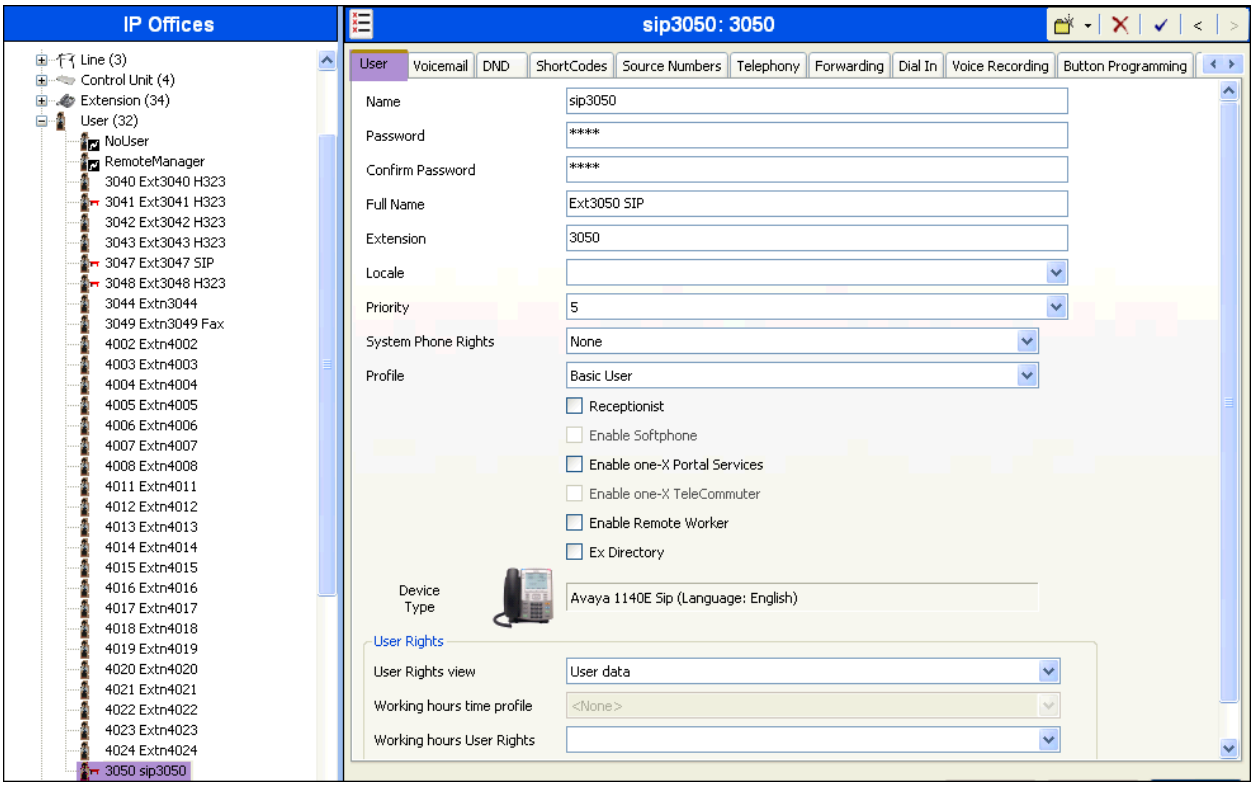
This section will summarize aspects of the completed configuration for the Avaya 1140E (the Avaya 1120 may also be use). A new SIP extension may be added by right-clicking on **Extension** in the Navigation pane and selecting **New SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1140E. The **Base Extension** field is populated with 3050, the extension assigned to the Avaya 1140E. Ensure the **Force Authorization** box is checked.



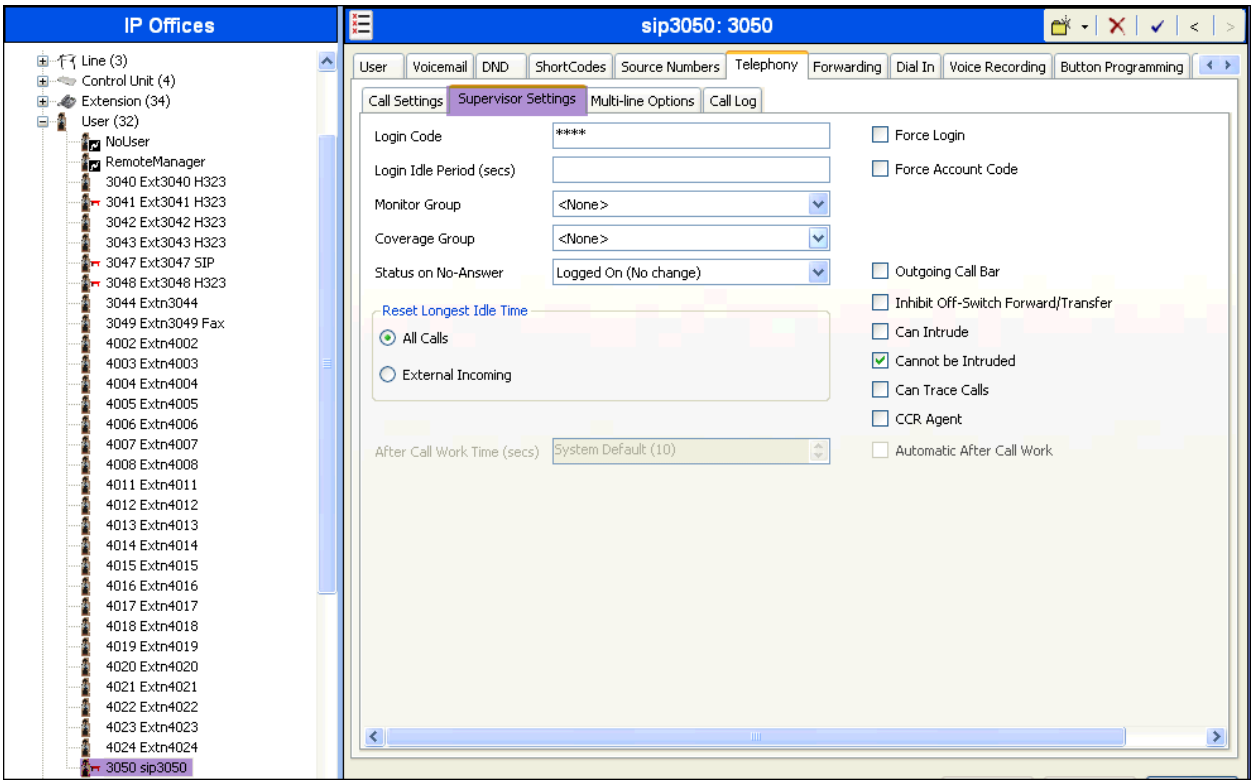

The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. The new **Codec Selection** parameter may retain the default setting "System Default" to follow the system configuration shown in Section **5.2.6.** Alternatively, "Custom" may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs. Other fields may retain default values.



