



## **Avaya Solution & Interoperability Test Lab**

---

# **Application Notes for Configuring Avaya IP Office Release 7.0(5) with Alestra Enlace IP SIP Trunk Service – Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider Alestra in Mexico and Avaya IP Office 7.0(5). The official name of Alestra's SIP Trunk offering is **"Enlace IP"**.

Alestra's Enlace IP SIP Trunk Service provides PSTN access via SIP trunks 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.

Service provider 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.

# 1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) Trunking between the 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 7.0(5) with a Digital Expansion Module - Release 9.0(5), Avaya Voicemail Pro, Avaya IP Office Softphone (SIP), Avaya IP Office Phone Manager PC Softphone (H.323), Avaya IP Telephones (H.323), Avaya Digital Telephones, Analog Telephones and Fax machines.

Customers using Avaya IP Office with Alestra's Enlace IP SIP Trunk service are able to place and receive PSTN calls via the SIP protocol. The converged network solution is an alternative to traditional analog trunks and/or PSTN trunks such as ISDN-PRI. This approach generally results in lower cost for the enterprise.

## 2. General Test Approach and Test Results

The general test approach was to simulate an enterprise site in the Solution & Interoperability Test Lab by connecting an Avaya IP Office Release 7.0(5) to Alestra's Enlace IP SIP Trunk service via the public internet, as depicted in **Figure 1**.

### 2.1 Interoperability Compliance Testing

To verify SIP trunk interoperability, the following areas were tested for compliance:

- Static IP Authentication.
- SIP OPTIONS messages.
- Incoming calls to IP Office from the PSTN were routed to the DID numbers assigned by Alestra. Incoming PSTN calls were terminated to the following end points: Avaya IP Telephones (H.323), Avaya Digital Telephones, IP Office Softphone (SIP), IP Office Phone Manager PC Softphone (H.323), Analog Telephones and Fax machines.
- Outgoing calls from IP Office were routed via Alestra's GSX9000 Sonus 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 end points.
- 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.
- 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.
- 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 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).
- IP Office Mobility Twinning.
- Avaya IP Office Softphone (SIP).
- Avaya IP Office Phone Manager PC Softphone (H.323)
- Simultaneous active calls.
- Long duration calls (> one hour).
- Proper response/error treatment to all trunks busy.
- Proper response/error treatment when disabling SIP connection.

## 2.2 Test Results

Interoperability testing of Alestra's Enlace IP SIP trunk service with Avaya IP Office 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 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 settings are not matched.

### Example Configuration:

Using DID number **8128813093** from local PSTN as the twinned mobile extension.  
Using extension number 3041 in IP Office as the Host station.

### INVITE received by IP Office:

The "FROM" field in the invite sent by the Twinned mobile extension to IP Office when dialing the number of the Host station is in the following format:

**From:<sip:8128813093@xxx.xxx.xxx.9**

### IP Office settings:

User/3041/Mobility: Twinned Mobile Number = **8128813093**  
Incoming Call Route/Incoming CLI = **8128813093**

With the above settings calling party receives dial tone from 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 = **28813093**

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

2. Outgoing Call Transfers from IP Office to the PSTN will not complete if Short Code 9N; or ARS 9N are used. The problem is in the REFER message from IP Office to the Service Provider's network (Sonus). If Short Code 9N; or ARS 9N are used, IP Office will insert the leading digit "9" to the number in the "**Refer-to**" header (in Refer message), for example: **928813093@xxx.xxx.xxx.9** thus causing Sonus to generate a "**503 Service Unavailable**" response in the Notify message. With Alestra's Enlace IP SIP Trunk service, Sonus is not programmed to route calls with the leading digit "9".

The work around is to add Short Code 2N; in IP Office to send the digits in the Refer-to header in the following format: **28813093@xxx.xxx.xxx.9** (without the leading "9").

**Note:** Short Codes and ARS are two important features of IP Office, Short Code 9N; will allow users to dial **any** number after the "9" and ARS provides second dial tone, alternate routing, etc.

3. Incoming Attended Call transfers from the local Mexican PSTN to IP Office and back to the local Mexican PSTN will not complete. The problem is in the REFER message from IP Office to the Service Provider's network (Sonus). IP Office inserts "**81**" to the number in the "**Refer-to**" header (in Refer message), for example: **8128813093@xxx.xxx.xxx.9** thus causing Sonus to generate a "**503 Service Unavailable**" response in the Notify message. With Alestra's Enlace IP SIP Trunk service, Sonus was not programmed to route calls with leading digits "**81**".

The work around is to add routing based on leading digits "**81**" to Alestra's network (Sonus).

Note: **81** is the **area code** for Monterrey, Mexico, the **country code** for Mexico (not shown here) is **52**.

4. Outgoing Fax calls fail to connect. The problem is seen with outgoing Fax calls (IPO → PSTN) with T.38 Interworking with G.711-Alaw. Traces captured indicate successful conversion from **G.711-Alaw** to **T.38** but the "modem connection" fails to be established and the attempt times out. At the user level, both fax machines show "connecting", the tone is heard and the attempt is made to scan the 1<sup>st</sup> page but failure happens eventually due to connection time-out.

The work around is to use **G.729(a)** interworking with **T.38**

5. Testing was executed using **G.729(a)** and **G.711-Alaw** codec's as requested by Alestra (common codec's used in Mexico).
6. Call Display on transferred calls to PSTN – Caller ID display is not properly updated on PSTN phones involved with call transfers from IP Office to the PSTN. On Call Transfers from IP Office to the PSTN, after the call transfer is completed, the PSTN phone does not display the actual connected party but instead shows the ID of the host extension that initiated the call transfer.
7. Calls originating from PSTN telephones in the U.S. to Mexico DIDs assigned to 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 IP Office. Alestra also provided access to a WEB based GUI allowing feature changes to this local PSTN number.

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

8. 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.).

## 2.3 Support

### 2.3.1. Avaya

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

### 2.3.2. Alestra

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>

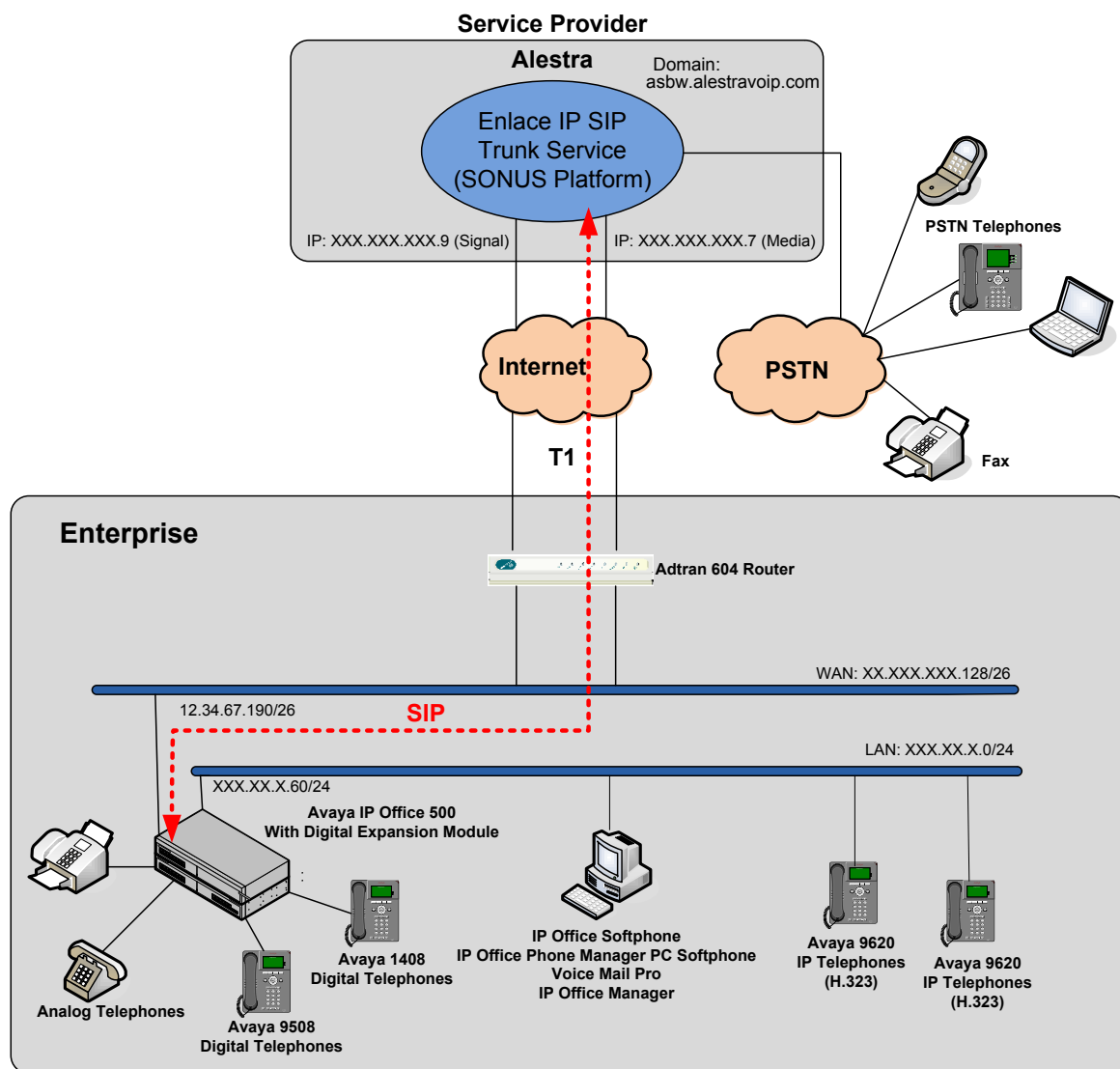
## 3. Reference Configuration

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

Located at the enterprise site is an Avaya IP Office 500v2 with a Digital expansion module. The LAN port of the Avaya IP Office is connected to the enterprise LAN while the WAN port is

connected to the public internet via a T1 to the local service provider. The T1 coming in to the simulated enterprise site from the local service provider is connected to an Adtran 604 router.

Actual customer configuration may also include additional network components between the service provider's network and the Avaya IP Office, such as a session border controller at the Service provider and firewalls at the enterprise site.



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

The following DIDs (for Monterrey, Mexico), US PSTN numbers, and end point types were used for testing:

<b>DID Number</b>	<b>Client Type</b>	<b>Notes</b>
(81) 1234-3040	Avaya 9620 IP Telephone (H323)	Registered with IP Office.
(81) 1234-3041	Avaya 9620 IP Telephone (H323)	Registered with IP Office.
(81) 1234-3042	Avaya 9620 IP Telephone (H323)	Registered with IP Office.
(81) 1234-3043	Avaya 9508 Digital Telephone	Registered with IP Office.
(81) 1234-3044	Avaya 9508 Digital Telephone	Registered with IP Office.
(81) 1234-3045	Not Used	---
(81) 1234-3046	Not Used	---
(81) 1234-3047	Avaya IP Office Softphone (SIP)	Registered with IP Office.
(81) 1234-3048	Avaya IP Office Phone Manager PC Softphone (H.323)	Registered with IP Office.
(81) 1234-3049	Not Used	---
(81) 1234-3093	SIP Softphone	Registered with the local PSTN in Monterrey, Mexico
<b>US PSTN Number</b>	<b>Client Type</b>	<b>Notes</b>
1-555-123-0788	Avaya 9641G IP Telephone	Registered with the local PSTN in the U.S.
1-555-123-0772	Analog Line/Phone	Connected to the Local PSTN in the U.S.

**Table 1 – DID / US PSTN Numbers and 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
<b>Avaya</b>	
Avaya IP Office 500v2	7.0 (5)
Avaya IP Office Digital Expansion Module	9.0 (5)
Avaya IP Office Manager	9.0 (5)
Avaya Voicemail Pro Client	7.0 (17)
Avaya 9620 Series IP Phones (H.323)	Avaya one-X Desk phone Edition 3.1
Avaya 1408 Series Digital Phone	--
Avaya 9508 Series Digital Phone	--
Lucent Analog Phone	--
Avaya IP Office Softphone (SIP)	3.1.2.17
Avaya IP Office Phone Manager PC Softphone (H.323)	4.2.39
Fax Machines	--
SIP based Softphone (For use at local PSTN in Monterrey, Mexico)	--
<b>Alestra</b>	
Sonus Network	6.4

**Table 2 – Hardware and Software Components Tested**

## 5. Configure IP Office

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [2]. From the IP Office Manager PC, select **Start → Programs → IP Office → Manager** to launch the Manager application. A screen that includes the following in the center may be displayed:

