Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Alestra Enlace IP SIP Trunk Service with Avaya IP Office 8.1 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Alestra in Mexico and Avaya IP Office 8.1.
The official name of Alestra's SIP Trunk offering is **"Enlace IP"**.

During the interoperability testing, Avaya IP Office was able to interoperate with the Alestra Sonus GSX 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 Avaya endpoints.

Alestra's Enlace IP SIP Trunking Service provides PSTN access via a SIP trunk between the enterprise and Alestra's network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Alestra in Mexico 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.

Introduction

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

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 8.1 with a Digital Expansion Module, Avaya Voicemail Pro, Avaya IP Office Softphone (SIP and H.323), Avaya IP Office Phone Manager PC Softphone (H.323), Avaya H.323 Telephones, Avaya SIP Telephones, Avaya Digital Telephones, Analog Telephones and a fax machines.

Alestra's Enlace IP SIP Trunking 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.

General Test Approach and Test Results

The approach used for the test was to connect a simulated enterprise site to Alestra Sonus GSX switch via SIP trunk 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.

Interoperability Compliance Testing

A simulated enterprise site with an Avaya IP Office was connected to Alestra's Enlace IP SIP Trunking service. To verify SIP trunk interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Response to SIP OPTIONS queries.
- Incoming calls to Avaya IP Office from the PSTN were routed to DID numbers assigned by Alestra. Incoming PSTN calls were terminated to the following endpoints: Avaya IP Telephones (H.323 and SIP), Avaya Digital Telephones, Avaya IP Office Softphone (H.323 and SIP), Avaya IP Office Phone Manager PC Softphone (H.323), Analog Telephones and a fax machine.
- Outgoing calls from Avaya IP Office were routed via Alestra's Sonus GSX network to the various PSTN destinations. A local PSTN extension in Monterrey & telephones in the Test Lab connected to the PSTN in the U.S. were used as PSTN endpoints.
- 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 endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Proper Codec negotiation and two way speech-path. G.729(a) codec and G.711-Alaw codec were tested as requested by Alestra (common codec's used in Mexico).
- Proper response/error treatment with no matching codec's between the network and the enterprise.
- Voice mail and DTMF tone support (RFC 2833).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response) systems.
- Outbound/Inbound local calls.
- International calls.
- Calls to special numbers (Alestra information: 040, etc.).
- Calling number blocking from Avaya IP Office and from the PSTN.
- Call Hold/Resume (long and short duration).
- Call Forward (unconditional, busy, no answer).
- Off-net call forwarding.
- Blind Call Transfers.
- Consultative Call Transfers.
- Use of SIP REFER for call transfer to the PSTN.
- 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 the SIP connection.

Test Results

Interoperability testing of Alestra's Enlace IP SIP trunk service with Avaya IP Office Release 8.1 was completed successfully with the following observations/limitations.

1. **Feature Name Extension (FNE)** – This is a Mobility Twinning feature for Mobile call control. This feature provides dial tone to twinned mobile devices (e.g., cell phone) directly from Avaya IP Office; once dial tone is received the user can perform dialing actions including making calls and activating Short Codes.

   To get the FNE feature to work with the dial plan specific to Mexico, the following settings had to be matched. Note: The user gets a Busy tone (486 Busy Here) when attempting to activate the feature if the settings are not matched.

   Example Configuration:
   Using DID number **8128811210** from the local PSTN as the twinned mobile extension.
   Using extension number 3041 in Avaya IP Office as the Host station.

<u>INVITE received by Avaya IP Office:</u>
The "FROM" field in the invite sent by the Twinned mobile extension to Avaya IP Office when dialing the number of the Host station is in the following format:
From:<sip:8128811210@192.168.10.9

<u>Avaya IP Office settings:</u>
User/3041/Mobility: Twinned Mobile Number = **8128811210**
Incoming Call Route/Incoming CLI = **8128811210**

With the above settings calling party receives dial tone from Avaya IP Office for dialing, but twinning doesn't work since the number is not in the correct format. For twinning to work the following setting is needed:

User/3041/Mobility: Twinned Mobile Number = **28811210**

This issue may be resolved by digit manipulation in Alestra's network.

2. Call Display on transferred calls to the 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 ID of the host extension that initiated the call transfer.

3. Calls originating from PSTN telephones in the U.S. to Mexican DIDs assigned to Avaya IP Office will display "Restricted/Unavailable". This is a PSTN restriction for all calls from the U.S. to Mexico. For testing, Alestra provided a local PSTN number in Monterrey, Mexico. A SIP Softphone was registered to this local PSTN number and was used to originate and terminate calls to and from the Mexican PSTN to Avaya IP Office. Alestra also provided access to a WEB based GUI allowing feature changes to this local PSTN number.

   Note: International long distance calls will be presented without Caller ID in accordance with International Rule between carriers.

4. Items not supported or not tested included the following:
   - Inbound toll-free calls.
   - 0, 0+10, 411,911, etc. are call types not supported in Mexico. Instead, calls to special numbers in Mexico were tested (e.g., information: 040, Denuncia: 089, etc.).

Support

For technical support on Alestra Enlace IP SIP Trunk service offer visit the online support site at http://www.alestra.com.mx/negocios.asp?id=206

Reference Configuration

Figure 1 below illustrates the test configuration used. The test configuration shows an enterprise site connected to Alestra's Enlace IP SIP Trunking service through the public internet.

For confidentiality and privacy purposes, actual public IP addresses and telephone numbers used during the testing have been masked and replaced with fictitious IP addresses and telephone numbers throughout the document.

Located at the enterprise site is an Avaya IP Office 500v2 with the MOD DGTL STA16 expansion which provides connections for 16 digital stations, the extension PHONE 8 card which provides connections for 8 analog stations as well as 64-channel VCM (Voice Compression Module) for VoIP codec support. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network. Endpoints includes Avaya 9600 Series IP Telephones (H.323), Avaya 1100 Series IP Telephones (SIP), Avaya 1400 Series Digital Telephones, Avaya 9500 Series Digital Telephones , Avaya Analog Telephones, Avaya IP Office Softphone (H.323 and SIP), Avaya IP Office Phone Manager PC Softphone (H.323) and fax machines. Avaya IP Office Manager run on a Windows XP machine, it's used to configure and administer Avaya IP Office.

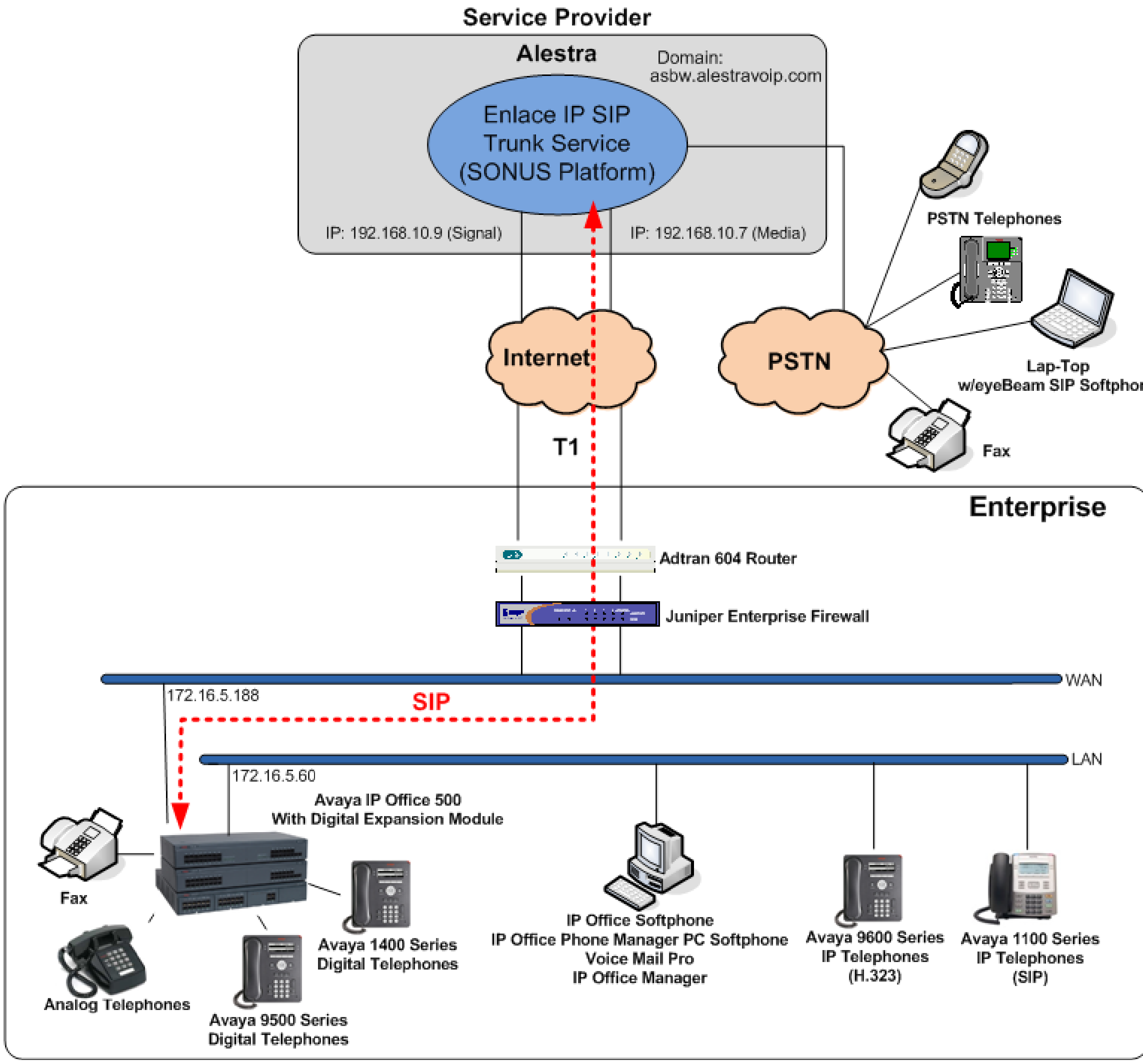


Figure 1: Test Configuration for Avaya IP Office with Alestra Enlace IP SIP Trunking Service

For the purposes of the compliance test, for outbound international calls from Avaya IP Office stations to PSTN stations in the U.S. the user dialed a short code of 9+001+10 digits to send digits across the SIP trunk to Alestra's network. The short code of 9 was stripped off by Avaya IP Office but the remaining 001+ 10 digits were sent unaltered to Alestra's network. For inbound international calls from PSTN stations in the U.S. to Avaya IP Office stations, the user would dial 01152812282+4 digits (4 digit DID numbers assigned by Alestra). For local calls from Avaya IP Office stations to an eyebeam SIP soft Client registered across the public internet to Alestra's network, the user dialed a short code of 9+28811234 to send digits across the SIP trunk to Alestra's network.

Equipment and Software Validated

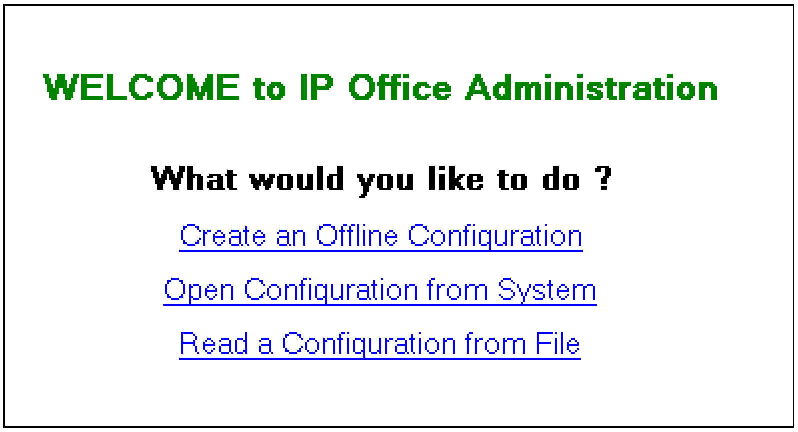
The following equipment and software/firmware were used for the sample configuration provided:

Avaya Telephony Components	
Equipment	**Release**
Avaya IP Office 500v2	8.1 (43)
Avaya IP Office Digital Expansion (DIG DCPx16 V2)	10.1 (43)
Avaya IP Office Manager	10.1 (43)
Avaya Voicemail Pro Client	10.1 (43)
Avaya 9600 Series IP Telephones (H.323)	Avaya one-X Desk phone Edition S3.104S
Avaya 1100 Series IP Telephones (SIP)	SIP1140 Load Ver.: 04.03.12.00
Avaya1400 Series Digital Telephones	N/A
Avaya 9500 Series Digital Telephones	N/A
Avaya Analog Phone	N/A
Avaya IP Office Softphone (H.323 and SIP)	3.2.3.20 64770
Avaya IP Office Phone Manager PC Softphone (H.323)	4.2.42
CounterPath eyebeam SIP Soft client	1.5.20.2 build 59031
Alestra Enlace IP SIP Trunking Service Components	
Component	Release
Sonus GSX soft switch	V07.03.06R003

Table 1 – Hardware and Software Components Tested

Configure Avaya IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to Alestra Enlace IP 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.



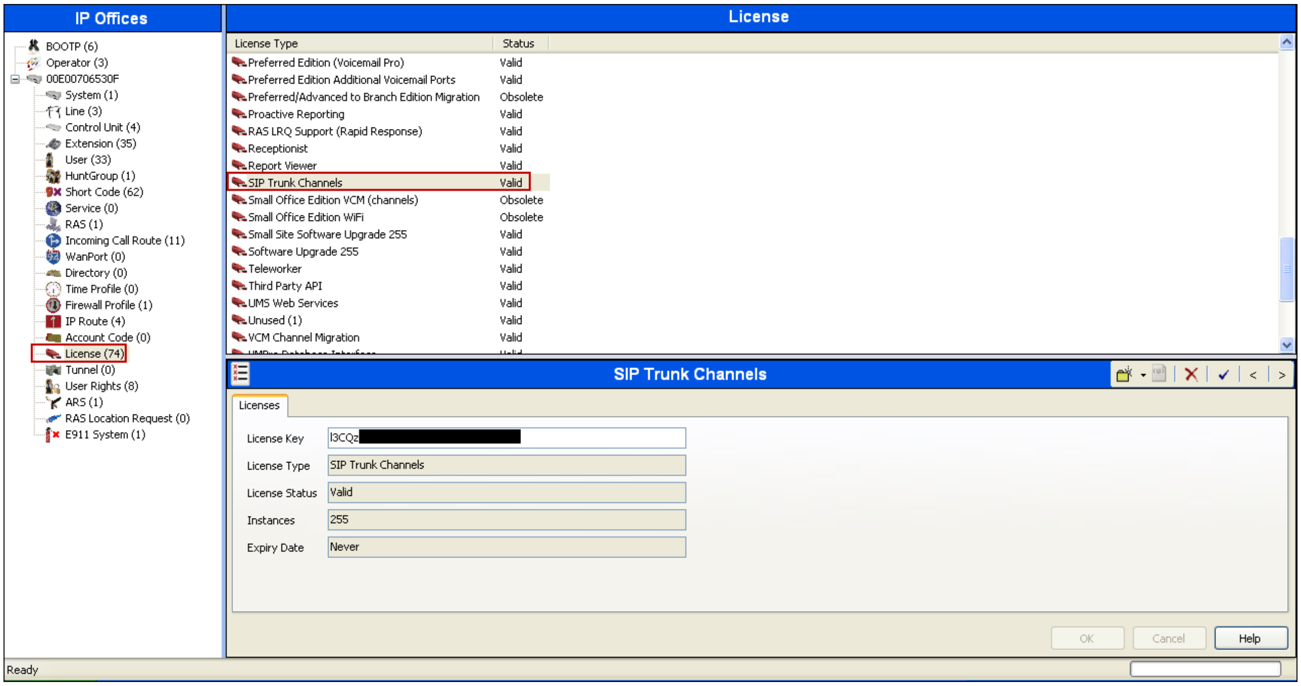
A management window will appear similar to the one shown in the next section.

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.

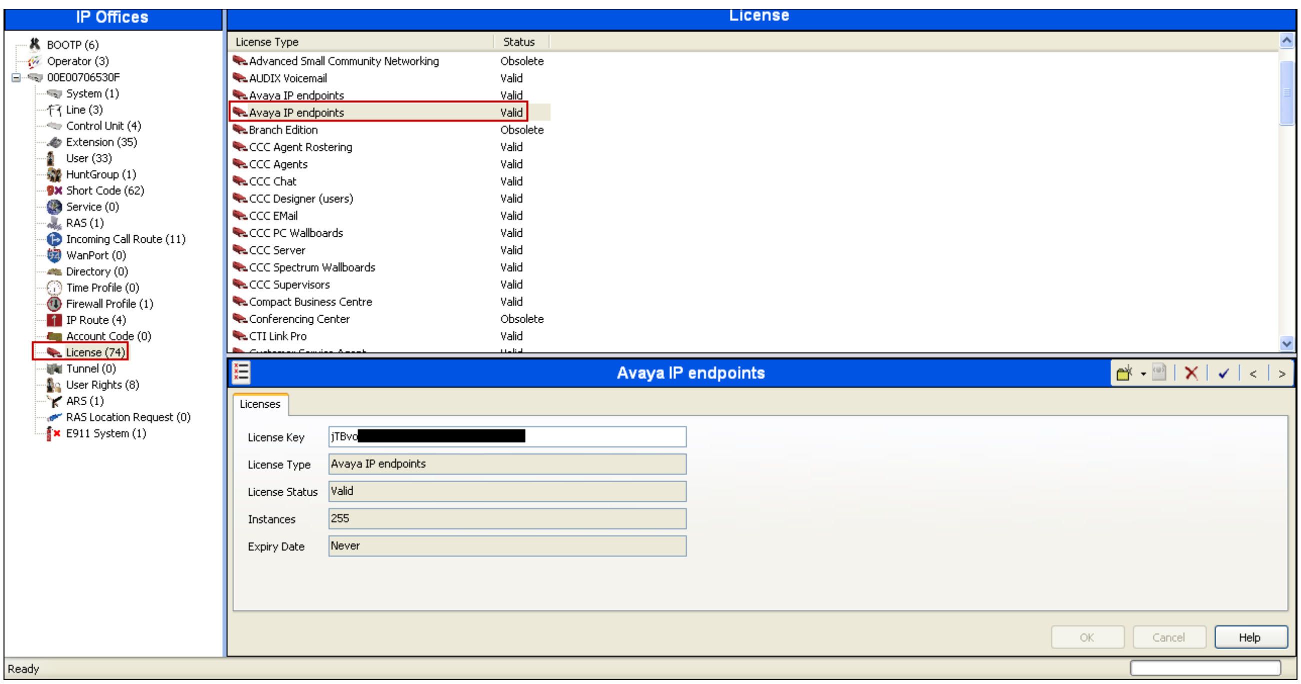
Licensing

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.



