

Avaya Solution & Interoperability Test Lab

# Application Notes for Avaya IP Office Release 8.1 with AT&T Business in a Box (BIB) with AT&T IP Flexible Reach - Enhanced Features – Issue 1.1

#### Abstract

These Application Notes describe the steps for configuring Avaya IP Office R8.1with the AT&T Business in a Box with AT&T IP Flexible Reach - Enhanced Features. AT&T Business in a Box with AT&T IP Flexible Reach - Enhanced Features supports AVPN or MIS/PNT transport connection.

The AT&T Business in a Box with AT&T IP Flexible Reach - Enhanced Features is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

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 R8.1 with the AT&T Business in a Box (BIB) with AT&T IP Flexible Reach - Enhanced Features.

In the reference configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 V2 Release 8.1, Avaya Voicemail Pro, Avaya IP Office Softphone (SIP), and Avaya H.323, SIP, and Analog desk phones.

The AT&T AT&T Business in a Box with IP Flexible Reach - Enhanced Features service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T AT&T Business in a Box with IP Flexible Reach - Enhanced Features service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

**Note** - The AT&T IP Flexible Reach - Enhanced Features service utilizes AVPN or MIS/PNT transport services, with or without BIB access. In addition, AVPN transport supports compressed RTP (cRTP), while MIS/PNT transport does not.

However, while the BIB access also supports AVPN and MIS/PNT transport, cRTP is not supported.

The AT&T IP Flexible Reach - Enhanced Features service will be referred to as IPFR-EF in the remainder of this document.

# 2. General Test Approach and Test Results

The test environment (see **Figure 1**) consisted of:

- A simulated enterprise with Avaya IP Office, Avaya IP Office telephones and fax machines (Ventafax application).
- AT&T Business in a Box with IPFR-EF service, to which the simulated enterprise was connected via AVPN transport. However, these Application Notes also certify MIS/PNT transport.
- The configurations described in these Application Notes certify Avaya IP Office and IPFR-EF, with and without BIB access.

The main test objectives were to verify the features and functionality described in Section 2.1.

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 and outbound call flows (see **Section 3.2** for examples) between Avaya IP Office and AT&T Business in a Box with IPFR-EF via AVPN transport. This 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 to and from the PSTN across the AT&T network. The following features were tested as part of this effort:

- Incoming and outgoing calls between PSTN, routed by the AT&T Business in a Box with IPFR-EF service, and Avaya IP Office. These calls are via the Avaya IP Office SIP Line and may be generated/answered by Avaya SIP telephones/Softphones, H.323 telephones, Analog telephones, Analog fax machines or via Hunt Groups. Coverage to Avaya IP Office Voicemail Pro, and Voicemail Pro auto-attendant applications, may also be used.
- Inbound / Outbound fax using T38 or G.711.
- Proper disconnect when the caller abandoned a call before answer for both inbound and outbound calls, and when the Avaya IP Office party or the PSTN party terminated an active call.
- Proper busy tone heard when an Avaya IP Office user called a busy PSTN user, or a PSTN user called a busy Avaya IP Office user (i.e., if no redirection was configured for user busy conditions).
- Various outbound PSTN call types were tested including long distance, international, and toll-free.
- Requests for privacy (i.e., caller anonymity) for Avaya IP Office outbound calls to the PSTN, as well as privacy requests for inbound calls from the PSTN to Avaya IP Office users.
- SIP OPTIONS monitoring of the health of the SIP trunk. In the reference configuration Avaya IP Office sent OPTIONS to the AT&T Business in a Box with IPFR-EF service Border Element and AT&T responded with 405 Method Not Allowed (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up. The AT&T Business in a Box with IPFR-EF service Border Element does not send OPTIONS to Avaya IP Office
- Incoming and outgoing calls using the G.729(a) and G.711 ULAW codecs.
- Call redirection with Diversion Header.
- Long duration calls.
- DTMF transmission (RFC 2833) for successful voice mail navigation, including navigation of a simple auto-attendant application configured on Avaya IP Office Voicemail Pro.
  - See Section 2.2, Item 1 for a DTMF issue regarding Avaya IP Office Softphone.
- Telephony features such as call waiting, hold, transfer, and conference.
- Mobile twinning to a mobile phone when the associated Avaya IP Office extension is called, as well as Mobility features such as Mobile Callback and Mobile Call Control.
- AT&T IPFR-EF service features such as:
  - o Simultaneous Ring
  - Sequential Ring

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- $\circ$  Call Forward Always
- Call Forward Busy
- Call Forward Ring No Answer

#### 2.2. Test Results/Known Limitations

Interoperability testing of the reference configuration was completed with successful results; however the following observations were noted:

- The AT&T IP Flexible Reach Enhanced Features service "blind transfer" feature (utilizing Refer for network call redirection) is not supported. While IP Office R8.1 does support Refer, it does not support this type of call redirection.
  - Refer must be disabled on IP Office R8.1 (see Section 5.4).
- Avaya IP Office Softphone did not generate RFC2833 DTMF when attempting to navigate menus with an Auto Attendant system utilizing SIP 183 Session Progress signaling. Some Auto Attendant systems send 183 Session Progress in response to an inbound call, and do not send 200 OK until a menu selection has been made. In these cases, Avaya IP Office Softphone did not generate DTMF SIP Telephone Events when menu selection was attempted (even though appropriate SIP Telephone Event SDP signaling was sent in the initial Avaya IP Office Invite and received in the network 183).
  - An MR was opened with Avaya IP Office support.
  - This issue does not occur with Avaya IP Office desk phones.
  - It should be noted that if the called Auto Attendant system answers at the onset with 200 OK (prior to a menu selection), then the Avaya IP Office Softphone does generate DTMF Telephone Events during menu selection.
- The Avaya IP Office fax feature "T.38 Fallback" (to G.711) is not supported in the reference configuration.
- Avaya IP Office only supports a packet size of 20 msecs, and does not specify a PTIME value in the SIP SDP (in either requests or responses). Network responses include MAXPTIME=20, and network requests include MAXPTIME=30. Although no issues were found during testing, the AT&T IPFR-EF service recommends a value of 30ms when AVPN transport is used.
- A no audio condition was found in the AT&T test network environment when an inbound PSTN call was forwarded back to PSTN or twinned to a mobile phone. This problem was apparently a result of media deadlock condition where the security aspects of the AT&T test network environment expected Avaya IP Office endpoint to send RTP packets first, even though there was no Avaya IP Office originated media involved (since the original inbound PSTN call had been re-directed to a second PSTN destination). A work around for this issue was accomplished by enabling RTP Keepalives in Avaya IP Office to send spurious RTP packets toward the AT&T test network at the start of the call, thus breaking the media deadlock. Although this issue may not occur in production AT&T Business in a Box with IPFR-EF service connections, customers experiencing loss of

audio for similar call flows over secure connections should follow the steps for enabling RTP Keepalives in Avaya IP Office as shown in **Section 5.3.3**.

- Avaya IP Office uses a fixed RFC2833 Telephone Event type of 101 in SIP requests. The network responses comply with a value of 101. However network SIP requests specify Telephone Event type 100 and Avaya IP Office complies with a value of 100. No issues were found during testing as a result of this behavior.
- Emergency 911/E911 Services Limitations and Restrictions Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.

While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at <u>http://new.serviceguide.att.com</http://new.serviceguide.att.com/</u>>. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.

\* NOTE: N11 (including 911) calls are not supported unless AT&T IP Flexible Reach Local Service is ordered!

#### 2.3. Support

#### 2.3.1. Avaya

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

# 2.3.2. AT&T

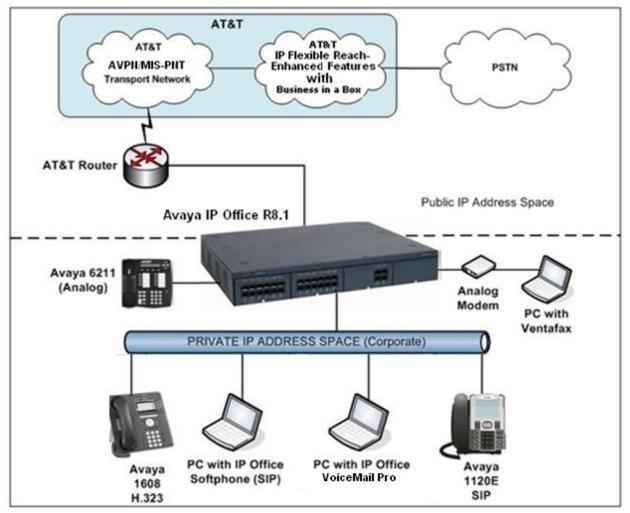
AT&T customers may obtain support for the AT&T Business in a Box with IPFR-EF service by calling (877) 288-8362 or by visiting: <u>http://www.business.att.com/enterprise/Service/voice-services/voip/business-in-a-box/</u>

# 3. Reference Configuration

Note – Documents used to provision the reference configuration are listed in Section 9. Specific references to these documents are indicated in the following sections by the notation reference [x], where x is the document reference number.

The reference configuration used in these Application Notes is shown in **Figure 1** below and consists of the following components:

- Avaya IP Office provides the voice communications services for a particular enterprise site. In the reference configuration, Avaya IP Office runs on an IP 500 V2. This solution is extensible to other Avaya IP Office hardware as well.
- Avaya "desk" telephones are represented with an Avaya 1608 H.323 set, an Avaya 6211 Analog set, an Avaya 1120E SIP set, and PC based Avaya IP Office SIP Softphone (in Default Mode). Fax endpoints are represented by PCs running Ventafax software connected by modem to an Avaya IP Office analog port.
- Avaya IP Office Voicemail Pro (running on a Windows 2003 server) provided the voice messaging capabilities in the reference configuration.
- Outbound and Inbound calls utilize telephone or fax User/Extensions provisioned on Avaya IP Office. Signaling is sent between Avaya IP Office and the AT&T Business in a Box via the IPFR-EF Border Element.
- The AT&T IPFR-EF service requires the following network settings between Avaya IP Office and the IPFR-EF Border Element:
  - UDP transport using port 5060
  - RTP port ranges 16384-32767
- AT&T provided the inbound and outbound access numbers (DID and DNIS) used in the reference configuration.



**Figure 1: Reference Configuration** 

### **3.1. Illustrative Configuration Information**

The specific values listed in **Table 1** 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 Business in a Box with IPFR-EF IP addressing shown in this document is an example. AT&T Customer Care will provide the actual IP addressing as part of the Business in a Box with IPFR-EF provisioning process.

Illustrative Value in these Application Notes
192.168.64.130
192.168.42.1
172.16.6.51
9
135.25.29.74
192.168.64.254

 Table 1: Illustrative Values Used in these Application Notes

#### 3.2. Call Flows

To understand how inbound and outbound AT&T Business in a Box with IPFR-EF service calls are handled by Avaya IP Office, four 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 Business in a Box with IPFR-EF service call that arrives on Avaya IP Office, which in turn routes the call to a hunt group, phone or a fax endpoint.

1. A PSTN phone originates a call to an IPFR-EF service number.

2. The PSTN routes the call to the AT&T Business in a Box with IPFR-EF service network.

3. The AT&T Business in a Box with IPFR-EF service routes the call to Avaya IP Office.

4. Avaya IP Office applies any necessary digit manipulations based upon the DID and routes the call to a hunt group, phone or a fax endpoint.

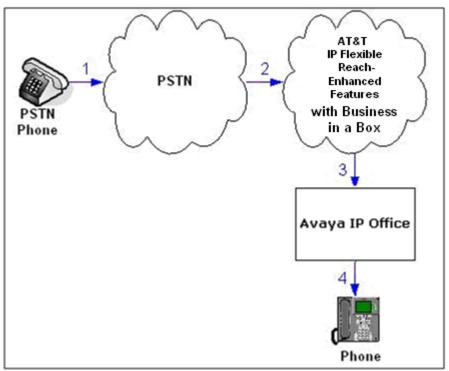


Figure 2: Inbound AT&T Business in a Box with IPFR-EF Call

### 3.2.2. Outbound

The second call scenario illustrated in the figure below is an outbound call initiated on Avaya IP Office for delivery to AT&T Business in a Box with IPFR-EF service.

1. An Avaya IP Office phone or fax endpoint originates a call to an AT&T Business in a Box with IP IPFR-EF service number for delivery to PSTN.

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2. Avaya IP Office applies any necessary origination treatment (verifying permissions, determining the proper route, selecting the outgoing trunk, etc.) and routes the call to AT&T Business in a Box with IPFR-EF service.

3. The AT&T Business in a Box with IPFR-EF service delivers the call to PSTN.

4. PSTN delivers the call to a phone or fax endpoint.

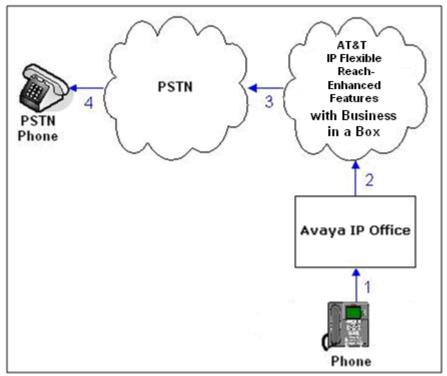


Figure 3: Outbound AT&T IP Business in a Box with IPFR-EF Call

### 3.2.3. Call Forward Re-direction<sup>1</sup>

The third call scenario illustrated in the figure below is an inbound AT&T Business in a Box with IPFR-EF service call destined for an Avaya IP Office station that has set Call Forwarding to an alternate destination. Without answering the call, Avaya IP Office immediately redirects the call back to the AT&T Business in a Box with IPFR-EF service for routing to the alternate destination.

**Note** – AT&T requires the Diversion header be used when a call is redirected to AT&T Business in a Box with IPFR-EF service (See **Section 5.4**).

1. Same as the first call scenario in Section 3.2.1.

2. Because the Avaya IP Office phone has set Call Forward to another AT&T Business in a Box with IPFR-EF service number, Avaya IP Office initiates a new call back out to the AT&T I Business in a Box with IPFR-EF service network. This ne SIP Invite will contain a Diversion Header.

<sup>&</sup>lt;sup>1</sup> Note that when the redirection is completed, RTP will still flow through Avaya IP Office

3. The AT&T Business in a Box with IPFR-EF service places a call to the alternate destination and upon answer, Avaya IP Office connects the calling party (PSTN Phone) to the target party (Target Phone).

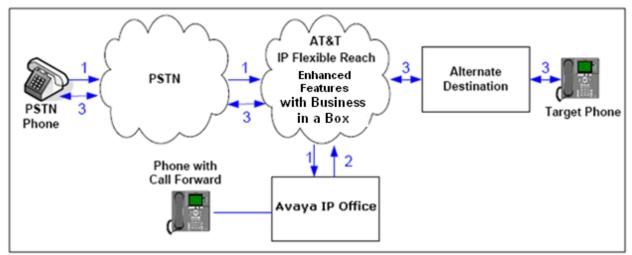


Figure 4: Re-directed (e.g. Call Forward) AT&T Business in a Box with IPFR-EF Call

### 3.2.4. Coverage to Voicemail

The call scenario illustrated in the figure below is an inbound call that is covered to Voicemail. In the reference configuration, the Voicemail system used is Avaya IP Office Voicemail Pro, running on a Windows 2003 server.

1. Same as the first call scenario in **Section 3.2.1**.

2. The Avaya IP Office phone does not answer the call, and the call covers to the external application Avaya IP Office Voicemail Pro.

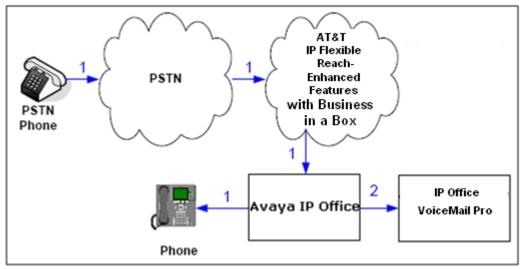


Figure 5: Coverage to Voicemail (Voicemail Pro)

# 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office 500 V2	R8.1 (43)
Avaya IP Office Manager	10.1 (43)
Avaya 1608 (H.323) Telephone	Ha1608ua1_3100.bin
Avaya 1120E (SIP) Telephone	04.03.12.00
Avaya IP Office Softphone (SIP)	3.2.3.20
Avaya 6211 Analog Telephone	-
Fax device	Ventafax 6.3
AT&T Business in a Box with IPFR-EF	VNI 23
Service via MIS/PNT transport service	
connections.	

#### Table 2: Equipment and Software Versions

**Note** - Testing was performed with IP Office 500 V2 R8.1, but it also applies to IP Office Server Edition R8.1 (single site configuration only).

# 5. Avaya IP Office Configuration

**Note** - This section describes attributes of the reference configuration, but is not meant to be prescriptive. The configuration steps described here are only for the fields where a value was changed. For all the other fields default values are used. Additionally, the screen shots referenced in these sections may not be complete at times. For more information on installing Avaya IP Office consult reference [1].

