



Avaya Solution & Interoperability Test Lab

Applications Notes for Avaya IP Office 8.0 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office 8.0 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** 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 8.0 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.

TABLE OF CONTENTS

1.	Introduction.....	4
2.	General Test Approach and Test Results.....	4
2.1.	Interoperability Compliance Testing	5
2.2.	Known Limitations/Test Results.....	5
2.3.	Support.....	5
3.	Reference Configuration	6
3.1.	Illustrative Configuration Information.....	7
3.2.	Call Flows.....	8
3.2.1.	Inbound.....	8
3.2.2.	Coverage to Voicemail.....	9
4.	Equipment and Software Validated.....	10
5.	Avaya IP Office.....	11
5.1.	Physical, Network, and Security Configuration.....	12
5.2.	Licensing.....	15
5.3.	System Settings	16
5.3.1.	System Tab.....	16
5.3.2.	LAN Settings.....	17
5.3.3.	Voicemail	21
5.3.4.	System Telephony Configuration	21
5.3.5.	Codecs	22
5.4.	SIP Line	22
5.4.1.	SIP Line - SIP Line Tab	22
5.4.2.	SIP Line - Transport Tab.....	23
5.4.3.	SIP Line - SIP URI Tab.....	24
5.4.4.	SIP Line - VoIP Tab	25
5.5.	Users, Extensions, and Hunt Groups	25
5.5.1.	Digital Telephone User 217	26
5.5.2.	IP Telephone User 501	28
5.5.3.	SIP Telephone User 520.....	31
5.5.4.	Hunt Groups.....	33
5.6.	Short Codes	35
5.6.1.	Call Center Codes	35
5.6.2.	Voicemail Retrieval Code	35
5.7.	Incoming Call Routes	36
5.8.	Call Center Provisioning in Voicemail Pro	37
5.9.	SIP OPTIONS Frequency.....	41
5.10.	Saving Configuration Changes to IP Office.....	42
6.	Verification Steps.....	43
6.1.	General.....	43
7.	Conclusion	43
8.	Additional References.....	44
9.	Appendix – Example SIP Trunk Template.....	45
9.1.	Configure IP Office Manager for Template Creation.....	45

To enable IP Office to create a SIP Trunk template, configure as follows on the desktop where the IP Office Manager is installed:	45
9.2. Generate a SIP Trunk Template	46
9.3. Create SIP Trunk from Template	48

1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office Preferred Edition 8.0 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** 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.

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 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 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 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 calls 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 8.0 and hence not supported.
3. Shuffling is not supported for SIP trunks in Avaya IP Office 8.0.
4. G.726 codec is not supported by Avaya IP Office 8.0.

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 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.

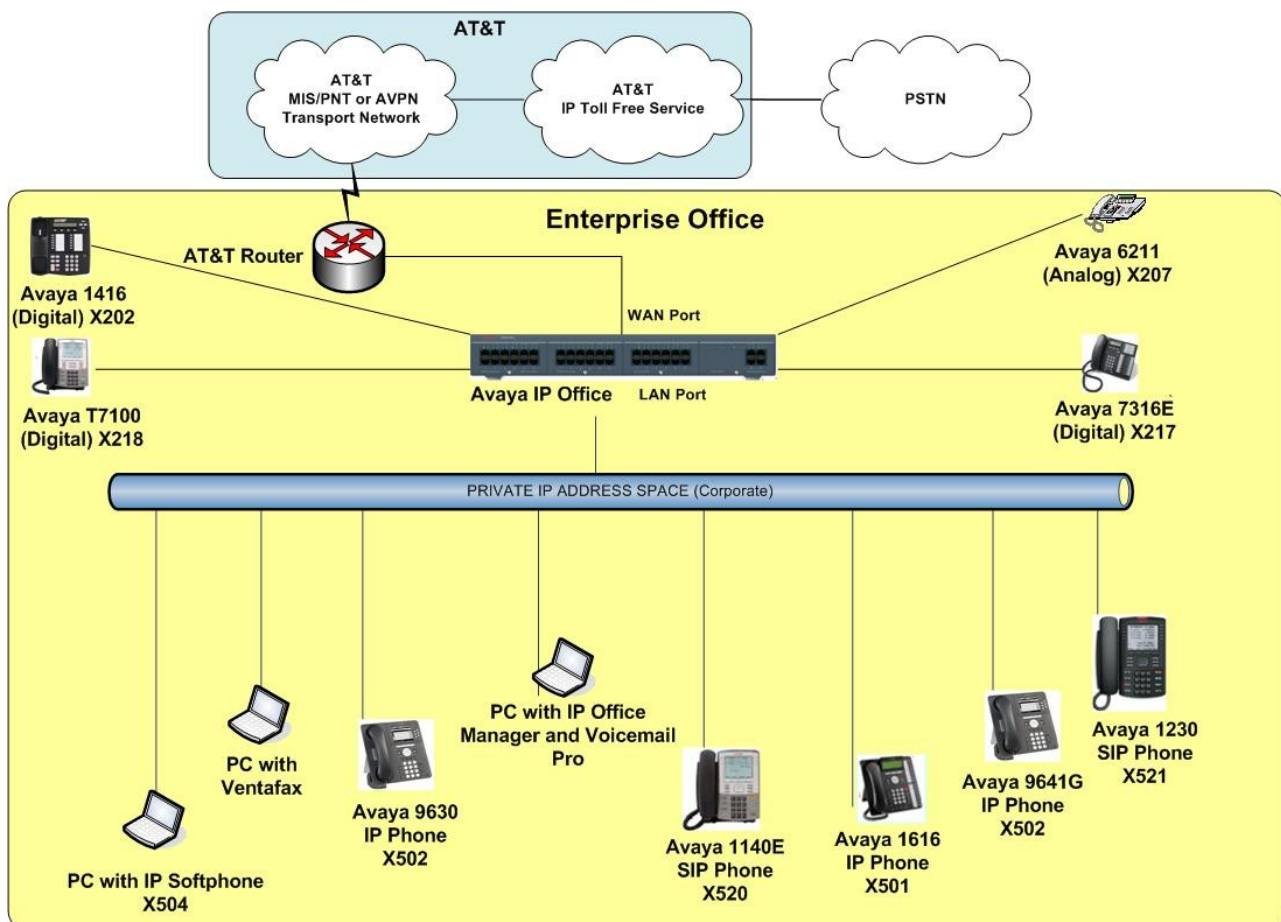


Figure 1: Reference configuration

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	192.168.62.59
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
AT&T IP Toll Free Service	
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 this Reference Configuration

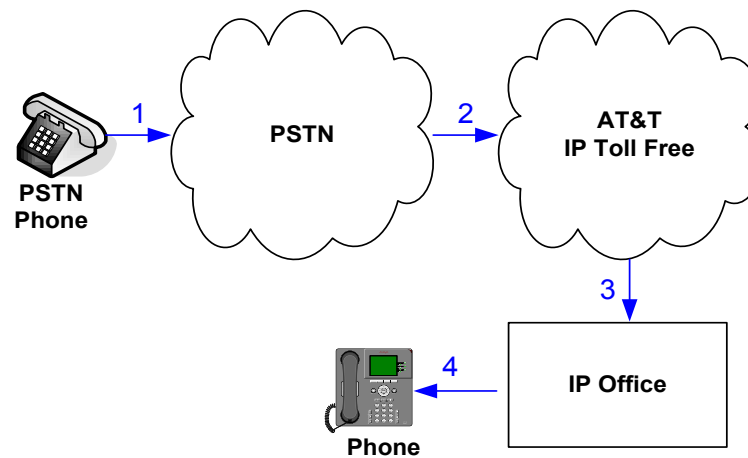
3.2. Call Flows

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.

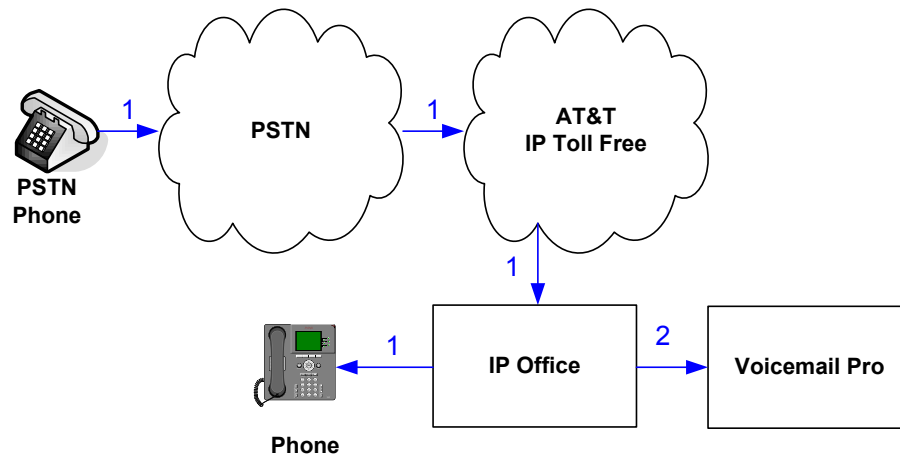
1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to IP Office.
4. Depending on the called number, IP Office routes the call to
 - A hunt group, which in turn, 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	Release 8.0 (42) (Preferred Edition)
Avaya IP Office Manager	Release 10.0 (42) (Preferred Edition)
Avaya IP Office Voicemail Pro	Release 8.0 (1009)
Avaya IP Office Voicemail Pro Client	Version 8.0 (1009)
Avaya 1616IP-Series Telephones (H.323)	Release 3.1
Avaya 9630 IP Telephone	Avaya one-X® Deskphone Edition H.323 Version S3.1.03
Avaya 9641G IP Telephone	R6.2.0.13
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	04.03.09.00 (SIP1140)
Avaya 1230 SIP Telephone	04.03.09.00 (SIP1230)
Fax device	Ventafax Home Version 6.2
AT&T IP Toll Free Service using MIS/PNT transport service connections.	VNI 22

Table 2: Equipment and Software Versions

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, 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.