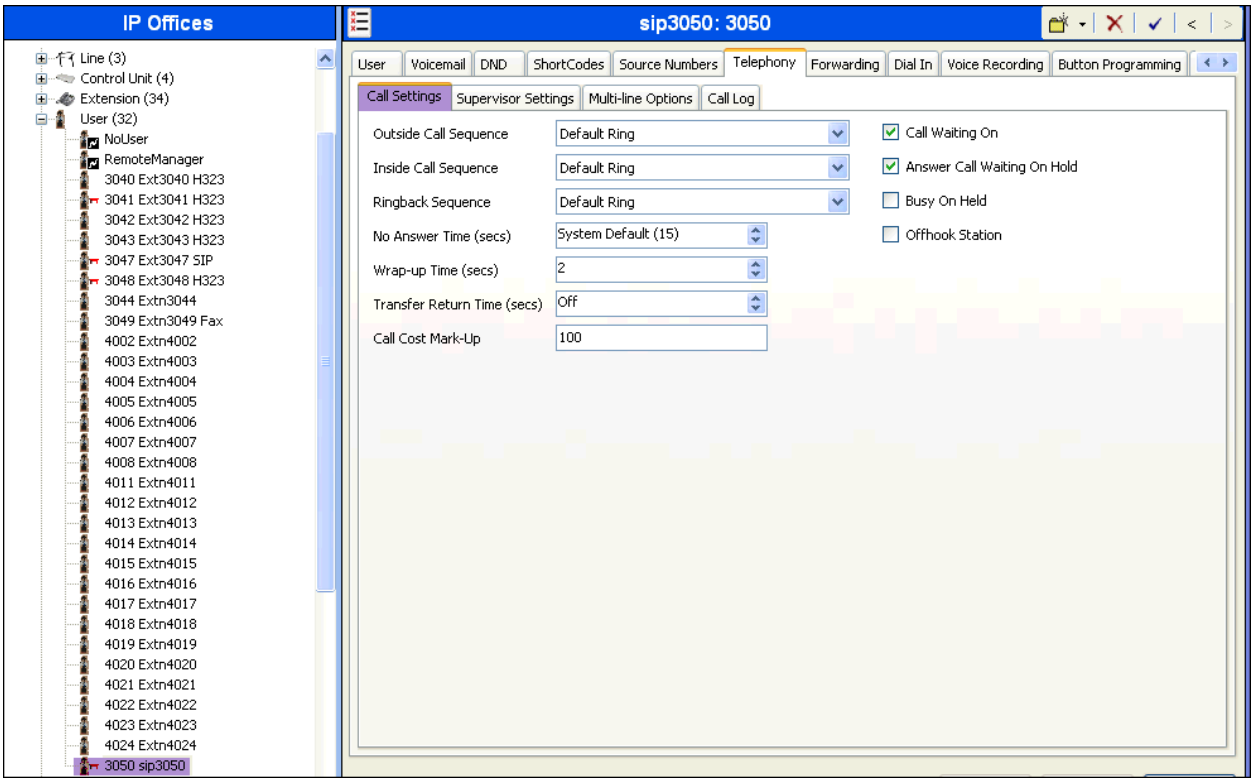
The following screen shows the **User** tab for User 3050 corresponding to an Avaya 1140E. The **Extension** parameter is populated with extension 3050.



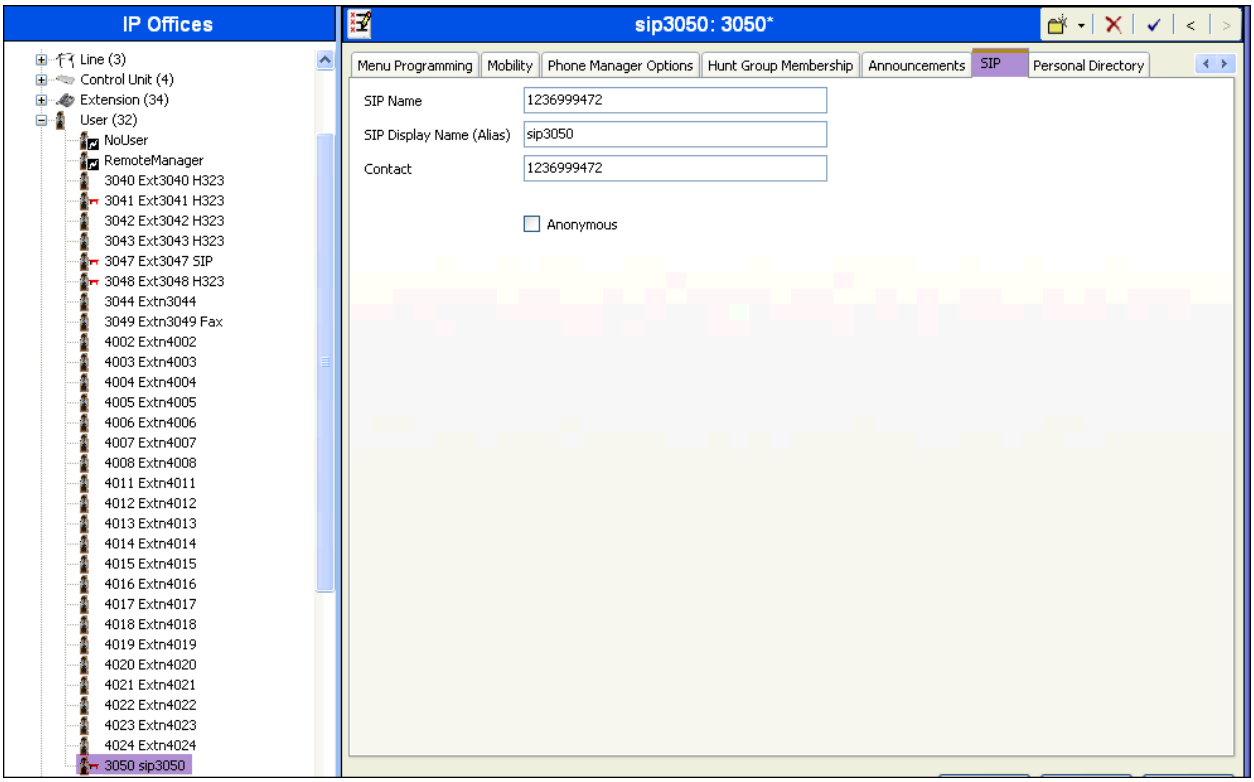
Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Avaya 1140E telephone user as the login password.



Remaining in the **Telephony** tab for the user, select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow multiple call appearances and call transfer operations.



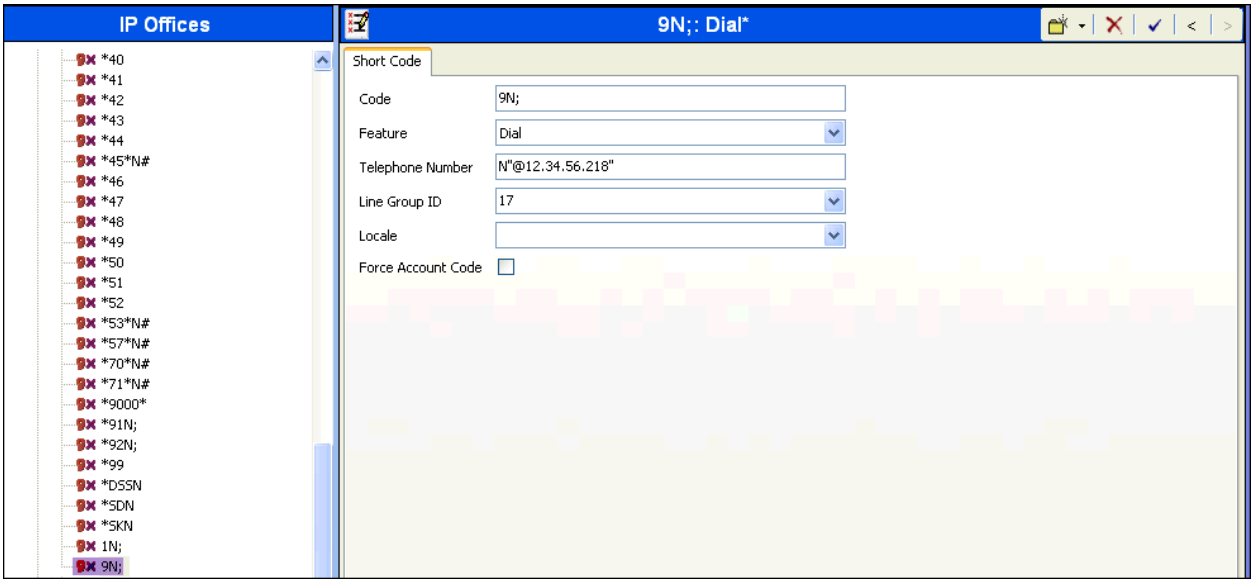
Like other users previously illustrated, the **SIP** tab for the user with extension 3050 is configured with a **SIP Name** and **Contact** specifying the user's TELUS IP Trunk service DID number.




## 5.8. Short Code

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case ***9N;*** this short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@12.34.56.218"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. The IP address of the TELUS SIP proxy server follows the @ sign in the above expression.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4**. This short code will use this line group when placing outbound calls.
- Default values may be used for all other parameters.

