



## Avaya Solution & Interoperability Test Lab

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# Applications Notes for Avaya IP Office 7.0 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

### Abstract

These Application Notes describe the steps for configuring Avaya IP Office 7.0 with the AT&T IP Toll Free service. The Avaya IP Office solution was tested with the AT&T IP Toll Free service using **MIS/PNT** transport.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks for business customers.

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program.

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## 1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office Preferred Edition 7.0 with the AT&T IP Toll Free service. The Avaya IP Office solution was tested with the AT&T IP Toll Free service using **MIS/PNT** transport.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks for business customers.

## 2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with IP Office, Avaya phones and fax machines (Ventafax application).
- A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise was connected via MIS/PNT transport.

The main test objectives were to verify the following features and functionality:

- Inbound AT&T IP Toll Free service calls to IP Office hunt groups/telephones.
- Call and two-way talk path establishment between PSTN and IP Office phones via the AT&T Toll Free service.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729 and G.711 codecs.
- T.38 and G.711 fax calls from AT&T IP Toll Free service/PSTN to Avaya IP Office G3 and SG3 fax endpoints.
- DTMF tone transmission using RFC 2833 between IP Office and the AT&T IP Toll Free service/PSTN automated access systems.
- Inbound AT&T IP Toll Free service calls to IP Office that are directly routed to stations, and unanswered, can be covered to Voicemail Pro.
- Long duration calls.

## 2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound and outbound call flows (see **Section 3.2** for examples) between Avaya IP Office and the AT&T IP Toll Free service.

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network. Calls were made from the PSTN across the AT&T network (see **Section 3.2** for sample call flows). The following features were tested as part of this effort:

- SIP trunking
- T.38 and G.711 fax
- Passing of DTMF events and their recognition by navigating automated voice menus
- PBX and AT&T IP Toll Free service features such as hold, resume, conference and transfer
- Legacy Transfer Connect
- Alternate Destination Routing

## 2.2. Known Limitations/Test Results

1. Avaya IP Office supports G.711 faxing only for inbound call and therefore it works with AT&T IP Toll Free service.
2. AT&T IP Transfer connect option of the AT&T IP Toll Free service was not verified with Avaya IP Office 7.0 and hence not supported.
3. Shuffling is not supported for SIP trunks in Avaya IP Office 7.0.
4. G.726 codec is not supported by Avaya IP Office 7.0.

The test objectives stated in **Section 2** with limitations as noted in this section were verified.

## 2.3. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (888) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on <http://support.avaya.com>) to directly access specific support and consultation services based upon their Avaya support agreements.

### 3. Reference Configuration

The reference configuration used in these Application Notes is shown in the figure below and consists of several components:

- IP Office provides the voice communications services for a particular enterprise site. In the reference configuration, IP Office runs on an IP 500 V2.
- Avaya “desk” phones are represented with Avaya 1616I, 4625 and 9630 IP Telephones running H.323 software, Avaya Digital Phones (1416, T7100 and 7316E), Avaya 6211 Analog Telephone, Avaya SIP Phones (1140E and 1230) and PC based IP Office Softphone.
- Voicemail Pro provides the voice messaging capabilities in the reference configuration and its provisioning is beyond the scope of this document.
- Inbound calls from PSTN were sent from AT&T IP Toll Free service to IP Office. IP Office terminated the call to the appropriate agent/phone or fax extension. Signaling is between IP Office public interface and the AT&T Border Element.
- Enterprise sites may have additional or alternate routes to PSTN using analog or digital TDM trunks. However these trunks were not used in this reference configuration.

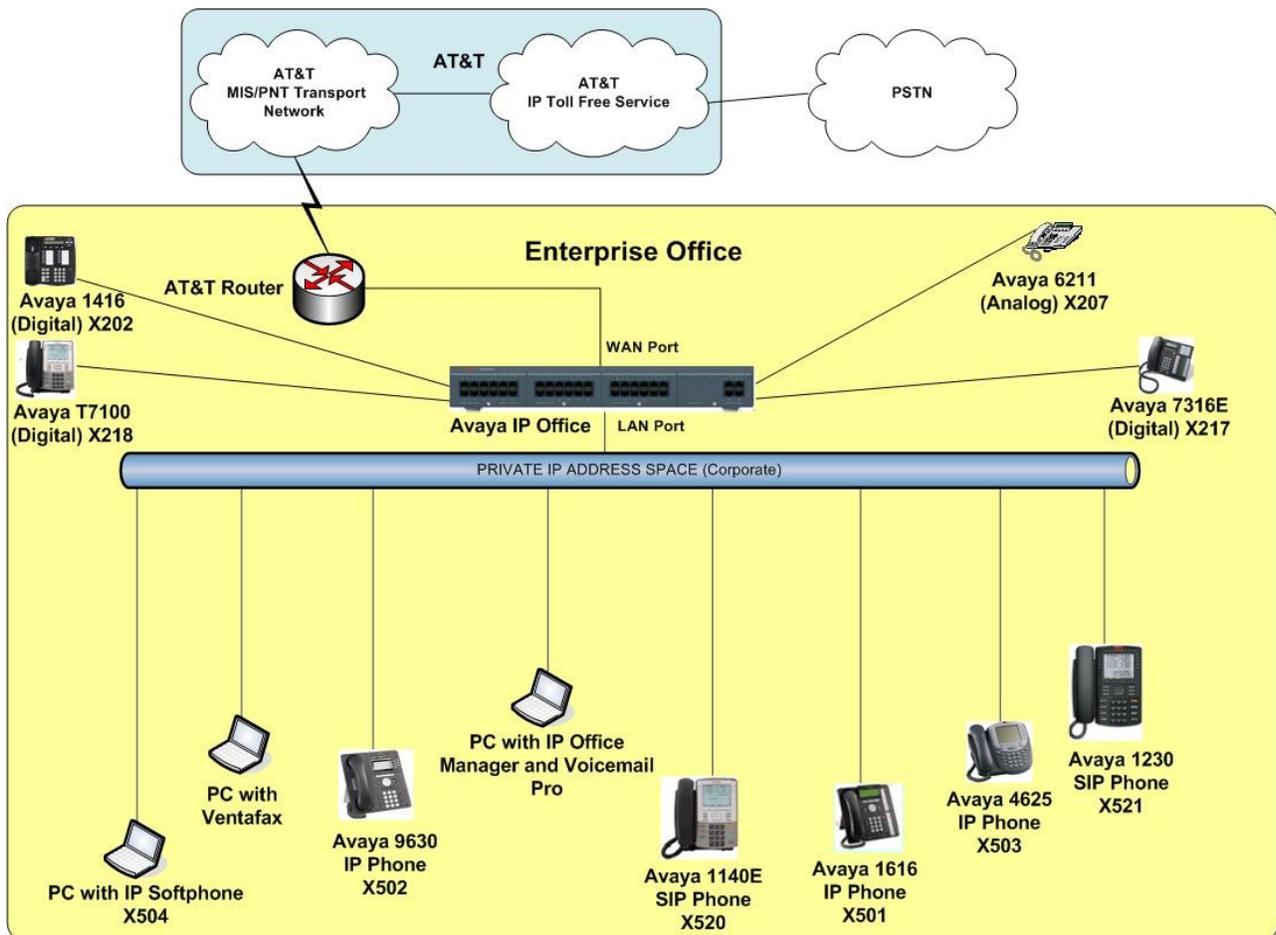


Figure 1: Reference configuration

### 3.1. Illustrative Configuration Information

The specific values listed in the table below and in subsequent sections are used in the reference configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

**Note** - The AT&T IP Toll Free service Border Element IP address shown in this document is only an example. AT&T Customer Care will provide the actual IP addresses as part of the AT&T IP Toll Free service provisioning process.

Component	Illustrative Value in these Application Notes
<b>Avaya IP Office</b>	
Public IP Address	192.168.62.59
Private IP Address	10.80.130.58
Avaya IP Office Extensions	207 = Analog 501,502,503=H323 202,217,218=Digital 504=Softphone 520,521= SIP phones
<b>AT&amp;T IP Toll Free Service</b>	
Border Element IP Address	135.242.225.200
Digits passed in SIP-URI Request	0000011001 – CPN Basic 0000021002 - CPN Restricted 0000031003 – Legacy Xfer Connect 0000041004 – ADR 0000051005 – ADR Secondary

**Table 1: Illustrative Values Used in these Application Notes**

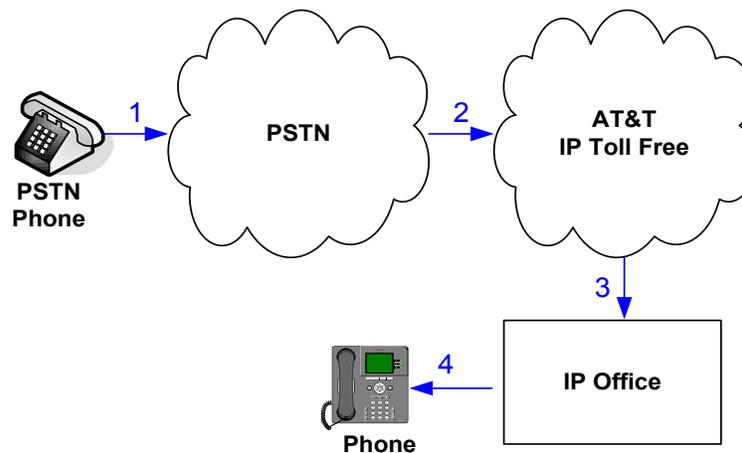
## 3.2. Call Flows

To understand how inbound AT&T IP Toll Free service calls are handled by IP Office, two basic call flows are described in this section.

### 3.2.1. Inbound

The first call scenario illustrated in the figure below is an inbound AT&T IP Toll Free service call that arrive on IP Office, which in turn routes the call to a hunt group, phone or a fax.

1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to IP Office.
4. Depending on the called number, IP Office routes the call to
  - A hunt group, which in turn, routes the call to an agent
  - Directly to an agent or a phone/fax extension

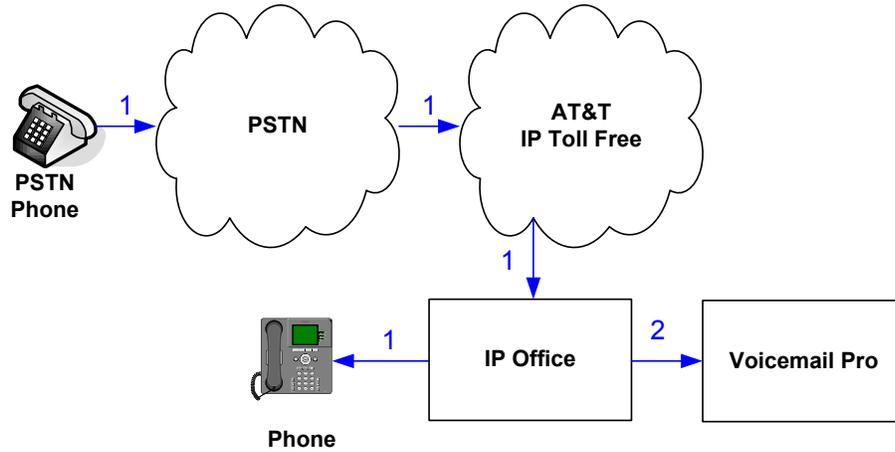


**Inbound - AT&T IP Toll Free**

### 3.2.2. Coverage to Voicemail

The call scenario illustrated in the figure below is an inbound call that is covered to voicemail. In this scenario, the voicemail system is Voicemail Pro software installed on a PC.

1. Same as the first call scenario in **Section 3.2.1**.
2. The IP Office phone does not answer the call, and the call covers to the phone's voicemail. IP Office forwards the call to Voicemail Pro.



**Coverage to Voicemail**

## 4. Equipment and Software Validated

The following equipment and software was used for the reference configuration described in these Application Notes.

**Note** – Although Avaya IP Office Preferred Edition was used during this testing; Avaya IP Office Essential and Advanced Editions are also supported.