The configuration steps shown in these application notes are not prescriptive 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 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.

The screenshot displays the IP Office configuration interface. On the left is the 'IP Offices' navigation pane, and on the right is the 'IP 500 V2' details pane.

IP Offices Navigation Pane:

- BOOTP (6)
- Operator (3)
- 00E00705C035
 - System (1)
 - Line (7)
 - Control Unit (4)
 - 1 IP 500 V2** (selected)
 - 2 VCM32
 - 3 COMBO6210/ATM4
 - 5 TCM8
 - Extension (30)
 - User (32)
 - HuntGroup (18)

IP 500 V2 Details Pane:

Unit	
Device Number	1
Unit Type	IP 500 V2
Version	8.0 (42)
Serial Number	00e00705c035
Unit IP Address	10.80.130.58
Interconnect Number	0
Module Number	Control Unit

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**).

IP Offices	0.0.0.0*														
<ul style="list-style-type: none"> BOOTP (5) Operator (3) 00E00705C035 <ul style="list-style-type: none"> System (1) 00E00705C035 <ul style="list-style-type: none"> Line (6) Control Unit (4) Extension (30) User (32) HuntGroup (9) Short Code (61) Service (0) RAS (1) Incoming Call Route (10) WanPort (0) Directory (0) Time Profile (0) Firewall Profile (1) IP Route (3) <ul style="list-style-type: none"> 0.0.0.0 10.80.130.0 	<table border="1"> <tr> <td colspan="2">IP Route</td> </tr> <tr> <td>IP Address</td> <td>0 . 0 . 0 . 0</td> </tr> <tr> <td>IP Mask</td> <td>0 . 0 . 0 . 0</td> </tr> <tr> <td>Gateway IP Address</td> <td>192 . 168 . 62 . 1</td> </tr> <tr> <td>Destination</td> <td>LAN2</td> </tr> <tr> <td>Metric</td> <td>0</td> </tr> <tr> <td colspan="2"><input type="checkbox"/> Proxy ARP</td> </tr> </table>	IP Route		IP Address	0 . 0 . 0 . 0	IP Mask	0 . 0 . 0 . 0	Gateway IP Address	192 . 168 . 62 . 1	Destination	LAN2	Metric	0	<input type="checkbox"/> Proxy ARP	
IP Route															
IP Address	0 . 0 . 0 . 0														
IP Mask	0 . 0 . 0 . 0														
Gateway IP Address	192 . 168 . 62 . 1														
Destination	LAN2														
Metric	0														
<input type="checkbox"/> Proxy ARP															

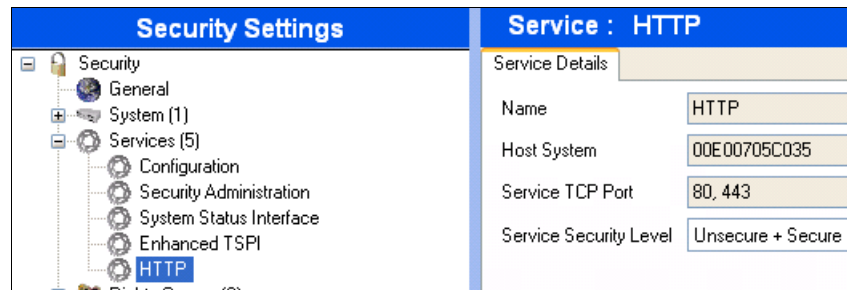
- Another route for **10.80.130.0** subnet was added for the enterprise side **LAN1** port (labeled as LAN port in Figure 1) as shown in the screen below. All the enterprise IP devices were part of this 10.80.130.x network in this reference configuration.

IP Offices	10.80.130.0														
<ul style="list-style-type: none"> BOOTP (5) Operator (3) 00E00705C035 <ul style="list-style-type: none"> System (1) Line (6) Control Unit (4) Extension (32) User (32) HuntGroup (9) Short Code (61) Service (0) RAS (1) Incoming Call Route (10) WanPort (0) Directory (0) Time Profile (0) Firewall Profile (1) IP Route (3) <ul style="list-style-type: none"> 0.0.0.0 10.80.130.0 	<table border="1"> <tr> <td colspan="2">IP Route</td> </tr> <tr> <td>IP Address</td> <td>10 . 80 . 130 . 0</td> </tr> <tr> <td>IP Mask</td> <td>255 . 255 . 255 . 0</td> </tr> <tr> <td>Gateway IP Address</td> <td>10 . 80 . 130 . 1</td> </tr> <tr> <td>Destination</td> <td>LAN1</td> </tr> <tr> <td>Metric</td> <td>0</td> </tr> <tr> <td colspan="2"><input type="checkbox"/> Proxy ARP</td> </tr> </table>	IP Route		IP Address	10 . 80 . 130 . 0	IP Mask	255 . 255 . 255 . 0	Gateway IP Address	10 . 80 . 130 . 1	Destination	LAN1	Metric	0	<input type="checkbox"/> Proxy ARP	
IP Route															
IP Address	10 . 80 . 130 . 0														
IP Mask	255 . 255 . 255 . 0														
Gateway IP Address	10 . 80 . 130 . 1														
Destination	LAN1														
Metric	0														
<input type="checkbox"/> Proxy ARP															

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

Security Service User Login	
IP Office :	00E00705C035 - IP 500 V2
Service User Name	security
Service User Password	••••••••
<input type="button" value="OK"/> <input type="button" value="Cancel"/> <input type="button" value="Help"/>	

- 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, 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.

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.

The screenshot shows the IP Office web interface. On the left, the 'IP Offices' navigation pane lists various features, with 'SIP Trunk Channels' selected at the bottom. The main area on the right is titled 'SIP Trunk Channels' and contains a 'Licenses' section. This section displays the following license information:

Licenses	
License Key	@yedRQt5EtjQhJPiJMpNLYZicZDUr509
License Type	SIP Trunk Channels
License Status	Valid
Instances	255
Expiry Date	Never

2. 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 screenshot shows the IP Office web interface. On the left, the 'IP Offices' navigation pane lists various features, with 'Avaya IP endpoints' selected at the bottom. The main area on the right is titled 'Avaya IP endpoints' and contains a 'Licenses' section. This section displays the following license information:

Licenses	
License Key	64u_@wd8MKv7ns6dBk8@cNsiPmNsEz1
License Type	Avaya IP endpoints
License Status	Valid
Instances	255
Expiry Date	Never

- The screenshot shows the IP Office software interface. On the left, a list of IP Office features is displayed, with 'Power User' selected. On the right, the 'Licenses' tab is active, showing the following details:

Field	Value
License Key	0UH34zyFLNs5nLsW12M_g@datEVXYMr9
License Type	Power User
License Status	Valid
Instances	255
Expiry Date	Never

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.

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.

