



Avaya Solution & Interoperability Test Lab

Applications Notes for Avaya IP Office 6.1 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

Abstract

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

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

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

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1. Introduction

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

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

2. General Test Approach and Test Results

The test environment consisted of:

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

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

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

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound and outbound call flows (see **Section 3.2** for examples) between 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 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. G.711 faxing is not supported between IP Office and the AT&T IP Toll Free service. T.38 faxing is supported, as is Group 3 and Super Group 3 fax.
2. AT&T IP Transfer connect option of the AT&T IP Toll Free service was not verified with Avaya IP Office 6.1 and hence not supported.
3. Shuffling is not supported for SIP trunks in Avaya IP Office 6.1.
4. G.726 codec is not supported Avaya IP Office 6.1.

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. This solution is extensible to other Avaya IP Office hardware too.
- Avaya “desk” phones are represented with Avaya 1616, 4625 and 9630 IP Telephones running H.323 software, Avaya 1416 digital phone, 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.

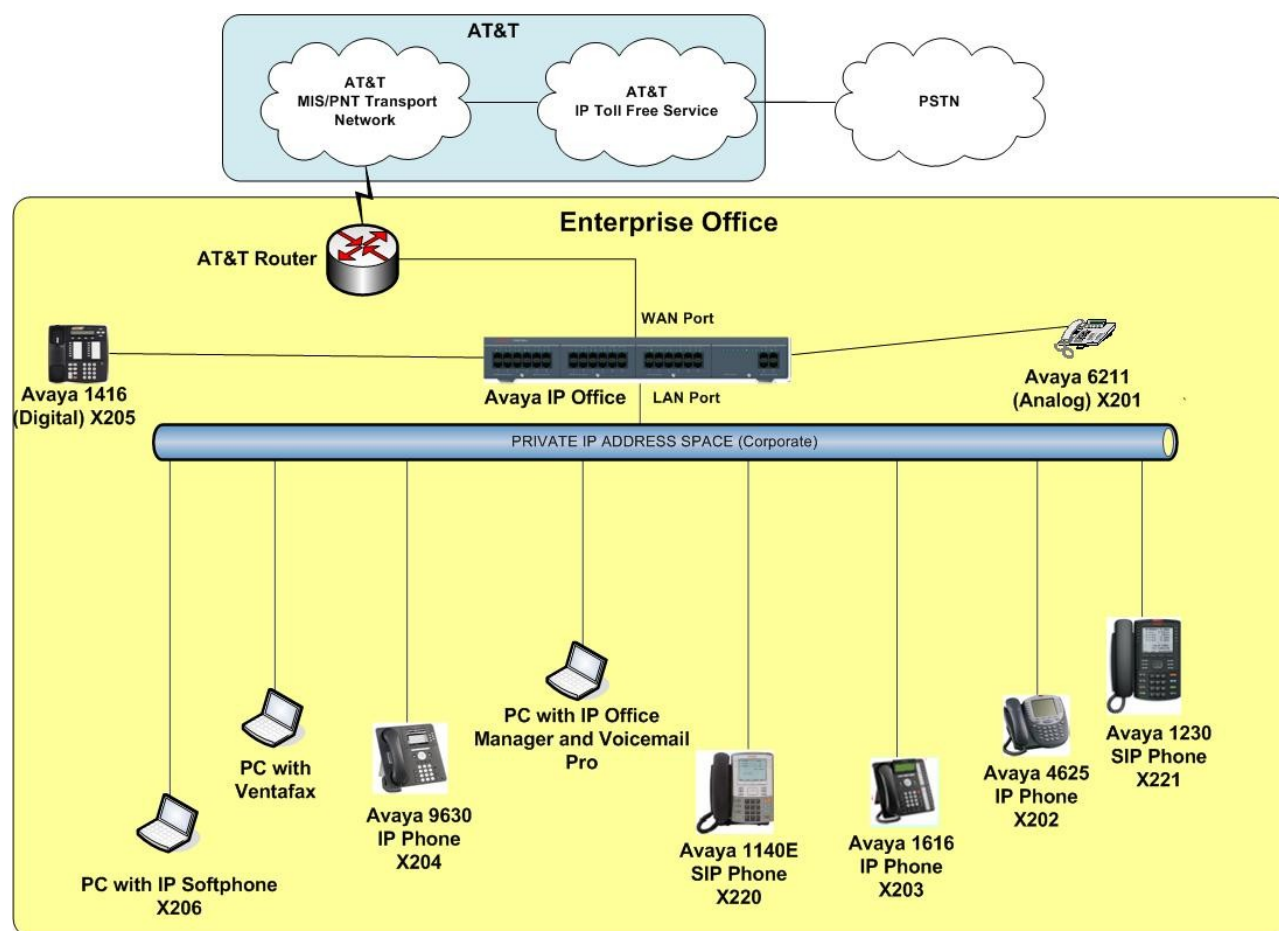


Figure 1: Reference configuration

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 addresses shown in this document are examples. 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	192.168.62.56
Private IP Address	10.80.130.56
Avaya IP Office Extensions	201 = Analog 202,203,204=H323 205=Digital 206=Softphone 220,221= SIP phones
AT&T IP Toll Free Service	
Border Element IP Address	135.242.225.200
Digits passed in SIP-URI Request	000001017 to 000001021

Table 1: Illustrative Values Used in these Application Notes

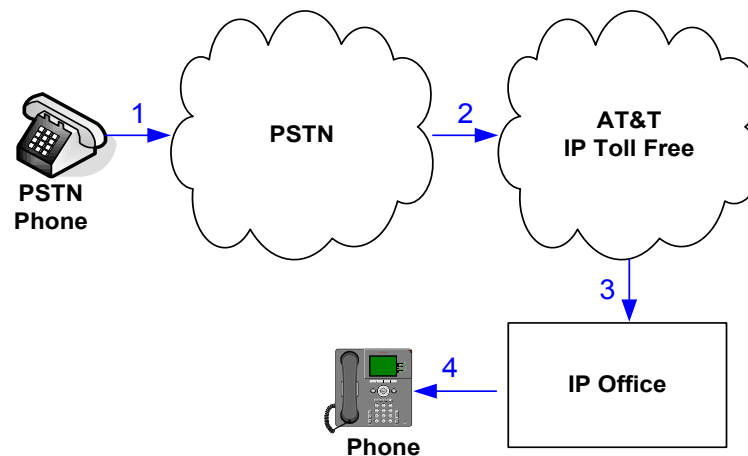
3.2. Call Flows

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

3.2.1. Inbound

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

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

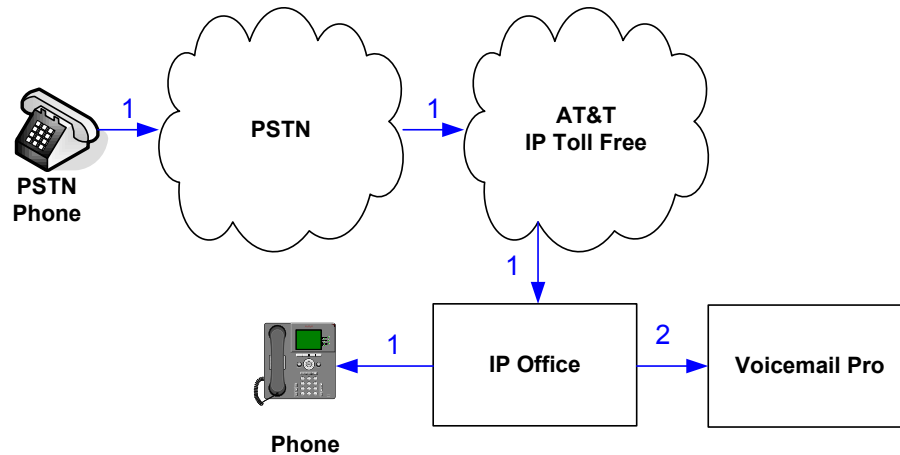


Inbound - AT&T IP Toll Free

3.2.2. Coverage to Voicemail

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

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



Coverage to Voicemail

4. Equipment and Software Validated

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

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

Component	Version
Avaya IP Office 500	Release 6.1 (6.1.5) (Preferred Edition)
Avaya IP Office Manager	Release 8.1 (8.1.5) (Preferred Edition)
Avaya IP Office Voicemail Pro	Release 6.1.16
Avaya IP Office Voicemail Pro Client	Version 6.1 (16)
Avaya 1616IP-Series Telephones (H.323)	Release 1.3
Avaya 9630 IP Telephone	Avaya one-X® Deskphone Edition H.323 Version S3.11
Avaya 4625SW IP Telephone	a25d01a2_9_1.bin
Avaya IP Office Softphone	Release 3.1.2.17 59616
Avaya 1416 Digital Telephone	-
Avaya 6211 Analog phone	-
Avaya 1140E SIP Telephone	04.00.04.00 (SIP1140)
Avaya 1230 SIP Telephone	04.00.04.00 (SIP1230)
Fax device	Ventafax Home Version 6.2
AT&T IP Toll Free Service using MIS/PNT transport service connections.	VNI 18

Table 2: Equipment and Software Versions

5. Avaya IP Office

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

WELCOME to IP Office Administration

What would you like to do ?