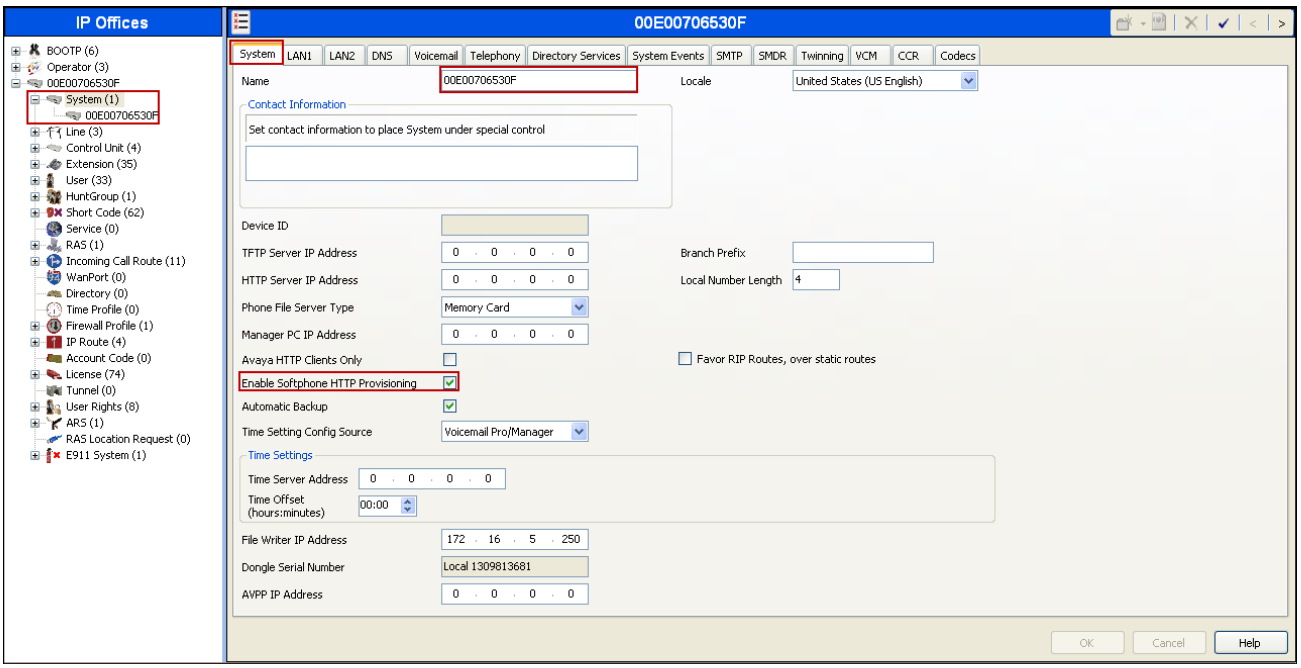
System Settings

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 8.0 and later.

In the sample configuration, the MAC address **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.

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.

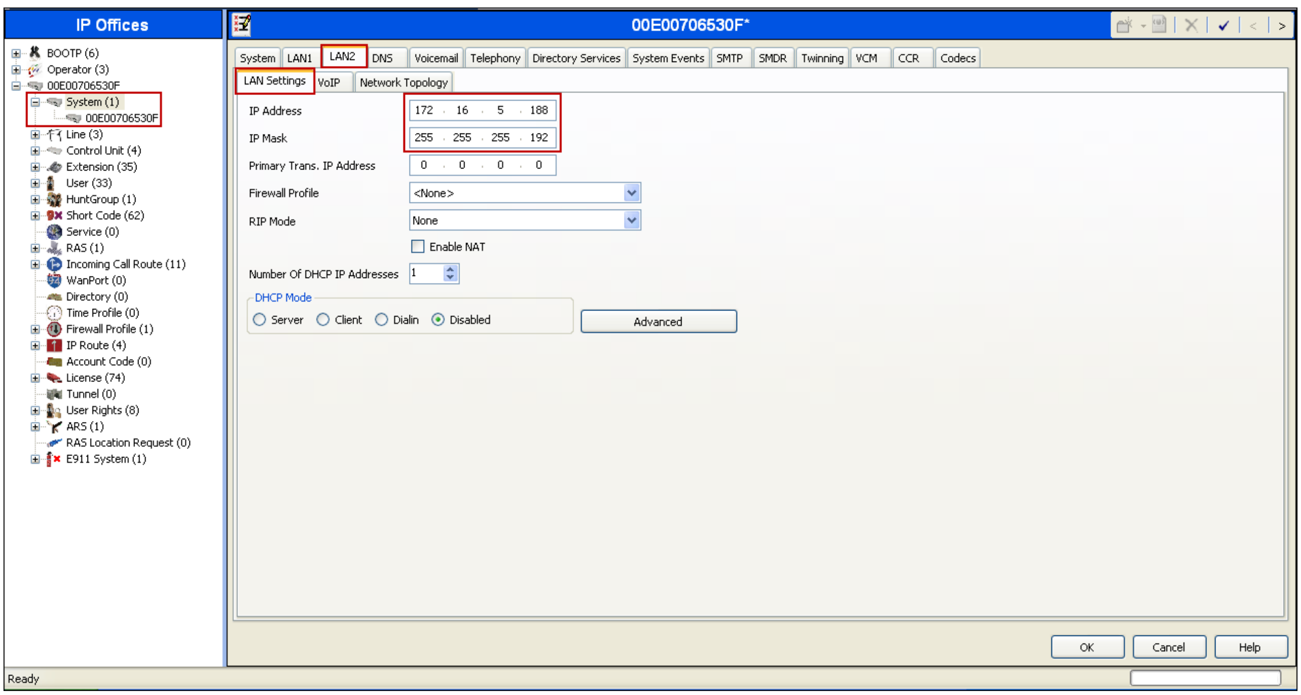


LAN1 Tab

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 Alestra Enlace IP SIP Trunk Service, and therefore is not described in these Application Notes. It should be noted that in this sample configuration telephones and soft clients use **LAN1** to register to Avaya IP Office, the configuration required for **LAN1** was done as part of the initial Avaya IP Office installation.

LAN2 Settings

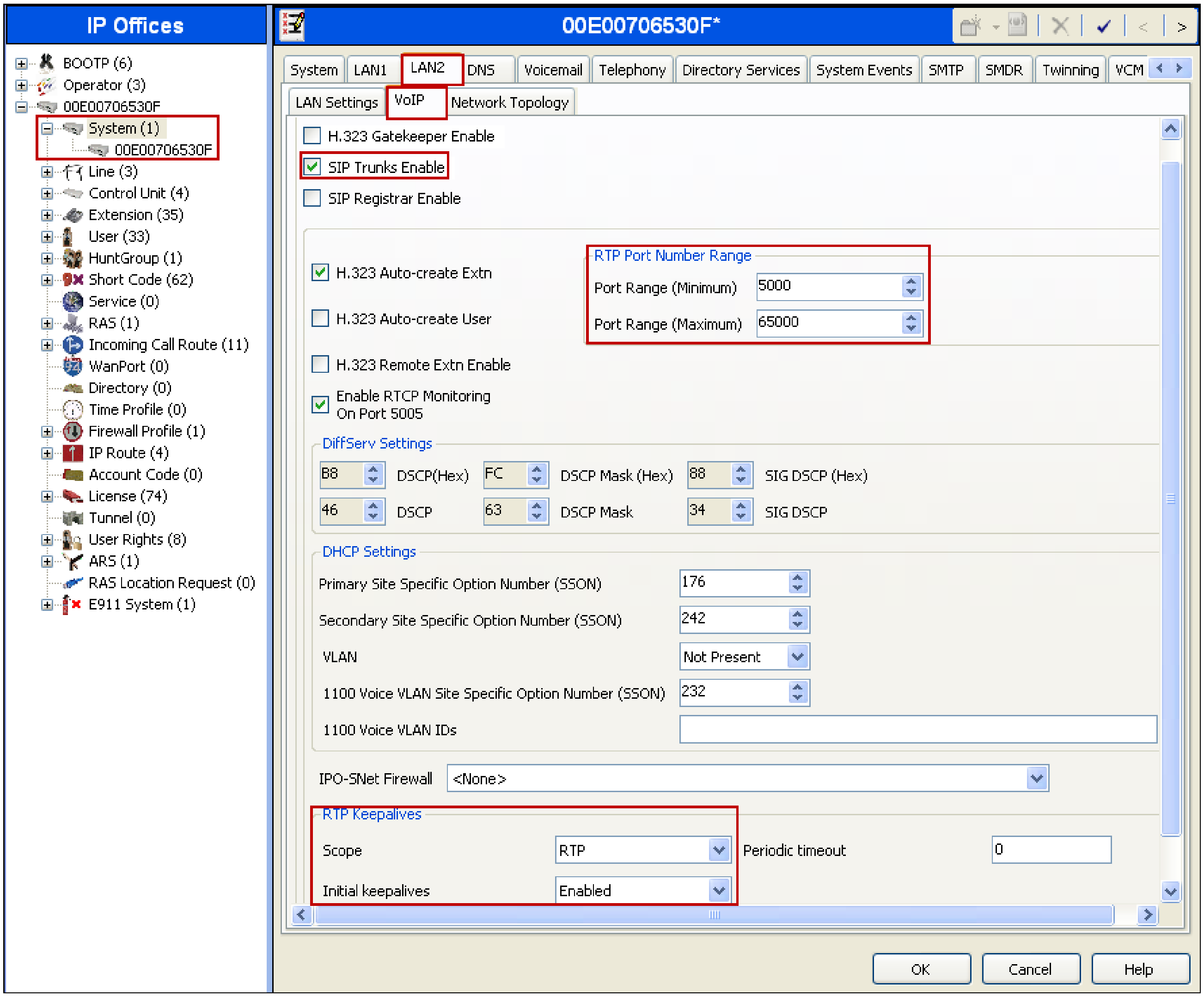
The LAN2 settings correspond to the WAN port on the Avaya IP Office; it was used to connect the Avaya IP Office to the public network. To access the **LAN2** settings, first navigate to **System (1) → 00E00706530F** in the Navigation and Group Panes and then navigate to the **LAN2→ LAN Settings** tab in the Details Pane. Set the **IP Address** field to the public 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.



Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Alestra. The **RTP Port Number Range** can be customized to a specific range of ports for the RTP media; port range 5000-65000 was used as requested by Alestra. 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.

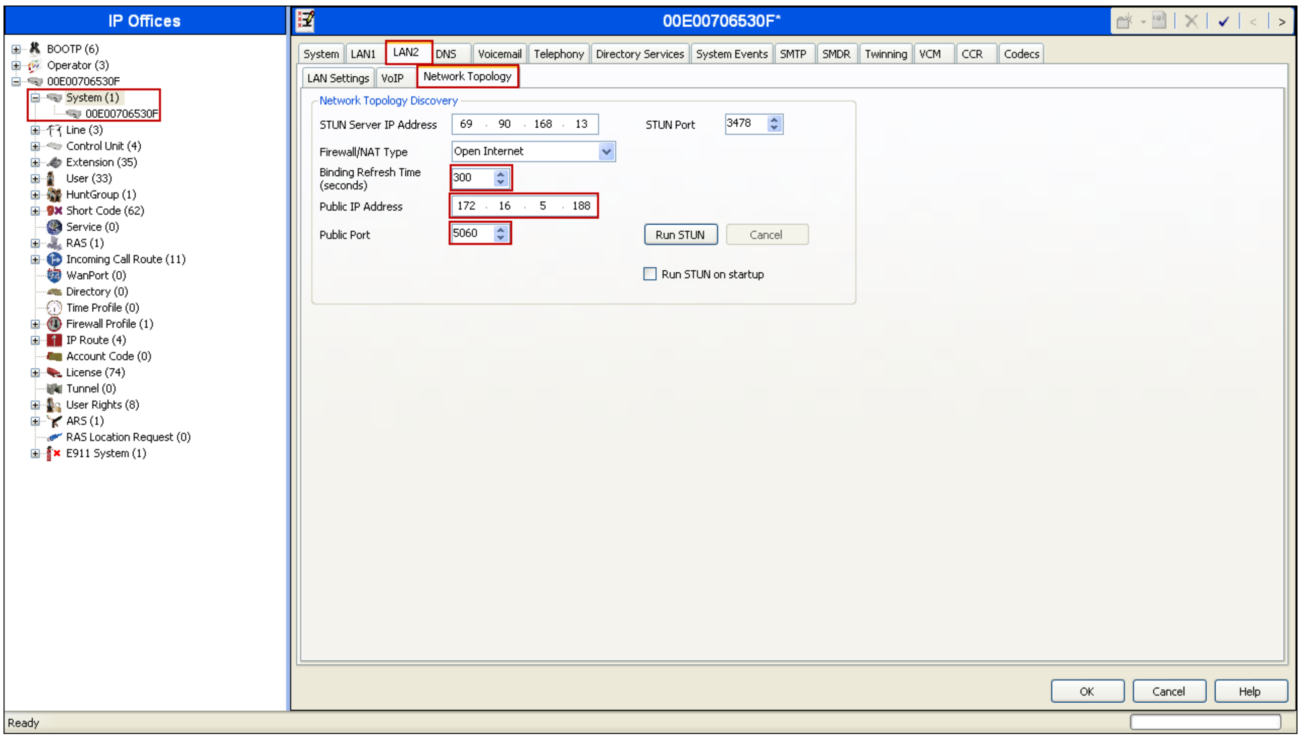
In the **RTP Keepalives** section at the bottom of the page, set the **Scope** field to **RTP**, and **Initial keepalives** to **Enabled**. This will cause the Avaya IP Office to send RTP keepalive packets at the beginning of the calls, to avoid problems of media deadlock that can occur with certain 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.


IP Offices
BOOTP (6)
Operator (3)
00E00706530F
System (1)
00E00706530F
Line (3)
Control Unit (4)
Extension (35)
User (33)
HuntGroup (1)
Short Code (62)
Service (0)
RAS (1)
Incoming Call Route (11)
WanPort (0)
Directory (0)
Time Profile (0)
Firewall Profile (1)
IP Route (4)
Account Code (0)
License (74)
Tunnel (0)
User Rights (8)
ARS (1)
RAS Location Request (0)
E911 System (1)
00E00706530F*
System
LAN1
LAN2
DNS
Voicemail
Telephony
Directory Services
System Events
SMTP
SMDR
Twinning
VCM
LAN Settings
VoIP
Network Topology
H.323 Gatekeeper Enable
SIP Trunks Enable
SIP Registrar Enable
H.323 Auto-create Extn
H.323 Auto-create User
RTP Port Number Range
Port Range (Minimum) 5000
Port Range (Maximum) 65000
H.323 Remote Extn Enable
Enable RTCP Monitoring On Port 5005
DiffServ Settings
B8 DSCP(Hex) FC DSCP Mask (Hex) 88 SIG DSCP (Hex)
46 DSCP 63 DSCP Mask 34 SIG DSCP
DHCP Settings
Primary Site Specific Option Number (SSON) 176
Secondary Site Specific Option Number (SSON) 242
VLAN Not Present
1100 Voice VLAN Site Specific Option Number (SSON) 232
1100 Voice VLAN IDs
IPO-SNet Firewall <None>
RTP Keepalives
Scope RTP
Periodic timeout 0
Initial keepalives Enabled
OK
Cancel
Help

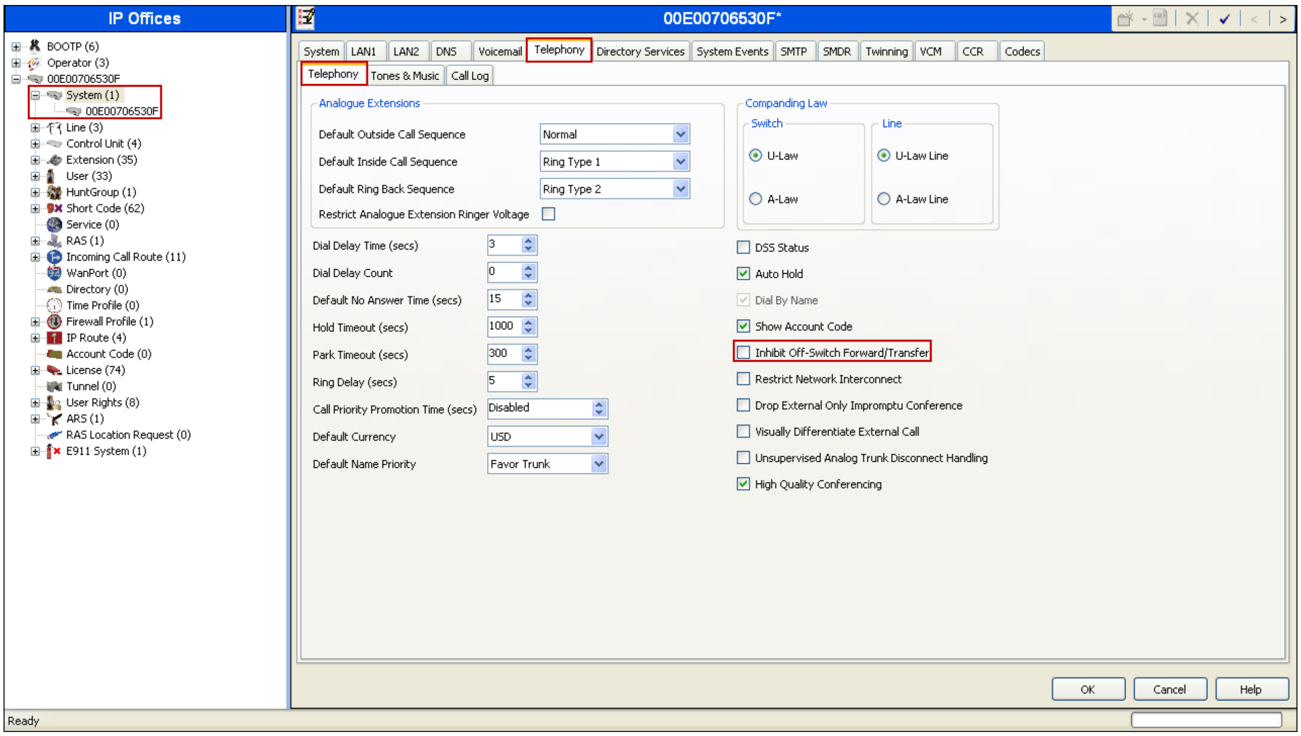
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test, so the parameter was set to **Open Internet**. With this configuration, STUN will not be used.
- Set **Binding Refresh Time (seconds)** to **300 (or every 5 minutes)**. For Avaya IP Office Release 8.1 this is the only setting that is required to set the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port. **Public Port** is set to **5060**
- All other parameters should be set according to customer requirements.