Component	Version
Avaya IP Office 500	Release 7.0 (5) (Preferred Edition)
Avaya IP Office Manager	Release 9.0 (5) (Preferred Edition)
Avaya IP Office Voicemail Pro	Release 7.0 (17)
Avaya IP Office Voicemail Pro Client	Version 7.0 (17)
Avaya 1616IP-Series Telephones (H.323)	Release 1.3
Avaya 9630 IP Telephone	Avaya one-X® Deskphone Edition H.323 Version S3.11
Avaya 4625SW IP Telephone	a25d01a2_9_1.bin
Avaya IP Office Softphone	Release 3.1.2.17 59616
Avaya 1416 Digital Telephone	-
Avaya T7100 Digital Phone	-
Avaya 7316E Digital Phone	-
Avaya 6211 Analog phone	-
Avaya 1140E SIP Telephone	04.00.13.00 (SIP1140)
Avaya 1230 SIP Telephone	04.00.13.00 (SIP1230)
Fax device	Ventafax Home Version 6.2
AT&T IP Toll Free Service using MIS/PNT transport service connections.	VNI 18

**Table 2: Equipment and Software Versions**

## 5. Avaya IP Office

This section describes attributes of the reference configuration, but is not meant to be prescriptive. The configuration steps described here are only for the fields where a value was changed. For all the other fields default values are used. Additionally, the screen shots referenced in these sections may not be complete at times. Consult reference [IPO-INSTALL] for more information on the topics in this section.

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [IPO-MGR]. 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:

### WELCOME to IP Office Administration

#### What would you like to do ?

[Create an Offline Configuration](#)

[Open Configuration from System](#)

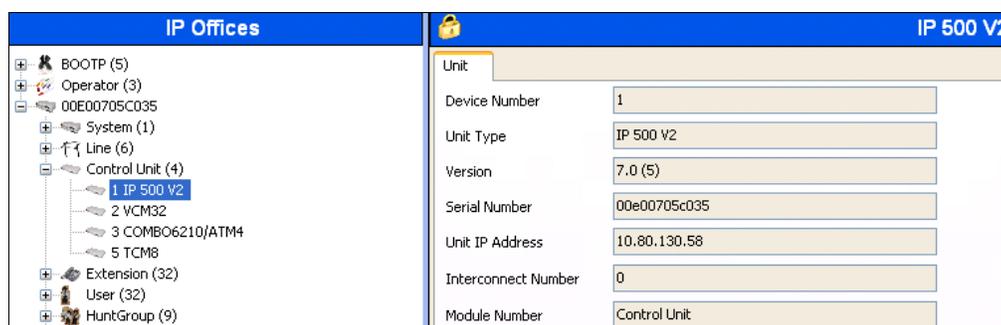
[Read a Configuration from File](#)

Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. 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.

## 5.1. Physical, Network, and Security Configuration

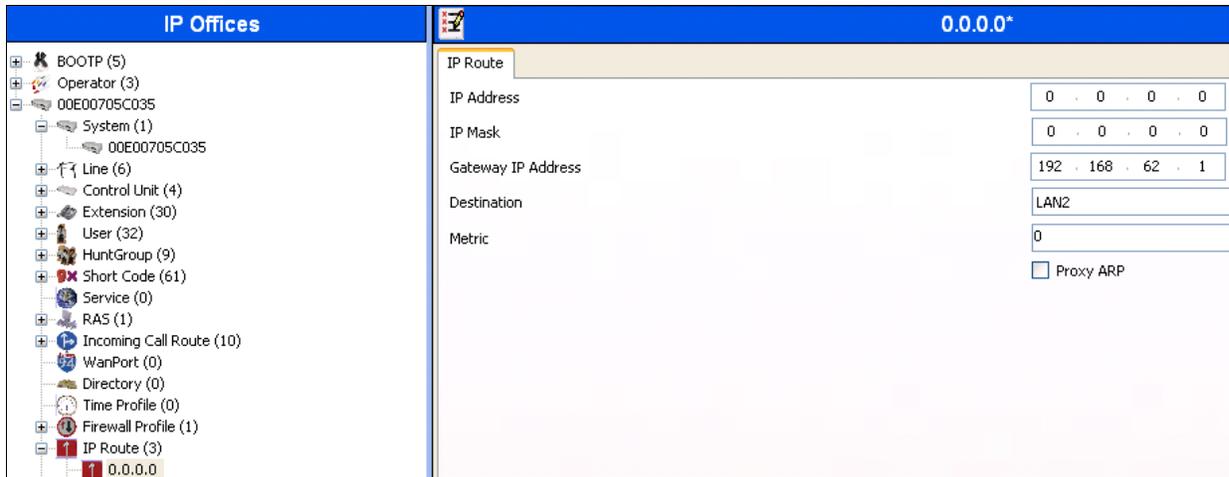
In the reference configuration, the IP Office 500 V2 contains a VCM32 module, COMBO6210/ATM4 module, and a TCM8 module. The VCM32 is a Voice Compression Module supporting VoIP codecs. The COMBO6210/ATM4 was used in this reference configuration to support digital and analog telephones or fax machines. The TCM8 module was used to support heritage Avaya/Nortel digital phone extensions.

The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Details pane. In the screen below, **IP 500 V2** is selected in the Navigation pane, revealing additional information about the IP 500 V2 in the Details pane.

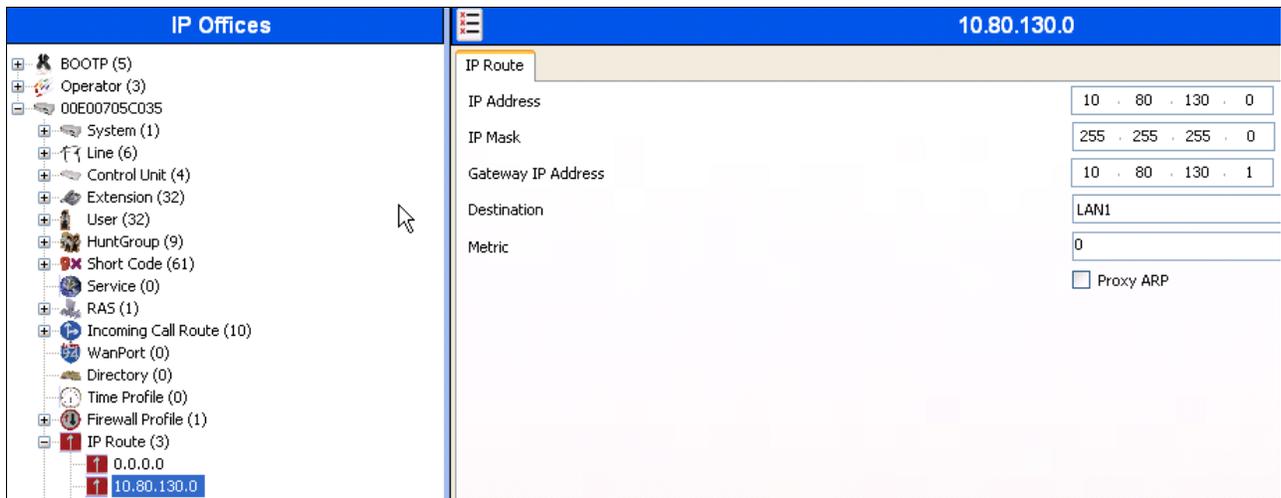


In this reference configuration, the IP Office **LAN2** port (labeled as WAN port in Figure 1) is physically connected to the public network at the IP Office customer site. The default gateway for this network is **192.168.62.1**.

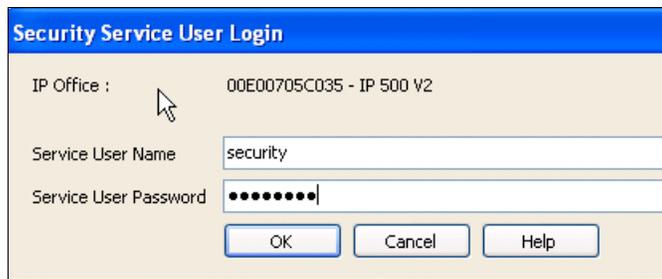
1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New** [not shown]. To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant default route using **LAN2** as configured in **Destination** field (Refer **Section 5.3.2**).



- Another route for **10.80.130.0** subnet was added for the enterprise side **LAN1** port (labeled as LAN port in Figure 1) as shown in the screen below. All the IP devices were part of this 10.80.130.x network in this reference configuration.

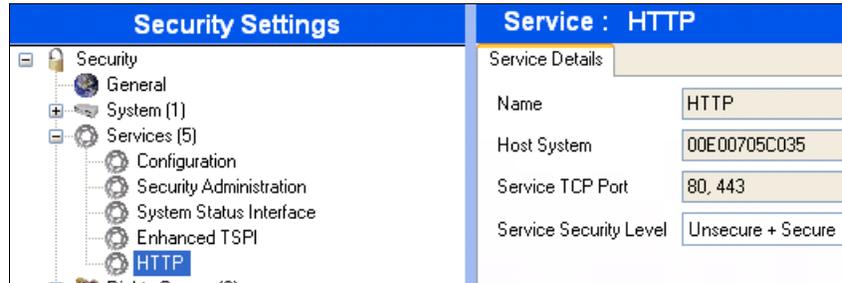


- For use of Avaya IP Office Softphone, navigate to **File** → **Advanced** → **Security Settings** and login with proper credentials in the screen shown below.





- After logging in, navigate to **Services** → **HTTP** and verify that **Service Security Level** field is set to **Unsecure + Secure**. Note that this action may be service disrupting.

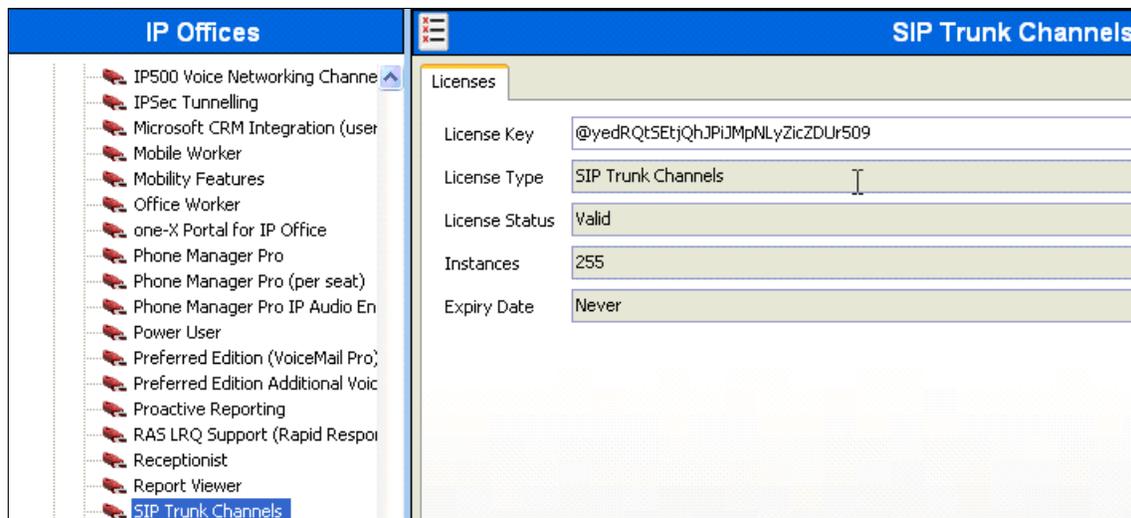


- When complete, select **File** → **Configuration** to return to configuration activities.