[Create an Offline Configuration](#)

[Open Configuration from System](#)

[Read a Configuration from File](#)

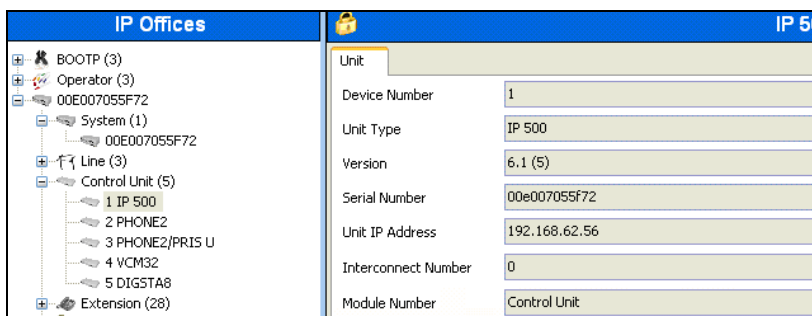
Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left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

5.1. Physical, Network, and Security Configuration

This section describes attributes of the reference configuration, but is not meant to be prescriptive. Consult reference [IPO-INSTALL] for more information on the topics in this section.

In the reference configuration, the IP Office 500 contains a Phone2 (analog), Phone2/PRIS U (Combo module), VCM32, and a DIGSTA8 (Digital) modules. The VCM32 is a Voice Compression Module supporting VoIP codecs. The Analog module (Phone2) was used in this reference configuration to support analog telephones or fax machines. The Digital module (DIGSTA8) was used to support digital phone extensions.

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



In the sample configuration, the IP Office LAN1 port (labeled as WAN in Figure 1) is physically connected to the public network at the IP Office customer site. The default gateway for this network is 192.168.62.1.

1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New**. 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 **Destination** LAN1 (Refer **Section 5.3.2**).

The screenshot displays the IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree structure including BOOTP (3), Operator (3), 00E007055F72, System (1), 00E007055F72, Line (3), Control Unit (5), Extension (28), User (20), HuntGroup (10), Short Code (65), Service (0), RAS (1), Incoming Call Route (11), WanPort (0), Directory (0), Time Profile (0), Firewall Profile (1), and IP Route (3). The 'IP Route' item is selected, showing a sub-item '0.0.0.0'. The main area on the right is titled '0.0.0.0*' and contains the 'IP Route' configuration details. The fields are: IP Address (0 . 0 . 0 . 0), IP Mask (0 . 0 . 0 . 0), Gateway IP Address (192 . 168 . 62 . 1), Destination (LAN1), and Metric (0). There is a checkbox for 'Proxy ARP' which is currently unchecked. A mouse cursor is visible over the main configuration area.

IP Route	
IP Address	0 . 0 . 0 . 0
IP Mask	0 . 0 . 0 . 0
Gateway IP Address	192 . 168 . 62 . 1
Destination	LAN1
Metric	0
<input type="checkbox"/> Proxy ARP	

2. Another route was added for the enterprise side (LAN2) as shown in the screen below. All the IP devices were part of this 10.80.130.x network in this reference configuration.

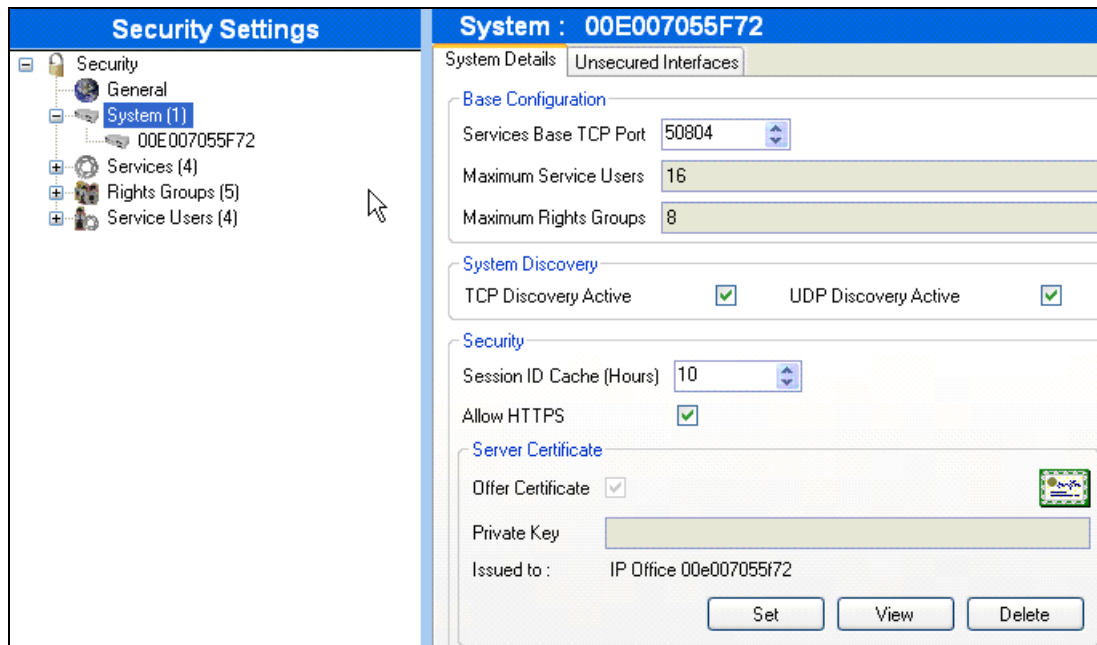
The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view under 'IP Offices' shows a hierarchy starting with 'BOOTP (3)', followed by 'Operator (3)', and then a specific office '00E007055F72'. Under this office, various components are listed, including 'System (1)', 'Line (3)', 'Control Unit (5)', 'Extension (28)', 'User (20)', 'HuntGroup (10)', 'Short Code (65)', 'Service (0)', 'RAS (1)', 'Incoming Call Route (11)', 'WanPort (0)', 'Directory (0)', 'Time Profile (0)', 'Firewall Profile (1)', and 'IP Route (3)'. The 'IP Route (3)' item is selected, and its configuration is shown on the right. The configuration includes fields for 'IP Address' (10 . 80 . 130 . 0), 'IP Mask' (255 . 255 . 255 . 0), 'Gateway IP Address' (10 . 80 . 130 . 1), 'Destination' (LAN2), and 'Metric' (0). A 'Proxy ARP' checkbox is also present and is unchecked.

3. For use of Avaya IP Office Softphone, https was enabled in this reference configuration. To check whether https is enabled, navigate to **File → Advanced → Security Settings** and login in with proper credentials in the screen below.

The screenshot shows a 'Security Service User Login' dialog box. It contains the following fields and controls:

- IP Office :** 00E007055F72 - IP 500
- Service User Name:** security
- Service User Password:** A field with masked characters (dots).
- Buttons:** OK, Cancel, and Help.

4. After logging in, select **System** from the Navigation pane and the appropriate IP Office system. In the Details pane, select the **System Details** tab. Verify that **Allow HTTPS** is checked. If not, check the box, click **OK**, and heed the on-screen prompts and warnings. Note that this action may be service disrupting.