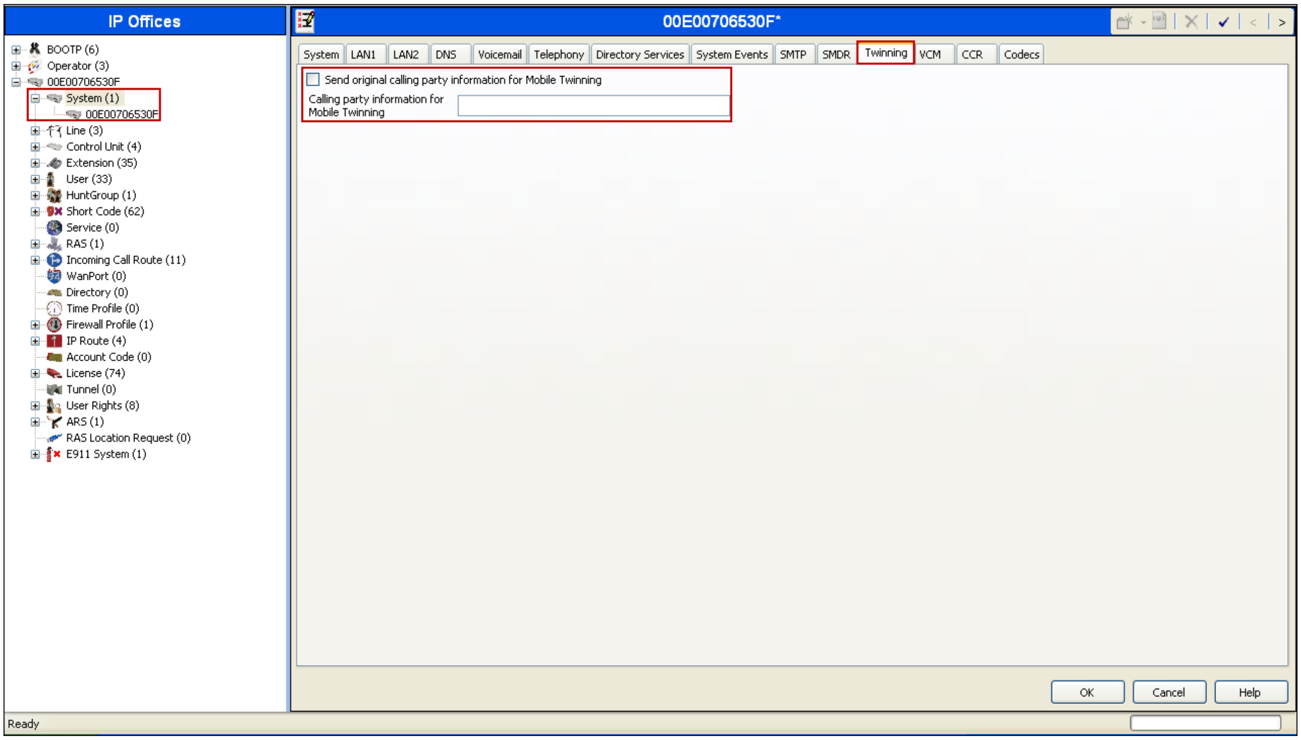
System Telephony Settings

Navigate to the **00E00706530F → System (1) → Telephony** Tab in the Details Pane. **Companding LAW** settings should not normally be changed from their defaults. They should only be used where 4400 Series phones (U-Law) are installed on systems which have A-Law digital trunks. Note that **U-Law** is also called **Mu-Law** or **µ-Law**. In the sample configuration the **Companding Law** was left as **default** (**U-Law**). Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk.



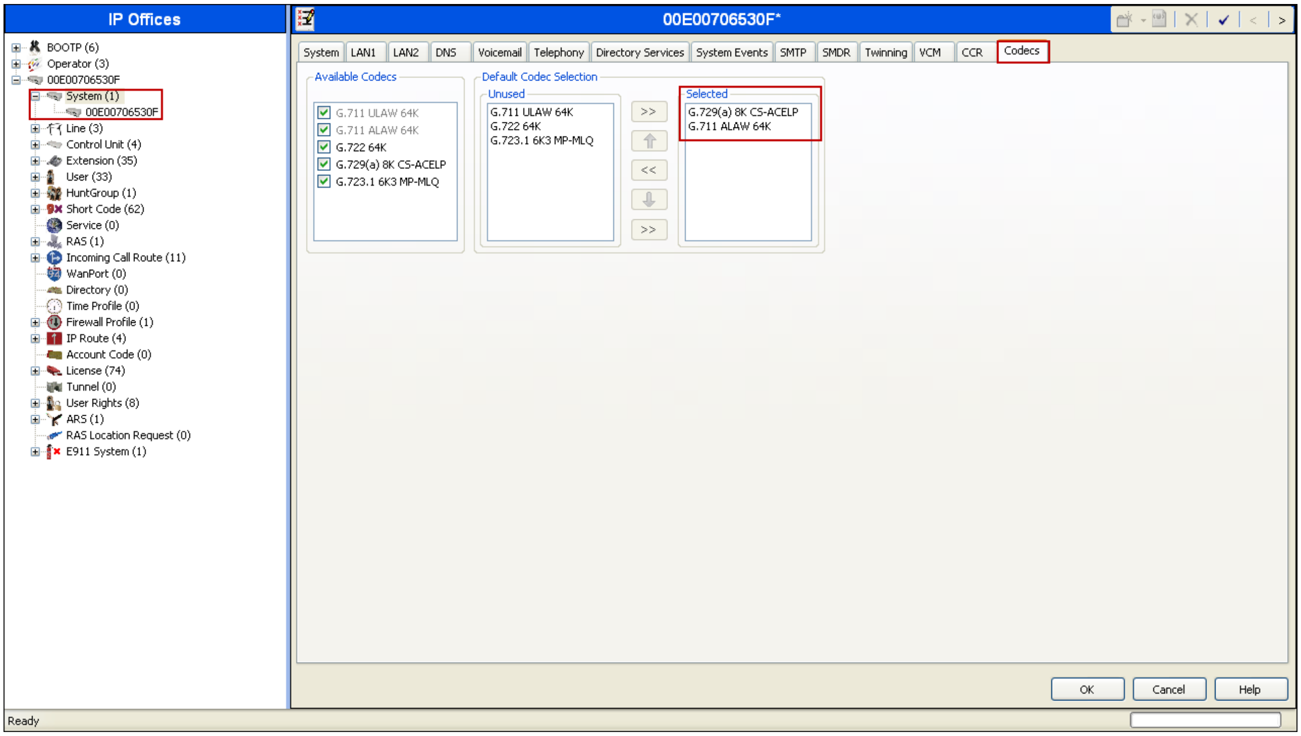
Twinning Calling Party Settings

The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. With this configuration, and related configuration of **Remote Party ID** on the SIP Line (Section **5.4**), the true identity of a PSTN caller can be presented to the twinning destination.



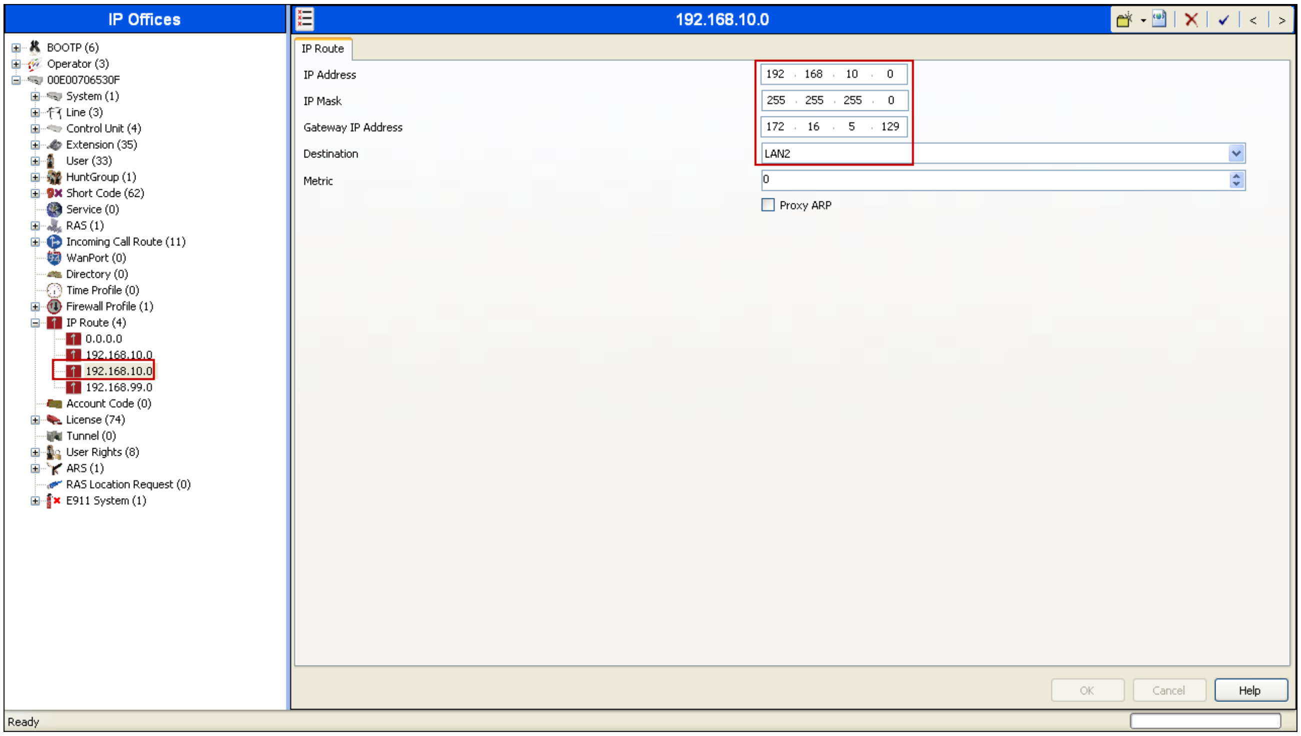
Codecs Tab

The **System →Codecs** tab is new in Avaya Release 8.0 and later. 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.



IP Route

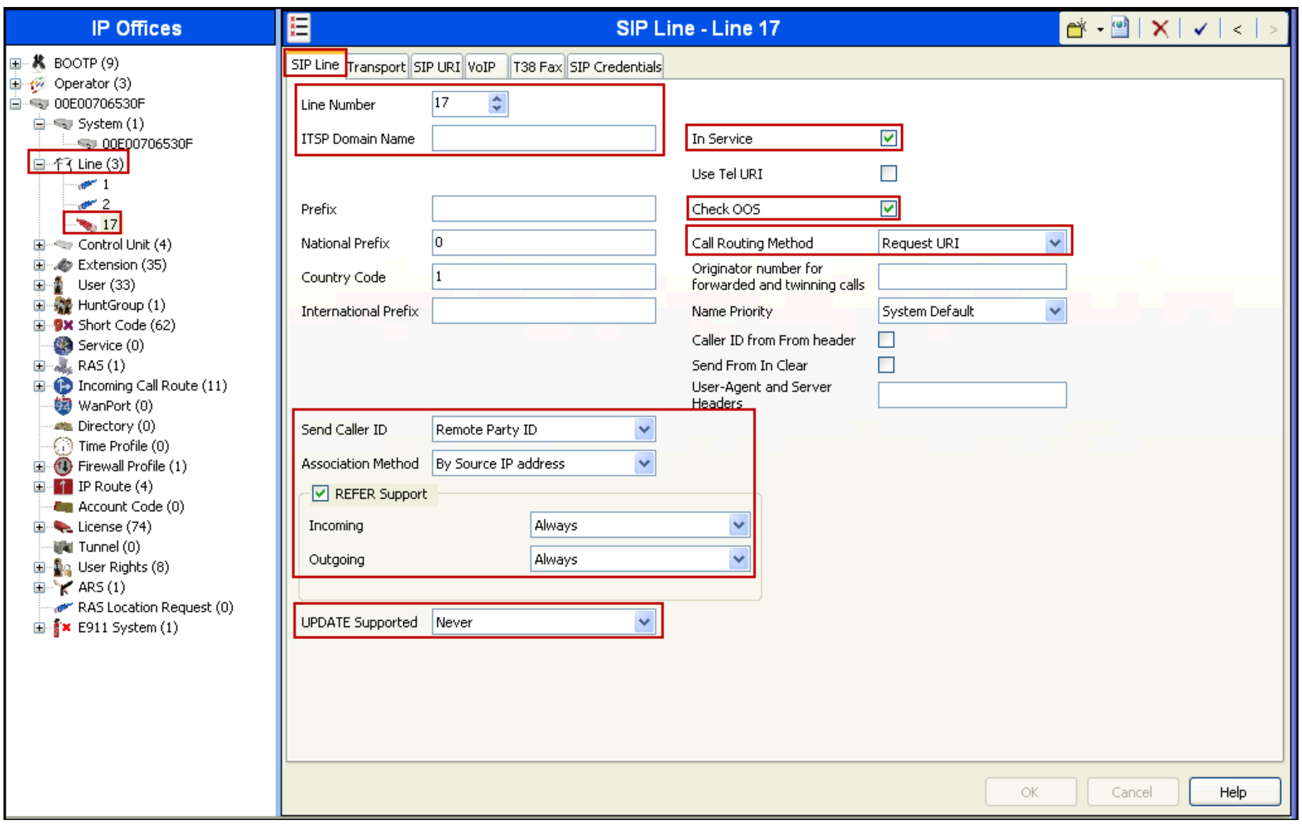
Create an IP route to specify the gateway or router address where the Avaya IP Office needs to send the packets, in order to reach the SIP proxy on Alestra's network. On the left navigation pane, right-click on **IP Route** and select **New**. The values used during the compliance test are shown on the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



Administer SIP Line

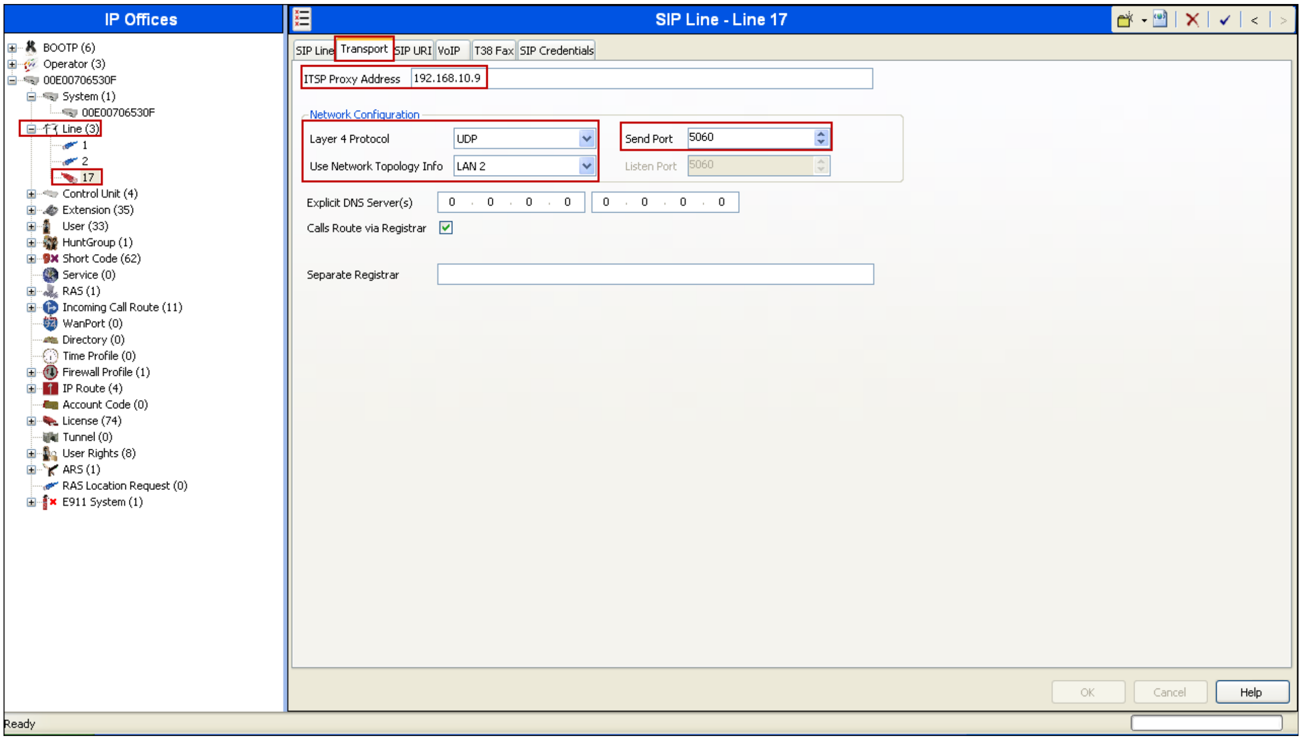
A SIP line is needed to establish the SIP connection between Avaya IP Office and Alestra Enlace IP SIP Trunking service. To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the enterprise domain so that Avaya IP Office uses this domain as the host portion of SIP URI in SIP headers such as the From header. This field was left blank since Alestra uses IP addresses instead of domain name.
- Set **Send Caller ID** to **Remote Party ID**.
- 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 the **Call Routing Method** to **Request URI** which is the default value.
- Check the **REFER Support** and set **Incoming** and **Outgoing** to **Always.**
- Set the **UPDATE Supported** to **Never** which is the default value.