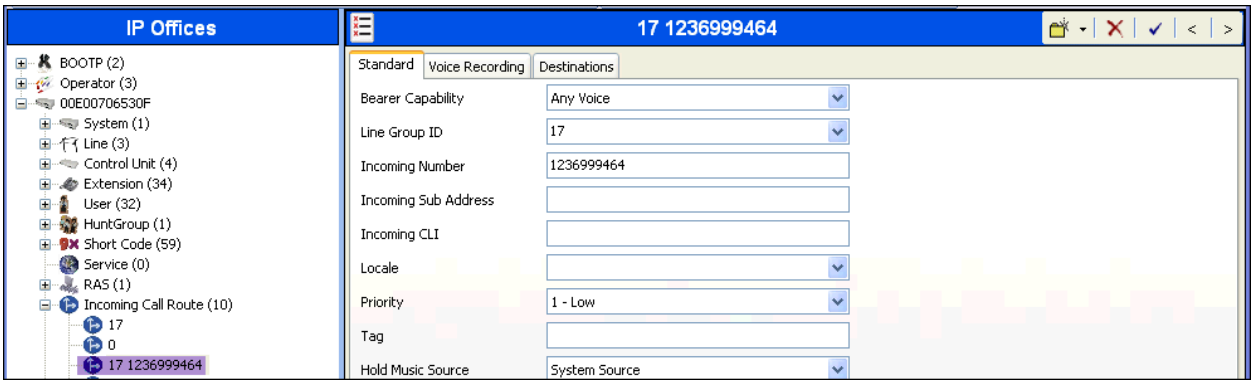



The simple "9N;" short code illustrated above, although effectively routes the outbound calls to the SIP trunk, does not provide the means of alternate routing if the configured SIP Line is out of service or temporarily unavailable. When alternate routing options and/or more customized features and analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. ARS was implemented and tested during the compliance tests, but its configuration is beyond the scope of these Application Notes.

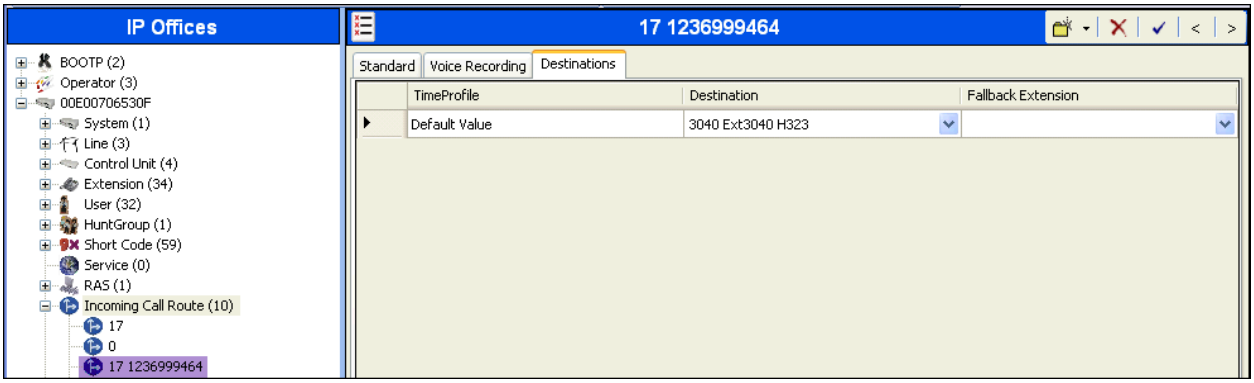
## 5.9. Incoming Call Routing

An incoming call route maps an inbound DID number on a specific line to an internal extension, hunt group, auto attendant, etc. in the Avaya IP Office. To create an incoming call route, right-click **Incoming Call Routes** in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

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



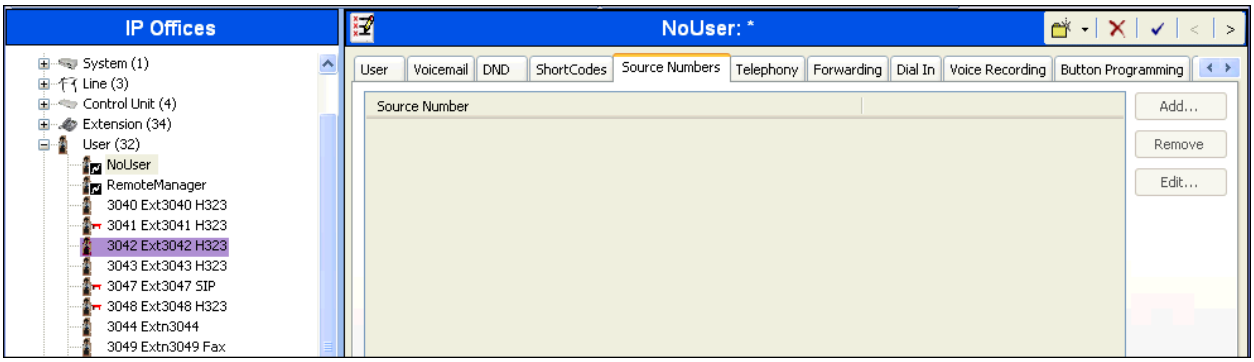

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 1236999464 on line 17 are routed to extension 3040.



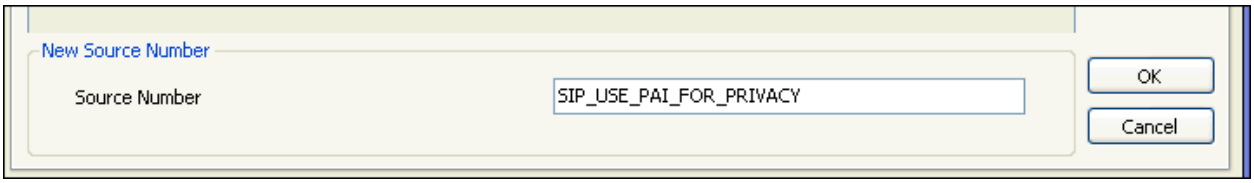

## 5.10. Privacy/Anonymous Calls

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

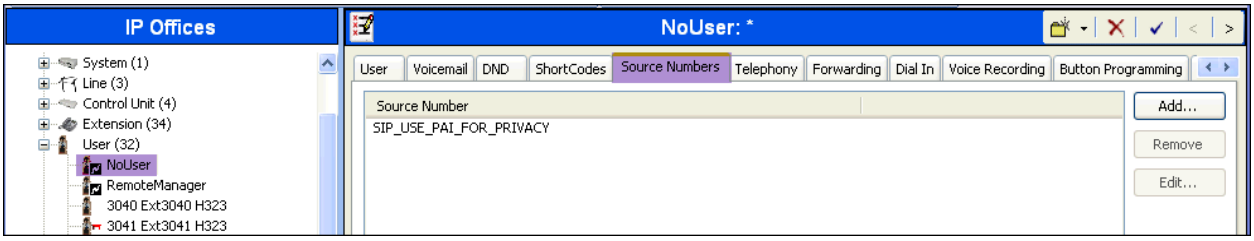
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User →NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.