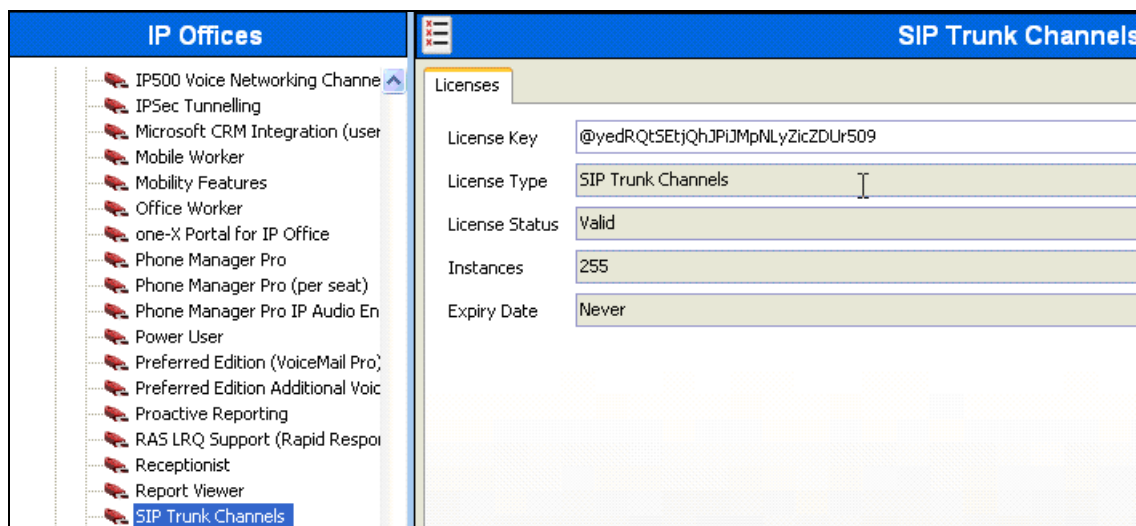


5. When complete, select **File → Configuration** to return to configuration activities.

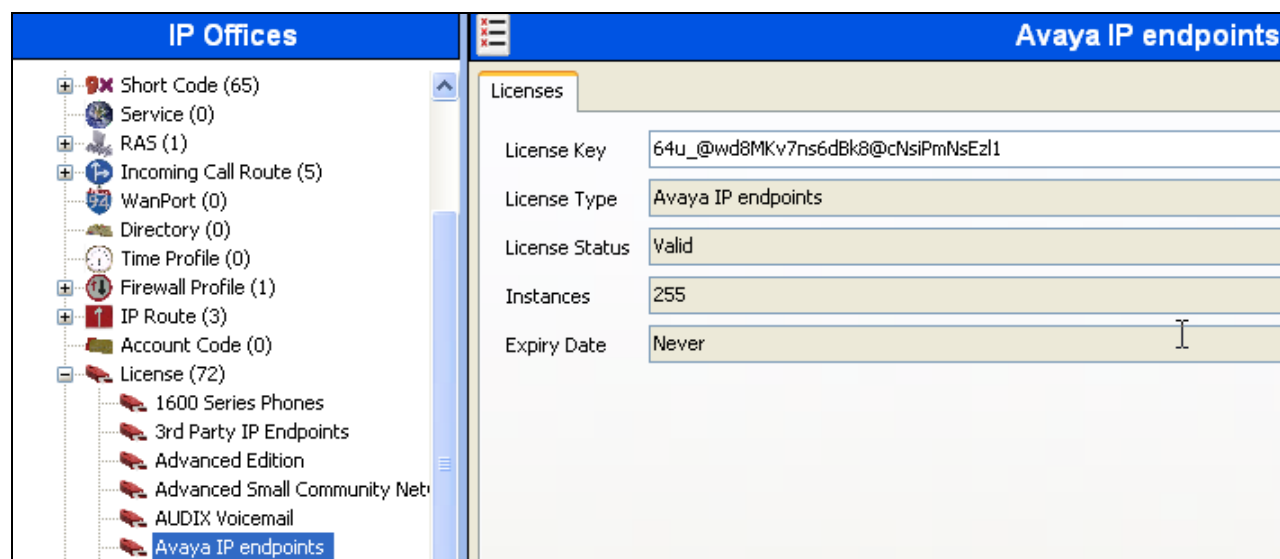
5.2. Licensing

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 there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient “Instances” (trunk channels) in the Details pane.



2. If Avaya IP Telephones will be used, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient “Instances” in the Details pane.



3. The following screen shows the availability of a valid license for **Power User** features. In the sample configuration, the user with extension **203** will be configured as a **Power User** and will be capable of using the IP Office Softphone.

IP Offices		Power User											
<ul style="list-style-type: none">IP500 Voice Networking ChannelIPSec TunnellingMicrosoft CRM Integration (user)Mobile WorkerMobility FeaturesOffice Workerone-X Portal for IP OfficePhone Manager ProPhone Manager Pro (per seat)Phone Manager Pro IP Audio EnPower User		<div>Licenses</div> <table><tr><td>License Key</td><td>0UH34zyFLNsSnLsW1ZM_g@datEVXYMr9</td></tr><tr><td>License Type</td><td>Power User</td></tr><tr><td>License Status</td><td>Valid</td></tr><tr><td>Instances</td><td>255</td></tr><tr><td>Expiry Date</td><td>Never</td></tr></table>		License Key	0UH34zyFLNsSnLsW1ZM_g@datEVXYMr9	License Type	Power User	License Status	Valid	Instances	255	Expiry Date	Never
License Key	0UH34zyFLNsSnLsW1ZM_g@datEVXYMr9												
License Type	Power User												
License Status	Valid												
Instances	255												
Expiry Date	Never												

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

The screenshot displays the 'IP Offices' configuration interface. On the left, a tree view under 'IP Offices' shows a hierarchy: BOOTP (3), Operator (3), 00E007055F72, System (1), and 00E007055F72. The 'System' tab is selected in the top right. The main configuration area on the right contains the following fields and settings:

- Name:** 00E007055F72
- Contact Information:** A section with a text box labeled '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:** Custom (dropdown menu)
- 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 (dropdown menu)

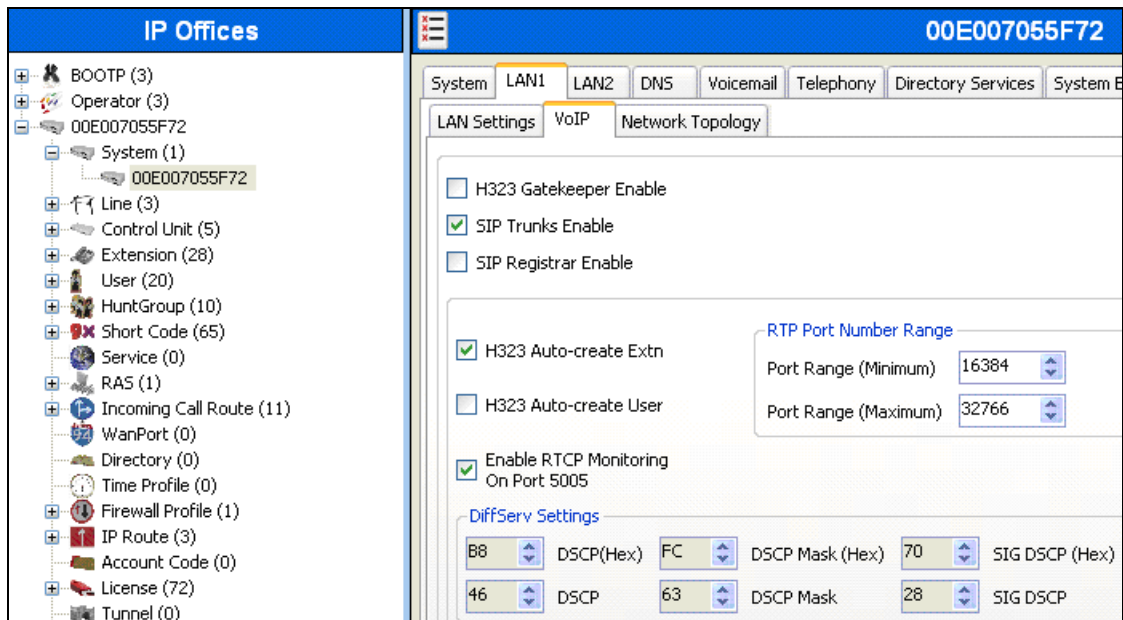
5.3.2. LAN Settings

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

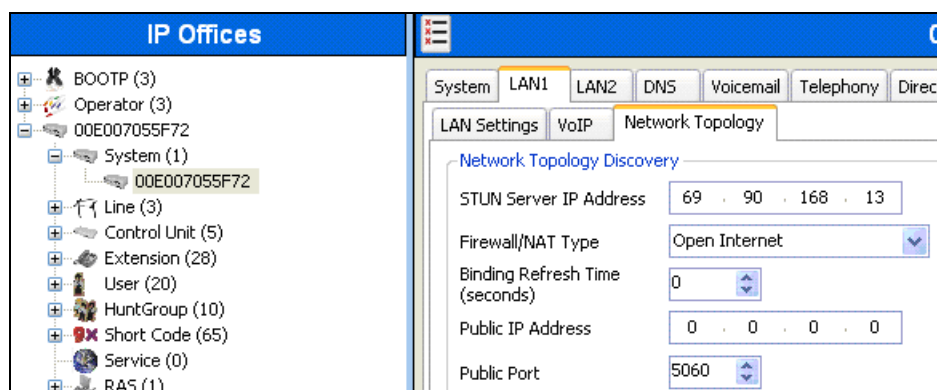
1. Select the **LAN1** tab followed by the **LAN Settings** tab and configure as follows:
 - **IP Address** – Set to **192.168.62.56** which is the IP address of IP Office known to AT&T network
 - **IP Mask** – Set to a valid value e.g. 255.255.255.0
 - **Primary Trans. IP Address** – Set to **0.0.0.0**
 - **DHCP Mode** – Select the **Disabled** radio button