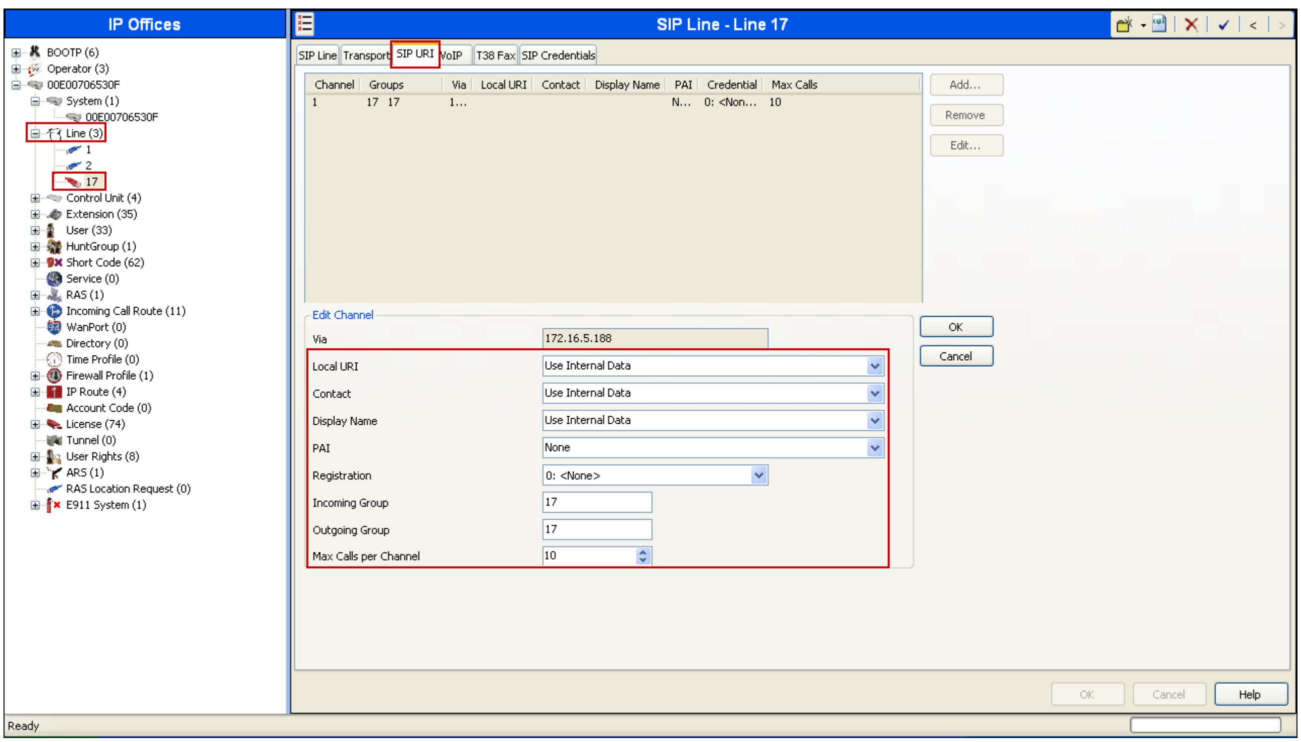
Select the **Transport** tab and set the following:

- Set the **ITSP Proxy Address** to the **Signaling** IP address of Alestra's 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.



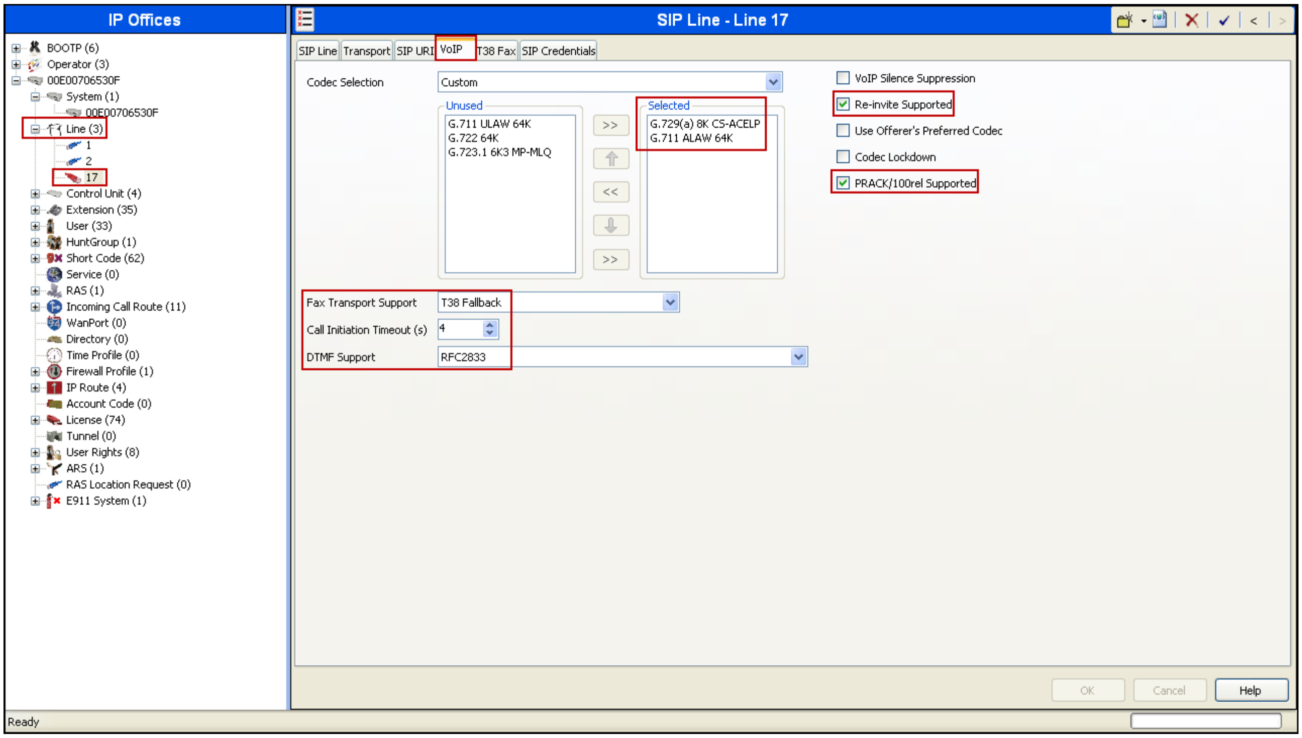
Select the **SIP URI** tab. A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Under the **SIP URI** tab, click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in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

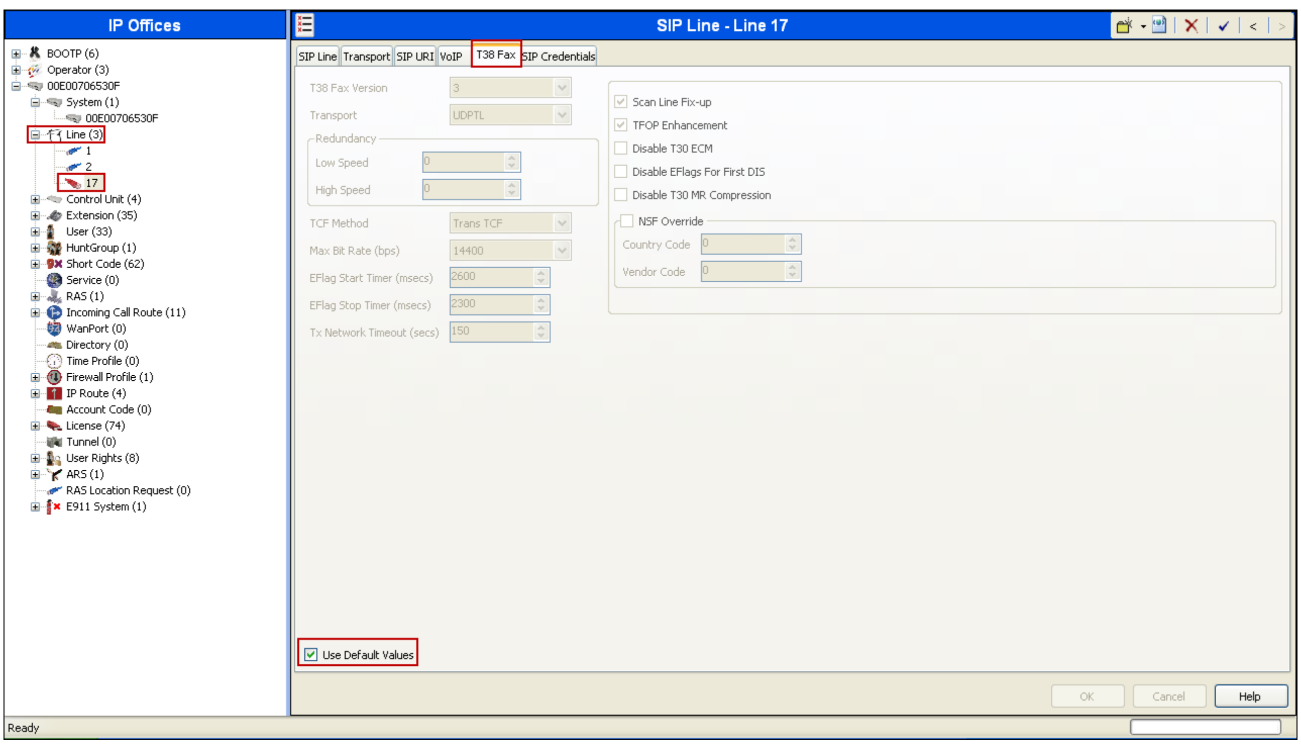


Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- In the sample configuration, the **Codec Selection** was configured using the **Custom** option, allowing an explicit ordered list of codecs to be modified for the testing, different from the system default defined in **Section 5.2.6**. The buttons allows configuration of an explicit list of codecs to be used on the line, in that specific order of preference. During normal circumstances the codec selection option can be set to **System Default** instead of **Custom**, this way the system default defined in **Section 5.2.6** will be used.
- 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.
- 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.



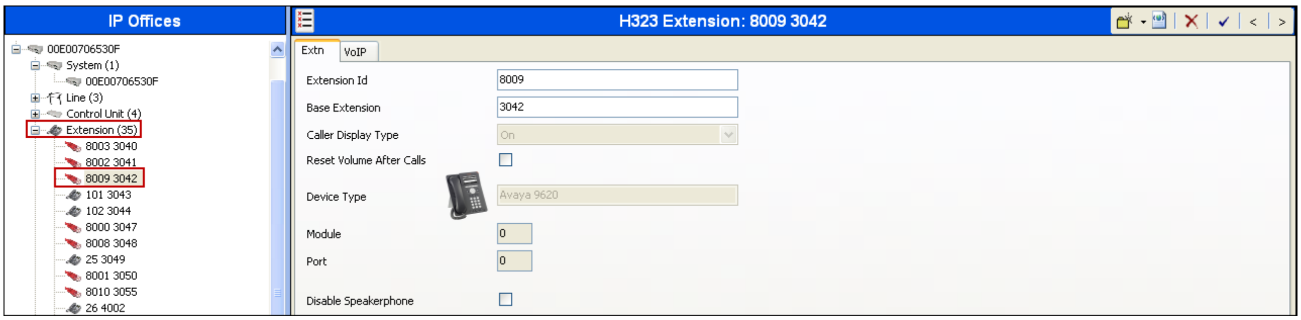
Select the **T.38 Fax** tab, check **Use Default Values.**



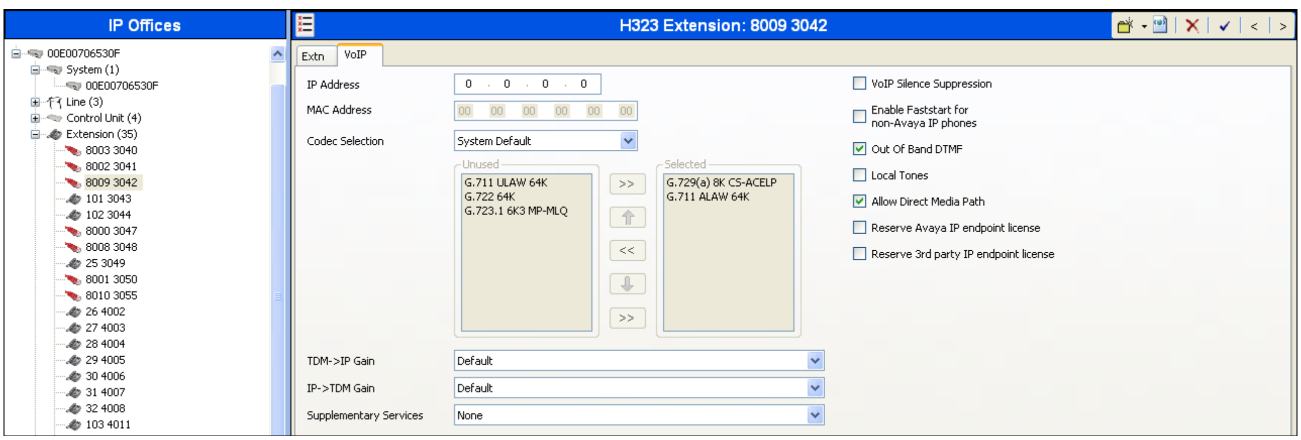
Extension

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.

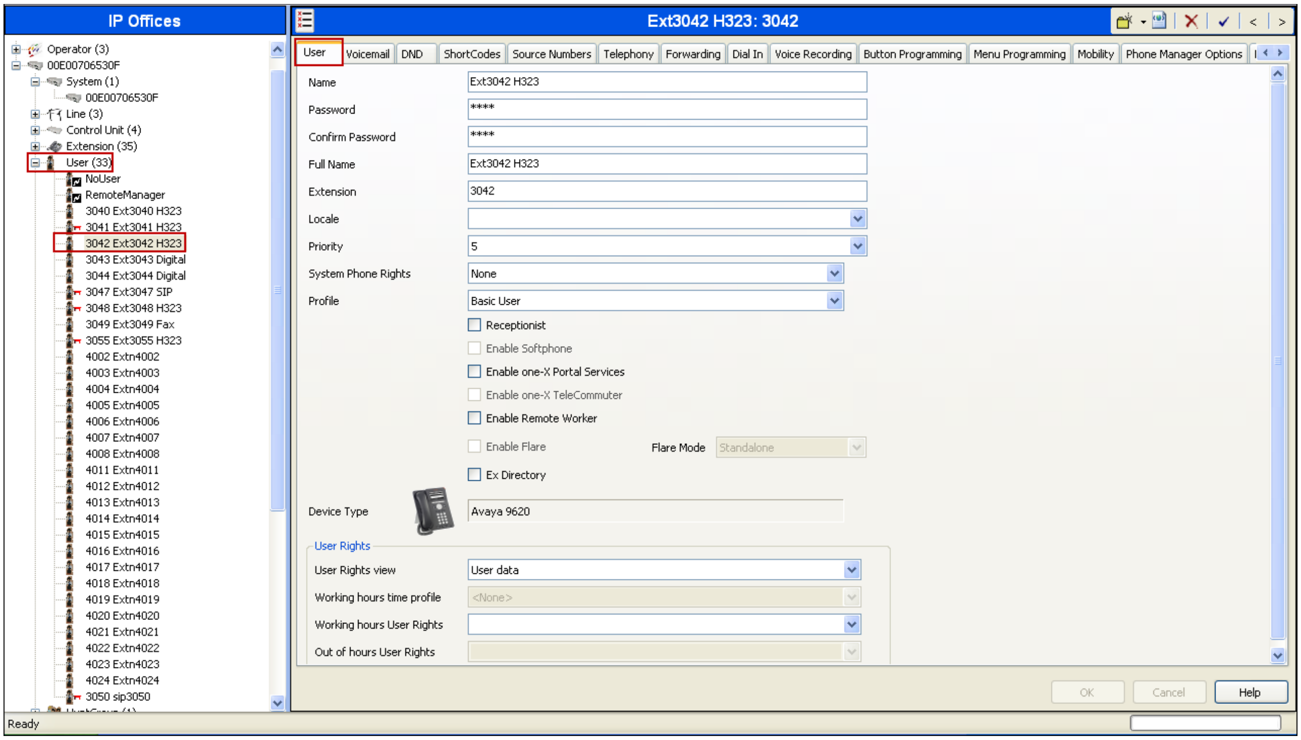


Select the **VOIP** tab. Use default values on VoIP tab. Following is an example for Extension 3042; this extension corresponds to an H.323 extension.