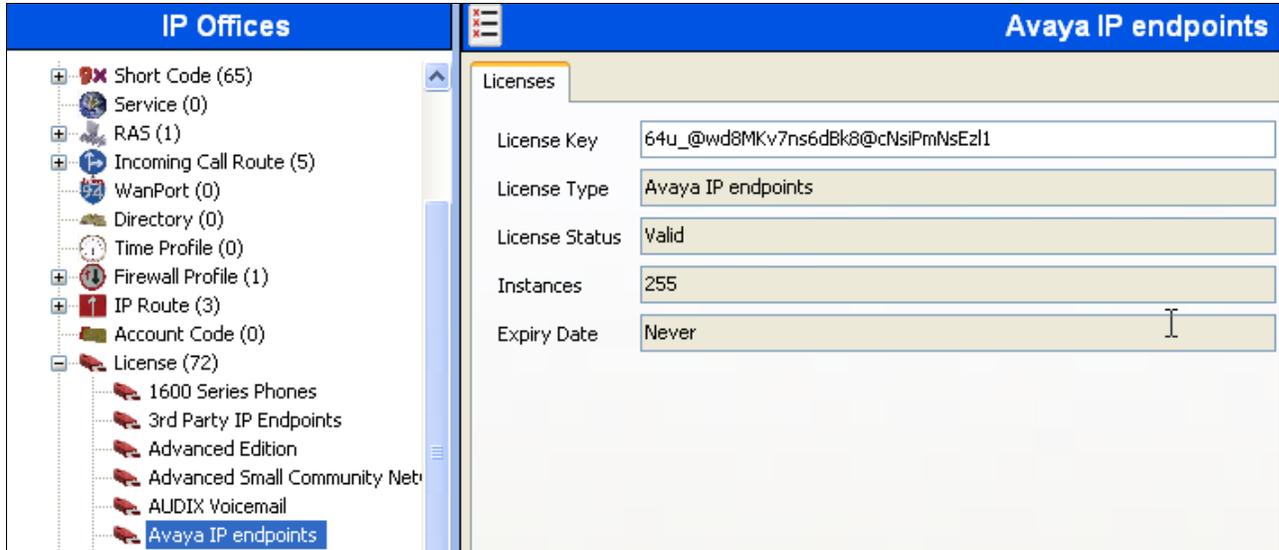
## 5.2. Licensing

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

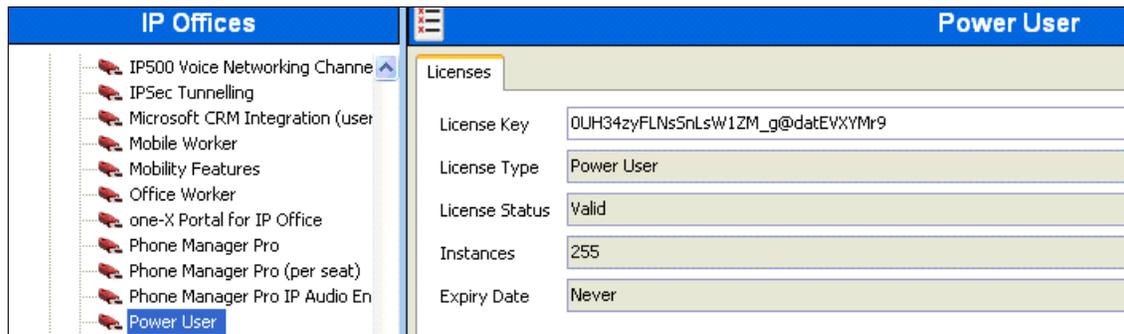
- To verify that SIP Trunk Channels has sufficient capacity, navigate to **License** → **SIP Trunk Channels** in the Navigation pane and confirm a valid license with sufficient **Instances** (trunk channels) exist in the Details pane.



- To verify Avaya IP endpoints with sufficient capacity, navigate to **License** → **Avaya IP endpoints** in the Navigation pane and confirm a valid license with sufficient **Instances** exist in the Details pane.



- The following screen shows the availability of a valid license for **Power User** features. In this reference configuration, the user with extension **501** (**Section 5.5.2**) is configured as a **Power User** and is capable of using the IP Office Softphone too.

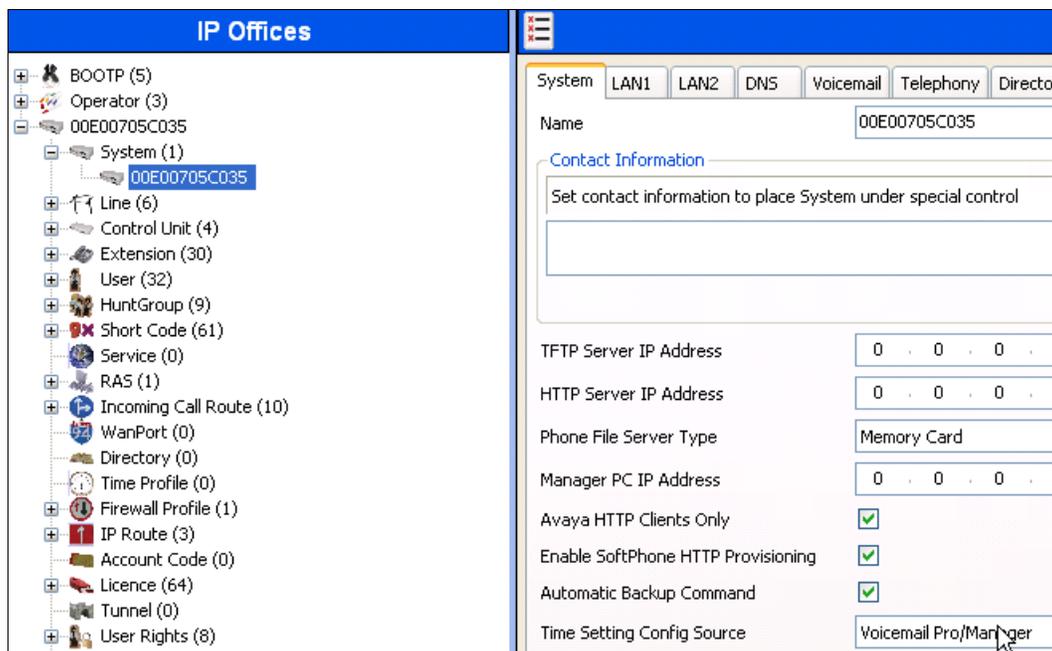


### 5.3. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The configuration in following sections is for reference purposes only.

#### 5.3.1. System Tab

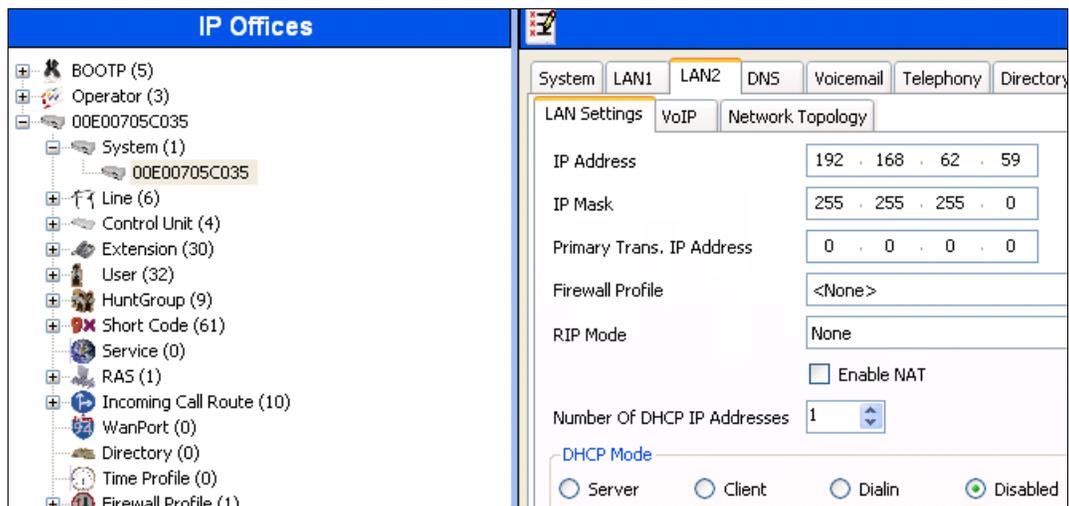
With the proper system name selected in the Navigation pane, select the **System** tab in the Details pane. The following screen shows a portion of the **System** tab. The **Name** field is used for a descriptive name of the system. In this case, the MAC address is used as the name. The **Avaya HTTP Clients Only** and **Enable SoftPhone HTTP Provisioning** boxes are checked to facilitate IP Office Softphone usage.



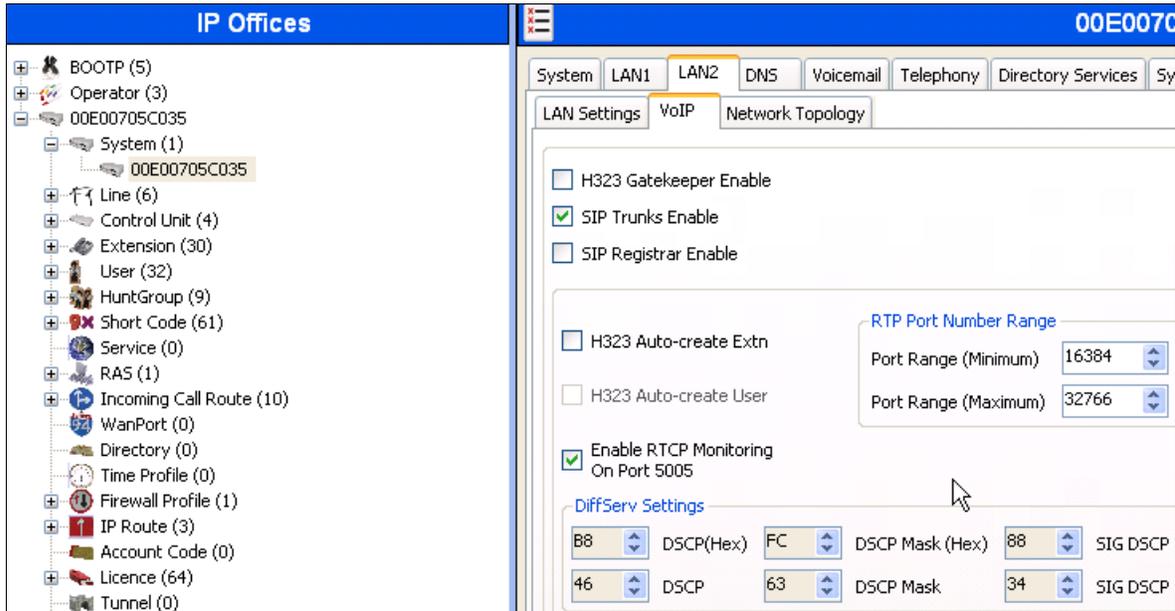
### 5.3.2. LAN Settings

In the sample configuration, **LAN2** was used to connect the IP Office to AT&T Network and **LAN1** was used to connect to the enterprise network.

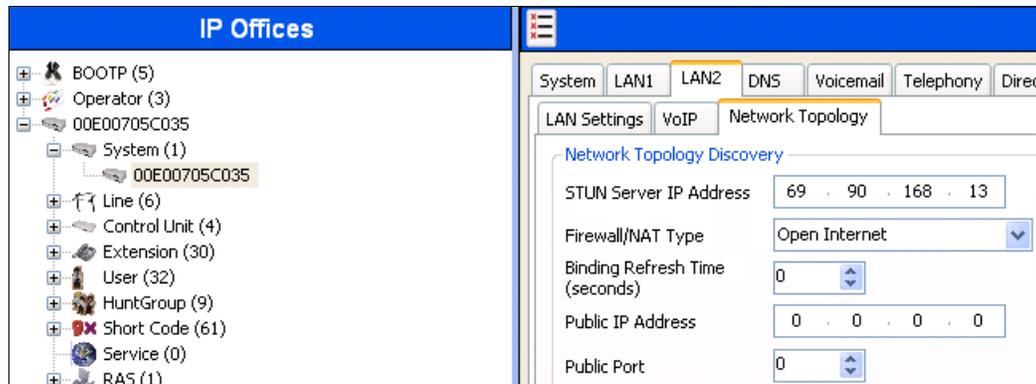
1. Select the **LAN2** tab followed by the **LAN Settings** tab and configure as follows:
  - **IP Address** – Set to **192.168.62.59** which is the IP address of IP Office known to AT&T network
  - **IP Mask** – Set to a valid value e.g **255.255.255.0**
  - **Primary Trans. IP Address** – Set to **0.0.0.0**
  - **DHCP Mode** – Select the **Disabled** radio button