The screenshot displays the IP Office configuration interface. On the left, a tree view under 'IP Offices' shows a hierarchy: BOOTP (3), Operator (3), 00E007055F72, System (1), 00E007055F72, Line (3), Control Unit (5), Extension (28), User (20), HuntGroup (10), Short Code (65), Service (0), RAS (1), Incoming Call Route (11), WanPort (0), and Directory (0). The '00E007055F72' office is selected. The right pane shows the configuration for this office, with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, and System Events. The 'LAN1' tab is active, and within it, the 'LAN Settings' sub-tab is selected. The configuration fields are as follows: IP Address is 192 . 168 . 62 . 56; IP Mask is 255 . 255 . 255 . 0; Primary Trans. IP Address is 0 . 0 . 0 . 0; RIP Mode is set to 'None' in a dropdown menu; there is an unchecked checkbox for 'Enable NAT'; and the Number Of DHCP IP Addresses is set to 200 in a spinner. At the bottom, the 'DHCP Mode' section has four radio buttons: Server, Client, Dialin, and Disabled. The 'Disabled' radio button is selected. An 'Advanced' button is located at the bottom right of the configuration pane.

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 requires an even number to be entered in this field



3. Select the **Network Topology** tab as shown in the following screen and configure as follows:
 - **Firewall/NAT Type** - Select **Open Internet** from the drop-down list. With this configuration, STUN will not be used but make sure to leave **STUN Server IP Address** to its default value.
 - **Public IP Address** – Set to 0.0.0.0 (default)
 - **Public Port** – Set to 5060 (default)



4. Select the **LAN2** tab followed by the **LAN Settings** tab and set **IP Address** of the IP Office on the enterprise side to **10.80.130.56** and **IP Mask** to **255.255.255.0**.

The screenshot shows the IP Office configuration interface. On the left is a tree view of the system components. The main panel on the right is titled '00E00' and has several tabs: System, LAN1, LAN2 (selected), DNS, Voicemail, Telephony, and Directory Services. Under the LAN2 tab, there are sub-tabs: LAN Settings (selected), VoIP, Network Topology, and SIP Registrar. The LAN Settings tab contains the following configuration fields:

- IP Address: 10 . 80 . 130 . 56
- IP Mask: 255 . 255 . 255 . 0
- Primary Trans. IP Address: 0 . 0 . 0 . 0
- Firewall Profile: <None>
- RIP Mode: None
- Enable NAT: ☐
- Number Of DHCP IP Addresses: 200
- DHCP Mode: ☐ Server ☐ Client ☐ Dialin ☒ Disabled

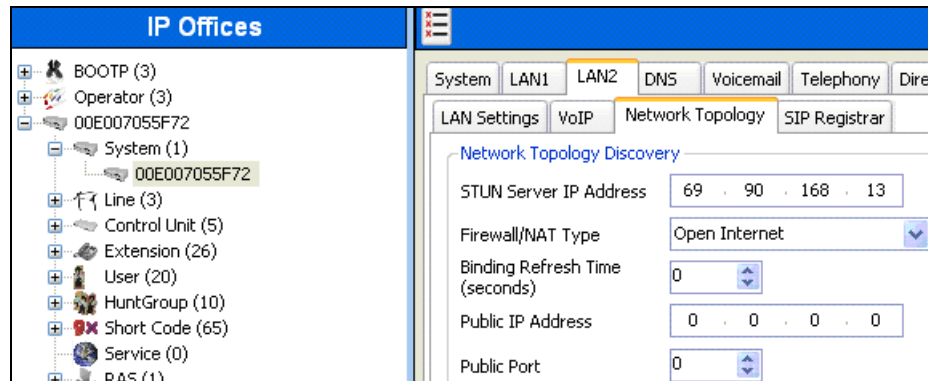
5. Select the **VoIP** tab as shown in the following screen and configure as follows:
 - **H323 Gatekeeper Enable** – Check this box to allow the use of Avaya IP Phones
 - **SIP Registrar Enable** – Check this box to allow SIP phones and IP Office Softphone usage

The screenshot shows the IP Office configuration interface with the 'VoIP' tab selected under the 'LAN2' tab. The main panel is titled '00E007055F72'. The configuration fields are as follows:

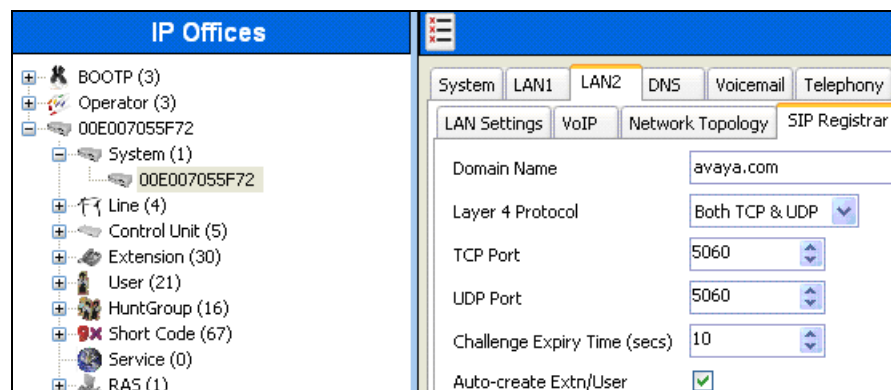
- ☒ H323 Gatekeeper Enable
- ☐ SIP Trunks Enable
- ☒ SIP Registrar Enable
- ☒ H323 Auto-create Extn
- ☐ H323 Auto-create User
- Enable RTCP Monitoring On Port 5005: ☒
- RTP Port Number Range:
 - Port Range (Minimum): 49152
 - Port Range (Maximum): 53246
- DiffServ Settings:

88	DSCP(Hex)	FC	DSCP Mask (Hex)	88	SIG DSCP (Hex)
46	DSCP	63	DSCP Mask	34	SIG DSCP

6. The Network Topology screen is set the same as it was set for LAN1.



7. Set the **Domain Name** field to **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

The settings presented here simply illustrate this reference configuration and are not intended to be prescriptive. Select **Voicemail** tab and configure as follows:

- **Voicemail Type** – Set to **Voicemail Lite/Pro** from the drop-down list
- **Voicemail IP Address** – Set to **10.80.130.152**, the IP Address of the PC running the Voicemail Pro software.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree shows a hierarchy starting with '00E007055F72' and 'System (1)', leading to the selected office '00E007055F72'. The right pane shows the configuration for this office, with the 'Voicemail' tab selected. The settings are as follows:

System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	Sy
Voicemail Type: Voicemail Lite/Pro							
Voicemail Destination: [Empty]							
Voicemail IP Address: 10 . 80 . 130 . 152							
Backup Voicemail IP Address: 0 . 0 . 0 . 0							
Voicemail Channel Reservation							
Unreserved Channels: 259							
Auto-Attendant: 0 Voice Recording: 0 Mandatory Voic							
Announcements: 0 Mailbox Access: 0							
DTMF Breakout							
Reception / Breakout (DTMF 0): [Empty]							
Breakout (DTMF 2): [Empty]							
Breakout (DTMF 3): [Empty]							

5.3.4. System Telephony Configuration

The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. Select the **Telephony** tab and **Telephony** sub-tab as shown and configure as follows:

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

The screenshot displays the Avaya System Manager configuration interface for System 00E007055F72. The left sidebar shows a tree view of system components, including IP Offices, BOOTP, Operator, System, Line, Control Unit, Extension, User, HuntGroup, Short Code, Service, RAS, Incoming Call Route, WanPort, Directory, Time Profile, Firewall Profile, IP Route, Account Code, License, Tunnel, User Rights, ARS, RAS Location Request, and E911 System. The main panel is titled '00E007055F72' and contains several tabs: System, LAN1, LAN2, DNS, Voicemail, Telephony (selected), Directory Services, System Events, SMTP, SMDR, and Twinning. The Telephony tab is further divided into sub-tabs: Telephony, Tones & Music, and Call Log. The 'Analogue Extensions' section includes settings for Default Outside Call Sequence (Normal), Default Inside Call Sequence (Ring Type 1), and Default Ring Back Sequence (Ring Type 2). The 'Companding Law' section shows two columns: 'Switch' and 'Line'. In the 'Switch' column, the 'ULAW' radio button is selected, and in the 'Line' column, the 'ULAW Line' radio button is selected. Other settings include Dial Delay Time (secs) set to 4, Dial Delay Count set to 0, Default No Answer Time (secs) set to 15, Hold Timeout (secs) set to 120, Park Timeout (secs) set to 300, Ring Delay (secs) set to 5, Call Priority Promotion Time (secs) set to Disabled, Default Currency set to USD, and Automatic Codec Preference set to G.729(a) 8K CS-ACELP. A list of checkboxes on the right includes DSS Status, Auto Hold (checked), Dial By Name (checked), Show Account Code (checked), Inhibit Off-Switch Forward/Transfer (unchecked), Restrict Network Interconnect (unchecked), Drop External Only Impromptu Conference (unchecked), and Visually Differentiate External Call (unchecked).