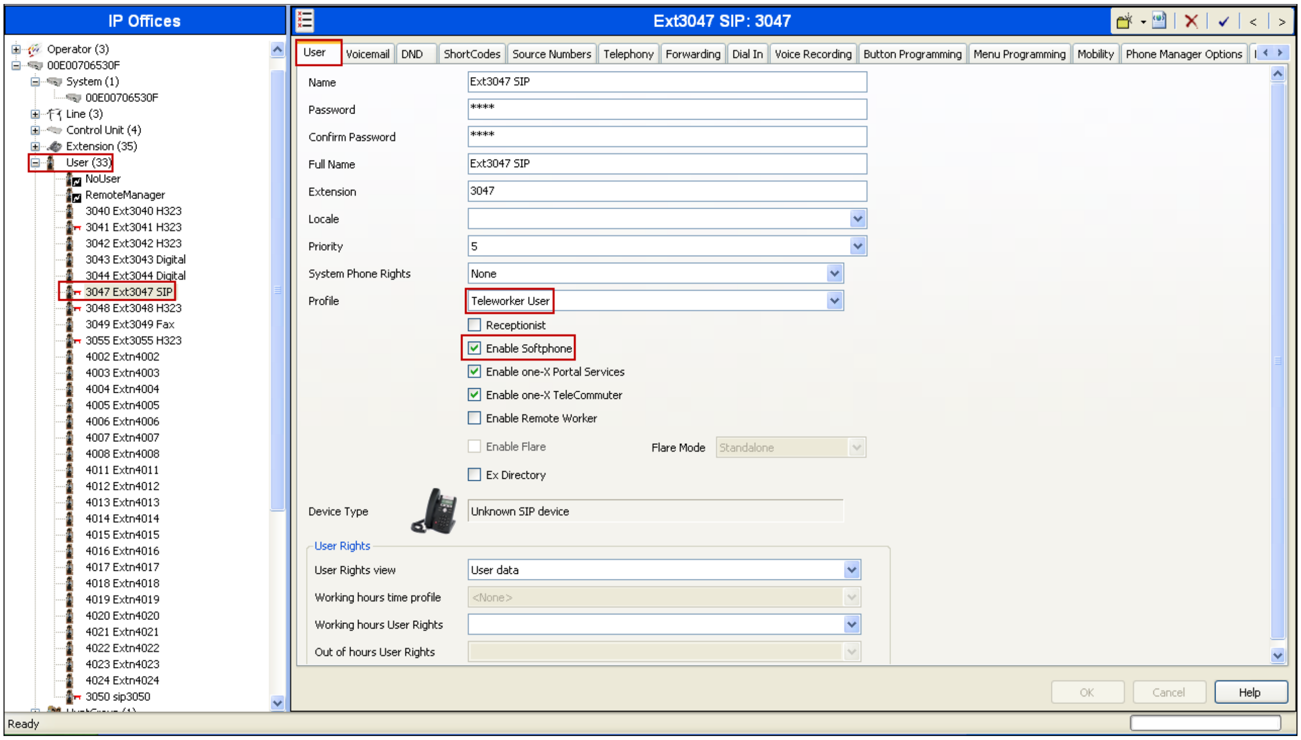


User

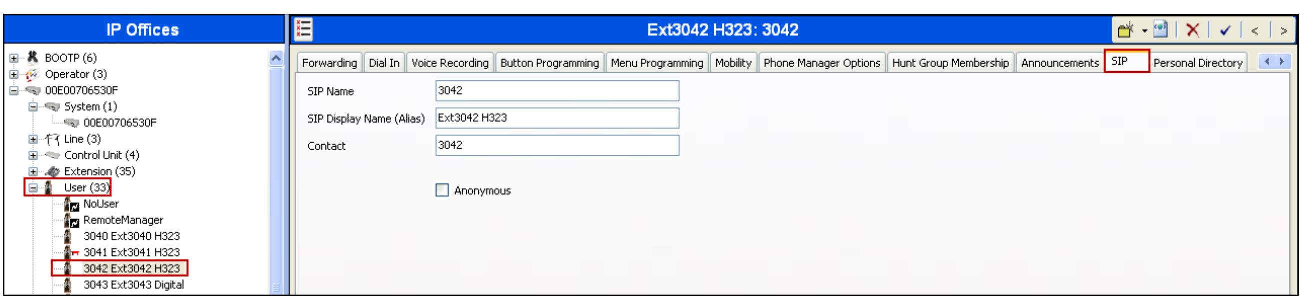
Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. To configure these settings, first expand **User** in the left Navigation Pane, and then select the name of the user to be modified. In the example below, the name of the user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



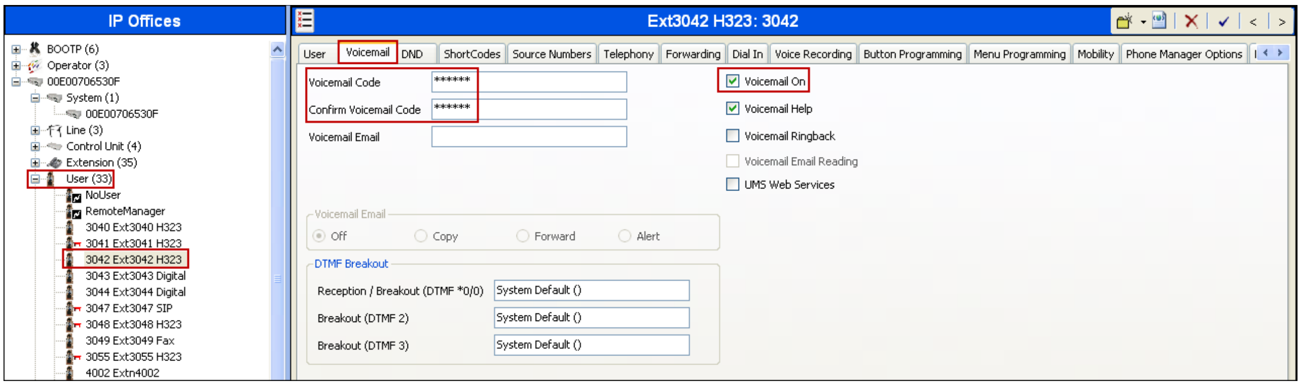
In the example below, the name of the user is “Ext3047 SIP”. This is an Avaya IP Office SIP Softphone user, set the Profile to **Teleworker User** and check **Enable Softphone**.



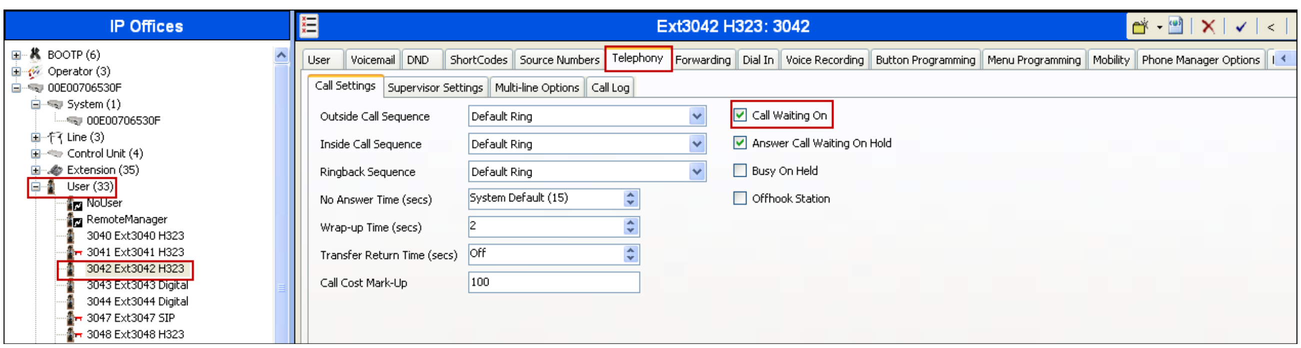
Select the **SIP** tab. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the “From” header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for 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. Note that only 4 digits of the DID number were assigned; this is because in this sample configuration Alestra’s network was configured to send only 4 digits 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.



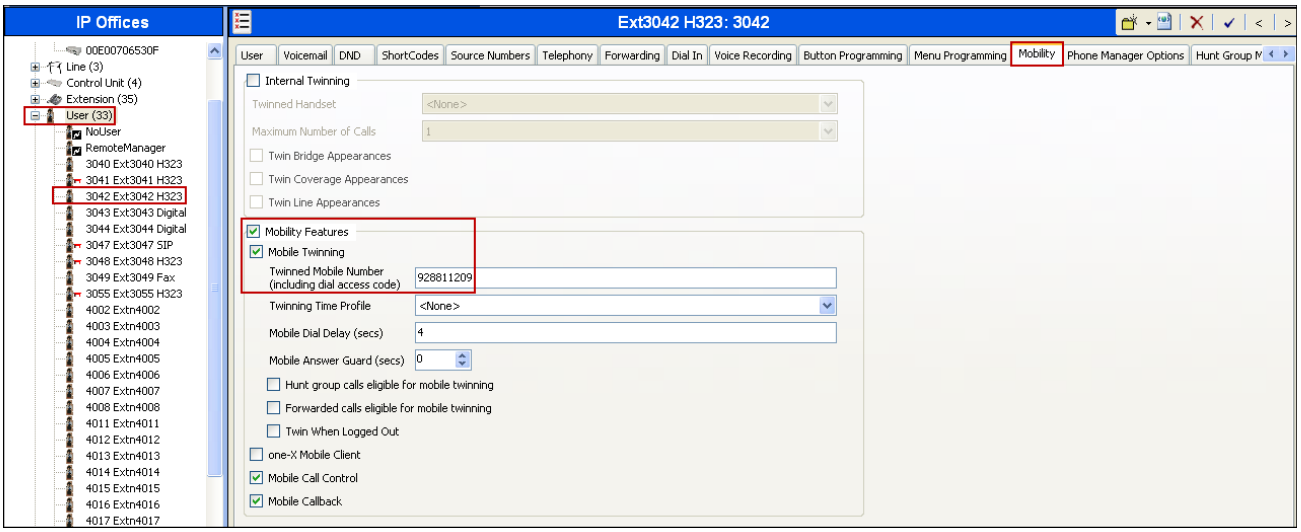
Select the **Voice Mail** tab. The following screen shows the **Voicemail** tab for the user with extension 3042. The **Voicemail On** box is checked. Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from Alestra Enlace IP 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.



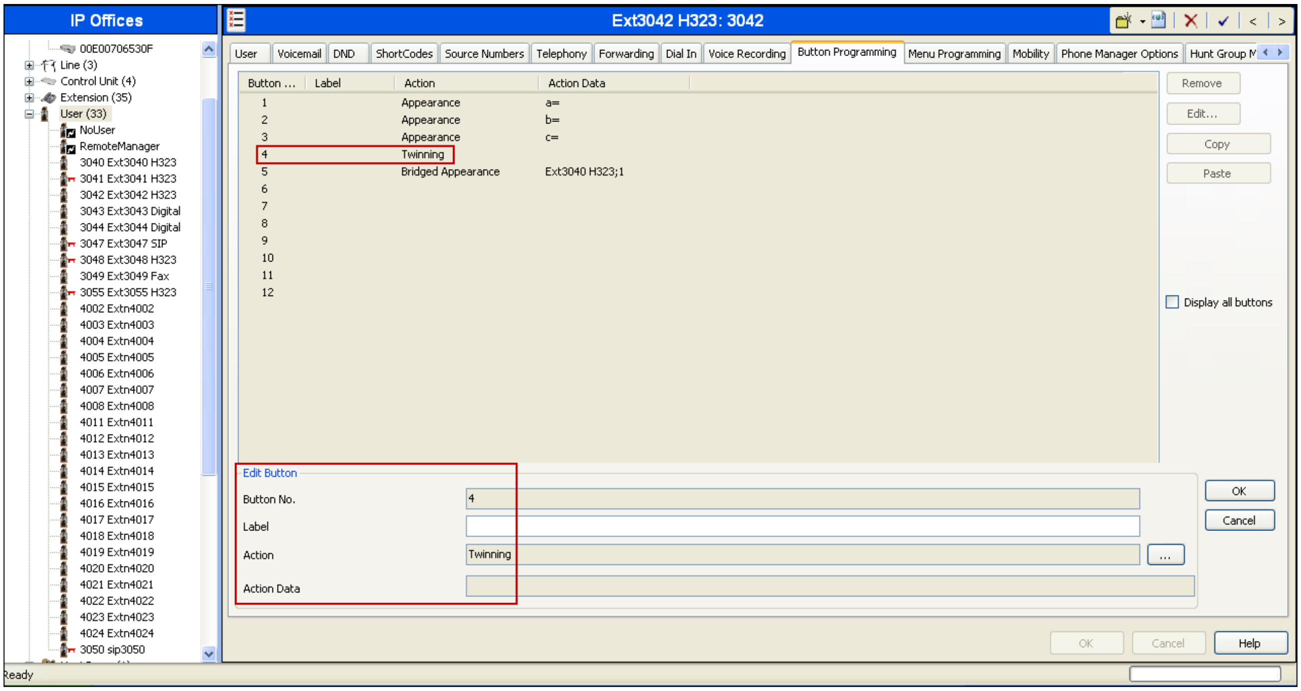
Select the **Telephony** tab, then **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an Avaya IP Office phone logged in as this extension to have multiple call appearances. Note: **Call Waiting On** is necessary for call transfer.



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 **928811209**. Other options can be set according to customer requirements.

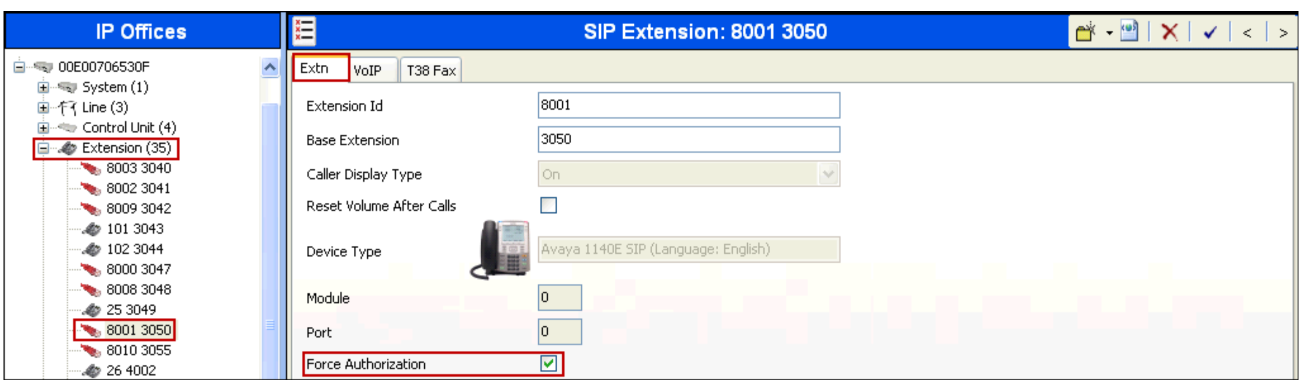


To program a key on the telephone to turn Mobil Twinning on and off, select the **Button Programming** tab on the user, then select the button to program to turn Mobil Twinning on and off, click on **Edit → Emulation → Twinning**. In the sample below, button **4** was programmed to turn Mobil Twinning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

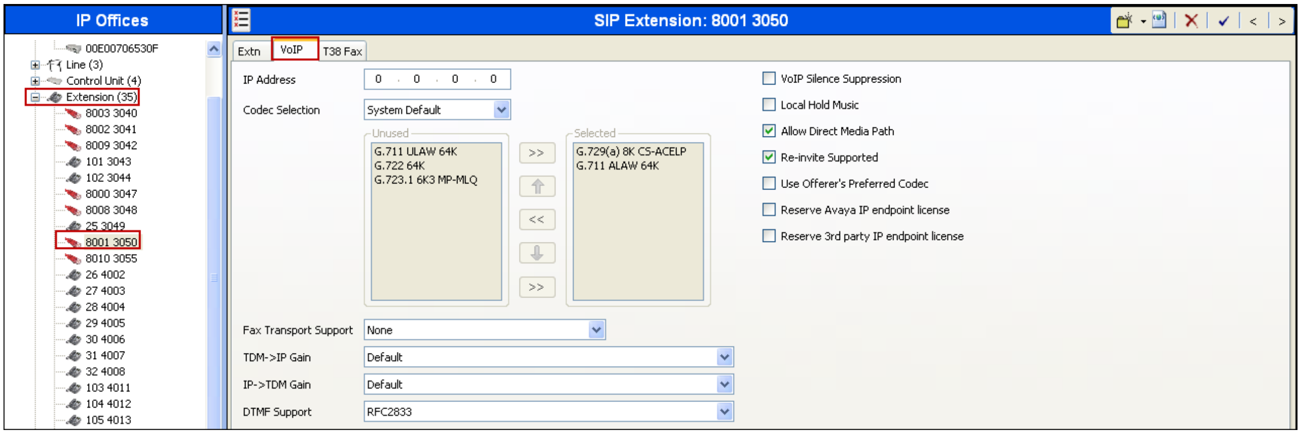


SIP Telephone Users (Avaya 1140E)

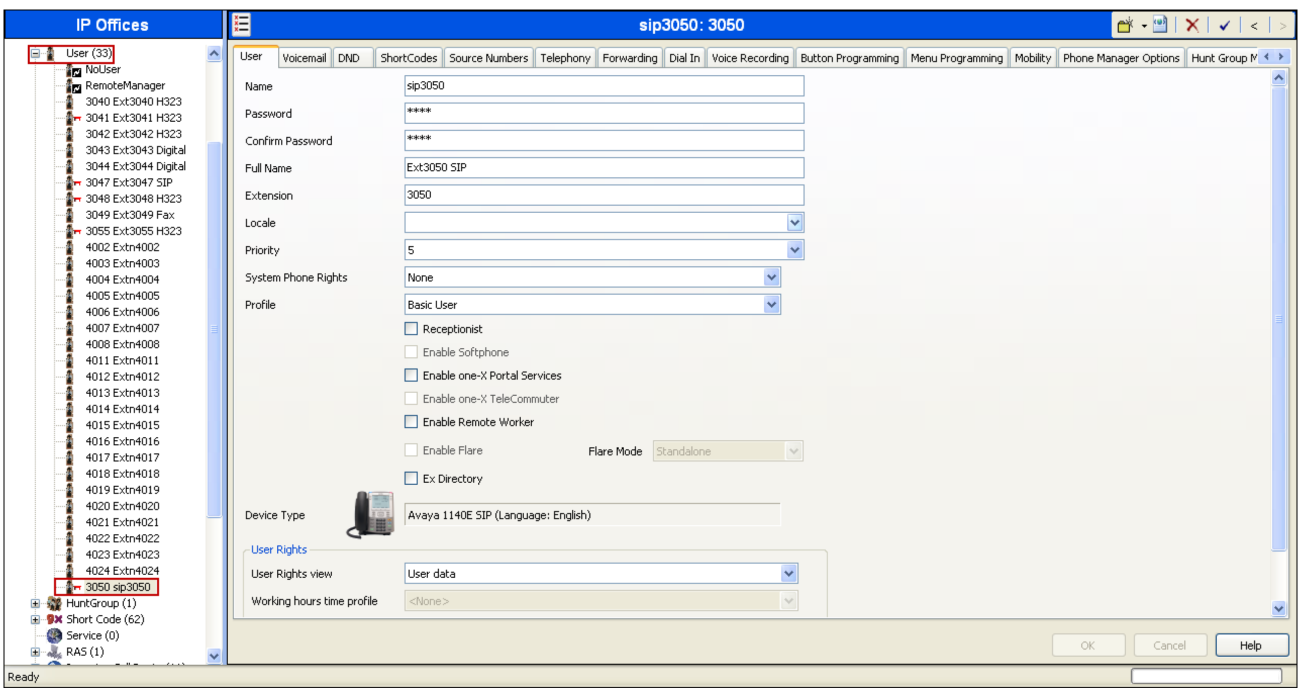
This section will summarize aspects of the completed configuration for the Avaya 1140E (the Avaya 1120 may also be use). A new SIP extension may be added by right-clicking on **Extension** in the Navigation pane and selecting **New SIP Extension**. Alternatively, an existing SIP extension may be selected 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.



