Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office 8.1 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office 8.1 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** or **AVPN** 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. Avaya IP Office 8.1 is a telephony application server and is the point of connection between the enterprise endpoints and AT&T IP Toll Free service.

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 8.1 with the AT&T IP Toll Free service. The IP Office solution was tested with the AT&T IP Toll Free service using **MIS/PNT** or **AVPN** 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.

Note: ***Testing was performed with IP Office 500 v2 R8.1, but it also applies to IP Office Server Edition R8.1 (single site configuration only)***.

2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise site 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 site was connected via MIS/PNT or AVPN 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 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 for accessing/navigating automated voice 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.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for examples) between 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. IP Office supports G.711 faxing only for inbound calls and therefore it works with AT&T IP Toll Free service.
2. Shuffling is not supported for SIP trunks in IP Office V2 R8.1 but the IP Office Server Edition R8.1 supports shuffling.
3. G.726 codec is not supported by IP Office 8.1.
4. Alternate Destination Routing – Ring No Answer is not supported by AT&T IP Toll Free service.
5. AT&T IP Transfer Connect option of the AT&T IP Toll Free service was not verified with IP Office 8.1 and hence not supported.

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 1616, 9641G 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.

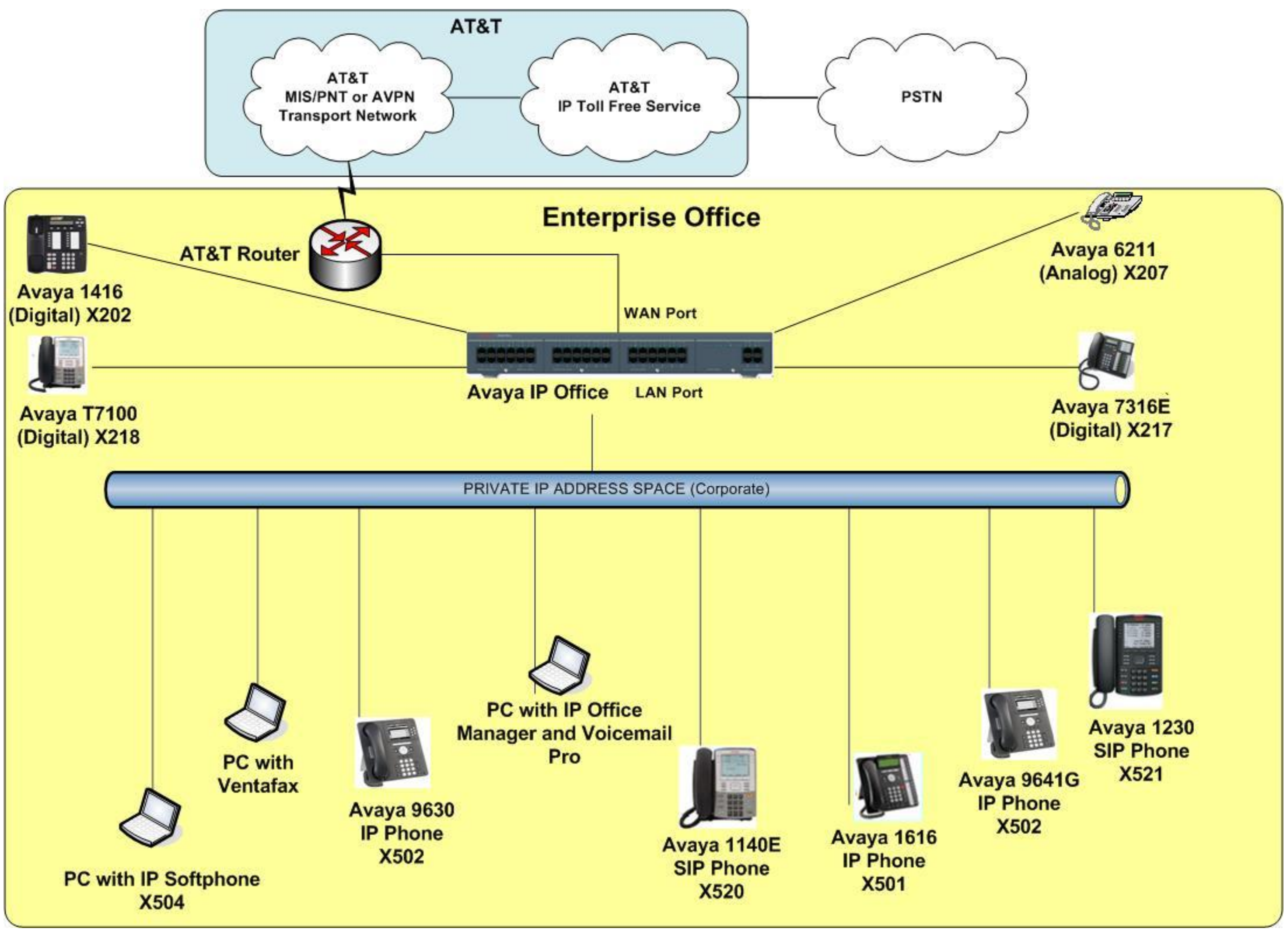


Figure 1: Reference configuration

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
Avaya IP Office	
Public IP Address	192.168.62.50
Private IP Address	10.80.130.58
Avaya IP Office Extensions	207 = Analog 501,502=H323 202,217,218=Digital 504=Softphone 520,521= SIP phones
AT&T IP Toll Free Service	
Border Element IP Address	135.242.225.210
Digits passed in SIP-URI Request	000004153571057 – CPN Basic 000004153581058 - CPN Restricted 000004153591059 – Legacy Xfer Connect 000004153501060 – ADR 000004153511061 – ADR Secondary

Table 1: Illustrative Values Used in this Reference Configuration

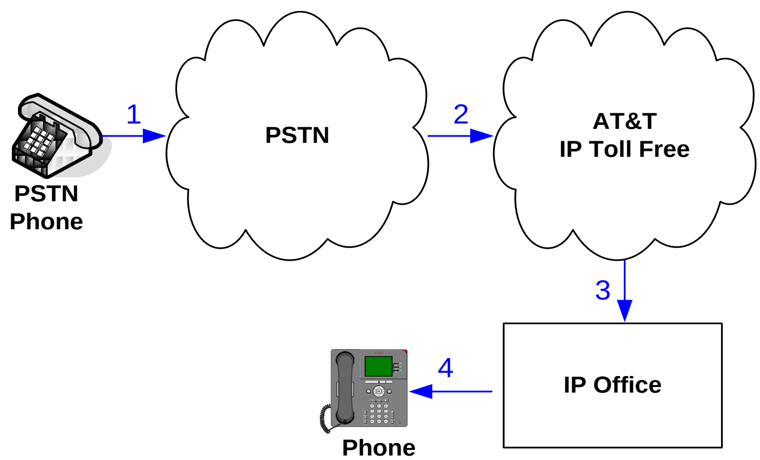
3.2. Call Flows

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 arrives at IP Office, which in turn routes the call to a hunt group, phone or a fax.

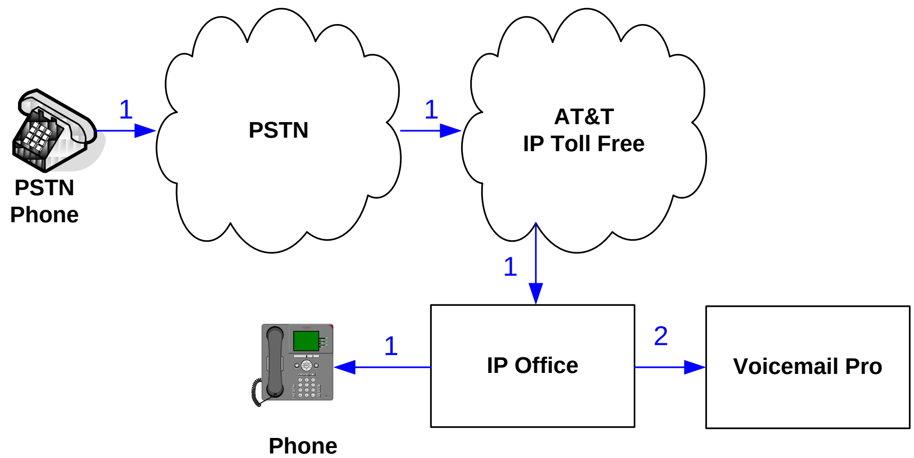
1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to IP Office.
4. Depending on the called number, IP Office routes the call to
    - A hunt group, which in turn,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



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.



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.

Equipment/Software	Release/Version
Avaya IP Office 500 V2	Release 8.1 (52) (Preferred Edition)
Avaya IP Office Manager	Release 10.0 (52) (Preferred Edition)
Avaya IP Office Voicemail Pro	Release 8.1 (1003.0)
Avaya IP Office Voicemail Pro Client	Version 8.1 (1003.0)
Avaya 1616IP-Series Telephones (H.323)	Release 3.2
Avaya 9641G IP Telephone	R6.2.2.09U
Avaya IP Office Softphone	Release 3.2.3.15 64595
Avaya 1416 Digital Telephone	-
Avaya T7100 Digital Phone	-
Avaya 7316E Digital Phone	-
Avaya 6211 Analog phone	-
Avaya 1140E SIP Telephone	04.03.12.00 (SIP1140)
Avaya 1230 SIP Telephone	04.03.12.00 (SIP1230)
Fax device	Ventafax Home Version 6.2
AT&T IP Toll Free Service using MIS/PNT or AVPN transport service connections	VNI 23

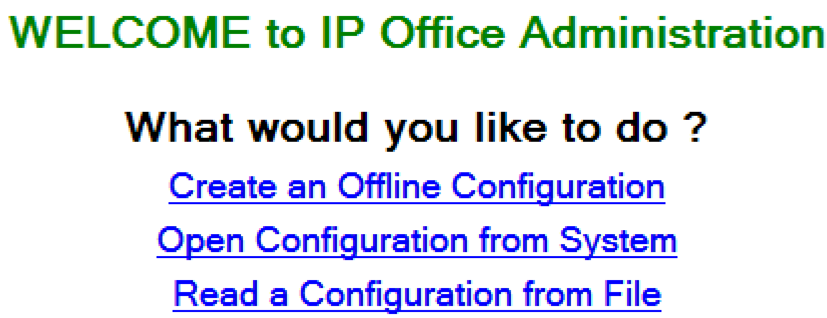
Table 2: Equipment and Software Versions

Testing was performed with IP Office 500 V2 R8.1, but it also applies to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R8.1 to support analog or digital endpoints or trunks.

5. Avaya IP Office

This section describes specific settings of the reference configuration, but is not meant to be prescriptive. The configuration steps described here are only for specific fields where a value was changed. For all the other fields default values were 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, navigate to **Start→Programs →IP Office →Manager** to launch the Manager application. A screen that includes the following in the center may be displayed:



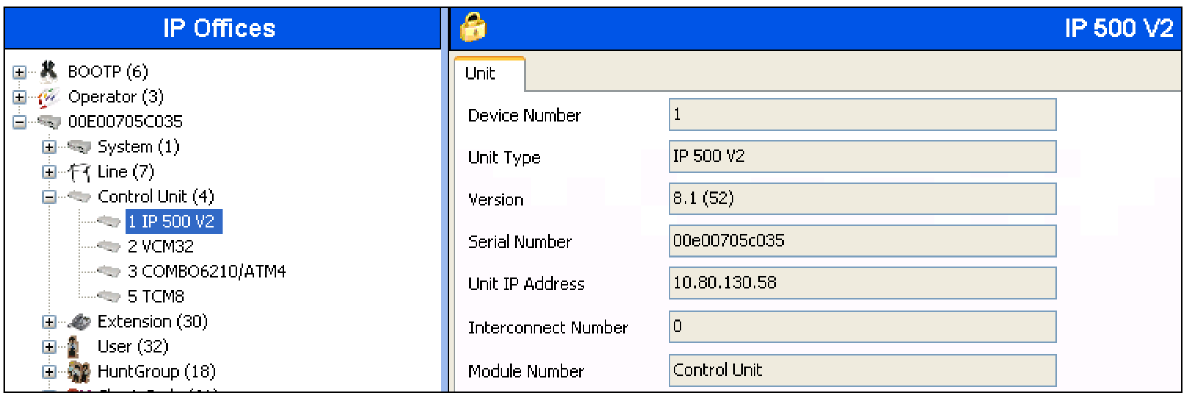
Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, and the Details pane on the right side.

The configuration steps shown in these application notes are not presciptive in nature; they only demonstrate a way to perform this configuration. Configuration is done only for the field values required for this testing. Default values were used for all the other fields.

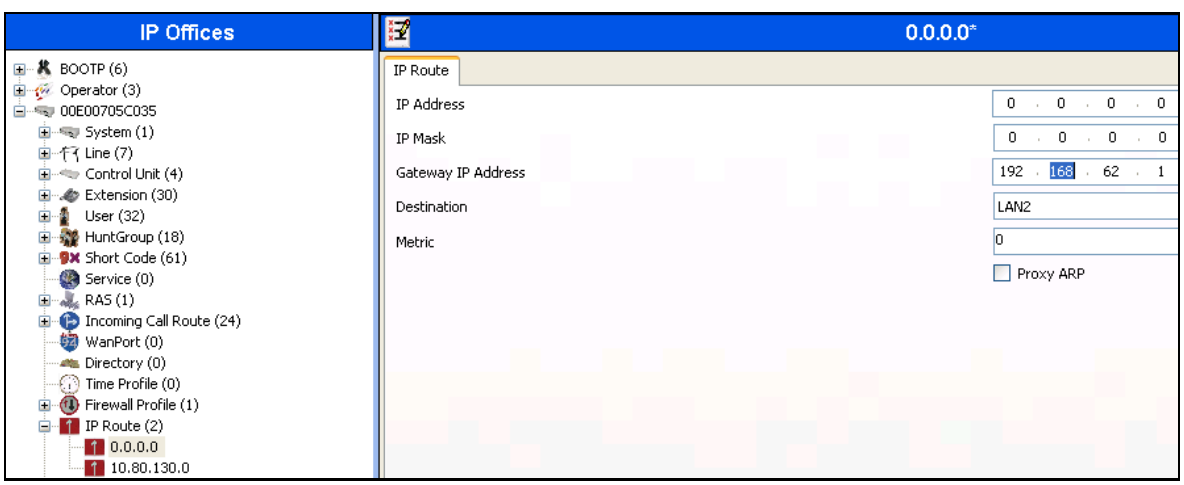
5.1. Physical, Network, and Security Configuration

In the reference configuration, the IP Office 500 V2 contains a VCM32 module, a 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.