5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 6.1. To add a new SIP Line, right click on **Line** in the Navigation pane, and select **New → SIP Line**. A new Line Number will be assigned automatically. The settings presented here simply illustrate this reference configuration and are not intended to be prescriptive.

5.4.1. SIP Line - SIP Line Tab

Select **SIP Line** tab as shown below for Line Number 10 used for AT&T and configure as follows:

- **ITSP Domain Name** – Set to the IP Office LAN1 address (192.168.62.56) 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 header and Diversion header.
- **In Service** – Default is checked
- **Check OOS** – Uncheck this box. 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)

The screenshot displays the IP Office configuration interface. On the left is the 'IP Offices' navigation pane with a tree structure including BOOTP (3), Operator (3), System (1), Line (5), Control Unit (5), Extension (28), User (21), HuntGroup (16), Short Code (67), Service (0), RAS (1), Incoming Call Route (22), WanPort (0), Directory (0), and Time Profile (0). The 'Line (5)' folder is expanded, showing lines 5, 9, 10, 11, and 17. Line 10 is selected. The main configuration area is titled 'SIP Line - Line 10*' and contains several tabs: SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'SIP Line' tab is active. It contains the following fields and controls:

- Line Number: 10 (dropdown)
- ITSP Domain Name: 192.168.62.56 (text box)
- In Service: ☒ (checkbox)
- Use Tel URI: ☐ (checkbox)
- Prefix: (empty text box)
- Check OOS: ☐ (checkbox)
- National Prefix: 0 (text box)
- Call Routing Method: Request URI (dropdown)
- Country Code: (empty text box)
- Originator number for forwarded and twinning calls: (empty text box)
- International Prefix: 011 (text box)
- Send Caller ID: Diversion Header (dropdown)
- REFER Support: ☐ (checkbox)
- Incoming: Auto (dropdown)
- Outgoing: Auto (dropdown)

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 1** configured in **Section 5.3.2**. Default values are used for the other fields.

The screenshot shows the 'SIP Line - Line 10*' configuration window with the 'Transport' tab selected. On the left, the 'IP Offices' tree is visible, showing a hierarchy: BOOTP (3), Operator (3), 00E007055F72, System (1), Line (5) (with sub-items 5, 9, 10, 11, 17), Control Unit (5), Extension (28), User (21), HuntGroup (16), and Short Code (67). The 'Line (5)' item is expanded, and 'Line 10' is selected. The main configuration area on the right contains the following fields:

- ITSP Proxy Address:** 135.242.225.200
- Network Configuration:**
 - Layer 4 Protocol:** UDP (dropdown)
 - Send Port:** 5060
 - Use Network Topology Info:** LAN 1 (dropdown)
 - Listen Port:** 5060
- Explicit DNS Server(s):** 0 . 0 . 0 . 0 . 0 . 0 . 0 . 0
- Calls Route via Registrar:** ☒
- Separate Registrar:** (empty text field)

5.4.3. SIP Line - SIP URI Tab

Select the **SIP URI** tab and click the **Add...** button 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 000001017 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 10*' configuration window. On the left is a tree view of 'IP Offices' containing various system components like BOOTP, Operator, System, Line, Control Unit, Extension, User, HuntGroup, Short Code, Service, RAS, Incoming Call Route, WanPort, Directory, Time Profile, Firewall Profile, IP Route, Account Code, License, Tunnel, User Rights, ARS, RAS Location Request, and E911 System. The main area shows the 'SIP URI' tab with a table of channels. Below the table is an 'Edit Channel' dialog box.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	110 110	2...	000001017			None	0: <Non...	10
2	110 110	2...	000001018			None	0: <Non...	10
3	110 110	2...	000001019			None	0: <Non...	10
4	110 110	2...	000001020			None	0: <Non...	10
5	110 110	2...	000001021			None	0: <Non...	10
6	110 110	2...	0000017			None	0: <Non...	10
7	110 110	2...	00000017			None	0: <Non...	10
8	110 110	2...	000003171017			None	0: <Non...	10
9	110 110	2...	00000415317...			None	0: <Non...	10

Edit Channel

Via: 205.168.62.56

Local URI: 000001017

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: None

Registration: 0: <None>

Incoming Group: 110

Outgoing Group: 110

Max Calls per Channel: 10

5.4.4. SIP Line - VoIP Tab

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

- **Compression Mode** – Set to **Automatic Select** from the drop-down list
- **DTMF Support** - Set to the default value **RFC2833**.
- **Re-invite Supported** – Check to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk
- **Use Offerer's Preferred Codec** – Check this box so that the top codec offered to IP Office is used if IP Office supports that codec
- **Fax Transport Support** – Check to allow T38 fax support.

The screenshot shows the 'SIP Line - Line 10*' configuration window with the 'VoIP' tab selected. On the left, the 'IP Offices' tree shows a hierarchy: BOOTP (3) > Operator (3) > 00E007055F72 > System (1) > Line (5) > 5, 9, 10. The main configuration area has tabs for SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'VoIP' tab is active, showing 'Compression Mode' set to 'Automatic Select' (with an 'Advanced' button), 'Call Initiation Timeout (s)' set to '4', and 'DTMF Support' set to 'RFC2833'. On the right, four checkboxes are checked: 'VoIP Silence Suppression', 'Fax Transport Support', 'Re-invite Supported', and 'Use Offerer's Preferred Codec'.

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

5.5. Users, Extensions, and Hunt Groups

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

5.5.1. Digital Telephone User 205

The following screen shows the **User** tab for User 205. This user corresponds to a digital phone.

The screenshot displays the IP Office configuration interface. On the left is the 'IP Offices' navigation pane, showing a tree structure with categories like BOOTP, Operator, System, Line, Control Unit, Extension, and User. The 'User' category is expanded, and '205 Extn205' is selected. The main area on the right is titled 'Extn205: 205' and contains several tabs: 'User', 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', and 'Voice Recording'. The 'User' tab is active, showing fields for Name (Extn205), Password, Confirm Password, Full Name, Extension (205), 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'. A 'Device Type' section shows an icon of a telephone and the text 'Avaya 1416'. At the bottom is the 'User Rights' section, which includes 'User Rights view' (set to 'User data'), 'Working hours time profile' (set to '<None>'), 'Working hours User Rights', and 'Out of hours User Rights'.

IP Offices	
BOOTP (3)	
Operator (3)	
00E007055F72	
System (1)	
Line (2)	
Control Unit (5)	
Extension (26)	
User (19)	
NoUser	
RemoteManager	
206 Extn206	
201 Extn201	
202 Extn202	
203 Extn203	
204 Extn204	
205 Extn205	
210 Extn210	
32006 Extn32006	
32008 Extn32008	
32009 Extn32009	
6665013 Extn6665013	
6665014 Extn6665014	
6665015 Extn6665015	
6665016 Extn6665016	
6665017 Extn6665017	
6665018 Extn6665018	
6665019 Extn6665019	
HuntGroup (4)	
Short Code (65)	
Service (0)	
RAS (1)	
Incoming Call Route (5)	
WanPort (0)	
Directory (0)	
Time Profile (0)	

Extn205: 205	
User	
Voicemail	
DND	
ShortCodes	
Source Numbers	
Telephony	
Forwarding	
Dial In	
Voice Recording	
Name	Extn205
Password	
Confirm Password	
Full Name	
Extension	205
Locale	
Priority	5
System Phone Rights	None
Profile	Basic User
<input type="checkbox"/> Receptionist	
<input type="checkbox"/> Enable SoftPhone	
<input type="checkbox"/> Enable one-X Portal Services	
<input type="checkbox"/> Enable one-X TeleCommuter	
<input type="checkbox"/> Ex Directory	
Device Type	Avaya 1416
User Rights	
User Rights view	User data
Working hours time profile	<None>
Working hours User Rights	
Out of hours User Rights	

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

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'Extension (26)' selected, and '73 205' highlighted. The main panel, titled 'Digital Extension: 73', shows the configuration for this extension. The 'Extn' tab is active, displaying the following fields:

Field	Value
Extension Id	73
Base Extension	205
Caller Display Type	Off
Reset Volume After Calls	<input type="checkbox"/>
Device type	Avaya 1416
Module	BD4
Port	1
Disable Speakerphone	<input type="checkbox"/>

5.5.2. IP Telephone User 203

The following screen shows the **User** tab for User 203. This user corresponds to an Avaya 1616 IP Telephone that is configured as power user with IP Office Softphone features enabled as shown below.