Avaya IP Office is configured via the Avaya IP Office Manager program. For more information on Avaya IP Office Manager, consult reference [2]. From the Avaya IP Office Manager PC, select Start  $\rightarrow$  Programs  $\rightarrow$  Avaya IP Office  $\rightarrow$  Manager to launch the Manager application. Enter the appropriate credentials.

Configuration Service User Login					
IP Office :	00E007058008 - IP 500 V2				
<u>S</u> ervice User Name Service User Password	K Cancel Help				

#### 5.1. Physical, Network, and Security Configuration

This section describes attributes of the reference configuration, but is not meant to be prescriptive.

In the reference configuration the Avaya IP Office 500 V2 platform contained a COMBO6210/ATM4 module. The COMBO6210/ATM4 is used to add a combination of ports to an IP500 V2 control unit and is not supported by IP500 control units. The module supports 10 voice compression channels. Codec support is G.711mu, G729a, G.723 with 64ms echo cancellation and G.722 (supported by Avaya IP Office Release 8.0 and higher). It will support 6 Digital Station ports for digital stations in slots 1-6 (except 3800, 4100, 4400, 7400, M and T-Series), 2 Analog Extension ports in slots 7-8, and 4 Analog Trunk ports in slots 9-12.

The following screen shows the modules in the Avaya IP Office used in the reference configuration. In the screen below, **IP 500 V2** is selected in the Group pane, revealing additional information about the IP 500 V2 in the Details pane.

🖬 Avaya IP Office R8.1 Manager 00E007058008 [8.1(43)] [Administrator(Administrator)]						
File Edit View Tools Help						
: 2 🖻 - 🖬 i 🗖 🔜 🔝 🚺	/ ॐ ₽	100E00705800	8 -	Control Unit	2 COMBO6210/ATM4	
IP Offices		Control Unit		6	COMBO6210/ATM4	📸 - 🔤   🗙   🗸   <   >
	Dev No.	Dev Type	Version	Unit		
- 🖗 Operator (3)	⇒1	IP 500 V2	8.1 (43)			_
OE007058008	<b>~</b> 2	COMBO6210/ATM4	8.1 (43)	Device Number	2	
				Unit Type	COMBO6210/ATM4	
				one type		
Control Unit (2)				Version	8.1 (43)	
Extension (12)					00-007050000	
User (13)				Serial Number	00e007058008	
🖓 HuntGroup (5)				Unit IP Address	0.0.0.0	
Short Code (67)						
- 🍪 Service (0)				Interconnect Number	0	
Incoming Call Route (8)				Module Number	Control Unit	
- 🚧 WanPort (0)						

The Avaya IP Office 500 V2 has two Ethernet ports on the back of the chassis, labeled **WAN** and **LAN**. In the reference configuration, the WAN port is connected to the AT&T network and the LAN port is connected to a local Ethernet switch. The Avaya H.323 and SIP telephones as well as the Administration/Softphone PC used in the reference configuration are also connected to this Ethernet switch.

In the Avaya IP Office Manager, the WAN port is identified as **LAN2** and the LAN port is identified as **LAN1**. Provisioning for these interfaces is described in **Section 5.3.2** and **5.3.3**.

In order for the Avaya IP Office system to be able to route data to/from the AT&T network, a default route must be added specifying the reference configuration AT&T gateway (192.168.64.254). To add an IP Route in Avaya 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 relevant default route using **Destination**  $\rightarrow$  LAN2.

🖬 Avaya IP Office R8.1 Manager 00E007058008 [8.1(43)] [Administrator(Administrator)]						
	i 🕹 🗢 🍖 🕴 00E00705800		• 0.0.0.0	•		
IP Offices           ▲ BOOTP (1)           ② Operator (3)           ■ ③ 005007058008           ● ③ 005007058008           ● ③ 005007058008           ● ③ 005007058008           ● ③ 005007058008           ● ③ 005007058008           ● ③ 005007058008           ● ③ 005007058008           ● ③ 0050070058008           ● ③ Control Unit (2)           ● ③ Extension (12)           ● ③ User (13)           ● ④ HuntGroup (5)           ● ● Short Code (67)           ● ③ Service (0)           ● ▲ RAS (1)           ● ● Incoming Call Route (8)           ● ● WanPort (0)           ● ○ ○ Time Profile (3)           ● ● Firewall Profile (1)           ● ● In Route (4)	IP Rot IP Address IP Mask 10.0.0.0 255.255.25 192.168.99.0 255.255.25	Gateway 5.0 192.168.64.254	IP Route IP Address IP Mask Gateway IP Address Destination Metric	LAN	· 0 · 0 · 0 · 255 · 255 · 0 · 168 · 64 · 254	

#### 5.2. Licensing

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

To verify that 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.

IP Offices	License	III	SIP Trunk Channels
	License Type	Licenses	
— 💯 Operator (3)	🛼 IP500 Upgrade Standard to Professi		
🖃 🤜 Verizon1	🐜 IP500 Voice Networking Channels	License Key	
	🐜 IP500 Voice Networking Channels		
(=? Line (5)	👟 IPSec Tunnelling	License Type	SIP Trunk Channels
	Nicrosoft CRM Integration (users)	License Status	Valid
Extension (22)	🐜 Mobile Worker		
User (22)	🐜 Mobility Features	Instances	255
HuntGroup (4)	💊 Office Worker	Duraine Data	Never
Service (0)	🍬 one-X Portal for IP Office	Expiry Date	Never
RAS (1)	🗫 Phone Manager Pro		
Incoming Call Route (8)	👟 Phone Manager Pro (per seat)		
WanPort (0)	🐜 Phone Manager Pro IP Audio Enable		
— A Directory (1)	🐜 Power User		
- 💮 Time Profile (0)	🐜 Preferred Edition (VoiceMail Pro)		
	🐜 Preferred Edition Additional VoiceMa		
IP Route (4)	Neferred/Advanced to Branch Editic		
Account Code (0)	🐜 Proactive Reporting		
License (76)	🍬 RAS LRQ Support (Rapid Response)		
Tunnel (0)	🐜 Receptionist		
User Rights (8)	🐜 Report Viewer		
Auto Attendant (0)	🐜 SIP Trunk Channels		
	👟 Small Office Edition VCM (channels)		

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. Note that in some cases duplicate license entries may be listed (e.g, Avaya IP endpoints below). One will display a key sequence in the License Key field while the other will display "Virtual".

IP Offices	License	×××	Avaya IP endpoints
	License Type	Licenses	
- 👰 Operator (3)	🛼 1600 Series Phones		
😑 🤜 Verizon1	🛼 3rd Party IP Endpoints	License Key	
	🛼 Advanced Edition		
	🔍 Advanced Small Community Network	License Type	Avaya IP endpoints
	🛼 AUDIX Voicemail	License Status	Valid
🛶 🛷 Extension (22)	🔍 Avaya IP endpoints 👘		Valid
📲 User (22)	Avaya IP endpoints	Instances	255
	Sranch Edition		
Short Code (62)		Expiry Date	Never
Service (0)	a coordigent hereoformig		

The following screen shows the availability of a valid license for **Power User** features. In the reference configuration, the user with extension 500 will be configured as a "Power User".

IP Offices	License	×××		Power User
BOOTP (1)	License Type		Licenses	
Verizon1	IP500 Upgrade Standard to Professi IP500 Voice Networking Channels		License Key	
- (7 Line (5)	🐜 IP500 Voice Networking Channels		License Type	Power User
Control Unit (3)	Reference Funder State S		License Status	Valid
User (22)	Mobile Worker		Instances	255
9x Short Code (62)	Nobility Features		Expiry Date	Never
	🎭 one-X Portal for IP Office			
Incoming Call Route (8)	🗫 Phone Manager Pro 🔍 Phone Manager Pro (per seat)			
	None Manager Pro IP Audio Enable			
Time Profile (0)	Power User			
Firewall Profile (1) 1 IP Route (4)	Referred Edition (VoiceMail Pro)			
Account Code (0)	Neferred/Advanced to Branch Editic			
🔍 🍋 License (76)	Reporting			

### 5.3. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings.

#### 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 default system serial number is used as the name. The **Avaya HTTP Clients Only** and **Enable SoftPhone HTTP Provisioning** boxes are checked to facilitate Avaya IP Office Softphone usage.

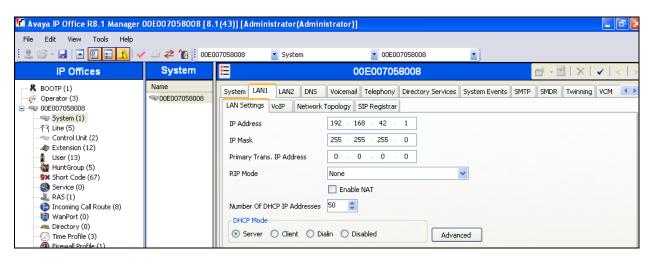
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IP Offices	System		00E007058008	
<ul> <li>BOOTP (1)</li> <li>Operator (3)</li> <li>Operator (3)</li> <li>System (1)</li> <li>-↑↑ Line (5)</li> <li>Control Unit (2)</li> <li>User (13)</li> <li>HuntGroup (5)</li> <li>Short Code (67)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Incoming Call Route (8)</li> <li>Directory (0)</li> <li>Time Profile (3)</li> <li>Firewall Profile (1)</li> <li>Firewall Profile (1)</li> <li>Firewall Profile (1)</li> <li>User Rights (8)</li> <li>Vane Rafs (2)</li> <li>RAS (2)</li> <li>RAS (2)</li> <li>RAS (1)</li> </ul>	Name	System         LAN1         LAN2         DN5         Voi           Name         Contact Information         Set contact information to place System         Set contact information to place System           Device ID         TFTP Server IP Address         HTTP Server IP Address           Phone File Server Type         Manager PC IP Address         Avaya HTTP Clients Only           Enable Softphone HTTP Provisioning         Automatic Backup         Time Setting Config Source	00E007058008	System Events SMTP SMDR Twinning VCM <

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#### 5.3.2. LAN 1 Settings

In the reference configuration, LAN1 was used to connect the Avaya IP Office to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab, and enter the following:

- **IP Address:** As shown in **Figure 1**, the IP Address of the Avaya IP Office in the reference configuration is **192.168.42.1**.
- **DHCP Mode** is also set to **Server** so that IP phones will get an IP Address from the Avaya IP Office Server. Other parameters on this screen may be set according to customer requirements.



• Click the **OK** button (not shown).

Select the LAN1  $\rightarrow$  VoIP tab as shown in the following screen. The following settings were used in the reference configuration:

- The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series Telephones used in the reference configuration.
- The **SIP Registrar Enable** box is checked to allow Avaya 11xx (SIP) and Avaya IP Office Softphone (SIP) usage.
- **RTP Port Number Range:** For each VoIP call, a receive port for incoming Real Time Protocol (RTP) traffic is selected from a defined range of possible ports, using the even numbers in that range. The Real Time Control Protocol (RTCP) traffic for the same call uses the RTP port number plus 1, that is the odd numbers. For control units and Avaya H.323 IP phones, the default port range used is 49152 to 53246. On some installations, it may be a requirement to change or restrict the port range used. It is recommended that only port numbers between 49152 and 65535 are used.
  - **Port Range (Minimum):** Default = **49152**. Range = 1024 to 64510. This sets the lower limit for the RTP port numbers used by the system.

- **Port Range (Maximum):** Default = **53246.** Range = 2048 to 65534. This sets the upper limit for the RTP port numbers used by the system. The gap between the minimum and the maximum must be at least 1024.
- **DiffServ Settings** (optional): If desired, Avaya IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Service policies. The default values were used in the reference configuration.
- Note that on this interface, **RTP Keepalives/Scope** is set to **Disabled** (default).
- Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button (not shown).

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IP Offices	System	🗄 00E007058008 🗃 - 🖻   🗙   🗸   <
<ul> <li>BOOTP (1)</li> <li>○ Operator (3)</li> <li>○ Operator (3)</li> <li>○ System (1)</li> <li>- ↑ Line (5)</li> <li>○ Control Unit (2)</li> <li>○ Extension (12)</li> <li>○ User (13)</li> <li>○ HuntGroup (5)</li> <li>○ Short Code (67)</li> <li>○ Service (0)</li> <li>○ RAS (1)</li> <li>○ Directory (0)</li> <li>○ Time Profile (3)</li> <li>○ Firewall Profile (1)</li> <li>○ IP Route (4)</li> <li>○ Account Code (0)</li> <li>○ License (77)</li> <li>○ User Rights (8)</li> <li>○ ARS (2)</li> <li>○ ARS (2)</li> <li>○ ARS (2)</li> <li>○ ARS Location Request (0)</li> <li>○ E11 System (1)</li> </ul>	Name	System       LAN1       LAN2       DNS       Voicemail       Telephony       Directory Services       System Events       SMTP       SMDR       Twinning       VCM         LAN Settings       VoiP       Network Topology       SIP Registrar         Image: Hand State       YolP       Network Topology       SIP Registrar         Image: Hand State       Port Range       Port Range       Port Range         Image: Hand State       Port Range       Port Range       Port Range         Image: Hand State       Port Range       Port Range       Port Range         Image: Hand State       Port Range (Maximum)       State       Port Range         Image: Hand State       Port Range (Maximum)       State       Port Range         Image: Hand State       Port Range (Maximum)       State       Port Range (Maximum)       State         Image: Maximum       State       Differv Settings       Port State       Por

Select the LAN1  $\rightarrow$  Network Topology tab as shown in the following screen, and enter the following:

- **Public IP Address**: The **0.0.0** default value is used. This means Avaya IP Office will use the LAN1 IP address specified on the LAN1 **LAN Settings** tab described above (192.168.42.1).
- **Public Port** to **5060**.

- **Firewall/NAT Type** is set to **Open Internet**. With this configuration, STUN will not be used.
- Click the **OK** button (not shown).

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**Note:** The **Firewall/NAT Type** parameter may need to be different, depending on the type of firewall or Network Address Translation device used at the customer premise.

If SIP endpoints are used, select the **SIP Registrar** tab. The following screen shows the settings used in the reference configuration. Note that if the **Domain Name** field is left blank (default) the LAN IP address is used for registration.

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IP Offices	System	🗄 00E007058008 🖆 - 🖭   🗙   🗸	<pre>/   &lt;   &gt;</pre>
<ul> <li>♣ BOOTP (1)</li> <li>✓ Operator (3)</li> <li>✓ ODE00758008</li> <li>✓ System (1)</li> <li>← [7] Line (5)</li> <li>✓ Control Unit (2)</li> <li>▲ Extension (12)</li> <li>▲ User (13)</li> <li>₩ HuntGroup (5)</li> <li>♥ Short Code (67)</li> <li>④ Service (0)</li> <li>♣ RAS (1)</li> <li>➡ Incoming Call Route (8)</li> </ul>	Name	System       LAN1       LAN2       DN5       Voicemail       Telephony       Directory Services       System Events       SMTP       SMDR       Twinning       V         LAN Settings       VoIP       Network Topology       SIP Registrar	'CM <b>« &gt;</b>

### 5.3.3. LAN 2 Settings

In the reference configuration, LAN2 was used to connect the Avaya IP Office to the AT&T network. To view or configure the **IP Address** of LAN2, select the **LAN2** tab followed by the **LAN Settings** tab, and enter the following:

- **IP Address:** This is the IP address of the Avaya IP Office, known to AT&T. **192.168.64.130** was used in the reference configuration.
- **DHCP Mode** is set to **Disabled** since DHCP is unnecessary towards AT&T.
- Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button.

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Select the LAN2  $\rightarrow$  VoIP tab as shown in the following screen and verify/enter the following:

- H323 Gatekeeper Enable and SIP Registrar Enable: These boxes are unchecked since IP telephones will not be registering on this link.
- **SIP Trunks Enable:** This box must be checked to enable the configuration of SIP trunks to AT&T.
- **RTP Port Number Range:** The AT&T IPFR-EF service requires that the RTP use the port range 16384 to 32767.
  - 16384 is entered in the Port Range (Minimum) field.
  - **32766** is entered in the **Port Range (Maximum)** field, as this field requires even numbers. See **Section 5.3.2** for more information on the RTP settings.

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IP Offices	System		📝 00E007058008* 🖻 🖓 🖓 🗸 🖓	
♣ BOOTP (1)           ♀ Operator (3)           ♀ 00E007058008           ♀ System (1)           -↑? Line (5)           ~ © Control Unit (2)           ~ ♥ Extension (14)           ↓ User (13)           ↓ HuntGroup (5)           ● X Short Code (67)           ● Service (0)           ▲ RAS (1)           ● Directory (0)           ● Firewall Profile (3)           ● Firewall Profile (1)           ■ PRoute (4)           ▲ Account Code (0)           ▲ License (77)           ● User Rights (8)           ↓ ARS (2)	Name		System LANI       LANI2       DNS       Yoicemail       Telephony       Directory Services       System Events       SMTF          LAN Settings       YoIP       Network Topology         H.323       Gatekeeper Enable          SIP Trunks Enable       SIP Registrar Enable         H.323       Auto-create Extn       Port Number Range         Port Range (Minimum)       16384          H.323       Auto-create User       Port Range (Maximum)       32766         H.323       Remote Extn Enable           Enable RTCP Monitoring       On Port S005           DiffServ Settings       B8       DSCP(Hex)       FC       DSCP Mask (Hex)       88       SIG DSCP (Hex)         46       DSCP       63       DSCP Mask       34       SIG DSCP	

- As described in Section 2.2 Item 4, a no audio condition was observed in the AT&T test network environment when an inbound PSTN call was forwarded back to PSTN or twinned to a mobile phone. If a similar situation is observed, a workaround for this issue is to enable RTP Keepalives on the LAN2 → VoIP tab. Scrolling down to the bottom of the form, enter the following:
  - Scope: Select RTP
  - **Periodic Timeout**: Enter **30**
  - Initial keepalives: Select Enabled

-RTP Keepalives			
Scope	RTP	Periodic timeout	30
Initial keepalives	Enabled	~	

- Let all other values default.
- Click the **OK** button.