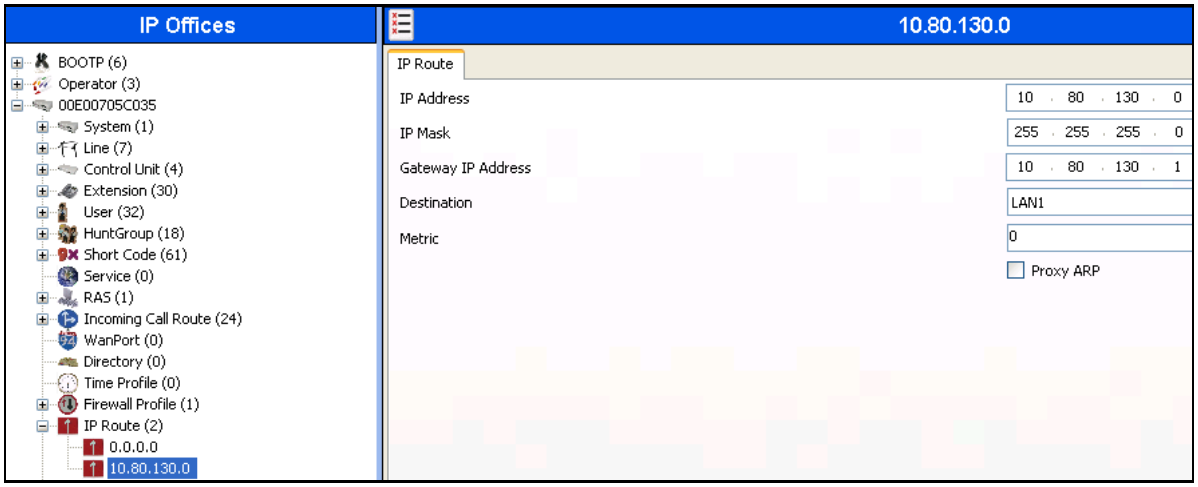
1. The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select a module under **Control Unit** in the Navigation pane. The module information appears in the Details pane. In the screen below, **IP 500 V2** is selected in the Navigation pane,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



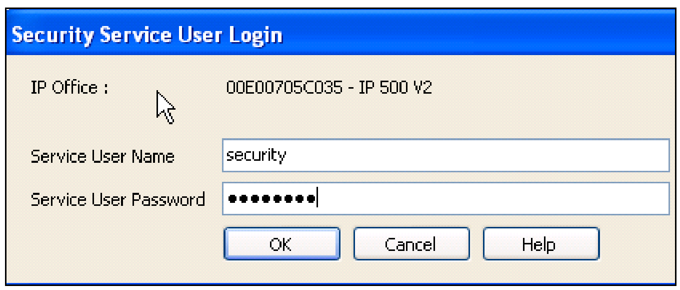
2. In this reference configuration, the IP Office **LAN2** port (labeled as WAN port in Figure 1) is physically connected to the public network at the IP Office customer site. The default gateway for this network is **192.168.62.1**. To add an IP Route in IP Office, right-click **IP Route** from the 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**).



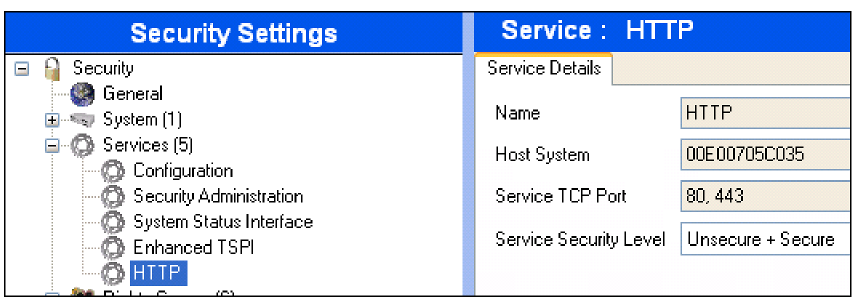
3. Another route for **10.80.130.0** subnet was added for the enterprise side **LAN1** port (labeled as LAN port in Figure 1) as shown in the screen below. All the enterprise IP devices were part of this **10.80.130.x** network in this reference configuration.



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



5. 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.

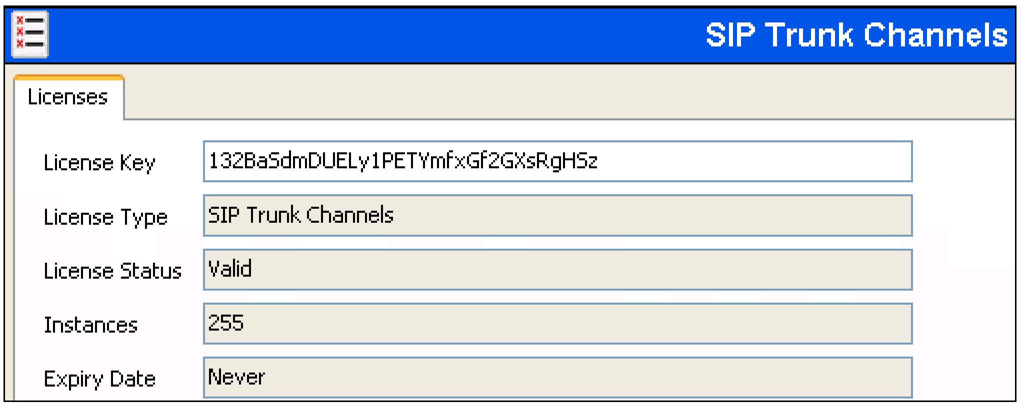


6. When complete, navigate to **File → Configuration** to return to configuration activities.

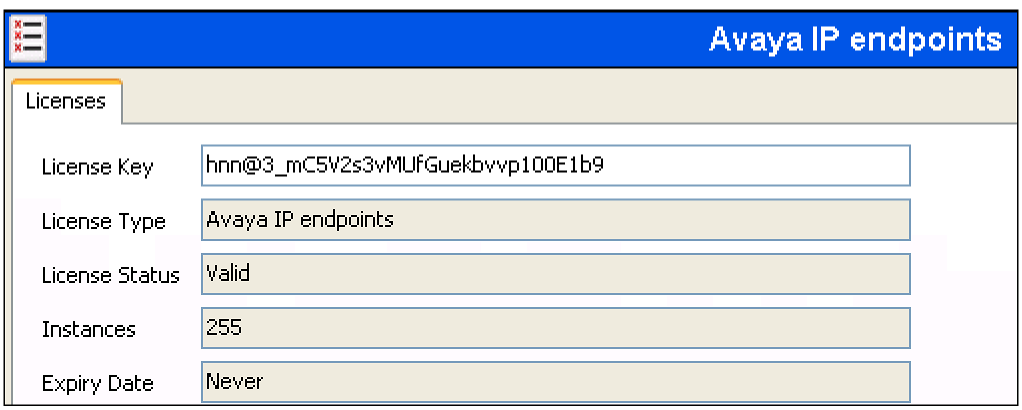
5.2. Licensing

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.

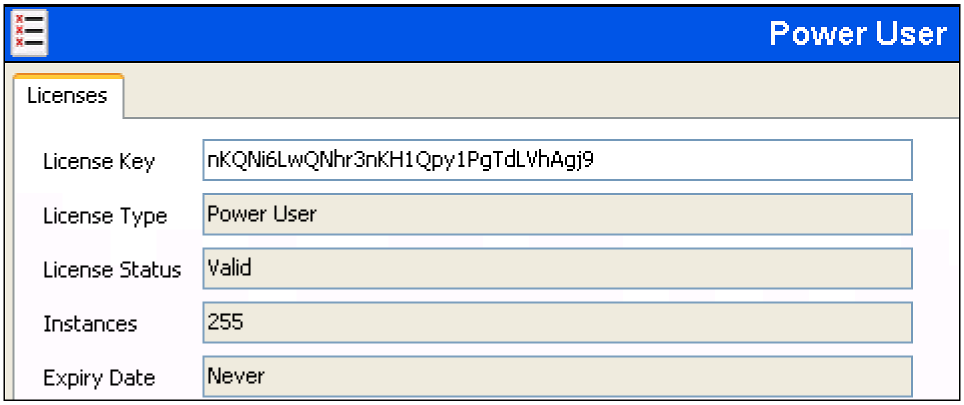
1. To verify that SIP Trunk Channels has sufficient capacity, navigate to **License → SIP Trunk Channels** in the Navigation pane and confirm a valid license with sufficient **Instances** (trunk channels) exist in the Details pane.



2. To verify Avaya IP endpoints have 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.



3. 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.

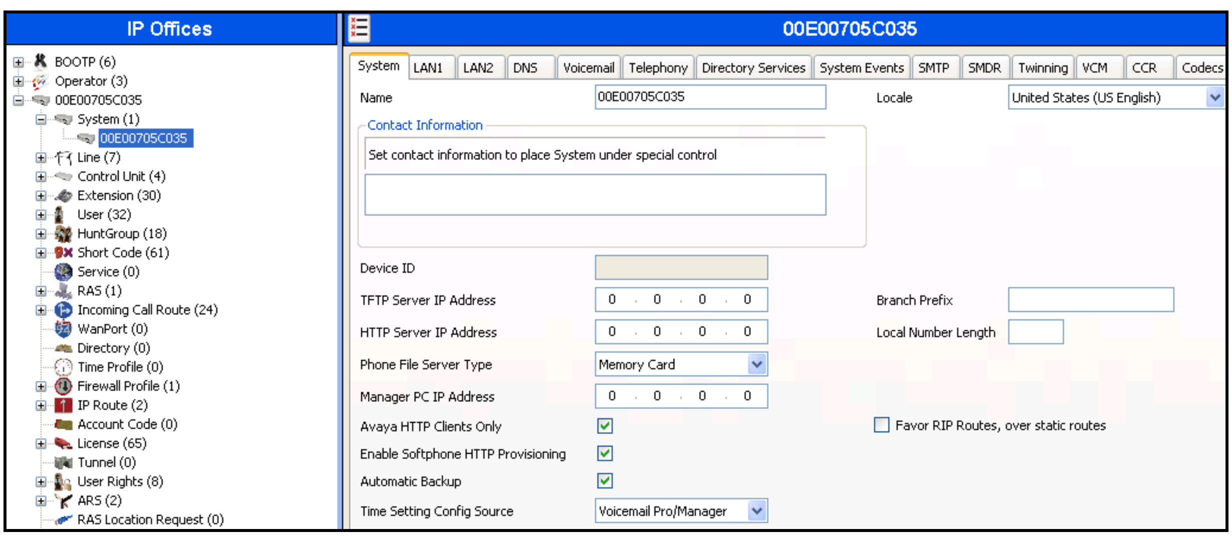


5.3. System Settings

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 as shown below, 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.

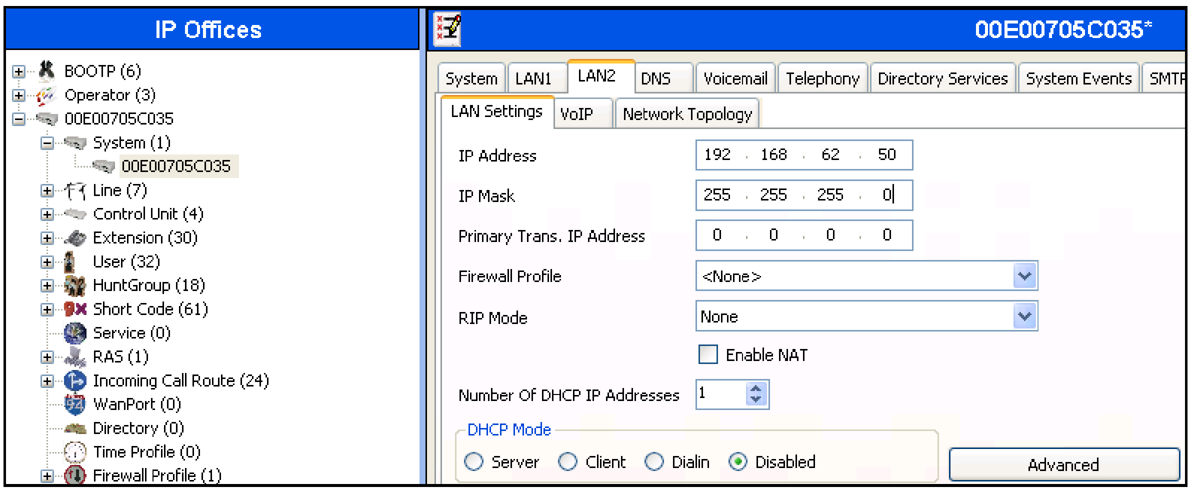


5.3.2. LAN Settings

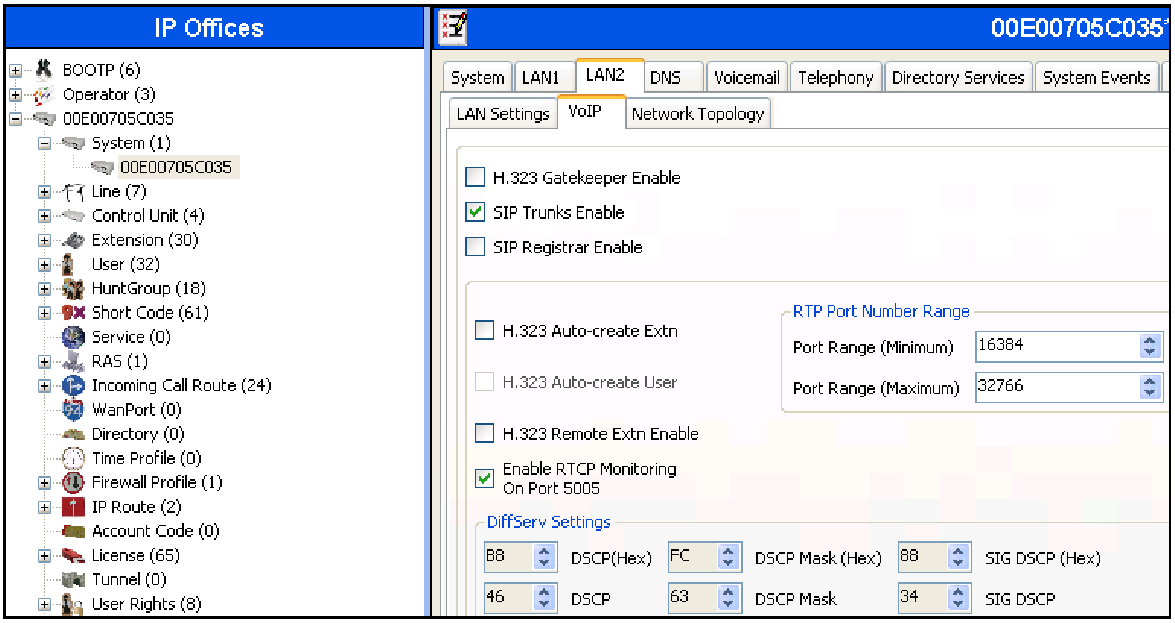
In the sample configuration, **LAN2** was used to connect the IP Office to the AT&T Network and **LAN1** was used to connect to the enterprise network.