The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view under 'IP Offices' shows a hierarchy including BOOTP, Operator, System, Line, Control Unit, Extension, and User. The 'User (20)' folder is expanded, and '203 Extn203' is selected. The main panel on the right is titled 'Extn203: 203' and contains several tabs: 'User', 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', and 'Voice Record'. The 'User' tab is active, showing fields for Name (Extn203), Password (masked with asterisks), Confirm Password (masked with asterisks), Full Name, Extension (203), Locale, Priority (5), System Phone Rights (None), and Profile (Power User). Below these fields, there are checkboxes for 'Receptionist' (unchecked), 'Enable SoftPhone' (checked), 'Enable one-X Portal Services' (checked), 'Enable one-X TeleCommuter' (checked), and 'Ex Directory' (unchecked). A 'Device Type' field shows 'Avaya 1616L' with a small image of the phone. At the bottom, the 'User Rights' section includes 'User Rights view' (User data), 'Working hours time profile' (<None>), 'Working hours User Rights', and 'Out of hours User Rights'.

IP Offices	
BOOTP (3)	
Operator (3)	
00E007055F72	
System (1)	
Line (3)	
Control Unit (5)	
Extension (26)	
User (20)	
NoUser	
RemoteManager	
206 Extn206	
201 Extn201	
202 Extn202	
203 Extn203	
204 Extn204	
205 Extn205	
210 Extn210	
32006 Extn32006	
32008 Extn32008	
32009 Extn32009	
6665013 Extn6665013	
6665014 Extn6665014	
6665015 Extn6665015	
6665016 Extn6665016	
6665017 Extn6665017	
6665018 Extn6665018	
6665019 Extn6665019	
250 Robert	
HuntGroup (10)	
Short Code (65)	
Service (0)	
RAS (1)	
Incoming Call Route (11)	
WanPort (0)	
Directory (0)	

Extn203: 203	
User Voicemail DND ShortCodes Source Numbers Telephony Forwarding Dial In Voice Record	
Name	Extn203
Password	*****
Confirm Password	*****
Full Name	
Extension	203
Locale	
Priority	5
System Phone Rights	None
Profile	Power User
<input type="checkbox"/> Receptionist	
<input checked="" type="checkbox"/> Enable SoftPhone	
<input checked="" type="checkbox"/> Enable one-X Portal Services	
<input checked="" type="checkbox"/> Enable one-X TeleCommuter	
<input type="checkbox"/> Ex Directory	
Device Type	Avaya 1616L
User Rights	
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 using the **Voicemail Code** and **Confirm Voicemail Code** parameters.

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

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

The screenshot shows the IP Office configuration interface. On the left, a tree view under 'IP Offices' shows the hierarchy: BOOTP (3), Operator (3), 00E007055F72, System (1), Line (3), Control Unit (5), Extension (26), and User (20). The 'User' folder is expanded, showing 'NoUser', 'RemoteManager', and several extensions including '203 Extn203'. The main panel is titled 'Extn203: 203' and has tabs for 'User', 'Voicemail', 'DND', 'ShortCodes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', 'Voice Recording', and 'Button Program'. The 'Call Settings' sub-tab is active. It contains the following settings:

Outside Call Sequence	Default Ring	<input checked="" type="checkbox"/> Call Waiting On
Inside Call Sequence	Default Ring	<input checked="" type="checkbox"/> Answer Call Waiting On Hold (Analogue)
Ringback Sequence	Default Ring	<input type="checkbox"/> Busy On Held
No Answer Time (secs)	15	<input checked="" type="checkbox"/> Offhook Station
Wrap-up Time (secs)	2	
Transfer Return Time (secs)	Off	
Call Cost Mark-Up	100	

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

The screenshot shows the IP Office configuration interface. On the left, a tree view under 'IP Offices' shows the hierarchy: BOOTP (3), Operator (3), 00E007055F72, System (1), Line (3), Control Unit (5), Extension (26), and User (20). The 'Extension' folder is expanded, showing extensions including '8005 203'. The main panel is titled 'H323 Extension: 8005 203' and has tabs for 'Extn' and 'VoIP'. The 'VoIP' sub-tab is active. It contains the following settings:

IP Address	0 . 0 . 0 . 0	<input type="checkbox"/> VoIP Silence Suppression
MAC Address	00 00 00 00 00 00	<input type="checkbox"/> Enable Faststart for non-Avaya IP phones
Compression Mode	Automatic Select	<input checked="" type="checkbox"/> Out Of Band DTMF
TDM->IP Gain	Default	<input type="checkbox"/> Local Tones
IP->TDM Gain	Default	<input checked="" type="checkbox"/> Allow Direct Media Path
Supplementary Services	None	<input type="checkbox"/> Reserve Avaya IP endpoint license
		<input type="checkbox"/> Reserve 3rd party IP endpoint license

5.5.3. SIP Telephone User 220

The following screen shows the **User** tab for User 220. This user corresponds to an Avaya 1140E SIP Telephone and configured as below:

The screenshot displays the Avaya system configuration interface. On the left, a tree view under 'IP Offices' shows the hierarchy: BOOTP (3), Operator (3), 00E007055F72, System (1), 00E007055F72, Line (4), Control Unit (5), Extension (30), and User (21). Under 'User (21)', 'Extn220' is selected. The main panel shows the 'User' tab for 'Extn220: 220'. The configuration fields are as follows:

Field	Value
Name	Extn220
Password	*****
Confirm Password	*****
Full Name	Avaya 1140
Extension	220
Locale	[Dropdown]
Priority	5
System Phone Rights	None
Profile	Basic User
Receptionist	<input type="checkbox"/>
Enable SoftPhone	<input type="checkbox"/>
Enable one-X Portal Services	<input type="checkbox"/>
Enable one-X TeleCommuter	<input type="checkbox"/>
Ex Directory	<input type="checkbox"/>
Device Type	Avaya 1140E Sip

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**.

The screenshot displays the Avaya system configuration interface for 'SIP Extension: 8001 220'. The left tree view shows the hierarchy: IP Offices, BOOTP (3), Operator (3), 00E007055F72, System (1), 00E007055F72, Line (4), Control Unit (5), Extension (30), and 8001 220. The main panel shows the 'SIP Extension: 8001 220' configuration with the 'VoIP' tab selected. The configuration fields are as follows:

Field	Value
IP Address	10 . 80 . 130 . 51
Compression Mode	Automatic Select
TDM->IP Gain	Default
IP->TDM Gain	Default
DTMF Support	RFC2833
VoIP Silence Suppression	<input type="checkbox"/>
Fax Transport Support	<input type="checkbox"/>
Local Hold Music	<input type="checkbox"/>
Allow Direct Media Path	<input checked="" type="checkbox"/>
Re-invite Supported	<input checked="" type="checkbox"/>
Use Offerer's Preferred Codec	<input type="checkbox"/>
Reserve Avaya IP endpoint license	<input checked="" type="checkbox"/>
Reserve 3rd party IP endpoint license	<input type="checkbox"/>

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 the an agent with the right skill set. To configure a new hunt group, right-click **HuntGroup** from the Navigation pane, and select **New**. 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 **Rotary**, this will enable the telephones to ring in a round robin fashion. Click the **Edit** button to change the **User List**. Once a user is part of a hunt group **User List**, it can be enabled/disabled by checking/unchecking the box in the Extension field.

The screenshot shows the 'Rotary Group' configuration window for the 'Receivables' hunt group. The left pane shows the 'IP Offices' tree with 'HuntGroup (16)' selected, and '11 Receivables' highlighted. The right pane shows the configuration for the 'Receivables' hunt group. The 'Name' field is set to 'Receivables' and the 'Extension' field is set to '11'. The 'Ring Mode' is set to 'Rotary'. The 'Overflow Mode' is set to 'Call'. The 'Hold Music Source' is set to 'No Change'. The 'Agent's Status on No-Answer Applies To' is set to 'None'. The 'User List' table shows the following data:

Extension	Name
<input checked="" type="checkbox"/> 201	Extn201
<input checked="" type="checkbox"/> 202	Extn202
<input type="checkbox"/> 210	Extn210
<input type="checkbox"/> 32008	Extn32008
<input type="checkbox"/> 32009	Extn32009
<input checked="" type="checkbox"/> 205	Extn205

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 'Rotary Group Receivables' configuration window with the 'Queuing' tab selected. The 'Queuing On' checkbox is checked. The 'Queue Length' field is set to '3'. 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'. The 'Analog Extension to Notify' is set to '<None>'. The left pane shows the 'IP Offices' tree with 'HuntGroup (16)' selected, and '11 Receivables' highlighted.