IP Offices

- [-] [Icon] BOOTP (5)
- [-] [Icon] Operator (3)
- [-] [Icon] 00E00705C035
- [-] [Icon] System (1)
 - [-] [Icon] 00E00705C035
- [-] [Icon] Line (6)
- [-] [Icon] Control Unit (4)
- [-] [Icon] Extension (30)
- [-] [Icon] User (32)
- [-] [Icon] HuntGroup (9)
- [-] [Icon] Short Code (61)
- [-] [Icon] Service (0)
- [-] [Icon] RAS (1)
- [-] [Icon] Incoming Call Route (10)
- [-] [Icon] WanPort (0)
- [-] [Icon] Directory (0)
- [-] [Icon] Time Profile (0)
- [-] [Icon] Firewall Profile (1)
- [-] [Icon] IP Route (3)
- [-] [Icon] Account Code (0)
- [-] [Icon] Licence (64)
- [-] [Icon] Tunnel (0)
- [-] [Icon] User Rights (8)

System
LAN1
LAN2
DNS
Voicemail
Telephony
Director

Name

Contact Information

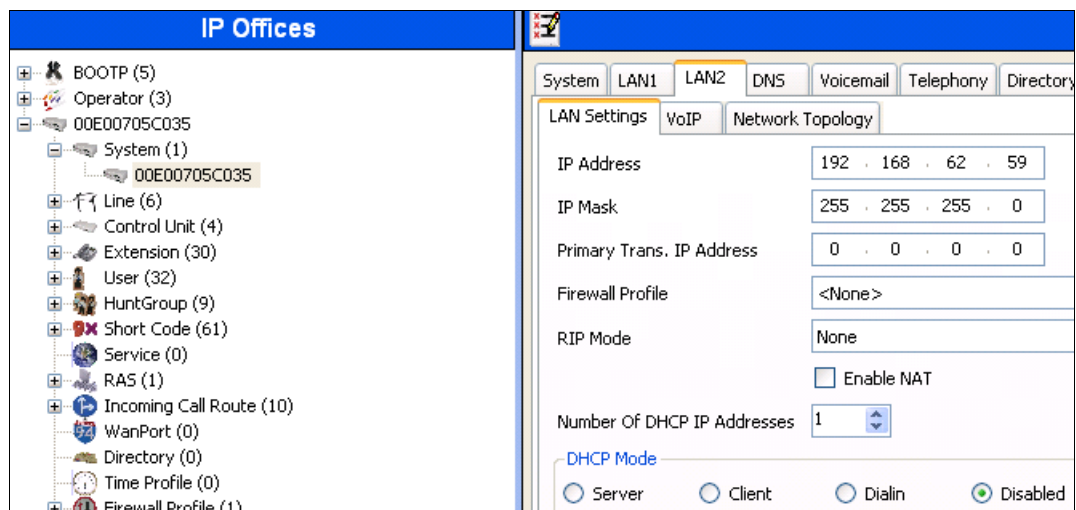
Set contact information to place System under special control

TFTP Server IP Address	0 . 0 . 0 . 0
HTTP Server IP Address	0 . 0 . 0 . 0
Phone File Server Type	Memory Card
Manager PC IP Address	0 . 0 . 0 . 0
Avaya HTTP Clients Only	✓
Enable SoftPhone HTTP Provisioning	✓
Automatic Backup Command	✓
Time Setting Config Source	Voicemail Pro/Manager

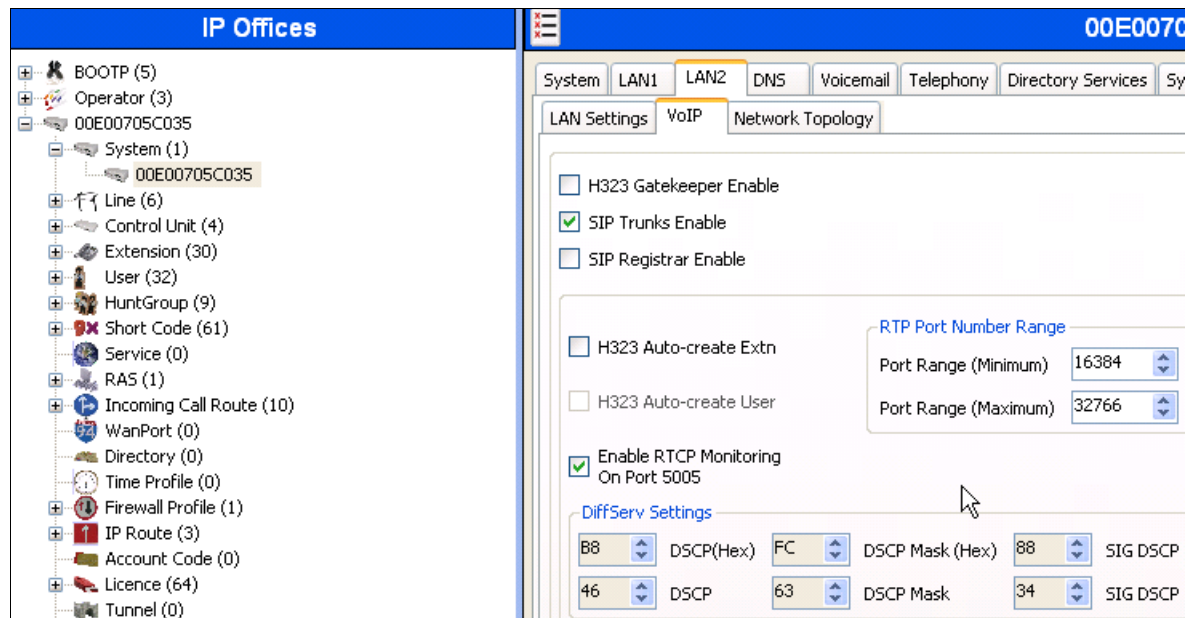
5.3.2. LAN Settings

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.

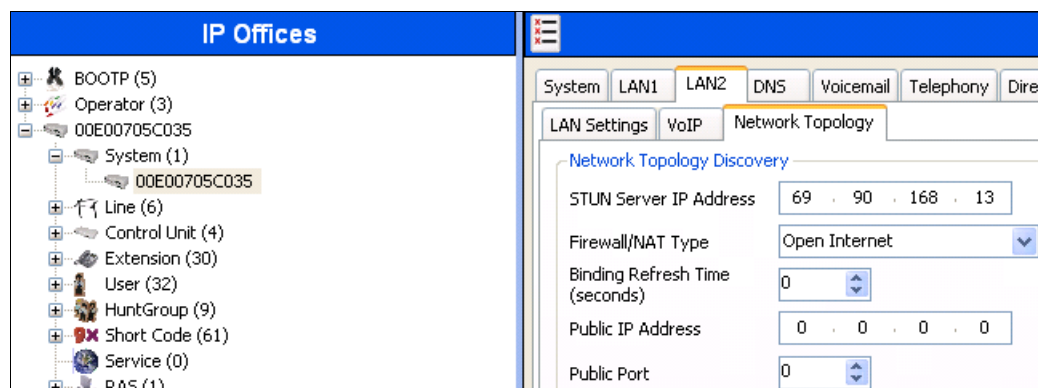
1. Navigate to **LAN2 → LAN Settings** and configure as follows:
 - **IP Address** – Set to **192.168.62.59** which is the IP address of IP Office known to AT&T network
 - **IP Mask** – Set to a valid value e.g **255.255.255.0**
 - **Primary Trans. IP Address** – Set to **0.0.0.0**
 - **DHCP Mode** – The **Disabled** radio button was selected in this reference configuration



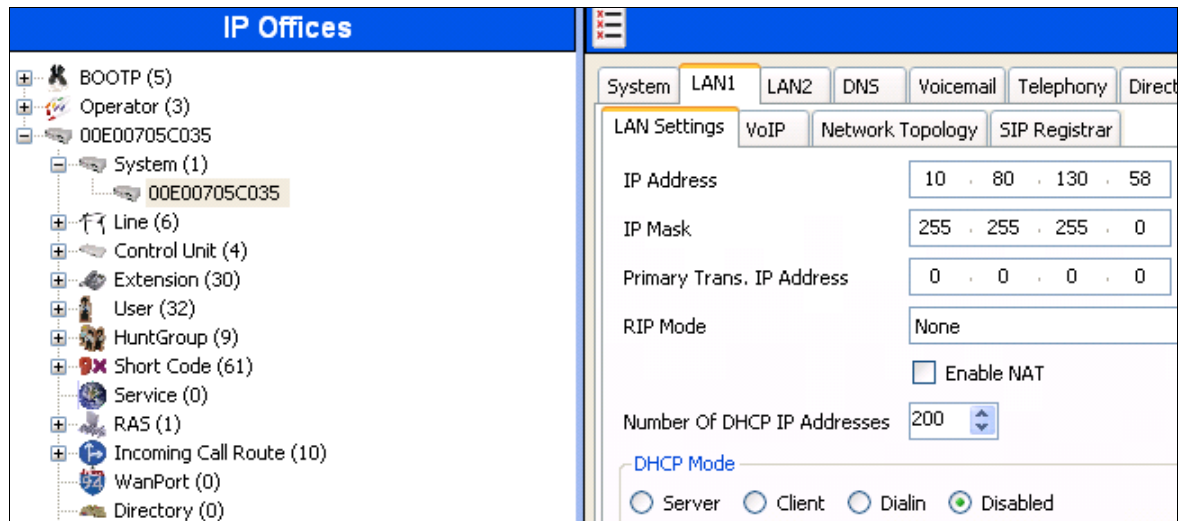
2. Select the **VoIP** tab as shown in the following screen and configure as follows:
 - **SIP Trunks Enable** – Check this box to enable the configuration of SIP trunks
 - **RTP Port Range (Minimum)** – Set to **16384** (As required by AT&T)
 - **RTP Port Range (Maximum)** – Set to **32766** (As required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