1. Navigate to **LAN2 → LAN Settings** and configure as follows:

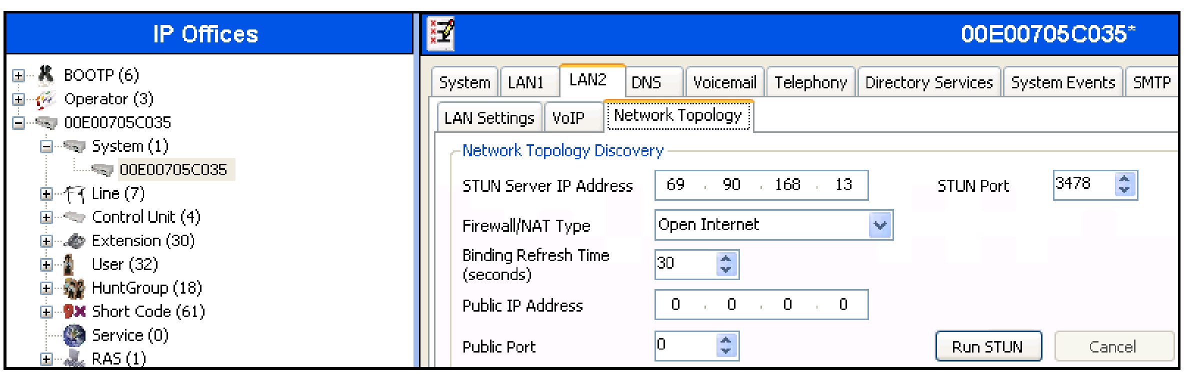
- **IP Address** – Set to **192.168.62.50** which is the IP address of IP Office known to AT&T network
- **IP Mask** – Set to a valid value e.g **255.255.255.0**
- **Primary Trans**. **IP Address** – Set to **0.0.0.0**
- **DHCP Mode** – The **Disabled** radio button was selected in this reference configuration



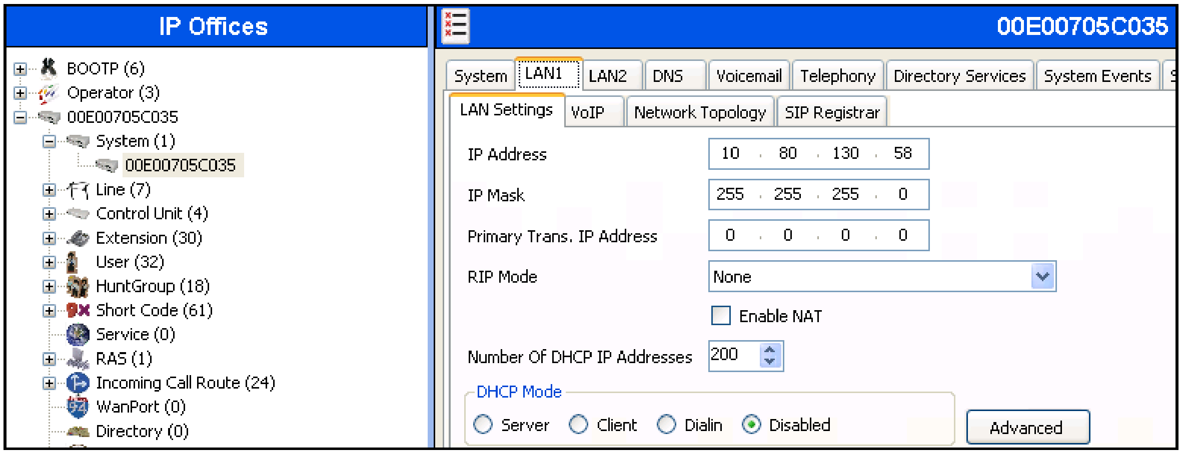
2. Select the **VoIP** tab as shown in the following screen and configure as follows:
   - **SIP Trunks Enable** – Check this box to enable the configuration of SIP trunks
   - **RTP Port Range (Minimum)** – Set to **16384** (As required by AT&T)
   - **RTP Port Range (Maximum)** – Set to **32766** (As 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 otherwise it sets the port range to its default value.



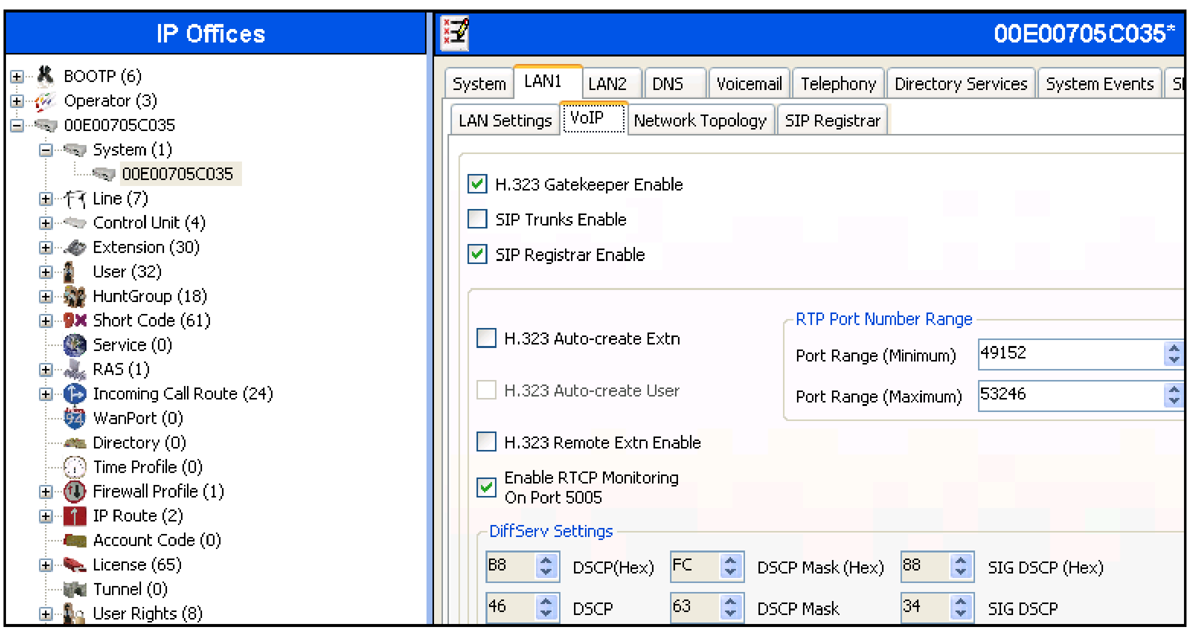
3. Select the **Network Topology** tab as shown in the following screen and set **Firewall/NAT Type** field to **Open Internet**. With this configuration, STUN will not be used but make sure to leave **STUN Server IP Address** to its default value.



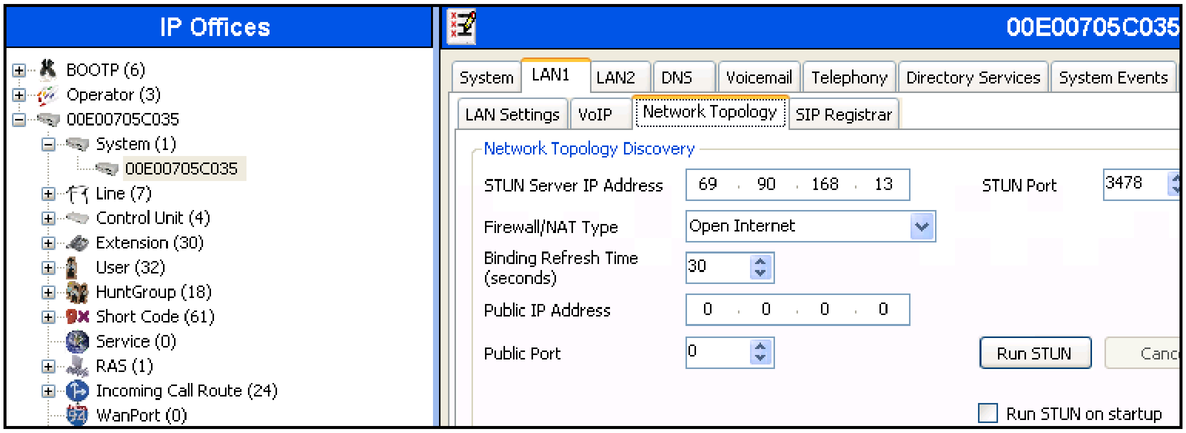
4. Navigate to the **LAN1→ LAN Settings** and configure as follows:
   - **IP Address** – Set to **10.80.130.58**, the IP address of the enterprise side connected to IP Office
   - **IP Mask** – Set to **255.255.255.0**
   - **DHCP Mode** - Set to **Disabled** in this reference configuration



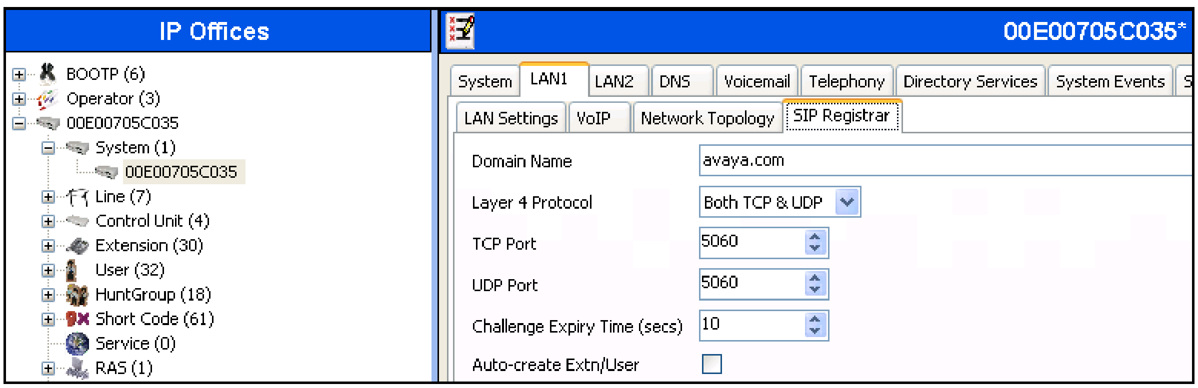
5. Select the **VoIP** tab as shown in the following screen and configure as follows:
   - **H323 Gatekeeper Enable** – Check this box to allow the use of Avaya IP Phones
   - **SIP Registrar Enable** – Check this box to allow SIP phones and IP Office Softphone usage



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.

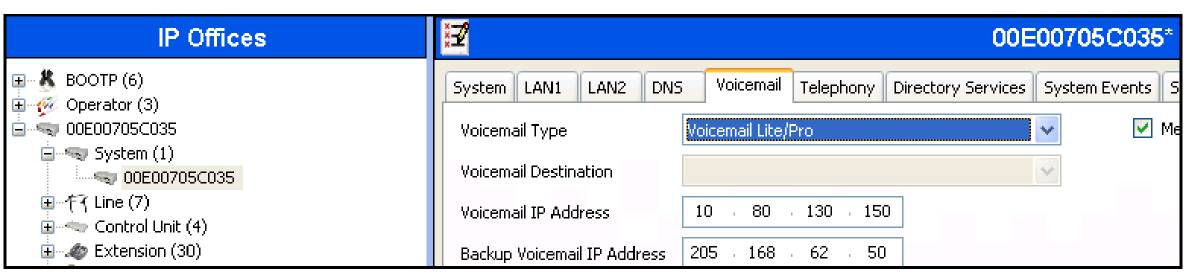


8. Click **OK** [not shown] to commit.

5.3.3. Voicemail

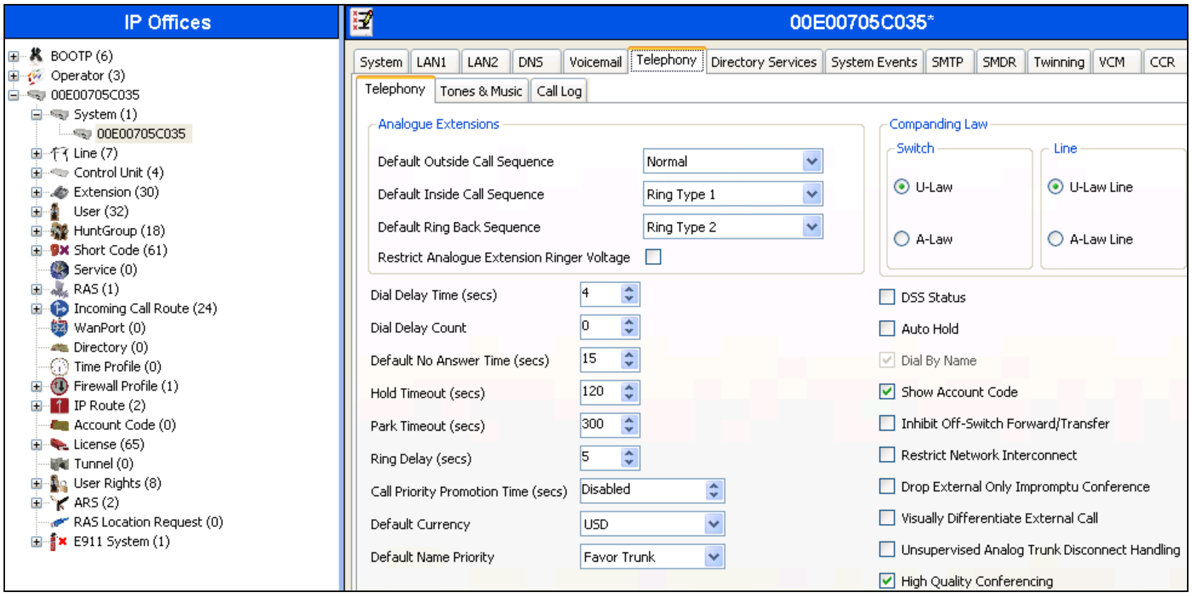
Select **Voicemail** tab and configure as follows:

- **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.



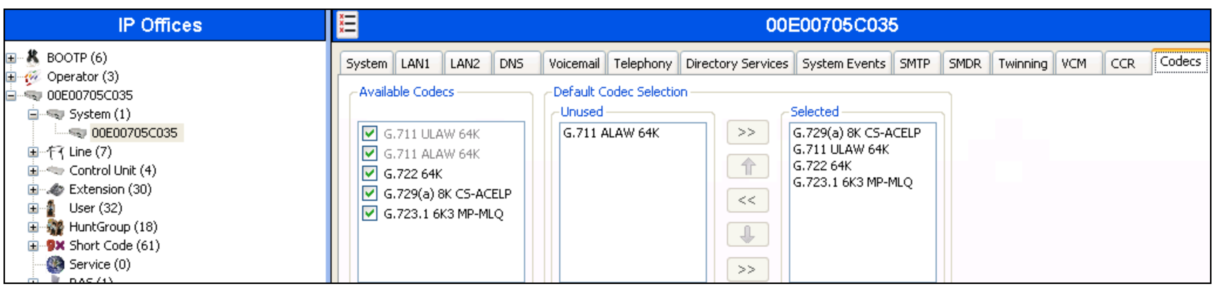
5.3.4. System Telephony Configuration

Navigate to **Telephony → Telephony** and check **ULAW** box under **Switch** in **Companding Law** and **ULAW Line** box under the **Line** in **Companding Law**.



5.3.5. Codecs

Select the **Codecs** tab and set the order as shown in the **Selected** box.



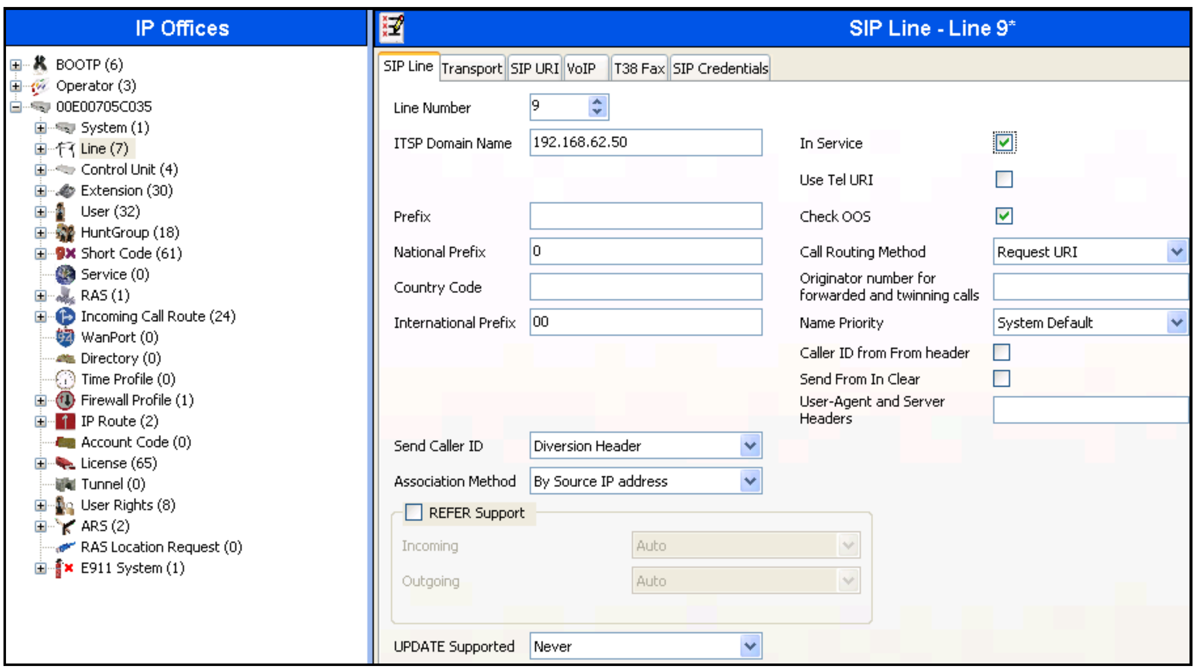
5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office. 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

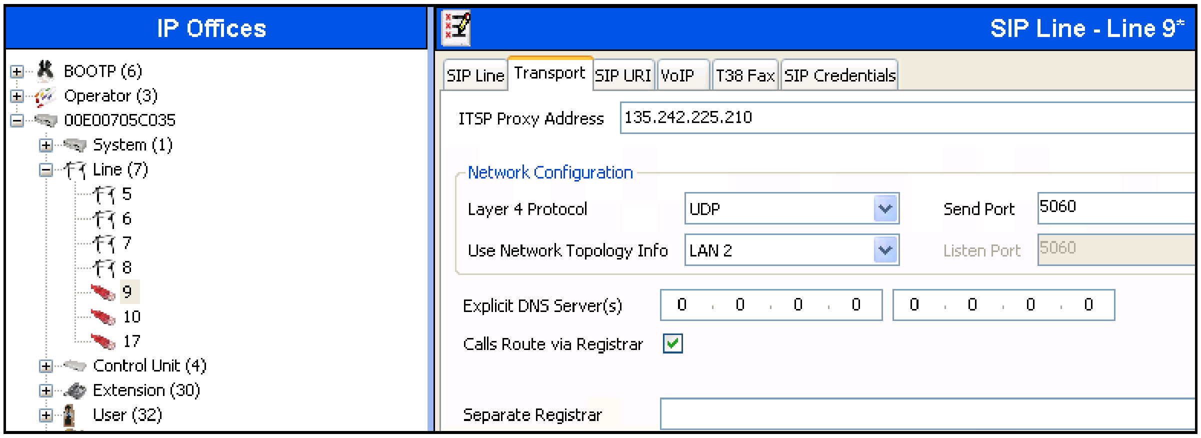
Select **SIP Line** tab as shown below for Line Number **9** used for AT&T and configure as follows:

- **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 this 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



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.