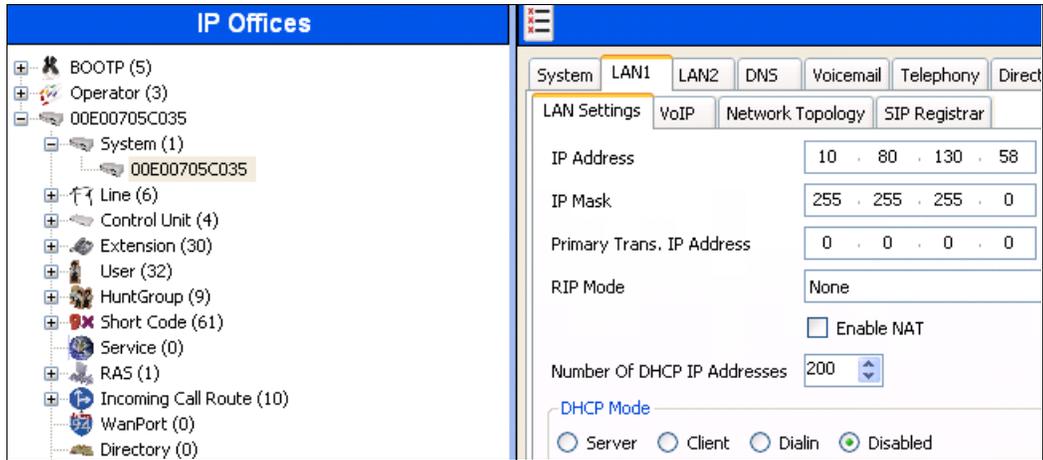
2. Select the **VoIP** tab as shown in the following screen and configure as follows:
  - **SIP Trunks Enable** – Check this box to enable the configuration of SIP trunks
  - **RTP Port Range (Minimum)** – Set to **16384** (As required by AT&T)
  - **RTP Port Range (Maximum)** – Set to **32766** (As required by AT&T). Although AT&T requires the maximum value to be **32767**, IP Office needs an even number to be entered in this field



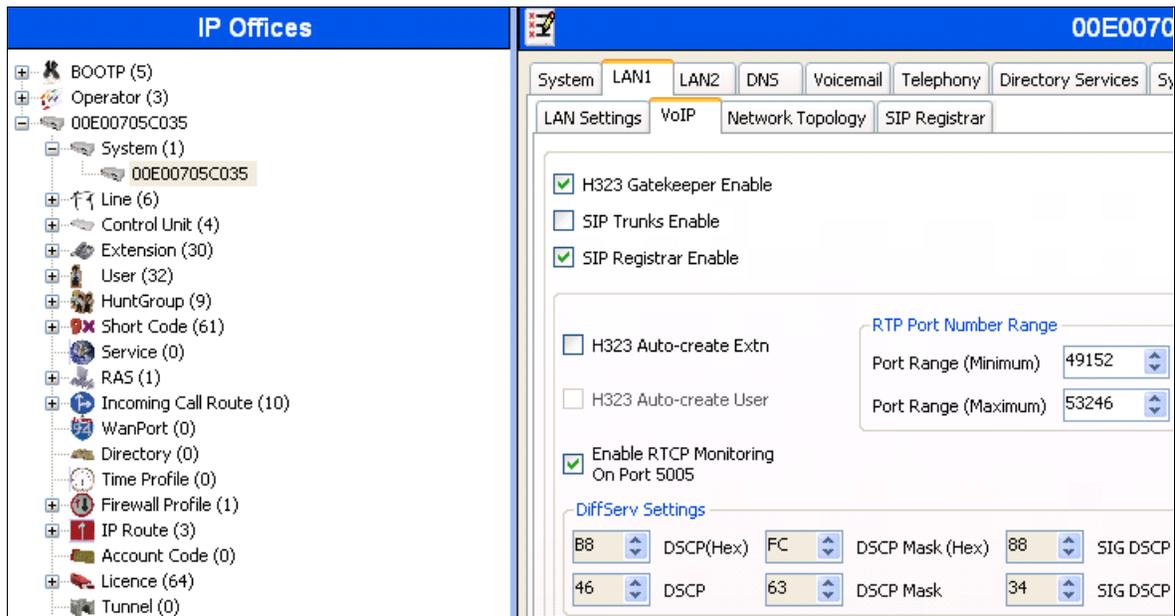
3. Select the **Network Topology** tab as shown in the following screen and set **Firewall/NAT Type** field to **Open Internet**. With this configuration, STUN will not be used but make sure to leave **STUN Server IP Address** to its default value.



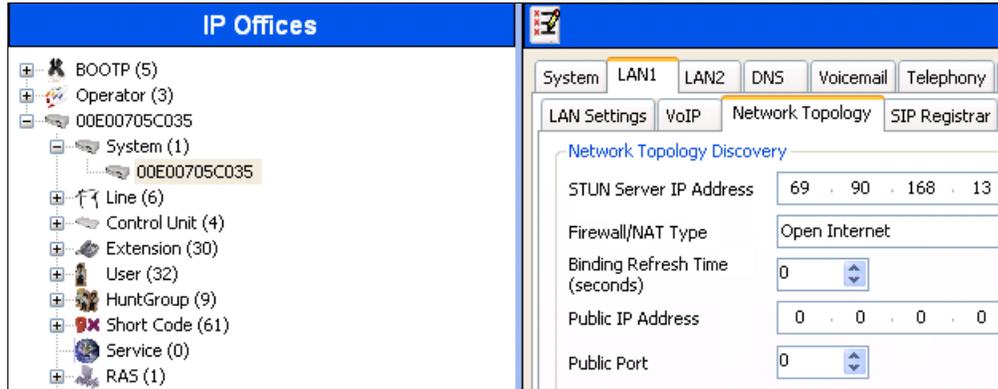
- Select the **LAN1** tab followed by the **LAN Settings** tab and set **IP Address** of the IP Office on the enterprise side to **10.80.130.58** and **IP Mask** to **255.255.255.0**.



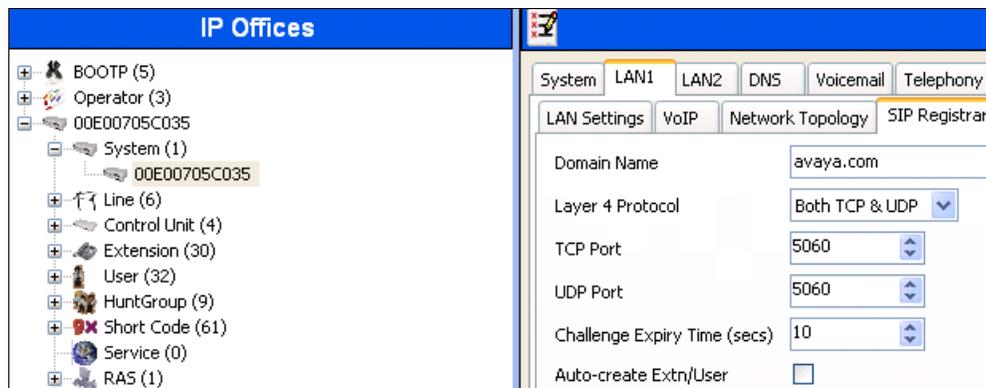
- Select the **VoIP** tab as shown in the following screen and configure as follows:
  - H323 Gatekeeper Enable** – Check this box to allow the use of Avaya IP Phones
  - SIP Registrar Enable** – Check this box to allow SIP phones and IP Office Softphone usage



6. The Network Topology screen is set the same as it was set for LAN2 in Step 3.



7. Select the SIP Registrar tab and set the **Domain Name** field to the enterprise SIP domain (e.g. **avaya.com**) and leave all the other fields to their default values. This domain name is used to register the SIP telephones. Also, make sure that the **Layer 4 Protocol** field is set to **Both TCP & UDP** as Avaya IP Softphone uses UDP and the SIP phones require TCP.

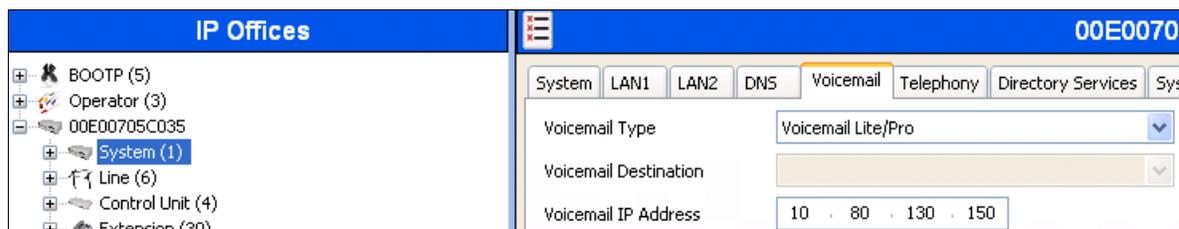


8. Click **OK** [not shown] to commit.

### 5.3.3. Voicemail

Select **Voicemail** tab and configure as follows:

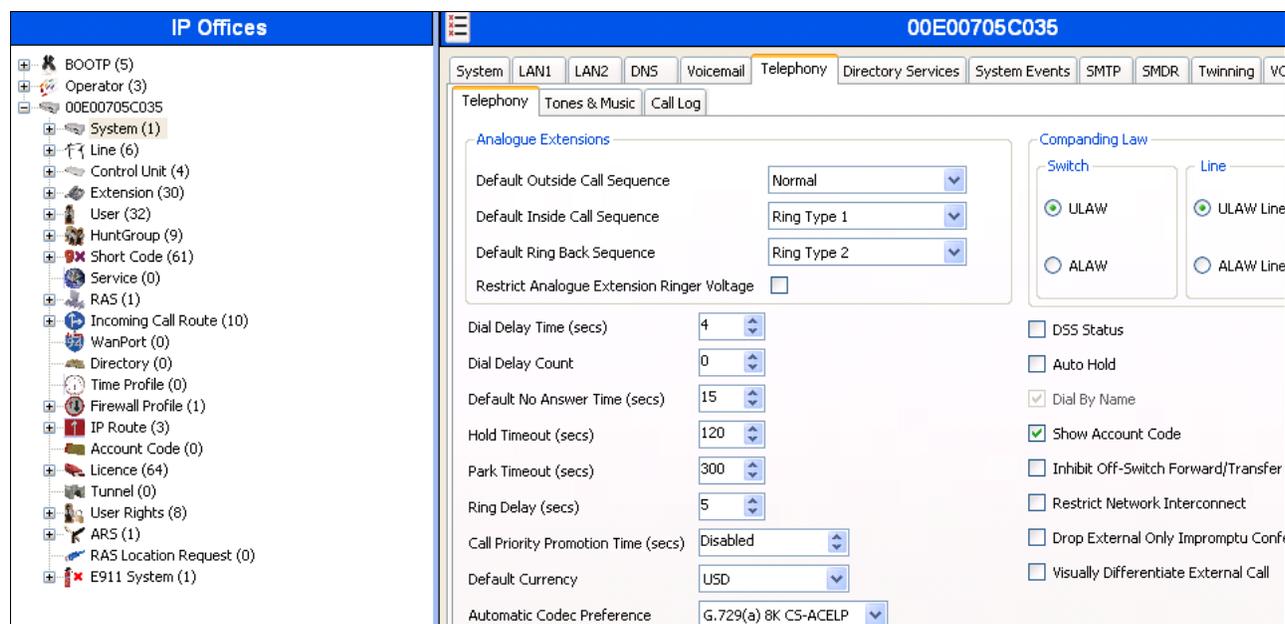
- **Voicemail Type** – Set to **Voicemail Lite/Pro** from the drop-down list
- **Voicemail IP Address** – Set to **10.80.130.150**, the IP Address of the PC running the Voicemail Pro software.



### 5.3.4. System Telephony Configuration

Select the **Telephony** tab and in **Telephony** sub-tab configure as follows:

- **Companding Law** – Check the **ULAW** box for **Switch** field and **ULAW Line** box for the **Line** field.
- **Inhibit Off-Switch Forward/Transfer** – Uncheck this box so that call forwarding and call transfer to PSTN destinations via the AT&T Toll Free service can be tested.
- **Automatic Codec Preference** – Use the default value of **G.729(a) 8K CS-ACELP**.



## 5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 7.0. To add a new SIP Line, right click on **Line** in the Navigation pane, and select **New** → **SIP Line** [not shown]. A new Line Number is assigned automatically.