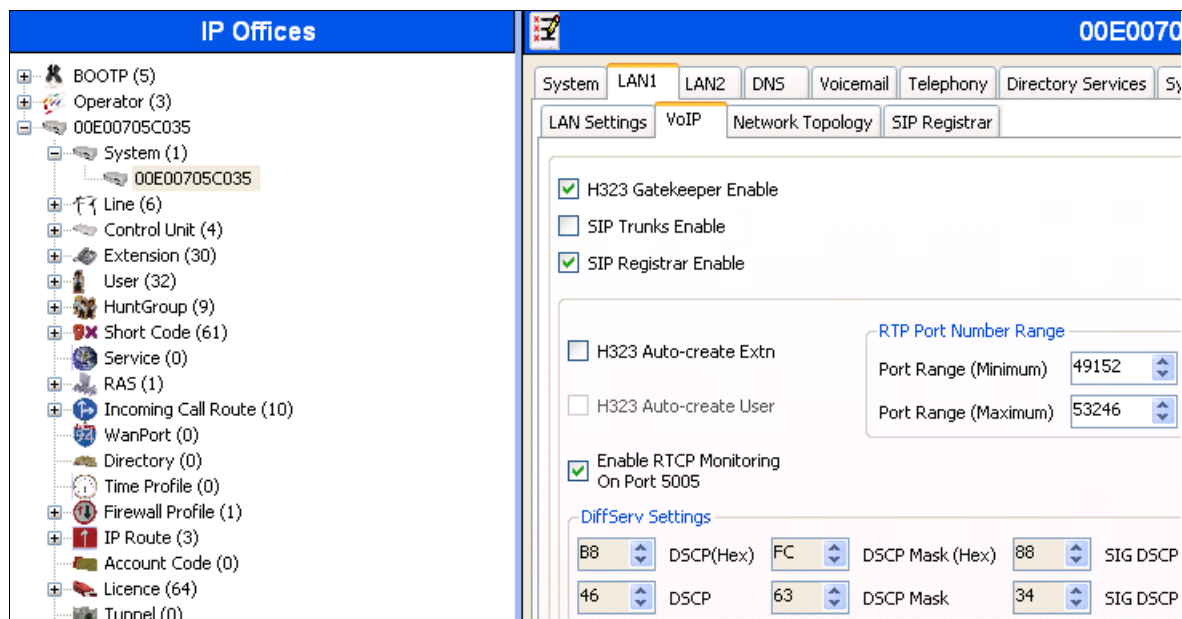
3. Select the **Network Topology** tab as shown in the following screen and set **Firewall/NAT Type** field to **Open Internet**. With this configuration, STUN will not be used but make sure to leave **STUN Server IP Address** to its default value.



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**.

The screenshot shows the 'IP Offices' configuration interface. On the left is a tree view of the system hierarchy. On the right, the 'Network Topology' tab is selected under the 'LAN2' section. The configuration fields are as follows:

Field	Value
STUN Server IP Address	69 . 90 . 168 . 13
Firewall/NAT Type	Open Internet
Binding Refresh Time (seconds)	0
Public IP Address	0 . 0 . 0 . 0
Public Port	0

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.

The screenshot shows the 'IP Offices' configuration interface with the 'SIP Registrar' tab selected. The configuration fields are as follows:

Field	Value
Domain Name	avaya.com
Layer 4 Protocol	Both TCP & UDP
TCP Port	5060
UDP Port	5060
Challenge Expiry Time (secs)	10
Auto-create Extn/User	<input type="checkbox"/>

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.

The screenshot shows the 'Voicemail' configuration tab for system 00E00705C035. The left sidebar shows a tree view with 'System (1)' selected. The main panel has three tabs: 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', and 'Sys'. The 'Voicemail' tab is active, showing the following settings:

Voicemail Type	Voicemail Lite/Pro
Voicemail Destination	
Voicemail IP Address	10 . 80 . 130 . 150

5.3.4. System Telephony Configuration

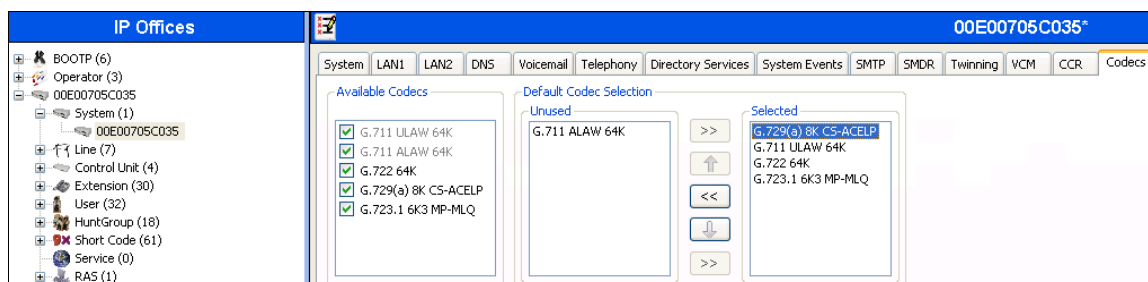
Navigate to **Telephony** → **Telephony** and check **ULAW** box under **Switch** in **Companding Law** and **ULAW Line** box under the **Line** in **Companding Law**.

The screenshot shows the 'Telephony' configuration tab for system 00E00705C035. The left sidebar shows a tree view with 'System (1)' selected. The main panel has tabs: 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', 'SMTP', 'SMDR', 'Twinning', 'VCM', 'CCR', and 'Codecs'. The 'Telephony' tab is active, showing the following settings:

Analog Extensions	
Default Outside Call Sequence	Normal
Default Inside Call Sequence	Ring Type 1
Default Ring Back Sequence	Ring Type 2
Restrict Analogue Extension Ringer Voltage	<input type="checkbox"/>
Dial Delay Time (secs)	4
Dial Delay Count	0
Default No Answer Time (secs)	15
Hold Timeout (secs)	120
Park Timeout (secs)	300
Ring Delay (secs)	5
Call Priority Promotion Time (secs)	Disabled
Default Currency	USD
Default Name Priority	Favor Trunk
Companding Law	
Switch	Line
<input checked="" type="radio"/> U-Law	<input checked="" type="radio"/> U-Law Line
<input type="radio"/> A-Law	<input type="radio"/> A-Law Line
<input type="checkbox"/> DSS Status	
<input type="checkbox"/> Auto Hold	
<input checked="" type="checkbox"/> Dial By Name	
<input checked="" type="checkbox"/> Show Account Code	
<input type="checkbox"/> Inhibit Off-Switch Forward/Transfer	
<input type="checkbox"/> Restrict Network Interconnect	
<input type="checkbox"/> Drop External Only Impromptu Conference	
<input type="checkbox"/> Visually Differentiate External Call	
<input type="checkbox"/> Unsupervised Analog Trunk Disconnect Handling	
<input checked="" type="checkbox"/> High Quality Conferencing	

5.3.5. Codecs

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



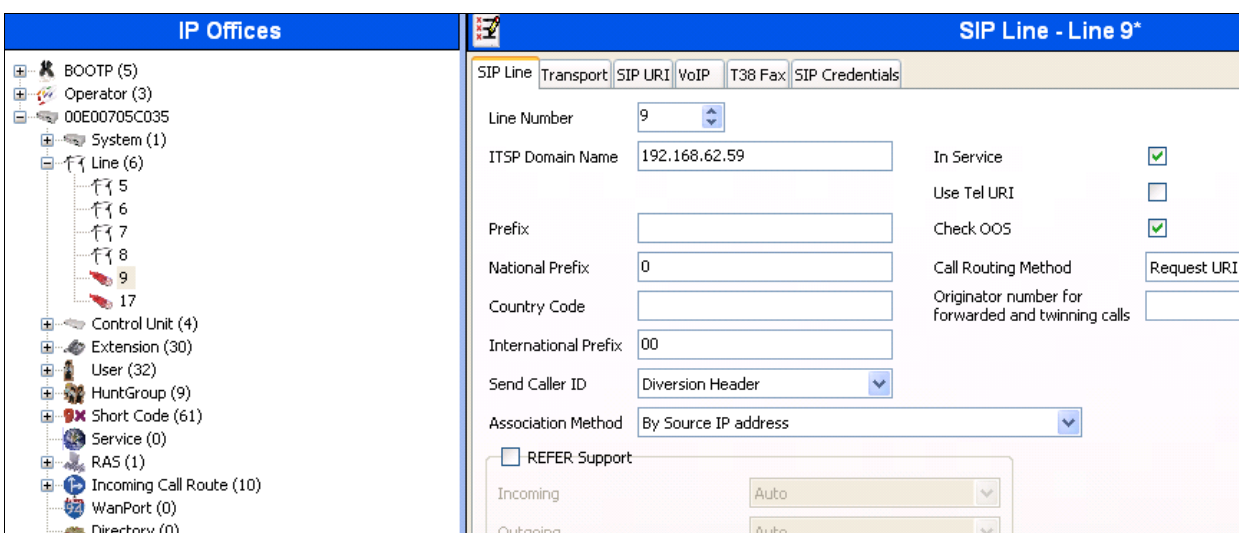
5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 8.0. 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 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.