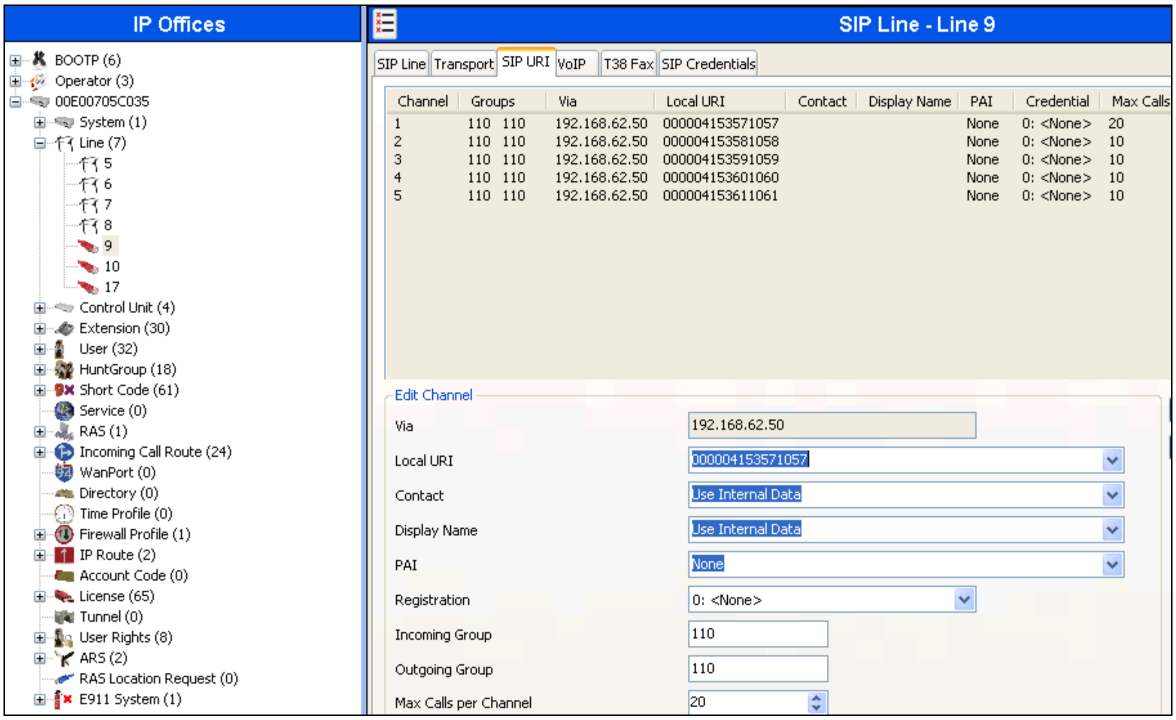


5.4.3. SIP Line - SIP URI Tab

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:

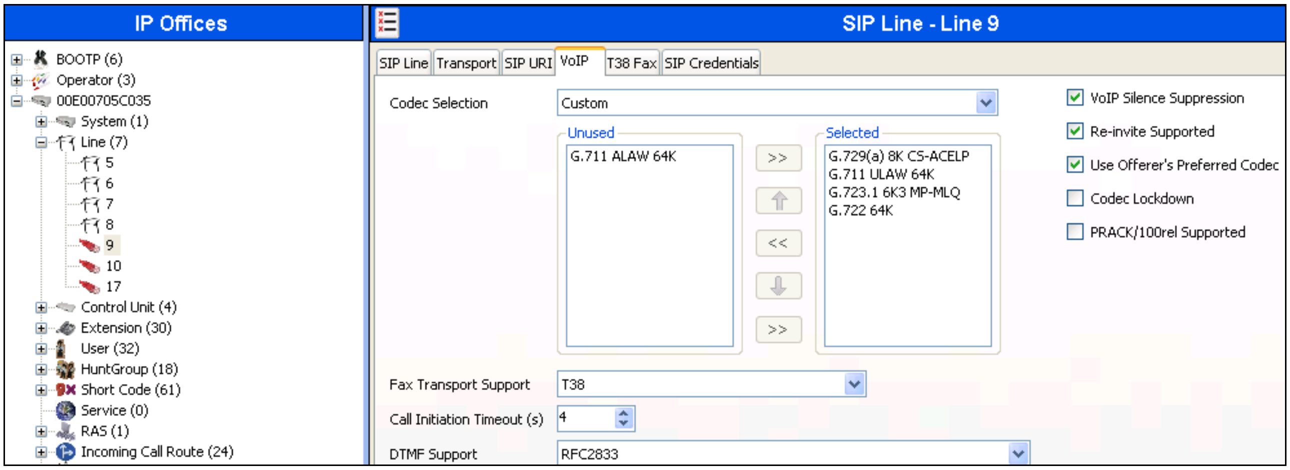
- **Local URI** – Set to the DNIS sent by AT&T IP Toll Free service in the SIP URI. In this example it is set to **000004153571057** which is one of the DNIS mentioned in **Table 1**.
- **Registration** - Set to **0: <None>**
- **Incoming Group** and **Outgoing Group** – Set to **110**

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



5.4.4. SIP Line - VoIP Tab

Select the **VoIP** tab. In this reference configuration the **Codec Selection** was set to **System Default** which indicates that it will use the same selection and order set in **Section 5.3.5**. This order can be changed at the trunk level if so desired. In this reference configuration **Fax Transport Support** was tested both with T.38 and G.711MU.



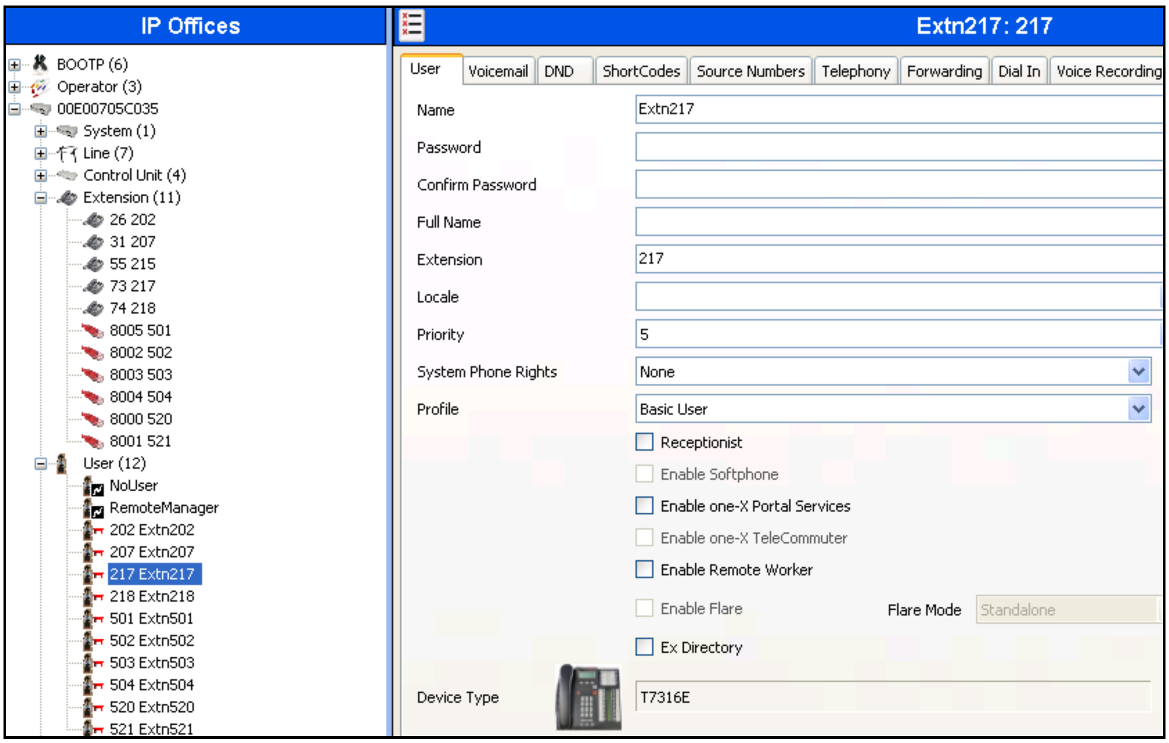
Since default values were used for T.38 fax and AT&T IP Toll Free service does not require registration, the **T38 Fax** and **SIP Credentials** tabs need not be visited. Click **OK** [not shown] to commit the SIP Line configuration.

5.5. Users, Extensions, and Hunt Groups

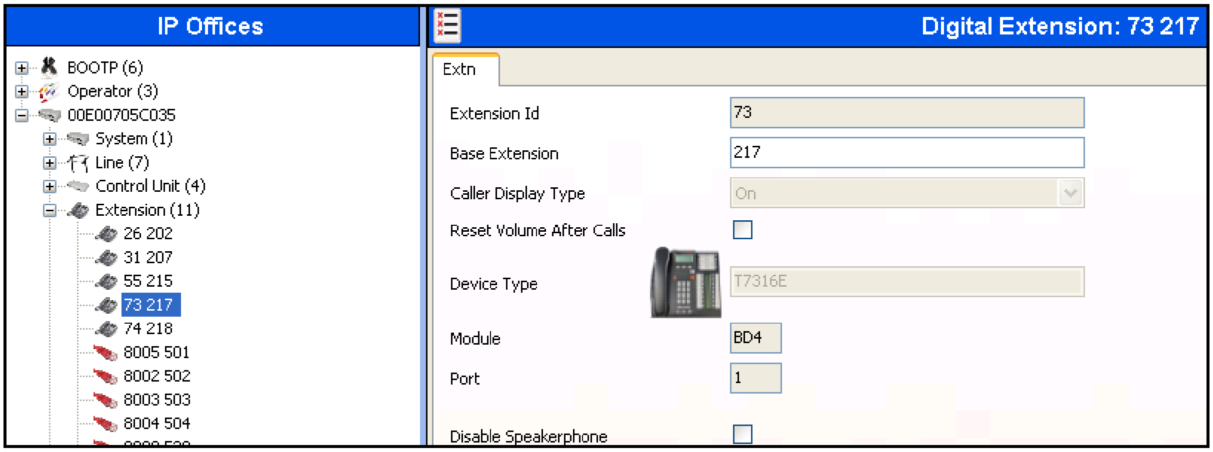
In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions used will be presented, since the configuration can be easily extrapolated. To add a User, right click on **User** in the Navigation pane, and select **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.

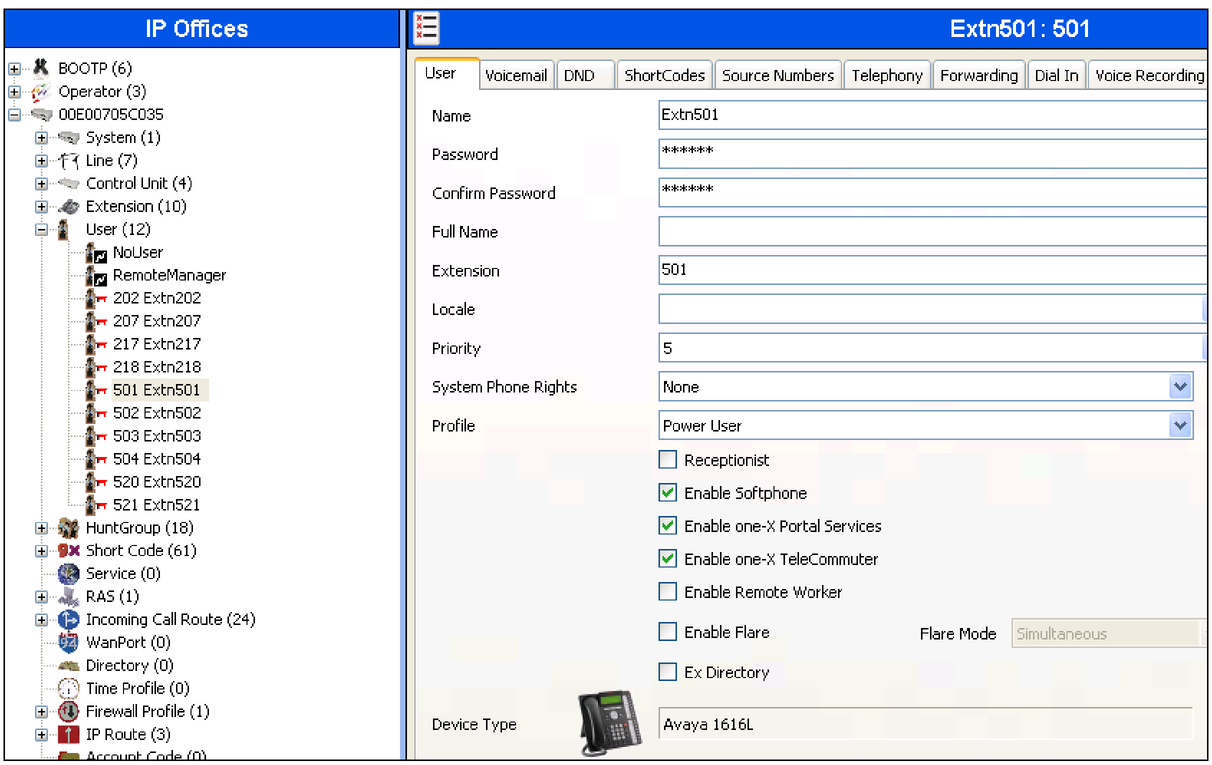


The following screen shows the Extension information for this user. To view, select **Extension** and the appropriate extension in the Navigation pane.