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




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




## 5.11. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2.3** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **NoUser** user. The OPTIONS period is determined in the following manner:

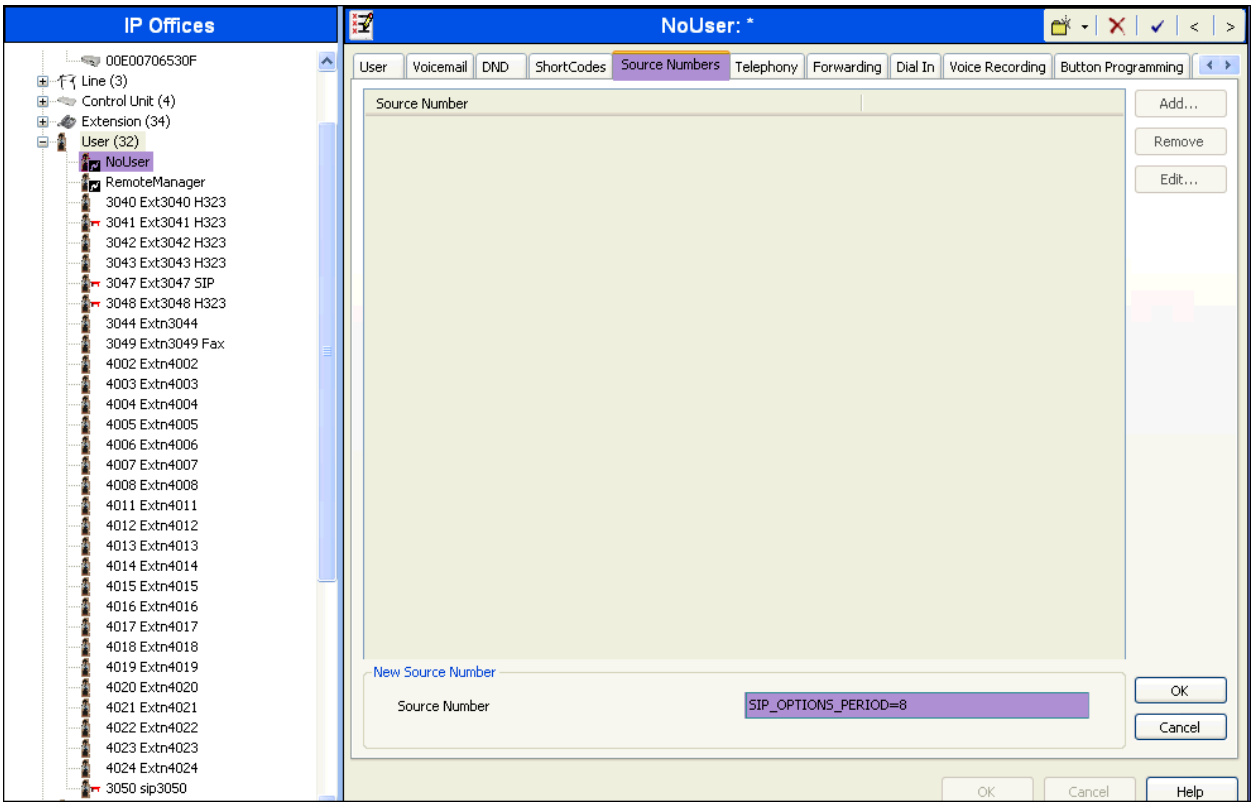
- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 300 seconds is used.

- To establish a period less than 300 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 300 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

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

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

For the compliance test, an OPTIONS period of 5 minutes was used. The **Binding Refresh Time** was set to **300** seconds (5 minutes) in **Section 5.2.3**. The **SIP_OPTIONS_PERIOD** was set to **8** minutes. Avaya IP Office chose the Binding Refresh Time of 300 seconds as the smaller of these two values. Click the **OK** button.

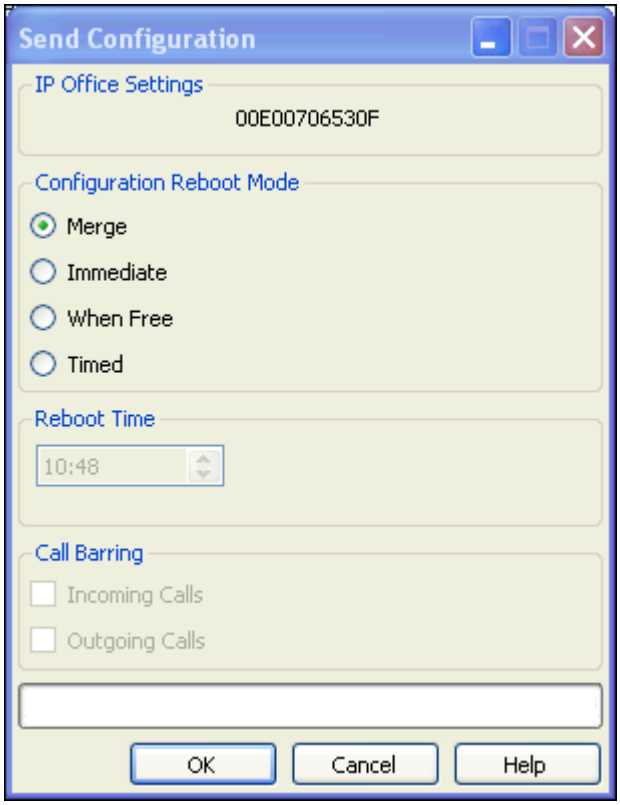


## 5.12. Save Configuration

When desired, send the configuration changes made in Avaya IP Office Manager to the Avaya IP Office server to cause 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.



# 6. TELUS SIP Trunk Service Configuration

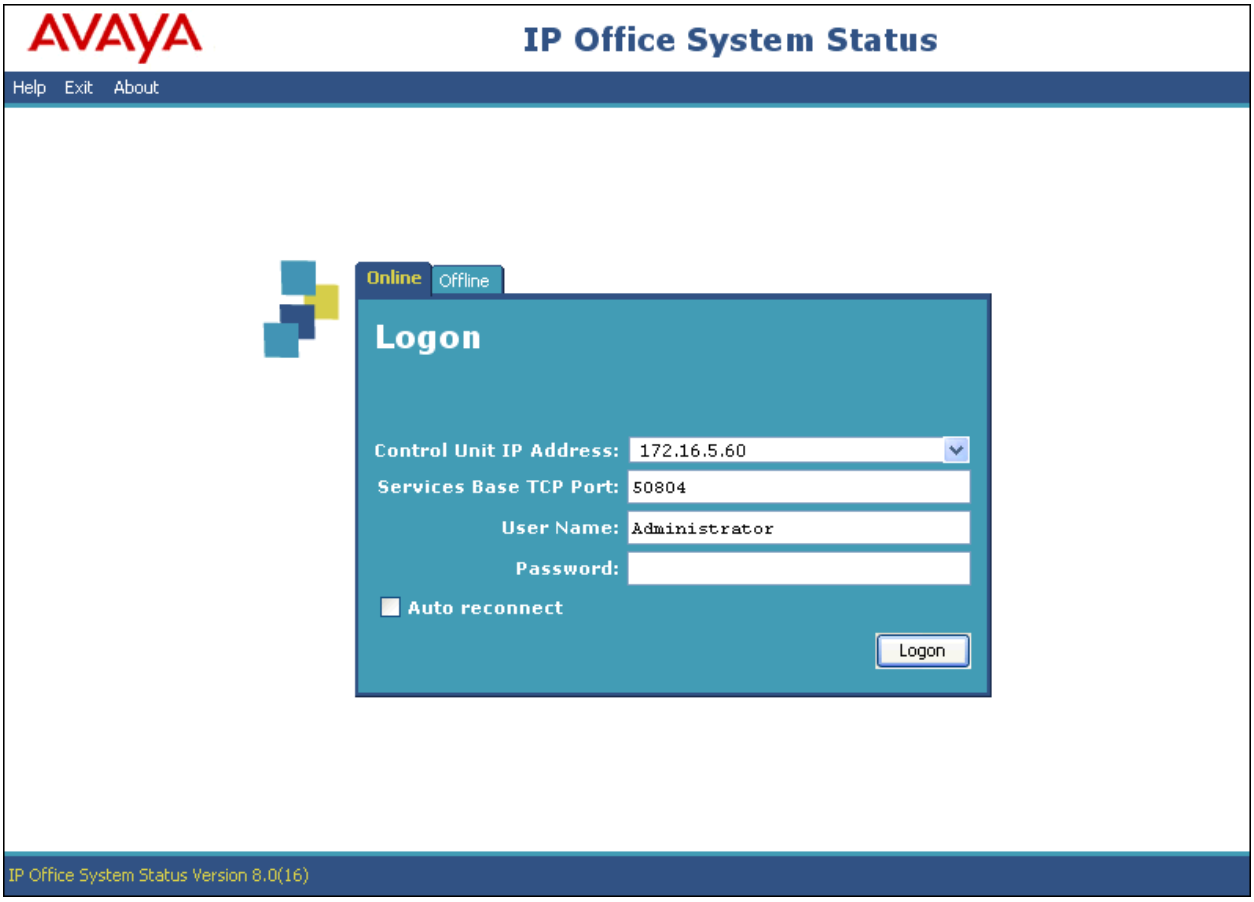
TELUS is responsible for the configuration of the SIP Trunk service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise site. TELUS will provide the customer the necessary information to configure the Avaya IP Office SIP connection to TELUS, including:

- IP address of the TELUS SIP Proxy server.
- Supported codecs.
- DID numbers.
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

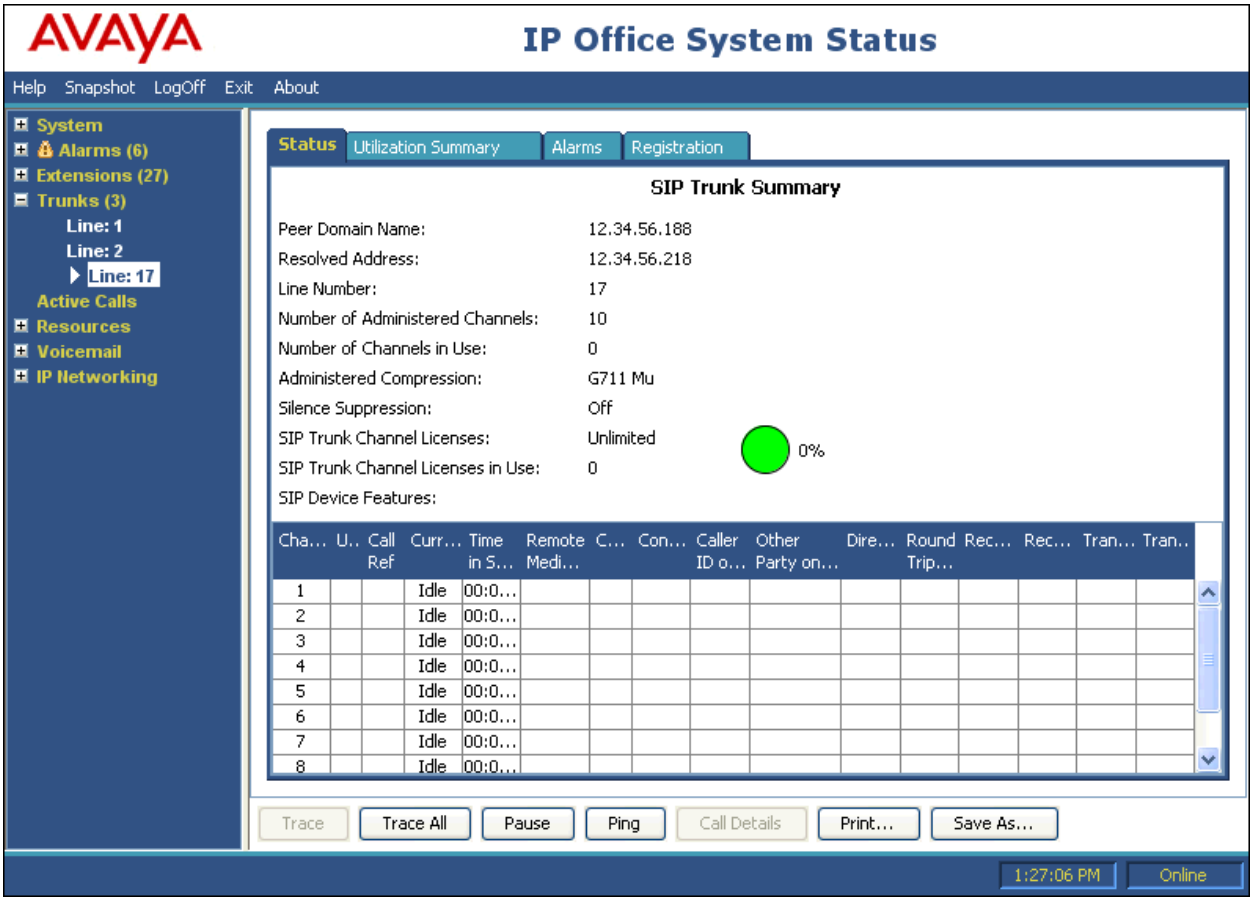
# 7. Verification Steps

The following steps may be used to verify the configuration:

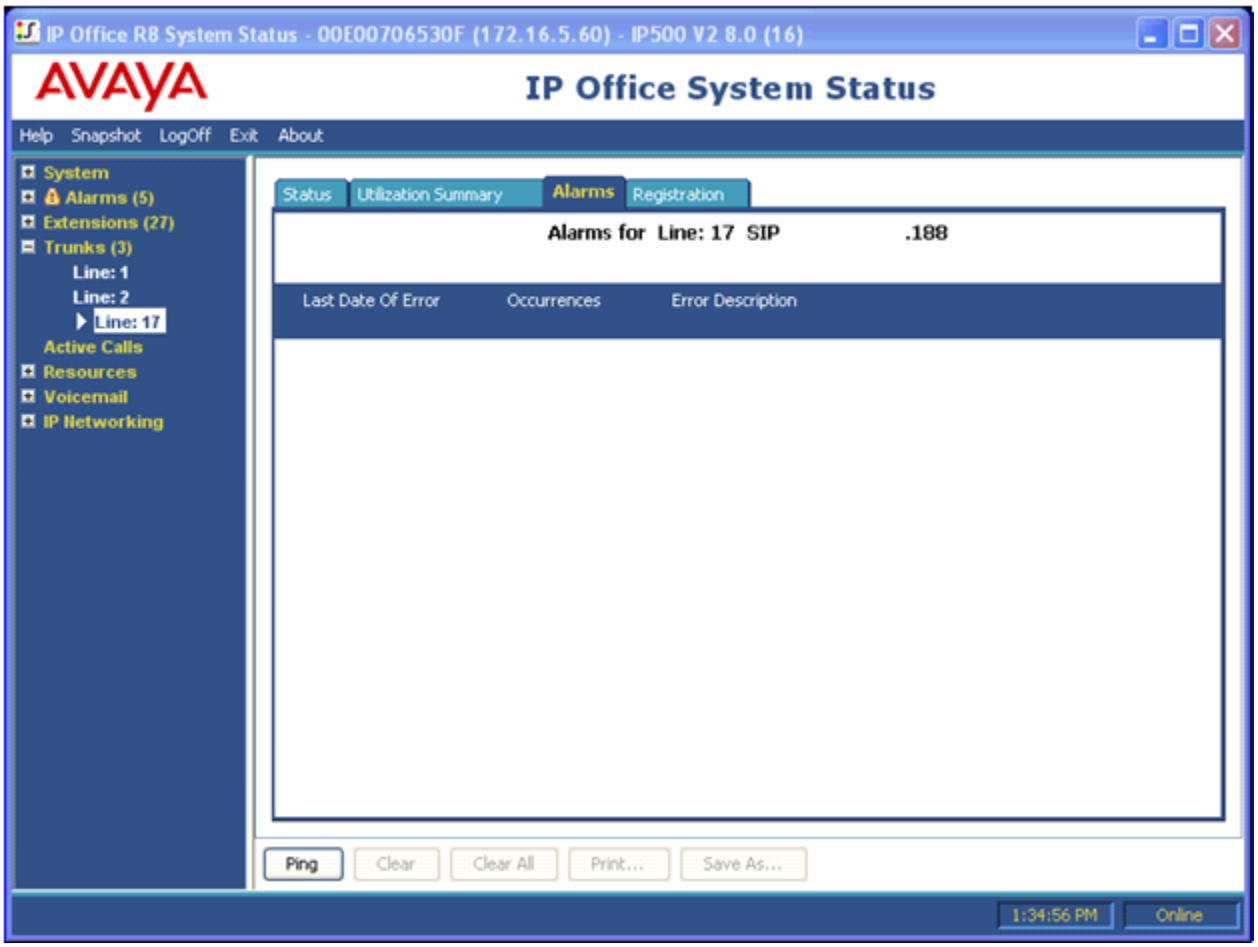
Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start →Programs→IP Office→System Status** on the PC where Avaya IP Office Manager was installed. Log in using the appropriate credentials.