The screenshot shows the 'SIP Line - Line 9*' configuration window with the 'Transport' tab selected. On the left, the 'IP Offices' tree shows a hierarchy: BOOTP (5), Operator (3), 00E00705C035, System (1), Line (6) (containing 5, 6, 7, 8, 9, 17), Control Unit (4), Extension (30), User (32), and HuntGroup (9). The 'Line (6)' folder is expanded, and a mouse cursor is pointing at the '9' entry. The main configuration area on the right has tabs for 'SIP Line', 'Transport' (selected), 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'ITSP Proxy Address' is set to '135.242.225.200'. The 'Network Configuration' section includes 'Layer 4 Protocol' set to 'UDP', 'Send Port' set to '5060', 'Use Network Topology Info' set to 'LAN 2', and 'Listen Port' set to '5060'. The 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

SIP Line - Line 9*	
SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials	
ITSP Proxy Address 135.242.225.200	
Network Configuration	
Layer 4 Protocol	UDP
Send Port	5060
Use Network Topology Info	LAN 2
Listen Port	5060
Explicit DNS Server(s)	0 . 0 . 0 . 0 0 . 0 . 0 . 0
Calls Route via Registrar	<input checked="" type="checkbox"/>
Separate Registrar	

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 'SIP Line - Line 9*' configuration window. The left pane shows the 'IP Offices' tree with 'Line (6)' expanded. The right pane shows the 'SIP URI' tab with a table of channels. Below the table is the 'Edit Channel' dialog.

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

Edit Channel

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. 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.

The screenshot shows the 'SIP Line - Line 9' configuration window with the 'VoIP' tab selected. The left pane shows a tree view of the system configuration, including 'IP Offices', 'BOOTP (6)', 'Operator (3)', 'System (1)', 'Line (7)', 'Control Unit (4)', 'Extension (30)', 'User (32)', 'HuntGroup (18)', 'Short Code (61)', 'Service (0)', 'RAS (1)', and 'Incoming Call Route (19)'. The main pane displays the 'VoIP' configuration for 'Line 9'. The 'Codec Selection' is set to 'System Default'. Below this, there are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.711 ALAW 64K'. The 'Selected' list contains 'G.729(a) 8K CS-ACELP', 'G.711 ULAW 64K', 'G.722 64K', and 'G.723.1 6K3 MP-MLQ'. There are buttons for moving items between the lists: '>>', '<<', and arrows. Below the lists, the 'Fax Transport Support' is set to 'T38', 'Call Initiation Timeout (s)' is set to '4', and 'DTMF Support' is set to 'RFC2833'. On the right side, there are several checkboxes: 'VoIP Silence Suppression' (checked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (checked), 'Codec Lockdown' (unchecked), and 'PRACK/100rel Supported' (unchecked).

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 screenshot displays the Avaya IP Office configuration 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). Under 'User (32)', a list of users is shown, with '217 Extn217' highlighted. The main panel on the right is titled 'Extn217: 217' and contains several tabs: 'User' (selected), 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', and 'Dial In'. The 'User' tab contains the following fields and options:

- Name: Extn217
- Password: [Empty field]
- Confirm Password: [Empty field]
- Full Name: [Empty field]
- Extension: 217
- Locale: [Empty field]
- 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)

Below these fields is a section titled 'User Rights' with the following fields:

- User Rights view: User data
- Working hours time profile: <None>
- Working hours User Rights: [Empty field]
- Out of hours User Rights: [Empty field]

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. On the left, a navigation tree shows the hierarchy: System (1), Line (6), Control Unit (4), and Extension (30). The extension '73 217' is selected and highlighted. The main panel, titled 'Digital Extension: 73 217', shows the configuration for this extension. The 'Extn' tab is active, displaying the following settings:


Field	Value
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 IP Office configuration 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, showing a list of 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 fields:

- Name: Extn501
- Password: *****
- Confirm Password: *****
- Full Name:
- Extension: 501
- Locale:
- 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

Below the main configuration fields is a section titled 'User Rights' with the following fields:

- User Rights view: User data
- Working hours time profile: <None>
- Working hours User Rights:
- Out of hours User Rights:

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.

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

Navigate to **Telephony** → **Call Settings** and 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 screenshot shows the IP Office configuration interface. On the left, the 'IP Offices' tree is expanded to 'User (32)', showing a list of extensions from 201 to 205. The main panel is titled 'Extn501: 501' and has tabs for 'User', 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', 'Voice Recording', and 'Bu'. The 'Telephony' tab is selected, and the 'Call Settings' sub-tab is active. The settings include:

- Outside Call Sequence: Default Ring
- Inside Call Sequence: Default Ring
- Ringback Sequence: Default Ring
- No Answer Time (secs): 15
- Wrap-up Time (secs): 2
- Transfer Return Time (secs): Off
- Call Cost Mark-Up: 100
- Call Waiting On: ☒ (checked)
- Answer Call Waiting On Hold: ☒ (checked)
- Busy On Held: ☐ (unchecked)
- Offhook Station: ☐ (unchecked)

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**.

The screenshot shows the IP Office configuration interface. On the left, the 'IP Offices' tree is expanded to 'Extension (30)', showing a list of extensions from 25 to 80. The main panel is titled 'H323 Extension: 8006 501' and has tabs for 'Extn' and 'VoIP'. The 'VoIP' tab is selected. The settings include:

- IP Address: 0 . 0 . 0 . 0
- MAC Address: 00 00 00 00 00 00
- Codec Selection: System Default
- Unused codecs: G.711 ALAW 64K
- Selected codecs: G.729(a) 8K CS-ACELP, G.711 ULAW 64K, G.722 64K, G.723.1 6K3 MP-MLQ
- VoIP Silence Suppression: ☐ (unchecked)
- Enable Faststart for non-Avaya IP phones: ☐ (unchecked)
- Out Of Band DTMF: ☒ (checked)
- Local Tones: ☐ (unchecked)
- Allow Direct Media Path: ☒ (checked)
- Reserve Avaya IP endpoint license: ☐ (unchecked)
- Reserve 3rd party IP endpoint license: ☐ (unchecked)
- TDM->IP Gain: Default
- IP->TDM Gain: Default
- Supplementary Services: None

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 screenshot displays the Avaya SIP Telephone User configuration interface. The left pane, titled "IP Offices", shows a hierarchical tree structure with the following nodes: BOOTP (5), Operator (3), 00E00705C035, System (1), Line (6), Control Unit (4), Extension (30), and User (32). The "User (32)" node is expanded, showing a list of users from "NoUser" to "501 Extn501". The right pane, titled "Extn520: 520*", contains the configuration details for User 520. The "User" tab is selected, showing fields for Name (Extn520), Password (*****), Confirm Password (*****), Full Name, Extension (520), Locale, Priority (5), System Phone Rights (None), and Profile (Basic User). Below these fields are checkboxes for "Receptionist", "Enable SoftPhone", "Enable one-X Portal Services", "Enable one-X TeleCommuter", and "Ex Directory". The "Device Type" is set to "Avaya 1140E Sip". The "User Rights" section includes fields for "User Rights view" (User data), "Working hours time profile" (<None>), "Working hours User Rights", and "Out of hours User Rights".

IP Offices		Extn520: 520*	
User (32)		User	
NoUser		Name: Extn520	
RemoteManager		Password: *****	
201 Extn201		Confirm Password: *****	
202 Extn202		Full Name:	
203 Extn203		Extension: 520	
204 Extn204		Locale:	
205 Extn205		Priority: 5	
206 Extn206		System Phone Rights: None	
207 Extn207		Profile: Basic User	
208 Extn208		<input type="checkbox"/> Receptionist	
209 Extn209		<input type="checkbox"/> Enable SoftPhone	
210 Extn210		<input type="checkbox"/> Enable one-X Portal Services	
211 Extn211		<input type="checkbox"/> Enable one-X TeleCommuter	
212 Extn212		<input type="checkbox"/> Ex Directory	
213 Extn213		Device Type: Avaya 1140E Sip	
214 Extn214		User Rights	
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

- BOOTP (6)
- Operator (3)
- 00E00705C035
- System (1)
- Line (7)
- Control Unit (4)
- Extension (30)
 - 25 201
 - 26 202
 - 27 203
 - 28 204
 - 29 205
 - 30 206
 - 31 207
 - 32 208
 - 49 209
 - 50 210
 - 51 211
 - 52 212
 - 53 213
 - 54 214
 - 55 215
 - 56 216
 - 73 217
 - 74 218
 - 75 219
 - 76 220
 - 77 221
 - 78 222
 - 79 223
 - 80 224
 - 8006 501
 - 8003 502
 - 8004 503
 - 8005 504
 - 8000 520**

SIP Extension: 8000 520

Extn VoIP T38 Fax

IP Address: 10 . 80 . 130 . 51

Codec Selection: System Default

Unused: G.711 ALAW 64K

Selected: G.729(a) 8K CS-ACELP, G.711 ULAW 64K, G.722 64K, G.723.1 6K3 MP-MLQ

Fax Transport Support: None

TDM->IP Gain: Default

IP->TDM Gain: Default

DTMF Support: RFC2833

☐ VoIP Silence Suppression
☐ Local Hold Music
☒ Allow Direct Media Path
☐ Re-invite Supported
☐ Use Offerer's Preferred Codec
☒ Reserve Avaya IP endpoint license
☐ 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** [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**.