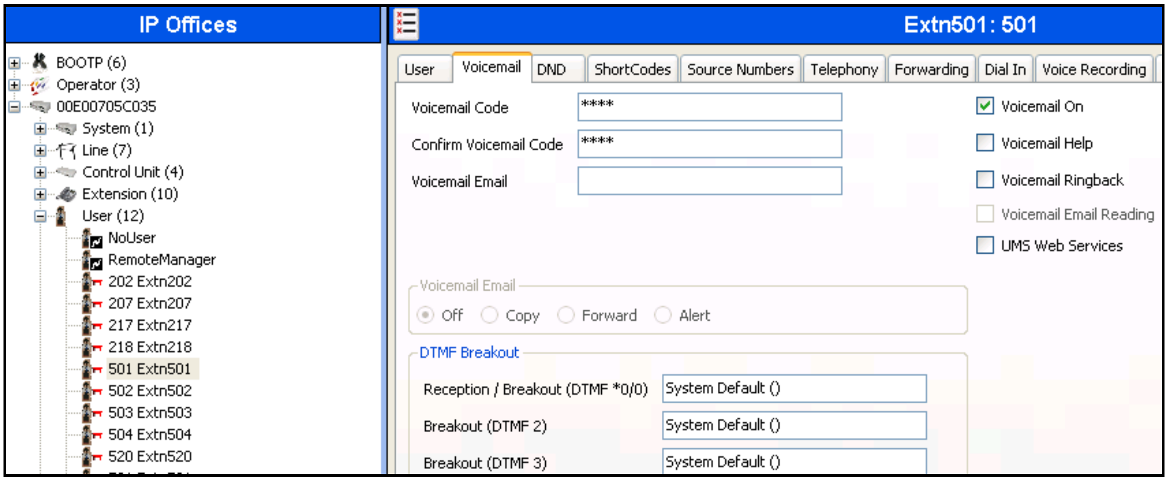


5.5.2. IP Telephone User 501

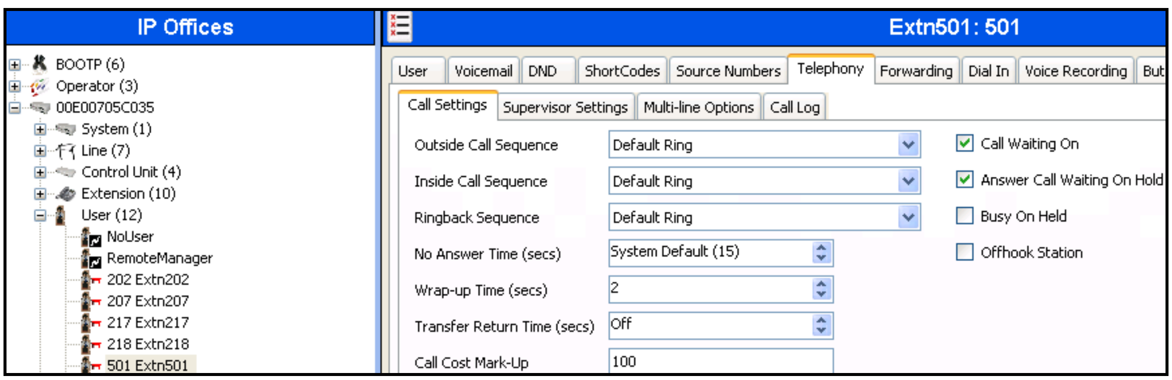
The following screen shows the **User** tab for User **501**. This user corresponds to an Avaya 1616 IP Telephone that is configured as power user with IP Office Softphone features enabled as shown below.



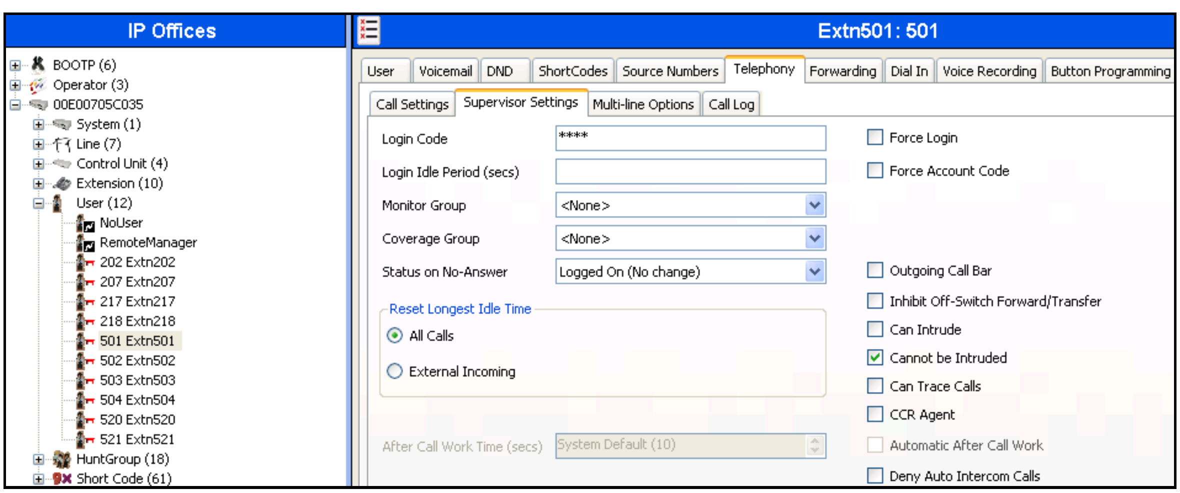
The following screen shows the **Voicemail** tab for this user. The **Voicemail On** box is checked, and a voicemail password is entered in the **Voicemail Code** and **Confirm Voicemail Code** fields.



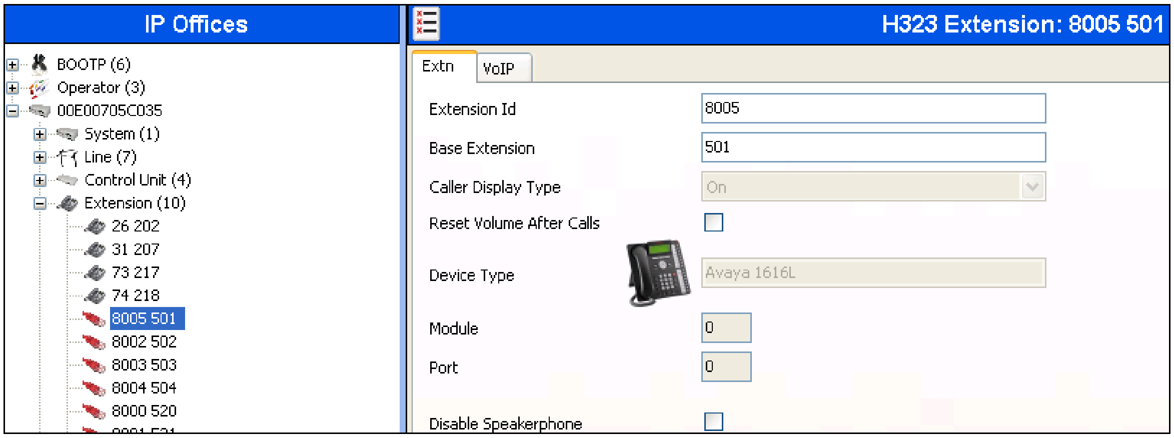
Navigate to **Telephony→ Call Settings** and check the **Call Waiting On** box to allow an IP Office Softphone to have multiple call appearances (necessary for call transfer).



Navigate to **Telephony→ Supervisor Settings** and enter a **Login Code** to allow hot-desking.

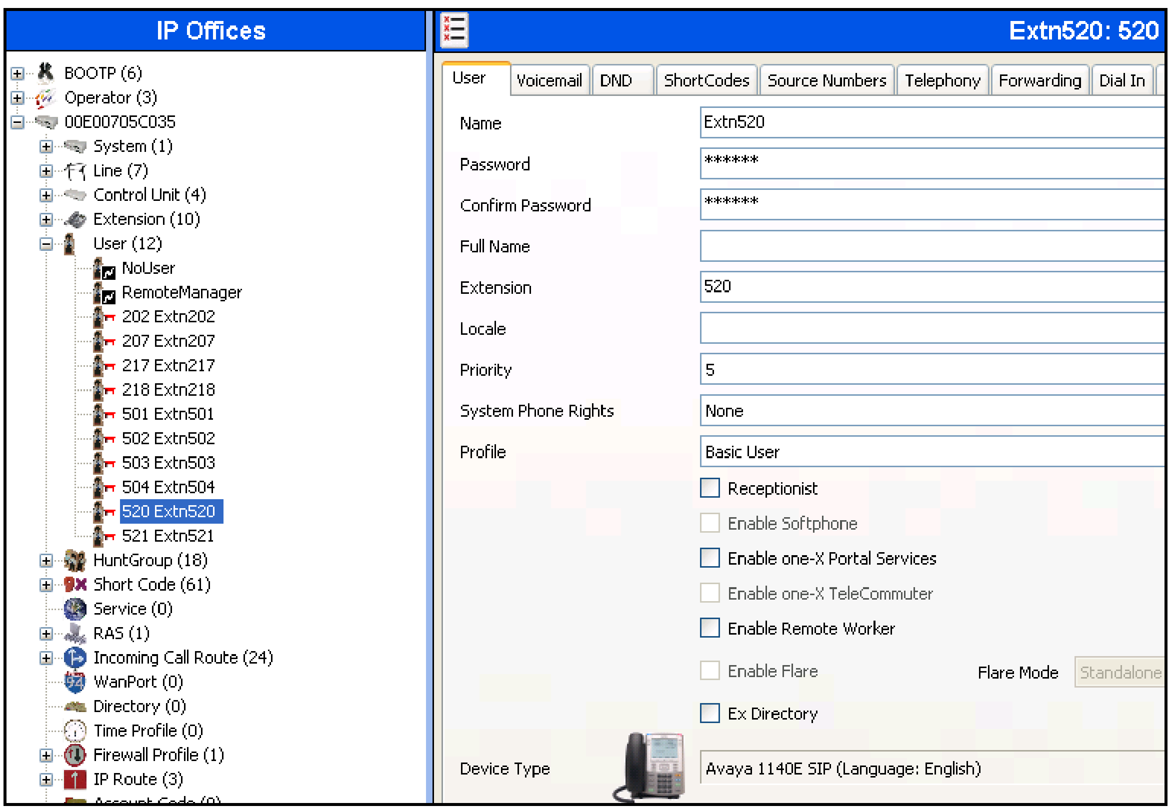


The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for an IP Telephone. In this reference configuration **Codec Selection** was set to **System Default** configured in **Section 5.3.5**.

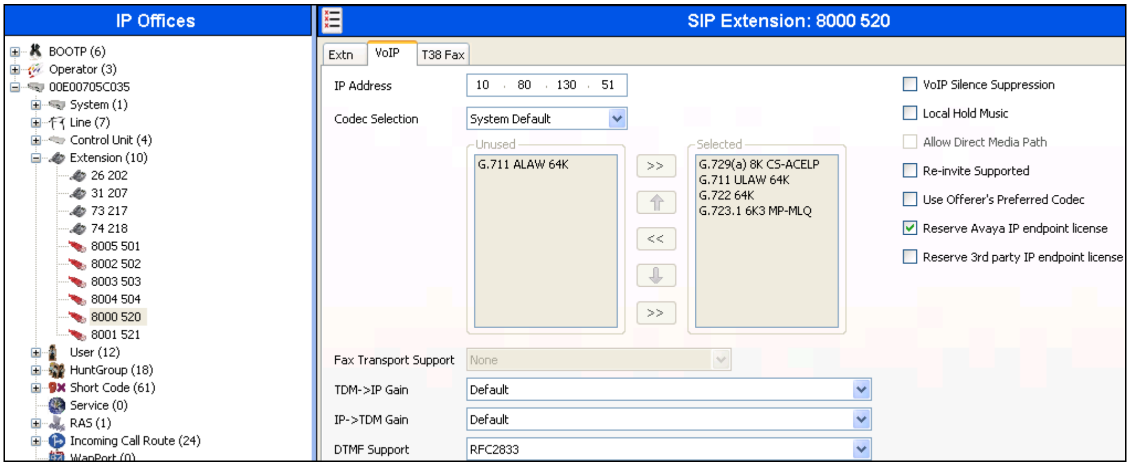


5.5.3. SIP Telephone User 520

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



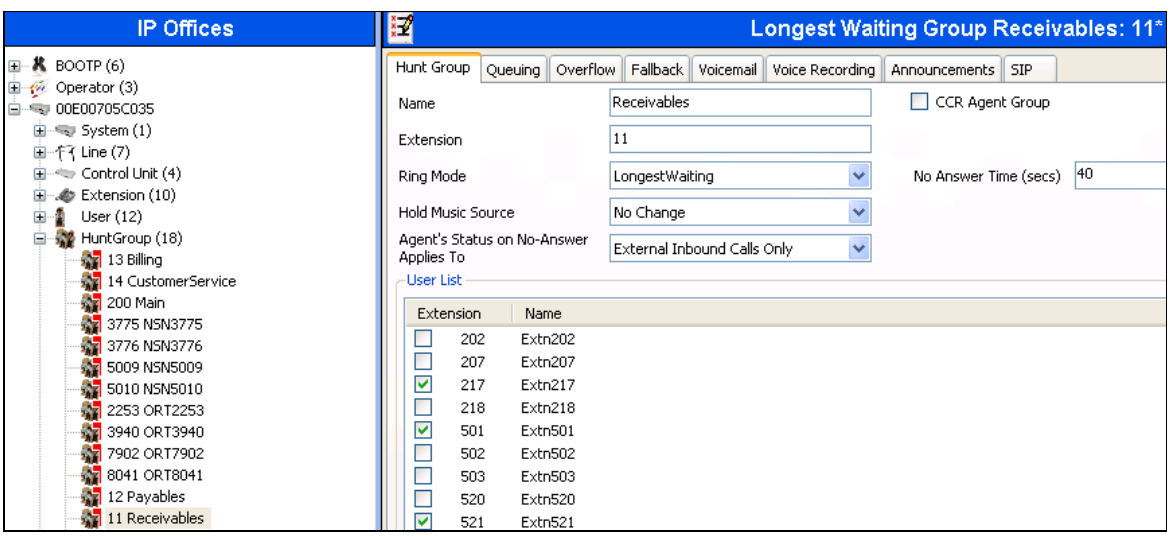
The following screen shows the Extension information for this user. Note that for a SIP telephone, the IP Address configured for the phone needs to be specified. In this example, **10.80.130.51** was assigned to the Avaya 1140E telephone. All other screens are configured the same way as in **Section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



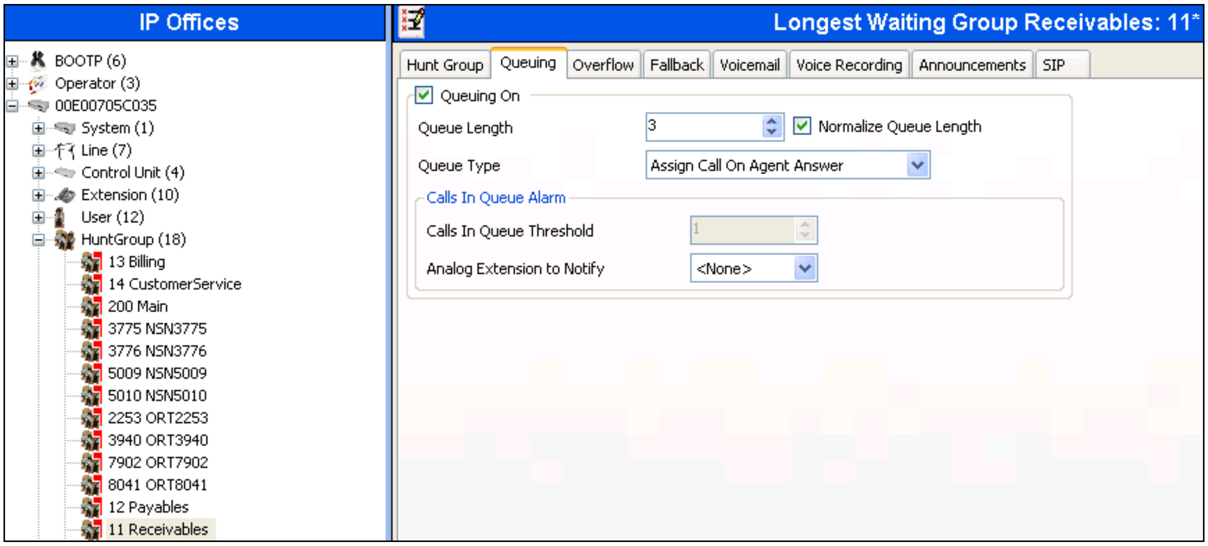
5.5.4. Hunt Groups

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 a hunt group from the Navigation pane.