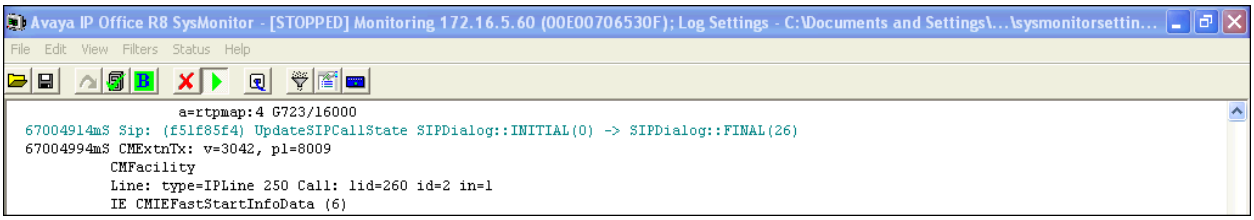
Select the SIP line configured from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is **Idle** for each channel (assuming no active calls at present time).



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



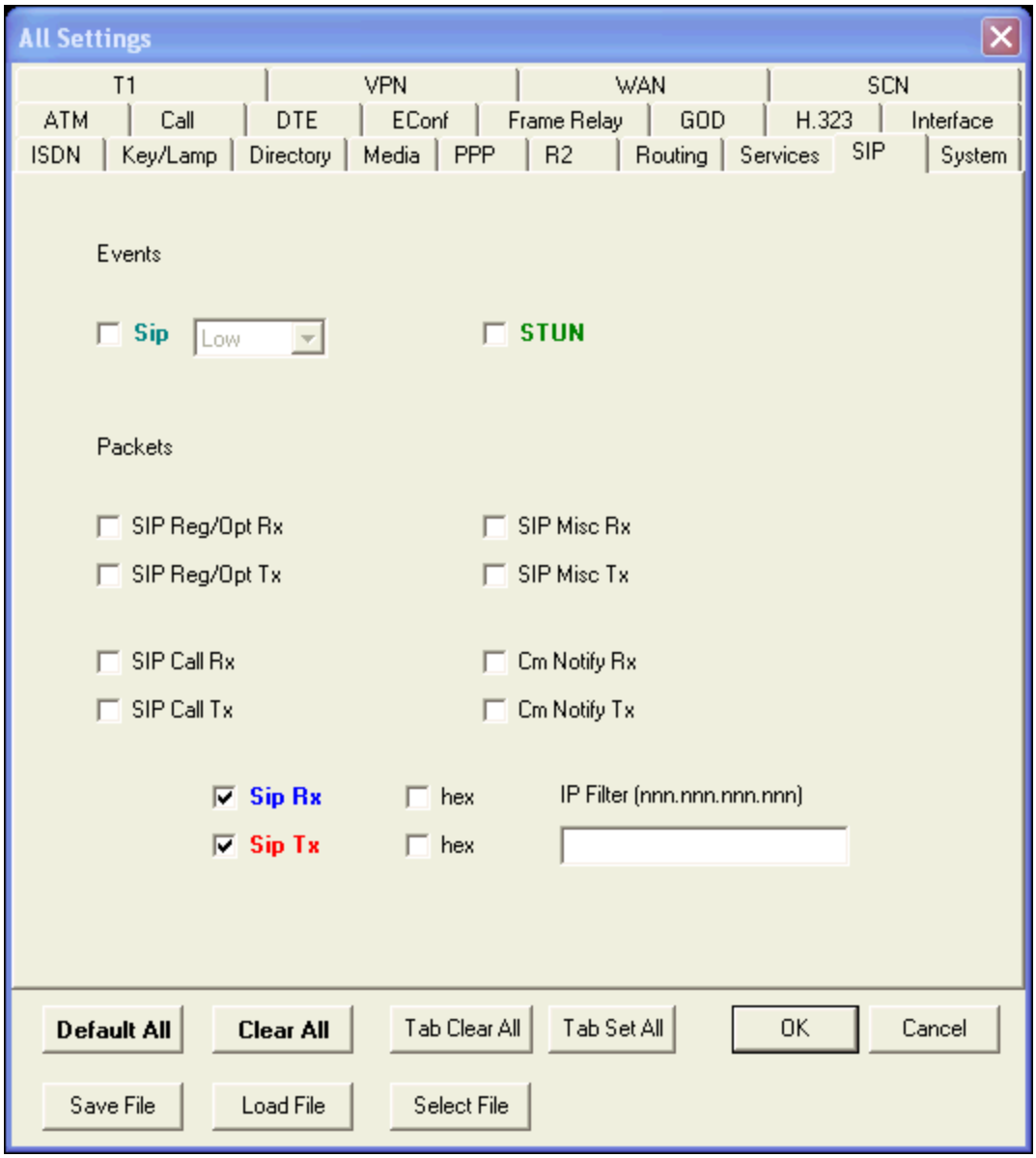

The System Monitor application can also be used to monitor or troubleshoot. The System Monitor application can typically be accessed from **Start→Programs→IP Office→Monitor**. See reference [7] for more information. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters→Trace Options**.




The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. To customize colors, right-click on **SIP Rx** or **SIP Tx** and select the desired color. In this example, all received SIP messages will appear in

the trace with the color blue, and all transmitted SIP messages will appear in the trace with the color Red.

.



# 8. Conclusion

The TELUS SIP Trunk Service passed compliance testing. These Application Notes describe the procedures required to configure the SIP trunk connectivity between Avaya IP Office 8.0 and the TELUS SIP Trunk Service, as shown in **Figure 1**.

# 9. Additional References

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

[1] IP Office 8.0 Installation Manual, Document Number 15-601042, December 2011.
[2] IP Office Manager Manual 10.0, Document Number 15-601011, January 2012.
[3] IP Office System Status Application, Document Number 15-601758, November 2011
[4] IP Office Release 8.0 Implementing Voicemail Pro, Document Number 15-601064, December, 2011
[5] IP Office Softphone Installation, Issue 3c, October, 2011.
[6] RFC 3261 SIP: Session Initiation Protocol, http://www.ietf.org/
[7] IP Office System Monitor, Document Number 15-601019, November, 2008

# Appendix: SIP Line Template

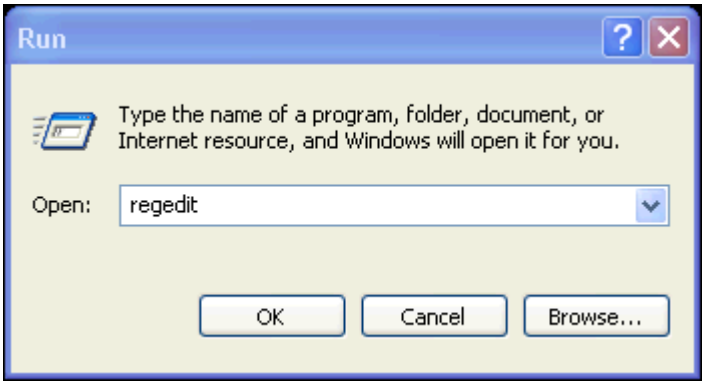
Avaya IP Office Release 8.0 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Not all of the configuration information is included in the SIP Line Template, therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported, and additional configuration be supplemented using the settings provided in this Application Notes.