The screenshot shows the 'Hunt Group' configuration window for 'Longest Waiting Group Receivables'. The 'Name' field is set to 'Receivables' and the 'Extension' field is set to '11'. The 'Ring Mode' is set to 'LongestWaiting' and the 'No Answer Time (secs)' is set to '40'. The 'Hold Music Source' is set to 'No Change' and the 'Agent's Status on No-Answer Applies To' is set to 'External Inbound Calls Only'. The 'User List' table shows the following data:

Extension	Name
<input type="checkbox"/> 202	Extn202
<input checked="" type="checkbox"/> 207	Extn207
<input type="checkbox"/> 217	Extn217
<input type="checkbox"/> 218	Extn218
<input checked="" type="checkbox"/> 501	Extn501
<input type="checkbox"/> 502	Extn502
<input type="checkbox"/> 503	Extn503
<input type="checkbox"/> 520	Extn520
<input checked="" type="checkbox"/> 521	Extn521

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.

The screenshot shows the 'Hunt Group' configuration window for 'Longest Waiting Group Receivables: 11*'. The 'Queuing' tab is selected. The 'Queuing On' checkbox is checked. The 'Queue Length' is set to '3' and the 'Normalize Queue Length' checkbox is checked. The 'Queue Type' is set to 'Assign Call On Agent Answer'. The 'Calls In Queue Alarm' section shows the 'Calls In Queue Threshold' set to '1' and the 'Analog Extension to Notify' set to '<None>'. The 'User List' table is also visible, showing the same data as the previous screenshot.

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, the 'IP Offices' tree shows a hierarchy including BOOTP (5), Operator (3), System (1), Line (6), Control Unit (4), Extension (30), User (32), and HuntGroup (9). The HuntGroup (9) is expanded, showing sub-groups like 13 Billing, 14 CustomerService, 200 Main, 5009 NSN5009, 5010 NSN5010, 12 Payables, and 11 Receivables. The right pane shows the configuration for the 'Longest Waiting Group Receivables: 11*' hunt group, with 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 illustrates the announcement sequence: Play 1st announcement → Post announcement tone → Play 2nd announcement → Repeat last announcement (looping back to Play 2nd announcement).

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 Offices' configuration window. On the left, a list of short codes is shown, with '*93' selected. The right pane shows the configuration for '*93: Voicemail'. The 'Short Code' tab is active, and the following fields are visible:

Field	Value
Code	*93
Feature	Voicemail Collect
Telephone Number	"CallCenter"
Line Group Id	0
Locale	
Force Account Code	<input type="checkbox"/>

5.6.2. Voicemail Retrieval Code

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

The screenshot displays the 'Short Code' configuration window for '*17: Voicemail Collect'. The 'Short Code' tab is active, and the following fields are visible:

Field	Value
Code	*17
Feature	Voicemail Collect
Telephone Number	?U
Line Group Id	0
Locale	
Force Account Code	<input type="checkbox"/>

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	
BOOTP (5)		Standard	Voice Recording
Operator (3)		Bearer Capability	Any Voice
00E00705C035		Line Group Id	110
System (1)		Incoming Number	0000011001
Line (6)		Incoming Sub Address	
Control Unit (4)		Incoming CLI	
Extension (30)		Locale	
User (32)		Priority	1 - Low
HuntGroup (9)		Tag	
Short Code (61)		Hold Music Source	System Source
Service (0)			
RAS (1)			
Incoming Call Route (10)			
0			
110 0000011001			
110 0000021002			
110 0000031003			

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.

110 0000011001*		
Standard	Voice Recording	Destinations
TimeProfile	Destination	Fallback Extension
Default Value	502 Extn502	
	502 Extn502	
	503 Extn503	
	521 Extn521	
	504 Extn504	
	200 Main	
	4893 SonusTDM4893	
	4894 SonusTDM4894	
	5009 NSN5009	

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	

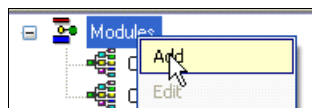
The following screen displays another mechanism to access the Call Center functionality without using the short code. The Call Center functionality is configured in Voicemail Pro as detailed in **Section 5.8**.

TimeProfile	Destination	Fallback Extension
Default Value	VM:CallCenter	

5.8. Call Center Provisioning in Voicemail Pro

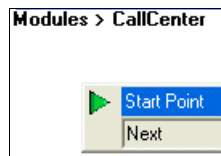
The call center functionality was configured in Voicemail Pro. Following steps highlight the configuration of this functionality. For further information, consult [IPO-VMPRO].

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

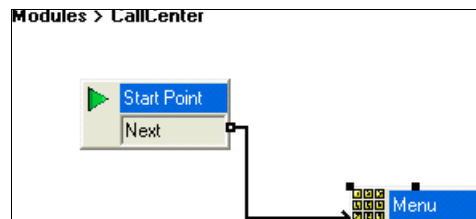


2. In the screen below, enter **CallCenter** in the **Name** field and click OK.

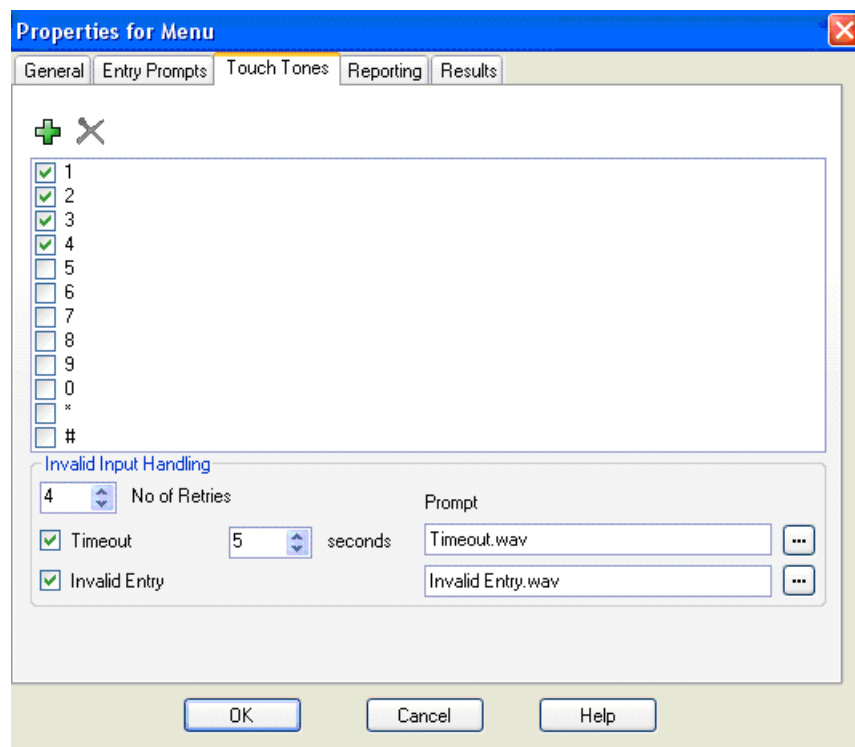
3. Following screen is displayed indicating the starting point for the Call Center functionality.