Select the LAN2  $\rightarrow$  Network Topology tab as shown in the following screen, and enter the following:

- **Public IP Address**: Enter the Avaya IP Office LAN2 IP address **192.168.64.130** defined in the LAN2 **LAN Settings** tab above.
- **Firewall/NAT Type** is set to **Open Internet**. With this configuration, STUN will not be used.
- Binding Refresh Time can be configured to vary SIP OPTIONS timing. A value of 60 seconds was used in the reference configuration (see Section 5.11 for more information on SIP Options timing between Avaya IP Office and AT&T).
   Note In the reference configuration Avaya IP Office sent OPTIONS to the AT&T Business in a Box with IPFR-EF service Border Element and AT&T responded with 405 *Method Not Allowed* (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up. The AT&T Business in a Box with IPFR-EF service Border Element does *not* send OPTIONS to Avaya IP Office.
- Public Port to 5060.
- Click the **OK** button (not shown).

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IP Offices	System	📝 00E007058008* 📑 - 🔤   🗙   🗸   <   >		
BOOTP (1)     Operator (3)     Greater (3)     System (1)	Name	System LAN1       LAN2       DNS       Voicemail       Telephony       Directory Services       System Events       SMTP       SMDR       Twinnin < >         LAN Settings       VoIP       Network Topology       Network Topology       STUN Server IP Address       69       90       168       13       STUN Port       3478       \$         Firewall/NAT Type       Open Internet       V       \$		

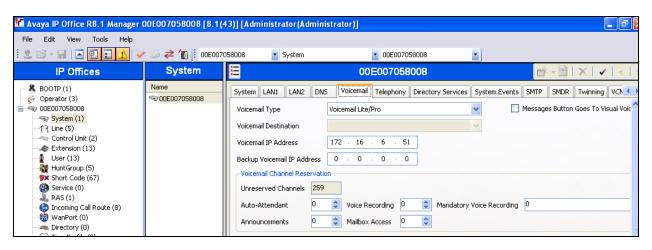
Solution & Interoperability Test Lab Application Notes ©2013 Avaya Inc. All Rights Reserved. 23 of 71 IPOR81BIBEIPFR Note that since **SIP Registrar Enable** was unchecked on the LAN2 **VOIP** tab, the SIP Registrar Tab is not present for LAN2.

### 5.3.4. Voicemail

As described in **Sections 1** and **2**, Avaya IP Office Voicemail Pro was used in the reference configuration, running on a Windows 2003 Server. The installation and provisioning of Avaya IP Office Voicemail Pro is beyond the scope of this document. See reference **[4]** & **[5]** for more information on installing and provisioning Avaya IP Office Voicemail Pro.

To view or change Avaya IP Office Voicemail settings, select the **Voicemail** tab as shown in the following screen. The settings presented here simply illustrate the reference configuration and are not intended to be prescriptive.

- Set Voicemail Type: Voicemail Lite/Pro.
- Set Voicemail IP Address: to the IP address of the platform running Voicemail Pro.
- Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button.



### 5.3.5. System Telephony Configuration

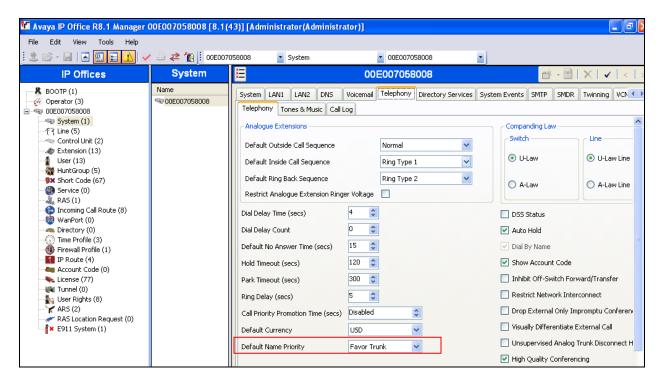
To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. The settings presented here simply illustrate the reference configuration and are not intended to be prescriptive.

In the reference configuration, the **Inhibit Off-Switch Forward/Transfer** box is unchecked so that call forwarding and call transfer to PSTN destinations via the AT&T Business in a Box with IPFR-EF service can be tested.

The **Companding Law** parameters are set to **U-LAW** as is typical in North America. Other parameters on this screen may be set according to customer requirements.

OPTIONAL: The **Default Name Priority** can be relevant to SIP Trunking. The option to **Favor Trunk** or **Favor Directory** can be set system-wide using the screen below, or set uniquely for each line. **Favor Trunk** was used in the reference configuration. With the option

set to **Favor Directory**, Avaya IP Office will prefer to display names found in a personal or system directory over those arriving from the far-end, if there is a directory match to the caller ID. This capability is also defined in the **SIP Line** tab in **Section 5.4**. A user's personal directory example is shown in **Section 5.5.2**.



#### 5.3.6. System Twinning Configuration

To view or change Twinning settings, select the **Twinning** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked, and the **Calling party information for Mobile Twinning** is left blank, in the reference configuration.

With this configuration, and related configuration of Diversion Header on the SIP Line (Section 5.4), the true identity of a PSTN caller can be presented to the twinning destination (e.g., a user's mobile phone) when a call is twinned out via the AT&T Business in a Box with IPFR-EF service.

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BOOTP (1)     Goperator (3)     O0E007058008     System (1)     -     「↑ Line (5)	Name	System LAN1 LAN2 DN Send original calling party Calling party information for Mobile Twinning	S Voicemail Telephony Directory Services	System Events SMTP SMDR Twinning VCN <>

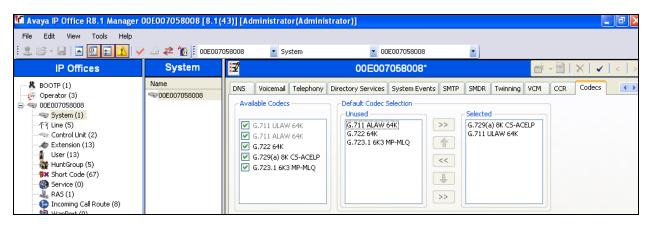
### 5.3.7. System Codecs Configuration

Navigate to the **System**  $\rightarrow$  **Codecs** tab. On the left, observe the list of **Available Codecs**. By selecting codecs in this column, they will appear in the **Default Codec Selection**  $\rightarrow$  **Unused** column. Codecs may be selected from the **Unused** list and moved to the **Selected** column by use

of the >> button, thereby making the selected codecs available in other screens where codec configuration may be performed (e.g., Sip Lines and Extensions).

The up and down arrow buttons are used to order the selected codecs. By default, all IP (SIP and H.323) lines (included the SIP Line) and extensions will assume the system default **Selected** codec list, unless configured otherwise for the specific line or extension. When completed, click on **OK** (not shown).

**Note** - In the reference configuration the System and Extension (see Section 5.5) codec lists specify G.711mu and G.729A (in that order), and the SIP Line (see Section 5.4.3) offers G.729A and G.711mu (in that order). In this manner local Avaya IP Office calls will offer G.711mu first, and SIP trunk calls will offer G.729A first.



# 5.4. SIP Line

The **SIP Line** tab in the Details pane is shown below for **Line Number 17**, used for the AT&T SIP Trunk. Note, if no SIP Line exists, right click on the **Line** item in the **Navigation** pane and select **New**  $\rightarrow$  **SIP Line**. SIP Line 17 will be the first SIP Line number created. The SIP Line form is completed as follows:

- **ITSP Domain Name:** Set to the IP Office LAN2 address as defined in **Section 5.3.3** (192.168.64.130).
- In Service and Check OOS: These boxes are checked (default).
  - Note that the Out Of Service (OOS) option is used in conjunction with SIP OPTIONS (see Section 5.11).
- Call Routing Method: Specify matching for Incoming Call Routes (Section 5.7). Matched values based on the Request URI, or To Header contents, may be selected. In the reference configuration, the default Request URI setting was used.
- **Country Code:** Use the default <blank>.

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- Send Caller ID: Set to Diversion Header. This is required by the AT&T Business in a Box with IPFR-EF service for call redirection scenarios (e.g., Call Forward, Mobile Twinning).
- **REFER Support:** Verify that this open is *not* selected (default). See Section 2.2, item 1.
- Use the default values for the other fields.
- Click **OK** (not shown).

**Optional:** As described in **Section 5.3.5**, the **Name Priority** parameter may retain the default **Favor Trunk** setting, or can be configured to **Favor Directory**. The default **Favor Trunk** setting was used in the reference configuration.

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#### 5.4.1. SIP Line - Transport Tab

Select the **SIP Line**  $\rightarrow$  **Transport** tab and configure the following:

- **ITSP Proxy Address:** Set to the AT&T Business in a Box with IPFR-EF Border Element IP address (e.g., **135.25.29.74**).
- Network Configuration  $\rightarrow$  Layer 4 Protocol. Set to UDP.
- Network Configuration  $\rightarrow$  Send Port: Set to 5060 (default).
- Network Configuration → Use Network Topology Info: Set to LAN 2. This associates the SIP Line with the parameters in the System → LAN2 → Network Topology tab.
- Calls Route via Registrar: Enabled (default).
- Click OK (not shown).

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BOOTP (1)     Operator (3)     Operator (3)     System (1)	Lin Line Type †11 Analogue Trunk †12 Analogue Trunk †13 Analogue Trunk †14 Analogue Trunk 17 SIP Line	ITSP Proxy Address 135.25.29.74

# 5.4.2. SIP Line - SIP URI Tab

Select the **SIP Line**  $\rightarrow$  **SIP URI** tab. On this form a list of the DNIS digits delivered by AT&T is created. To add a new SIP URI, click the **Add...** button. In the bottom of the screen, a **New Channel** area will be opened. Entries may be specified in two ways:

1. A "wild-card" entry that will use the contents of SIP headers containing "calling" information.

**Note** - When this method is used, the inbound AT&T DNIS digits must be specified for an Avaya IP Office User or Hunt Group on its corresponding **SIP** tab (see **Section 5.5**). Otherwise the call may be denied.

In this method the following information is specified:

- The VIA field will automatically be populated with the IP address defined in the ITSP Domain Name field defined in Section 5.4.
- Local URI, Contact, Display Name, and PAI: Set to Use Internal Data.
- **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set here to **100**. This value references the **Incoming Call Routes** in **Section 5.7**.
- **Outgoing Group:** Set to **100**. This will be used for routing outbound calls to AT&T via the **Short Codes** (Section 5.6) or ARS configuration (Section 5.8).
- Max Calls per Channel: In the reference configuration this was set to 10. This sets the maximum number of simultaneous calls that can use the URI before Avaya IP Office returns busy to any further calls.
- Click **OK** to save the information.

🜃 Avaya IP Office R8.1 Manager	00E007058008 [8.1(4	43)] [Administrator(Administrator)]	_ <b>d</b> 🛛
File Edit View Tools Help			
i 2. 🗁 - 🖬 i 🛋 🔝 🔝 🚹 🗸	🖌 🍰 🛹 🗽 🚺 00E0070	058008 🔽 Line 🔽 17	
IP Offices	Line	🗄 SIP Line - Line 17 📸 - 🕋 🗙 🗎	<ul> <li></li> <li>&gt;</li> </ul>
<ul> <li>BOOTP (1)</li> <li>✓ Operator (3)</li> <li>✓ Operator (3)</li> <li>✓ Operator (3)</li> <li>✓ System (1)</li> <li>✓ Line (5)</li> <li>✓ Control Unit (2)</li> <li>✓ Extension (13)</li> <li>✓ User (13)</li> <li>✓ HuntGroup (5)</li> <li>✓ Short Code (67)</li> <li>✓ Short Code (67)</li> <li>✓ Service (0)</li> <li>✓ RAS (1)</li> <li>✓ Incoming Call Route (8)</li> <li>✓ WanPort (0)</li> <li>✓ Time Profile (3)</li> <li>✓ Firewall Profile (1)</li> <li>✓ In P Route (4)</li> <li>✓ Account Code (0)</li> <li>✓ License (77)</li> <li>✓ Tunnel (0)</li> <li>✓ User Rights (8)</li> <li>✓ ARS (2)</li> <li>✓ RAS Location Request (0)</li> <li>✓ E911 System (1)</li> </ul>	Lin Line Type †↑1 Analogue Trunk †↑2 Analogue Trunk †↑4 Analogue Trunk ↑↑4 Analogue Trunk ▶17 SIP Line	Channel         Groups         Via         Local URI         Contact         Display Name         PAI         Credential         Max Calls           1         100         100         1         0: <non< td="">         10           Edit Channel         Via         192 · 168 · 64 · 130         100         100</non<>	Add Remove Edit

2. A specific entry that will match inbound DNIS digits from AT&T.

**Note** – This method must be used for Avaya IP Office call destinations other than Users or Hunt Groups or the calls will be denied.

In this method the following information is specified:

- Local URI, Contact, and Display Name: Set to an AT&T DNIS number (e.g, 3145555386).
- **PAI**: Set to **Use Internal Data**. When set, Avaya IP Office will use PAI for privacy signaling (see **Section 5.6.3**).
- **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set here to **100**. This value references the **Incoming Call Routes** in **Section 5.7**.
- **Outgoing Group:** Set to **100**. This will be used for routing outbound calls to AT&T via the **Short Codes** (Section 5.6) or ARS configuration (Section 5.8).
- Max Calls per Channel: In the reference configuration to this was set to 10. This sets the maximum number of simultaneous calls that can use the URI before Avaya IP Office returns busy to any further calls.
- Click **OK** to save the information.

IP Offices		Line	12		SIP	Line - Lin	e 17*			<b>- 1</b>	1   X	(   ✔   <   >
BOOTP (1)	Lin	Line Type	SIP Line T	ransport SIP URI	VoIP T38 Fax S	IP Credentials						
Operator (3)     Operator (3)     Operator (3)     Operator (3)     System (1)     -     -     -     -     -     System (1)     -	行1 行2 行3 行4 <b>》</b> 17	Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk SIP Line	Channi 1 2	el Groups 100 100 100 100	Via 192.168.64.130 192.168.64.130		Contact 3145555386	Display Name 3145555386	PAI	Credential 0: <non 0: <non< td=""><td>10</td><td>Add Remove Edit</td></non<></non 	10	Add Remove Edit
			Outgoir	RI : Name		192.168.64.13 3145555386 3145555386 3145555386 Use Internal D 0: <none> 100 100 10</none>	-			v v v		OK Cancel

- To edit an existing entry, click an entry in the list (e.g., **Channel 2**  $\rightarrow$  3145555386), and click the **Edit...** button. In the bottom of the screen, the **Edit Channel** area will be opened.
- When all SIP URI entries have been added/edited, click **OK** at the bottom of the screen (not shown).

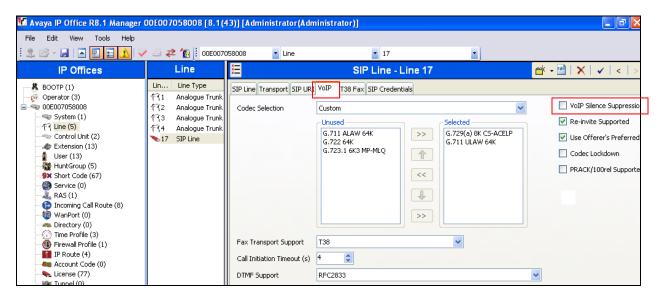
#### 5.4.3. SIP Line - VoIP Tab

Select the VoIP tab. The Codec Selection drop-down box  $\rightarrow$  System Default will list all available codecs, in the reference configuration, Custom was selected and G729(a) 8K CS-ACELP, and G.711 ULAW 64K were specified. This will cause Avaya IP Office to include these codecs in the Session Description Protocol (SDP) offer, and in the order specified. Note that G.729A is set as the preferred codec on the connection to the AT&T Business in a Box with IPFR-EF network.

- T.38 fax was used in the reference configuration. Set the **Fax Transport Support** dropdown menu to **T38**. Note that the **T.38 Fallback** option is *not* supported in the reference configuration (see **Section 2.2**). Note that Error Correction Mode (ECM) is enabled by default on the T.38 Fax tab (**Section 5.4.4**). ECM is supported by the AT&T Business in a Box with IPFR-EF service. G.711 fax also worked in the reference configuration; however T.38 is the preferred method.
- The **DTMF Support** parameter can remain set to the default value **RFC2833**.
- The **Re-invite Supported** parameter can be checked to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Click **OK** (not shown).