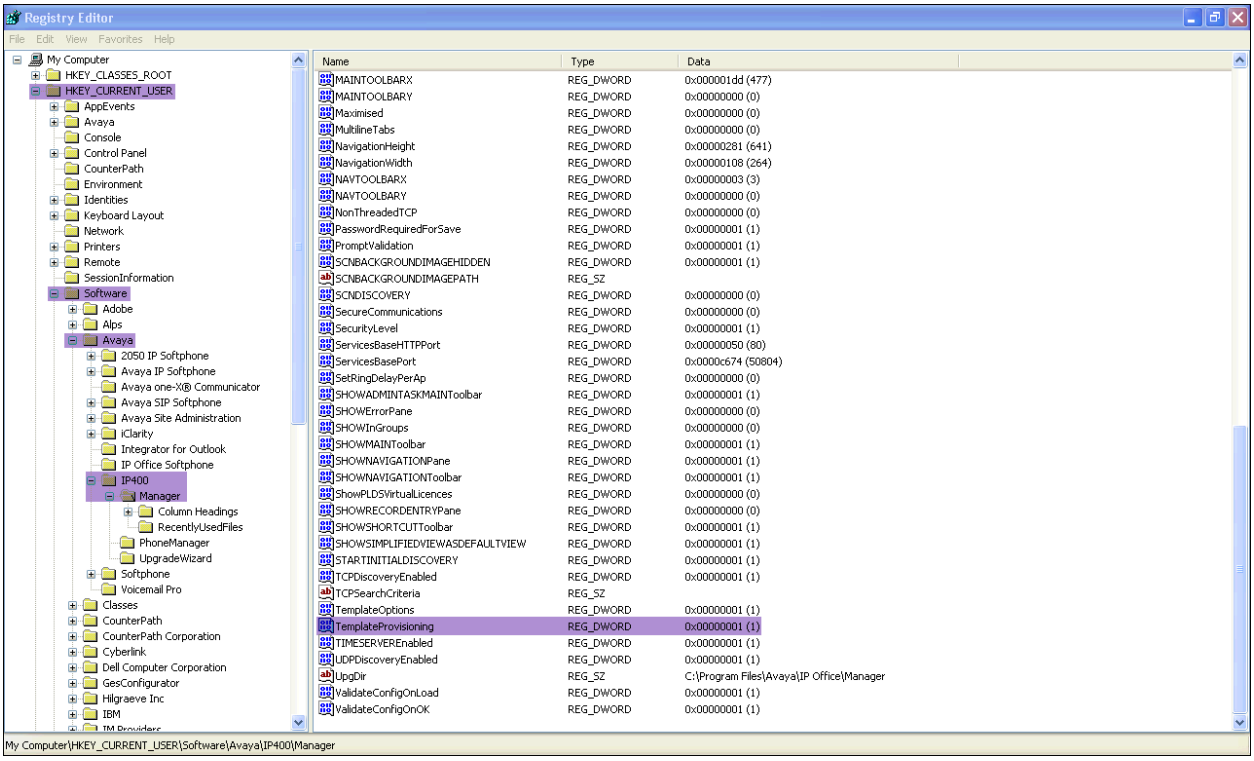
To create a SIP Line Template from the configuration described in these Application Notes, configure the parameters as described below.

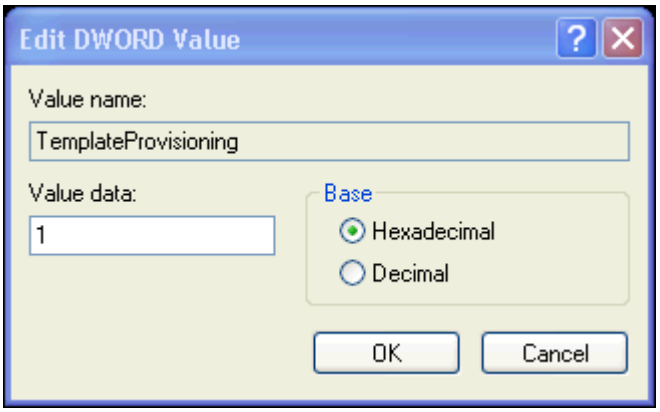
Create a new registry entry called **TemplateProvisioning** and set the **Value data** to **1**, as follows:

Select **Start**, and then **Run**. Type **regedit** as shown below

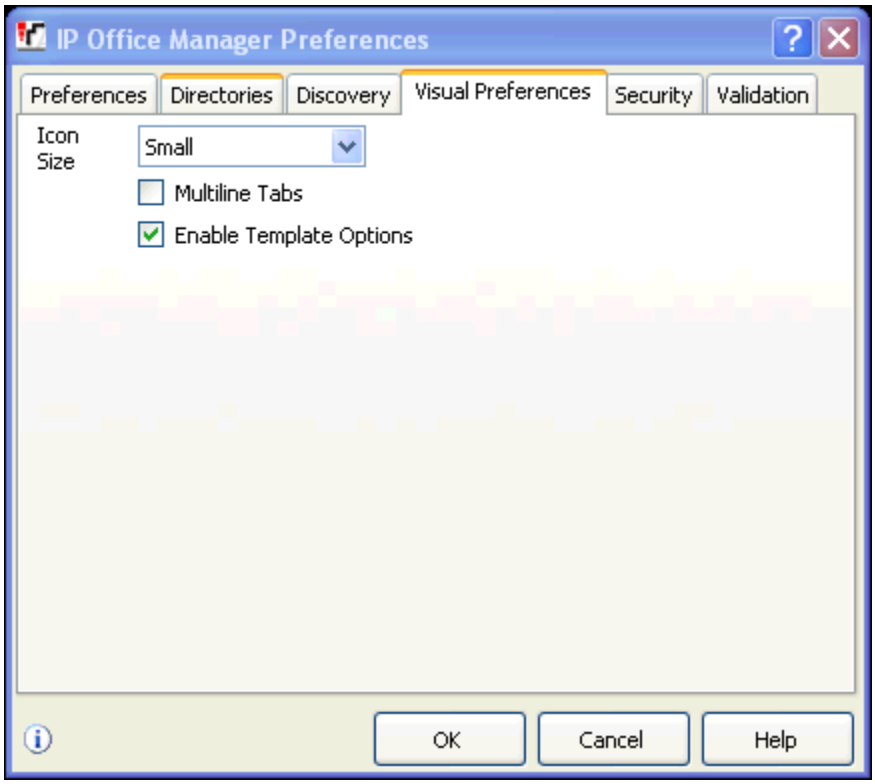


Under **HKEY_CURRENT_USER, Software, Avaya, IP400**, right click on **Manager**, then select **New, DWORD value,** then rename the newly created entry to: **TemplateProvisioning.** Right click on the newly created entry and select **Modify,** change the value under **Value Data** from "**0**" to "**1**". **Reboot the computer.**