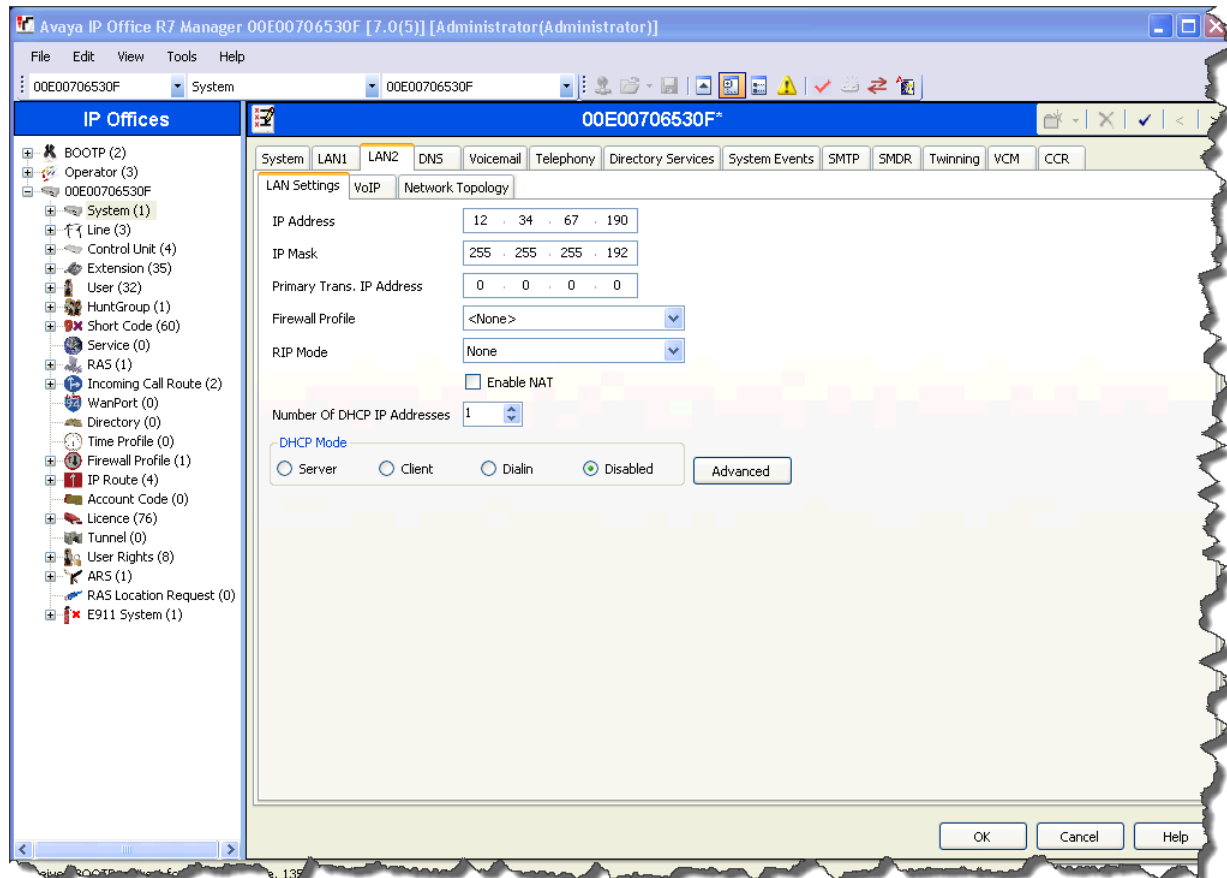


Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials.

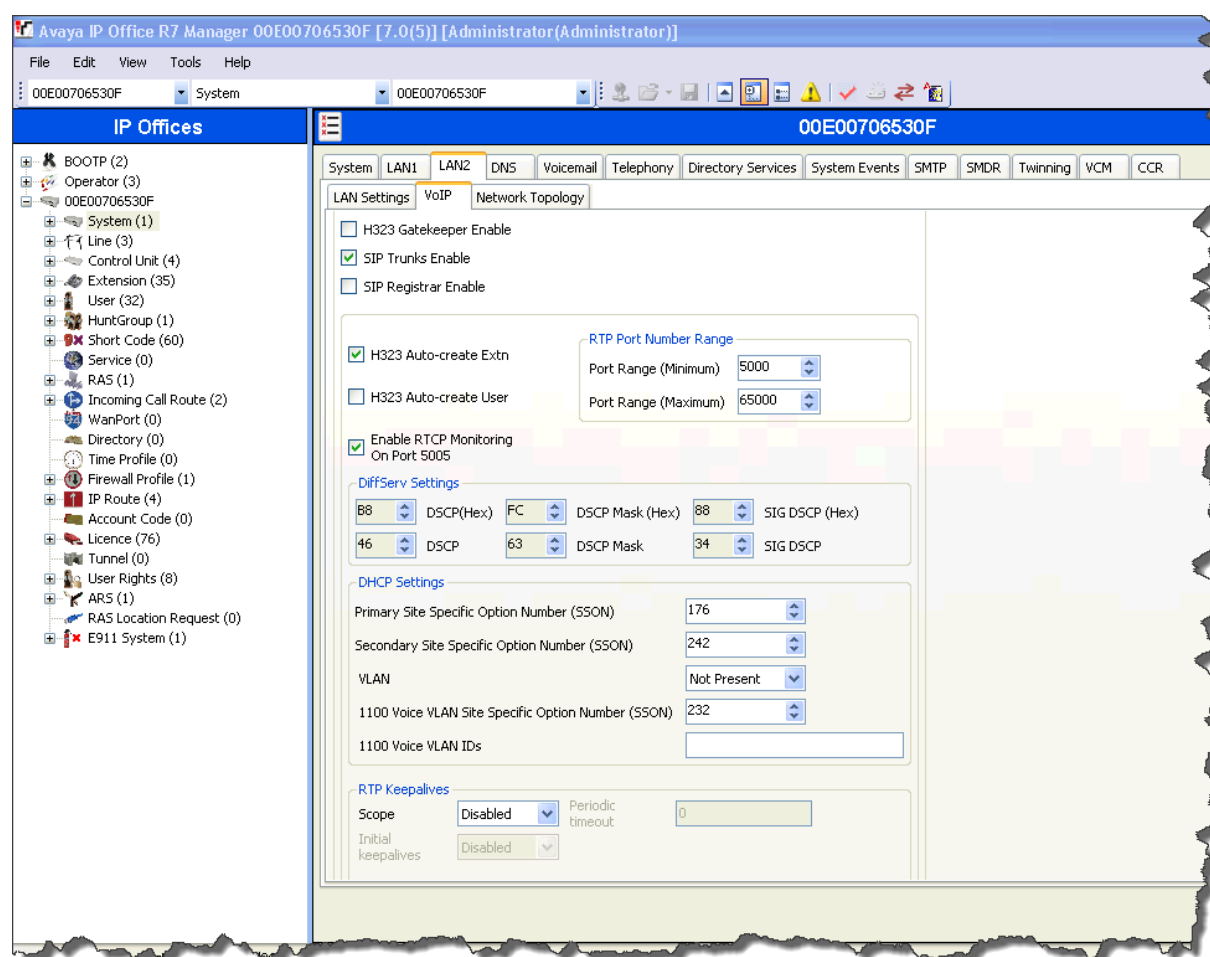
A management window will appear similar to the one shown in the next section. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side and the Details pane on 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 interface with the service provider (such as LAN1 interface to the enterprise site and IP Office Softphone support) is assumed to be already in place.

## 5.1 LAN2 Settings

In the sample configuration, the MAC address **00E00706530F** was used as the system name and the WAN port was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office. To access the LAN2 settings, first navigate to **00E00706530F → System (1)** in the Navigation Pane and then navigate to the **LAN2** Settings tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port, this is a public address. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.

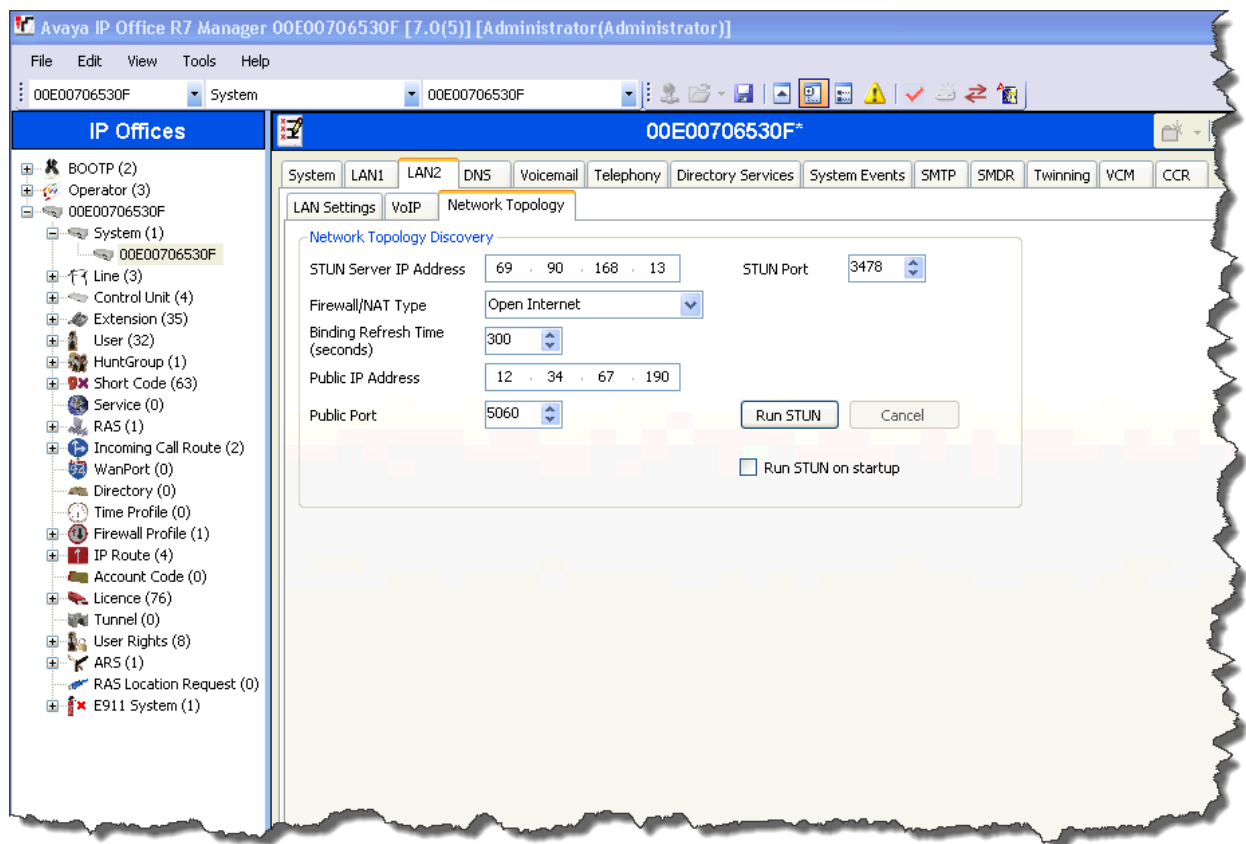


Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Alestra's Enlace IP SIP Trunk service. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on 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. All other parameters should be set according to customer requirements.



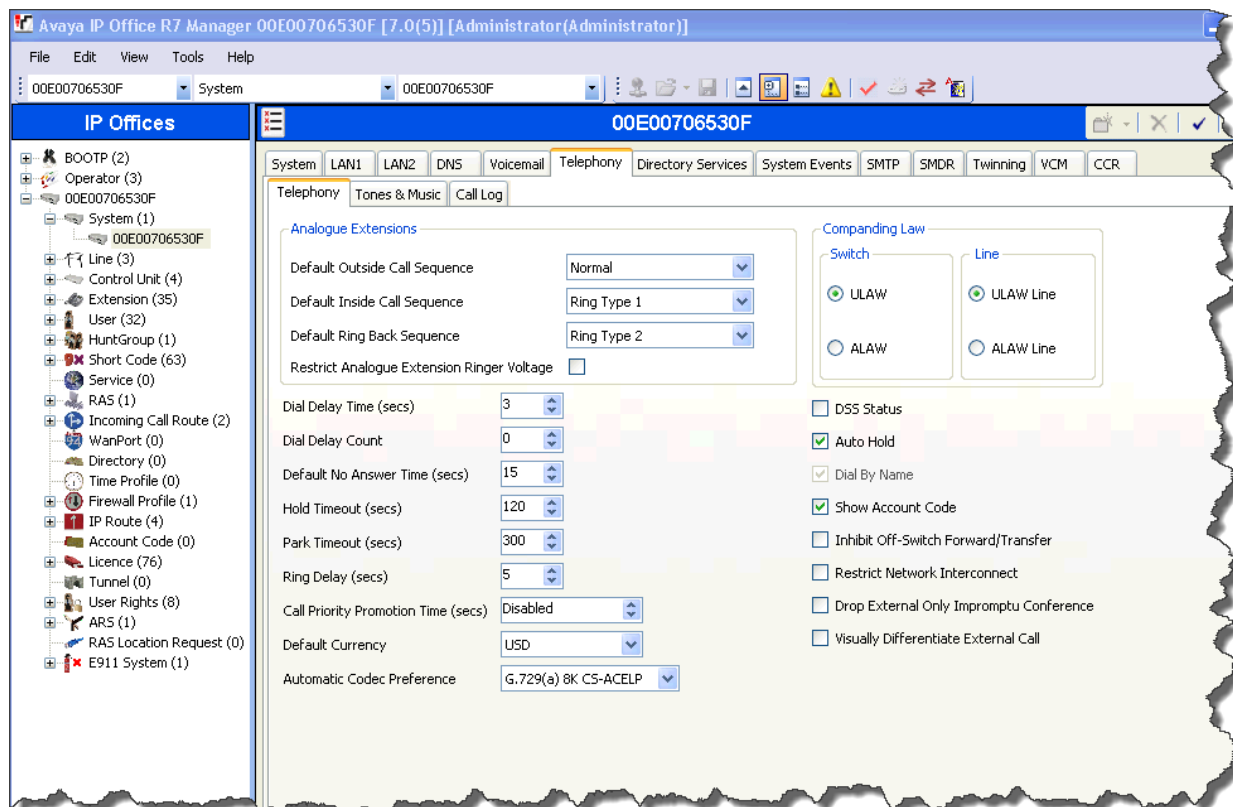
Select the **Network Topology** tab as shown in the following screen, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**. With this configuration, STUN will not be 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 OPTIONS messages to the service provider. Refer to section **5.10** for additional parameters that need to be set for SIP OPTIONS.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- All other parameters should be set according to customer requirements.