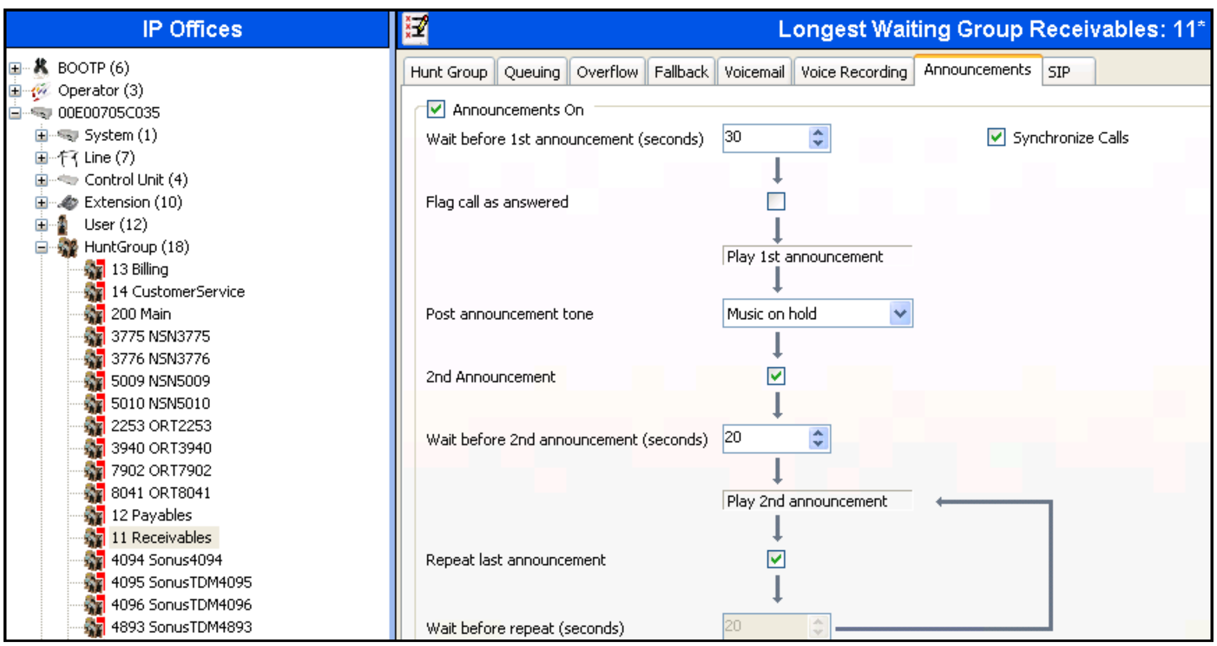
The following screen shows the **Hunt Group** tab for hunt group 11. The group name is set to **Receivables**. Several extensions/agents are part of this hunt group. Since the **Ring Mode** field is set to **LongestWaiting**, this will enable to ring the least used extension in the hunt group. Click the **Edit** button [not shown] to change the **User List** [not shown]. Once a user is part of a hunt group, it can be enabled/disabled by checking/unchecking the box by the Extension field in **User List**.



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.



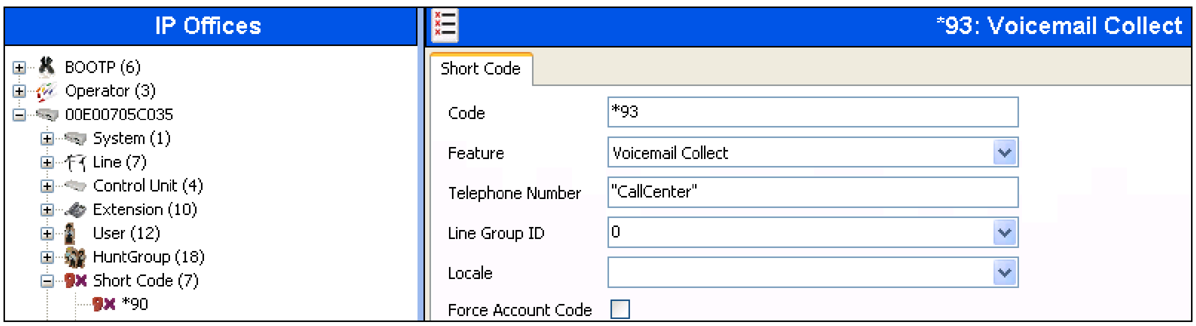
Similarly, additional hunt groups **Billing**, **Payables** and **Customer Service** are created in this reference configuration to exercise the Call Center functionality within IP Office.

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.

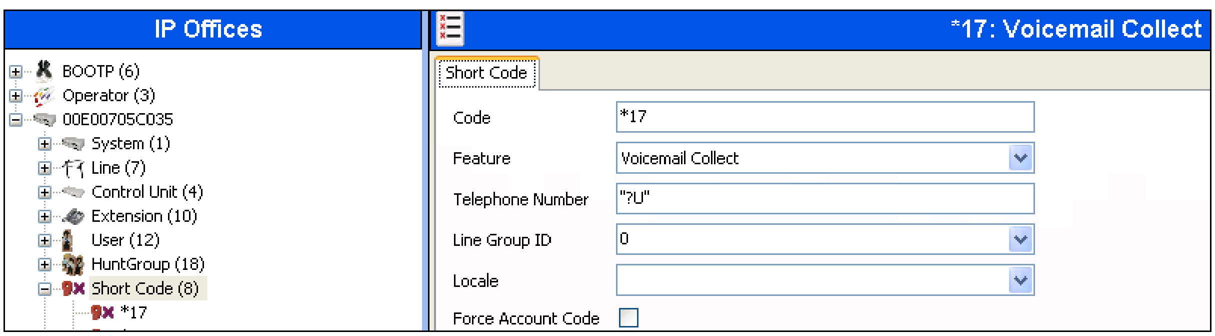
5.6.1. Call Center Codes

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 code set to access this functionality.



5.6.2. Voicemail Retrieval Code

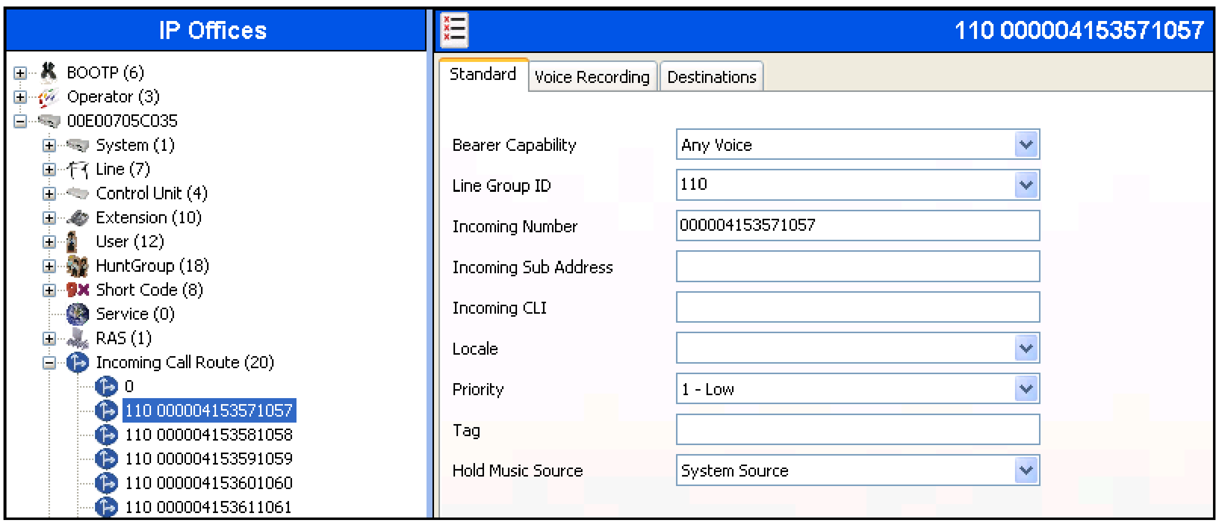
To retrieve voicemails left in individual mailboxes, the following code was configured in this reference configuration. When a user enters, ***17**, they can retrieve the messages in their mailbox.



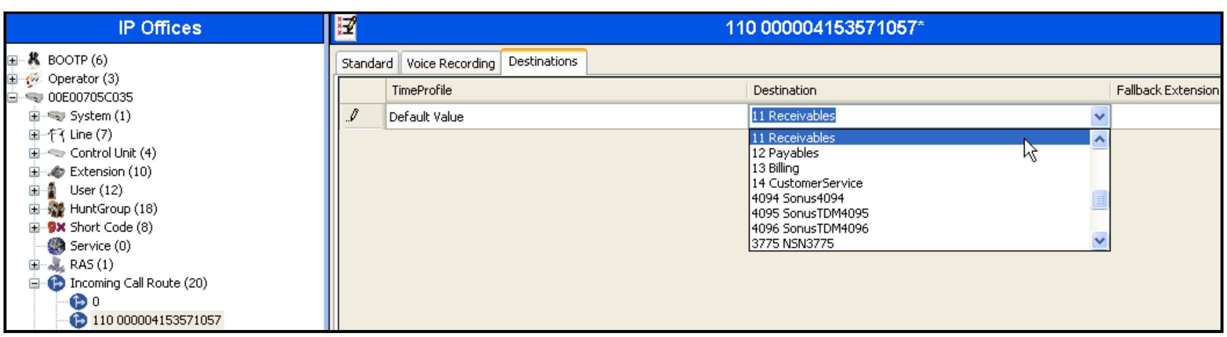
5.7. Incoming Call Routes

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.

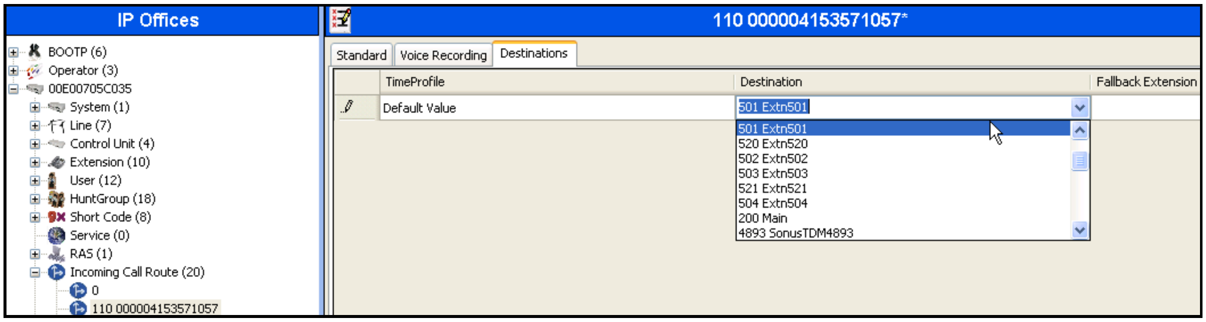
The screen shown below matches the AT&T IP Toll Free DID **00000415351057** in the **Incoming Number** field on the **Line Group Id** (**110**). The **Line Group Id** matches the **Incoming Group** field and **Incoming Number** matches the **Local URI** field configured in the **SIP URI** tab for the SIP Line to AT&T IP Toll Free service in **Section 5.4.3**.



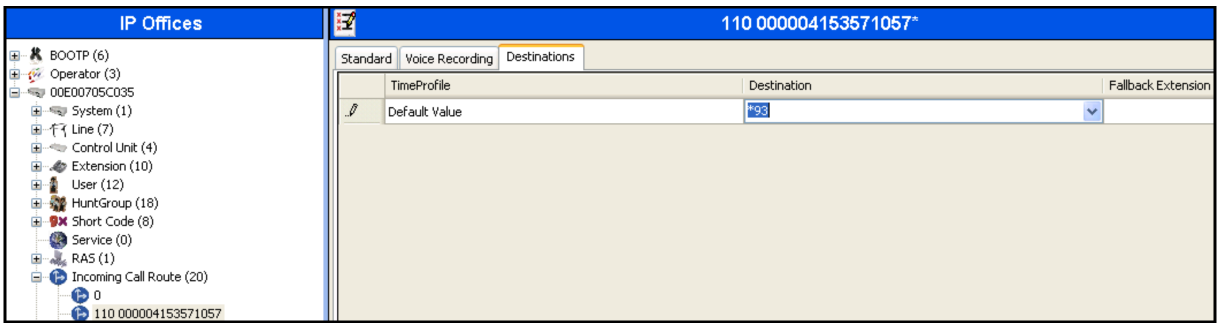
Select the **Destinations** tab and a value can be either selected from the drop-down list or manually entered. In the screen shown below, the hunt group configured in **Section 5.5.4** was selected.



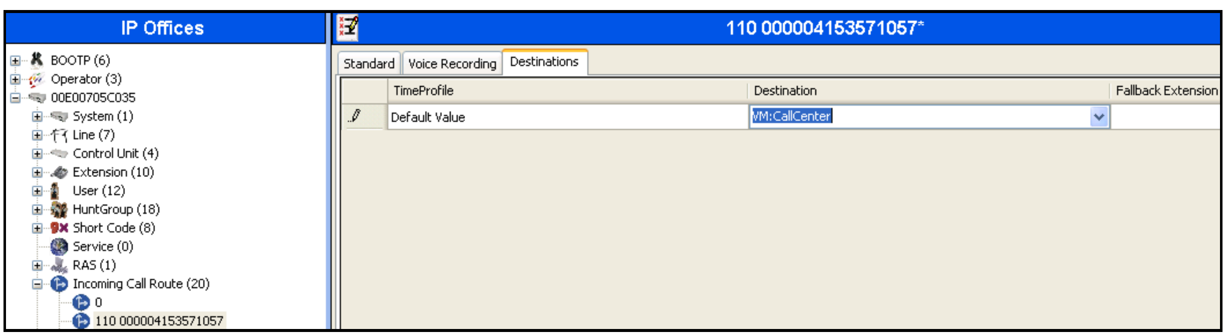
Similarly, in the screen below, an extension configured in **Section 5.5.2** was selected.



The following screen 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**.



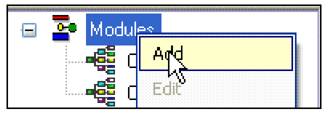
The following screen displays another mechanism to access the Call Center functionality without using the short code. The Call Center functionality is configured in Voicemail Pro as detailed in **Section 5.8**.