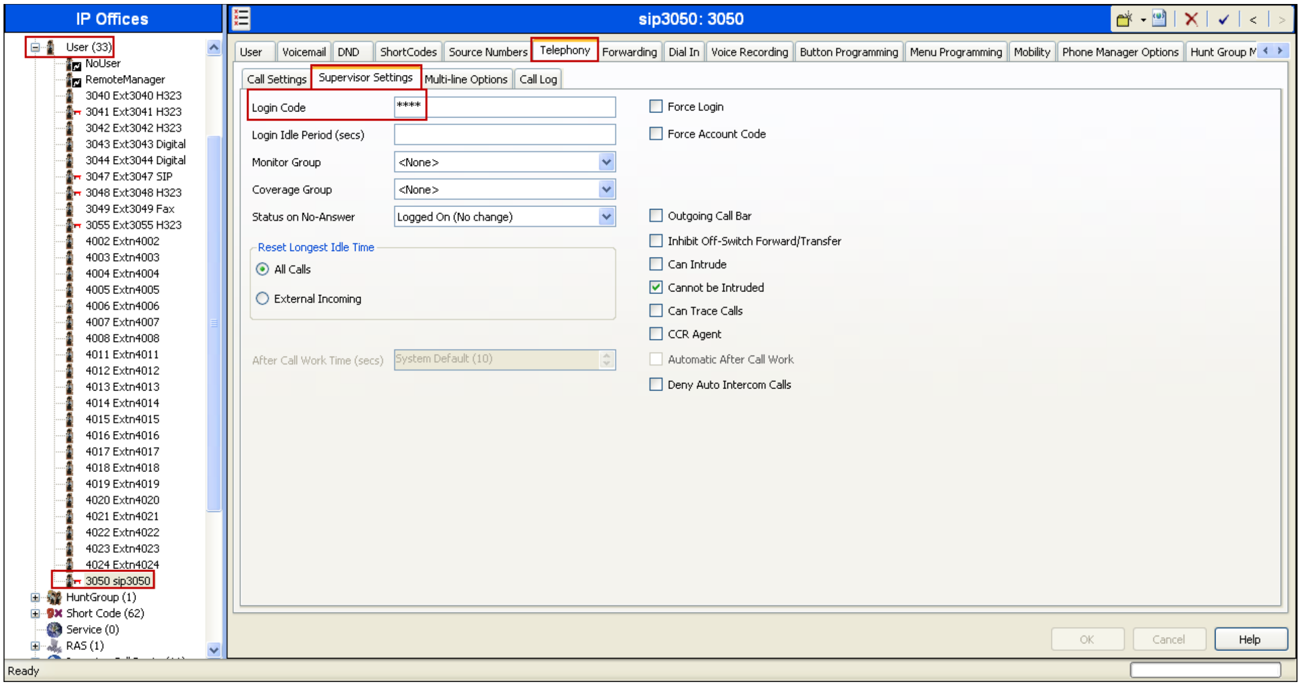
The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. The new **Codec Selection** parameter may retain the default setting “System Default” to follow the system configuration shown in **Section 5.2.6.** Alternatively, “Custom” may be selected to allow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



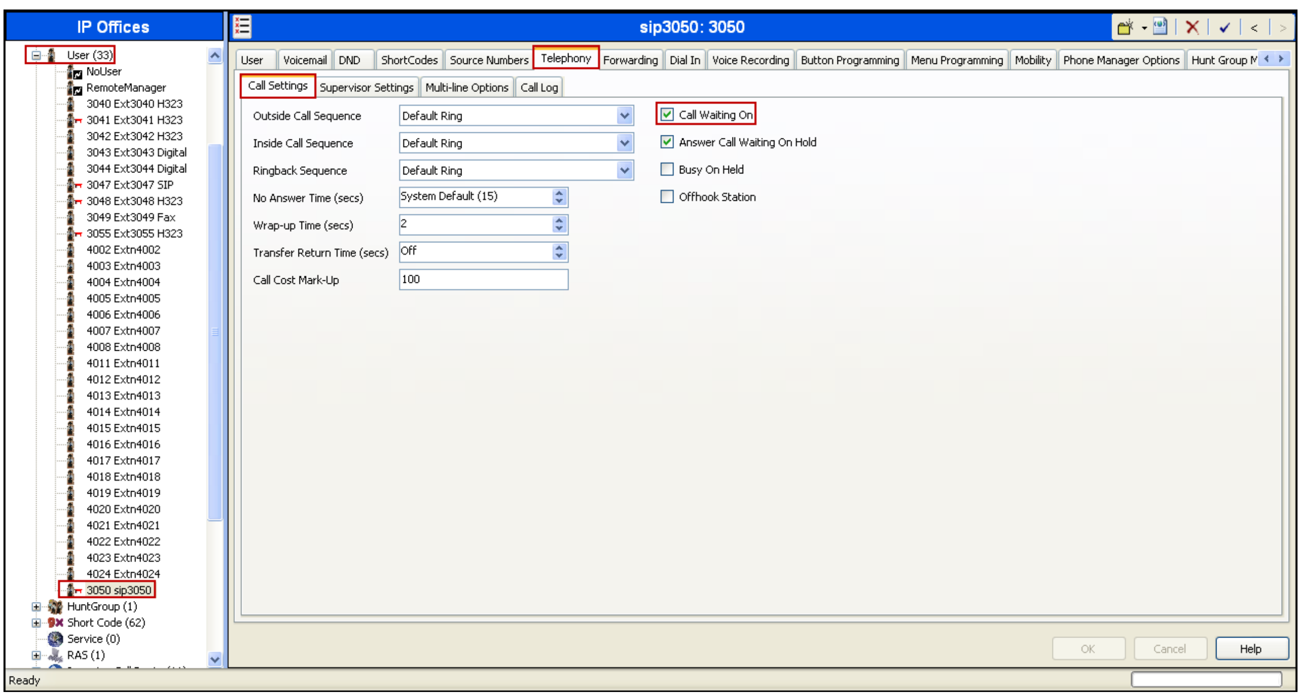
The following screen shows the **User** tab for User 3050 corresponding to an Avaya 1140E. The **Extension** parameter is populated with extension 3050.



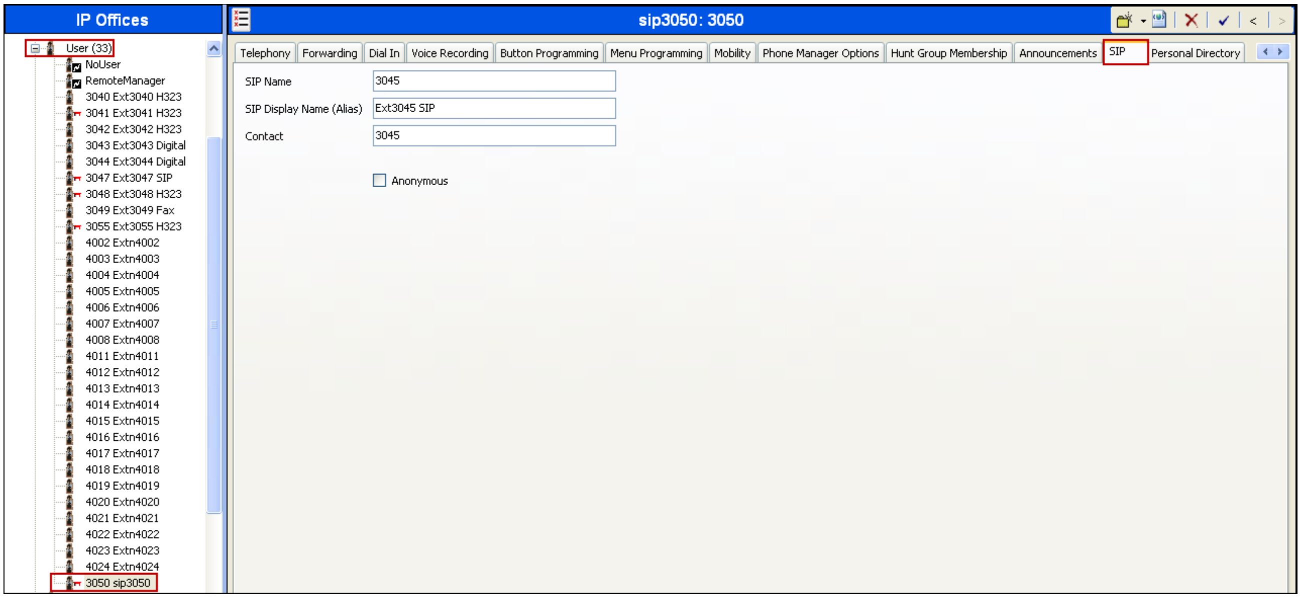
Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Avaya 1140E telephone user as the login password.



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.



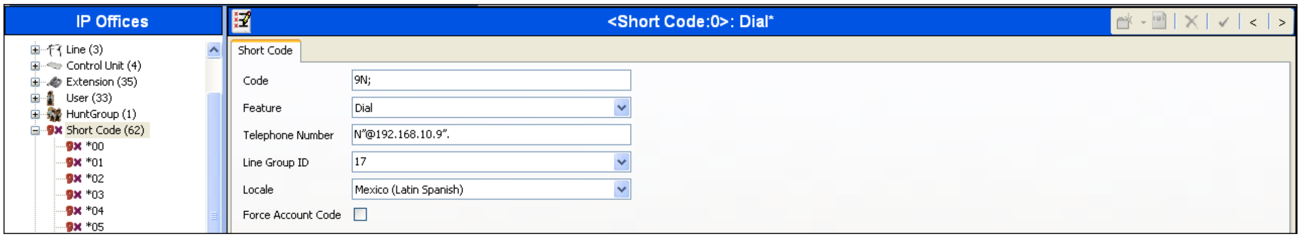
Like other users previously illustrated, the **SIP** tab for the user with extension 3050 is configured with a **SIP Name** and **Contact** specifying one of the DID number provided by Alestra. Note that only 4 digits of the DID number were assigned; this is because in this sample configuration Alestra's network was configured to send only 4 digits to the enterprise.



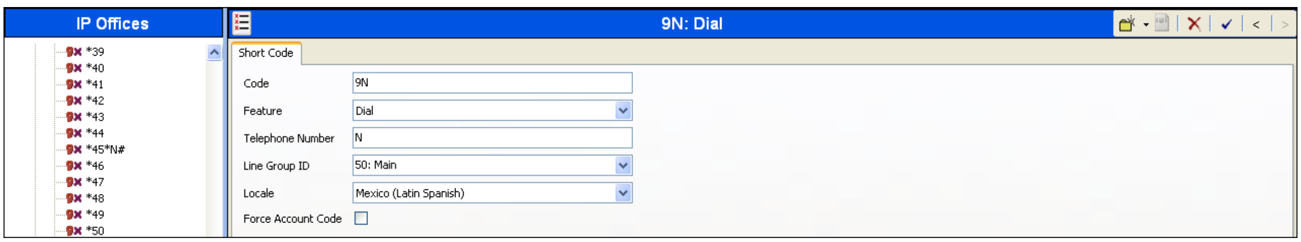
Short Code

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"@192.168.10.9"**. 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 Alestra 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.



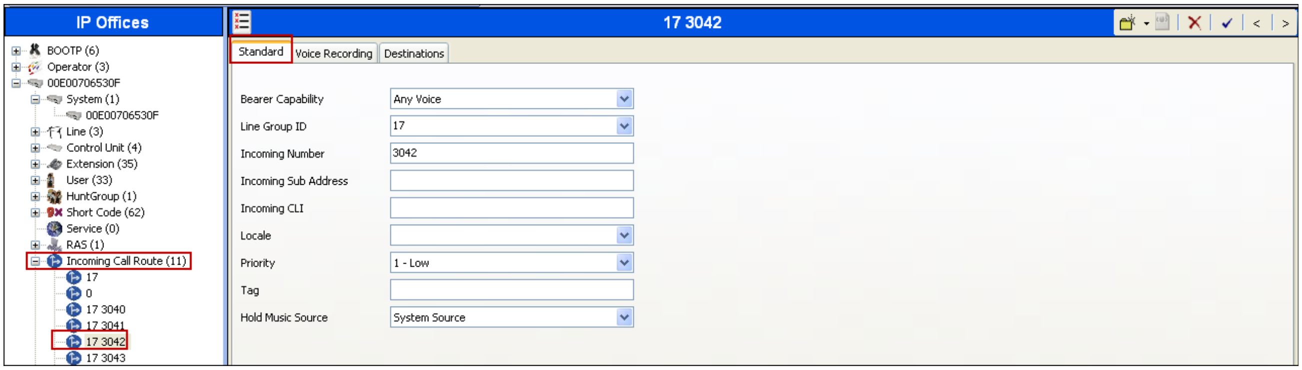
The simple "9N;" short codes illustrated above does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code 9N is illustrated for access to ARS. When the Avaya IP Office user dials 9 plus any number N, rather than being directed to a specific **Line Group Id**, the call is directed to **Line Group ID** "50: Main", configurable via ARS. See **Section 5.11** for example ARS route configuration for "50: Main" as well as a backup route. Both methods were used during the compliance testing.



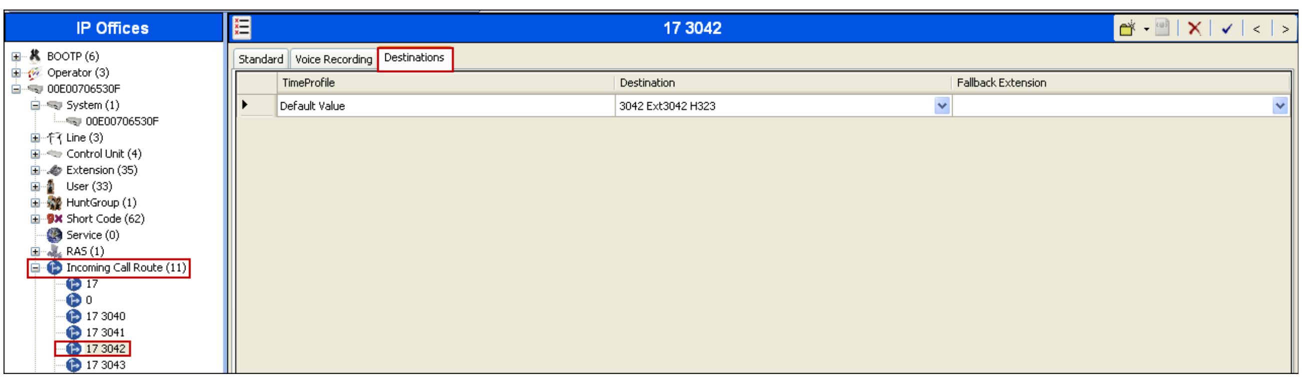
Incoming Call Routing

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 3042 on line 17 are routed to extension 3042. An incoming call route must be added for each DID number assigned to internal extensions.



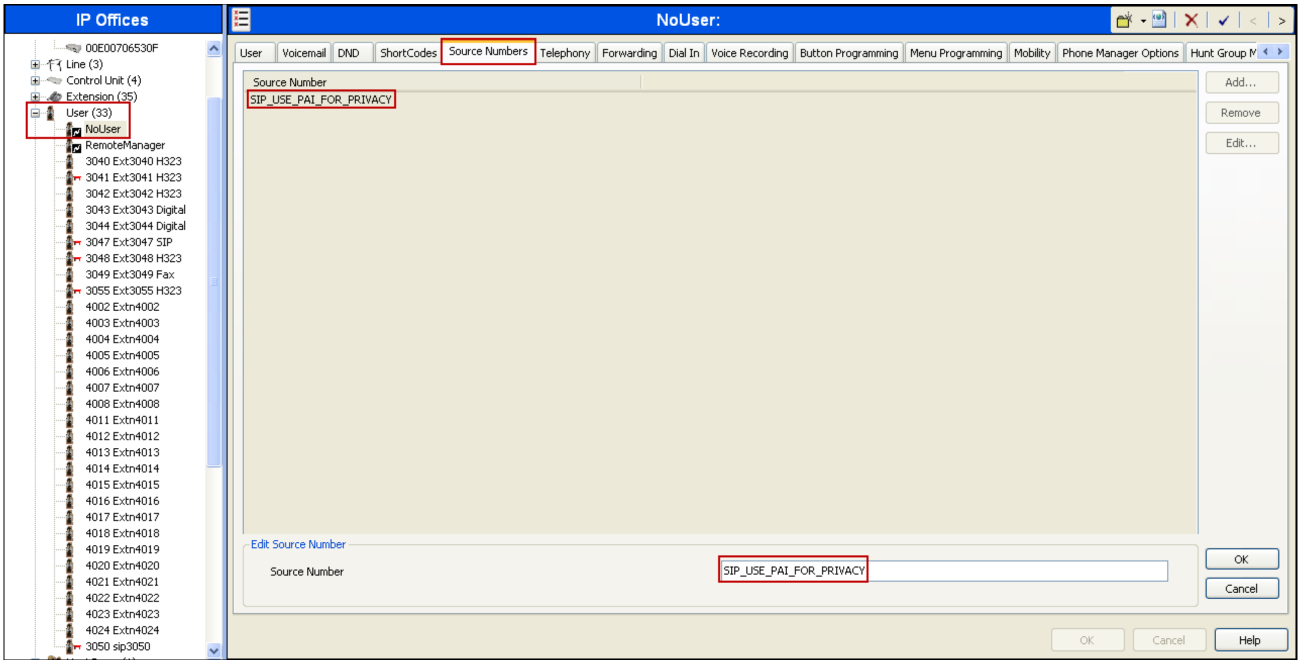
Privacy/Anonymous Calls

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.