Since the AT&T Business in a Box with IPFR-EF service does not require registration, the **SIP Credentials** tab need not be visited.

Note that by default the VoIP Silence Suppression box is not checked. This disables the use of the G.729B codec. If silence suppression is desired, check this box, and enable the **VoIP Silence Suppression** option on the Extension form **VoIP** tab (see **Section 5.5.2**).



#### 5.4.4. SIP Line - T38 Fax

**Note** - The settings on this tab are only accessible if **Re-invite Supported** and a **Fax Transport Support** option are selected on the **VoIP** tab (**Section 5.4.3**).

**Note** - All default values were used in the reference configuration. Therefore the **Use Default Values** box is checked (not shown). If different settings are needed, uncheck this box to unlock the form.

00E007058008 Line	<ul> <li>17</li> <li>SIP Line - Line 17</li> <li>▲ - ●   ×   &lt;   &gt;</li> </ul>
X	
	SIP Line - Line 17 🛛 📩 🚽 🕹 🖓 🖌 🗸 🕹 🕹
ue Trunk Iue Trunk Iue Trunk e Trunk Tansport Iue Trunk Redundancy Low Speed III High Speed TCF Method TCF Method III EFlag Start Timer (msecs) Zef	
	Max Bit Rate (bps) EFlag Start Timer (msecs)

#### 5.5. Users, Extensions, and Hunt Groups

In this section, examples of Avaya IP Office Users, Extensions, and Hunt Groups will be illustrated. 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. Analog User Extn207

The following screen shows the **User** tab for User **Extn207**. This user corresponds to the Avaya Analog 6211 set.

IP Offices	User	🗄 Extn207: 207 📑 🗸 🛛
BOOTP (1)           Operator (3)           System (1)           T Line (5)           Control Unit (2)           Extension (11)           Extension (10)           Extension (10)	Name         Extension           Extn201         201           Extn202         202           Extn203         203           Extn204         204           Extn205         205           Extn206         206           Extn207         207           Extn208         208           Extn200         500           mExtn200         600           mExtn700         700           mRemote         Remote	User       Voicemail       DND       ShortCodes       Source Numbers       Telephony       Forwarding       Dial In       Voice Recording       Buttor         Name       Extn207         Password

The following screen shows the **SIP** tab for User **Extn207** (use the arrow buttons in the upper right corner to navigate to the SIP tab). The **SIP Name** and **Contact** parameters are configured with the associated AT&T DNIS number of the user, (e.g., **7325554892**). These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls, without having to enter this number as an explicit SIP URI for the SIP Line (see Section 5.4.2).

The SIP Display Name (Alias) parameter can optionally be configured with a descriptive name.

If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network (see **Section 5.10**). See **Section 5.6** for a method of using a Short Code (rather than static user provisioning) to place an anonymous call.

🚻 Avaya IP Office R8.1 Manager	00E007058008 [8.1	(43)] [Administrator(Ad	ministrator)]			- @ X
File Edit View Tools Help						
i & 🖻 - 🖬 🖪 🔜 🔜 认 🗸	🖊 🍰 🛹 🗽 🗓 OOEOO	7058008 🔽 User	207 Extn207	-		
IP Offices	User	12	Extn207: 207*		📸 - 🔛   🗙   🗸	<   >
BOOTP (1)	Name Extn201	Mobility Phone Manager	Options Hunt Group Membership Announceme	nts SIP	Personal Directory	< >
🖃 🐳 00E007058008	Extn201	SIP Name	7325554892			
	Extn203 Extn204	SIP Display Name (Alias)	Analog Phone			
Control Unit (2)	Extn205	Contact	7325554892			
User (13)	Extn206 Extn207					
With HuntGroup (5)	Extn208		Anonymous			
	Extn600					
Incoming Call Route (8)	Extn700					
- 🧐 WanPort (0) 	RemoteManager					

The following screen shows the Extension information for this user. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane (e.g., **207**).

00E007058008 Ext	ension	- 7	7 207 🔹 🗄 🖄 🗁 - 🖬 🖬 💽 💼 🔔 🗸 🖉 差 👔	
IP Offices	Extension		Analogue Extension: 7 207	
BOOTP (1) Operator (3) OUE007058008 System (1) 'f' Line (5) Control Unit (2) Extension (11) User (13) HuntGroup (2) Service (0) RAS (1) Incoming Call Route (	Id         Extension           ▲ 1         201           ▲ 2         202           ▲ 3         203           ▲ 4         204           ▲ 5         205           ▲ 6         206           ● 7         207           ▲ 8         208           ● 8000         600           ● 8003         500           ● 8004         700	Modu BD1 BD1 BD1 BD1 BD1 BD1 BD1 BD1 BP1 0 0 0 0	Extn       Analogue         Extension Id       7         Base Extension       207         Caller Display Type       On         Device type       Analogue Handset         Module       BP1         Port       7	

#### 5.5.2. IP Phone User Extn500

To create a new extension, right click on **Extension** from the Navigation pane and select **New** and **H.323**. Alternatively edit an existing extension by selecting an extension in the Group pane. The following screen shows a 1608 IP Telephone provisioned in the **User** tab for User **Extn500**. In the reference configuration, this user will be granted "Power User" features.

- **Password**: This password is used by user applications such as SoftConsole, Phone Manager and TAPI. It is also used for users with Dial In access. Note that this is *not* the user's phone log in code (see the information on the **Telephony** tab → **Supervisor Settings** below), or their Voicemail mailbox password (see information on the **Voicemail** tab below).
- The **Profile** parameter is set to **Power User**.
- The Enable Softphone box is checked, along with other advanced capabilities.

00E007058008 • Us	er 🔹	500 Extn500	2 🗠 - 🖬 🖪 💽 🖬 🗸 🗸 🖉	
IP Offices	User	E	Extn500: 500	<b>≝ -   X   √  </b> <   >
<ul> <li>BOOTP (1)</li> <li>Operator (3)</li> <li>Operator (3)</li> <li>System (1)</li> <li>Line (5)</li> <li>Control Unit (2)</li> <li>Extension (11)</li> <li>User (13)</li> <li>HuntGroup (2)</li> <li>Short Code (79)</li> <li>Frewall Profile (3)</li> <li>Directory (0)</li> <li>Time Profile (3)</li> <li>Frewall Profile (1)</li> <li>IP Route (3)</li> <li>Excount Code (0)</li> <li>License (76)</li> <li>User Rights (8)</li> <li>Auto Attendant (2)</li> <li>RAS (2)</li> <li>E911 System (1)</li> </ul>	Name         Extension           Extn201         201           Extn202         202           Extn203         203           Extn204         204           Extn205         205           Extn206         206           Extn208         208           Extn208         208           Extn208         208           Extn208         200           Extn2	User Voicemail DND Sho Name Password Confirm Password Full Name Extension Locale Priority System Phone Rights Profile	ttCodes Source Numbers Telephony Forwarding Extn500  ******  H323 Phone  500  S None  Power User  Receptionist Exable Softphone Exable softphone Exable softphone Exable softphone Exable softphone Exable softphone Aveya 1608	Dial In Voice Recording Button Program

Like the Analog Extn207 user, the **SIP** tab (use the arrow buttons in the upper right corner to navigate to the SIP tab) for User Extn500 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **7325554893**).

j 2 🖻 - 🖬 🖪 🔜 🖬 🚺 🗸	1 🐸 < 🌠 🕺 00E00705800	)8	🗾 User	<b>500 Extn500</b>	<b>_</b> ]	
IP Offices	User		<b>1</b>	Extn500: 500*		📸 • 🔛   🗙   🗸   <   >
<ul> <li>BOOTP (1)</li> <li>✓ Operator (3)</li> <li>✓ 00E007058008</li> <li>✓ System (1)</li> <li>– ↑ Line (5)</li> <li>✓ Control Unit (2)</li> <li>✓ Extension (13)</li> <li>✓ User (13)</li> <li>✓ User (13)</li> <li>✓ HuntGroup (5)</li> <li>✓ Short Code (67)</li> <li>✓ Short Code (67)</li> <li>✓ RAS (1)</li> <li>← Incoming Call Route (8)</li> <li>✓ WanPort (0)</li> <li>✓ Directory (0)</li> </ul>	Name         Extn           Extn201         201           Extn202         202           Extn203         203           Extn204         204           Extn205         205           Extn206         206           Extn207         207           Extn208         208           Extn200         500           Extn200         600           Extn200         700           Extn200 <th></th> <th>SIP Name SIP Display Name (Alias) Contact</th> <th>Diptions Hunt Group Membership 7325554893 H323 Phone 7325554893 Anonymous</th> <th>Announcements SIP</th> <th>Personal Directory</th>		SIP Name SIP Display Name (Alias) Contact	Diptions Hunt Group Membership 7325554893 H323 Phone 7325554893 Anonymous	Announcements SIP	Personal Directory

User **Extn500** will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User Extn500 (use the arrow buttons in the upper right corner to navigate to the Mobility tab).

The Mobility Features, Mobile Twinning, Mobile Call Control, and Mobile Callback boxes are checked.

The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **917325552438** (note that the outbound call access code **9** is specified, as described for the Short Code 9N; in **Section 5.6.1**).

The Mobile Call Control and Mobile Callback features are accessed via Short Codes (as shown in Section 5.6) and Incoming Call Routes (as shown in Section 5.7).

00E007058008 Us	er	🔹 500 Extri500 🔹 🗄 🔮 - 📓 🛋 💽 📰 🛕 🛹 🖄 🛹 🌆	
IP Offices	User	🖾 Extn500: 500*	<   >
<ul> <li>BOOTP (1)</li> <li>Operator (3)</li> <li>Operator (3)</li> <li>System (1)</li> <li>Line (5)</li> <li>Control Unit (2)</li> <li>Extension (11)</li> <li>User (13)</li> <li>HurtGroup (2)</li> <li>Short Code (79)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Directory (0)</li> <li>Time Profile (3)</li> <li>Frewall Profile (3)</li> <li>Frewall Profile (3)</li> <li>Extense (76)</li> <li>User Rights (8)</li> <li>Auto Attendant (2)</li> <li>RAS Location Request</li> <li>E911 System (1)</li> </ul>	Name         Extension           Extn201         201           Extn202         202           Extn203         203           Extn204         204           Extn205         205           Extn206         206           Extn207         208           Extn208         208           Extn200         500           Extn200         700           NoUser         Remote	n Mobility Phone Manager Options Hunt Group Membership Announcements SIP Personal Directory  Internal Twinning  Twinned Handset CNone> Madmum Number of Calls  Twin Coverage Appearances  V Mobile Twinning  Twinned Mobile Number (including dail access code) 917325552438  Twinning Time Profile Abone> V Mobile Calls V Mobile Clent Mobile Clent Mobile Callback:	

Avaya IP Office offers a feature where users can define names in a Personal Directory, and display these names, based on the inbound calling number.

The following screen shows the **Personal Directory** tab for User Extn500. With the configuration shown below, if Extn500 receives an inbound AT&T call from the telephone number 7325552438, the phone will display the name "PSTN Phone" (along with the number), even if AT&T provided a different name in the SIP INVITE message sent to Avaya IP Office.

Note – In the reference configuration, the SIP Line is configured with Name Priority  $\rightarrow$  Favor **Trunk** (see Section 5.4). This setting needs to be changed to Name Priority  $\rightarrow$  Favor **Directory**, to enable this feature.

File Edit View Tools	Help										
00E007058008 Use	м	-	500 E	xtn500	• : 2	🗠 - 🖬 i 🖪 💽 🖬	▲ 🗸 🕹 孝	10			
IP Offices	Us	er	B	1		Extn500: 50	0*		<u>e*</u> - 1 3	×	<   >
BOOTP (1) Ø Operator (3)	Name Extra201	Extension 201		fobility	Phone Manager Options	Hunt Group Membership	Announcements SI	p P	ersonal Directory		
⊟ <= 00E007058008	Extn202	202		Index	Name	Number				Add.	
	Extn203 Extn204	203 204		01	PSTN Phone	7325552438				Remo	we
Control Unit (2)	Extn205 Extn206	205 206								Edt.	

The following screen shows the Voicemail tab for User Extn500. The Voicemail On box is checked, and a Voicemail password can be configured using the Voicemail Code and Confirm Voicemail Code parameters.

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Voice mail navigation and retrieval were performed locally and from PSTN telephones, to test DTMF using RFC 2833, and to test assignment of an AT&T DNIS number to the "Voicemail Collect" feature (e.g., via the \*17 Short Code shown in Section 5.6). Note that the second configuration option described in the SIP Line  $\rightarrow$  SIP URI tab (Section 5.4.2) is required for this type of inbound call to work.

🗄 00E007058008 🔹 User 🔹 500 Extr500 💽 🗄 🙁 🐨 - 🔙 🖃 💽 🔂 🗸 🗸 🏠							
IP Offices	User	12	Extn500: 500*	<b>≝</b> * -   <b>X</b>   <b>√</b>   <   >			
BOOTP (1)     Coperator (3)     Operator (3)     Operator (3)     Operator (3)     Control Unit (2)     Operator (1)     User (13)     User (13)     WarPort (0)     Directory (0)     Time Profile (3)     Operator (3)     Account Code (0)     Leense (76)	Name         Extension           Extn201         201           Extn202         202           Extn203         203           Extn204         204           Extn205         205           Extn206         206           Extn207         207           Extn208         208           Petrh208         200           Petrh200         500           Petrh200         700           NoUser         Remote	User Voicemail DND Voicemail Code	Copy Porward Alert	rding Dial In Voice Recording Button Progra			

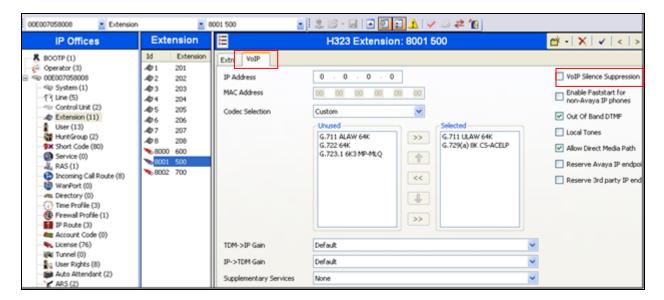
Select the **Telephony**  $\rightarrow$  **Supervisor Settings** tab as shown below. The **Login Code** will be used by the telephone user as the phone login password.

🖬 Avaya IP Office RB.1 Manager 00E007058008 [8.1(43)] [Administrator(Administrator)]									
File Edit View Tools Help									
2 🖉 - 🖬 🖪 🔜 🖬 🗸 🗸	iii 🗢 👔 📙 OOE007058008	3 🔹 User	<ul> <li>500 Extn500</li> </ul>						
IP Offices	User		Extn500: 500*	📸 🗕 🔛 🛛 🗙 🛛 🖌 🕹 🕹					
> Door if (7)           > 00E007058008           > System (1)           - f ~ Line (5)           - Control Unit (2)           - & Extension (13)           - & Extension (13)           - & Barrise (13)           - & Warn (10)           - & RAS (1)           - & Ticoming Call Route (8)           - & WanPort (0)	Jame         Exte           Extn201         201           Extn202         202           Extn203         203           Extn204         204           Extn205         205           Extn206         206           Extn207         207           Extn208         208           Extn500         500           Extn700         700           g RemoteManager         3	User Voicemail DND S	hortCodes Source Numbers Telephony tting Multi-line Options Call Log ******* (None> (None> Logged On (No change) System Default: (10)	Forwarding       Dial In       Voice Recording       Butte <					

The following screen shows the Extension information for this user, to illustrate the **VoIP** tab available for an IP Telephone. To view, select **Extension** from the Navigation pane, and the appropriate extension (**500**) from the Group pane. Select **VoIP** in the Details pane.

- Use the **IP Address** field default value (0.0.0.0).
- Note that the same codec list as shown in **Section 5.3.7** is used.
- Use defaults for the remaining fields.

Note that by default the VoIP Silence Suppression box is not checked (the same applies to provisioned SIP phones as shown in **Section 5.5.3**). This disables the use of the G.729B codec. If silence suppression is desired, check this box, and enable the Silence Suppression option on the SIP Line form VoIP tab (see **Section 5.4.3**).



# 5.5.3. SIP Telephone Users (Avaya 1120E and Avaya IP Office SoftPhone)

In the reference configuration, an Avaya 1120E SIP telephone and Avaya IP Office SoftPhone were provisioned as SIP users. To create a new extension, right click on **Extension** from the Navigation pane and select **New** and **SIP**. Alternatively edit an existing extension by selecting an extension in the Group pane.

## 5.5.3.1 SIP Avaya 1120E

The following screen shows an1120E Telephone provisioned in the User tab for User Extn600.