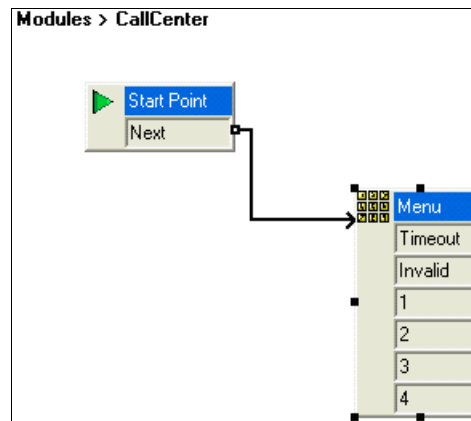
4. Under the **Actions** tab, select **Basic Actions** [not shown]. Select **Menu** and place it on the right side of the pane and then connect the **Start Point** to **Menu** as shown below:



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:

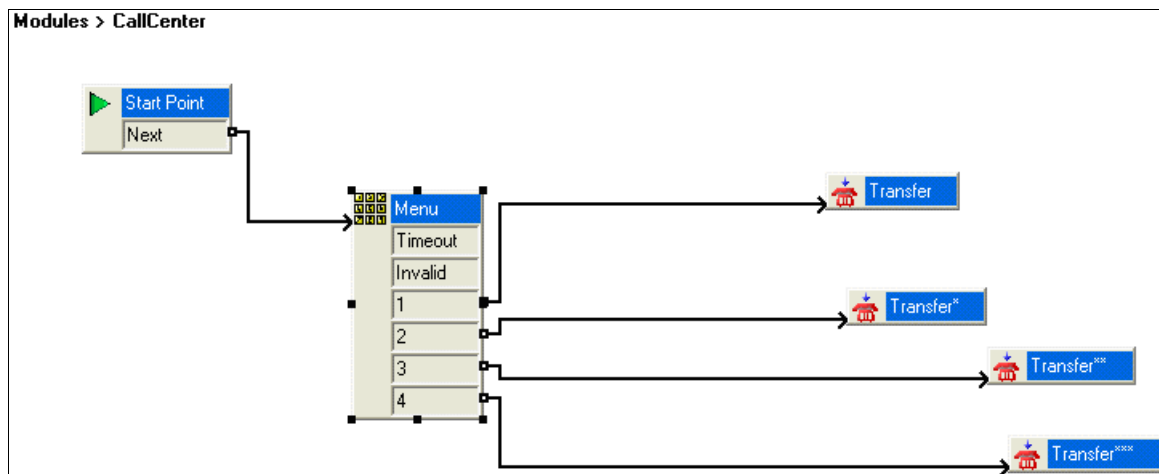


7. Right click on the **Menu** action and select the **Entry Prompts** tab and click on the **+** sign [not shown]. On the following screen, enter the **Extension** where the recording is done and the filename for the recording in the **Please select a file or enter a new file name** field and press 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.

The screenshot shows the "Wave Editor" dialog box. It has a title bar with a close button. The main area contains the following fields and controls:

- Use which media device?**: A dropdown menu set to "Telephony Handset".
- Extension**: A text field containing "203".
- Please select a file or enter a new file name**: A text field containing "Start.wav" and a browse button (...).
- Relative to:** "C:\Program Files\Avaya\NP Office\Voicemail Pro\WM\Wavs\"
- Wave Information**: A section with the following fields:
 - Wave Length: 9.3 seconds
 - Sample: 16 bit
 - Sample Rate: 8 KHz
 - Channels: Mono
- Controls**: A play button (green triangle), a stop button (grey square), and a record button (red circle with a white dot).
- Buttons**: "Close" and "Help" buttons at the bottom.

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.

The screenshot shows a dialog box titled "Properties for Transfer" with a close button (X) in the top right corner. The dialog has five tabs: "General", "Entry Prompts", "Specific" (which is selected), "Reporting", and "Results". The "Specific" tab contains the following fields and options:

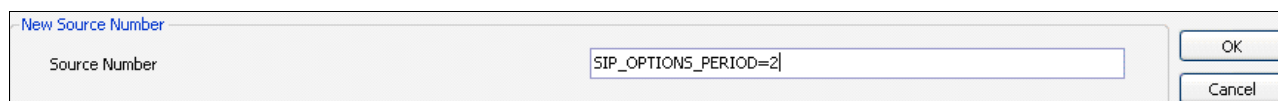
- Transfer call to** (header)
- Destination**: A text field containing "11" and a button with three dots.
- Source of transfer (displayed on phone)**: A text field and a button with three dots.
- Description (displayed on phone)**: A text field and a button with three dots.
- ☐ **Set Caller Priority**: A checkbox with a dropdown menu set to "Low".
- ☐ **Notify Caller of Transfer to Target**: A checkbox.

At the bottom of the dialog are three buttons: "OK", "Cancel", and "Help".

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_PAID_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

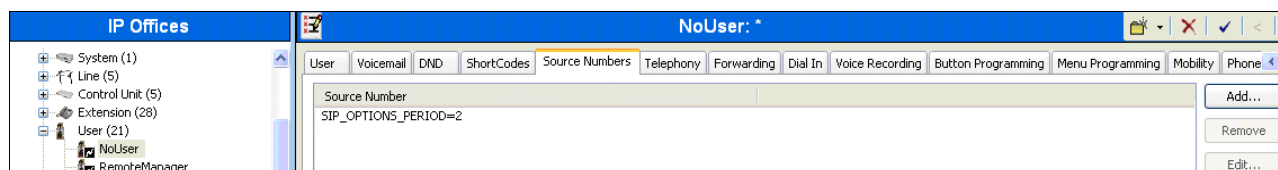
Source Number

SIP_OPTIONS_PERIOD=2

OK

Cancel

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



IP Offices

System (1)

Line (5)

Control Unit (5)

Extension (28)

User (21)

NoUser

RemoteManager

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

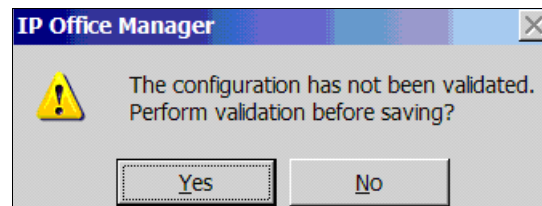
Add...

Remove

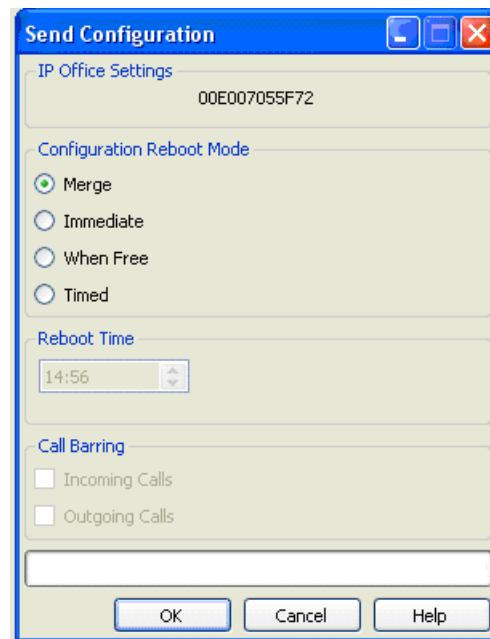
Edit...

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:

- 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 Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start→Programs→IP Office→System Status** on the PC where Avaya IP Office Manager 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 Avaya 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 Avaya IP Office Manager is installed.

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 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 8.0 Installation Manual, Issue 25d, April 9, 2012
Document Number 15-601042
<https://downloads.avaya.com/css/P8/documents/100150370>

[IPO-MGR] IP Office Release 8.0 Manager 10.0 Issue 28n, May 21, 2012
Document Number 15-601011
<https://downloads.avaya.com/css/P8/documents/100150305>

[IPO-SYSSTAT] IP Office Release 6.0 System Status Application, Issue 06b, November 12, 2011
Document Number 15-601758
<http://downloads.avaya.com/css/P8/documents/100150298>

[IPO-VMPRO] IP Office Release 8.0 Voicemail Pro Administration, Issue 27b, April 6, 2012
Document Number 15-601063
<https://downloads.avaya.com/css/P8/documents/100153495>