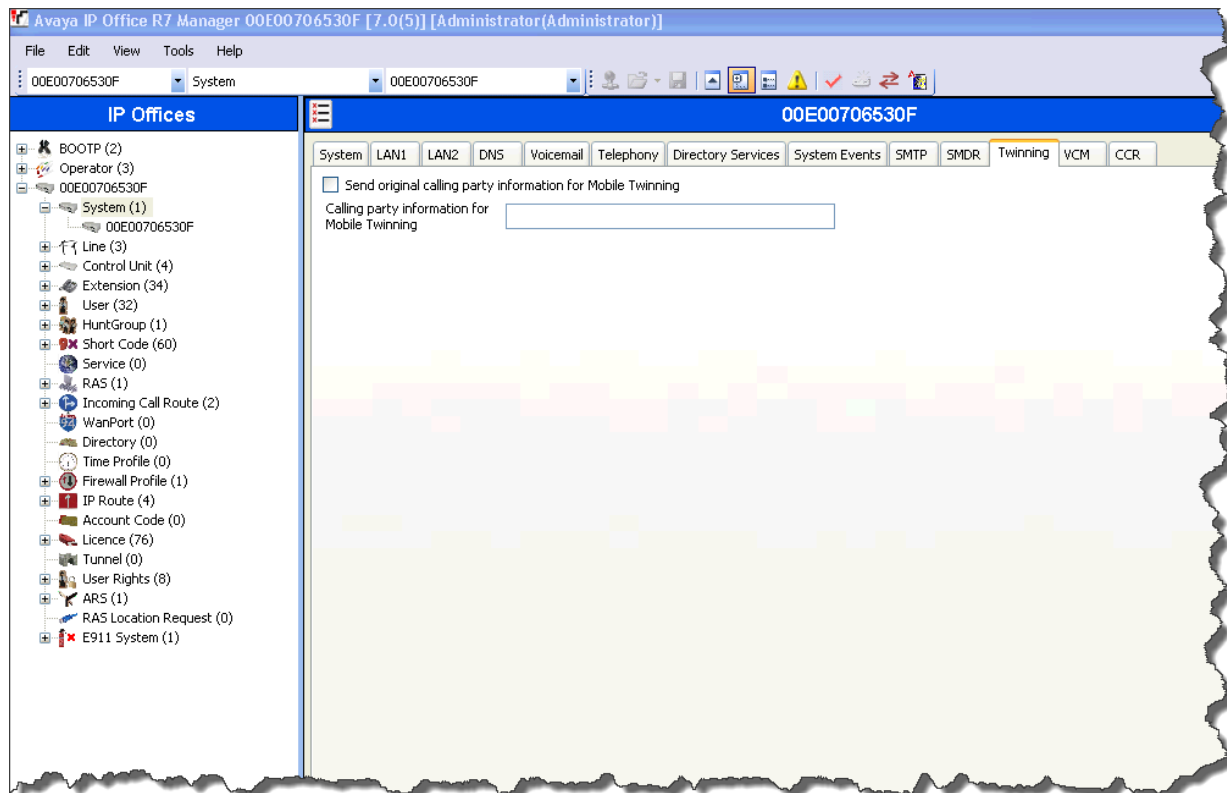
## 5.2 System Telephony Settings

Navigate to the **00E00706530F → System (1) → Telephony** Tab in the Details Pane. Set the **Automatic Codec Preference** for the default codec to be used for intra-enterprise traffic. This setting is used by all IP lines and extensions where the **Compression Mode** has been set to **Automatic Select**. The codec selected here will be the first codec the line or extension uses for codec negotiation. **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**. 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.



## 5.3 Twinning Settings

To view or change Twinning settings, select the **Twining** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. 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 (e.g., a user's mobile phone).



## 5.4 SIP Line

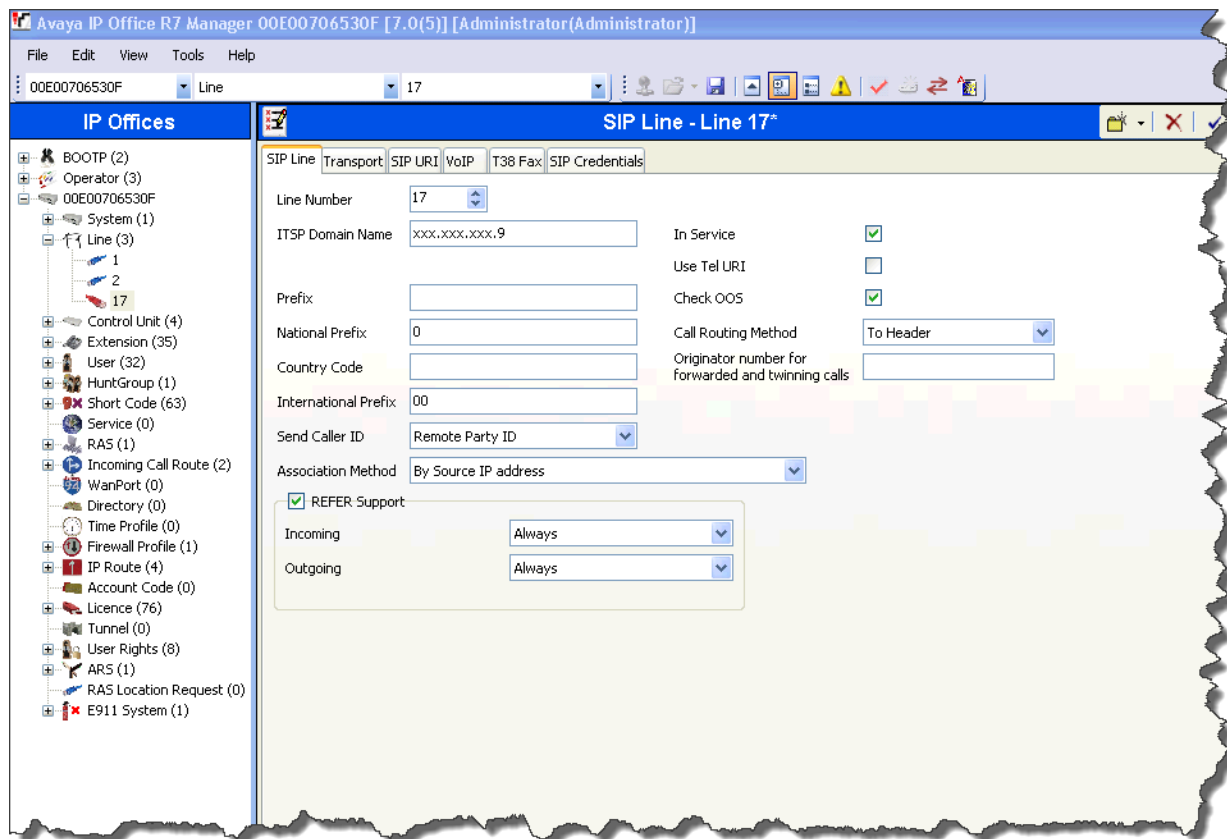
A SIP line is needed to establish the SIP connection between Avaya IP Office and Alestra's Enlace IP SIP Trunk service. To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click and select **New** → **SIP Line**, IP Office automatically assigns a Line Number.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

- Set **ITSP Domain Name** to the **SIP Proxy IP Address** of the service provider (Alestra), this IP Address is xxx.xxx.xxx.9 (Signaling IP address), the same IP address used in the Transport tab. IP Office uses this field on the host portion of SIP URI in SIP headers such as the "From" header.
- Set **Send Caller ID to Remote Party ID**. IP Office will include the Remote Party ID for calls that are directed via Mobile Twinning out the SIP Line to Alestra's network. Remote Party ID will contain the number associated with the party originating the call and will be populated with the true calling party identity, allowing the twinning destination (e.g., mobile phone) to see the true caller id.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Set **Call Routing Method** to **To Header**
- Default values may be used for all other parameters.

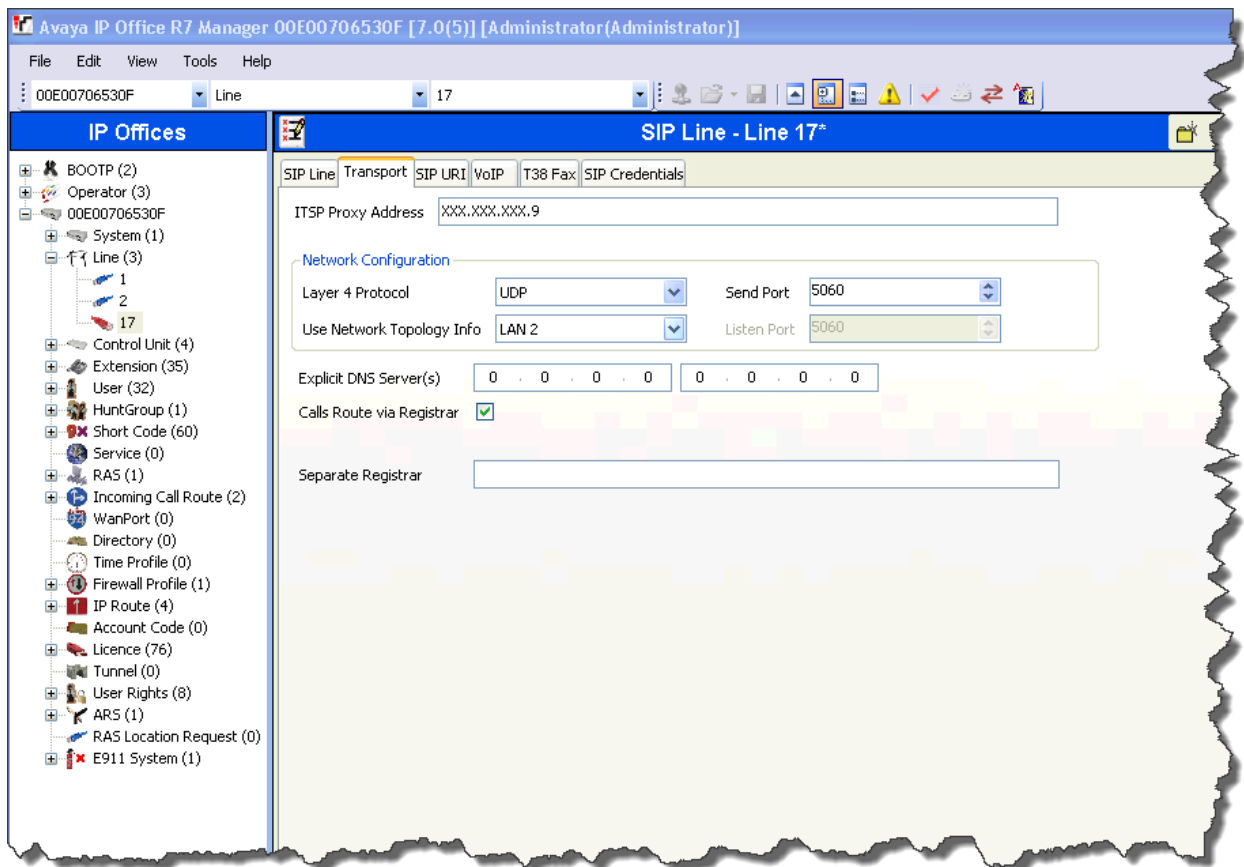
The **Association Method** parameter was introduced in IP Office Release 7.0, and the screen below shows the default value, which is sufficient in the sample configuration.

The area of the screen entitled **REFER Support** is used to enable/disable SIP REFER for call transfers. The default values of "Auto" for **Incoming** and **Outgoing** effectively disable use of SIP REFER. To enable SIP REFER, select "Always" from the drop-down menu for **Incoming** and **Outgoing**. In the compliance test this feature was enabled to test transfers of call between a PSTN phone and an enterprise phone to a second PSTN phone.

By default, the **In Service** and **Check OOS** boxes are checked. In the sample configuration, IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN2, as shown in **Section 5.1 LAN2 Settings** in the **Network Topology** tab. See **Section 5.10** for additional information related to configuring the periodicity of SIP OPTIONS.

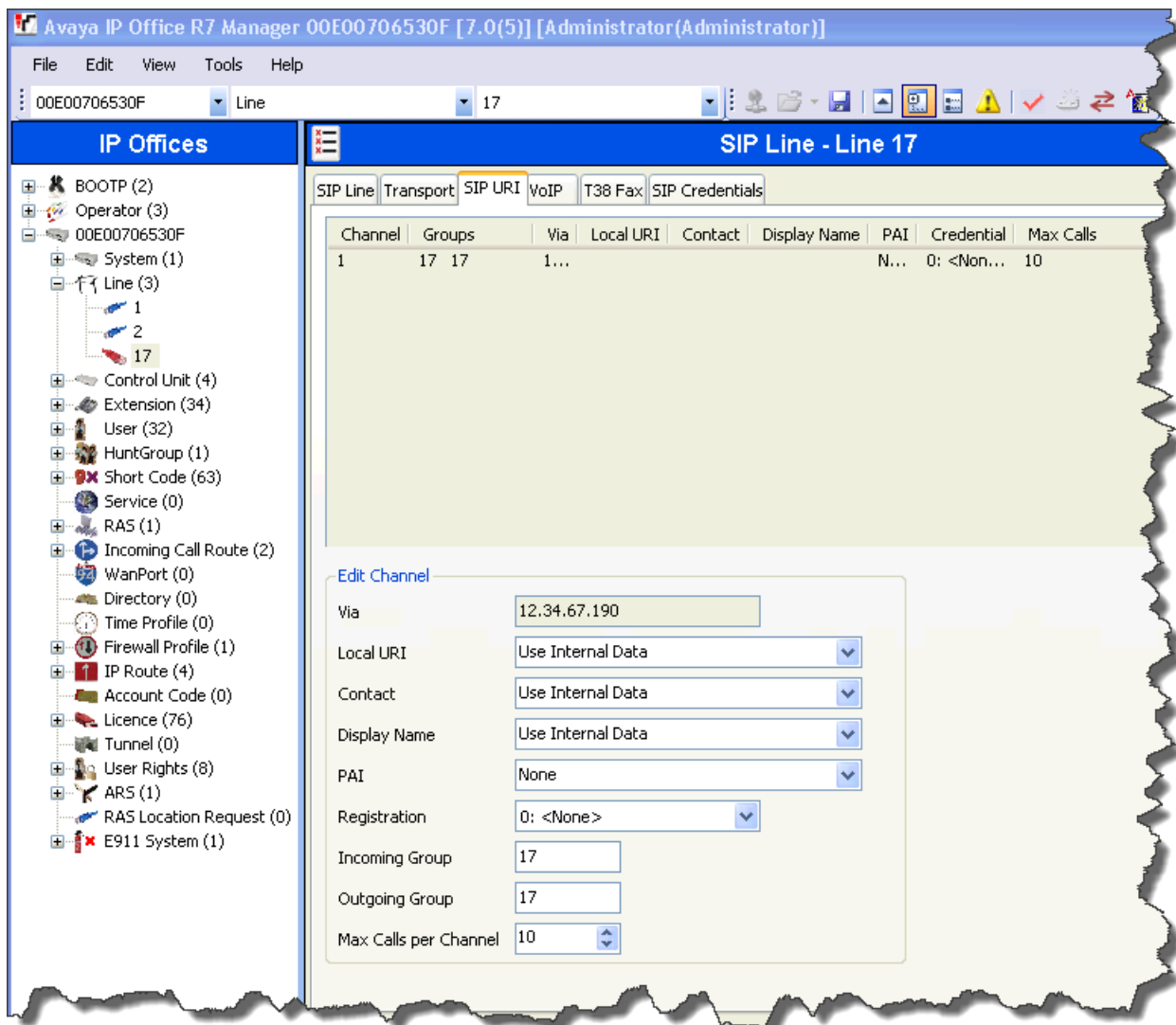


Select the **Transport** tab. The **ITSP Proxy Address** is set to the service provider SIP Proxy IP Address (provided by Alestra). As shown in **Figure 1**, this IP Address is xxx.xxx.xxx.9 (Signaling IP address). In the **Network Configuration** area, **UDP** is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by Alestra (5060). The **Use Network Topology Info** parameter is set to **LAN 2**. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab. Other parameters may retain default values as in the screen below.