### 5.4.1. SIP Line - SIP Line Tab

Select **SIP Line** tab as shown below for Line Number **9** used for AT&T and configure as follows:

- **ITSP Domain Name** – Set to the IP Office LAN1 address (**192.168.62.59**) configured in **Section 5.3.2** so that IP Office uses this IP address in the host portion of SIP headers such as the From and Diversion headers
- **In Service** – Verify that box is checked (default)
- **Check OOS** – If this box is checked, it enables IP Office to use the SIP OPTIONS method to periodically check the SIP Line and if no response is received, the SIP line is taken out of service. See **Section 5.9** for additional information related to configuring the periodicity of SIP OPTIONS
- **Call Routing Method** – Set to **Request URI** (default)
- **REFER Support** – Uncheck the box

The screenshot displays the IP Office configuration interface. On the left is the 'IP Offices' navigation pane, and on the right is the 'SIP Line - Line 9\*' configuration window. The 'SIP Line' tab is selected, showing various configuration options.

Field	Value	Field	Value
Line Number	9	In Service	<input checked="" type="checkbox"/>
ITSP Domain Name	192.168.62.59	Use Tel URI	<input type="checkbox"/>
Prefix		Check OOS	<input checked="" type="checkbox"/>
National Prefix	0	Call Routing Method	Request URI
Country Code		Originator number for forwarded and twinning calls	
International Prefix	00		
Send Caller ID	Diversion Header		
Association Method	By Source IP address		
REFER Support	<input type="checkbox"/>		
Incoming	Auto		
Outgoing	Auto		

## 5.4.2. SIP Line - Transport Tab

Select the **Transport** tab and set the **ITSP Proxy Address** to the AT&T Border Element IP Address. The **Use Network Topology Info** parameter is set to **LAN 2** configured in **Section 5.3.2**. Default values are used for the other fields.

The screenshot displays the configuration interface for a SIP Line, specifically the 'Transport' tab for 'Line 9\*'. The interface is divided into two main sections: a tree view on the left and a configuration panel on the right.

**Left Panel (IP Offices):** A tree view showing the hierarchy of network elements. The 'Line (6)' folder is expanded, showing lines 5, 6, 7, 8, and 9. Line 9 is highlighted with a mouse cursor. Other folders include BOOTP (5), Operator (3), System (1), Control Unit (4), Extension (30), User (32), and HuntGroup (9).

**Right Panel (SIP Line - Line 9\*):** The configuration panel for the selected line. It features several tabs: SIP Line, Transport (selected), SIP URI, VoIP, T38 Fax, and SIP Credentials. The configuration includes:

- ITSP Proxy Address:** A text field containing the IP address 135.242.225.200.
- Network Configuration:** A section with two rows of settings:
  - Layer 4 Protocol: A dropdown menu set to UDP.
  - Send Port: A spin box set to 5060.
  - Use Network Topology Info: A dropdown menu set to LAN 2.
  - Listen Port: A spin box set to 5060.
- Explicit DNS Server(s):** Two IP address input fields, both containing 0 . 0 . 0 . 0.
- Calls Route via Registrar:** A checkbox that is checked.
- Separate Registrar:** An empty text field.

### 5.4.3. SIP Line - SIP URI Tab

Select the **SIP URI** tab and click the **Add...** button in Details Pane [not shown] to add a new SIP URI. Configure the **New Channel** section displayed as follows:

- **Local URI** – Set to the DNIS sent by AT&T IP Toll Free service in the SIP URI. In this example it is set to **0000011001** which is one of the DNIS mentioned in **Table 1**.
- **Registration** - Set to **0: <None>**
- **Incoming Group** and **Outgoing Group** – Set to **110**

Repeat above steps for other DNIS provided by AT&T.

The screenshot displays the Avaya SIP Line configuration interface. On the left is a tree view of 'IP Offices' containing various system components like BOOTP, Operator, System, Line, Control Unit, Extension, User, etc. The main pane is titled 'SIP Line - Line 9\*' and has several tabs: SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'SIP URI' tab is active, showing a table with 5 channels. Below the table is an 'Edit Channel' form with the following fields:

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	110 110	2...	0000011001			None	0: <None>	20
2	110 110	2...	0000021002			None	0: <None>	10
3	110 110	2...	0000031003			None	0: <None>	10
4	110 110	2...	0000041004			None	0: <None>	10
5	110 110	2...	0000051005			None	0: <None>	10

The 'Edit Channel' form shows the following values:

- Via: 205.168.62.59
- Local URI: 0000011001
- Contact: Use Internal Data
- Display Name: Use Internal Data
- PAI: None
- Registration: 0: <None>
- Incoming Group: 110
- Outgoing Group: 110
- Max Calls per Channel: 20

#### 5.4.4. SIP Line - VoIP Tab

Select the **VoIP** tab and configure as follows:

- **Compression Mode** – Set to **Automatic Select** from the drop-down list
- **DTMF Support** - Set to the default value **RFC2833**.
- **Re-invite Supported** – Check 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
- **Use Offerer's Preferred Codec** – Check this box so that the top codec offered to IP Office is used if IP Office supports that codec

The screenshot shows the configuration window for a SIP Line (Line 9\*). The left pane shows a tree view of IP Offices, including BOOTP (5), Operator (3), 00E00705CD35, System (1), and Line (6) with sub-items 5, 6, 7, and 8. The right pane is titled 'SIP Line - Line 9\*' and has tabs for SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The VoIP tab is active, showing the following settings:

Setting	Value	Checkbox
Compression Mode	Advanced / Automatic Select	<input checked="" type="checkbox"/> VoIP Silence Suppression
Fax Transport Support	T38	<input checked="" type="checkbox"/> Re-invite Supported
Call Initiation Timeout (s)	4	<input checked="" type="checkbox"/> Use Offerer's Preferred Codec
DTMF Support	RFC2833	<input type="checkbox"/> Codec Lockdown

Since default values were used for T38 fax and AT&T IP Toll Free service does not require registration, the **T38 Fax** and **SIP Credentials** tabs need not be visited. Click **OK** [not shown] to commit the SIP Line configuration.

## 5.5. Users, Extensions, and Hunt Groups

In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions used will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the Navigation pane, and select **New** [not shown]. To edit an existing User, select **User** and select the appropriate user in the Navigation pane.

### 5.5.1. Digital Telephone User 217

The following screen shows the **User** tab for User **217**. This user corresponds to a digital phone.

The screenshot displays the IP Office configuration interface. On the left, a navigation tree under 'IP Offices' shows a hierarchy: BOOTP (5), Operator (3), 00E00705C035, System (1), Line (6), Control Unit (4), Extension (30), and User (32). Under 'User', a list of users is shown, with '217 Extn217' selected and highlighted in blue. The main area on the right is titled 'Extn217: 217' and contains several tabs: 'User', 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', and 'Dial In'. The 'User' tab is active, showing the following configuration fields:

- Name: Extn217
- Password: [Empty]
- Confirm Password: [Empty]
- Full Name: [Empty]
- Extension: 217
- Locale: [Empty]
- Priority: 5
- System Phone Rights: None
- Profile: Basic User
  - Receptionist
  - Enable SoftPhone
  - Enable one-X Portal Services
  - Enable one-X TeleCommuter
  - Ex Directory
- Device Type: T7316E (with a phone icon)
- User Rights
  - User Rights view: User data
  - Working hours time profile: <None>
  - Working hours User Rights: [Empty]
  - Out of hours User Rights: [Empty]

The following screen shows the Extension information for this user. To view, select **Extension** and the appropriate extension in the Navigation pane.

The screenshot displays the 'IP Offices' management console. The left-hand 'Navigation pane' shows a hierarchical tree structure: '00E00705C035' (System) -> 'System (1)' -> 'Line (6)' -> 'Control Unit (4)' -> 'Extension (30)'. A list of extensions is shown, with '73 217' selected and highlighted in blue. The right-hand pane, titled 'Digital Extension: 73 217', shows the configuration for this extension. The 'Extn' tab is active, and the configuration includes:

Extension Id	73
Base Extension	217
Caller Display Type	On
Reset Volume After Calls	<input type="checkbox"/>
Device type	T7316E
Module	BD4
Port	1
Disable Speakerphone	<input type="checkbox"/>

## 5.5.2. IP Telephone User 501

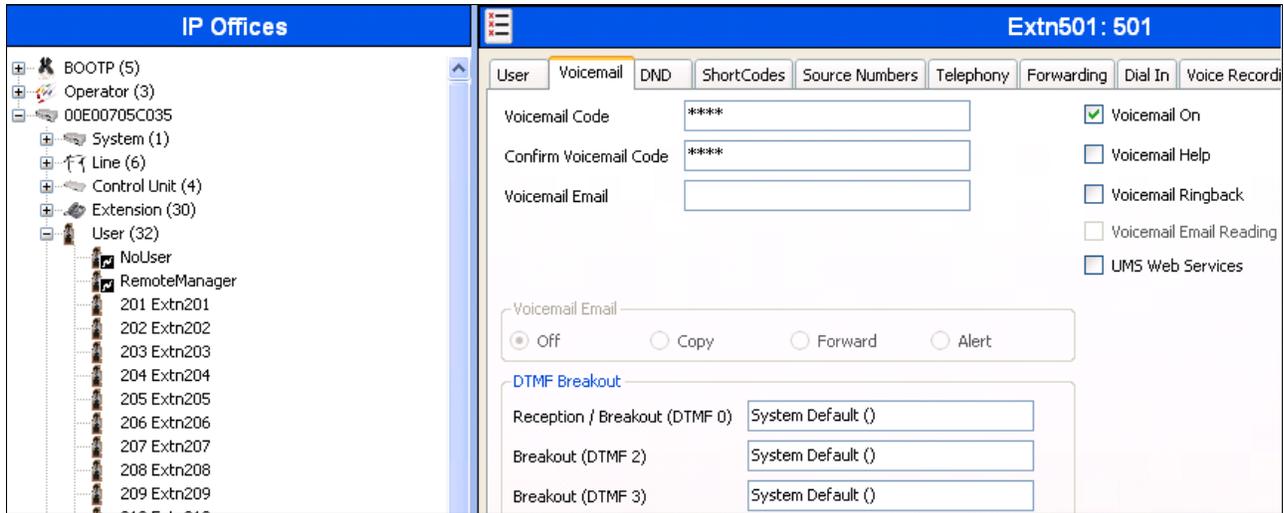
The following screen shows the **User** tab for User **501**. This user corresponds to an Avaya 1616 IP Telephone that is configured as power user with IP Office Softphone features enabled as shown below.

The screenshot displays the Avaya IP Office administration interface. On the left, a tree view under 'IP Offices' shows a hierarchy: BOOTP (5), Operator (3), 00E00705C035, System (1), Line (6), Control Unit (4), Extension (30), and User (32). The 'User' folder is expanded, listing users from 'NoUser' to '501 Extn501'. The '501 Extn501' user is selected and highlighted in blue.