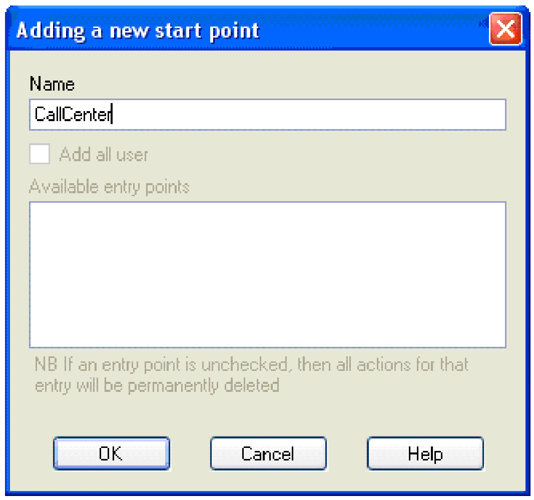
5.8. Call Center Provisioning in Voicemail Pro

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

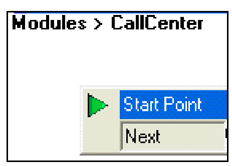
1. Navigate to **Start→Voicemail Pro Client** and right click on modules and select **Add** to add a new module.



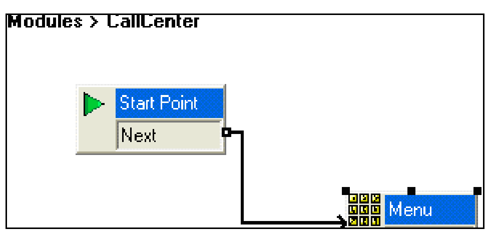
2. In the screen below, enter **CallCenter** in the **Name** field and click OK.



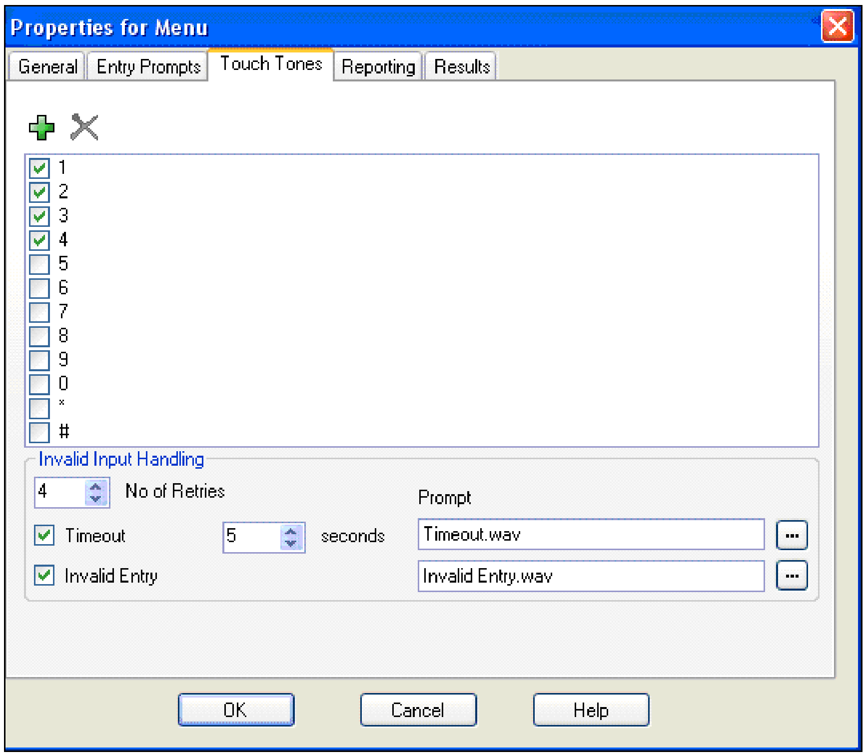
3. Following screen is displayed indicating the starting point for the Call Center functionality.



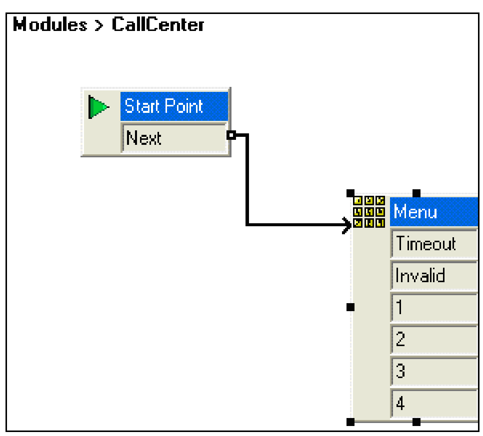
4. Under the **Actions** tab, select **Basic Actions** [not shown]. Select **Menu** and place it on the right side of the pane and then connect the **Start Point** to **Menu** as shown below:



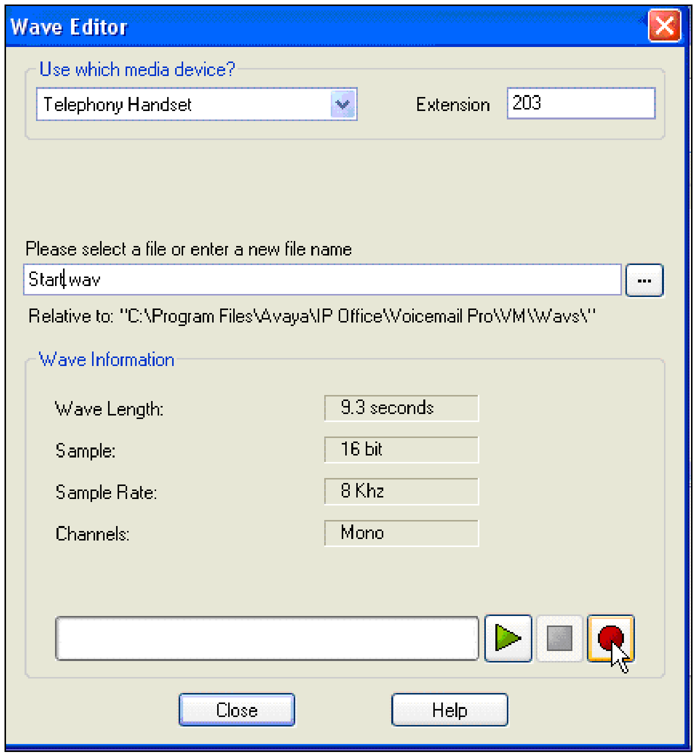
5. Right click on **Menu** and select **Touch Tones** tab. Check the appropriate boxes. In this reference configuration, 1, 2, 3, 4, Timeout and Invalid Entry boxes were selected. This allows caller to enter any of the digits from 1 to 4 to go to the appropriate agent. Digit **1** was used for **Receivables**, 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.



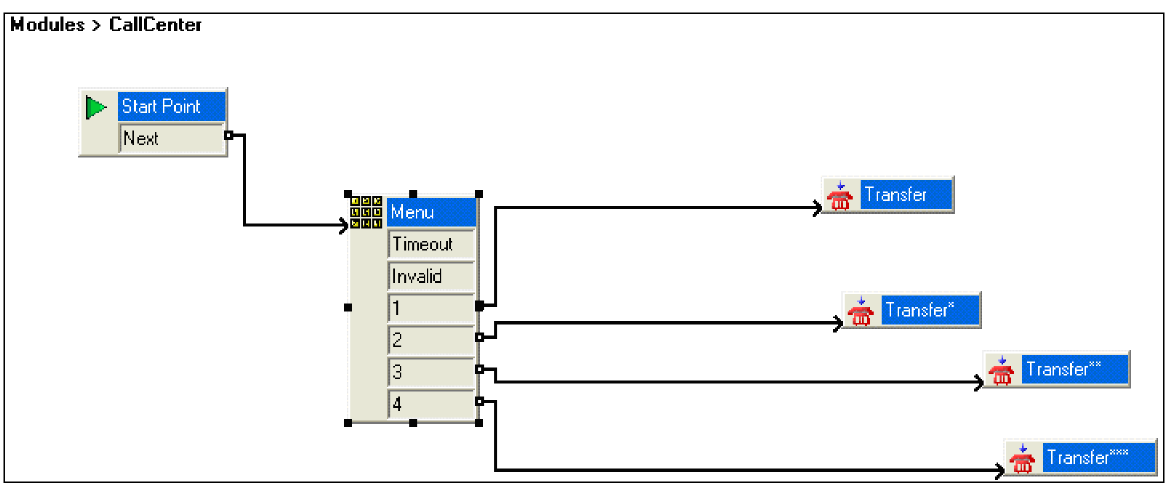
6. Click **OK** and following screen is displayed:



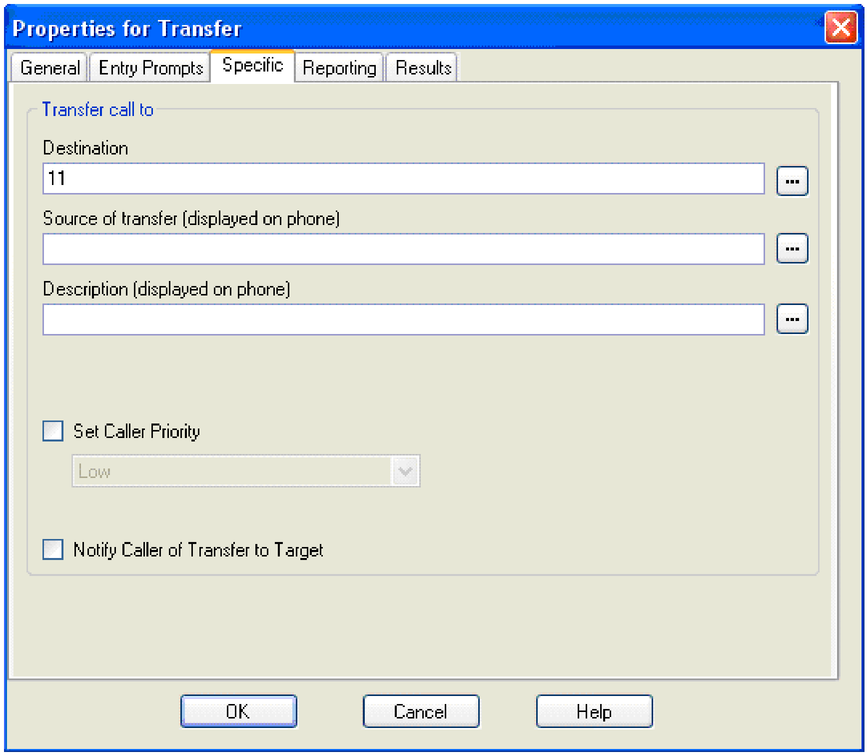
7. Right click on the **Menu** action and select the **Entry Prompts** tab and click on the ✚ sign [not shown]. On the following screen, enter the **Extension** where the recording is done and the filename for the recording in the **Please select a file or enter a new file name** field and press the 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 when a call comes into the CallCenter. The green button is used to verify the recording.



8. 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.

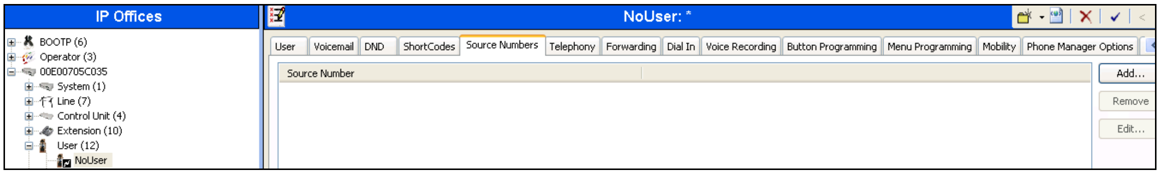


Right click on the **Transfer** action and select the **Specifics** tab. In the **Destination** field enter the hunt group/skill number created in **Section 5.5.4** and click **OK**. This will enable the call to be routed to the appropriate skill. Repeat this step for all the **Transfer** actions.

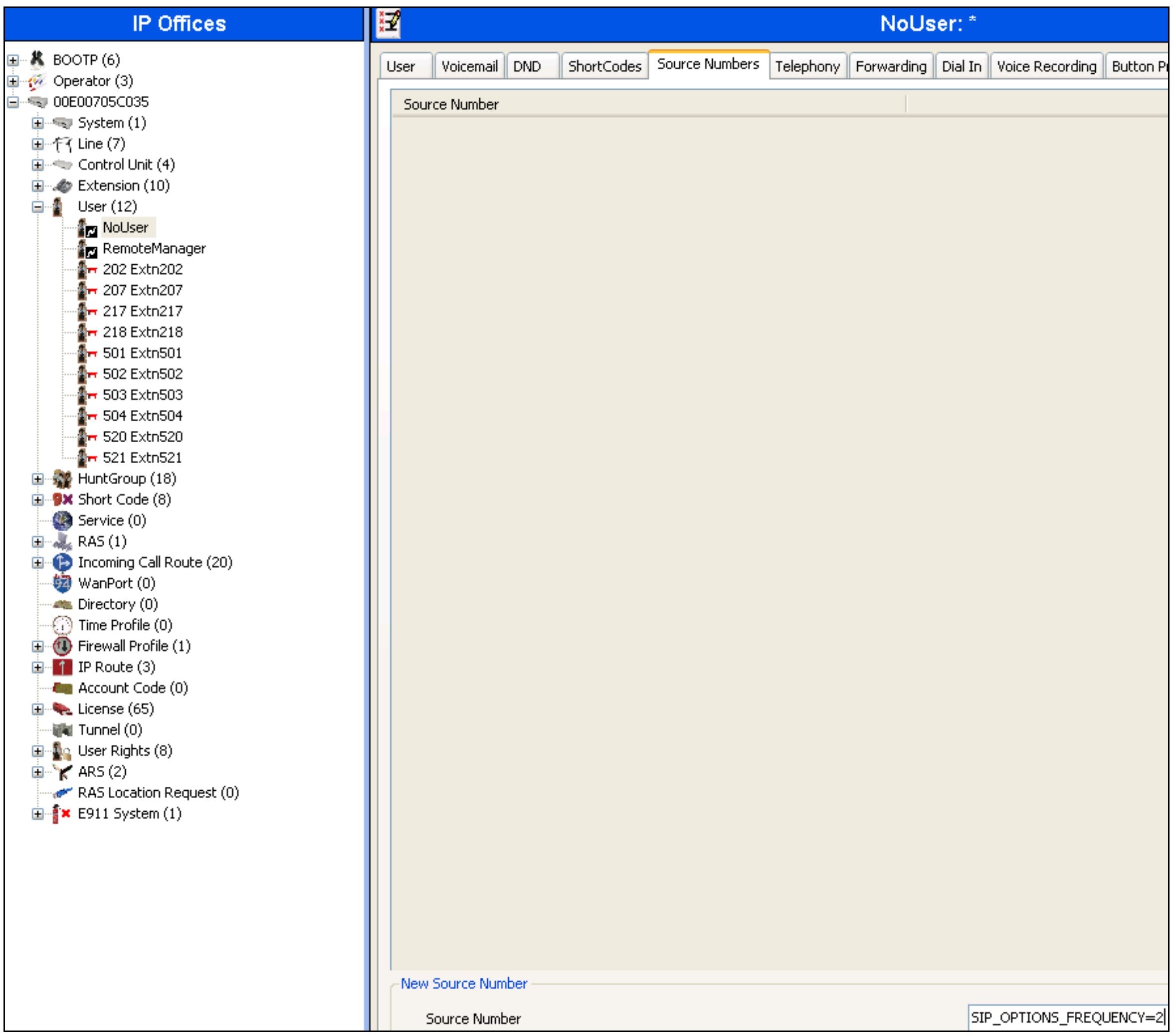


5.9. SIP OPTIONS Frequency

- From the Navigation pane, navigate to **User→NoUser**. In the NoUser Details pane shown below, select the tab **Source Numbers** and press the **Add…** button.

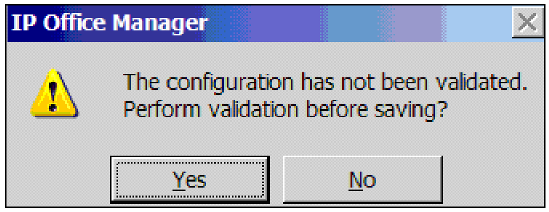


- In the following screen, enter **SIP_OPTIONS_PERIOD=2** in the **Source Number** field and click **OK** [not shown]. This will set the frequency of the SIP OPTIONS message sent by IP Office to 2 minutes.