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 IP Office Release 6.1, and the value “None” is shown selected from the drop-down menu. With PAI set to “None”, IP Office Release 6.1 and 7.0 will behave like IP Office Release 6.0 with respect to the SIP P-Asserted-Identity header (e.g., 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’s Enlace 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.

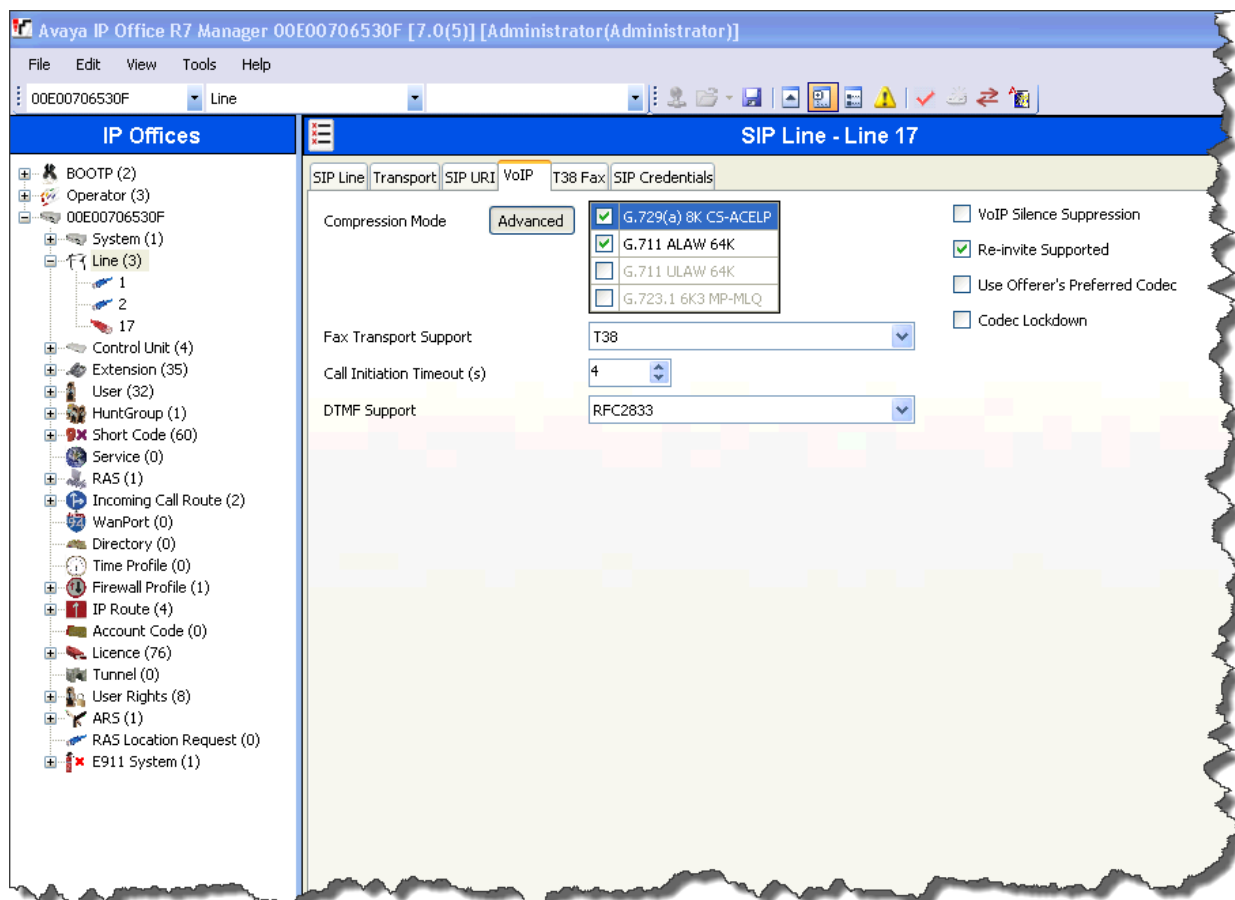


Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as follows:

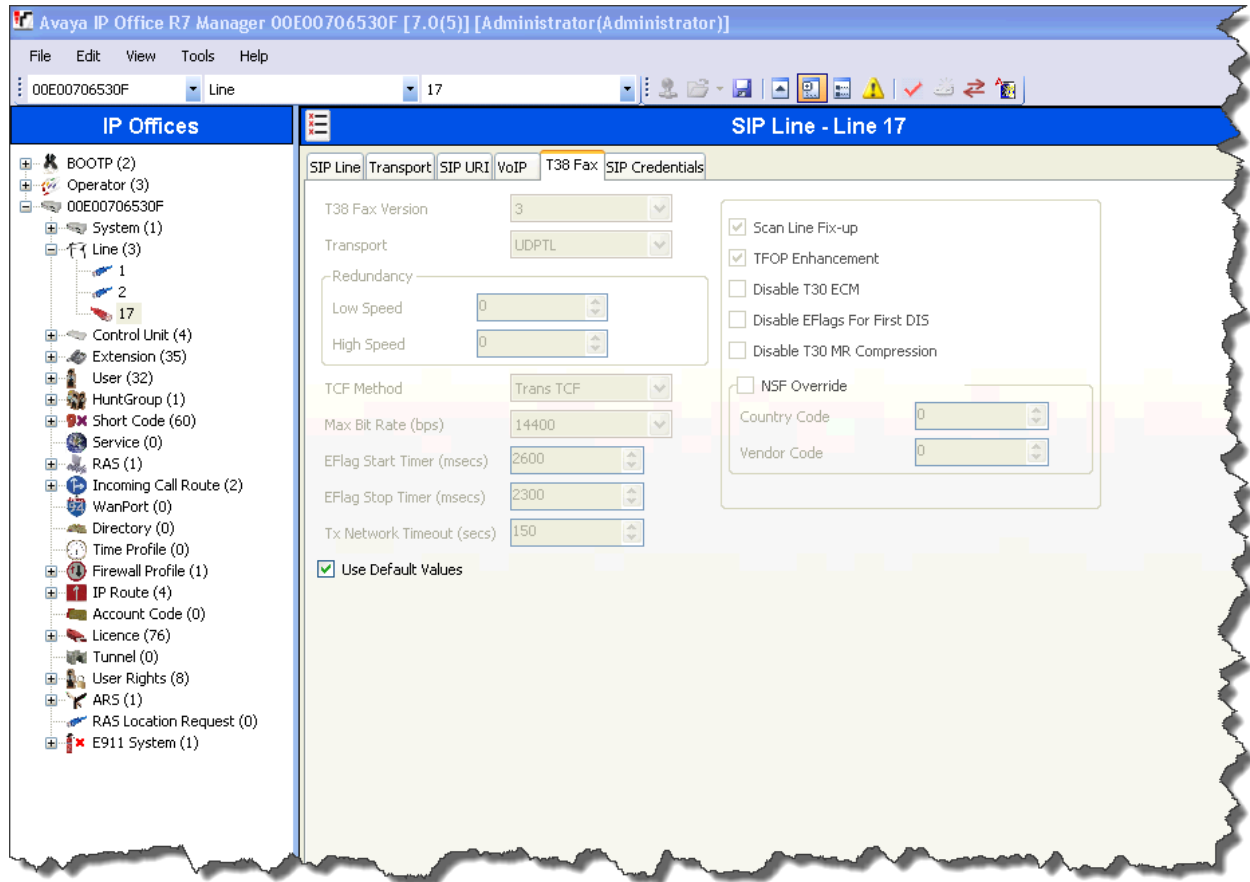
- The **Compression Mode** was configured using the **Advanced** button, allowing an explicit ordered list of codec's to be specified. Place check marks next to **G.729 (a) 8K CS-ACELP and G.711 ALAW 64K** which are the codec's supported by the Alestra's Enlace IP SIP Trunk service, in that specific order.
- **VoIP Silence Suppression:** Set to off (un-checked), default value. When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This feature is not used on IP lines using G711 between systems.
- The **Re-invite Supported** parameter should be checked 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. The **Re-invite Supported** parameter

must also be checked to allow the **Fax Transport Support** parameter to take on values other than “none”, or to check the **Codec Lockdown** parameter.

- **Use Offerer’s Preferred Codec:** Set to off (un-checked), default value. Normally for SIP calls the initiator of the calls sends a SIP SDP offer which includes the supported codec’s in order of preference. The SIP response includes the codec the callee wants to use for the call. This option can be used to override that behavior and use the codec preference offered by the caller.
- Select **T.38** for **Fax Transport Support**. Alestra supports T.38 faxing.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters.



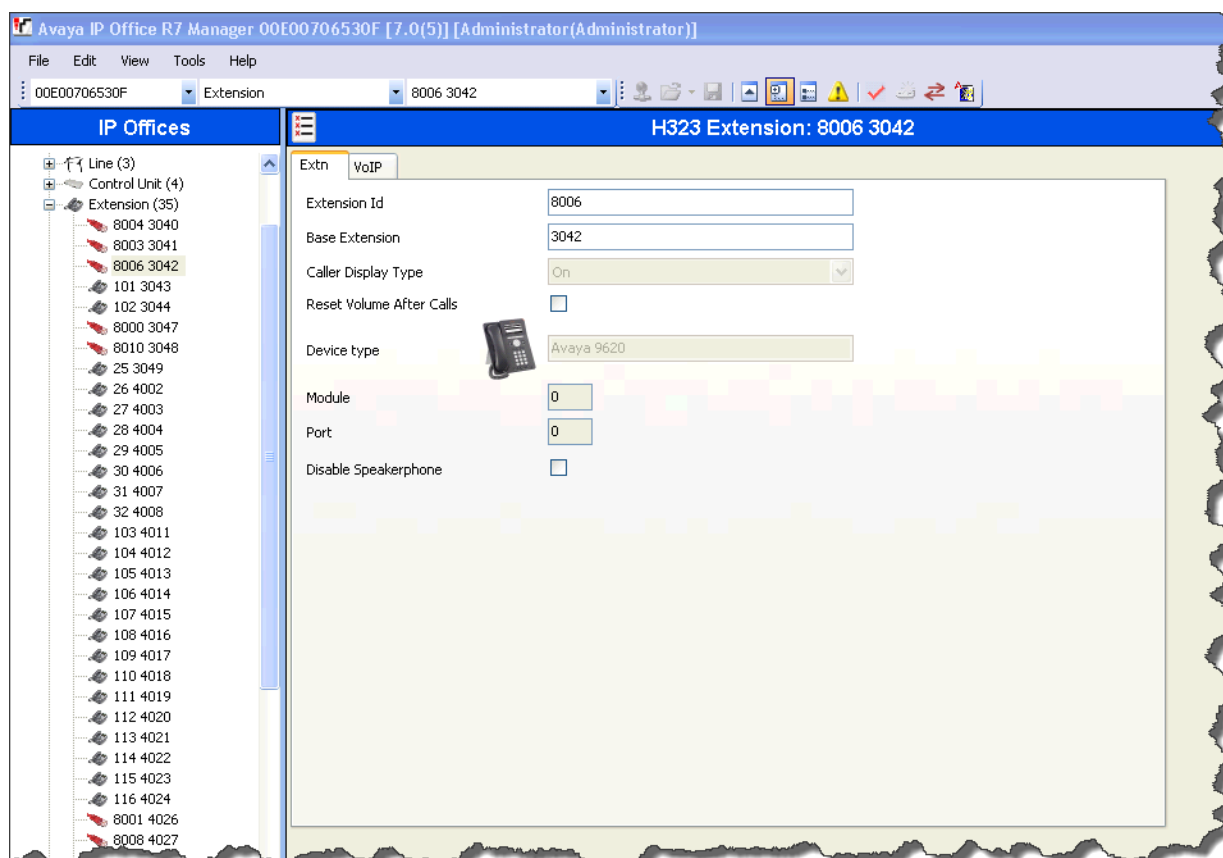
Select the **T38** tab. The setting on this tab are only accessible if **Re-invite Supported** and **Fax Transport Support** are selected on the **VoIP** tab. Use Default Value should be checked (default). If selected, all the fields are set to their default values and will be displayed as grayed out.