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

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree shows a hierarchy: BOOTP (3) > Operator (3) > 00E007055F72 > System (1) > 00E007055F72 > Line (4) > Control Unit (5) > Extension (30) > User (21) > HuntGroup (16). The 'HuntGroup' is expanded, showing various extensions and skills, with '11 Receivables' highlighted.

The main window is titled 'Rotary Group Receivables: 11*'. It has several tabs: Hunt Group, Voicemail, Fallback, Queuing, Voice Recording, **Announcements**, and SIP. The 'Announcements' tab is active, showing the following configuration:

- ☒ **Announcements On**
- Wait before 1st announcement (seconds): 40
- ☐ Synchronise Calls
- Flag call as answered: ☐
- Play 1st announcement
- Post announcement tone: Music on hold
- 2nd Announcement: ☒
- Wait before 2nd announcement (seconds): 20
- Play 2nd announcement
- Repeat last announcement: ☒
- Wait before repeat (seconds): 20

A flow diagram on the right illustrates the announcement sequence: 'Play 1st announcement' leads to 'Post announcement tone', which leads to 'Play 2nd announcement'. From 'Play 2nd announcement', a loop arrow returns to the 'Repeat last announcement' checkbox, which then loops back to 'Play 2nd announcement'.

Additional hunt groups Billing, Payables and Customer Service were 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 will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New**. 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 for 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, this code was configured in this reference configuration. When a user enters, ***17**, they can retrieve the messages in their mailbox. Additionally, this short code can be also in the Incoming Call Route configured in **Section 5.7**.

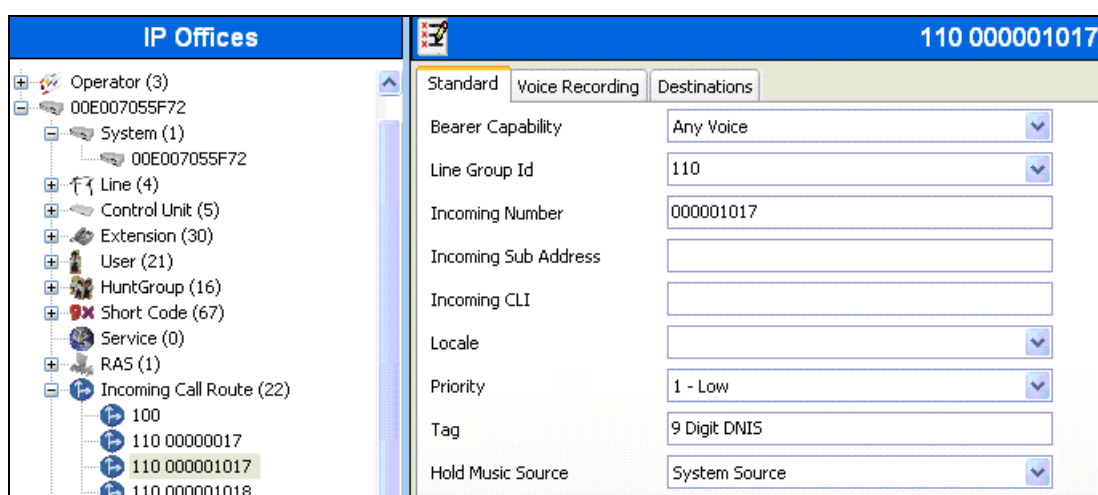
The screenshot displays the 'Short Code' configuration window for '*17: Voicemail Collect'. The 'Short Code' tab is active, and the following fields are visible:

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**. 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 **000001017** 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**.



IP Offices		110 000001017*	
Operator (3)		Standard	
00E007055F72		Voice Recording	
System (1)		Destinations	
00E007055F72		Bearer Capability: Any Voice	
Line (4)		Line Group Id: 110	
Control Unit (5)		Incoming Number: 000001017	
Extension (30)		Incoming Sub Address:	
User (21)		Incoming CLI:	
HuntGroup (16)		Locale:	
Short Code (67)		Priority: 1 - Low	
Service (0)		Tag: 9 Digit DNIS	
RAS (1)		Hold Music Source: System Source	
Incoming Call Route (22)			
100			
110 00000017			
110 000001017			
110 000001018			

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 000001017*		
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 selection.

TimeProfile	Destination	Fallback Extension
Default Value	203 Extn203	

The following screens displays how a short code can be manually assigned in the **Destination** field to route the call for voicemail retrieval and access Call Center functionality.

TimeProfile	Destination	Fallback Extension
Default Value	*17	

TimeProfile	Destination	Fallback Extension
Default Value	*93	

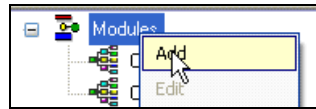
The following screen displays another mechanism to access the Call Center functionality without using the short code. The Call Center functionality is configured in Voicemail Pro as detailed in **Section 5.8**.

TimeProfile	Destination	Fallback Extension
Default Value	VM:CallCenter	

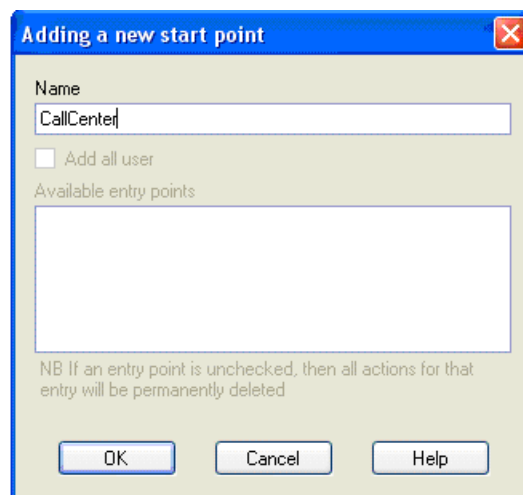
5.8. Call Center Provisioning in Voicemail Pro

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

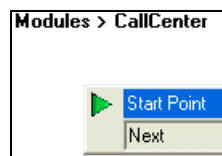
1. Navigate to **Start→Voicemail Pro Client** and right click on modules and select **Add** to add a new module.



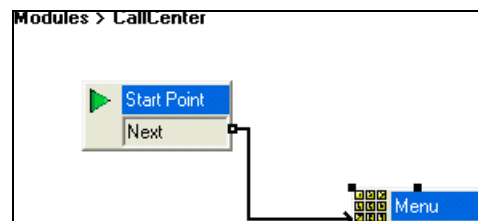
2. In the screen below, enter **CallCenter** in the **Name** field and click OK.



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



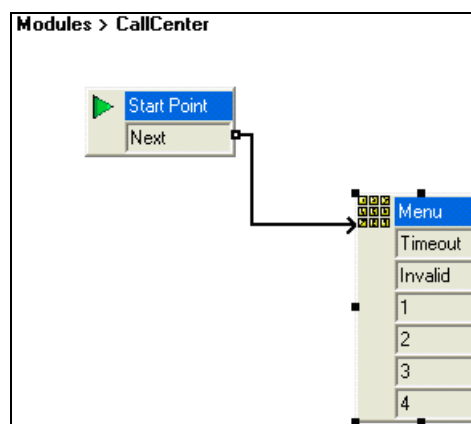
4. Under the **Actions** tab, select **Basic Actions** [not shown]. Select **Menu** and place it on the right side of the pane and then connect the **Start Point** to **Menu** as shown below:



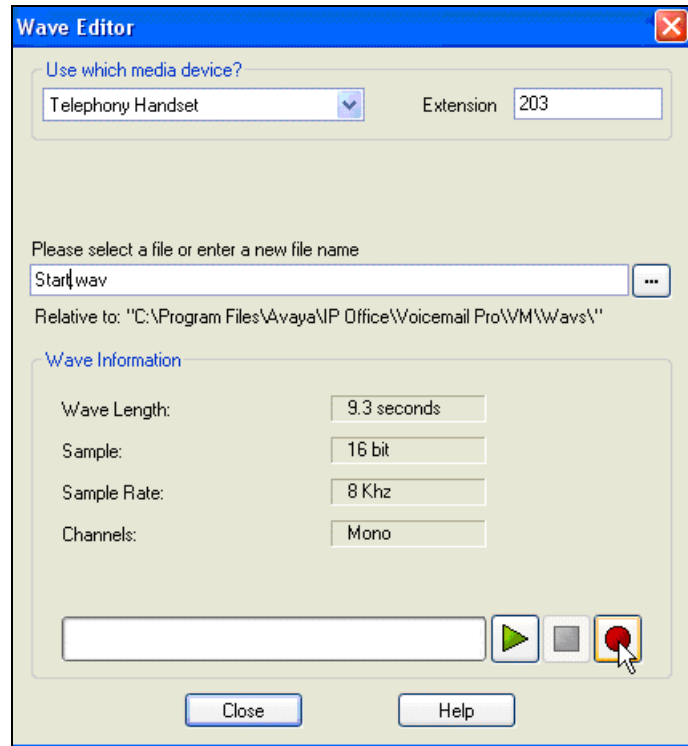
5. Right click on **Menu** and select **Touch Tones** tab. Check the appropriate boxes. In this reference configuration, 1, 2, 3, 4, Timeout and Invalid boxes were selected. This allows caller to enter any of the digits from 1 to 4 to go to the appropriate agent. Digit **1** was used for **Receivables**,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