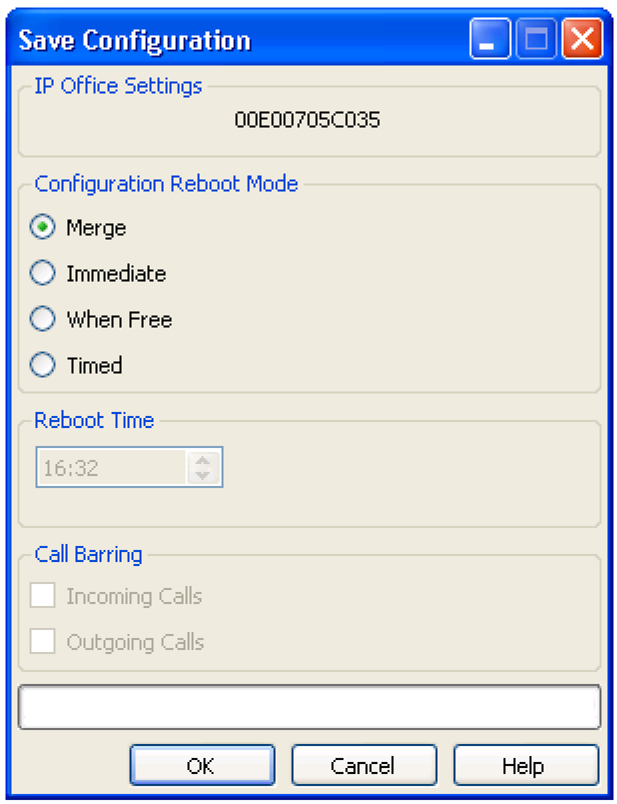


5.10. Saving Configuration Changes to IP Office

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 icon. 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:

- 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 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 IP Office Manager is installed. 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). Additionally, **System Status** application can also be used to verify the extension status, alarms and call status.
- Use the IP Office System Monitor application to monitor activity on IP Office including tracing a call. Launch the application from **Start→Programs→IP Office→Monitor** on the PC where IP Office Manager is installed.

7. Conclusion

As illustrated in these Application Notes, IP Office can be configured to interoperate successfully with the AT&T IP Toll Free service. This solution provides users of IP Office the ability to support inbound toll free calls over an AT&T IP Toll Free SIP trunk service connection via MIS/PNT or AVPN 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. Additional 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 Basic Edition – Partner Mode Installation Manual, Issue 26n, October 26, 2012
Document Number 15-601042
https://downloads.avaya.com/css/P8/documents/100162530

[IPO-MGR] IP Office Release 8.1 Manager 10.1 Issue 28n, August 3, 2012
Document Number 15-601011
https://downloads.avaya.com/css/P8/documents/100162522

[IPO-SYSSTAT] IP Office Release 8.1 System Status Application, Issue 07a, November 26, 2012
Document Number 15-601758
http://downloads.avaya.com/css/P8/documents/100150298

[IPO-VMPRO] IP Office Release 8.1 Administering Voicemail Pro, Issue 27b, June 5, 2012
Document Number 15-601063
https://downloads.avaya.com/css/P8/documents/100162853

[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/

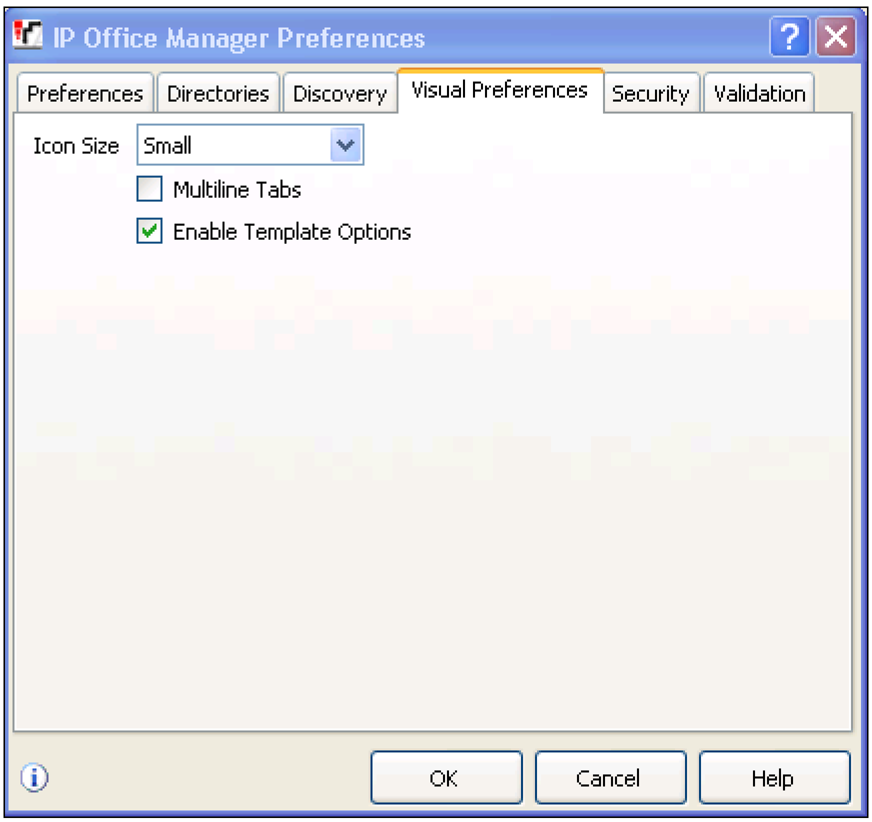
9. Appendix – Example SIP Trunk Template

IP Office Release 8.1 supports SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors. Note that not all of the configuration information, particularly items relevant to specific installation environment, is included in the SIP Line Template. Therefore it is critical that the SIP Line configuration be verified/updated after a template has been imported and additional configuration be supplemented using **Section 5.4** in these Application Notes as a reference.

9.1. Configure IP Office Manager for Template Creation

To enable IP Office to create a SIP Trunk template, configure as follows on the desktop where the IP Office Manager is installed:

1. Navigate to **File → Preferences** on the IP Office Manager and select the **Visual Preferences** tab. Check the **Enable Template Options** box as shown below and click **OK**.

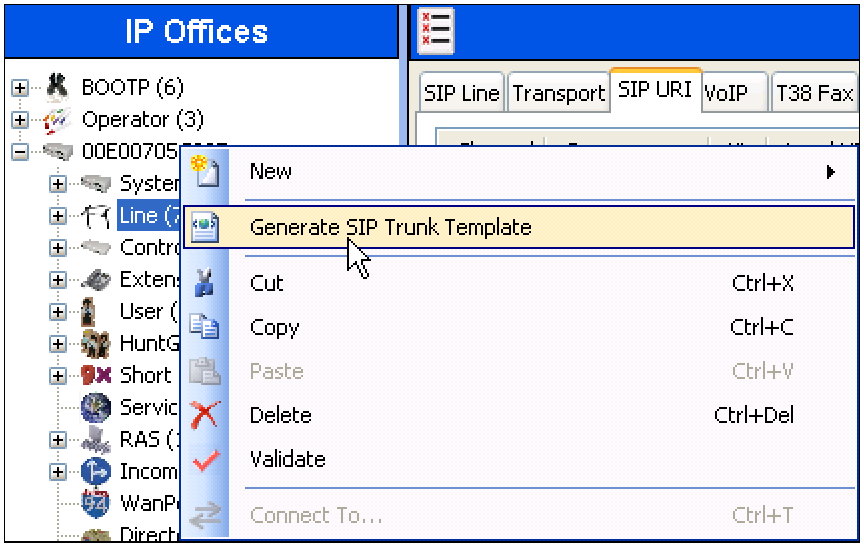


2. Run **regedit** on the desktop and navigate to **HKEY_CURRENT_USER/Software/IP400/Manager** and add a **DWORD** value **TemplateProvisioning** and set its value to **1**.

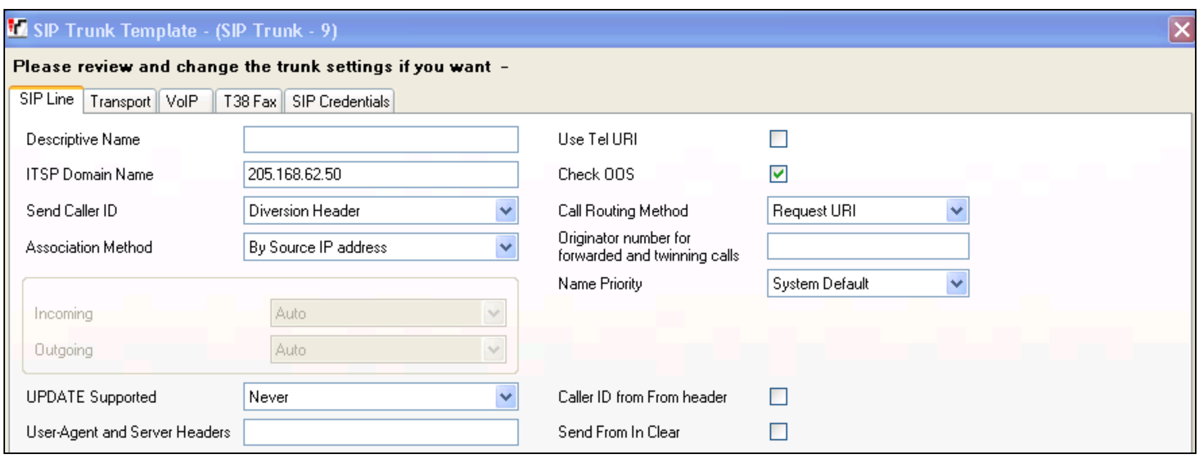
9.2. Generate a SIP Trunk Template

To generate a SIP Trunk template from an existing SIP trunk, execute the following steps:

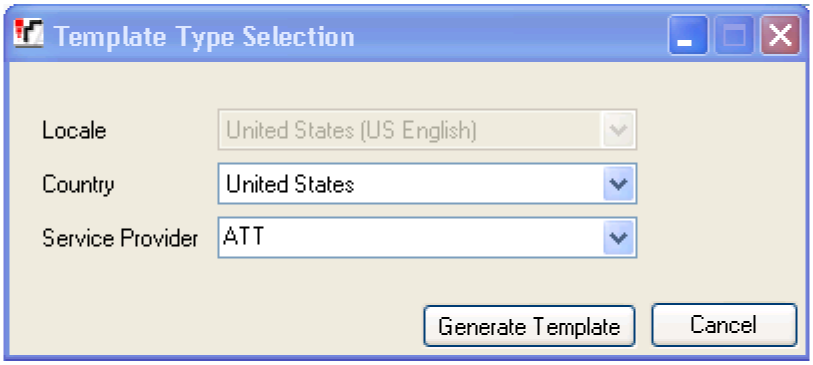
1. Select the SIP trunk under line and right click on the SIP line numbed for which the SIP trunk template is to be generated and then click **Generate SIP Trunk Template**.



2. In the SIP Trunk Template screen shown below, enter a template name in the **Description Name** field and click **Export** [not shown].



3. In the **Template Type Selection** screen, enter **Country** and **Service Provider** and click **Generate Template**.



4. A popup screen shows up [not shown] asking where the template is to be stored. This section shows an example SIP Trunk Template generated from the configuration presented in this document.

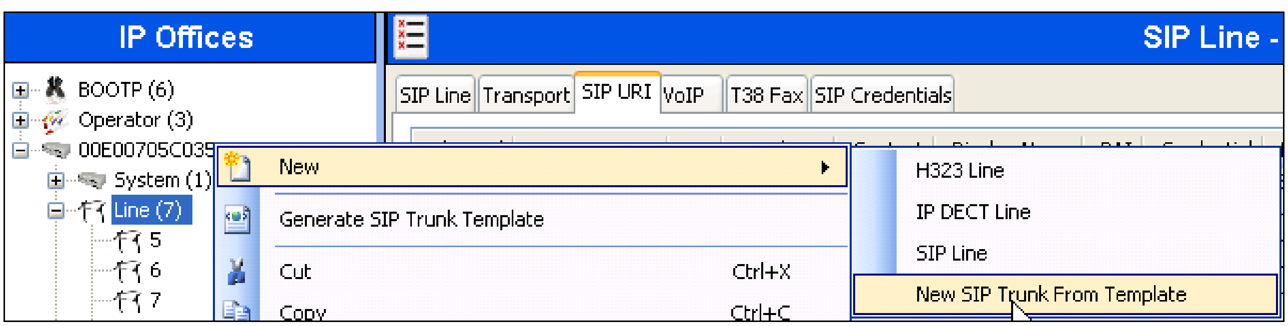

```xml
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
<TemplateType>SIPTrunk</TemplateType>
<Version>20121211</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>ATT</DescriptiveName>
<ITSPDomainName>192.168.62.50</ITSPDomainName>
<SendCallerID>CallerIDDIV</SendCallerID>
<ReferSupport>false</ReferSupport>
<ReferSupportIncoming>2</ReferSupportIncoming>
<ReferSupportOutgoing>2</ReferSupportOutgoing>
<RegistrationRequired>false</RegistrationRequired>
<UseTelURI>false</UseTelURI>
<CheckOOS>true</CheckOOS>
<CallRoutingMethod>1</CallRoutingMethod>
<OriginatorNumber />
<AssociationMethod>SourceIP</AssociationMethod>
<LineNamePriority>SystemDefault</LineNamePriority>
<UpdateSupport>UpdateNever</UpdateSupport>
<UserAgentServerHeader />
<CallerIDfromFromheader>false</CallerIDfromFromheader>
<PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
<ITSPProxy>135.242.225.210</ITSPProxy>
<LayerFourProtocol>SipUDP</LayerFourProtocol>
<SendPort>5060</SendPort>
<ListenPort>5060</ListenPort>
<DNSServerOne>0.0.0.0</DNSServerOne>
<DNSServerTwo>0.0.0.0</DNSServerTwo>
<CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
```

```
<SeparateRegistrar />
<CompressionMode>AUTOSELECT</CompressionMode>
<UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
<AdvCodecPref>G.729(a) 8K CS-ACELP,G.711 ULAW 64K,G.723.1 6K3
  MP-MLQ,G.722 64K</AdvCodecPref>
<CallInitiationTimeout>4</CallInitiationTimeout>
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<VoipSilenceSupression>true</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_T38</FaxTransportSupport>
<UseOffererPrefferedCodec>true</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown>
<Rel100Supported>false</Rel100Supported>
<T38FaxVersion>3</T38FaxVersion>
<Transport>UDPTL</Transport>
<LowSpeed>0</LowSpeed>
<HighSpeed>0</HighSpeed>
<TCFMethod>Trans_TCF</TCFMethod>
<MaxBitRate>FaxRate_14400</MaxBitRate>
<EflagStartTimer>2600</EflagStartTimer>
<EflagStopTimer>2300</EflagStopTimer>
<UseDefaultValues>true</UseDefaultValues>
<ScanLineFixup>true</ScanLineFixup>
<TFOPEnhancement>true</TFOPEnhancement>
<DisableT30ECM>false</DisableT30ECM>
<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>false</DisableT30MRCompression>
<NSFOverride>false</NSFOverride>
  </Template>
```


9.3. Create SIP Trunk from Template

To create a SIP Trunk from template shown above, execute the following steps:

1. Right click on **Line**, select **New** and click **New SIP Trunk From Template**.



2. In the **Template Type Selection** screen displayed, verify that **Country** is pre-populated with **United States** and **Service Provider** is set to **ATT**. Click **Create new SIP Trunk**.

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.