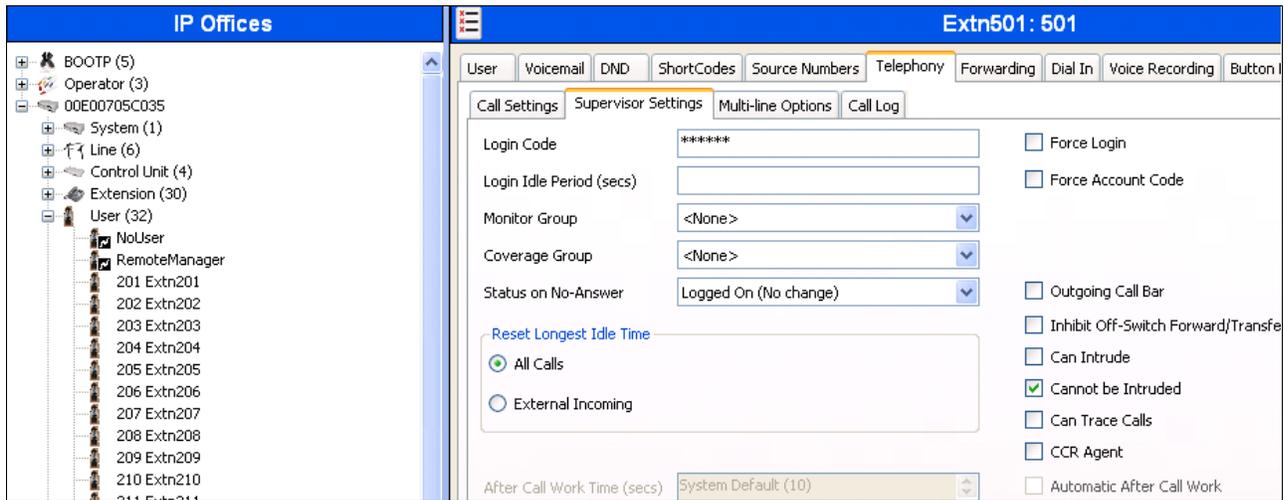
The main panel on the right is titled 'Extn501 : 501' and contains several tabs: 'User', 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', and 'Dial In'. The 'User' tab is active, showing the following configuration details:

- Name: Extn501
- Password: \*\*\*\*\*
- Confirm Password: \*\*\*\*\*
- Full Name: (empty)
- Extension: 501
- Locale: (empty)
- Priority: 5
- System Phone Rights: None
- Profile: Power User
  - Receptionist
  - Enable SoftPhone
  - Enable one-X Portal Services
  - Enable one-X TeleCommuter
  - Ex Directory
- Device Type:  Avaya 1616L
- User Rights
  - User Rights view: User data
  - Working hours time profile: <None>
  - Working hours User Rights: (empty)
  - Out of hours User Rights: (empty)

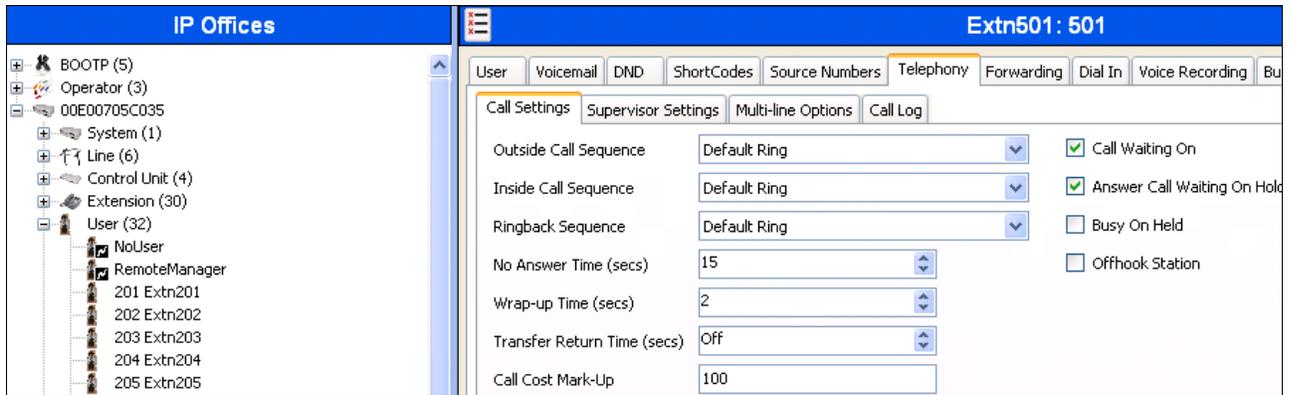
The following screen shows the **Voicemail** tab for this user. The **Voicemail On** box is checked, and a voicemail password can be configured in the **Voicemail Code** and **Confirm Voicemail Code** fields.



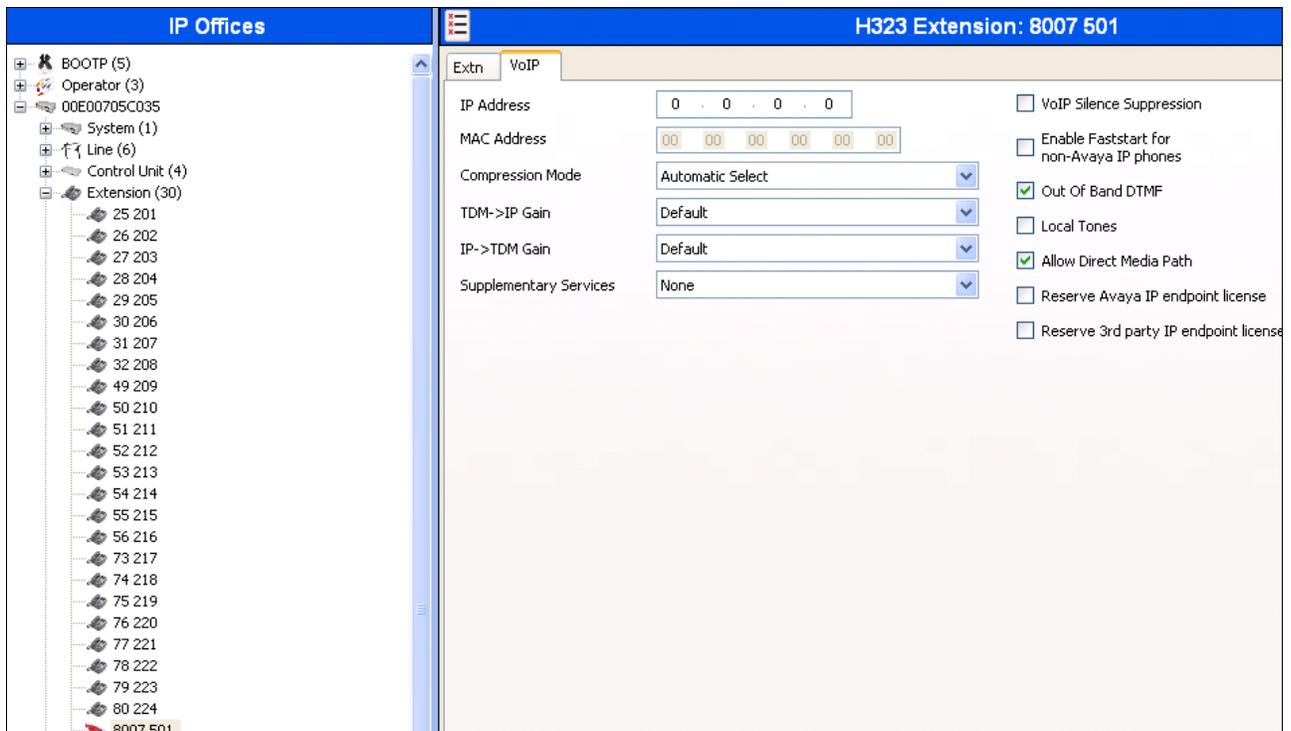
Select the **Telephony** tab and **Supervisor Settings** sub-tab as shown below. To allow hot desking, enter a **Login Code**.



Select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an IP Office Softphone logged in as this extension to have multiple call appearances (necessary for call transfer).



The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for an IP Telephone.



### 5.5.3. SIP Telephone User 520

The following screen shows the **User** tab for User **520**. This user corresponds to an Avaya 1140E SIP Telephone.

IP Offices		Extn520: 520*	
BOOTP (5)		User	
Operator (3)		Voicemail	
00E00705C035		DND	
System (1)		ShortCodes	
Line (6)		Source Numbers	
Control Unit (4)		Telephony	
Extension (30)		Forwarding	
User (32)		Dial In	
NoUser		Voice Recordin	
RemoteManager		Name: Extn520	
201 Extn201		Password: *****	
202 Extn202		Confirm Password: *****	
203 Extn203		Full Name:	
204 Extn204		Extension: 520	
205 Extn205		Locale:	
206 Extn206		Priority: 5	
207 Extn207		System Phone Rights: None	
208 Extn208		Profile: Basic User	
209 Extn209		<input type="checkbox"/> Receptionist	
210 Extn210		<input type="checkbox"/> Enable SoftPhone	
211 Extn211		<input type="checkbox"/> Enable one-X Portal Services	
212 Extn212		<input type="checkbox"/> Enable one-X TeleCommuter	
213 Extn213		<input type="checkbox"/> Ex Directory	
214 Extn214		Device Type: Avaya 1140E Sip	
215 Extn215		User Rights view: User data	
216 Extn216		Working hours time profile: <None>	
217 Extn217		Working hours User Rights:	
218 Extn218		Out of hours User Rights:	
219 Extn219			
220 Extn220			
221 Extn221			
222 Extn222			
223 Extn223			
224 Extn224			
501 Extn501			

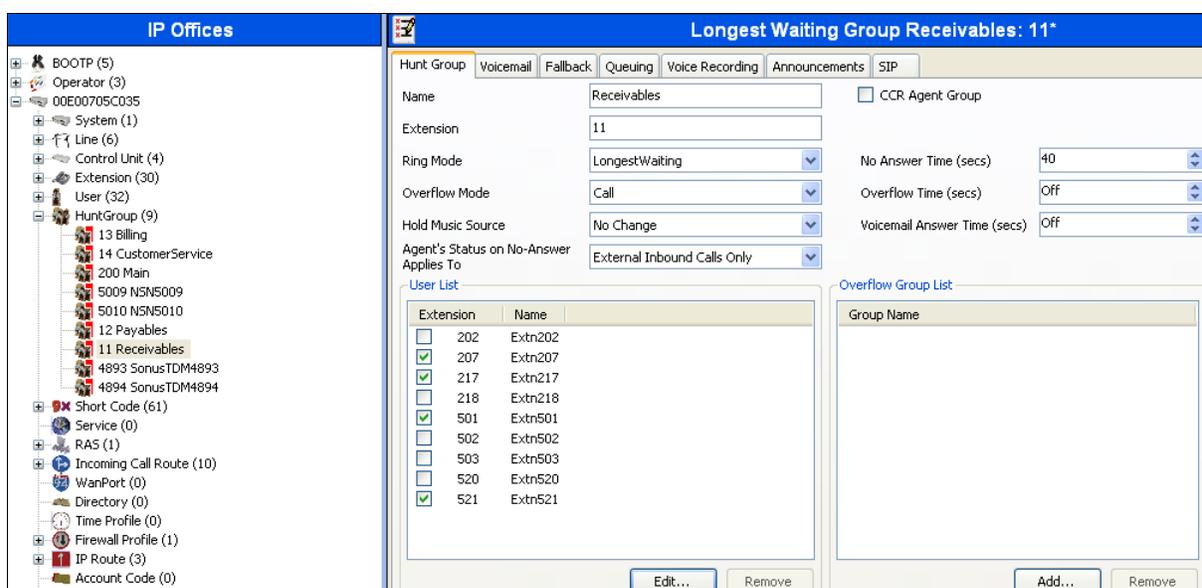
The following screen shows the Extension information for this user. Note that for a SIP telephone, the IP Address configured for the phone needs to be specified. In this example, **10.80.130.51** was assigned to the Avaya 1140E telephone. All other screens are configured the same way as in **Section 5.5.2**.