ARS and Alternate Routing

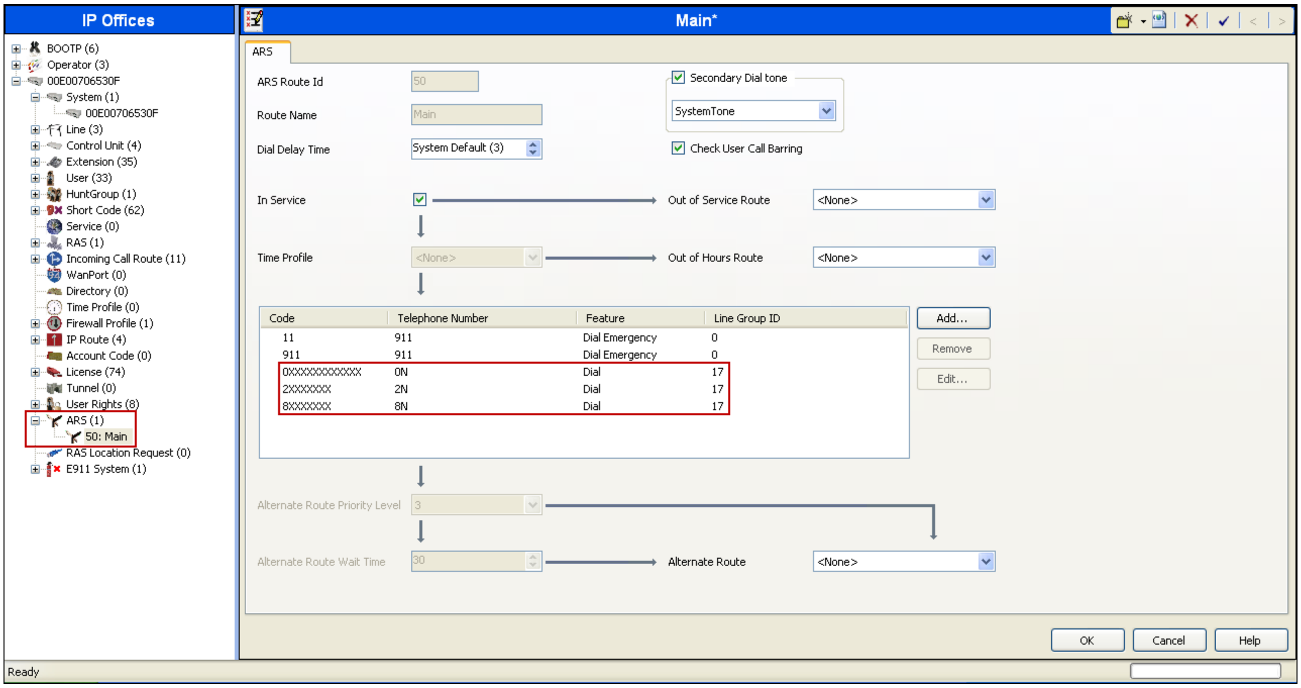
While detailed coverage of Automatic Route Selection (ARS) is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations. ARS is illustrated here mainly to illustrate alternate routing should the SIP Line be out of service or temporarily not responding.

Optionally, ARS can be used rather than the simple "9N;" short code approach documented in **Section 5.8**. With ARS, a secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, 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. ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code.

To add a new ARS route, right-click on **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named "Main". The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter.

Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., 9N in **Section 5.8**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen shown below, the user can dial any number after dialing 90, by doing so the call would be directed to Line Group 17, the SIP Line configured and described in these Application Notes. Per the example screen shown below the user can also dial any number after dialing 92 and 98.

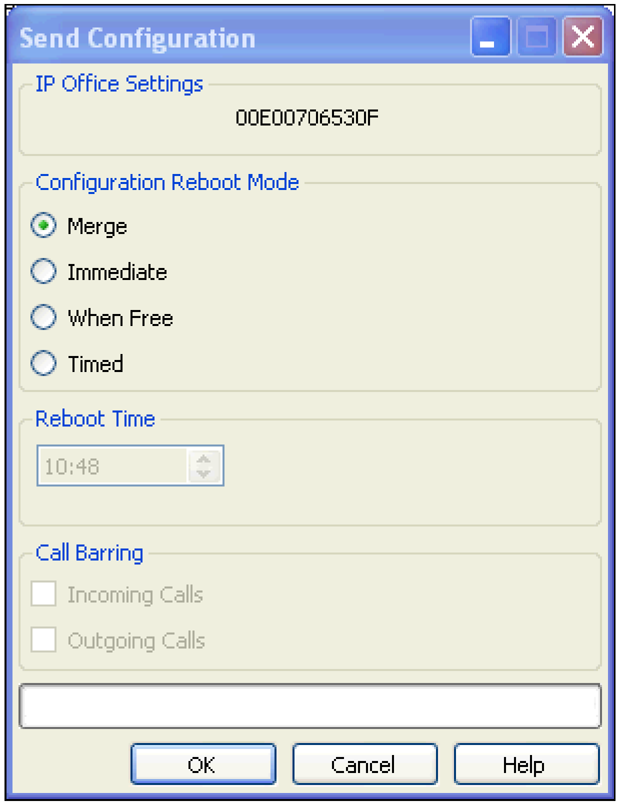


Save Configuration

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.



Alestra Enlace IP SIP Trunk Service Configuration

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

- Signaling IP address of Alestra SIP proxy Server.
- Supported codecs
- DID numbers

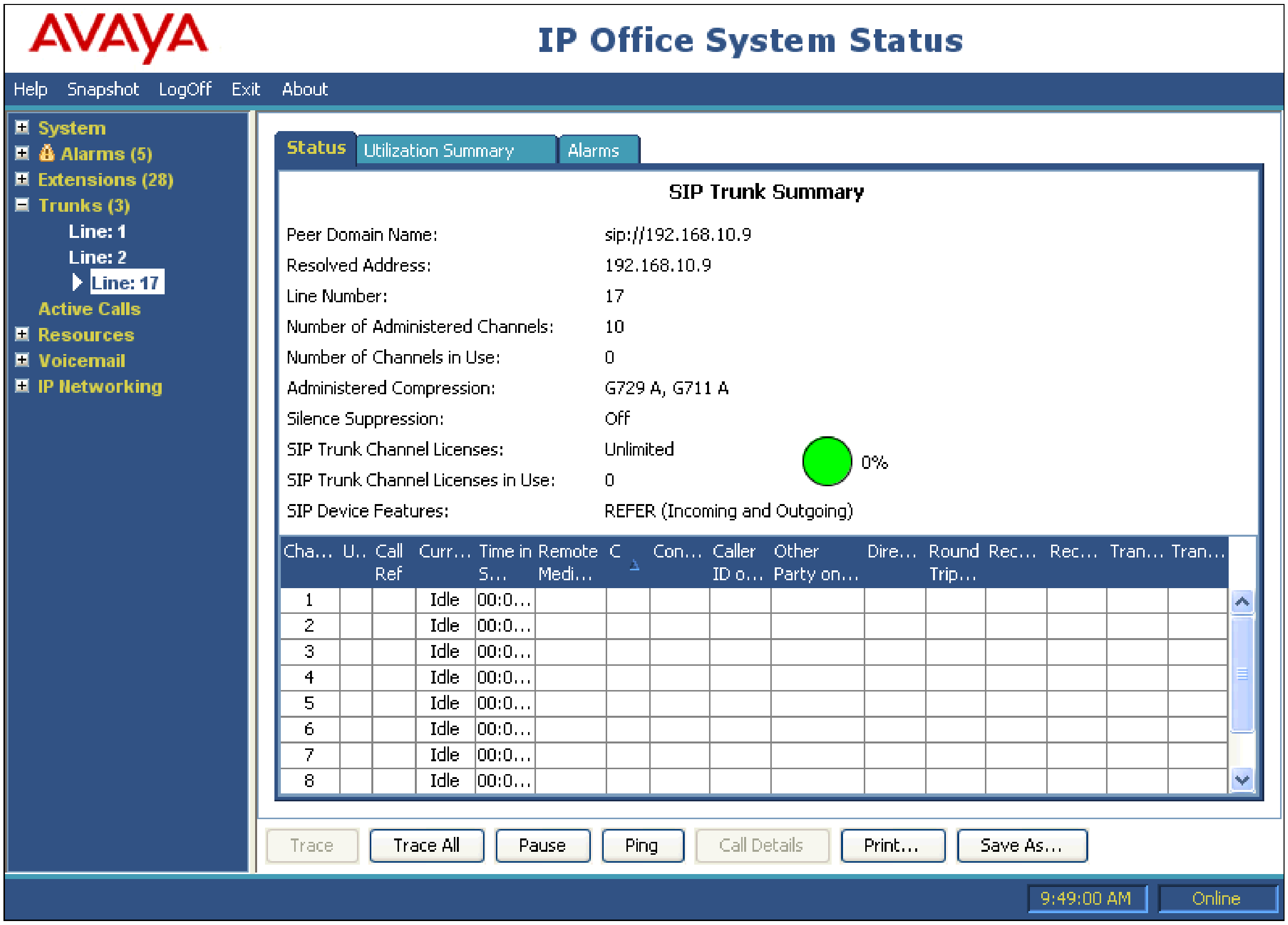
- IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

Verification Steps

The following steps may be used to verify the configuration:

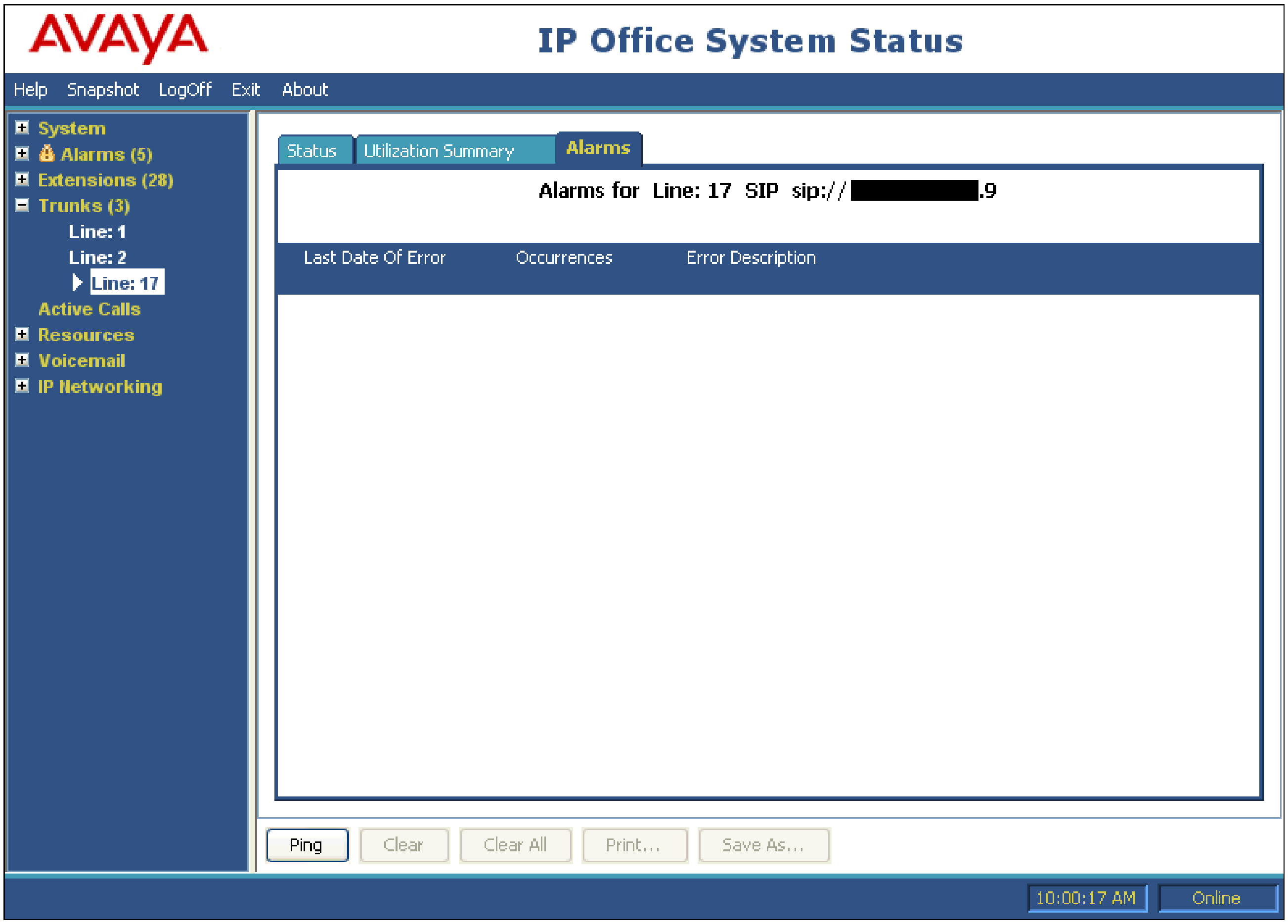
Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start →Programs→IP Office→System Status** on the PC where Avaya IP Office Manager was installed. Log in using the appropriate credentials.

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).



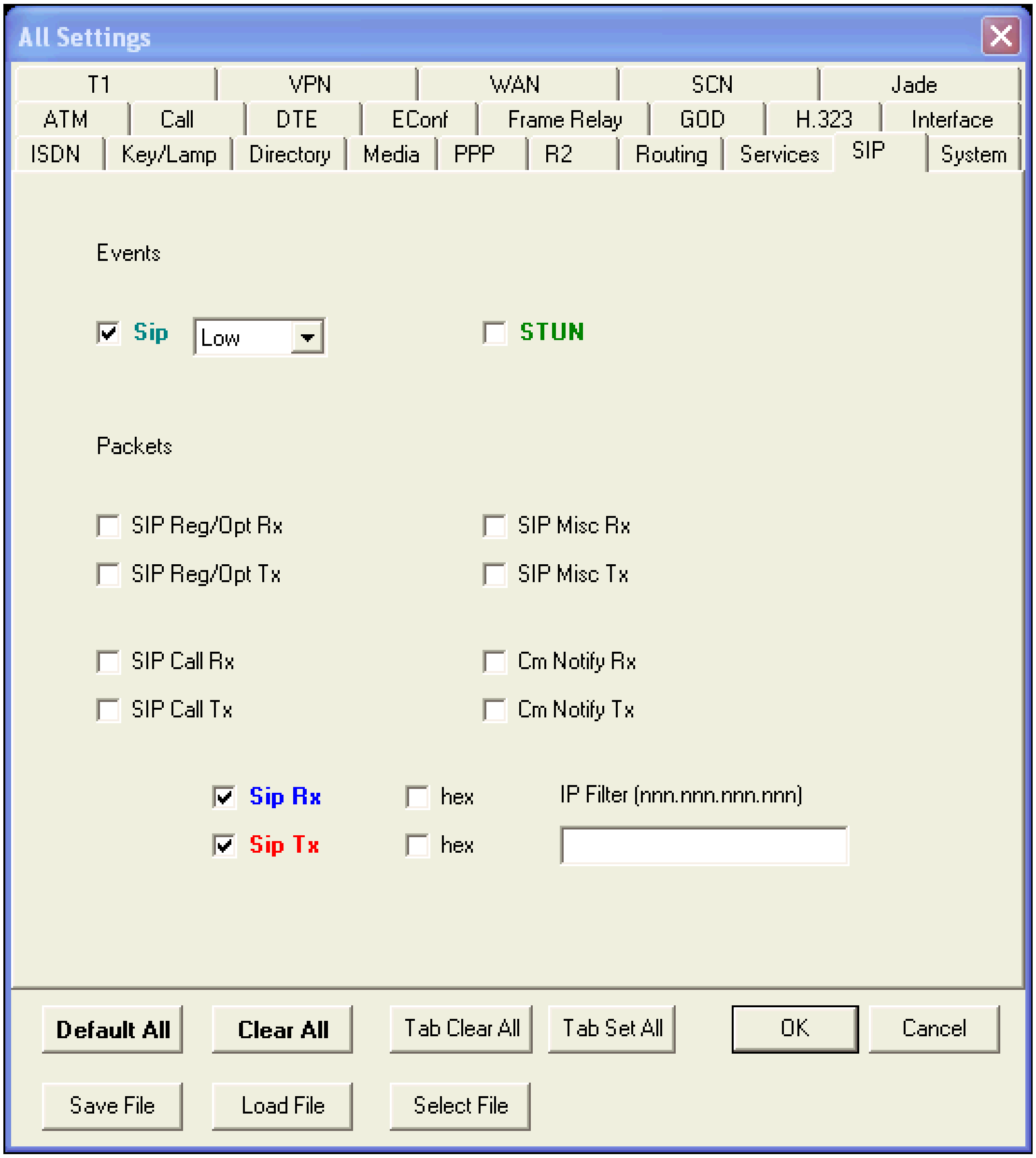
Cha...	U..	Call Ref	Curr...	Time in S...	Remote Medi...	C	Con...	Caller ID o...	Other Party on...	Dire...	Round Trip...	Rec...	Rec...	Tran...	Tran...
1			Idle	00:0...											
2			Idle	00:0...											
3			Idle	00:0...											
4			Idle	00:0...											
5			Idle	00:0...											
6			Idle	00:0...											
7			Idle	00:0...											
8			Idle	00:0...											