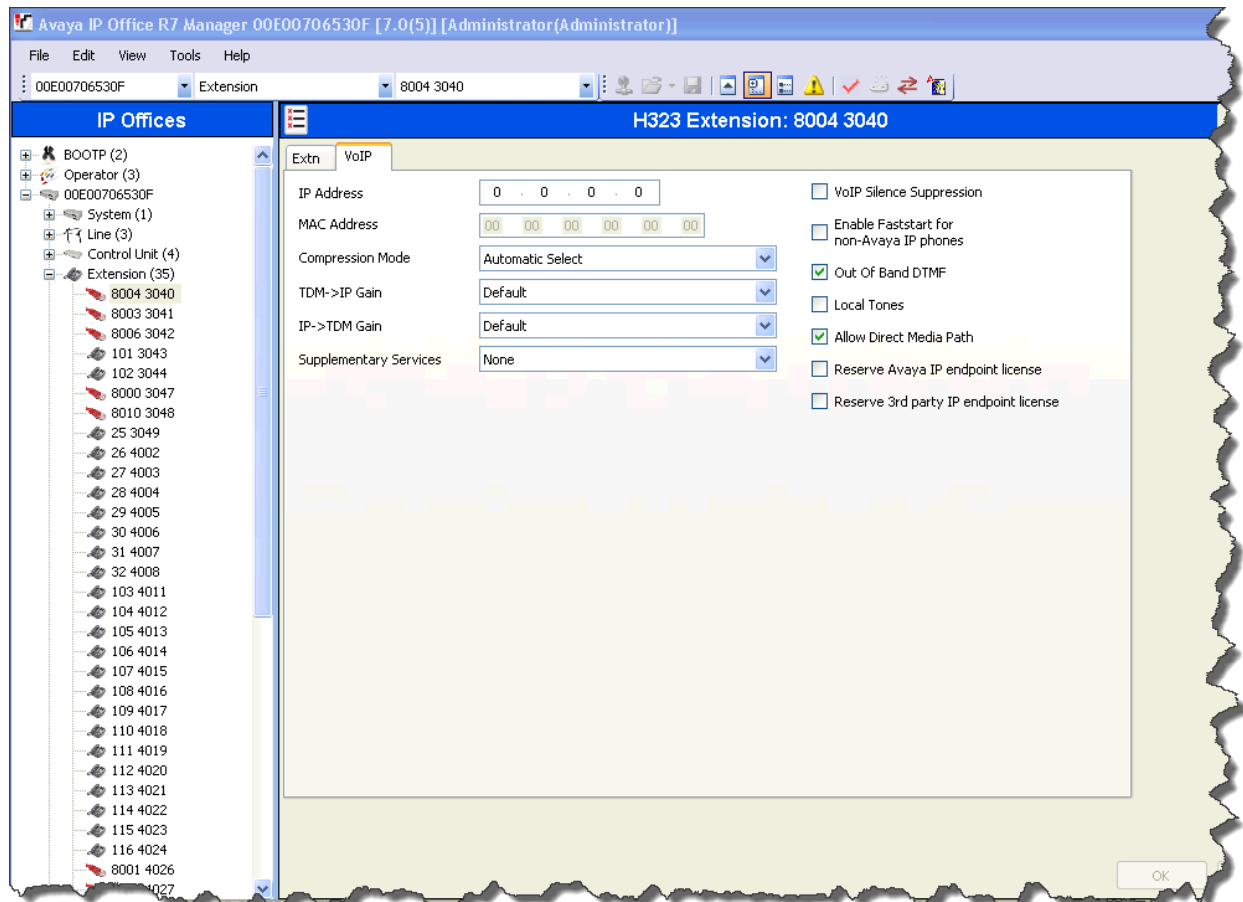
## 5.5 Extension

In this section, examples of 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.

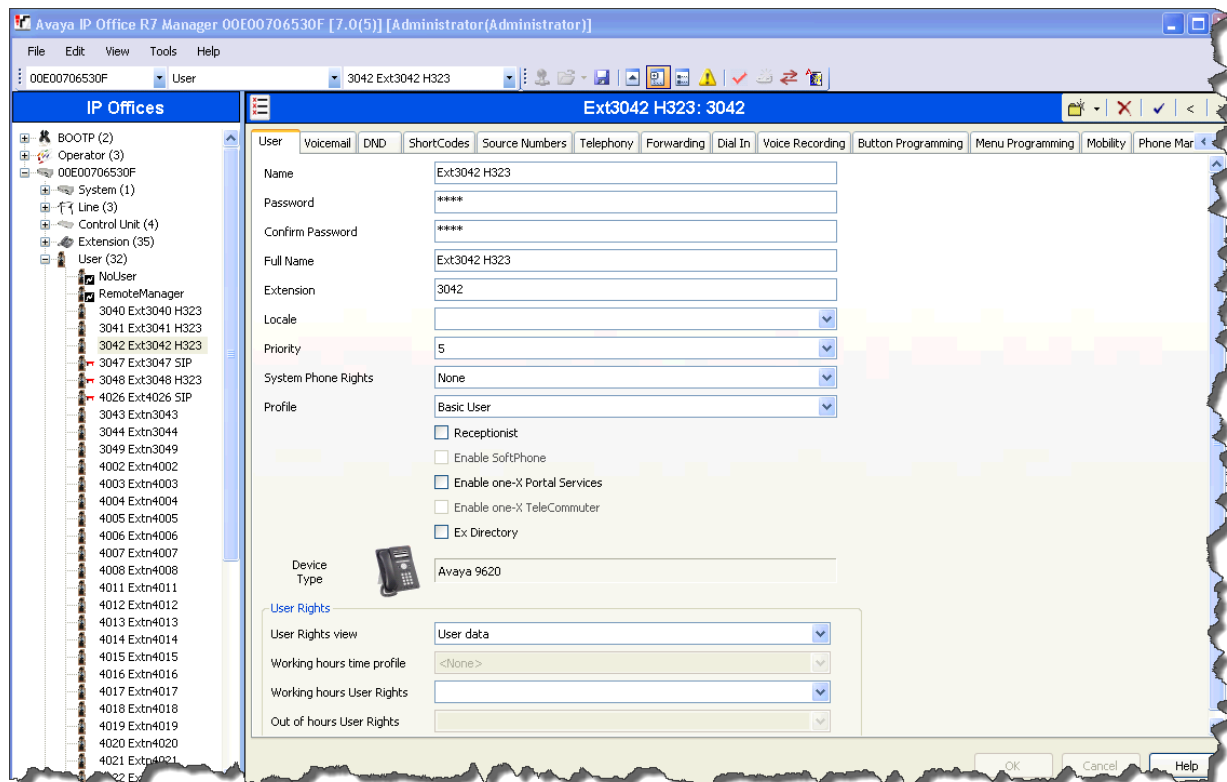


Select the VOIP tab. Use default values on VoIP tab. Following is an example for Extension 3042; this extension corresponds to an H.323 extension.



## 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, 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).

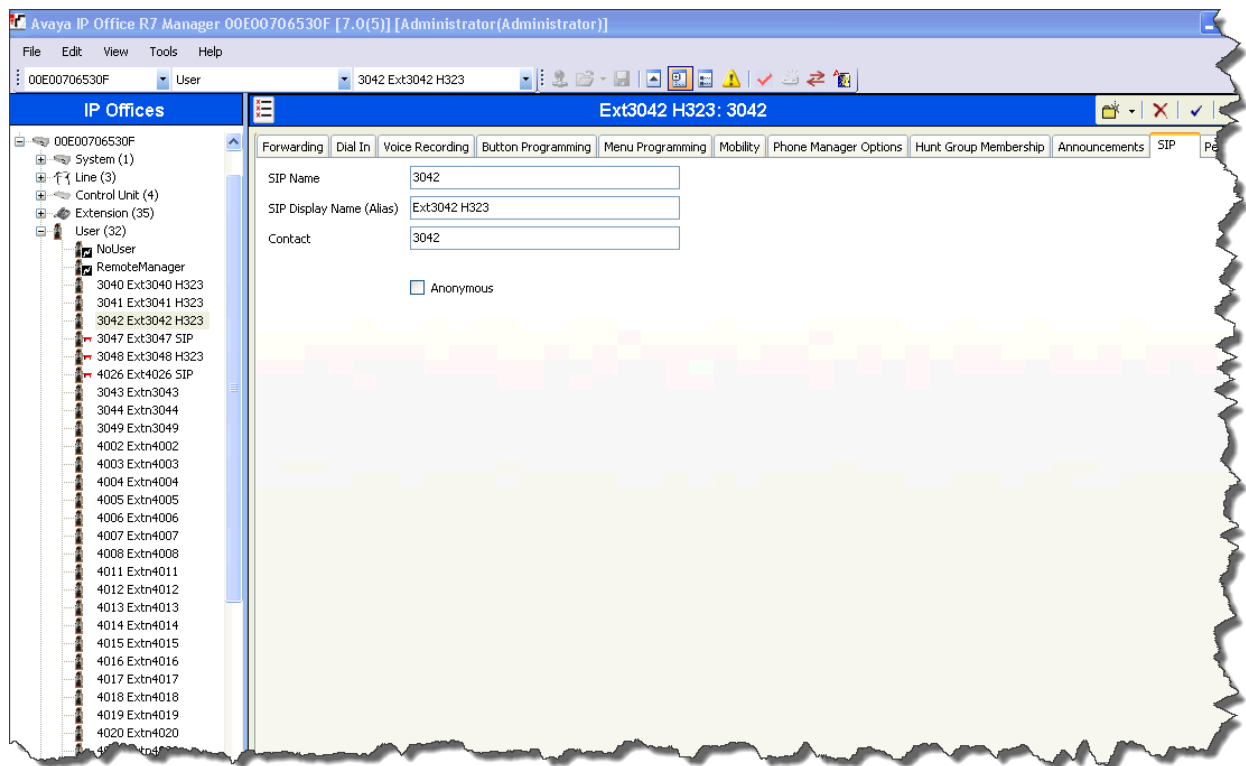


In the example below, the name of the user is “Ext3047 SIP”. This is an IP Office SIP Softphone user, set the Profile to **Teleworker User** and check **Enable Softphone**.

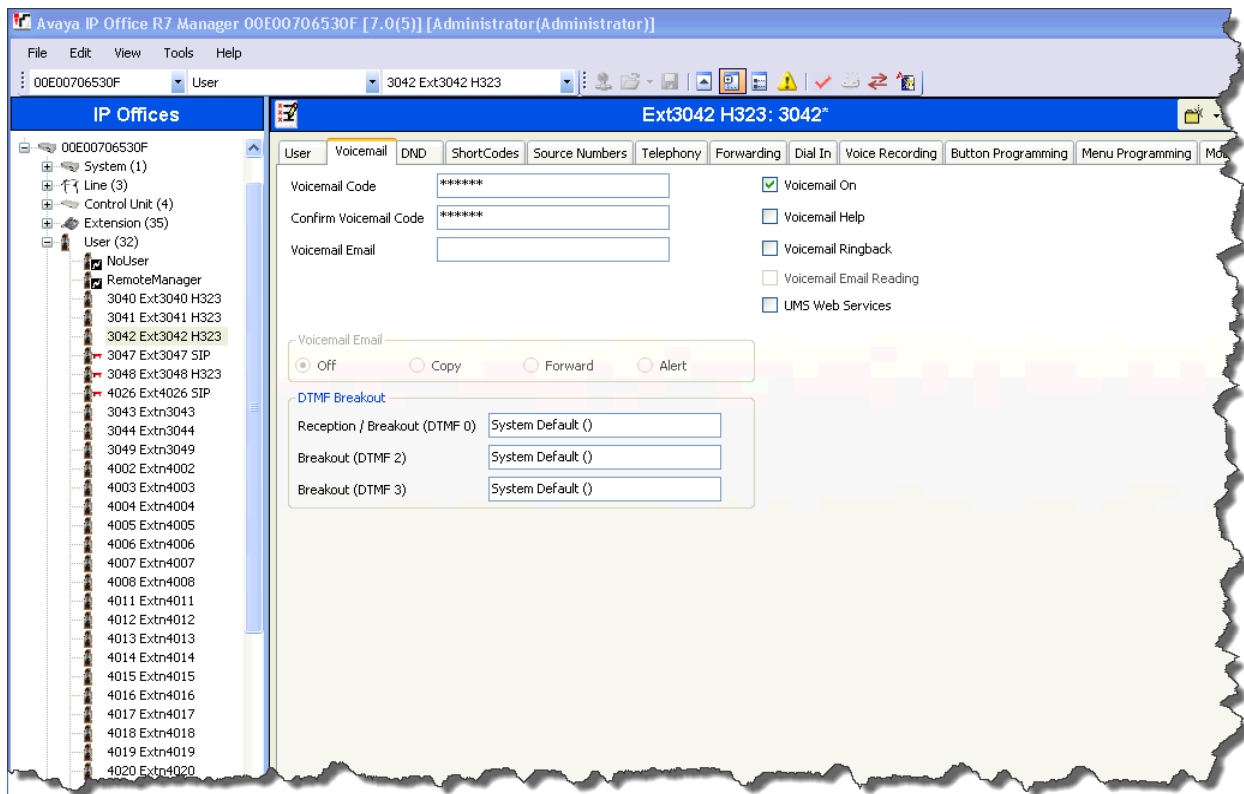
The screenshot shows the Avaya IP Office R7 Manager interface. The title bar reads "Avaya IP Office R7 Manager 00E00706530F [7.0(5)] [Administrator/Administrator]". The main window has a menu bar (File, Edit, View, Tools, Help) and a toolbar. Below the toolbar, there's a navigation pane on the left titled "IP Offices" showing a tree structure: Operator (3), System (1), Line (3), Control Unit (4), Extension (34), and User (32). The "User" folder is expanded, showing a list of users including "3047 Ext3047 SIP". The main pane displays the configuration for "Ext3047 SIP: 3047". The "User" tab is selected, showing fields for Name, Password, Confirm Password, Full Name, Extension, Locale, Priority, System Phone Rights, Profile, Device Type, and User Rights. The "Profile" is set to "Teleworker User" and "Enable SoftPhone" is checked. The "User Rights" section shows "User data" selected for "User Rights view".