- **Password**: This password is used by user applications such as SoftConsole, Phone Manager and TAPI. It is also used for users with Dial In access. Note that this is *not* the user's phone log in code (see the information on the **Telephony** tab → **Supervisor Settings** below), or their Voicemail mailbox password (see the **Voicemail** tab below).
- In the reference configuration, the **Profile** parameter is set to **Basic User** (default). User Extn600 does not have the Mobile feature capabilities in the reference configuration.

IP Offices	User	🗄 Extn600: 600 📸 🚽 🗙					
<ul> <li>BOOTP (1)</li> <li>Operator (3)</li> <li>OUE007058008</li> <li>System (1)</li> <li>Control Unit (2)</li> <li>Extension (11)</li> <li>User (13)</li> <li>HuntGroup (2)</li> <li>Short Code (79)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Incoming Call Route (i</li> <li>WanPort (0)</li> <li>Directory (0)</li> <li>Time Profile (3)</li> <li>Firewall Profile (1)</li> <li>IP Route (3)</li> <li>Account Code (0)</li> <li>License (76)</li> <li>License (76)</li> <li>User Rights (8)</li> <li>Auto Attendant (2)</li> <li>RAS (2)</li> <li>RAS Location Request</li> <li>E911 System (1)</li> </ul>	Name         Extension           Extn201         201           Extn202         202           Extn203         203           Extn204         204           Extn205         205           Extn206         206           Extn207         207           Extn208         208           Extn200         500           Extn200         600           Extn200         700           Extn200         700           Remote         Free State St	User Voicemail DND Sho Name Password Confirm Password Full Name Extension Locale Priority System Phone Rights Profile		ding Button Proc			

Like the H.323 Extn500 user, the **SIP** tab (use the arrow buttons in the upper right corner to navigate to the SIP tab) for User Extn600 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **7325554385**). Optionally a user can be set to use privacy for all calls by selecting the **Anonymous** option.

IP Offices	U	ser	🛃 Extn600: 600* 📑 🗸 🗸 🗸	>
	Name	Extension	Mobility Phone Manager Options Hunt Group Membership Announcements SIP Personal Directory	< >
- 👰 Operator (3)	🛔 Extn201	201		
00E007058008	🛔 Extn202	202	SIP Name 7325554385	
- System (1)	🛔 Extn203	203		
(F) Line (5)	Extn204	204	SIP Display Name (Alias) SIP Phone	
	Extn205	205	Contact 7325554385	
📣 Extension (11)	Extn206	206	Condee	
User (13)	Extn207	207		
HuntGroup (2)	Extn208	208	Anonymous	

Like the H.323 Extn500 user, Extn600 also utilized the external Voicemail Pro server. The **Voicemail On** box is checked, and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.

Image: System (1)         Extn203         203         Confirm Voicemail Code	
BOOTP (1)         Name         Extension           Coperator (3)         Extn201         201           Coperator (3)         Extn202         202           System (1)         Extn203         203           Control Unit (2)         Extn205         205           Extension (11)         Extn205         205           Extension (11)         Extn206         206           Extension (11)         Extn206         206           Extension (12)         Extn207         207           HurkGroup (2)         Extn208         208	
Coperator (3)         Extn201         201           Coperator (3)         Extn201         201           Coperator (3)         Extn202         202           Coperator (3)         Extn202         202           Coperator (3)         Extn202         202           System (1)         Extn203         203           Confron Unit (2)         Extn204         204           Confron Unit (2)         Extn205         205           Diser (13)         Extn207         207           HuntGroup (2)         Extn208         208	<b>≝ •   X   √</b>   <
9X Short Code (79)       Image: Extn800       500         Image: Service (0)       Image: Extn800       500         Image: RAS (1)       Image: Extn800       600         Image: RAS (1)       Image: Extn800 <th></th>	

Select the **Telephony**  $\rightarrow$  **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.

00E007058008 User	<ul> <li>600 Extn600</li> </ul>	• : 2: 23 - 24   • Q = A   • 24 24 19
IP Offices	User	🗄 Extn600: 600 📸 - 🗙 🗸 🗸 🕹
♣ BOOTP (1) ♥ Operator (3) ♥ 00E007055008 ♥ System (1) ••• Control Unit (2) ••• Extension (11) ● User (13) ● HuntGroup (2) ● X Short Code (80) ● Service (0) ••• RAS (1) ••• Comming Cal Route (12) ••• WarPort (0) ••• Directory (0) ••• Time Profile (3)	Name         Extension           Extra201         201           Extra202         202           Extra203         203           Extra204         204           Extra205         205           Extra206         206           Extra207         208           Extra208         208           Extra209         208           Extra600         600           Extra600         600           Extra600         Extra60           Extra600         Extra60           Extra600         Extra60           Extra600         Ext	User Voicemail DND ShortCodes Source Numbers Telephony Forwarding Dial In Voice Recording Butt Call Settings Supervisor Settings Multi-line Options Call Log Outside Call Sequence Default Ring Inside Call Sequence Default Ring Ringback Sequence Default Ring No Answer Time (secs) System Default (15) Wrap-up Time (secs) 2 Transfer Return Time (secs) Off Call Cost Mark-Up 100

Select the **Telephony** $\rightarrow$  **Supervisor Settings** tab as shown below. The **Login Code** will be used by the telephone user as the phone login password.

00E007058008 User	τ	<b>*</b> 600	Extn600	). 🖂 - 🖃 💽 💽 🛦 i 🗸	' 🏐 孝 🔞	
IP Offices	U	ser	<b>1</b> 2	Extn600: 600*		🖆 -   🗙   🗸   <
BOOTP (1)           Operator (3)           ODE007058008           System (1)           Clune (5)           Control Unit (2)           Extension (11)           User (13)           HunkGroup (2)           Sk Short Code (79)           Service (0)           RAS (1)           Discoming Call Route (1)           WanPort (0)           Time Profile (3)           Frewall Profile (1)           P Route (3)           Account Code (0)           License (76)           Tunnel (0)	Name Extn201 Extn202 Extn203 Extn204 Extn206 Extn206 Extn206 Extn207 Extn200 E	Extension 201 202 203 204 205 206 207 206 500 600 700		ttings Multi-line Options Call Log e===== <pre>cNone&gt; </pre> <pre>cNone&gt; Logged On (No change)</pre>		N Voice Recording Button Programming Voice Login Force Login Force Account Code Outgoing Call Bar Inhibit Off-Switch Forward/Transfer Can Intrude Cannot be Intruded Can Trace Calls CCR Agent Automatic After Call Work.

A new SIP extension may be added by right-clicking on **Extension** in the Navigation pane and selecting **New** and **SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1120E.

- The **Extension ID** is automatically assigned by Avaya IP Office (e.g., **8000**). The **Base Extension** field is manually populated with the desired extension (e.g., **600**).
- Ensure the Force Authorization box is checked.

Exte	- 800	600		- 2 🖻	- 🖬 🖪 💽 🔝 🔺 🗸 🖉		
IP Offices	Ext	ension				SI	P Extension: 8000 600
BOOTP (1)     Operator (3)     Operator (3)     Operator (3)     Operator (3)     Operator (1)     Ope	▲ 1 2 ▲ 2 2 ▲ 3 2 ▲ 4 2 ▲ 5 2 ▲ 6 2 ▲ 7 2 ▲ 8 2 ► 8000 6	02 03 04 05 06 07 08 00 00	Module 8D1 8D1 8D1 8D1 8D1 8D1 8P1 8P1 0 0	Extrn Volf Extension Id Base Extens Caller Displa Reset Volun Device type Module Port Force Author	d ion y Type ne After C		8000 600 On V Avaya 1120E Sip (Language: English) 0 0 V

The following screen shows the **VoIP** tab for extension 600.

- The **IP Address** default value is used (**0.0.0.**).
- Check the **Reserve Avaya IP endpoint license** box.
- In the reference configuration the **Codec Selection** parameter is set to **Custom**, and the same codec list used in **Section 5.3.7** is used.
- Other fields may retain default values.

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IP Offices Extens	×=	H323 Extension: 8002 500	📸 • 🔛   🗙   🗸   <   >
* BOOTP (1)       Id       Ext         * OPEOTP (3)       1       201         * OPEOTP (3)       2       202         * System (1)       3       203         * Fit Line (5)       4       204         * Control Unit (2)       5       205         * Extension (12)       6       206         * User (13)       7       207         * Short Code (67)       8       8000         * Short Code (67)       8       8000         * Briccing Call Route (10)       8001       700         * Firewall Profile (3)       1       8004       702         * Directory (0)       1       8004       702         * License (77)       1       Luner (0)       804       702         * License (77)       1       User Rights (8)       4       4         * AS Location Request (0)       *       4       5	Extn VoIP IP Address MAC Address Codec Selection TDM->IP Gain IP->TDM Gain Supplementary Services	0       0       0       0         Custom       Image: Custom and the second s	<ul> <li>VoIP Silence Suppression</li> <li>Enable Faststart for non-Avaya IP phones</li> <li>Out Of Band DTMF</li> <li>Local Tones</li> <li>Allow Direct Media Path</li> <li>Reserve Avaya IP endpoint license</li> <li>Reserve 3rd party IP endpoint licen</li> </ul>

#### 5.5.3.2 SIP Avaya IP Office Softphone

**NOTE** – During testing, a DTMF issue was discovered with the Avaya IP Office Softphone (see Section 2.2, Item 1).

Repeat the steps shown in Section 5.5.3.1 with the following settings.

- Defining a User
  - User tab
    - The Avaya IP Office SoftPhone was provisioned as User Extn700.
    - The **Profile** parameter is set to **Power User** (see **Section 5.5.2**).
    - The **Enable Softphone** box is checked, the other enabled advanced capabilities shown, are optional.
  - **SIP** tab
    - **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **7325554386**).
  - Voicemail tab
    - User Extn700 also utilized the embedded Voicemail. The Voicemail On box is checked, and a Voicemail password can be configured using the Voicemail Code and Confirm Voicemail Code parameters.
  - **Telephony**→ **Call Settings** tab.
    - Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.
  - Telephony→ Supervisor Settings tab
    - The **Login Code** is the softphone login password.

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00E007058008 🔹 User 🔹 700 Extn700 💽 🕄 🖄 🖓 - 🔙 🛛 🔜 🔝 🕰 🗸 🗸 🖓								
IP Offices	User	🗄 Extn700: 700 💣 🗸 🗙						
<ul> <li>BOOTP (1)</li> <li>Operator (3)</li> <li>Operator (3)</li> <li>Operator (3)</li> <li>System (1)</li> <li>T Line (5)</li> <li>Control Unit (2)</li> <li>Extension (11)</li> <li>User (13)</li> <li>HuntGroup (2)</li> <li>Short Code (79)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Incoming Call Route (10)</li> <li>Directory (0)</li> <li>Time Profile (3)</li> <li>Firewall Profile (1)</li> <li>IP Route (3)</li> <li>Firewall Profile (1)</li> <li>IP Route (3)</li> <li>Account Code (0)</li> <li>License (76)</li> <li>Tunnel (0)</li> <li>License (76)</li> <li>RAS (2)</li> <li>RAS (2)</li> <li>RAS (2)</li> <li>RAS (2)</li> <li>RAS (2)</li> <li>RAS (2)</li> <li>Service (1)</li> </ul>	ing NoUser ing Remote	User       Voicemail       DND       ShortCodes       Source Numbers       Telephony       Forwarding       Dial In       Voice Recording       Button P         Name       Extn700         Password       *******         Confirm Password       *******         Full Name       IPO Softphone         Extension       700         Locale       United States (US English)         Priority       5         System Phone Rights       None         Profile       Power User         Receptionist       Enable Softphone         Enable Softphone       Enable Softphone         Enable one-X Portal Services       Enable one-X TeleCommuter         Enable Remote Worker       Ex Directory						
		N N						

- Defining an Extension
  - SIP Extension tab
    - The **Extension ID** is automatically assigned by Avaya IP Office (e.g., **8004**). The **Base Extension** field is manually populated with the desired extension (e.g., **700**).

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🗄 00E007058008 🔹 Extension 🔹 8004 700 🔹 🗟 🗁 - 🗐 💽 🔜 🔥 🛹 🖉								
IP Offices	Extension		H3	23 Extension: 8004 700				
BOOTP (1)	Id Extension	M BD	Extn VoIP					
□ < 00E007058008	A 2 202	вс	Extension Id	8004				
	<i>▲</i> か3 203 <i>▲</i> か4 204	BC BC	Base Extension	700				
Control Unit (2)	<i>4</i> 25 205	вс	Caller Display Type	On 💉				
User (13)	▲06 206 ▲07 207	BC BR	Reset Volume After Calls					
W HuntGroup (2) W Short Code (79) Service (0)	<i>4</i>	BF 0	Device type	Unknown IP handset				
RAS (1)	≫8003 500 ≫8004 700	0	Module	0				
			Port	0				
Time Profile (3)			Disable Speakerphone					

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- **SIP Extension**  $\rightarrow$  **VoIP** tab for extension 700.
  - The **IP Address** default value is used (**0.0.0.**).
  - In the reference configuration the **Codec Selection** parameter is set to **Custom**, and the same codec list used in **Section 5.3.7** is used.
  - Other fields may retain default values.

🕼 Avaya IP Office R8 Manager 00E007058008 [8.0(16)] [Administrator(Administrator)]						
File Edit View Tools Help						
i 00E007058008 🔹 Extension	<b>-</b> 8	002 700	- 1 - 2 - 2 - 2 - 2 - 2 - 2 - 2 - 2 - 2			
IP Offices	Extension	III IIII	H323 Extension: 8002 700	<b>☆</b> •   <b>×</b>   •   <   >		
BOOTP (1)     Operator (3)     Operator (3)     Operator (3)     Operator (3)     Operator (3)     System (1)     (7 ( line (5)     Control Unit (2)     Extension (11)     User (13)     Wer (13)     Wer (13)     Wer (13)     Wer (13)     Directory (0)     Operator (0)     Directory (0)     Operator (0)     Firewall Profile (3)     Operator (3)     Econse (76)     User Rights (8)     War Part (2)     War Part (2)     Wer Rights (8)     Wer Rights (8)	Id         Extension           ▲ 1         201           ▲ 2         202           ▲ 3         203           ▲ 4         204           ▲ 5         205           ▲ 6         206           ▲ 7         207           ▲ 8         208           ▶ 8000         600           ▶ 8001         500           ▶ 8002         700	Extn VoIP IP Address MAC Address Codec Selection TDM->IP Gain IP->TDM Gain Supplementary Services	0       0	VoIP Silence Suppression Cono-Avaya IP phones Vout Of Band DTMF Cocal Tones Vallow Direct Media Path Reserve Avaya IP endpoi Reserve 3rd party IP end V		

## 5.5.4. Hunt Groups

Users may also receive incoming calls as members of a hunt group. 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** from the Navigation pane, and the appropriate hunt group from the Group pane.

The following screen shows the **Hunt Group** tab for hunt group **4386**. This hunt group was configured to contain the Analog telephone (Extn207), the H.323 telephone (Extn500), and the SIP telephone Extn600 (1120E). In the reference configuration, these telephones extensions are rung based on idle time, due to the **Ring Mode** setting **LongestWaiting**. Click the **Edit** button to change the **User List** included in the Hunt Group from the list of available users.

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🗄 🚨 - 🖬 🖬 🔜 🔜 🔔	🗸 🛎 🤻	<b>≥ 1</b> 2   o		_	• H	untGroup	▼ 4	386 Local Hunt	•		
IP Offices	H	untGrou	р	17		Longe	st Waiting O	Group Local Hi	unt: 4386*	ď	- □   ×   ✓   <   >
BOOTP (1)     Operator (3)     System (1)     Operator (1)     Opera	Syst ស្ត្រី ស្ត្រី ស្ត្រី ស្តិរី ស្តិរី	Name Parts Sales Local Hunt Main	Extensic 1003 1002 1000 4386 200	Jame Extensi Ring Mo Hold Mo Agent's Applies	on ode usic Sour Status To		flow Fallback (* Local Hunt 4386 LongestWaiting No Change None			ements SIP t Agent Group wer Time (secs)	System Default (15)

The following screen shows the **SIP** tab for hunt group **4386**. The **SIP Name** and **Contact** are configured with the AT&T DNIS number **4386** (note that for testing purposes in the reference configuration, AT&T delivered this four digit DNIS number). In **Section 5.7**, an **Incoming Call Route** will map 4386 to this hunt group.

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i 🚨 🗁 - 🔙 i 🖪 🔜 🔝 🚺	🗸 🍰 🛹 👔 🕴 00	E00705800	8 💽 HuntGroup	<ul> <li>4386 Local Hunt</li> </ul>	
IP Offices	HuntGroup	р	🛃 Lon	gest Waiting Group Local Hunt: 4386* 🛛 🛛 📸 🗸 🖻	🗙   ✓   <   >
BOOTP (1)     Operator (3)     ODE007050008     System (1)     -↑↑ (ine (5)     Control Unit (2)     Extension (11)     User (13)     Wer (13)     System (5)	Syst Name Parts Service Sales Local Hunt Main	Extensic 1003 1002 1000 4386 200	Hunt Group Queuing C SIP Name SIP Display Name (Alias) Contact	Verflow Fallback Voicemail Voice Recording Announcements SIP 4386 Local Hunt 4386 Anonymous	

# 5.6. Short Codes

Avaya IP Office provides predefined Short Codes, however new Short Codes may be defined to match number strings to an action. 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** in the Navigation pane, and the Short Code to be configured in the Group pane.