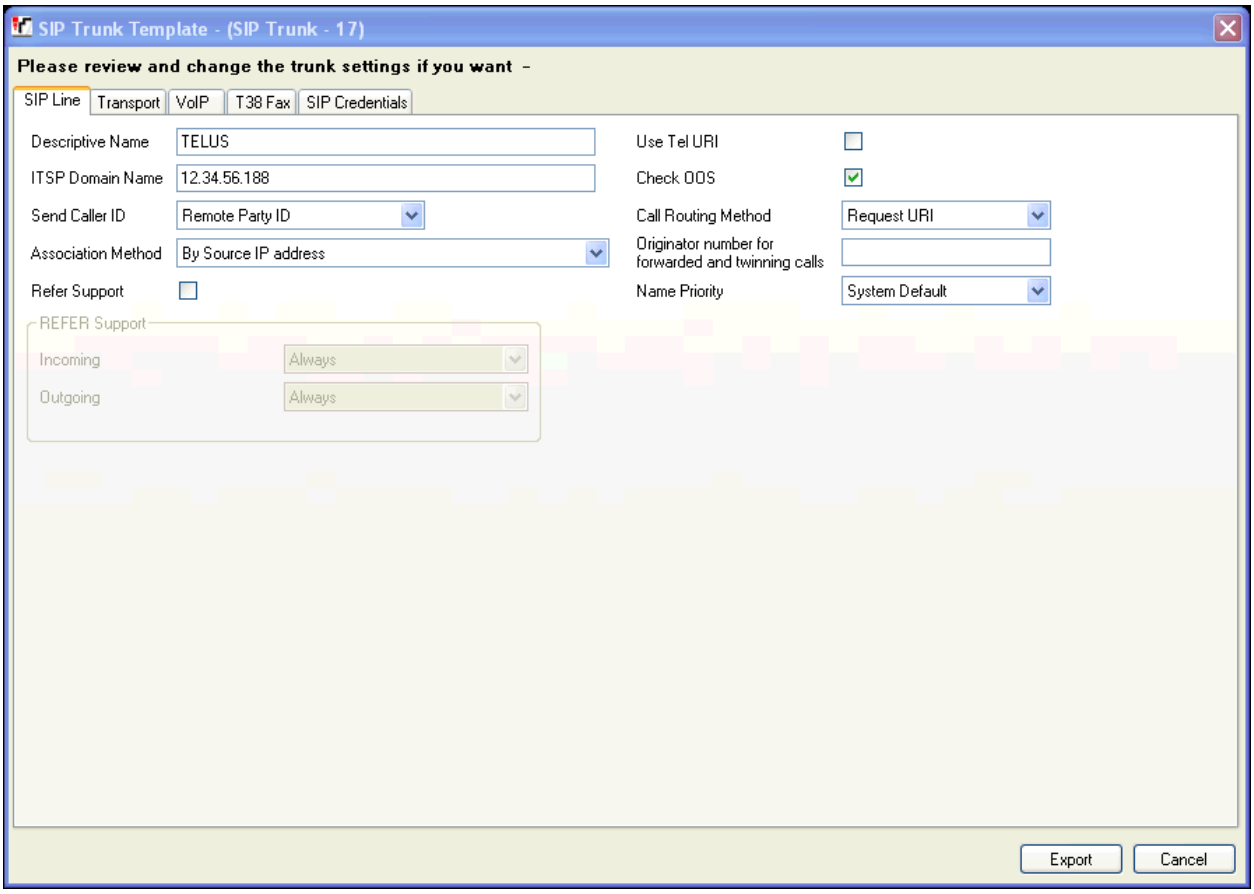



To enable template support, select **File**, then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.

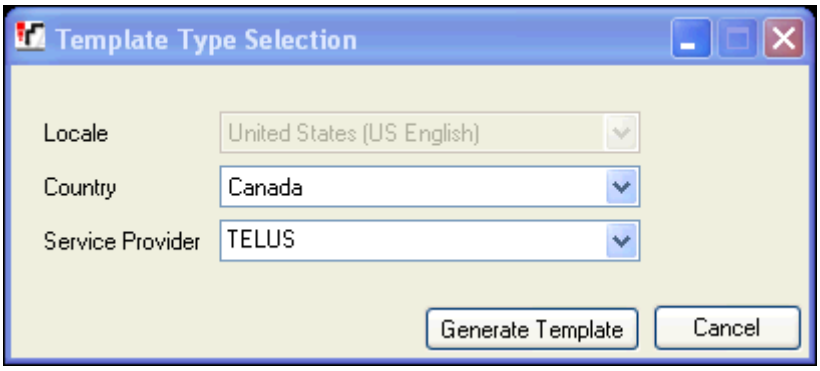


To create a SIP Line Template from the configuration, on the left Navigation Pane, right click on the Sip Line (**17**), and select **Generate SIP Trunk Template** (not shown)

Enter a descriptive name for the template, adjust the settings if required, and then click on **Export**.




On the next screen, **Template Type Selection**, select the **Country,** enter the name for the **Service Provider**, and click **Generate Template**.



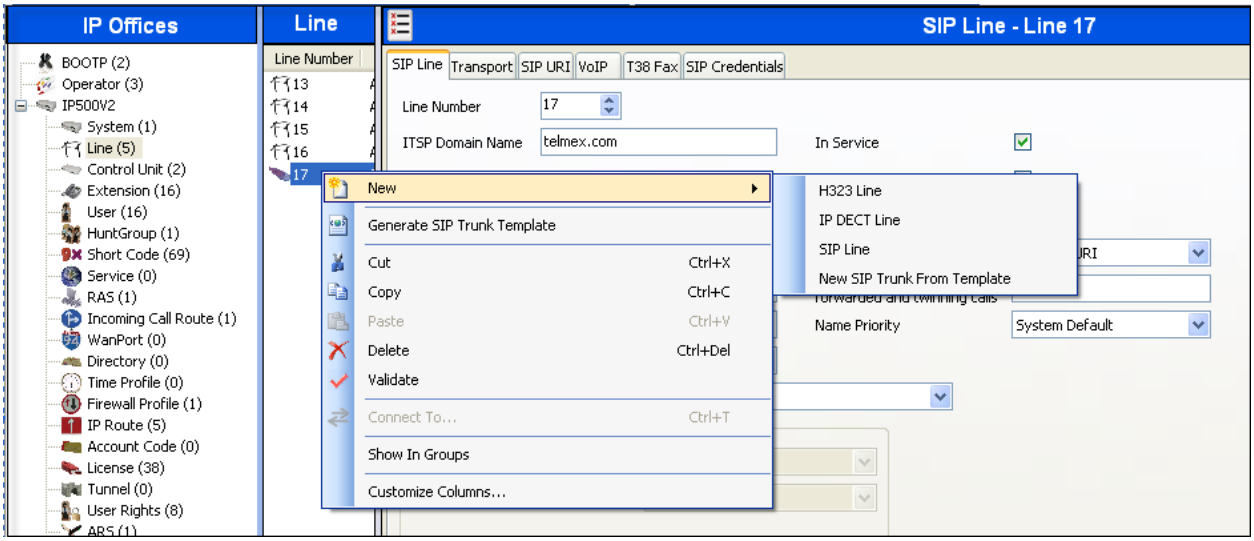
The following is the exported SIP Line Template file, **CA_TELUS_SIPTrunk.xml**:


```
- <Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20120228</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>TELUS</DescriptiveName>
  <ITSPDomainName>12.34.56.188</ITSPDomainName>
  <SendCallerID>CallerIDRPID</SendCallerID>
  <ReferSupport>false</ReferSupport>
  <ReferSupportIncoming>1</ReferSupportIncoming>
  <ReferSupportOutgoing>1</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <ITSPProxy>12.34.56.218</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>ALAW64K</CompressionMode>
  <UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
  <AdvCodecPref>G.711 ULAW 64K</AdvCodecPref>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_T38FB</FaxTransportSupport>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>true</Rel100Supported>
  <T38FaxVersion>3</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
  <HighSpeed>0</HighSpeed>
  <TCFMethod>Trans_TCF</TCFMethod>
  <MaxBitRate>FaxRate_14400</MaxBitRate>
  <EflagStartTimer>2600</EflagStartTimer>
  <EflagStopTimer>2300</EflagStopTimer>
  <UseDefaultValues>true</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
```

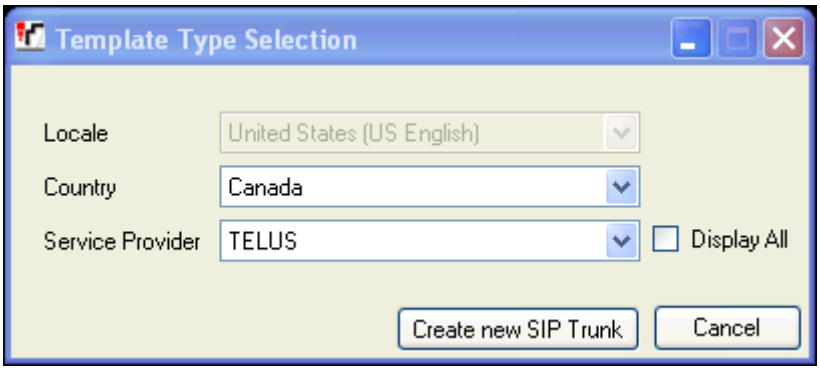
```
<TFOPEnhancement>true</TFOPEnhancement>
<DisableT30ECM>false</DisableT30ECM>
<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>false</DisableT30MRCompression>
<NSFOverride>false</NSFOverride>
- <SIPCredentials>
<Expiry>60</Expiry>
<RegistrationRequired>false</RegistrationRequired>
  </SIPCredentials>
  </Template>
```


To import the template into a new Avaya IP Office system, copy and paste the exported xml template file to the Templates directory (C:\Program Files\Avaya\IP Office\Manager\Templates) on the PC where Avaya IP Office Manager for the new system is running.

Next, import the template into the new Avaya IP Office system by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New, New SIP Trunk from Template**:



On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.



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.