Field	Value
Name	Ext3047 SIP
Password	****
Confirm Password	****
Full Name	Ext3047 SIP
Extension	3047
Locale	[Dropdown]
Priority	5
System Phone Rights	None
Profile	Teleworker User
Receptionist	<input type="checkbox"/>
Enable SoftPhone	<input checked="" type="checkbox"/>
Enable one-X Portal Services	<input checked="" type="checkbox"/>
Enable one-X TeleCommuter	<input checked="" type="checkbox"/>
Ex Directory	<input type="checkbox"/>
Device Type	Unknown
User Rights view	User data
Working hours time profile	<None>
Working hours User Rights	[Dropdown]

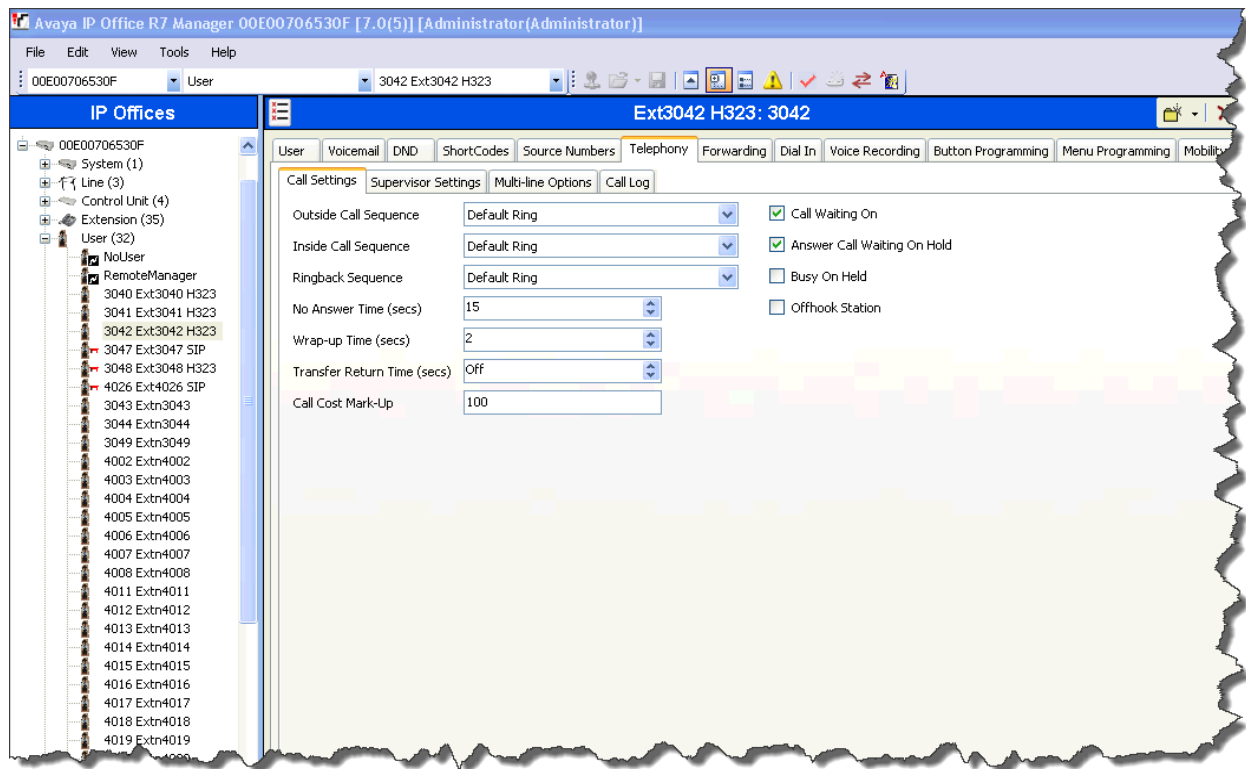
Select the **SIP** tab. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the “From” header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for 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.



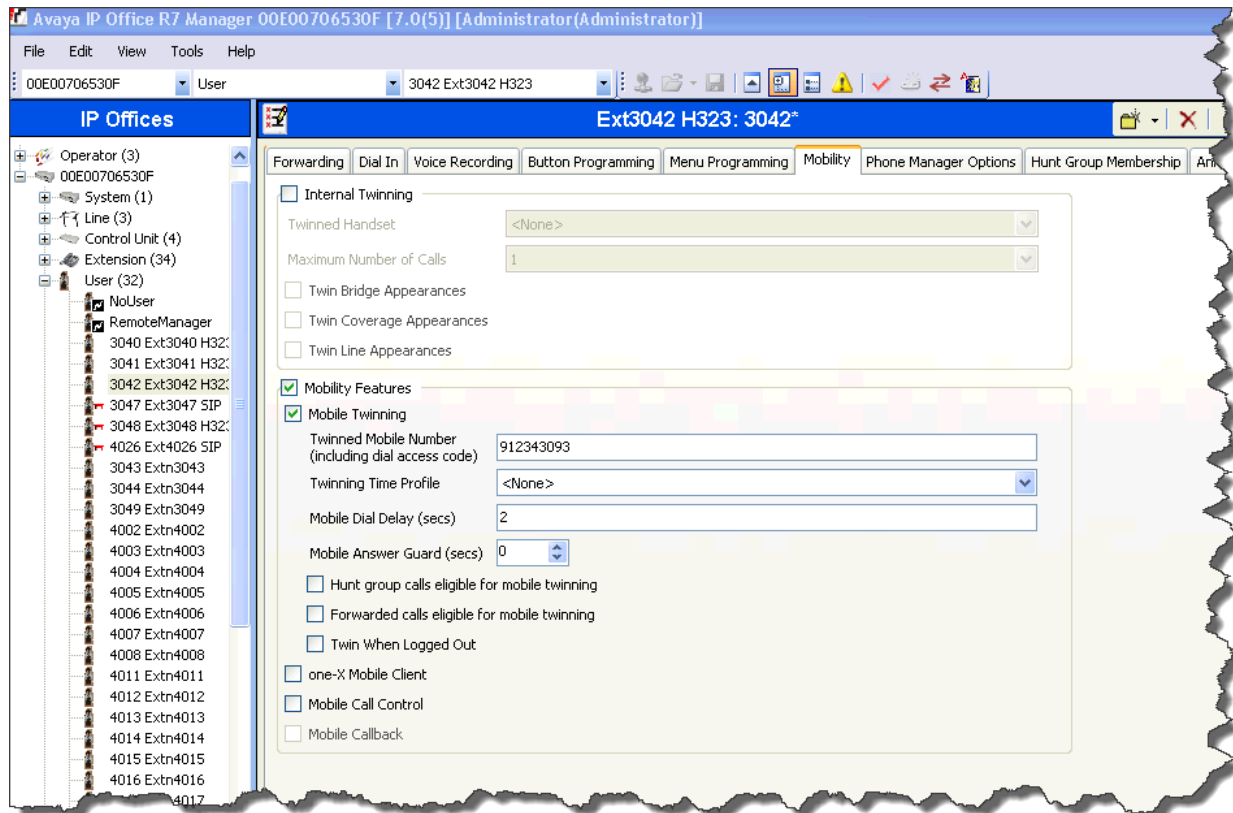
Select the Voice Mail tab. The following screen shows the **Voicemail** tab for the user with extension 3042. The **Voicemail On** box is checked. Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes,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



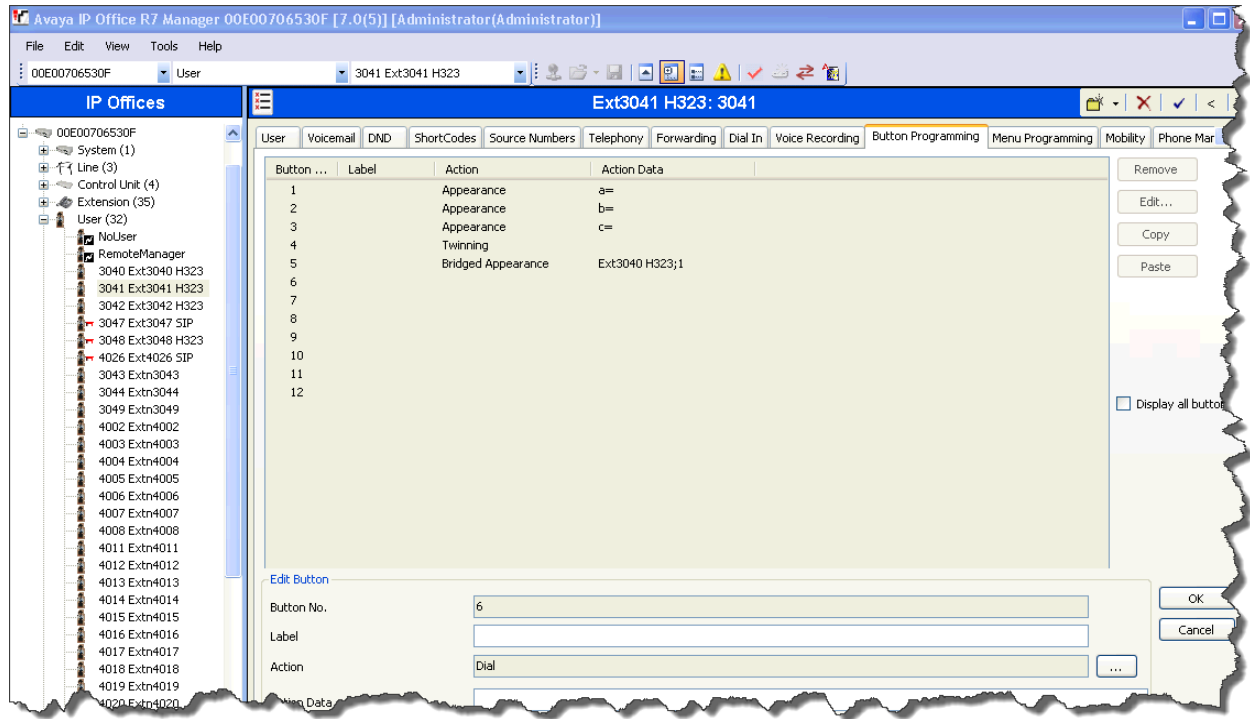
Select the Telephony tab, then **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an IP Office phone logged in as this extension to have multiple call appearances. Note: **Call Waiting On** is also necessary for call transfer.



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 **912343093**. Other options can be set according to customer requirements.



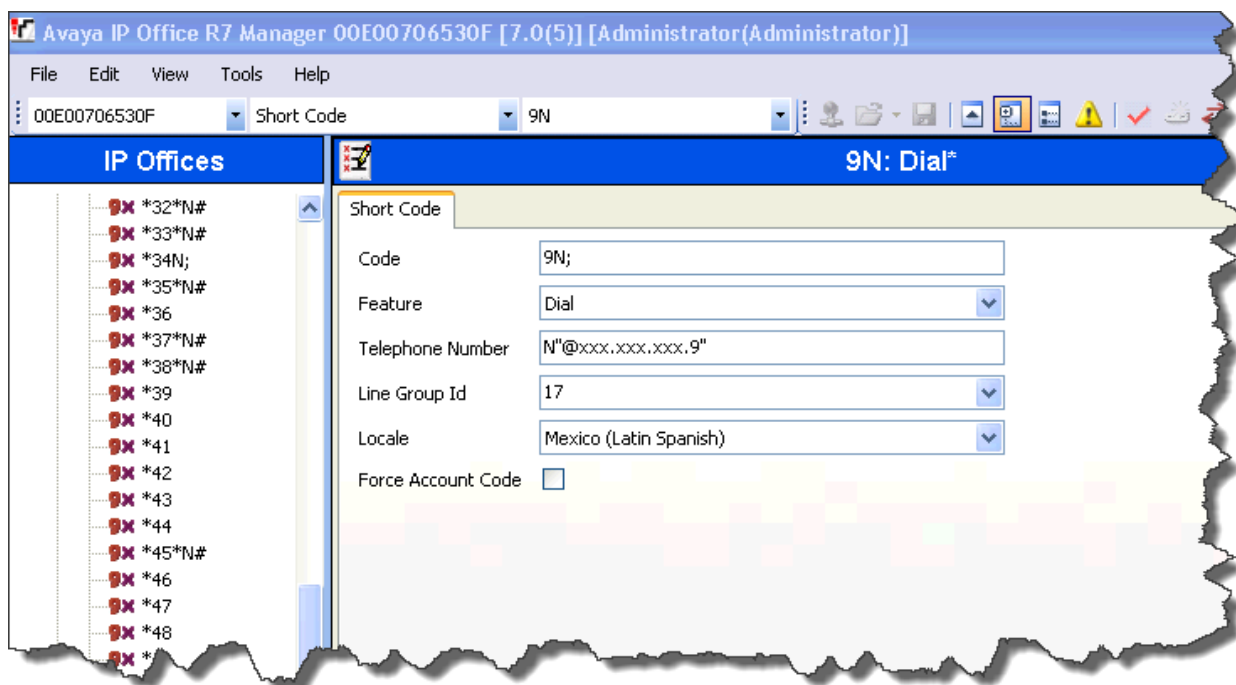
To program a key on the telephone to turn Mobil Twinning on and off, select the **Button Programming** tab on the user, then select the button to program to turn Mobil Twinning on and off, click on **Edit → Emulation → Twinning**. In the sample below button 4 was programmed to turn Mobil Twinning on and off on user 3041.