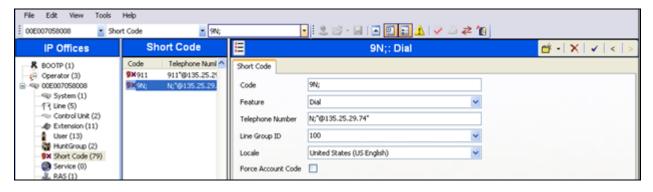
# 5.6.1. Short Code Direct Dialing (no ARS)

In the screen shown below, the Short Code **9N**; is illustrated (note the semicolon at the end of the string). This Short Code will allow an Avaya IP Office user to dial the digit 9 followed by any telephone number, symbolized by the letter **N**, to reach the SIP Line to AT&T. The variable

N can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 411), etc.

This Short Code does *not* use the ARS table, and this Short Code does not provide for alternate routing or an awareness of end of user dialing. When a users dial 9 plus the number, Avaya IP Office must wait for an end of dialing timeout before sending the SIP INVITE.

- The Code parameter is set to 9N;
- The **Feature** parameter is set to **Dial**
- The **Telephone Number** parameter is set to N"@135.25.29.74" with the text string beginning with @ in quotes (address 135.25.29.74 is the IP address of the AT&T Business in a Box with IPFR-EF border element used in the test environment). This field is used to construct the Request URI and To Header in the outgoing SIP INVITE message.
- The Line Group ID parameter is set to 100, matching the number of the Outgoing Group configured on the SIP URI tab of SIP Line 17 (see Section 5.4.2).
- $\circ~$  Click the  $\mathbf{OK}$  button (not shown).



## 5.6.2. Short Code Dialing via Automatic Route Selection (ARS)

In the reference configuration only a single connection existed to the AT&T Business in a Box with IPFR-EF service. Therefore alternate routing via the Avaya IP Office ARS capability was not used. However, for completeness, defining an ARS Short Code is described below.

In the screen shown below, the Short Code **8N**; is illustrated (note the semicolon at the end of the string). This simple Short Code will allow an Avaya IP Office user to dial the digit 8 followed by any telephone number, symbolized by the letter **N**, to reach the SIP Line to AT&T. However, Avaya IP Office will first consult the ARS table defined in **Section 5.8**. The variable **N** can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 411), etc.

- The Code parameter is set to 8N;
- The **Feature** parameter is set to **Dial**
- The Telephone Number parameter is set to N

- The **Line Group ID** parameter is set to **50:Main** (default value provided by Avaya IP Office).
- Click the **OK** button (not shown).

Short Code		
Code	8N;	
Feature	Dial	*
Telephone Numbe	er N	
Line Group ID	50: Main	*
Locale	United States (US English)	*
Force Account Co	de 🗌	

#### 5.6.3. Privacy Dialing

Optionally, add or edit a Short Code that can be used to access the SIP Line anonymously. In the screen shown below, the Short Code **3N**; is illustrated. This code is similar to the **9N**; code. The **Telephone Number** field begins with the letter **W**, which means "withhold the outgoing calling line identification".

**Note** - When a user dials 3 plus the number, Avaya IP Office will include the user's telephone number in the **P-Asserted-Identity** (PAI) header along with the **Privacy: Id** header. The **From** and **Contact** headers will contain **Anonymous**.

- The Code parameter is set to 3N;
- The **Feature** parameter is set to **Dial**
- The Telephone Number parameter is set to WN"@135.25.29.74"
- The Line Group ID parameter is set to 100, matching the number of the Outgoing Group configured on the SIP URI tab of SIP Line 17 (see Section 5.4.2).
- Click the **OK** button (not shown).

Short Code	
Code	3N;
Feature	Dial
Telephone Number	WN"@135.25.29.74"
Line Group ID	100 💌
Locale	United States (US English)
Force Account Code	

#### 5.6.4. Feature Dialing

Optionally, add or edit a Short Code that can be used to access Avaya IP Office features rather than a SIP Line.

#### 5.6.4.1 Voicemail Access

In this case, the Code \*17 is defined for Feature  $\rightarrow$  Voicemail Collect. This Short Code will be used as one means to allow an AT&T DNIS number to be programmed to route directly to voice messaging, (via inclusion of this Short Code as the destination of an Incoming Call Route in Section 5.7).

Short Code		
Code	*17	]
Feature	Voicemail Collect 🛛 👻	
Telephone Number	?U	]
Line Grot ID	0	]
Locale	×	
Force Account Code		

#### 5.6.4.2 Feature Name Extension (FNE) Access

Two Avaya IP Office Mobility features, **Mobile Call Control** and **Mobile Callback**, are shown as examples of this function.

The following screen illustrates the Code \*97 which is defined for Feature  $\rightarrow$  FNE Service and Telephone Number  $\rightarrow$  33 for the Avaya IP Office Mobility feature Mobile Callback (note that 33 is predefined in Avaya IP Office for this feature). Code, \*97 was used as the destination of an Incoming Call Route for an AT&T DNIS number. This enabled DID calls from a configured twinning destination to be dialed, and then hung up by the caller while hearing ring back. Avaya IP Office will then call the caller back using the number defined in the User  $\rightarrow$  Mobility  $\rightarrow$  Twinned Mobile Number field (see Section 5.5.2), via AT&T Business in a Box with IPFR-EF service.

Note that for this feature to work, the inbound calling number information must match the number provisioned in the associated User  $\rightarrow$  Mobility  $\rightarrow$  Twinned Mobile Number. For example, in Section 5.5.2 the number 917325552438 was defined for the H.323 set. Therefore, the inbound calling number must be 7325552438.

Short Code		
Code	*97	]
Feature	FNE Service	]
Telephone Number	33	]
Line Group ID	0	
Locale	· · · · · · · · · · · · · · · · · · ·	
Force Account Code		

The following screen illustrates another Mobility Short Code. In this case, the **Code \*98** is defined for **Feature**  $\rightarrow$  **FNE Service** and **Telephone Number**  $\rightarrow$  **31** for the Avaya IP Office Mobility feature **Mobile Call Control** (note that 31 is predefined in Avaya IP Office for this feature). Code **\*98** was used as the destination of an Incoming Call Route for an AT&T DNIS number. This enabled DID access to Mobile Call Control from configured twinning destinations, allowing the mobile user to make calls as if the calls were made from the user's Avaya IP Office extension in the office.

Short Code		
Code	*98	
Feature	FNE Service	
Telephone Number	31	
Line Group ID	0	
Locale		
Force Account Code		

#### 5.7. Incoming Call Routes

Each Incoming Call Route will map a specific AT&T DNIS number to a destination user, hunt group, or function on Avaya 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** in the Navigation pane, and the appropriate incoming call route to be configured in the Group pane.

In the screen shown below, the incoming call route for **Incoming Number**  $\rightarrow$  3145555384 is illustrated.

The Line Group ID is set to 100, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to AT&T in Section 5.4.2.

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E 00E007058008	ng Call Route	100 31455	55304	📕 🔍 🖄 - 🔚 I 🗖 🛛	1 🖬 🛦 🗸 🖉 🖉		
IP Offices	Inco	oming Call Ro	ute	E	100 3145555384		📸 •   🗙   🗸   <   >
BOOTP (1)     Greator (3)     Operator (1)     Operator (1)     User (13)     Operator (1)     Operator (1)     Operator (1)     Operator (0)     RAS (1)     Operator (0)     Operator (0)	Line Group ID © 100 © 100 © 100 © 100 © 100 © 100 © 100 © 100 © 100	Incoming Number 3145555385 3145555386 3145555384 3145555384 3145555384 3145555388 4386 7325554385 7325554384 7325554383	Destination *98 *97 500 Extn500 208 Extn500 600 Extn600 700 Extn700 500 Extn500 208 Extn208	Standard Voice Recording Bearer Capability Line Group ID Incoming Number Incoming Sub Address Incoming CLI Locale Priority Tag Hold Music Source	Destinations Any Yoice 100 3145555384	× ×	
Firewall Profile (1)				The state state	System Source		

Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when AT&T delivers DNIS digits 3145555384. In the reference configuration DNIS digits 3145555384 is associated with Avaya IP Office User **Extn500** (the 1608 H.323 telephone).

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IP Offices		ming Call Ro				100 3145			<b>≝ •   X   √  </b> <	>
R BOOTP (1)	Line Group ID			Standa	rd Voice Recording	Destinations				
- 🦗 Operator (3) 글 🖘 00E007058008	100	3145555385 3145555386	*98 *97		TimeProfile		Destination		Fallback Extension	
- System (1)	100	3145555384	500 Extn500	•	Default Value		500 Extn500	~		~
	100	3145555383	208 Extn208	*		*		~		~
Extension (11)     User (13)	100	7325554385 4386	600 Extn600 700 Extn700							
HuntGroup (2) Short Code (80)	100 100	7325554384 7325554383	500 Extn500 208 Extn208							

Repeat the process to route all AT&T DNIS numbers to their associated Avaya IP Office destinations.

Note that the **Destination** drop down menu may not contain all desired destinations (e.g., Short Codes for feature access as described in **Section 5.6**). In these cases the desired destination may be manually typed into the **Destination** field.

#### 5.8. Automatic Route Selection (ARS) and Alternate Routing

In the reference configuration only a single connection existed to the AT&T Business in a Box with IPFR-EF service. Therefore, alternate routing via the Avaya IP Office ARS capability was not used (the direct dial 9N; Short Code described in **Section 5.6.1** was used instead). While detailed coverage of ARS is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations, including alternate routing should the SIP Line be out of service or temporarily not responding.

**Section 5.6.2** shows an example of an ARS Short Code used to access the ARS table. With ARS, secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the

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primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

To add a new ARS route, right-click ARS in the Navigation pane, and select New. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route Main (ARS Route ID 50). The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the In Service box is un-checked, calls are routed to the ARS route name specified in the Out of Service Route parameter. Avaya IP Office Short Codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service Route, and the means to manually activate the Out of Service Route, can be helpful for scheduled maintenance or other known service-affecting events for the primary route.

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00E007058008 ARS		S0: Main	2 2		2 🖬 🔺 🗸	/ 🖾 🏞 🔞			
IP Offices	ARS	18		Main			<b>e</b> -	X   🗸   <	>
BOOTP (1)           Coperator (3)           System (1)	Name KMain KBackup	ARS ARS Route Id Route Name Dial Delay Time In Service Time Profile Code 11 911 0N; 1N; >N; >O0;00000000N	50 Main System Default (4) ↓ ↓ ↓ ↓ ↓ ↓ ↓ ↓ ↓ ↓ ↓ ↓ ↓	<b>0</b>	Feature Dial Emergency Dial Still Dial 3K1 Dial 3K1 Dial 3K1	Secondary Dial tone     SystemTone     Cut of Service Route     Out of Hours Route     Une Group ID     100     10     100     100     10     10     10     10     10	g S1: Badup <none></none>	Add Remove EdR	
		Alternate Route Priority Le	vel 3	×					
		Alternate Route Wait Time	30	¢ —		Alternate Route	51: Backup		~
		0							

Assuming the primary route is in-service, the number passed from the Short Code used to access ARS (e.g., 8N; in Section 5.6.2) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 8-1-732-555-2438, the call would be directed to Line Group 100. If Line Group 100 cannot be used, the call can automatically route to

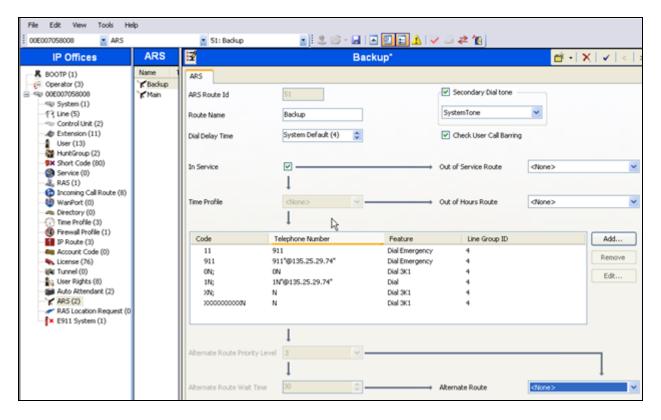
JF; Reviewed:	Solution & Interoperability Test Lab Application Notes	50 of 71
SPOC 1/14/2013	©2013 Avaya Inc. All Rights Reserved.	IPOR81BIBEIPFR

the route name configured in the **Alternate Route** parameter in the lower right of the screen (**Backup**).

Since alternate routing can be considered a privilege not available to all callers, Avaya IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

**Note** – No alternate network access was available in the reference configuration. The example below is included for informational purposes.

The following screen shows an example ARS configuration for the route named **Backup** (ARS **Route ID 51**). If the user dialed 8-1-732-555-2438, and the call could not be routed via the primary route **Main** described above, the call will be delivered to this **Backup** route. Per the configuration shown below, the call would be delivered to Line Group 4 (e.g., an additional SIP Line used for backup purposes, not shown). The configuration of the **Code**, **Telephone Number**, **Feature**, and **Line Group ID** for an ARS route is similar to the configuration already shown for Short Codes in **Section 5.6**.



If a primary route experiences a network outage such that no response is received to an outbound INVITE, Avaya IP Office routes the call via the alternate route. The user receives an audible tone when the re-routing occurs and may briefly see "Waiting for Line" on the display. The redirection will occur if a call is made right after a failure of the primary route, so Avaya IP Office had not yet marked the SIP Line out of service as a result of no response to SIP OPTIONS

JF; Reviewed: SPOC 1/14/2013 (see the note in **Section 5.3.3**). Alternatively calls can be delivered via the alternate route when the primary route was manually marked out-of-service, or known to be out-of-service due to prior failure of SIP OPTIONS.

#### 5.9. Auto Attendant (via Avaya IP Office Voicemail Pro)

In the reference configuration, Avaya IP Office Voicemail Pro (running on a Windows 2003 server), is used for Voicemail processing. The Avaya IP Office Voicemail Pro provides for basic Auto Attendant functionality.

**Note** - While Avaya IP Office Voicemail Pro provisioning and programming is beyond the scope of this document, a sample Auto Attendant basic configuration is shown below.

In the reference configuration an Auto Attendant was provisioned to prompt callers to select a numeric option (1, 2, or 3), that would forward the call to an associated Avaya IP Office extension (207, 500, and 600). This is accomplished via the following steps:

**Step 1** - Via Avaya IP Office Manager interface, follow the steps in **Section 5.5.4**, and create hunt groups for "Sales", "Service", and "Parts".

- In the **Hunt Group** tab, enter:
  - i. Name = Sales
  - ii. Extension = 1001
  - iii. **Ring Mode = Sequential**
  - iv. User List = Extension 207
  - v. Let all other values default.

🕼 Avaya IP Office R8.1 Manage	er 00E00	7058008 [	8.1(43)] [A	dministrator	Administrator	7)]			- 7 ×
File Edit View Tools Help		<b>≳^</b> (50) i n	DE007058008	<ul> <li>Hunt(</li> </ul>	iroup	<ul> <li>1001 Sales</li> </ul>	•		
IP Offices		HuntGro		X.	•	quential Group	Sales: 1001	-	A   X   √   <   >
BOOTP (1)           Operator (3)           Operator (3)           Operator (3)           System (1)	Syst	Name Parts Service Sales Local Hunt Main	Extension 1003 1002 1001 4386 200	Hunt Group Name Extension Ring Mode Hold Music Sc	Queuing Overfi	low Fallback Voiceme Sales 1001 Sequential System Source None	2		

- In the SIP tab, enter:
  - i. SIP Name = 1001
  - ii. SIP Display Name = Sales
  - iii. **Contact** = **1001**

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File Edit View Tools Help											
2 🖻 - 🖬   🖪 📰 🔔	🗸 🎿 🏞 🔞 🕴 o	0E007058008	- HuntGroup		•	1001 Sales	;	-			
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BOOTP (1)     Operator (3)     Oue007058008     System (1)	Syst Name Parts Service Sales Local Hunt Main	Extension 1003 1002 1001 4386 200	Hunt Group SIP Name SIP Display I Contact		1001		Voicemail	Voice Recording	Announceme	nts SIF	2

• Repeat the above steps to create the **Service** hunt group (group extension **1002** and associated with phone extension **500**) and the **Parts** hunt group (group extension **1003** and associated with phone extension **600**).