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**. 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.



Conclusion

The Alestra Enlace IP SIP Trunk Service passed compliance testing. These Application Notes describe the procedures required to configure the SIP trunk connectivity between Avaya IP Office Release 8.1 and the Alestra Enlace IP 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 8.1 Installation, Document Number 15-601042 Issue 26f, 30 July 2012.
[2] IP Office Manager 10.1, Document Number 15-601011 Issue 29o, August 2012.
[3] IP Office 8.1 Release 8.0 Administering Voicemail Pro, Document Number 15-601063 Issue 27b, June 2012
[4] RFC 3261 SIP: Session Initiation Protocol, http://www.ietf.org/

Appendix: SIP Line Template

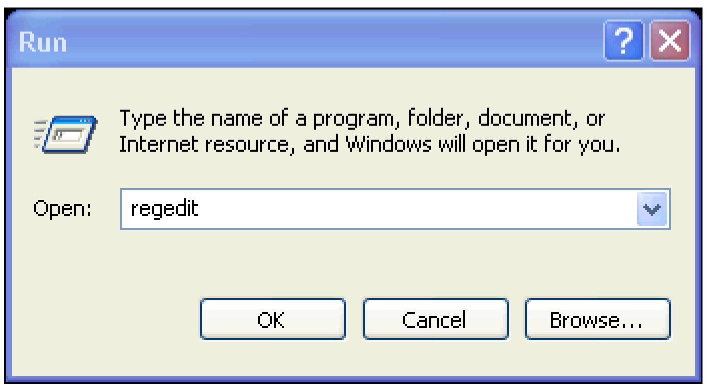
Avaya IP Office Release 8.1 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

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.

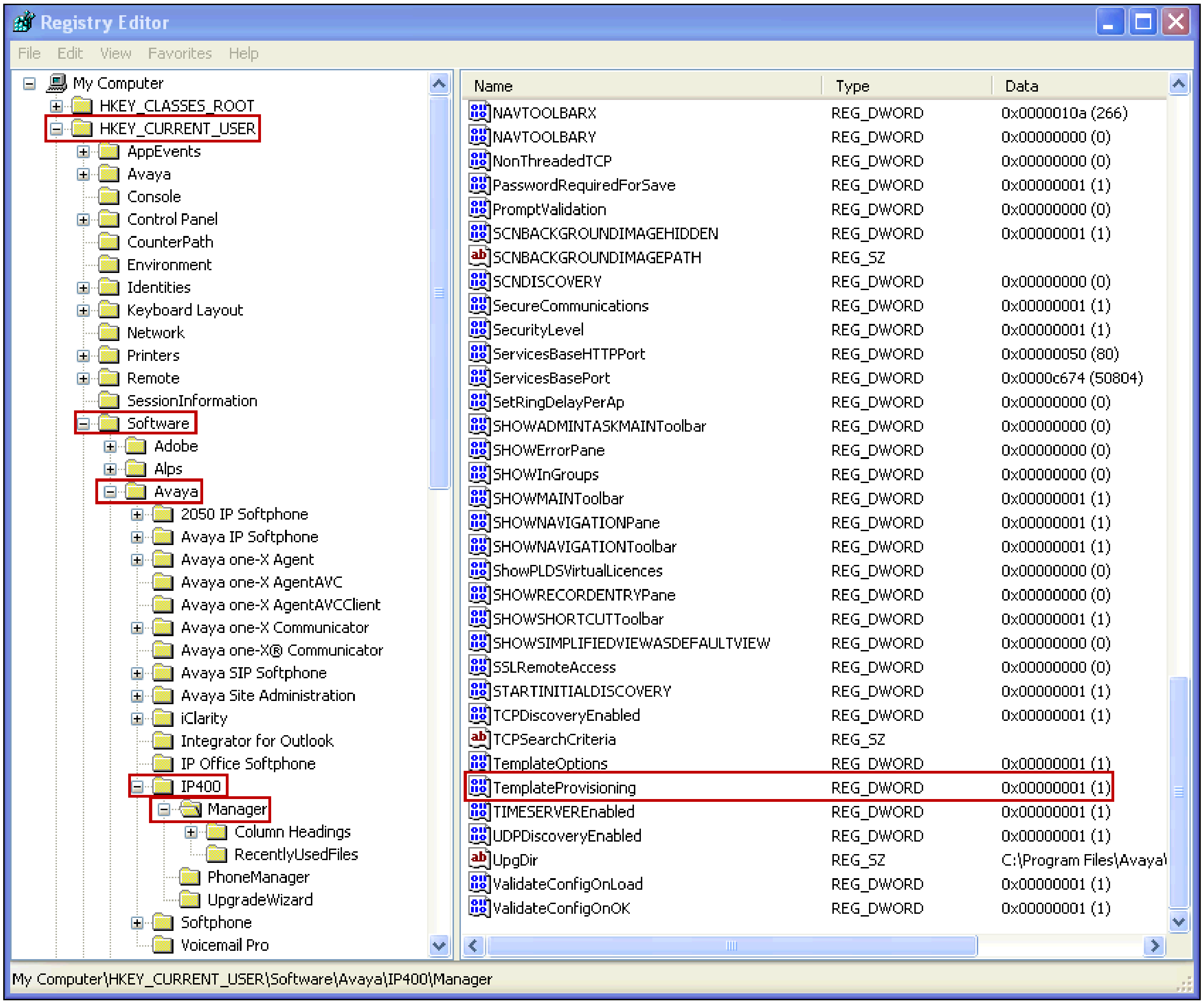
To create a SIP Line Template from the configuration described in these Application Notes, configure the parameters as described below.

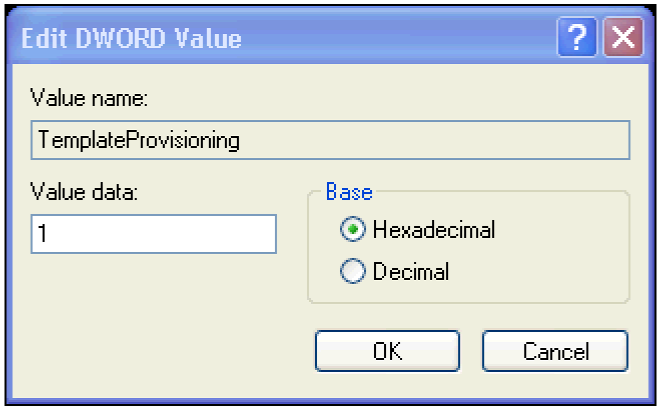
Create a new registry entry called **TemplateProvisioning** and set the **Value data** to **1**, as follows:

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

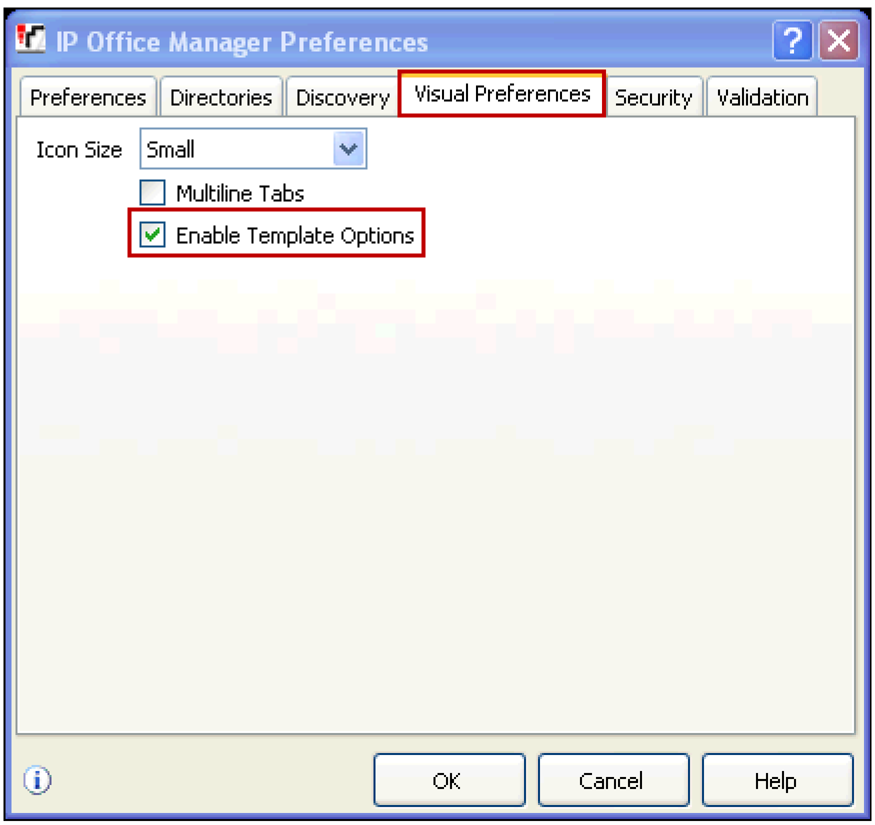


Under **HKEY_CURRENT_USER, Software, Avaya, IP400**, right click on **Manager**, then select **New, DWORD value,** then rename the newly created entry to: **TemplateProvisioning.** Right click on the newly created entry and select **Modify,** change the value under **Value Data** from "**0**" to "**1**".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





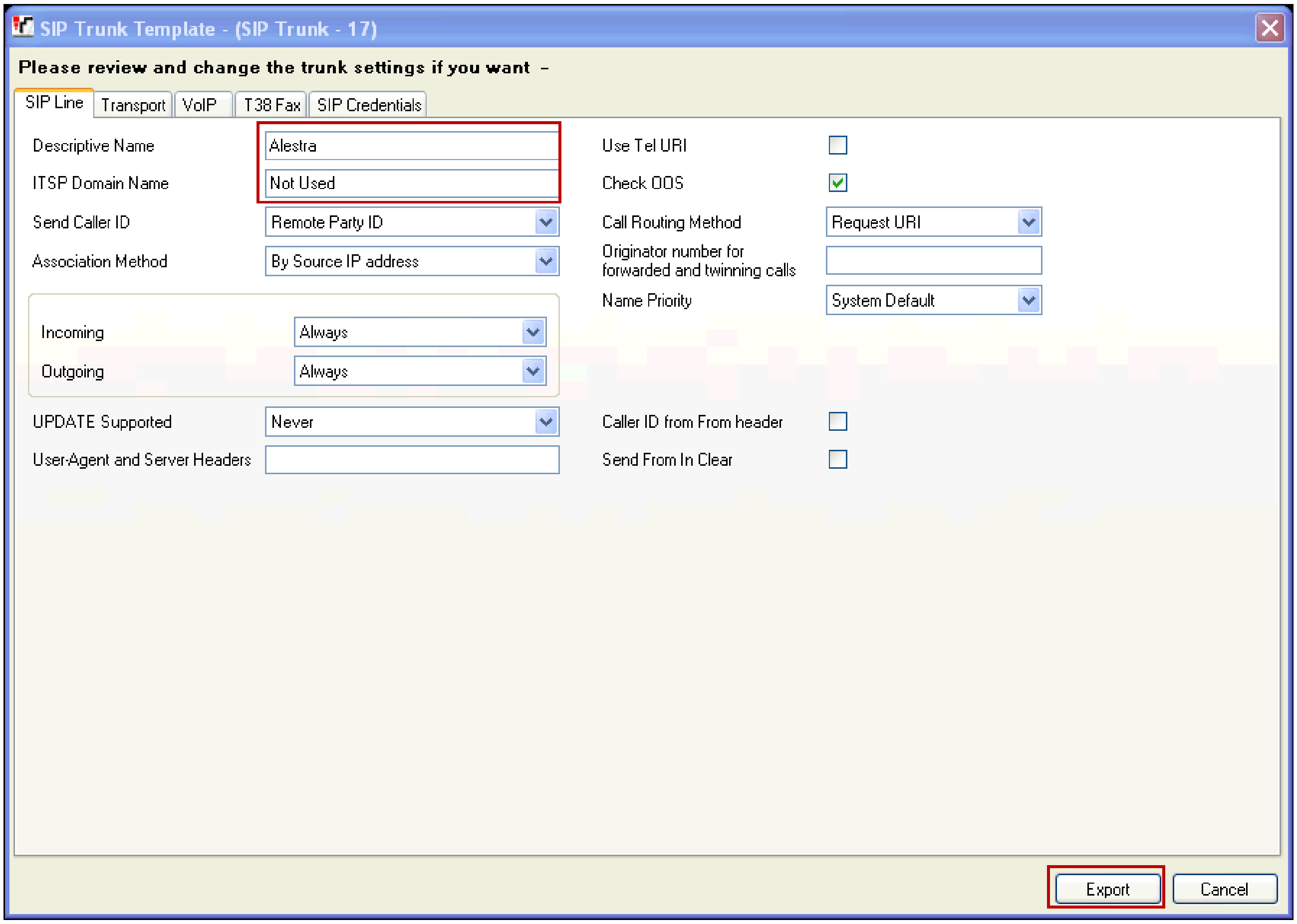
To enable template support go to **IP Office Manager**, select **File**, and then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.



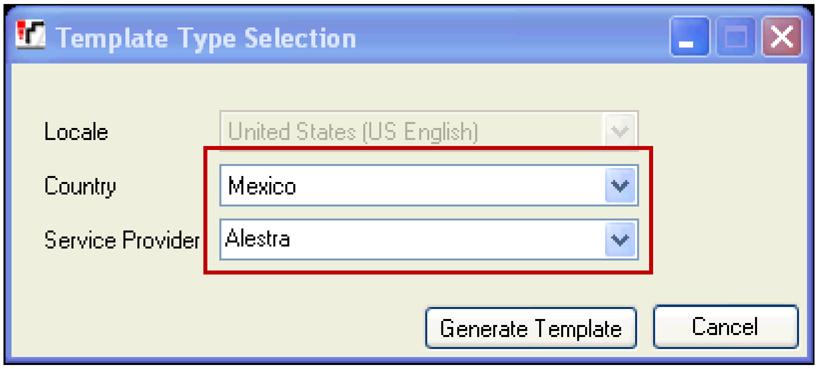
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click on the Sip Line (**17**), and select **Generate SIP Trunk Template** (not shown)

Enter a descriptive name; **Alestra** was used in the sample template. Note that for ITSP Domain Name **Not Used** was used (Alestra uses IP address instead of Domain name), an entry is required here or the template will not run. This entry (**Not Used**) should be removed after importing the configuration into a new Avaya IP Office installation.

To generate the template click on **Export**.



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

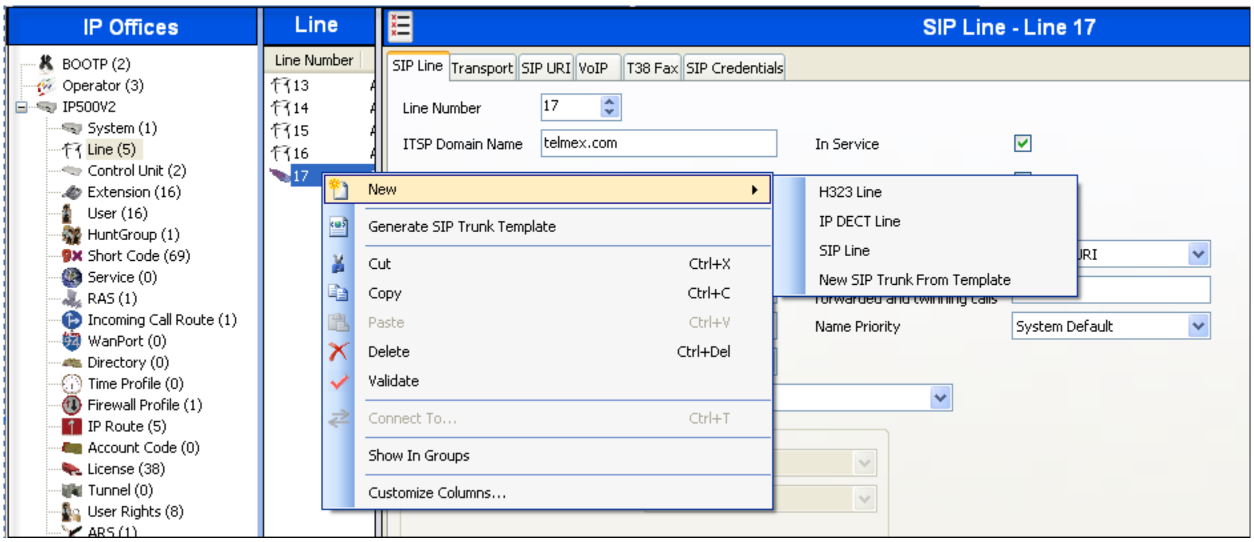


The following is the exported SIP Line Template file, **MX_Alestra_SIPTrunk.xml**:


```
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
<TemplateType>SIPTrunk</TemplateType>
<Version>20120904</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>Alestra</DescriptiveName>
<ITSPDomainName>Not Used</ITSPDomainName>
<SendCallerID>CallerIDRPID</SendCallerID>
<ReferSupport>true</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>
<UpdateSupport>UpdateNever</UpdateSupport>
<UserAgentServerHeader />
<CallerIDfromFromheader>false</CallerIDfromFromheader>
<PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
<ITSPProxy>192.168.10.9</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>AUTOSELECT</CompressionMode>
<UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
<AdvCodecPref>G.729(a) 8K CS-ACELP,G.711 ALAW 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>
= </Template>

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.