[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/>

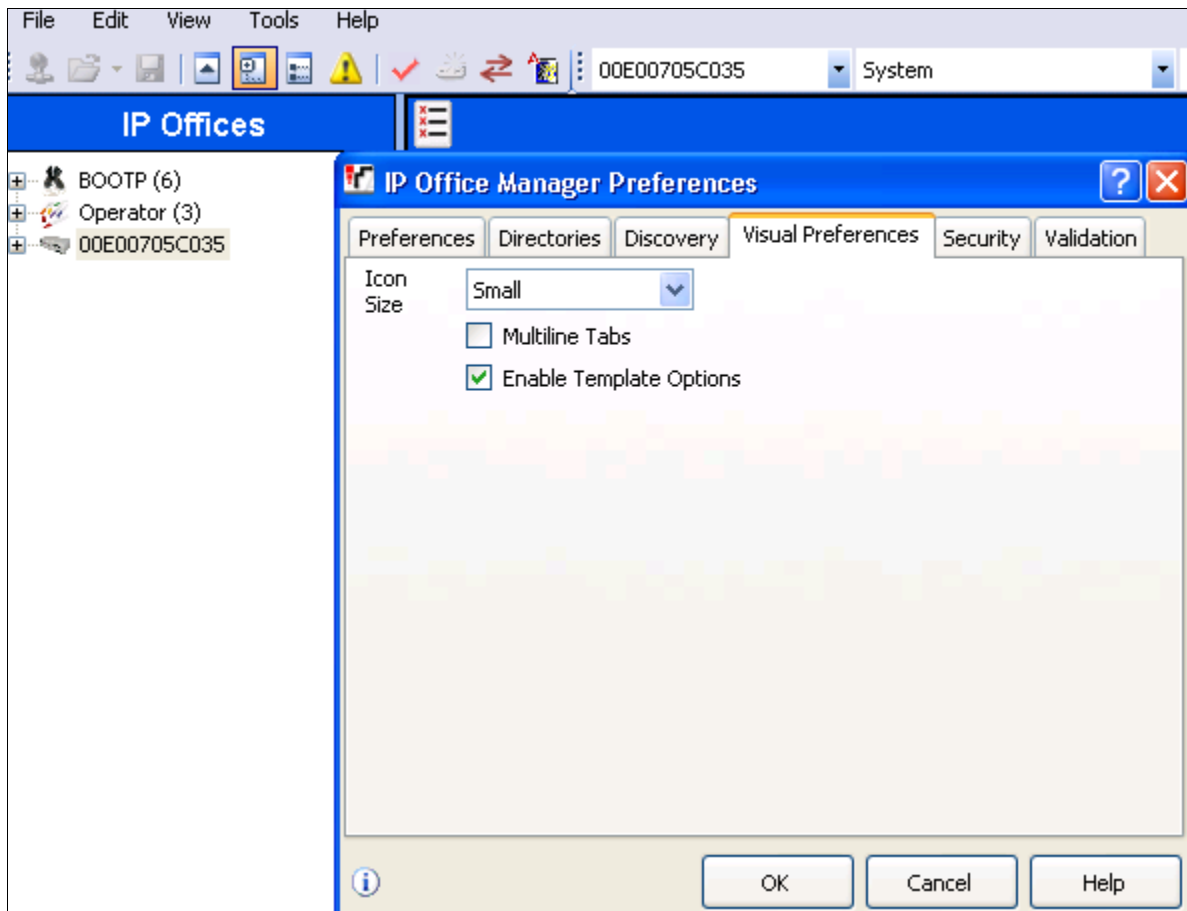
9. Appendix – Example SIP Trunk Template

IP Office Release 8.0 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.

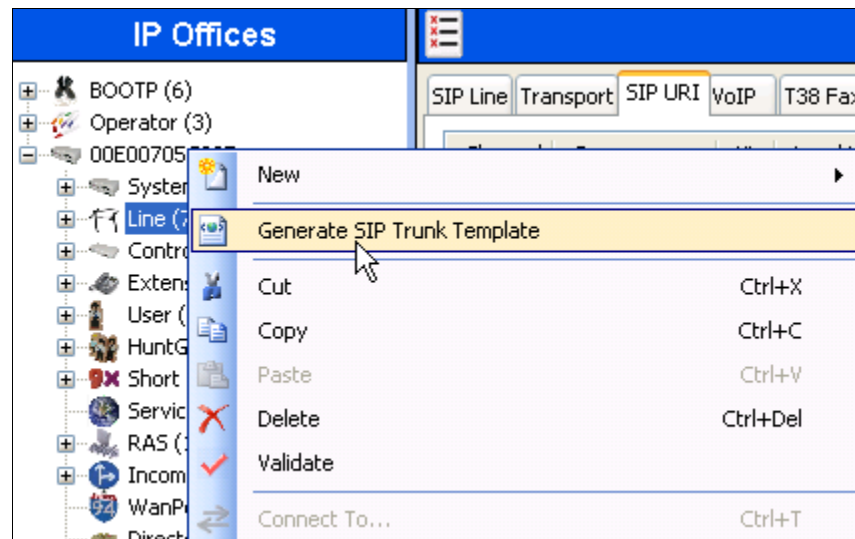


2. Run **regedit** on the desktop and navigate to **HKEY_CURRENT_USER/Software/IP400/Manager** and add a **DWORD** value **TemplateProvisioning** and set its value to **1**.

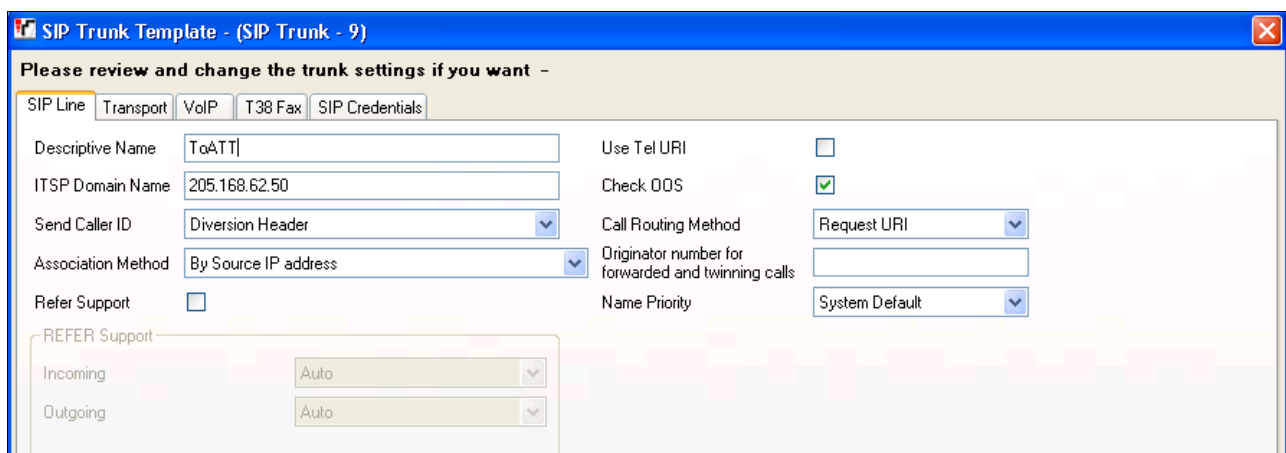
9.2. Generate a SIP Trunk Template

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

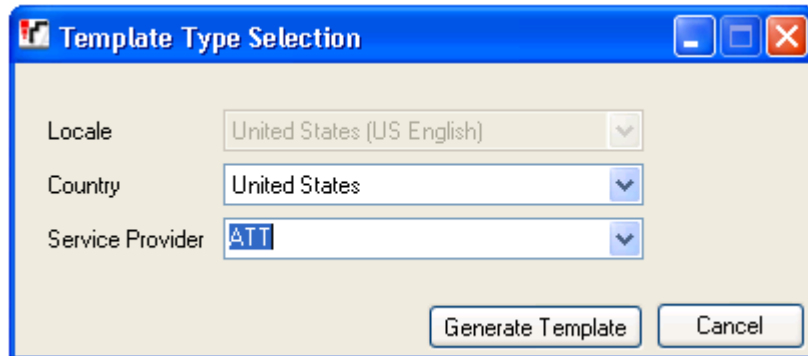
1. Select the SIP trunk under line and right click on the SIP line numbered for which the SIP trunk template is to be generated and then click **Generate SIP Trunk Template**.



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**.



The screenshot shows a 'Template Type Selection' window with the following settings:

- Locale: United States (US English)
- Country: United States
- Service Provider: ATT

Buttons: Generate Template, Cancel

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 version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
<TemplateType>SIPTrunk</TemplateType>
<Version>20120521</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>ToATT</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>
<ITSPProxy>135.242.225.200</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>false</UseAdvVoiceCodecPrefs>
<CallInitiationTimeout>4</CallInitiationTimeout>
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
```

```

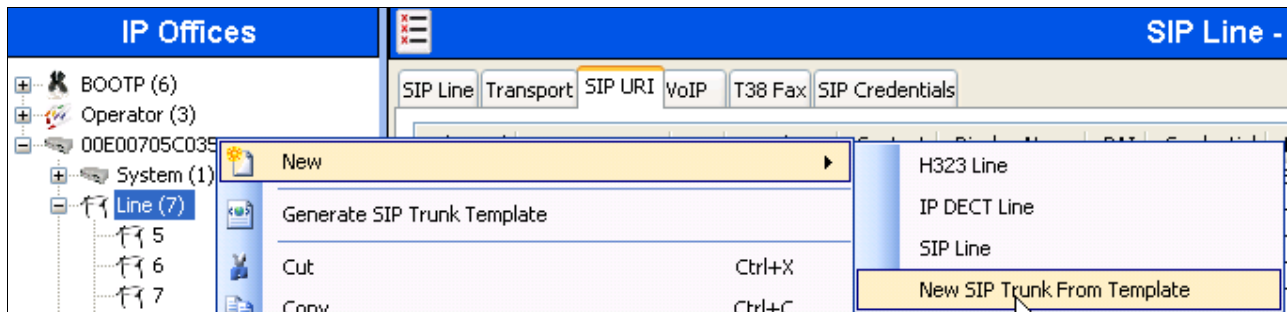
<VoipSilenceSupression>true</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_T38</FaxTransportSupport>
<UseOffererPrefferedCodec>true</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown>
<Rel100Supported>false</Rel100Supported>
<T38FaxVersion>3</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**.



Template Type Selection

Locale: United States (US English)

Country: United States

Service Provider: ATT

☐ Display All

Create new SIP Trunk Cancel

©2012 Avaya Inc. All Rights Reserved.

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

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