**Step 2** – Following the steps shown in **Section 5.6**, create a Short Code for the Auto Attendant.

- i. Code = Enter a Short Code value (e.g., \*63)
- ii. Feature = Select Voicemail Collect from the menu
- iii. **Telephone Number** = Enter "AutoAttend" (the quotes must be entered, and this text value must match the name of the Voicemail Pro Module created below).
- iv. **Line Group** = **0**

🌃 Avaya IP Office R8.1 Manage	er 00E0070	58008 [8.1(43)] [A	dministrator(Admini	strator)]	
File Edit View Tools Help					
ا 🏊 🖻 - 🗟 ا 🛋 🔜 🔝	✓ ॐ ₹	100E007058008	Short Code	*63	
IP Offices	Sh	nort Code	X	*63: Voicemail Collect	
BOOTP (1)	Code	Telephone Number	Short Code		
	<b>9X</b> *44	2	Code	*63	
	9×*45*N# 9×*46	N	Feature	Voicemail Collect	~
Control Unit (2)	9×*47 9×*48		Telephone Number	"AutoAttend"	
User (13)	<b>9×</b> *49		Line Group ID	0	*
Short Code (67)	9×*50 9×*51		Locale		*
	9×*52 9×*53*N#	N	Force Account Code		
Incoming Call Route (8) WanPort (0)	9×*57*N#	N			
Directory (0)	<b>9×</b> *63	"AutoAttend"			

**Step 3** - Following the steps shown in **Section 5.7**, create an Incoming Call Route that maps an AT&T DID to Short Code \*63 (e.g., the AT&T DID 3145555304).

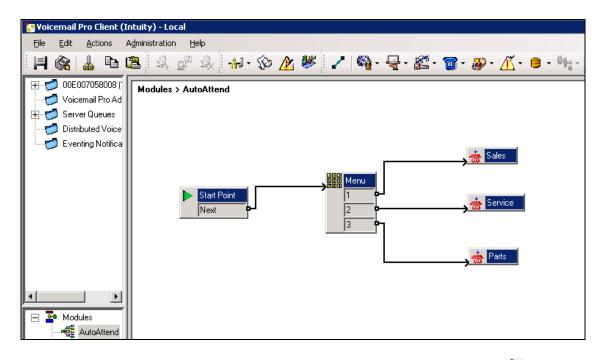
	oning Call Route	100 31455			28-11222		the second second second		
IP Offices	Inco	ming Call Ro	ute	E	100	3145	5555384		₫ • X 🗸 <
R 800TP (1)	Line Group 10			Rand	ard Voice Recording Destr	ations			
<ul> <li>Generator (3)</li> <li>Sector (3)</li> </ul>	© 100 © 100	3145555385 3145555386	*90		TmeProfile		Destination		Falback Extension
- 40 System (1)	100	3145555304	500 Extr600	•	Default Value		*63	1	8
	@100 @100	3145555303 7325554305	208 Extri208 600 Extri600	•		×		×	8
User (13)	<ul> <li>100</li> <li>100</li> <li>100</li> <li>100</li> </ul>	4306 7325554304 7325554303	700 Extr/700 500 Extr/500 208 Extr/208						

**Step 4** - Via the Voicemail Pro GUI interface:

- - i. Start Point object properties (let all other fields default)
    - 1. General tab, Token Name = Start Point
  - ii. Menu object properties (let all other fields default)
    - 1. General tab, Token Name = Menu
    - 2. Entry Prompt tab, Create an Entry Prompt that will tell the caller what digits to press to reach Sales, Service, and Parts (e.g., attendant.wav). To modify an existing recording, double click on the

.wav file and rerecord. If no .wav files exist, double click on the  $\mathbf{E}$  to open the .wav editor.

- 3. Touch Tone tab, Select 1, 2, and 3 as the possible entry digits.
- iii. Transfer Object "Sales" properties
  - 1. General tab, Token Name = Sales
  - 2. Specific tab, Destination  $\rightarrow$  Mailbox  $\rightarrow$  Sales
- iv. Transfer Object "Service" properties
  - 1. **General** tab, **Token Name = Service**
  - 2. Specific tab, Destination  $\rightarrow$  Mailbox  $\rightarrow$  Service
- v. Transfer Object "Parts" properties
  - 1. General tab, Token Name = Parts
  - 2. Specific tab, Destination  $\rightarrow$  Mailbox  $\rightarrow$  Parts
- Select the Connector icon 🖍 and:
  - i. Drag a connecting flow line from the **Start Point** box to the **Menu** box (see screen shot below).
  - ii. Drag connecting flow lines from each of the **Menu** options to their associated **Transfer** boxes (see screen shot below).



Step 5 - From the top menu select File  $\rightarrow$  Save & Make Live, or select the  $\bigotimes$  icon.

When the associated AT&T Business in a Box with IPFR-EF number is called from PSTN (e.g., 3145555304 from **Step 3**), the caller will be prompted to enter 1, 2, or 3 to access Sales, Service, or Parts. The associated Avaya IP Office extension (e.g., 207, 500, or 600) will then ring.

## 5.10. Privacy / Anonymous Calls

As described in Section 5.6, an Avaya IP Office user whose calling line identification is not typically withheld from the network, can request privacy in the reference configuration by dialing the Short Code 3N; to access the SIP Line. The Avaya 1600-Series IP Telephones can also request privacy for a specific call, without dialing a unique Short Code, using Features  $\rightarrow$  Call Settings  $\rightarrow$  Withhold Number, on the phone itself. Specific users may be configured to always withhold calling line identification by checking the Anonymous field in the SIP tab for the user (see Section 5.5.3).

#### 5.11. SIP Options Frequency

In the reference configuration, Avaya IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message. In **Section 5.4**, the SIP Line to AT&T is shown with the **Check OOS** box checked. If there is no response, Avaya IP Office can mark the trunk out of service. Although ARS (as shown in **Section 5.8**) can include alternate routes to complete calls even if the far-end is not responding, Avaya IP Office must wait for the outbound INVITE to timeout before route advance. Once the SIP OPTIONS maintenance recognizes that the SIP Line is out-of-service, new calls will no longer be delayed before route advance. Also, once the problem with the SIP Line is resolved, the SIP OPTIONS maintenance will automatically bring the link back to the in-service state. In addition, for secure networks, the periodic sending of

OPTIONS by Avaya IP Office *may* serve to keep network Firewall "pinholes" open preventing the blockage of inbound traffic to Avaya IP Office.

In the reference configuration, Avaya IP Office sourced SIP OPTIONS every 60 seconds, (the value configured in the **Binding Refresh Time** provisioned in **Section 5.3.3**).

**Note** – In the reference configuration Avaya IP Office sent OPTIONS to the AT&T Business in a Box with IPFR-EF service Border Element and AT&T responded with *405 Method Not Allowed* (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up. The AT&T Business in a Box with IPFR-EF service Border Element does *not* send OPTIONS to Avaya IP Office.

## 5.12. Saving Configuration Changes to Avaya IP Office

The provisioning changes made in Avaya IP Office Manager must be applied to the Avaya IP Office server in order for the changes to take effect. At the top of the Avaya IP Office Manager page click **File**  $\rightarrow$  **Save Configuration** (if that option is grayed out, no changes are pending).

A screen similar to the following will appear, with either **Merge** or **Immediate** automatically selected, based on the nature of the configuration changes. The **Merge** option will save the configuration change with no impact to the current system operation. The **Immediate** option will save the configuration and cause the Avaya IP Office server to reboot.

Click **OK** to execute the save.

Send Configuration	
- IP Office Settings	
Configuration Reboot Mode	
💿 Merge	
🔿 Immediate	
O When Free	
O Timed	
Reboot Time	
09:49	
L	
Call Barring	
Incoming Calls	
Outgoing Calls	
OK Cancel	Help

The active configuration may be saved to a file at any time by selecting File  $\rightarrow$  Save Configuration As.

# 6. AT&T Business in a Box with IPFR-EF Configuration

AT&T provides the Business in a Box with IPFR-EF service border element IP address, the access DID numbers, and the associated DNIS digits used in the reference configuration. In addition the AT&T EIPFR features are also assigned by AT&T. AT&T required that the Avaya IP Office LAN2 IP address be provided to the Business in a Box with IPFR-EF service, as part of the provisioning process.

# 7. Verification Steps

The following procedures may be used to verify the Avaya IP Office R8 with AT&T Business in a Box with IPFR-EF service configuration.

#### 7.1. General

The following scenarios may be executed to verify Avaya IP Office functionality with the AT&T Business in a Box with IPFR-EF service:

- Incoming calls from the PSTN routed by the AT&T Business in a Box with IPFR-EF service to the Avaya IP Office location. These incoming PSTN calls arrive via the SIP Line and may be answered by Avaya SIP telephones/Softphones, H.323 telephones, Analog telephones, Analog fax machines or via Hunt Groups. Coverage to Avaya IP Office Voicemail Pro, and Voicemail Pro auto-attendant applications, may also be used. The display of caller ID on display-equipped Avaya IP Office telephones can be verified.
- Outgoing calls from the Avaya IP Office location to the PSTN routed via the SIP Line to the AT&T Business in a Box with IPFR-EF service. These outgoing PSTN calls can be originated from Avaya SIP telephones/Softphones, H.323 telephones, or Analog endpoints. The display of caller ID on display-equipped PSTN telephones may be verified.
- Inbound / Outbound fax using T38 or G.711.
- Proper disconnect when the caller abandoned a call before answer for both inbound and outbound calls, and when the Avaya IP Office party or the PSTN party terminated an active call.
- Proper busy tone heard when an Avaya IP Office user called a busy PSTN user, or a PSTN user called a busy Avaya IP Office user (i.e., if no redirection was configured for user busy conditions).
- Various outbound PSTN call types were tested including long distance, international, and toll-free.
- Requests for privacy (i.e., caller anonymity) for Avaya IP Office outbound calls to the PSTN.
- Privacy requests for inbound calls from the PSTN to Avaya IP Office users.
- SIP OPTIONS monitoring of the health of the SIP trunk. In the reference configuration Avaya IP Office sent OPTIONS to the AT&T Business in a Box with IPFR-EF service Border Element and AT&T responded with 405 Method Not Allowed (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up. The AT&T Business in a Box with IPFR-EF service Border Element does not send OPTIONS to Avaya IP Office

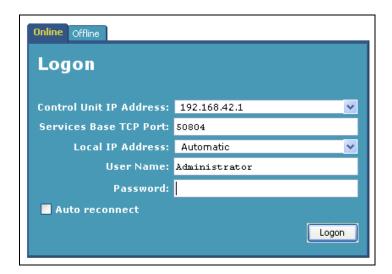
- Incoming and outgoing calls using the G.729(a) and G.711 ULAW codecs.
- DTMF transmission (RFC 2833) for successful voice mail navigation using G.729a and G.711MU for incoming and outgoing calls, including navigation of a simple autoattendant application configured on Avaya IP Office Voicemail Pro.
  - See Section 2.2, Item 1 for a DTMF issue regarding Avaya IP Office Softphone.
- Telephony features such as call waiting, hold, transfer, and conference.
- Mobile twinning to a mobile phone when the associated Avaya IP Office extension is called, as well as Mobility features such as Mobile Callback and Mobile Call Control may also be verified.
- AT&T IPFR-EF service features such as:
  - Simultaneous Ring
  - Sequential Ring
  - Call Forward Always
  - Call Forward Busy
  - Call Forward Ring No Answer

#### 7.2. Analysis/Troubleshooting

The following items may be used to analyze/troubleshoot Avaya IP Office operations.

## 7.2.1. System Status Application

The System Status application can be used to monitor or troubleshoot Avaya IP Office. The System Status application can typically be accessed from Start  $\rightarrow$  Programs  $\rightarrow$  Avaya IP Office  $\rightarrow$  System Status. The following screen shows an example Logon screen. Enter the Avaya IP Office IP address in the Control Unit IP Address field, and enter an appropriate User Name and Password. Click Logon.



After logging in, select **Trunks**  $\rightarrow$  **Line:** 17 from the left navigation menu. (SIP Line 17 is configured in Section 5.4 to communicate with the AT&T Business in a Box with IPFR-EF service). A screen such as the one shown below is displayed. In the lower left, the **Trace All** button may be pressed to display tracing information as calls are made using this SIP Line.

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-\V <i>F</i> \YF\						IP O	office	e Sys	tem Sta	tus						
Snapshot LogOff	Exit About							-								
System																
Marms (0)	Status	Utilization	Summary	Alarms												
xtensions (10) 'runks (5)							S	IP Trunk	Summary							
Lines: 1 - 4	Peer Dom	ain Name		19	2.168.64.13	30										
Line: 17	Resolved	Address:		13	35.25.29.74											
ctive Calls esources	Line Numb	ber:		17	,											
esources 'oicemail	Number o	f Administ	ered Chann	els: 90	)											
Networking	Number o	f Channel	s in Use:	0												
	Administe	red Comp	ression:	G	729 A, G711	Mu										
	Silence Su	ppression		0	ff											
	SIP Trunk	Channel	icenses:	Ur	nlimited		<b>_</b>									
			icenses in L	Jse: 0			09	0								
		Channel			PDATE (Inco	ming and	$\bigcirc$									
	SIP Trunk SIP Device	. Channel e Feature	s:	UF		-	Outgoing	)	Other Davis on	Divertion	Dauad	Deseive	Deseive	Transmit	Tupo emit	
	SIP Trunk	Channel e Feature URI Call	s: Current		PDATE (Incol Remote Media Ad	Codec	Outgoing	)	Other Party on Call	Direction of Call	Round Trip De	Receive . Jitter	Receive Packet	Transmit Jitter	Transmit Packet	
	SIP Trunk SIP Device Channel	Channel e Feature URI Call	s: Current	UF Time in	Remote	Codec	Outgoing	) Caller ID								•
	SIP Trunk SIP Device Channel Number 1 2	Channel e Feature URI Call	s: Current State Idle Idle	UF Time in State 03:12:05 03:14:28	Remote	Codec	Outgoing	) Caller ID								•
	SIP Trunk SIP Device Channel Number 1 2 3	Channel e Feature URI Call	S: Current State Idle Idle Idle	UF Time in State 03:12:05 03:14:28 03:14:28	Remote	Codec	Outgoing	) Caller ID								•
	SIP Trunk SIP Device Channel Number 1 2 3 4	Channel e Feature URI Call	s: Current State Idle Idle Idle Idle	Uf Time in State 03:12:05 03:14:28 03:14:28 03:14:28	Remote	Codec	Outgoing	) Caller ID								•
	SIP Trunk SIP Device Channel Number 1 2 3 4 5	Channel e Feature URI Call	s: Current State Idle Idle Idle Idle Idle	Uf Time in State 03:12:05 03:14:28 03:14:28 03:14:28 03:14:28	Remote	Codec	Outgoing	) Caller ID								•
	SIP Trunk SIP Device Channel Number 1 2 3 4	Channel e Feature URI Call	s: Current State Idle Idle Idle Idle	Uf Time in State 03:12:05 03:14:28 03:14:28 03:14:28	Remote	Codec	Outgoing	) Caller ID								•
	SIP Trunk SIP Device Channel Number 1 2 3 4 5 5 6	Channel e Feature URI Call	s: Current State Idle Idle Idle Idle Idle Idle	Uf Time in State 03:12:05 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28	Remote	Codec	Outgoing	) Caller ID								•
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	SIP Trunk SIP Device Channel Number 1 2 3 4 5 6 7 8 8 9 10 11	Channel e Feature URI Call	s: Current State Idle	UR Time in State 03:12:05 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28	Remote	Codec	Outgoing	) Caller ID								•
	SIP Trunk SIP Devic Channel Number 1 2 3 4 5 6 6 7 7 8 9 10	Channel e Feature URI Call	s: Current State Idle Idle Idle Idle Idle Idle Idle Idle Idle Idle Idle	Uf Time in State 03:12:05 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28 03:14:28	Remote	Codec	Outgoing	) Caller ID								•

The following screen shows an example outbound call where Avaya IP Office H.323 phone Extn500 called the PSTN.

Status Utilization Summary Ak	arms					
		SIP Trunk Summary				^
Peer Domain Name:	192.168.64.130					
Resolved Address:	135.25.29.74					
Line Number:	17					
Number of Administered Channels:	90					
Number of Channels in Use:	1					
Administered Compression:	G729 A, G711 Mu					
Silence Suppression:	Off					
SIP Trunk Channel Licenses:	Unlimited	0%				
SIP Trunk Channel Licenses in Use:	1					
SIP Device Features:	UPDATE (Incoming and Outgo	ping)				
Channel URI Call Current Time i	n Remote Co Conne.	Caller ID Other Party on	Direction Round	Receive Receive	Transmit Transmit	
Number G Ref State State	Media Ad	or Dial Call	of Call Trip D	Jitter Packet	Jitter Packet	
	26 135.25.2 G72 RTP Re.	Extn 500, Extn5	Outgoing 3ms		0ms 0%	~
2 Idle 03:19:	19					
Trace Output - All Channels: 174/12/244:08 PM-919ms Call Ket = 5,	LINDING STATE & LINDING LINDE	ELISAR LIESTIDMOD NAME ELI				
1/4/12 2:44:10 PM-620ms Line = 17, CP	2 21 11			45555386@192.16	8.64.130, To = 17325	5550
1/4/12 2:44:10 PM-621ms Call Ref = 5,				_		
1/4/12 2:44:10 PM-621ms Call Ref = 5, 1/4/12 2:44:13 PM-835ms Line = 17, CP					9 64 4 20 To - 47234	eeev.
1/4/12 2:44:13 PM-835ms Line = 17, Cr 1/4/12 2:44:13 PM-837ms Line = 17, Cr						
1/4/12 2:44:13 PM-841ms Call Ref = 5,			,	-		
1/4/12 2:44:13 PM-841ms Call Ref = 5,	Answered, Line = 17, Channel	= 1				~
<						>

#### 7.2.2. System Monitor Application

The System Monitor application can also be used to monitor or troubleshoot Avaya IP Office functionality (see reference [3]. The System Monitor application can typically be accessed from Start  $\rightarrow$  Programs  $\rightarrow$  Avaya IP Office  $\rightarrow$  Monitor.