The screenshot shows the 'Properties for Menu' dialog box with the 'Touch Tones' tab selected. The 'Invalid Input Handling' section is expanded, showing a list of digits (1-9, *, #) with checkboxes. Digits 1, 2, 3, and 4 are checked. Below the list, the 'No of Retries' is set to 4. The 'Timeout' checkbox is checked, with a value of 5 seconds. The 'Invalid Entry' checkbox is also checked. The 'Prompt' field for Timeout is set to 'Timeout.wav' and for Invalid Entry is set to 'Invalid Entry.wav'. The dialog has 'OK', 'Cancel', and 'Help' buttons at the bottom.

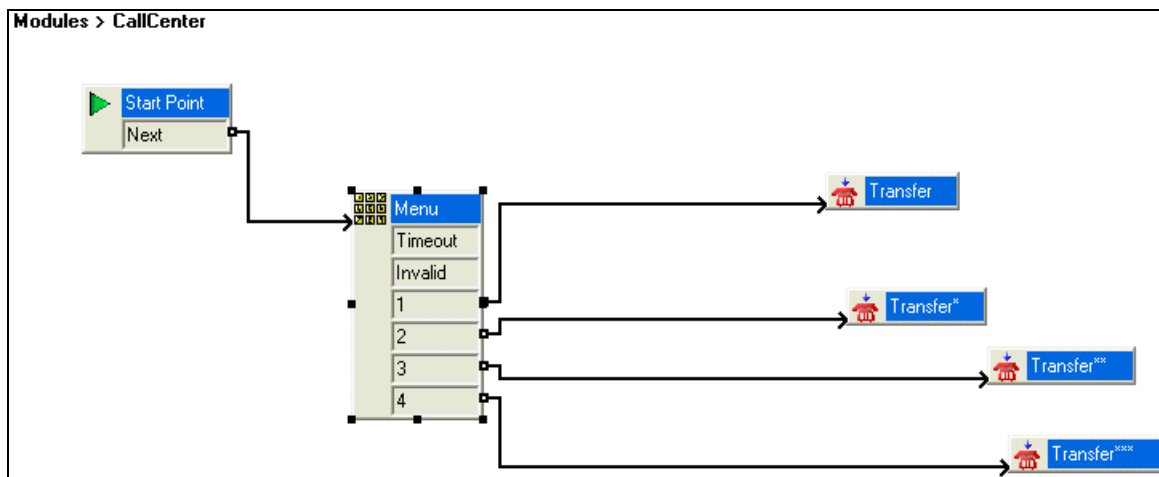
Click **OK** and following screen is displayed:



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



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



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

The screenshot shows the 'Properties for Transfer' dialog box with the 'Specific' tab selected. The 'Transfer call to' section contains the following fields and options:

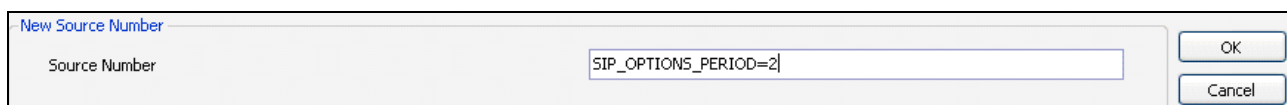
- Destination:** A text field containing '11' with a dropdown arrow on the right.
- Source of transfer (displayed on phone):** An empty text field with a dropdown arrow on the right.
- Description (displayed on phone):** An empty text field with a dropdown arrow on the right.
- Set Caller Priority:** An unchecked checkbox with a dropdown menu showing 'Low'.
- Notify Caller of Transfer to Target:** An unchecked checkbox.

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

5.9. SIP OPTIONS Frequency

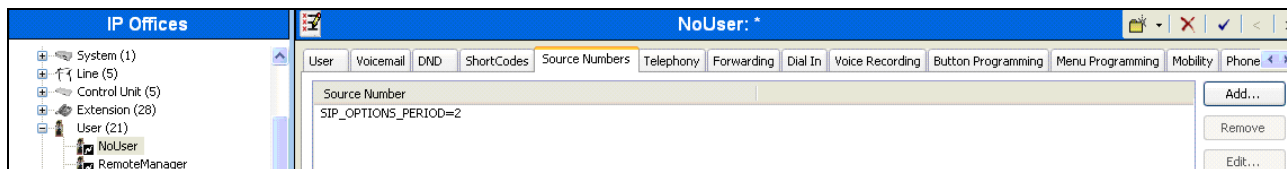
From the Navigation pane, select **User** and then select the user named **NoUser**. In the **NoUser** Details pane, select the tab **Source Numbers**. Press the **Add...** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field shown below, type **SIP_USE_PAI_FOR_PRIVACY**.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

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



A dialog box titled "New Source Number" with a text input field containing "SIP_OPTIONS_PERIOD=2" and "OK" and "Cancel" buttons.

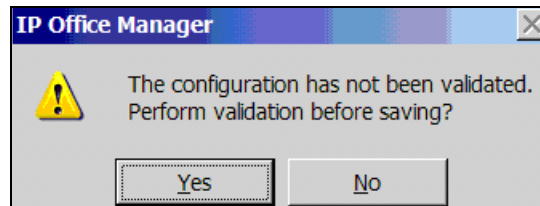
The following screen displays the Source Numbers configured in this reference configuration.



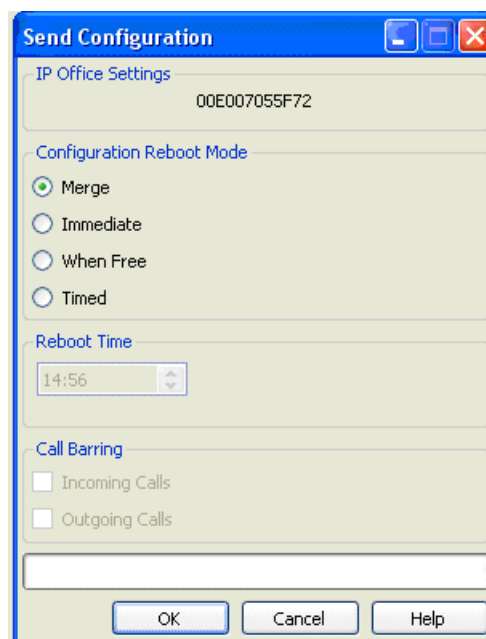
A screenshot of the "NoUser: *" configuration window. The "Source Numbers" tab is selected. The "Source Number" field contains "SIP_OPTIONS_PERIOD=2". The "Add..." button is visible on the right.

5.10. Saving Configuration Changes to IP Office

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



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



6. Verification Steps

The following steps may be used to verify the configuration:

6.1. General

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

7. Conclusion

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

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

8. References

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

[IPO-INSTALL] [IP Office 6.1](#) Installation Manual, Issue 22g, November 17 2010
Document Number 15-601042
<https://support.avaya.com/css/P8/documents/100119958>

[IPO-MGR] [IP Office Release 6.1 Manager 8.1](#), Issue 25i, November 23, 2010
Document Number 15-601011
<https://support.avaya.com/css/P8/documents/100119917>

[IPO-SYSSTAT] IP Office Release 6.0 System Status Application, Issue 05a, February 12, 2010
Document Number 15-601758
<http://support.avaya.com/css/P8/documents/100073300>

[IPO-VMPRO] IP Office Release 6.1 Voicemail Pro Installation and Maintenance, Issue 23c, November 5, 2010
Document Number 15-601063
<https://support.avaya.com/css/P8/documents/100119901>

[IPO-MON] IP Office System Monitor, Issue 02b, November 28, 2008
Document Number 15-601019
<http://support.avaya.com/css/P8/documents/100073350>

Additional IP Office documentation can be found at:
<http://marketingtools.avaya.com/knowledgebase/>

AT&T IP Toll Free Service Descriptions:

[1] *AT&T IP Toll Free*
<http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/>

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