## 5.7 Short Code

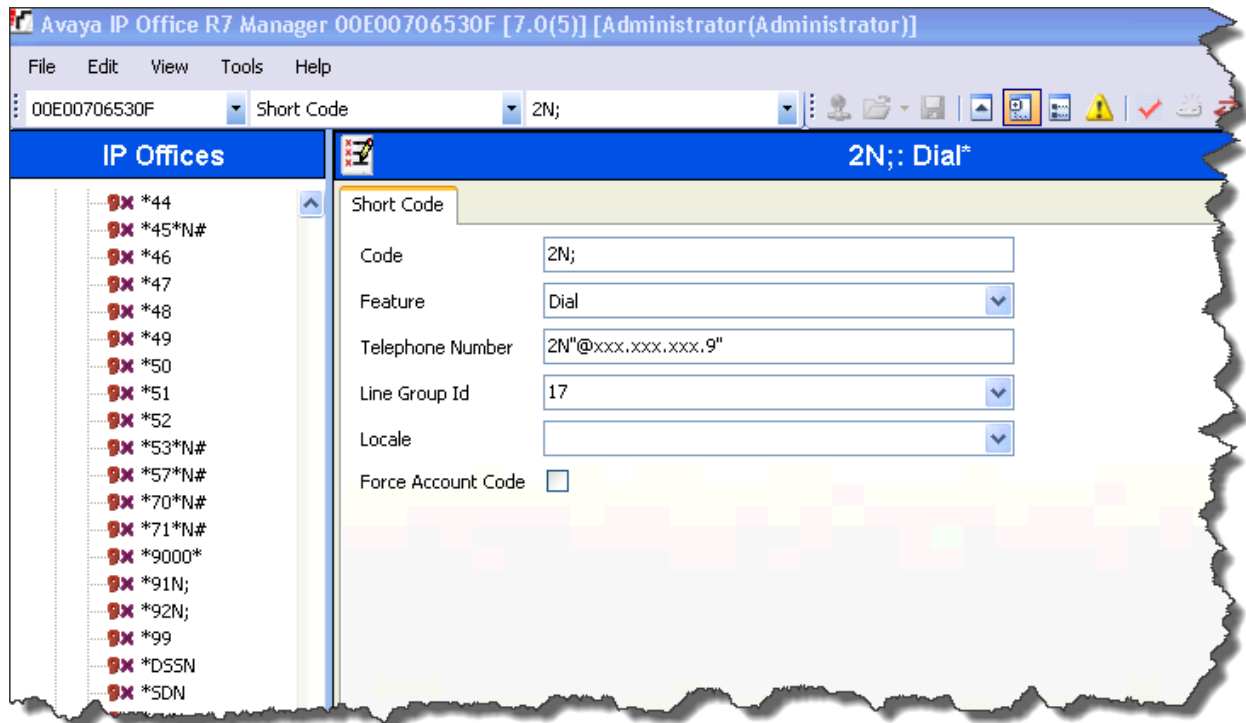
Define a Short Code to route outbound traffic to the SIP line. To create a Short Code, select **Short Code** in the left Navigation Pane, then right-click and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters for the new Short Code to be created as follows.

- 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"@xxx.xxx.xxx.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 the 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 the outbound call
- 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, or other features like Secondary Dial Tone. 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.

The following example shows Short Code 2N; This Short Code allows the user to dial any number starting with the digit “2” without having to dial “9” first.



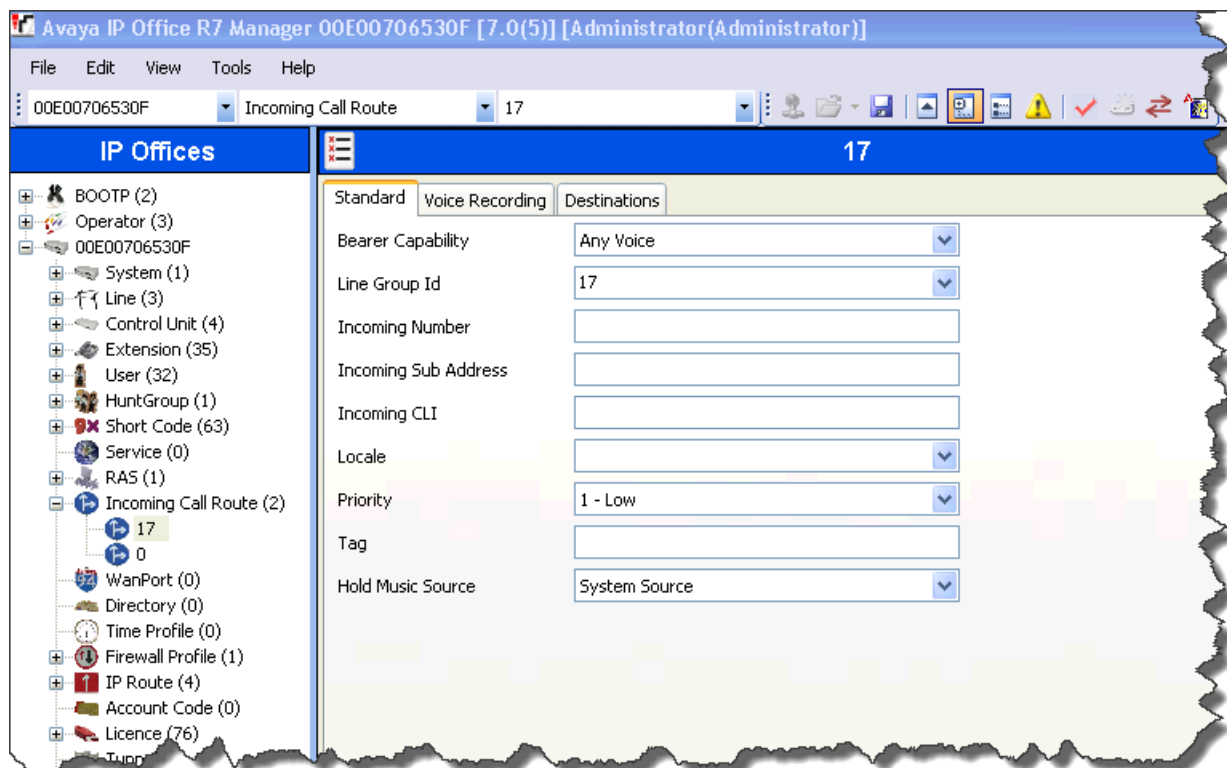
## 5.8 Incoming Call Route

Incoming Call Routes map inbound DID numbers on a specific line to internal extensions, hunt groups, Short Codes, voicemail, etc. in the IP Office. Incoming Call Routes should be defined for each DID number assigned by the service provider, matching incoming numbers on the SIP line with defined destinations in IP Office.

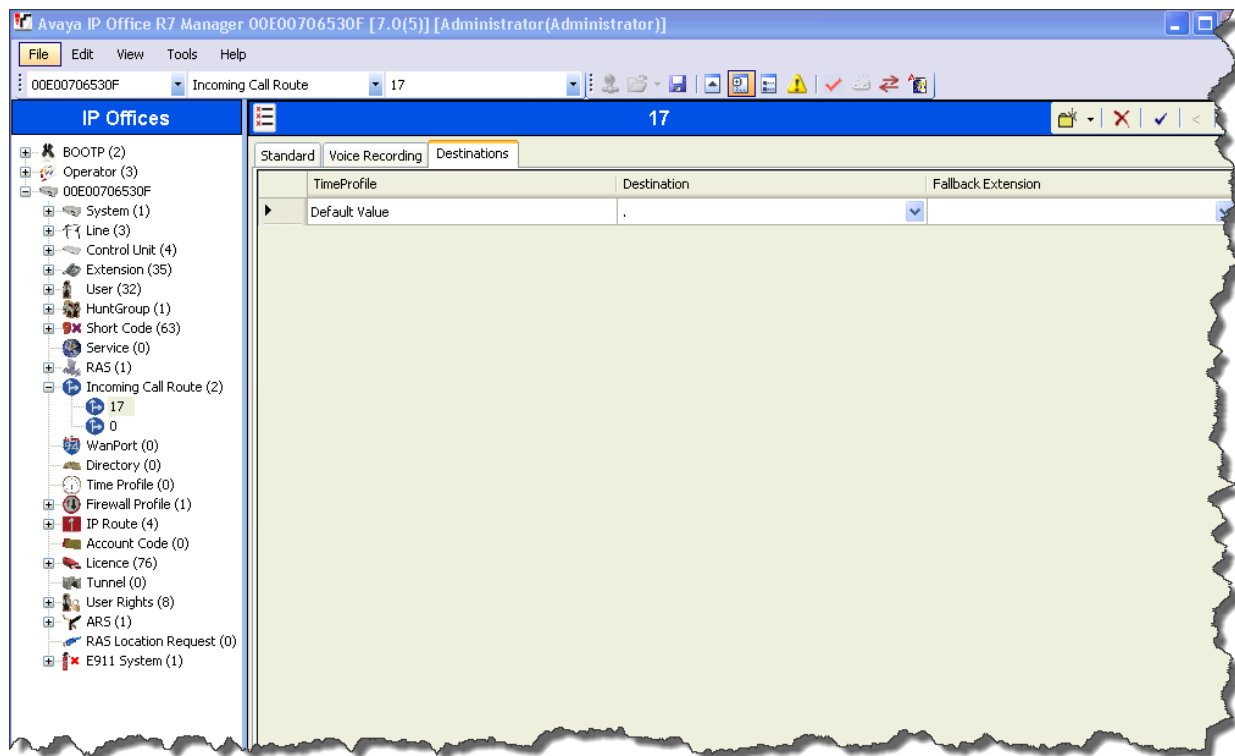
In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any extension in IP Office, and effectively performing all routing decision for the call based on the parameters already configured for **Call Routing Method**, set to **To Header (Section 5.4)**, and each user's **SIP Name** and **Contact** fields, already populated with the assigned DID numbers provided by Alestra (Section 5.6).

From the left Navigation Pane, right-click on **Incoming Call Route** and select **New**. On the Details Pane, under the **Standard** tab, set the parameters as show below:

- Set **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4**.
- Default values may be used for all other parameters.



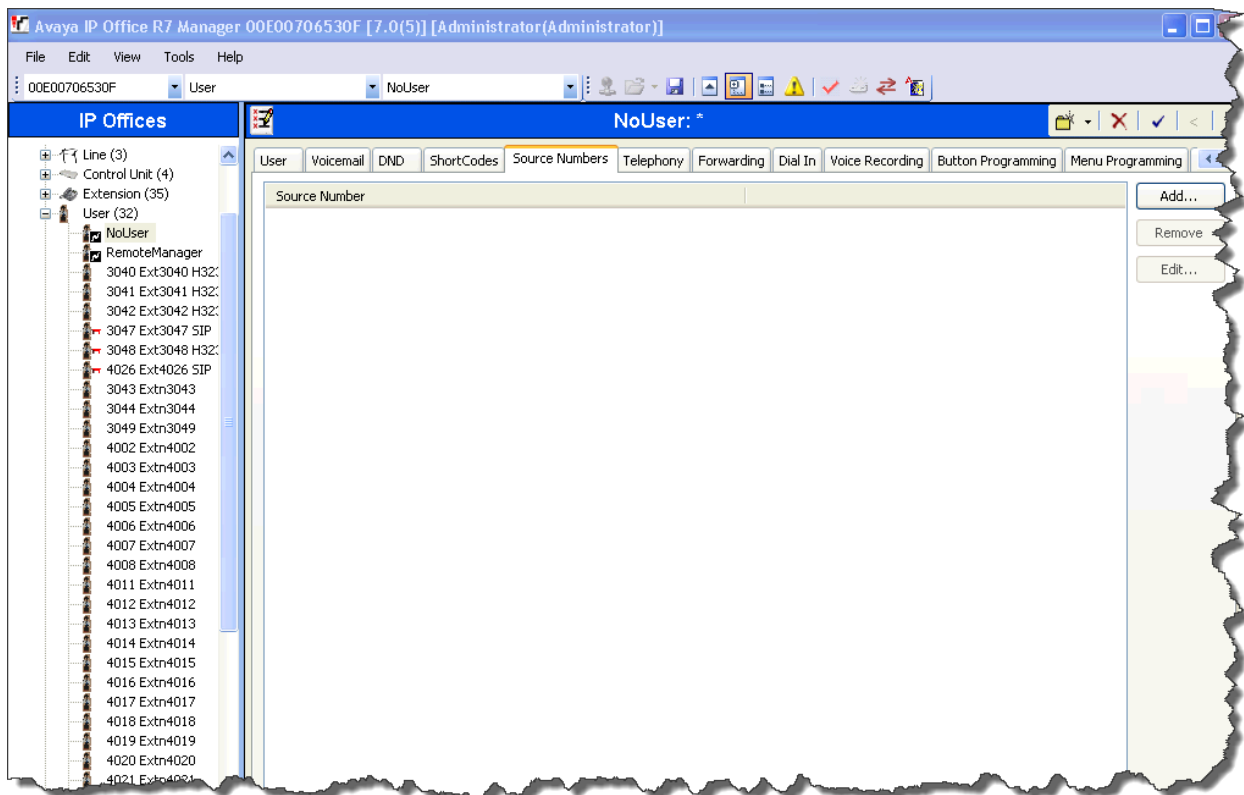
Under the **Destinations** tab, enter “.” for **Destination**. This will enable all incoming calls to be routed to any extension.