 IP Office R8 SysMonitor - Monitoring 19
View Filters Status Help
<u>∧§⊺ ×∥ q ∀≊</u> ■

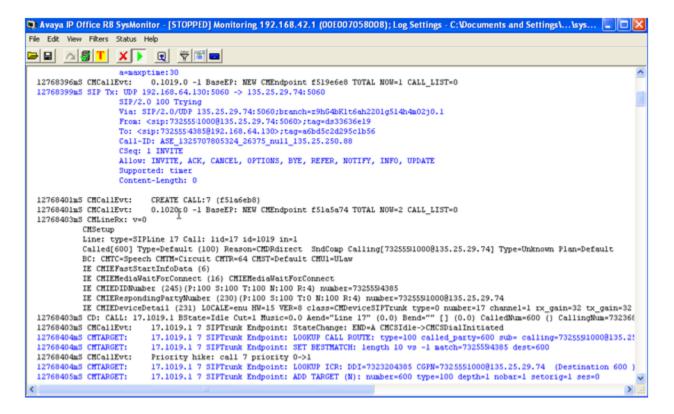
The Monitor will be active at startup. To pause the Monitor press the Pause III button.

The pause button will be replaced with the Start *b* button. Press this button to resume the

monitoring. To clear the Monitor display, press the Clear 🔀 button.

Below are samples of a monitored inbound call to Avaya IP Office SIP telephone Extn600. The Monitor will display SIP protocol (first image) as well as internal Avaya IP Office processing (second image).

📦 Avaya IP Office R8	3 SysMonitor - [STOPPED] Monitoring 192.168.42.1 (00E007058008); Log Settings - C:\Documents and Settings\\sys 📒	
File Edit View Filters	Status Help	
🛏 🖬 🔤 🔳		
	r: UDP 135,25,29,74:5060 -> 192,168,64,130:5060	~
	INVITE sip:73255543858192.168.64.130:5060 3IP/2.0	
	Via: SIP/2.0/UDP 135.25.29.74:5060;branch=z9hG4bK1t6ah2201g514h4a02j0.1	
	From: <sip:73255510008135.25.29.74:5060>;tag=ds33636e19</sip:73255510008135.25.29.74:5060>	
	To: <sip:732555143858192.168.64.130></sip:732555143858192.168.64.130>	
	Call-ID: ASE_1325707805324_26375_null_135.25.250.88	
	CSeq: 1 INVITE	
	Nax-Forwards: 66	
	Contact: <sip:73255510000135.25.29.74:5060;transport=udp></sip:73255510000135.25.29.74:5060;transport=udp>	
	Allow: INVITE,ACK,CANCEL,BYE,INFO,FRACK Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay, multipart/mixed	
	<pre>Accept: application/sdp, application/isup, application/dtmr, application/dtmr-relay, multipart/mixed P-Asserted-Identity: <sip:73255510000135.25.29.74:5060></sip:73255510000135.25.29.74:5060></pre>	
	Content Length: 262	
	Content-Disposition: session; handling=required	
	Content-Type: application/sdp	
	v=0	
	o=Sonus UAC 19987 14698 IN IP4 135.25.29.74	
	s=SIP Media Capabilities	
	c=IN IP4 135.25.29.74	
	t=0 0	
	n=audio 24634 RTP/AVP 18 0 100	
	a=rtpmap:18 G729/8000	
	a=rtpaap:0 PCMU/8000	
	a=ttpmap:100 telephone-event/8000	
	a-fmtp:100 0-15	
	a-sendzecv	
12768396mS CMCall	a=maxptime:30 1Evt: 0.1019.0 -1 BaseEP: NEW CMEndpoint f519e6e8 TOTAL NOW=1 CALL LIST=0	
	12V: 0.1019.0 -1 BBSEFF: MLW CHENGPOINT ESISEES IOIAL MOWEL CALL_LIST=0 (: UDD 19.168.64.130:5060 -> 135.25.29.74:5060	
12/0033543 311 1/	SIP/20.100 Trying	
	Via: SIP/2.0/UDP 135.25.29.74:5060;branch=z9h64bKlt6ah2201g514h4m02j0.1	-
1		- N -
<b>`</b>		<b>1</b>



The Monitor application allows the monitored information to be customized. To customize,

select the Options button  $\square$  that is third from the right in the screen above, or select **Filters**  $\rightarrow$  **Trace Options**. The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.

All Settings	×
ATM Call DTE	VPN WAN SCN EConf Frame Relay GOD H.323 Interface Media PPP R2 Routing Services SIP System
Events	
Sip High	T STUN
Packets	
SIP Reg/Opt Rx	SIP Misc Px
SIP Reg/Opt Tx	SIP Misc Tx
SIP Call Rx	Cm Notify Px
SIP Call Tx	Cm Notify Tx
ע Sip Rx ע Sip Tx	hex IP Filter (nnn.nnn.nnn)
Default All Clear All	Tab Clear All Tab Set All OK Cancel
Save File Load File	Select File

# 8. Conclusion

As illustrated in these Application Notes, Avaya IP Office Release 8.1 can be configured to interoperate successfully with AT&T Business in a Box and AT&T IP Flexible Reach - Enhanced Features, within the limitations described in **Section 2.2**. This solution provides users of Avaya IP Office R8.1 the ability to support inbound and outbound calls utilizing an AT&T Business in a Box with IPFR-EF SIP trunk service connection, via AVPN or MIS/PNT transport, using the service features listed in **Section 2**. 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.

# 9. References

#### Avaya:

In general, Avaya product documentation is available at <u>http://support.avaya.com</u>

[1] Avaya IP Office 8.1 Installation,15-601042 Issue 26i – (23 August 2012)

[2] Avaya IP Office R8.1 Manager, 10.115-601011 Issue 290 – (03 August 2012)

[3] Avaya IP Office System Monitor, Document Number 15-601019

[4] Avaya IP Office Voicemail Pro15-601063 Issue 20b - (11 July 2008)

[5] Avaya IP Office Voicemail Pro Example Exercises, Issue 4c (5th May 2004)

[6] Additional Avaya IP Office documentation can be found at: <u>http://marketingtools.avaya.com/knowledgebase/</u>

#### AT&T Business in a Box with IPFR-EF Service:

[7] http://www.business.att.com/enterprise/Service/voice-services/voip/business-in-a-box/

# 10. Appendix: SIP Line Template

Avaya IP Office Release 8.1 supports a 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.

Not all of the configuration information 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 the settings provided in this Application Notes.

To create a SIP Line Template from the configuration described in these Application Notes, configure the parameters as described below.

Create a new registry entry called **TemplateProvisioning** and set the **Value data** to **1**, as follows:

Select Start, and then Run. Type regedit as shown below

Run	? 🗙
-	Type the name of a program, folder, document, or Internet resource, and Windows will open it for you.
Open:	regedit 💌
	OK Cancel Browse

Under HKEY\_CURRENT\_USER, Software, Avaya, IP400, right click on Manager, then select New, DWORD value, then rename the newly created entry to: TemplateProvisioning. Right click on the newly created entry and select Modify, change the value under Value Data from "0" to "1".

Edit View Favorites Help					Ī
🚚 My Computer	~	Name	Туре	Data	
E HKEY_CLASSES_ROOT		RENAVTOOLBARX	REG_DWORD	0x0000010a (266)	
		B NAVTOOLBARY	REG DWORD	0x00000000 (0)	
AppEvents		Rei NonThreadedTCP	REG_DWORD	0×00000000 (0)	
🗈 🧰 Avaya		BeasswordRequiredForSave	REG_DWORD	0x00000001 (1)	
		B Prompt Validation	REG_DWORD	0×00000000 (0)	
Control Panel     CounterPath		B SCNBACKGROUNDIMAGEHIDDEN	REG_DWORD	0x00000001 (1)	
			REG_SZ		
		SCNDISCOVERY	REG DWORD	0×00000000 (0)	
Keyboard Layout		B SecureCommunications	REG DWORD	0x00000001 (1)	
Network		B SecurityLevel	REG DWORD	0x00000001(1)	
Printers		BervicesBaseHTTPPort	_ REG_DWORD	0x00000050 (80)	
🖬 🧰 Remote		ServicesBasePort	REG DWORD	0x0000c674 (50804)	
SessionInformation		BetRingDelayPerAp	REG DWORD	0x00000000 (0)	
🗐 🛄 Software		B SHOWADMINTASKMAINToolbar	REG_DWORD	0×00000000 (0)	
🕀 🧰 Adobe		B SHOWErrorPane	REG_DWORD	0×00000000 (0)	
🕒 🧰 Alps		B SHOWInGroups	REG_DWORD	0×00000000 (0)	
😑 🧰 Avaya	_	B SHOWMAINToolbar	REG_DWORD	0x00000001 (1)	
🗈 🦲 2050 IP Softphone			REG_DWORD	0x00000001 (1)	
🗈 🦲 Avaya IP Softphone			REG_DWORD	0x00000001 (1)	
🗈 🦲 Avaya one-X Agent			REG_DWORD	0x00000000 (0)	
Avaya one-X AgentAVC			REG_DWORD	0x00000000 (0)	
- 🛄 Avaya one-X AgentAVCClient 🕢 🧰 🧰 Avaya one-X Communicator			REG_DWORD	0x00000001 (1)	
Avaya one-X® Communicator		B SHOWSIMPLIFIEDVIEWASDEFAULTVIEW	REG_DWORD	0x00000000 (0)	
Avaya SIP Softphone		BSSLRemoteAccess	REG_DWORD	0x00000000 (0)	
Avaya Site Administration		B STARTINITIALDISCOVERY	REG_DWORD	0x00000001 (1)	
		TCPDiscoveryEnabled	REG_DWORD	0x00000001 (1)	
Integrator for Outlook		TCPSearchCriteria			
IP Office Softphone		TemplateOptions	REG DWORD	0×00000001 (1)	
😑 🧰 IP400		ReplateProvisioning	REG_DWORD	0×00000001 (1)	
😑 🔄 Manager		E TIMESERVEREnabled	REG_DWORD	0×00000001 (1)	
😟 🧰 Column Headings		UDPDiscoveryEnabled	REG_DWORD	0×00000001 (1)	
			REG_SZ	C:\Program Files\Ava	ıγ
PhoneManager			REG_DWORD	0×00000001 (1)	
UpgradeWizard		WalidateConfigOnOK	REG_DWORD	0×00000001 (1)	
🗈 🦲 Softphone					
Voicemail Pro	×	<			>

Edit DWORD Value	? 🗙
Value name: TemplateProvisioning	
Value data:	Base
1	<ul> <li>Hexadecimal</li> <li>Decimal</li> </ul>
	OK Cancel

#### **Reboot the computer.**

When the computer comes back up, enable the template by opening **IP Office Manager**, select **File**, and then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box, and click **OK**.

📶 IP Offic	ce Manager I	Preferenc	:es		? 🗙
Preferences	Directories	Discovery	Visual Preferences	Security	Validation
Icon Size	Small Multiline Ta	*		Securicy	Validation
<b>i</b>			ок Са	ancel	Help

To create a SIP Line Template from the configuration, on the left Navigation Pane, right click on the Sip Line (17), and select Generate SIP Trunk Template (not shown)

Enter a descriptive name; **ATT** was used in the sample template. Note that for ITSP Domain Name **Not Used** was used (AT&T uses IP addresses instead of Domain names), an entry is required here or the template will not run. This entry (**Not Used**) should be removed after importing the configuration into a new Avaya IP Office installation.

🐮 SIP Trunk Template - (Sil	P Trunk - 17)				×
Please review and change	the trunk settings if you want –				
SIP Line Transport VolP T3	88 Fax SIP Credentials				
Descriptive Name	ATT	Use Tel URI			
ITSP Domain Name	Not Used	Check OOS			
Send Caller ID	Diversion Header 🛛 🗸	Call Routing Method	Request URI 🗸 🗸		
Association Method	By Source IP address 💌	Originator number for forwarded and twinning calls		]	
		Name Priority	Favor Trunk 💌		
Incoming	Auto 🗸				
Outgoing	Auto 💌				
UPDATE Supported	Never	Caller ID from From header			
User-Agent and Server Headers		Send From In Clear			
				Export	Cancel

To generate the template click on **Export**.

On the next screen, **Template Type Selection**, select the **Country**, enter the name for the **Service Provider**, and click **Generate Template**.

🖬 Template Type Selection				
Locale	United States (US English)	·		
Country	United States	•		
Service Provider	AT&T	•		
	Generate Template	Cancel		

By default the template file is generated to the path \\**Program Files**\**Avaya**\**IP Office**\**Manager**\**Templates**.

The following is an example of the exported SIP Line Template file, US\_AT&T\_SIPTrunk.xml:

```
<?xml version="1.0" encoding="utf-8" ?>
- <Template xmlns="urn:SIPTrunk-schema">
 <TemplateType>SIPTrunk</TemplateType>
 <Version>20121130</Version>
 <SystemLocale>enu</SystemLocale>
 <DescriptiveName>ATT</DescriptiveName>
 <ITSPDomainName>Not Used</ITSPDomainName>
 <SendCallerID>CallerIDDIV</SendCallerID>
 <ReferSupport>true</ReferSupport>
 <ReferSupportIncoming>2</ReferSupportIncoming>
 <ReferSupportOutgoing>2</ReferSupportOutgoing>
 <RegistrationRequired>false</RegistrationRequired>
 <UseTelURI>false</UseTelURI>
 <CheckOOS>true</CheckOOS>
 <CallRoutingMethod>1</CallRoutingMethod>
 <OriginatorNumber />
 <AssociationMethod>SourceIP</AssociationMethod>
 <LineNamePriority>FavourTrunk</LineNamePriority>
 <UpdateSupport>UpdateNever</UpdateSupport>
 <UserAgentServerHeader />
 <CallerIDfromFromheader>false</CallerIDfromFromheader>
 <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
 <ITSPProxy>12.40.234.99</ITSPProxy>
 <LayerFourProtocol>SipUDP</LayerFourProtocol>
 <SendPort>5060</SendPort>
 <ListenPort>5060</ListenPort>
 <DNSServerOne>0.0.0.0</DNSServerOne>
```

<DNSServerTwo>0.0.0.0</DNSServerTwo>

<CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>

<SeparateRegistrar />

<CompressionMode>AUTOSELECT</CompressionMode>

<UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>

<AdvCodecPref>G.711 ULAW 64K,G.729(a) 8K CS-ACELP</AdvCodecPref>

<CallInitiationTimeout>4</CallInitiationTimeout>

<DTMFSupport>DTMF\_SUPPORT\_RFC2833</DTMFSupport>

<VoipSilenceSupression>false</VoipSilenceSupression>

<ReinviteSupported>true</ReinviteSupported>

<FaxTransportSupport>FOIP\_T38</FaxTransportSupport>

<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>

<CodecLockdown>false</CodecLockdown>

<Rel100Supported>true</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>false</UseDefaultValues>

<ScanLineFixup>true</ScanLineFixup>

<TFOPEnhancement>true</TFOPEnhancement>

<DisableT30ECM>false</DisableT30ECM>

<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>

<DisableT30MRCompression>false</DisableT30MRCompression>

<NSFOverride>false</NSFOverride>

</Template>

Next, import the template into the new Avaya IP Office system by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New, New SIP Trunk from Template**:

Lin Line Type 1월 1 - Analogue Tru	Line Sub	SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials					
行2 Analogue Tru	unk	Line Number 17					
f73 Analogue Tru f74 Analogue Tru		ITSP Domain Name			In Service	✓	
🔪 17 SIP Line 👝							
1	🔄 New	N	•	н	323 Line		
	Generate	SIP Trunk Template		IF	PDECT Line	1	
3	🔏 Cut		Ctrl+X	S	IP Line	e	
				N	New SIP Trunk From Template		
	🛅 Сору		Ctrl+C	_		-	
	🖺 Paste		Ctrl+V		Name Priority	Fai	
2	🔨 Delete		Ctrl+Del		Caller ID from From header		
	🧪 Validate				Send From In Clear		
	Connect T	·o	Ctrl+T		User-Agent and Server Headers		
	New from	Template (Binary)		~			
	Export as	Template (Binary)		~			
	Show In G	iroups					
	Customize	Columns			~		

On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.

📶 Template Ty		
Locale	United States (US English)	×
Country	United States	~
Service Provider	AT&T	~
	Generate Templa	te Cancel

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