IP Offices		SIP Extension: 8000 520	
BOOTP (5)		Extn	
Operator (3)		VoIP	
00E00705C035		T38 Fax	
System (1)		IP Address: 10 . 80 . 130 . 51	
Line (6)		Compression Mode: Automatic Select	
Control Unit (4)		Fax Transport Support: None	
Extension (30)		TDM->IP Gain: Default	
25 201		IP->TDM Gain: Default	
26 202		DTMF Support: RFC2833	
27 203		<input type="checkbox"/> VoIP Silence Suppression	
28 204		<input type="checkbox"/> Local Hold Music	
29 205		<input type="checkbox"/> Allow Direct Media Path	
30 206		<input type="checkbox"/> Re-invite Supported	
		<input type="checkbox"/> Use Offerer's Preferred Codec	
		<input checked="" type="checkbox"/> Reserve Avaya IP endpoint license	
		<input type="checkbox"/> Reserve 3rd party IP endpoint license	

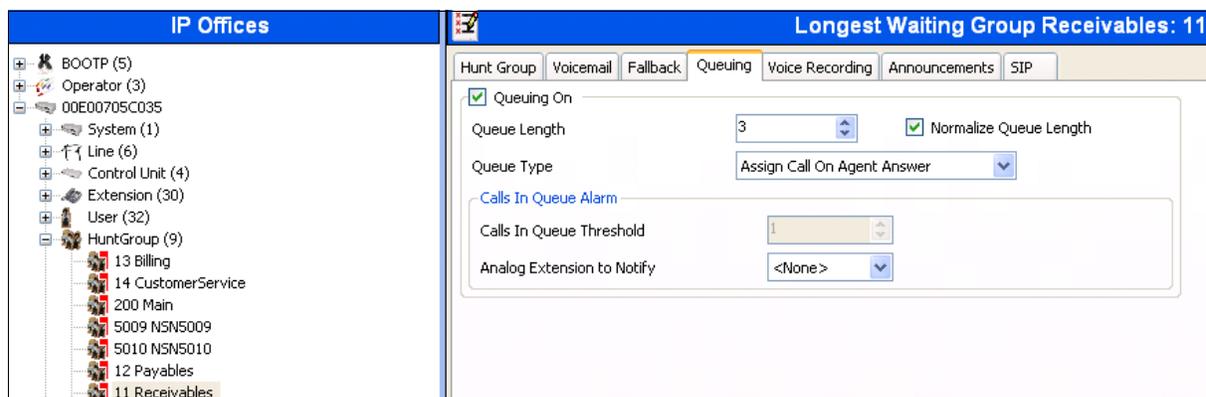
## 5.5.4. Hunt Groups

Hunt groups were used in this reference configuration to route the incoming calls on a SIP Trunk from AT&T Toll Free service to an agent with the right skill set. To configure a new hunt group, right-click **HuntGroup** from the Navigation pane, and select **New** [not shown]. To view or edit an existing hunt group, select **HuntGroup** and choose the appropriate hunt group from the Navigation pane.

The following screen shows the **Hunt Group** tab for hunt group 11. The group name is set to **Receivables**. Several extensions/agents are part of this hunt group. Since the **Ring Mode** field is set to **LongestWaiting**, this will enable the telephones to ring least used extension in the hunt group. Click the **Edit** button to change the **User List**. Once a user is part of a hunt group **User List**, it can be enabled/disabled by checking/unchecking the box in the Extension field.



Under the **Queuing** tab, check the **Queuing On** box and set the **Queue Length** field to any desirable value. Use the default values for all the other fields.



Under the **Announcements** tab, check the **Announcements On** box. The wait time can be set to any desirable value. Make sure that the **Synchronize Calls** box is checked. These announcements are played if an agent for a particular skill is unavailable.

The screenshot displays the IP Office configuration interface. On the left, a tree view shows the hierarchy of IP Offices, including a HuntGroup (9) with sub-groups like Billing, CustomerService, and Payables. The main window is titled 'Longest Waiting Group Receivables: 11\*' and has the 'Announcements' tab selected. The configuration includes:

- Announcements On:** Checked.
- Wait before 1st announcement (seconds):** 60.
- Synchronize Calls:** Checked.
- Flag call as answered:** Unchecked.
- Play 1st announcement:** Indicated by a downward arrow.
- Post announcement tone:** Music on hold.
- 2nd Announcement:** Checked.
- Wait before 2nd announcement (seconds):** 20.
- Play 2nd announcement:** Indicated by a downward arrow.
- Repeat last announcement:** Checked.
- Wait before repeat (seconds):** 20.

A flow diagram on the right side of the configuration window shows the sequence of events: 'Play 1st announcement' leads to 'Music on hold', which then leads to 'Play 2nd announcement'. A feedback loop arrow connects the 'Wait before repeat' field back to the 'Repeat last announcement' field, indicating that the process repeats if the agent remains unavailable.

Similarly, additional hunt groups Billing, Payables and Customer Service are created in this reference configuration to exercise the Call Center functionality within IP Office.

## 5.6. Short Codes

In this section, various examples of IP Office short codes are illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New** [not shown]. To edit an existing short code, click **Short Code** and select the code to be edited in the Navigation pane.

### 5.6.1. Call Center Codes

Call Center functionality is configured on Voicemail Pro. **Section 5.8** lists some of the configuration steps to provide this functionality. In order to access this functionality, short codes can be used. In this reference configuration, **CallCenter** was configured on Voicemail Pro. The following screen shows the short codes set to access this functionality.

The screenshot displays the IP Office configuration interface. On the left, a list of short codes is shown under the heading "IP Offices". The codes are: \*45\*N#, \*46, \*47, \*48, \*49, \*50, \*51, \*52, \*53\*N#, \*57\*N#, \*67, \*70\*N#, \*71\*N#, \*90, \*9000\*, \*91N;, \*92N;, and \*93. The code \*93 is highlighted in blue. On the right, the configuration details for the selected short code \*93 are shown under the heading "\*93: Voicemail". The fields are: Code (\*93), Feature (Voicemail Collect), Telephone Number ("CallCenter"), Line Group Id (0), Locale (empty), and Force Account Code (unchecked).

### 5.6.2. Voicemail Retrieval Code

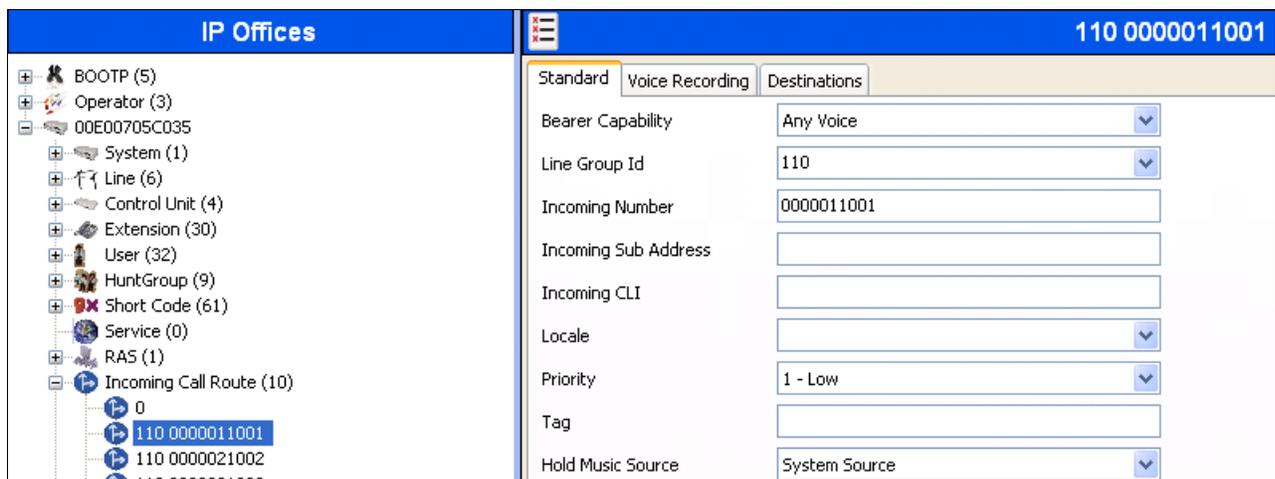
To retrieve voicemails left in individual mailboxes, this code was configured in this reference configuration. When a user enters, \*17, they can retrieve the messages in their mailbox.

The screenshot displays the IP Office configuration interface for the short code \*17. The heading is "\*17: Voicemail Collect". The fields are: Code (\*17), Feature (Voicemail Collect), Telephone Number (?U), Line Group Id (0), Locale (empty), and Force Account Code (unchecked).

## 5.7. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a specific AT&T IP Toll Free DNIS to a destination user, group, or function on IP Office. To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New** [not shown]. To edit an existing incoming call route, select **Incoming Call Route** and the appropriate route in the Navigation pane.

The screen shown below matches the AT&T IP Toll Free DID **0000011001** in the **Incoming Number** field on the **Line Group Id (110)**. The **Line Group Id** matches the **Incoming Group** field and **Incoming Number** matches the **Local URI** field configured in the **SIP URI** tab for the SIP Line to AT&T IP Toll Free service in **Section 5.4.3**.



IP Offices		110 0000011001			
Standard		Voice Recording		Destinations	
Bearer Capability	Any Voice				
Line Group Id	110				
Incoming Number	0000011001				
Incoming Sub Address					
Incoming CLI					
Locale					
Priority	1 - Low				
Tag					
Hold Music Source	System Source				

Select the **Destinations** tab and a value can be either selected from the drop-down list or manually entered. In the screen shown below, the hunt group configured in **Section 5.5.4** was selected.



110 0000011001*								
Standard			Voice Recording			Destinations		
	TimeProfile	Destination		Fallback Extension				
	Default Value	11 Receivables						

Similarly, in the screen below, an extension configured in **Section 5.5.2** was selected.

TimeProfile	Destination	Fallback Extension
Default Value	502 Extn502	

The following screens displays how a short code can be manually assigned in the **Destination** field to route the call to access Call Center functionality by entering a short code configured in **Section 5.6.1**.

TimeProfile	Destination	Fallback Extension
Default Value	*93	

The following screen displays another mechanism to access the Call Center functionality without using the short code. The Call Center functionality is configured in Voicemail Pro as detailed in **Section 5.8**.

TimeProfile	Destination	Fallback Extension
Default Value	VM:CallCenter	

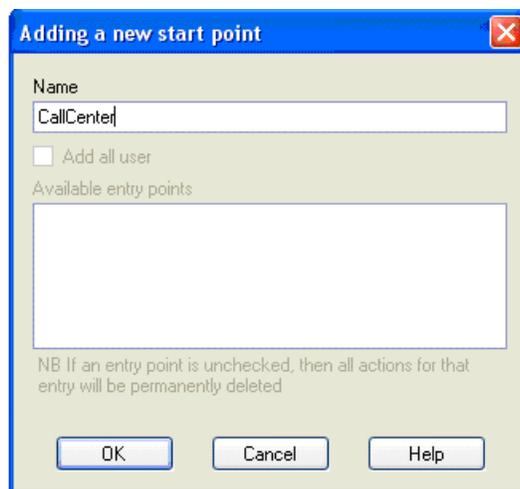
## 5.8. Call Center Provisioning in Voicemail Pro

The call center functionality was configured in Voicemail Pro. Following steps highlight the configuration of this functionality. For further information, consult [IPO-VMPRO].

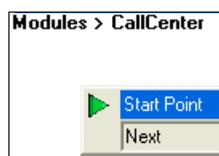
1. Navigate to **Start**→**Voicemail Pro Client** and right click on modules and select **Add** to add a new module.



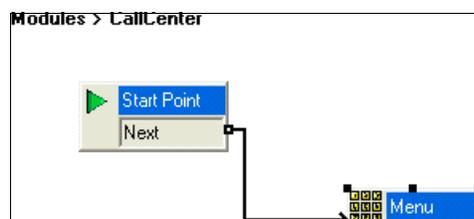
2. In the screen below, enter **CallCenter** in the **Name** field and click OK.



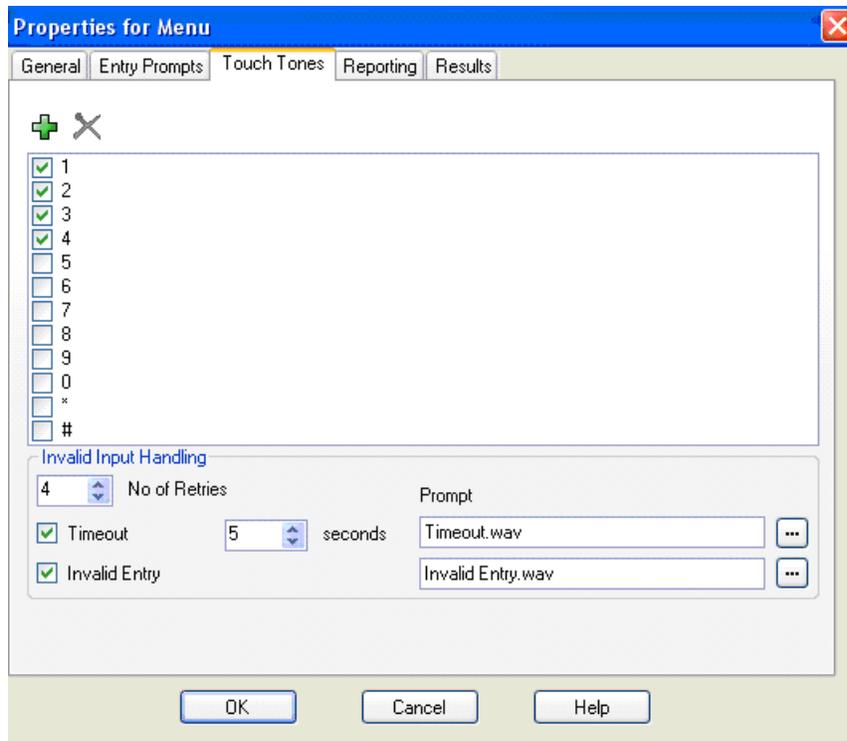
3. Following screen is displayed indicating the starting point for the Call Center functionality.