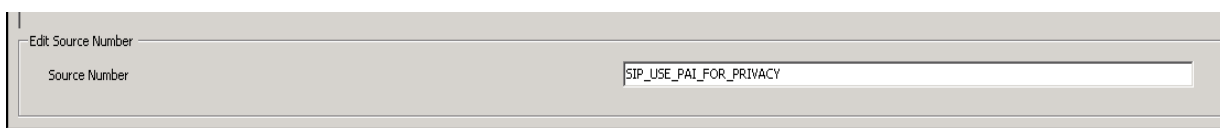
## 5.9 Privacy/Anonymous Calls

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.

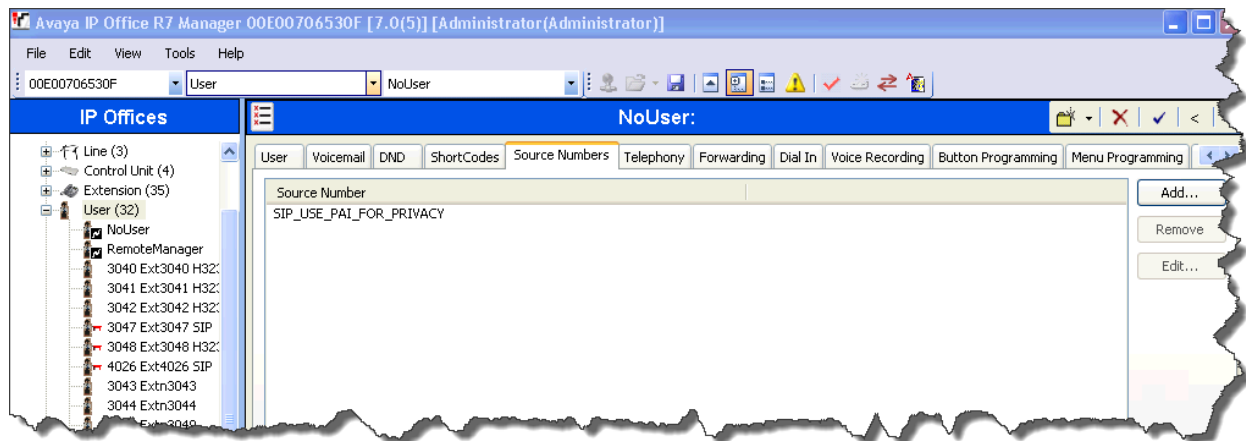
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User** → **NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP\_USE\_PA1\_FOR\_PRIVACY**. Click **OK**.



The **SIP\_USE\_PAIFOR\_PRIVACY** parameter will appear in the list of Source Numbers as shown below.

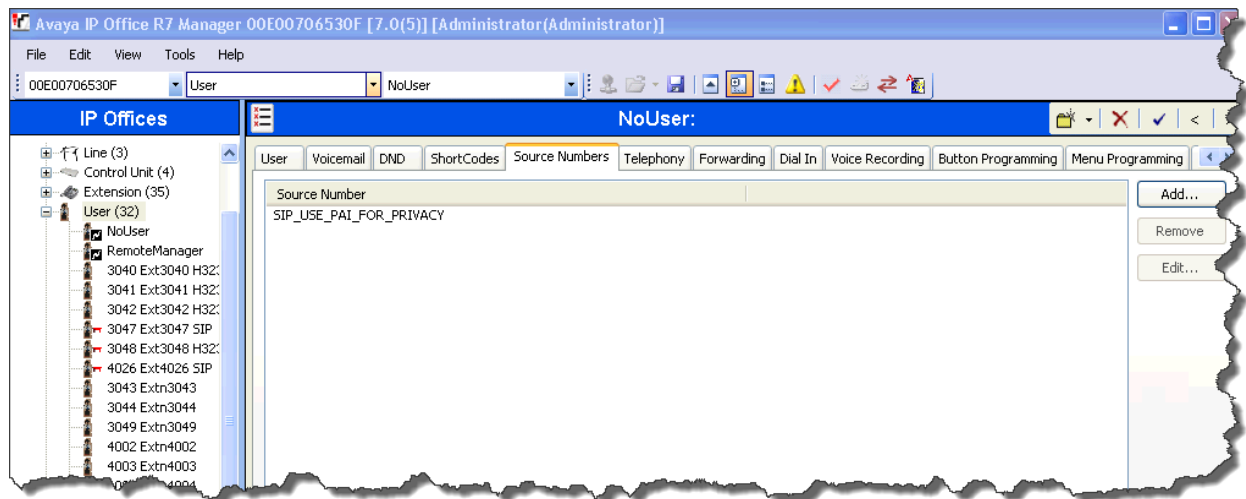


## 5.10 SIP Options Frequency

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 5.1** 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 44 seconds is used.
- To establish a period less than 42 seconds, do not define a **SIP\_OPTIONS\_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 42 seconds, a **SIP\_OPTIONS\_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 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.

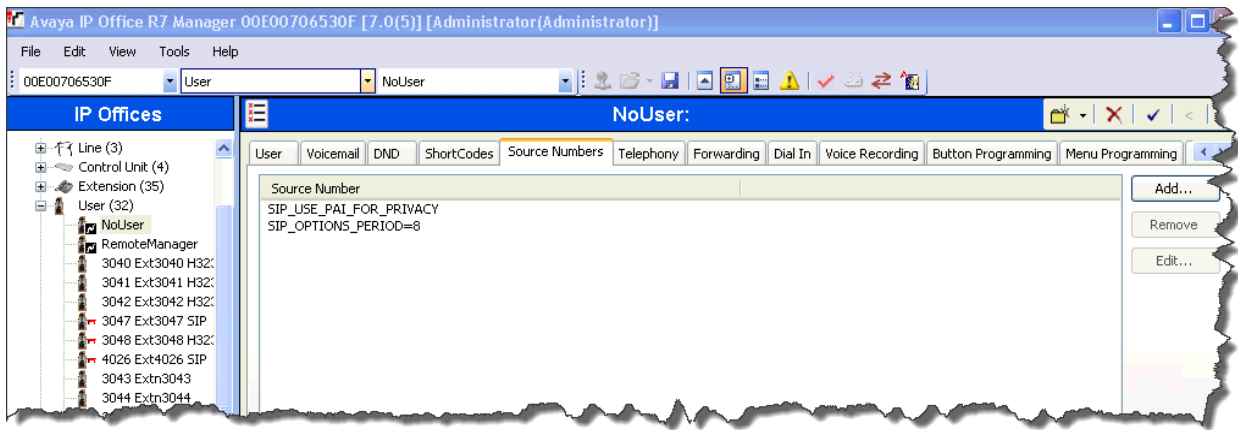


At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP\_OPTIONS\_PERIOD=X**, where **X** is the desired value in minutes. Click **OK**.

Edit Source Number

Source Number	SIP_OPTIONS_PERIOD=8
---------------	----------------------

For the compliance test, an OPTIONS period of 5 minutes was desired. The **Binding Refresh Time** was set to 300 seconds (5 minutes) in **Section 5.1**; the **SIP\_OPTIONS\_PERIOD** was set to 8 minutes. Avaya IP Office chose the **Binding Refresh Time** of 300 seconds (5 Minutes) as the smaller of these two values.

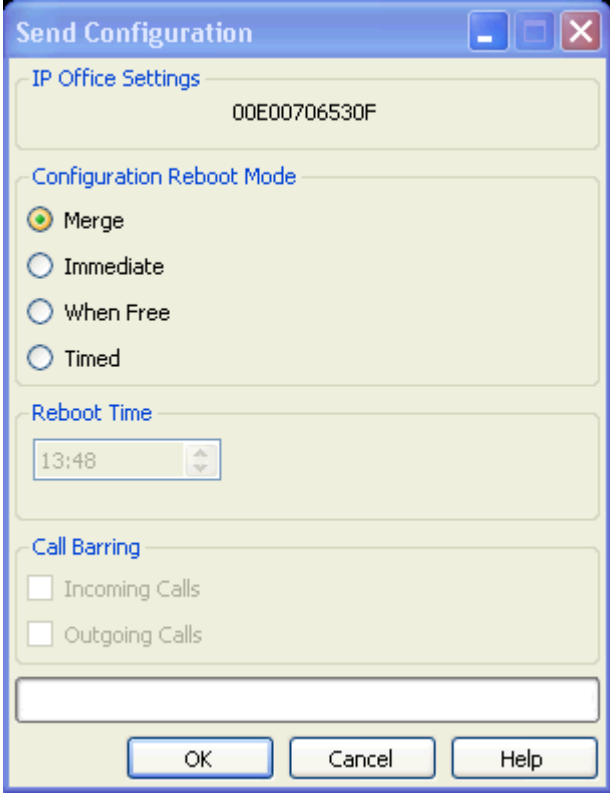


## 5.11 Save Configuration

When desired, send the configuration changes made in IP Office Manager to the 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 desire.



The image shows a Windows-style dialog box titled "Send Configuration". It contains several sections: "IP Office Settings" with a text field showing "00E00706530F"; "Configuration Reboot Mode" with four radio buttons: "Merge" (selected), "Immediate", "When Free", and "Timed"; "Reboot Time" with a time picker set to "13:48"; and "Call Barring" with two unchecked checkboxes: "Incoming Calls" and "Outgoing Calls". At the bottom is an empty text field and three buttons: "OK", "Cancel", and "Help".

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

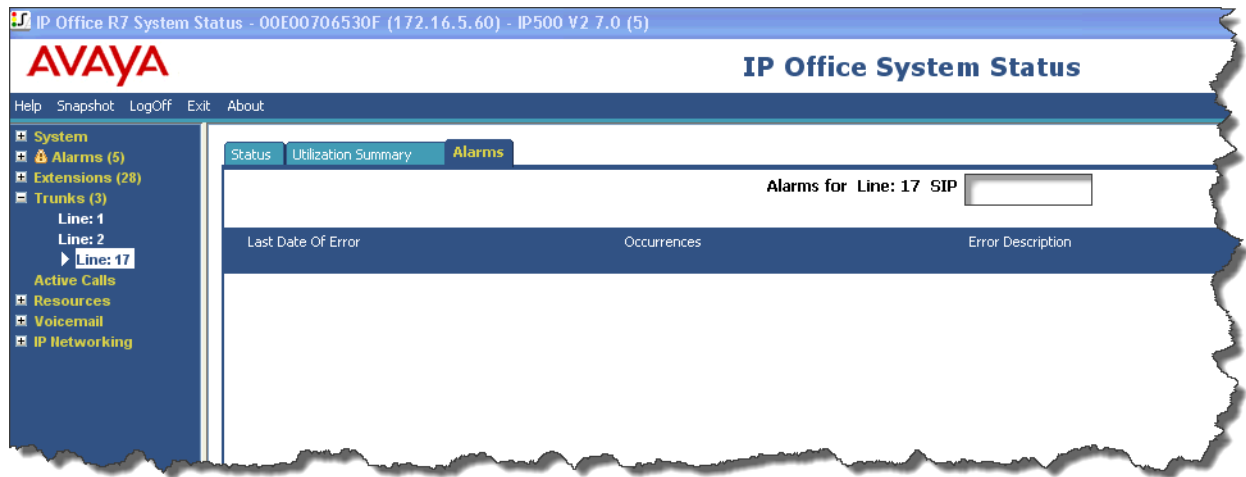
The screenshot displays the Avaya IP Office System Status application. The title bar indicates the application is running on IP Office R7 System Status - 00E00706530F (172.16.5.60) - IP500 V2 7.0 (5). The Avaya logo is visible in the top left. The main menu includes Help, Snapshot, LogOff, Exit, and About. The left sidebar contains a tree view with the following items: System, Alarms (5), Extensions (28), Trunks (3), Line: 1, Line: 2, Line: 17 (selected), Active Calls, Resources, Voicemail, and IP Networking. The main content area is titled 'IP Office System Status' and has three tabs: Status (selected), Utilization Summary, and Alarms. The Status tab displays the 'SIP Trunk Summary' with the following details:

- Peer Domain Name:
- Resolved Address:
- Line Number: 17
- Number of Administered Channels: 10
- Number of Channels in Use: 0
- Administered Compression: G711A
- Silence Suppression: Off
- SIP Trunk Channel Licences: Unlimited
- SIP Trunk Channel Licences in Use: 0 (0%)
- SIP Device Features: REFER (incoming and outgoing)

Below the summary is a table showing the current state of each channel:

Channel Number	Current State
1	Idle
2	Idle
3	Idle
4	Idle
5	Idle
6	Idle
7	Idle
8	Idle
9	Idle
10	Idle

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



- Verify that a phone connected to Avaya IP Office can successfully place calls to the PSTN with two-way audio.
- Verify that a phone connected to PSTN can successfully place calls to Avaya IP Office with two-way audio.

## 7. Conclusion

Alestra's 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 and Alestra's Enlace IP SIP Trunk Service, as shown in **Figure 1**

## 8. References

[1] *IP Office 7.0 IP Office Standard Version Installation*, Document number 15-601042, 22 May 2011.

<http://support.avaya.com/css/P8/documents/100129376>

[2] *IP Office Release 7.0 Manager 9.0*, Document number 15-601011, 22 May 2011.

<http://support.avaya.com/css/P8/documents/100129398>

[3] *IP Office Release 7.0 Voicemail Pro Installation*, Issue 01a, 01 May 2011.

<http://support.avaya.com/css/P8/documents/100134564>

[4] *IP Office Release 6.0 System Status Application*, Document number 15-601758, 12 February 2010.

<https://support.avaya.com/css/P8/documents/100073300>

[5] *IP Office Release 7.0 IP Office Softphone Installation*, Issue 2b, 21 February 2011.

<http://support.avaya.com/css/P8/documents/100129396>

Product documentation for Avaya products may be found at <http://support.avaya.com>.

Product documentation for Alestra's Enlace IP SIP Trunk Service is available from Alestra.

---

**©2011 Avaya Inc. All Rights Reserved.**

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

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