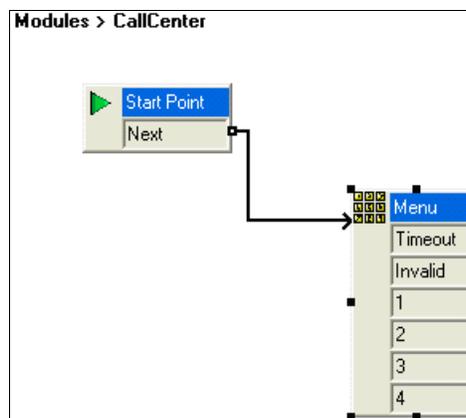
4. Under the **Actions** tab, select **Basic Actions** [not shown]. Select **Menu** and place it on the right side of the pane and then connect the **Start Point** to **Menu** as shown below:



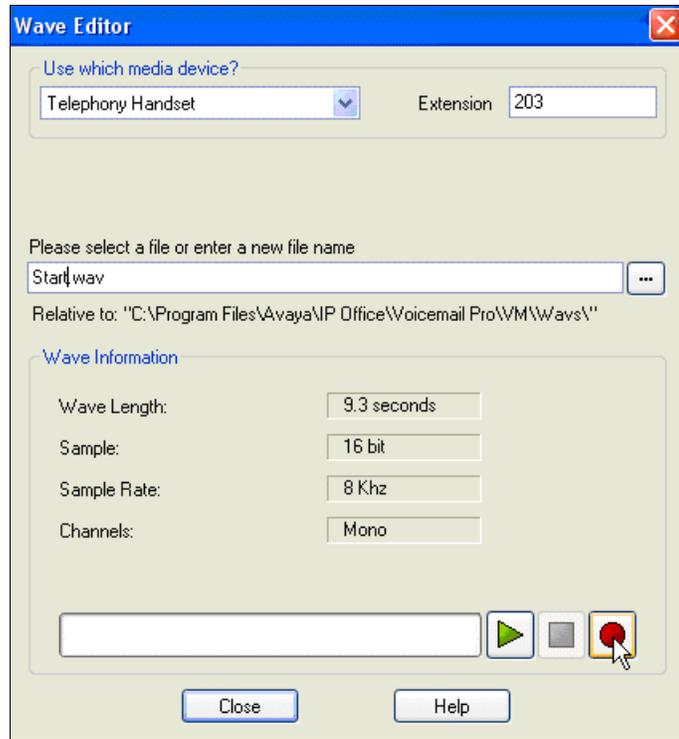
- Right click on **Menu** and select **Touch Tones** tab. Check the appropriate boxes. In this reference configuration, 1, 2, 3, 4, Timeout and Invalid boxes were selected. This allows caller to enter any of the digits from 1 to 4 to go to the appropriate agent. Digit **1** was used for **Receivables**, Digit **2** was used for **Payables**, Digit **3** was used for **Billing** and Digit **4** was used for **Customer Service** Hunt Groups/Skills in this reference configuration. Digits have to be entered within a certain time and within the specified range otherwise an error recording may be played. Enter any valid number in the **No. of Retries** field. This field dictates the number of retries allowed to the caller for entering a digit.



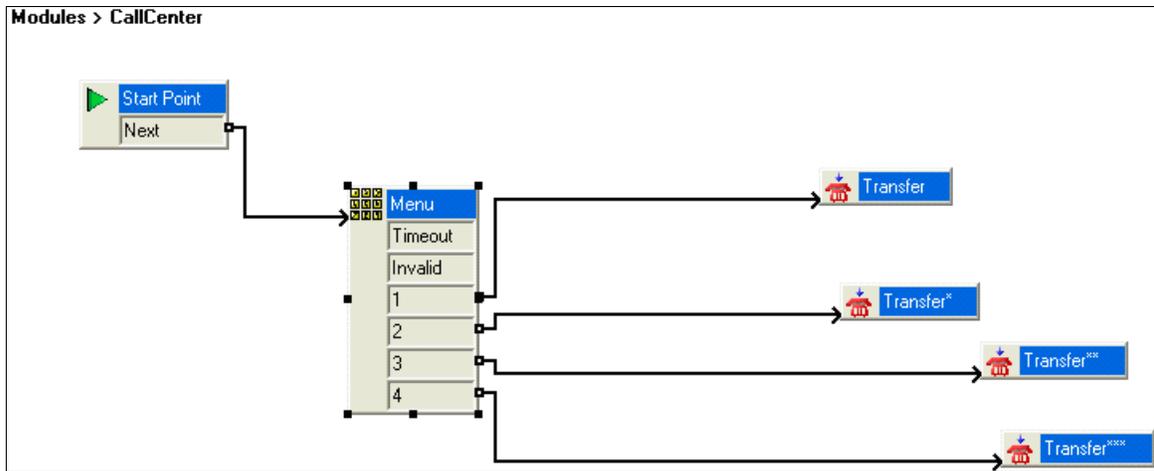
Click **OK** and following screen is displayed:



- Right click on the **Menu** action and select the **Entry Prompts** tab and click on the **+** sign [not shown]. On the following screen, enter the **Extension** where the recording is done and the filename for the recording in the **Please select a file or enter a new file name** field and press the Red record button as shown. In this reference configuration the phone at extension **203** rings and Voicemail Pro prompts the user to record an announcement which is played back to the caller when the call comes into IP Office.



- Under the **Actions** button, select **Transfer** [not shown]. Repeat this step for additional actions. In this reference configuration four **Transfer** actions were created for each of the selections in **Step 5** and connected to them.

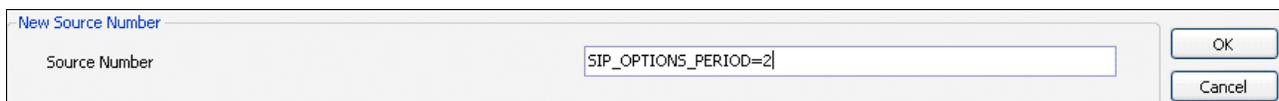


Right click on the **Transfer** action and select the Specifics tab. In the **Destination** field enter the hunt group/skill number created in **Section 5.5.4** and click **OK**. This will enable the call to be routed to the appropriate skill. Repeat this step for all the **Transfer** actions.

## 5.9. SIP OPTIONS Frequency

From the Navigation pane, select **User** and then select the user named **NoUser**. In the **NoUser** Details pane, select the tab **Source Numbers**. Press the **Add...** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field shown below, type **SIP\_USE\_PAI\_FOR\_PRIVACY**. Click **OK**.

Similarly, SIP OPTIONS frequency was configured by adding another source number by entering **SIP\_OPTIONS\_PERIOD=2** in the **Source Number** field. This will set the frequency of the SIP OPTIONS message sent by IP Office to 2 minutes.



New Source Number

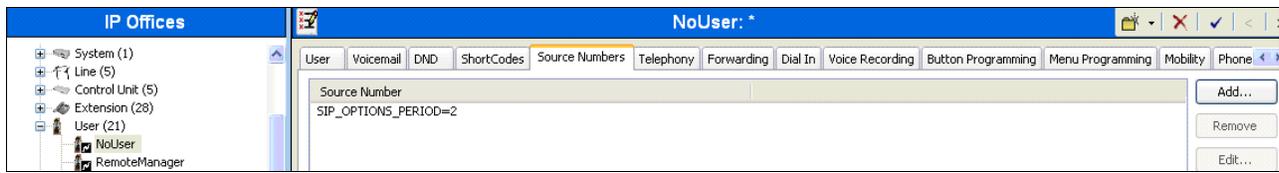
Source Number

SIP\_OPTIONS\_PERIOD=2

OK

Cancel

The following screen displays the Source Numbers configured in this reference configuration.



IP Offices

NoUser: \*

User Voicemail DND ShortCodes Source Numbers Telephony Forwarding Dial In Voice Recording Button Programming Menu Programming Mobility Phone

Source Number

SIP\_OPTIONS\_PERIOD=2

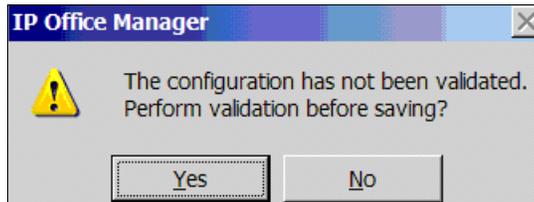
Add...

Remove

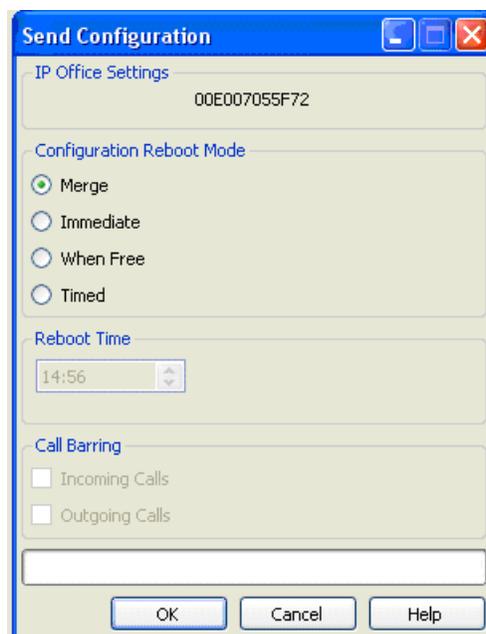
Edit...

## 5.10. Saving Configuration Changes to IP Office

When desired, send the configuration changes made in IP Office Manager to the IP Office server, to cause the changes to take effect. Click the “disk” icon that is the third icon from the left (i.e., common “save” icon with mouse-over help “Save Configuration File”). Click **Yes** to validate the configuration, if prompted.



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



## 6. Verification Steps

The following steps may be used to verify the configuration:

### 6.1. General

- Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.
- Place an inbound call to an agent or phone, but do not answer the call. Verify that the call covers to Voicemail Pro and messages can be retrieved using the appropriate short codes.
- Use the **System Status** application to verify the status of trunks, extensions and call progress.
- Use the **Monitor** application to monitor the activity on IP Office.

## 7. Conclusion

As illustrated in these Application Notes, Avaya IP Office can be configured to interoperate successfully with the AT&T IP Toll Free service. This solution provides users of Avaya IP Office the ability to support inbound toll free calls over an AT&T IP Toll Free SIP trunk service connection via MIS/PNT transport.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and is intended to provide **configuration guidance** to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program.

## 8. References

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

[IPO-INSTALL] IP Office 7.0 Installation Manual, Issue 23k, May 22, 2011  
Document Number 15-601042

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

[IPO-MGR] IP Office Release 7.0 Manager 9.0 Issue 26h, May 22, 2011  
Document Number 15-601011

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

[IPO-SYSSTAT] IP Office Release 6.0 System Status Application, Issue 05a, February 12, 2010  
Document Number 15-601758

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

[IPO-VMPRO] IP Office Release 7.0 Voicemail Pro Administration, Issue 26a, May 1, 2011  
Document Number 15-601063

<https://support.avaya.com/css/P8/documents/100129332>

[IPO-MON] IP Office System Monitor, Issue 02b, November 28, 2008  
Document Number 15-601019

<http://support.avaya.com/css/P8/documents/100073350>

Additional IP Office documentation can be found at:

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

AT&T IP Toll Free Service Descriptions:

[1] *AT&T IP Toll Free*

<http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/>

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